

Configuration Note

AudioCodes Professional Services – Interoperability Lab

Mediant™ E-SBC for BroadCloud SIP Trunk with Microsoft® Skype for Business Server 2015

Version 7.2



Table of Contents

1	Introduction	7
1.1	Intended Audience	7
1.2	About AudioCodes E-SBC Product Series.....	7
2	Component Information.....	9
2.1	AudioCodes E-SBC Version	9
2.2	BroadCloud SIP Trunking Version.....	9
2.3	Microsoft Skype for Business Server 2015 Version	9
2.4	Interoperability Test Topology	10
2.4.1	Environment Setup	11
2.4.2	Known Limitations.....	11
3	Configuring Skype for Business Server 2015.....	13
3.1	Configuring the E-SBC as an IP / PSTN Gateway	13
3.2	Configuring the "Route" on Skype for Business Server 2015.....	21
4	Configuring AudioCodes E-SBC.....	31
4.1	Step 1: IP Network Interfaces Configuration	32
4.1.1	Step 1a: Configure VLANs.....	33
4.1.2	Step 1b: Configure Network Interfaces.....	33
4.2	Step 2: Enable the SBC Application	35
4.3	Step 3: Configure Media Realms	36
4.4	Step 4: Configure SIP Signaling Interfaces.....	39
4.5	Step 5: Configure Proxy Sets	41
4.6	Step 6: Configure Coders	45
4.7	Step 7: Configure IP Profiles	49
4.8	Step 8: Configure IP Groups.....	53
4.9	Step 9: SIP TLS Connection Configuration	55
4.9.1	Step 9a: Configure the NTP Server Address.....	55
4.9.2	Step 9b: Configure the TLS version	56
4.9.3	Step 9c: Configure a Certificate for Operation with Microsoft Skype for Business Server 2015	57
4.9.4	Step 9d: Configure a Certificate for Operation with the BroadCloud SIP Trunk.....	63
4.10	Step 10: Configure SRTP	64
4.11	Step 11: Configure Maximum IP Media Channels	65
4.12	Step 12: Configure IP-to-IP Call Routing Rules	66
4.13	Step 13: Configure IP-to-IP Manipulation Rules.....	71
4.14	Step 14: Configure Message Manipulation Rules	73
4.15	Step 15: Configure Registration Accounts	82
4.16	Step 16: Miscellaneous Configuration.....	84
4.16.1	Step 16a: Configure Call Forking Mode	84
4.16.2	Step 16b: Configure SBC Alternative Routing Reasons	85
4.17	Step 17: Reset the E-SBC	86
A	AudioCodes INI File	87

This page is intentionally left blank.

Notice

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: May-14-2017

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
12381	Initial document release for Version 7.2.

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>.

This page is intentionally left blank.

1 Introduction

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

You can also use AudioCodes' SBC Wizard tool to automatically configure the E-SBC based on this interoperability setup. However, it is recommended to read through this document in order to better understand the various configuration options. For more information on AudioCodes' SBC Wizard including download option, visit AudioCodes Web site at <http://www.audicodes.com/sbc-wizard> (login required).

1.1 Intended Audience

The document is intended for engineers, or AudioCodes and BroadCloud Partners who are responsible for installing and configuring BroadCloud'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	<ul style="list-style-type: none"> ▪ Mediant 500 E-SBC ▪ Mediant 500L Gateway & E-SBC ▪ Mediant 800B Gateway & E-SBC ▪ Mediant 1000B Gateway & E-SBC ▪ Mediant 2600 E-SBC ▪ Mediant 4000 SBC ▪ Mediant 4000B SBC ▪ Mediant 9000 SBC ▪ Mediant Software SBC (SE and VE)
Software Version	SIP_7.20A.002
Protocol	<ul style="list-style-type: none"> ▪ SIP/UDP or SIP/TLS (to the BroadCloud SIP Trunk) ▪ SIP/TCP or SIP/TLS (to the S4B FE Server)
Additional Notes	None

2.2 BroadCloud SIP Trunking Version

Table 2-2: BroadCloud Version

Vendor/Service Provider	BroadCloud
SSW Model/Service	
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 6.0.9319.259
Protocol	SIP
Additional Notes	None

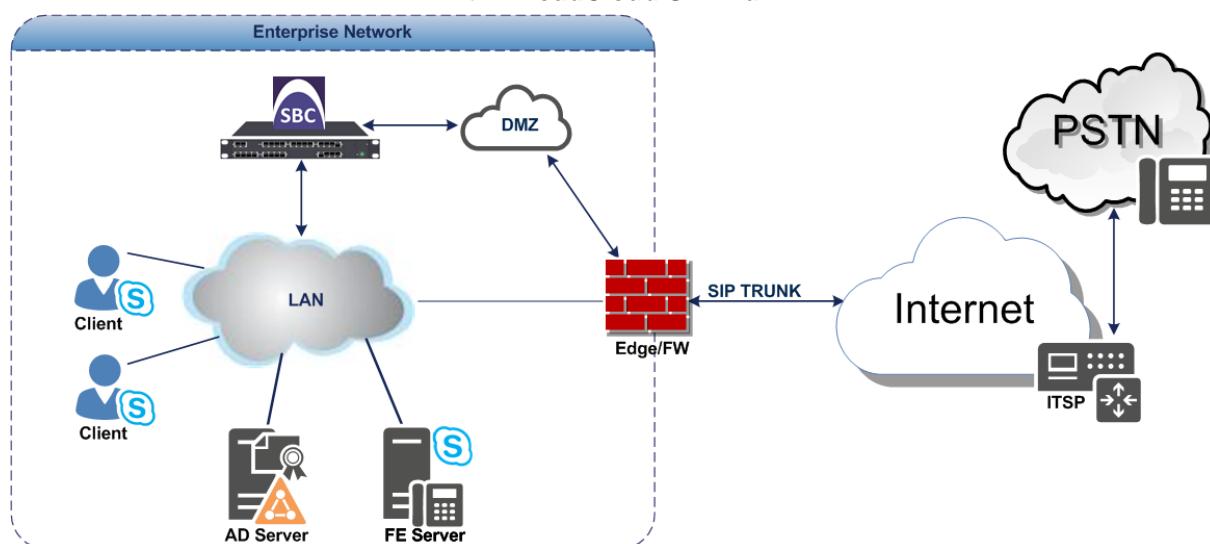
2.4 Interoperability Test Topology

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

The figure below illustrates this interoperability test topology:

Figure 2-1: Interoperability Test Topology between E-SBC and Microsoft Skype for Business with BroadCloud SIP Trunk



2.4.1 Environment Setup

The interoperability test topology includes the following environment setup:

Table 2-4: Environment Setup

Area	Setup
Network	<ul style="list-style-type: none"> ▪ Microsoft Skype for Business Server 2015 environment is located on the Enterprise's LAN ▪ BroadCloud SIP Trunk is located on the WAN
Signaling Transcoding	<ul style="list-style-type: none"> ▪ Both, Microsoft Skype for Business Server 2015 and BroadCloud SIP Trunk, operates with SIP-over-TLS transport type
Codecs Transcoding	<ul style="list-style-type: none"> ▪ Microsoft Skype for Business Server 2015 supports G.711A-law and G.711U-law coders ▪ BroadCloud SIP Trunk supports G.729, G.711A-law and G.711U-law coders
Media Transcoding	<ul style="list-style-type: none"> ▪ Both, Microsoft Skype for Business Server 2015 and BroadCloud SIP Trunk, operates with SRTP media type

2.4.2 Known Limitations

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

- If the Microsoft Skype for Business Server 2015 sends the '503 Service Unavailable' error response, the BroadCloud 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 '480 Temporarily Unavailable' response (see Section 4.14).

This page is intentionally left blank.

3 Configuring Skype for Business Server 2015

This chapter describes 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; however, they are beyond the scope of this document.

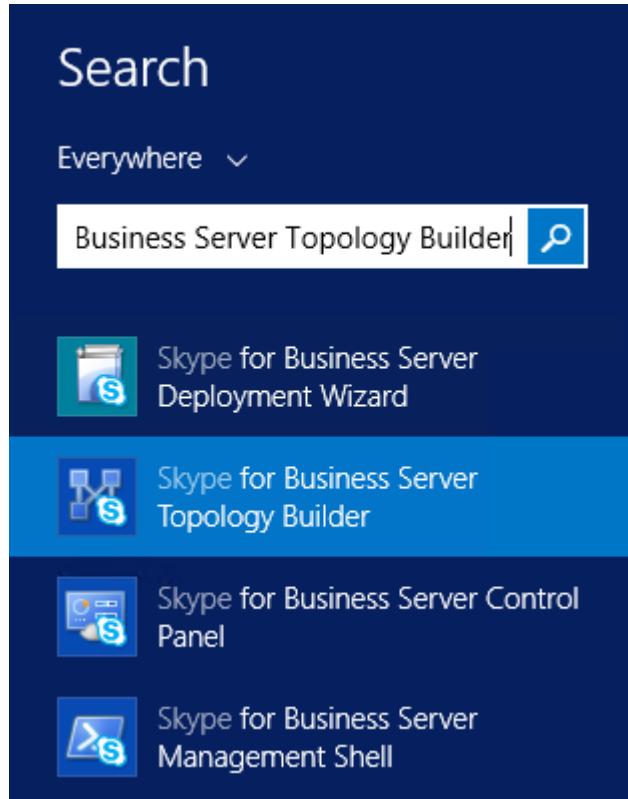
3.1 Configuring the E-SBC as an IP / PSTN Gateway

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

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

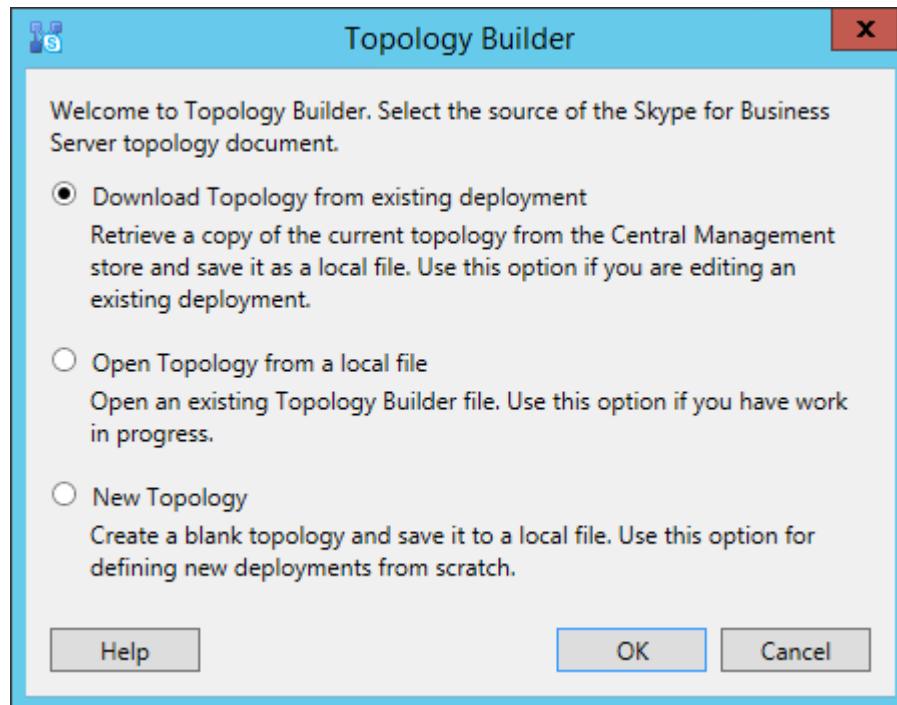
1. On the server where the Topology Builder is installed, start the Skype for Business Server 2015 Topology Builder (Windows Start menu > search for **Skype for Business Server Topology Builder**), as shown below:

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



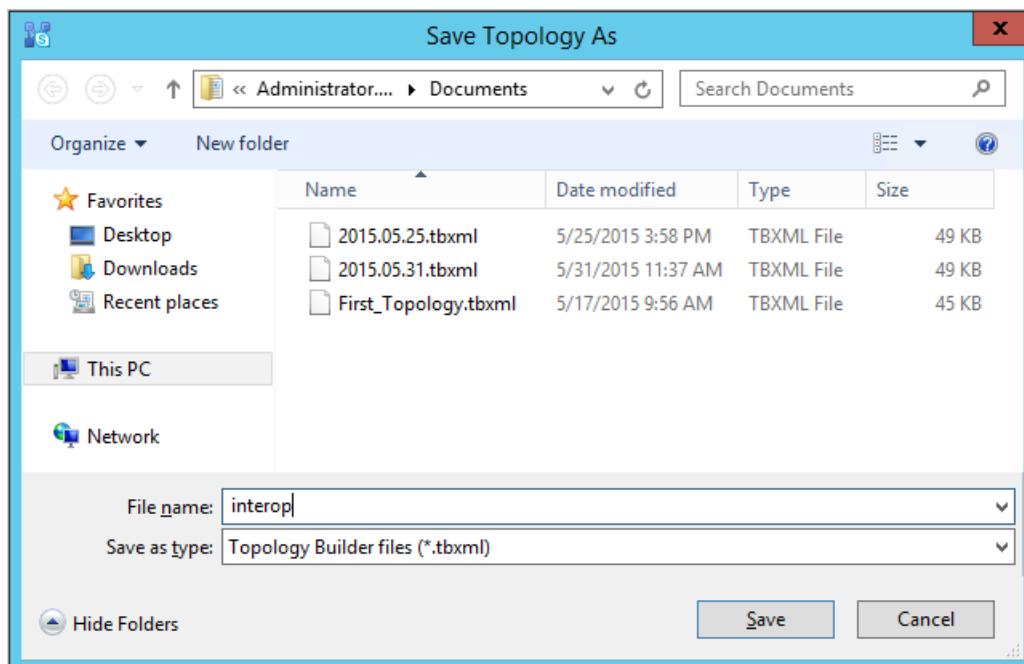
The following is displayed:

Figure 3-2: Topology Builder Dialog Box



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

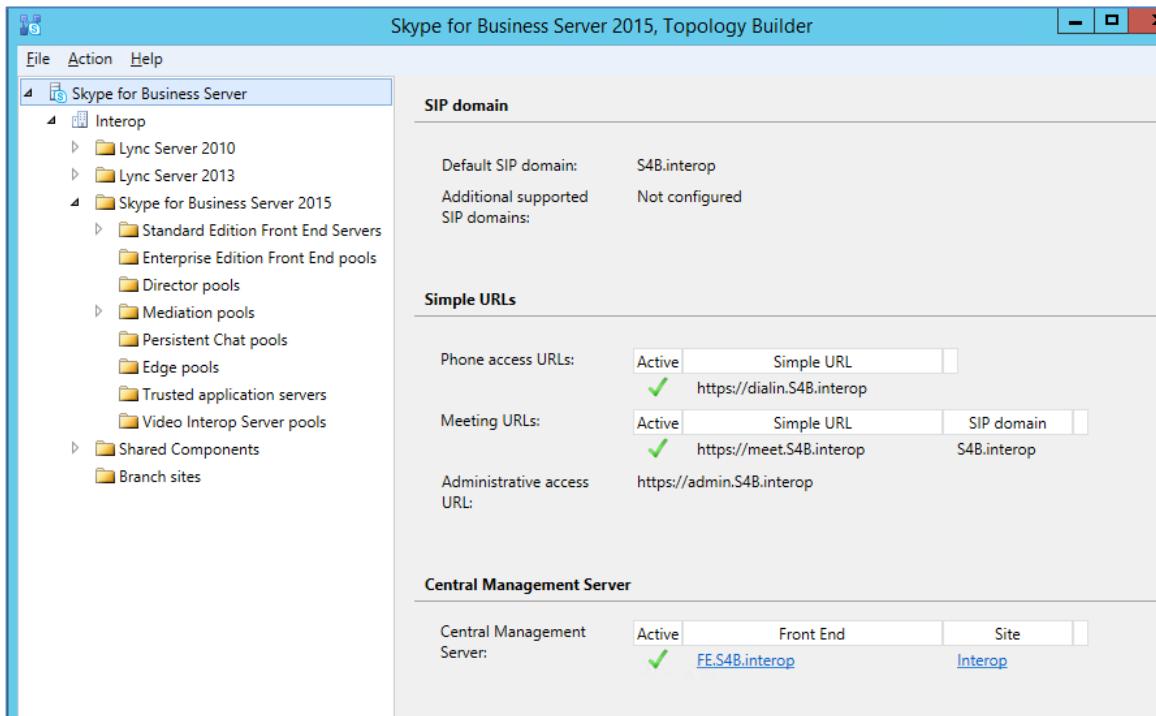
Figure 3-3: Save Topology Dialog Box



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

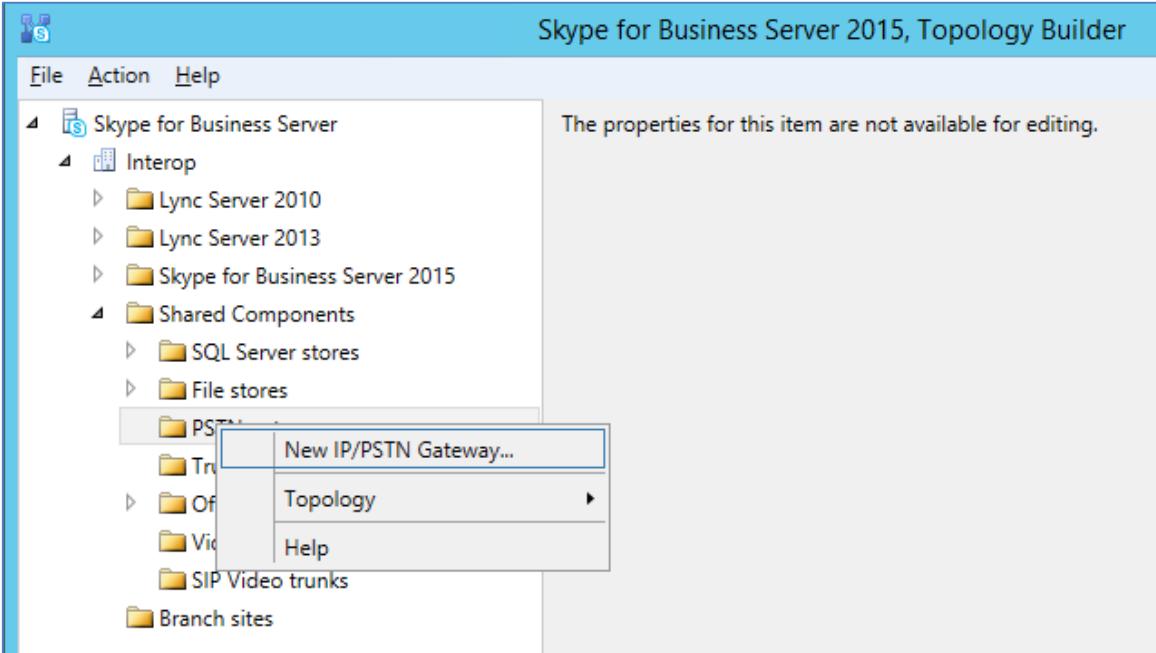
The Topology Builder screen with the downloaded Topology is displayed:

Figure 3-4: Downloaded Topology



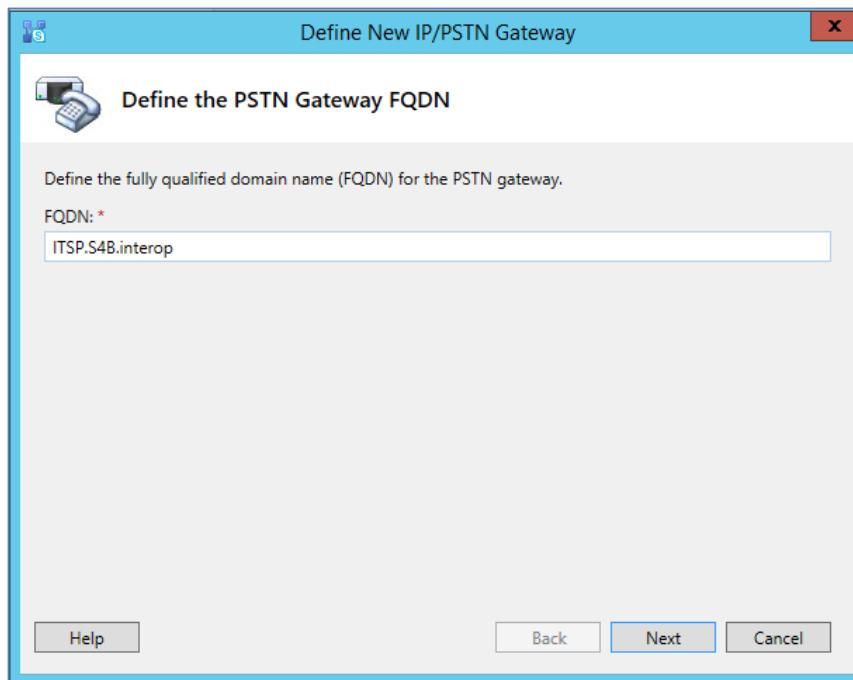
- Under the **Shared Components** node, right-click the **PSTN gateways** node, and then from the shortcut menu, choose **New IP/PSTN Gateway**, as shown below:

Figure 3-5: Choosing New IP/PSTN Gateway



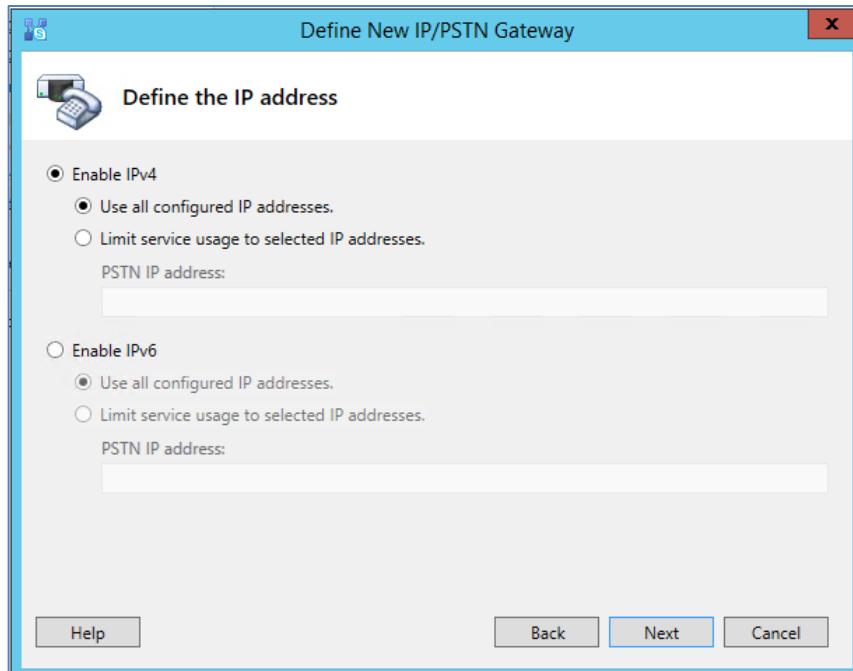
The following is displayed:

Figure 3-6: Define the PSTN Gateway FQDN



5. Enter the Fully Qualified Domain Name (FQDN) of the E-SBC (e.g., **ITSP.S4B.interop**). This FQDN should be equivalent to the configured Subject Name (CN) in the TLS Certificate Context (see Section 4.9.3 on page 57).
6. Click **Next**; the following is displayed:

Figure 3-7: Define the IP Address

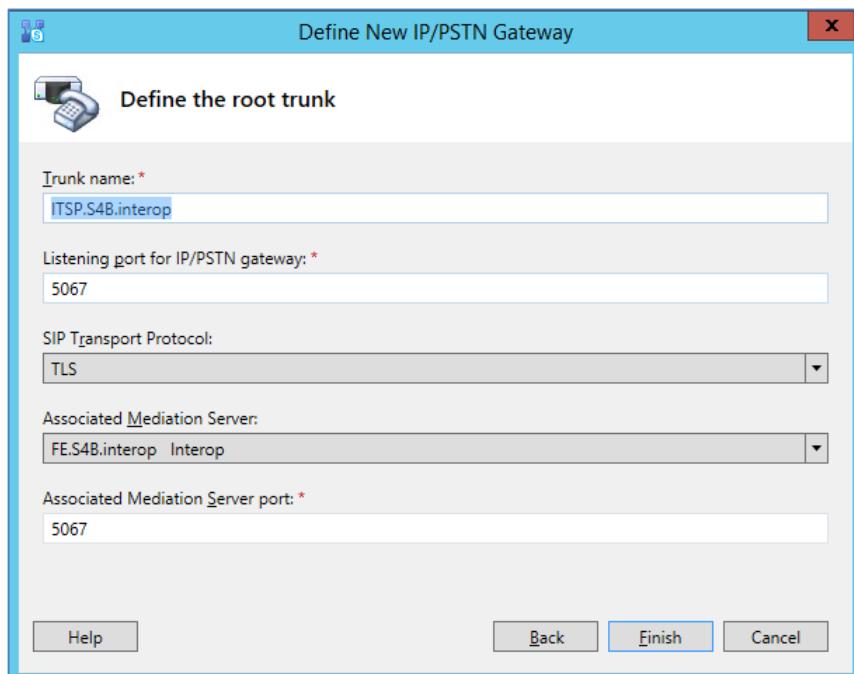


7. Define the listening mode (IPv4 or IPv6) of the IP address of your new PSTN gateway, and then click **Next**.

8. 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:**

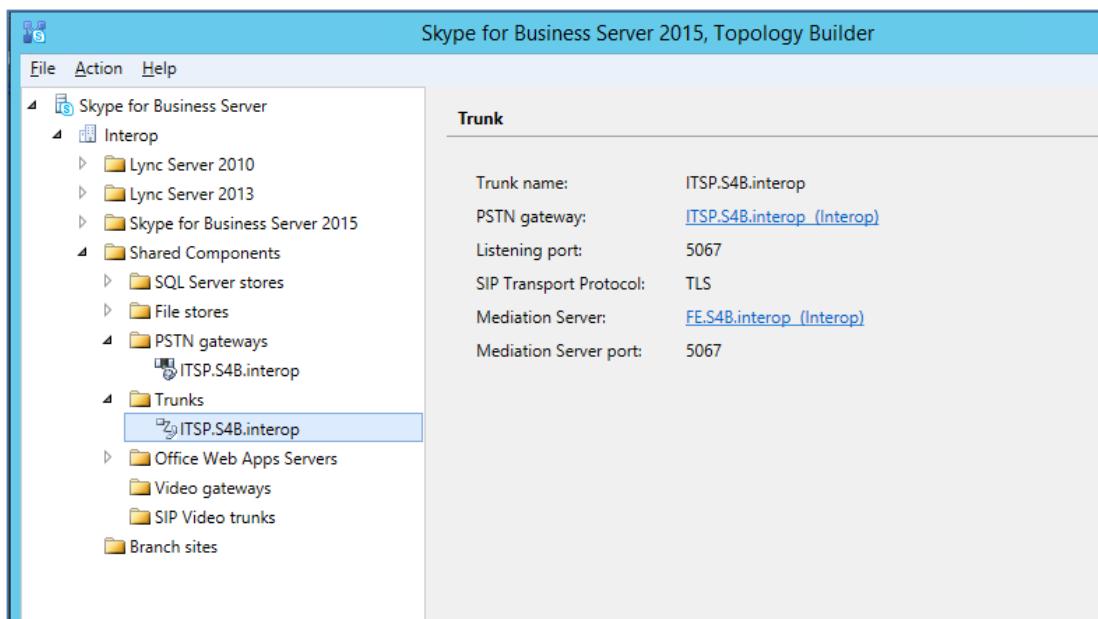
- When defining a PSTN gateway in Topology Builder, you must define a root trunk to successfully add the PSTN gateway to your topology.
- The root trunk cannot be removed until the associated PSTN gateway is removed.

Figure 3-8: Define the Root Trunk

- 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**). This parameter is later configured in the SIP Interface table (see Section 4.3 on page 36).
- In the 'SIP Transport Protocol' field, select the transport type (e.g., **TLS**) that the trunk uses. This parameter is later configured in the SIP Interface table (see Section 4.3 on page 36).
- In the 'Associated Mediation Server' field, select the Mediation Server pool to associate with the root trunk of this PSTN gateway.
- 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**).
- Click **Finish**.

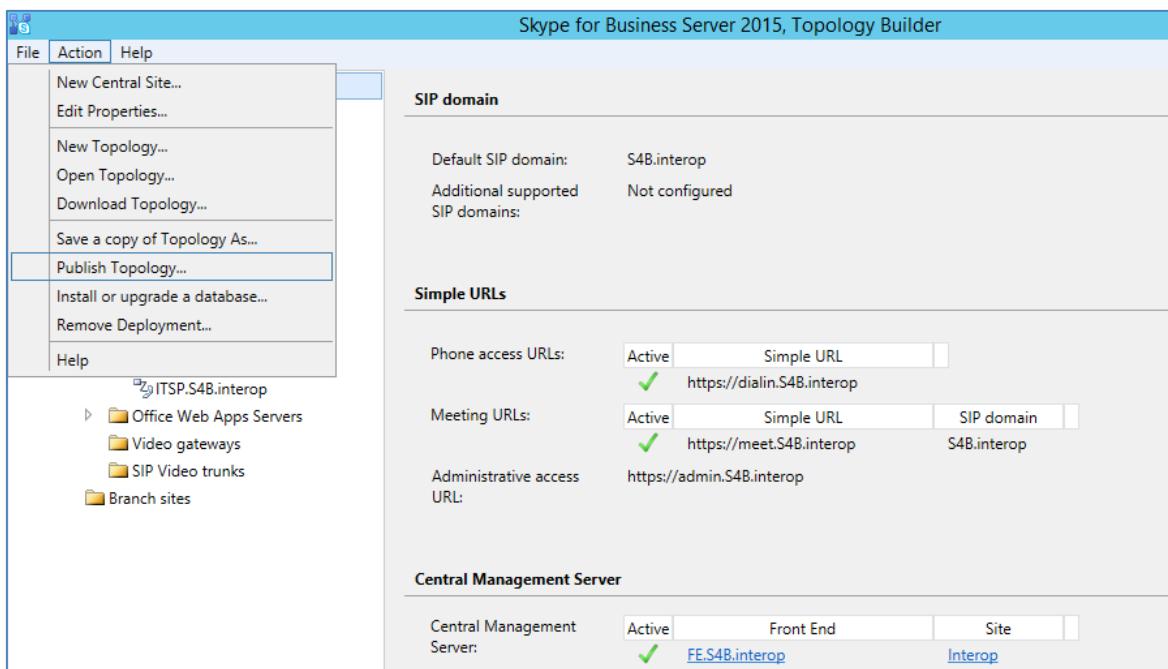
The E-SBC is added as a PSTN gateway, and a trunk is created as shown below:

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



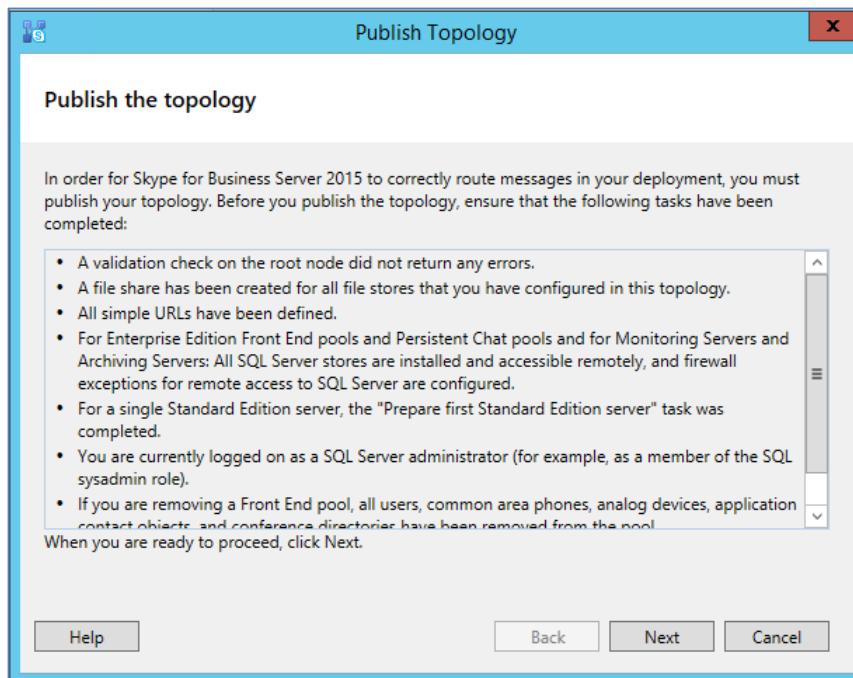
9. Publish the Topology: In the main tree, select the root node **Skype for Business Server**, and then from the **Action** menu, choose **Publish Topology**, as shown below:

Figure 3-10: Choosing Publish Topology



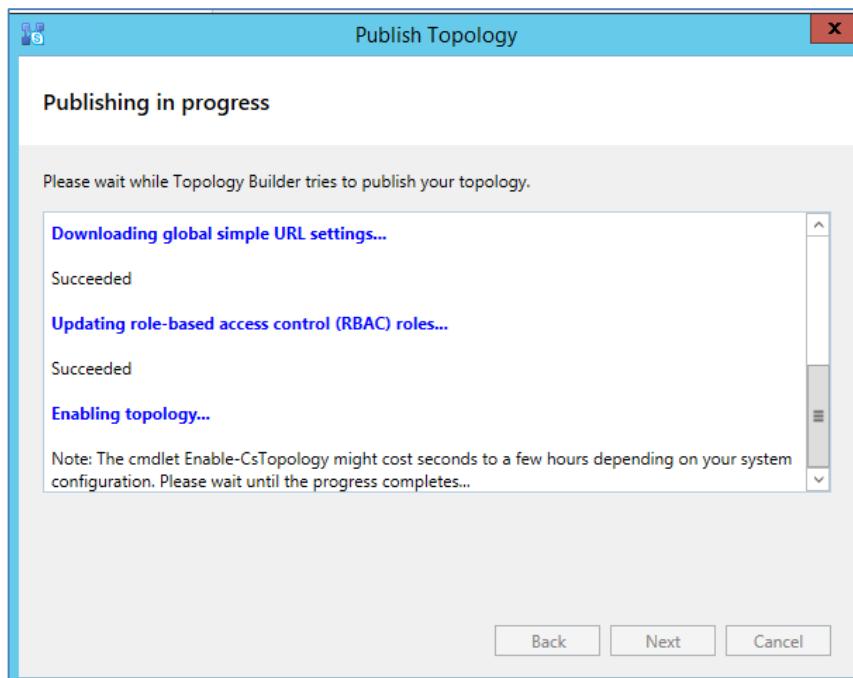
The following is displayed:

Figure 3-11: Publish the Topology



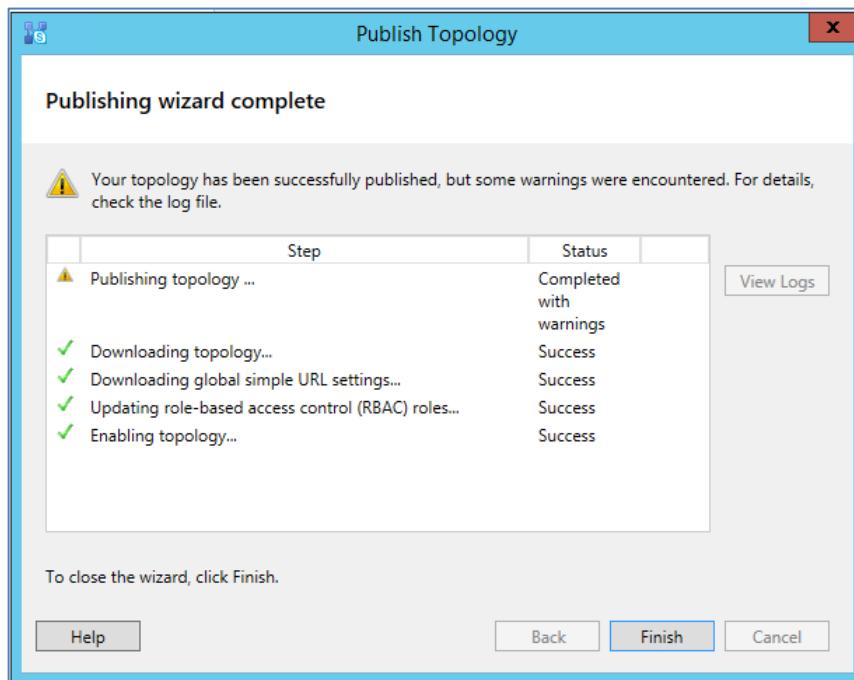
10. Click **Next**; the Topology Builder starts to publish your topology, as shown below:

Figure 3-12: Publishing in Progress



- 11.** Wait until the publishing topology process completes successfully, as shown below:

Figure 3-13: Publishing Wizard Complete



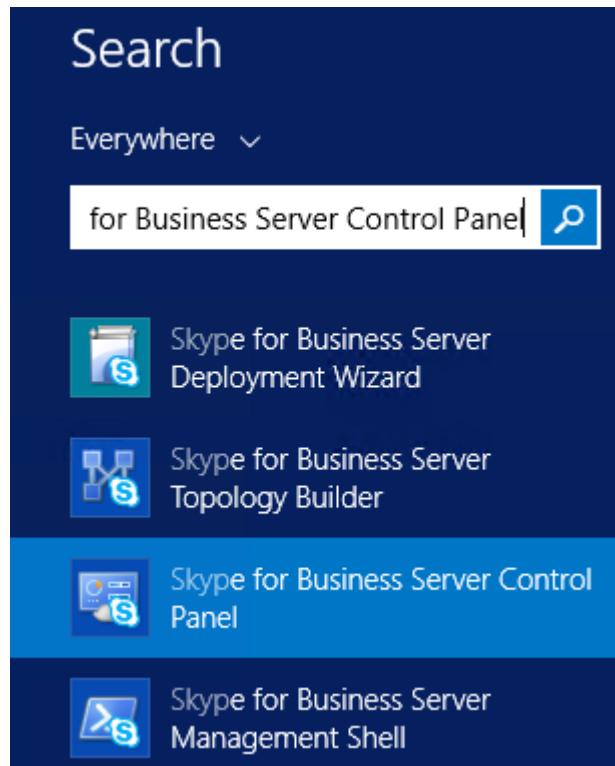
- 12.** Click **Finish**.

3.2 Configuring the "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.

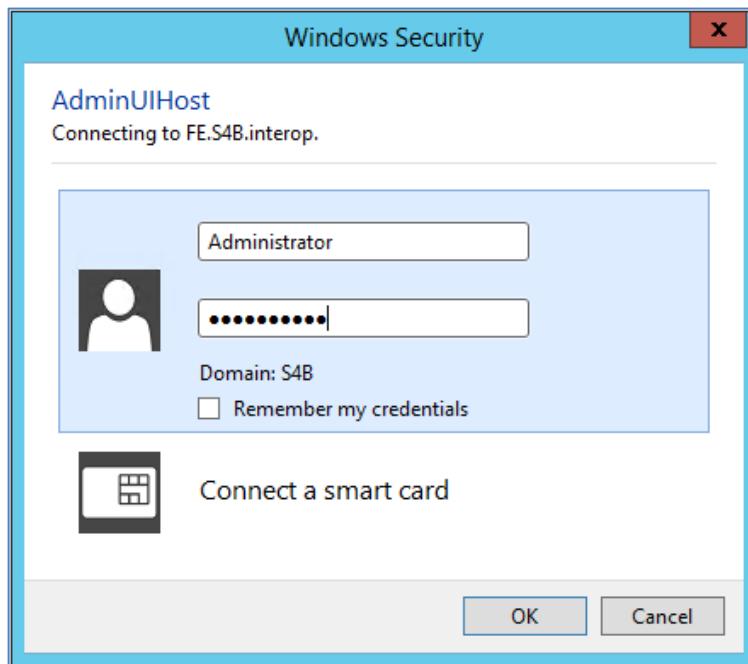
- **To configure the "route" on Skype for Business Server 2015:**
1. Start the Microsoft Skype for Business Server 2015 Control Panel (**Start** > search for **Microsoft Skype for Business Server Control Panel**), as shown below:

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



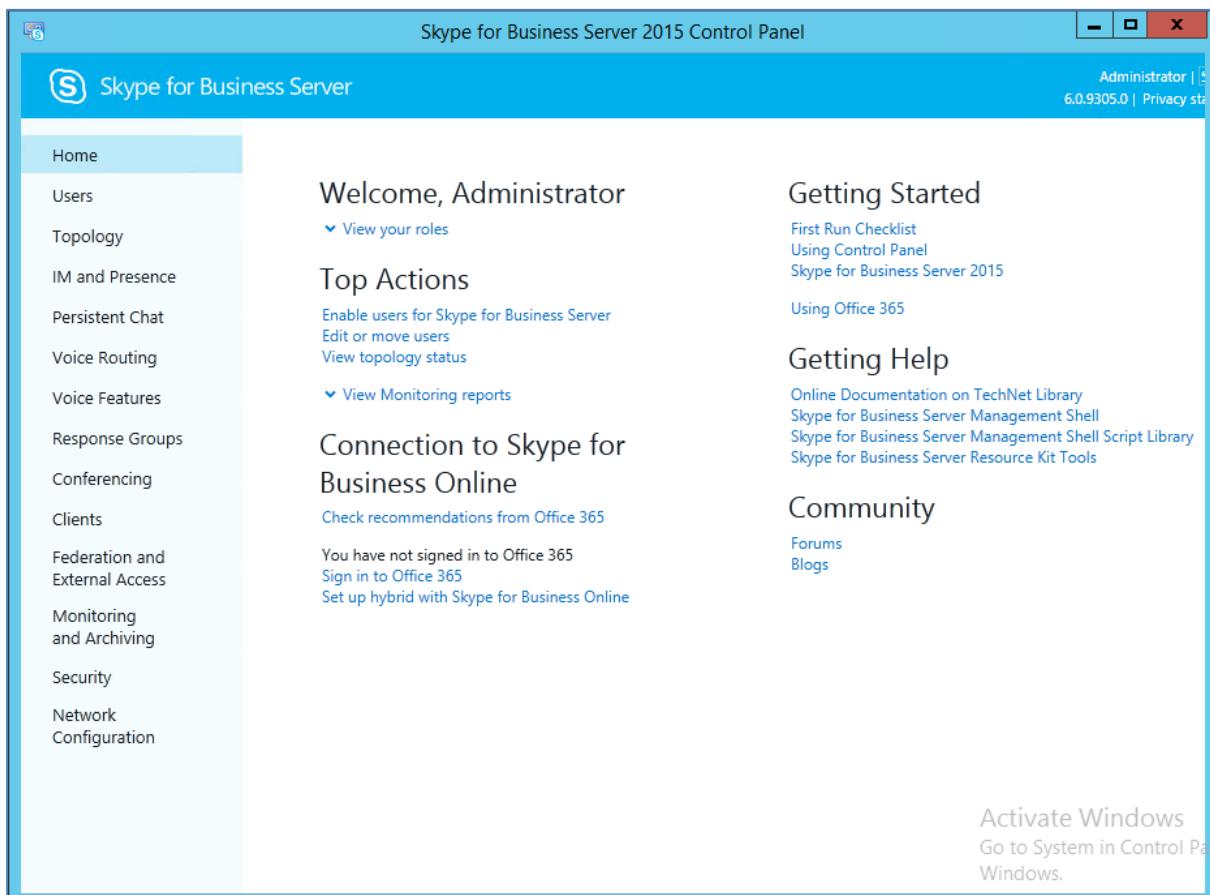
2. You are prompted to enter your login credentials:

Figure 3-15: Skype for Business Server Credentials



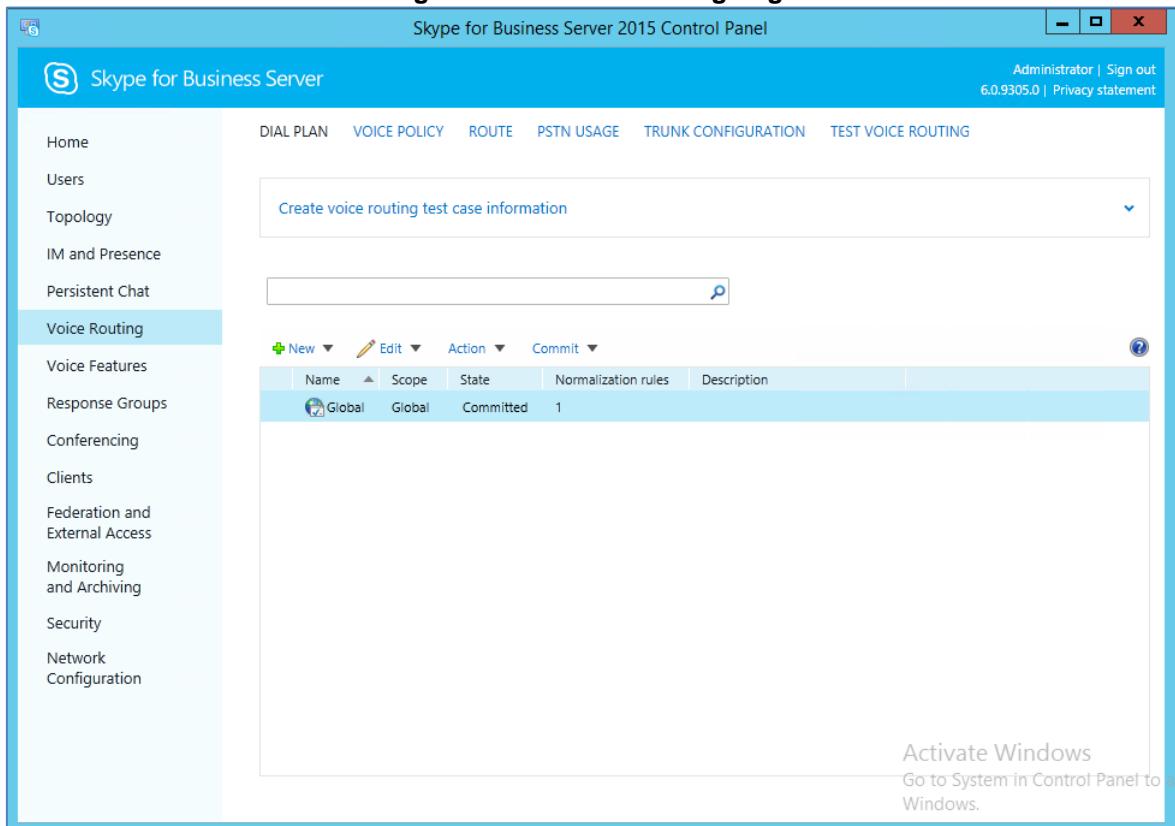
3. Enter your domain username and password, and then click **OK**; the Microsoft Skype for Business Server 2015 Control Panel is displayed:

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



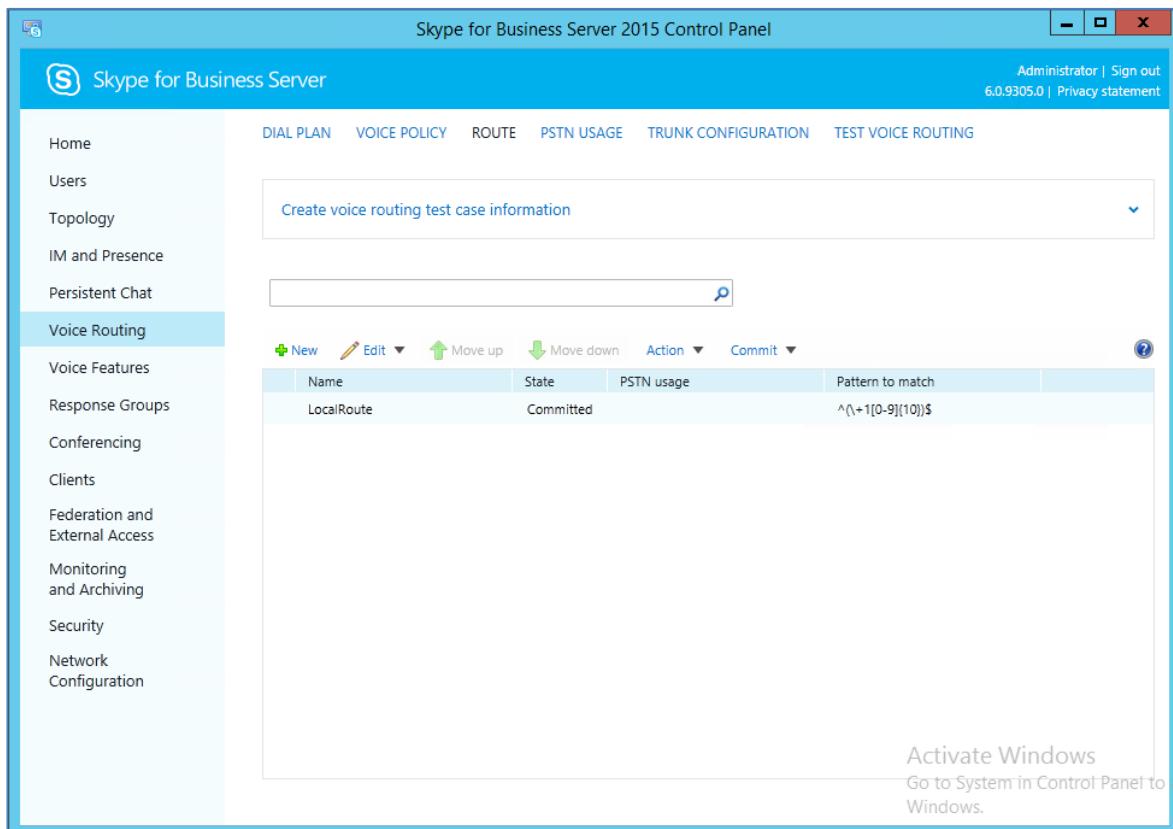
4. In the left navigation pane, select **Voice Routing**.

Figure 3-17: Voice Routing Page



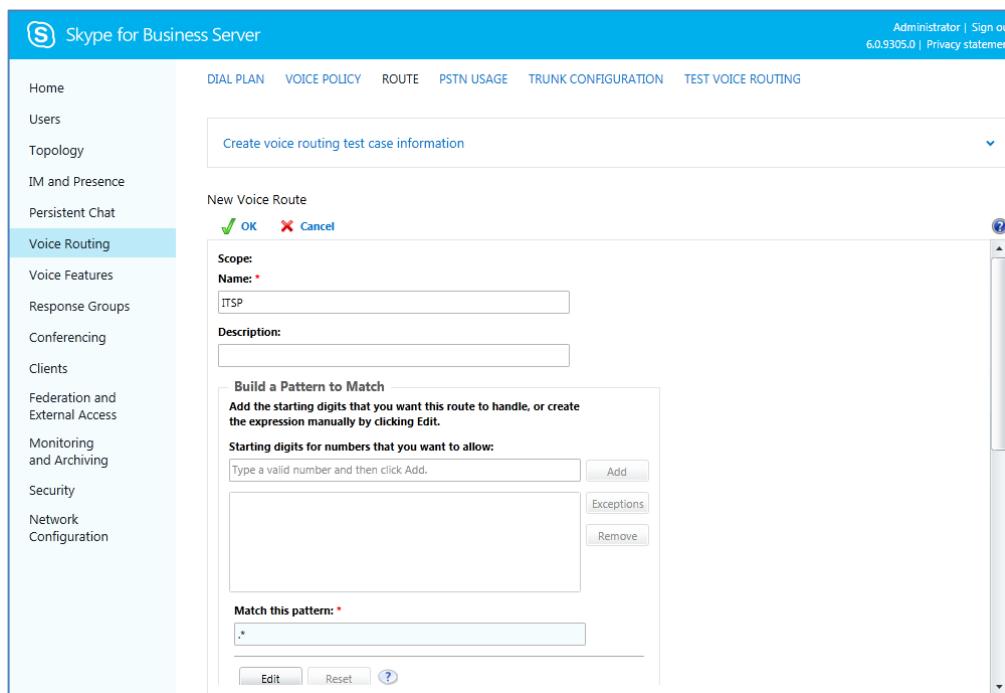
5. In the Voice Routing page, select the **ROUTE** tab.

Figure 3-18: Route Tab



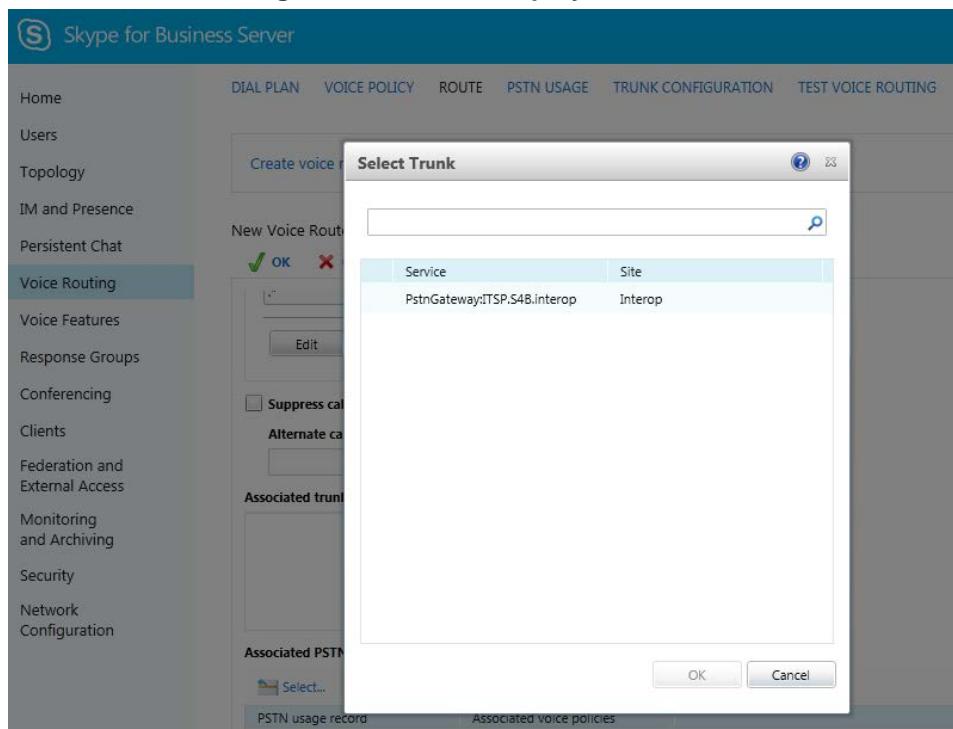
6. Click **New**; the New Voice Route page appears:

Figure 3-19: Adding New Voice Route



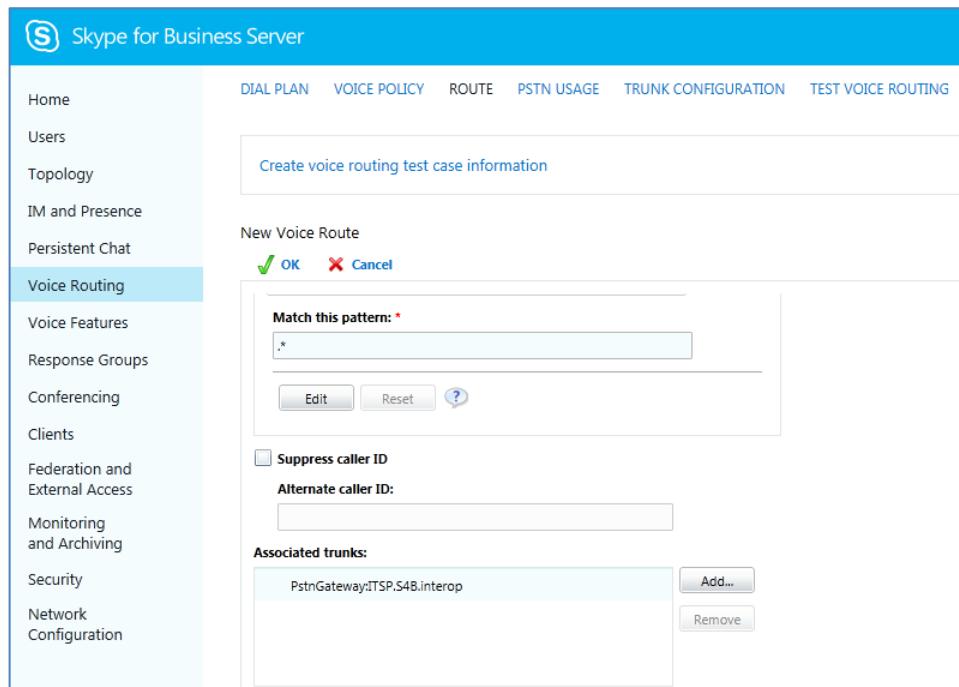
7. In the 'Name' field, enter a name for this route (e.g., **ITSP**).
8. In the 'Starting digits for numbers that you want to allow' field, enter the starting digits you want this route to handle (e.g., * to match all numbers), and then click **Add**.
9. Associate the route with the E-SBC Trunk that 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



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

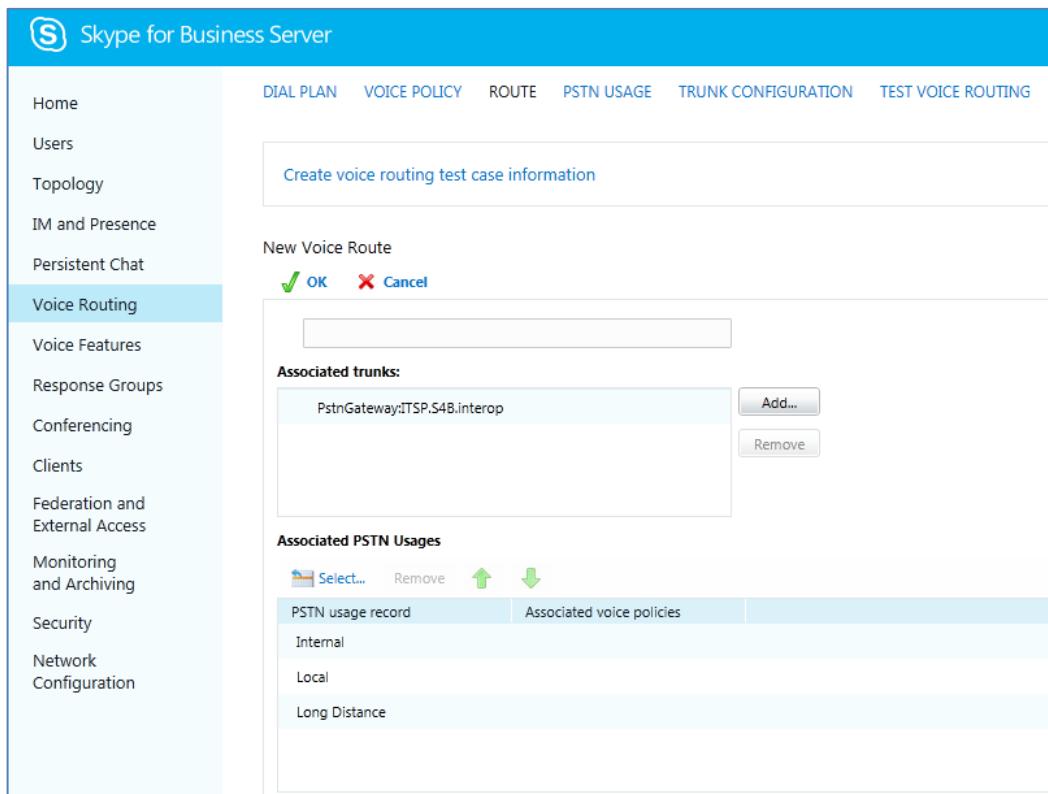
Figure 3-21: Selected E-SBC Trunk



10. Associate a PSTN Usage to this route:

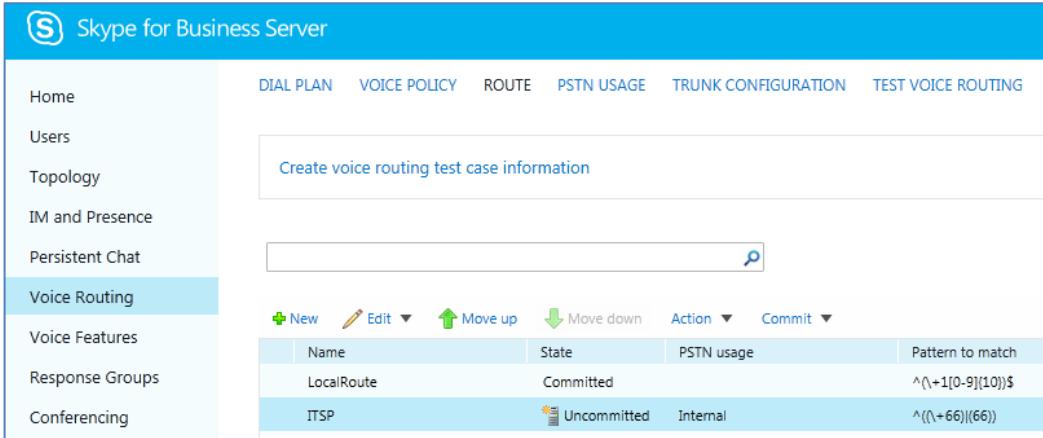
- Under the 'Associated PSTN Usages' group, click **Select** and then add the associated PSTN Usage.

Figure 3-22: Associating PSTN Usage to Route



- 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

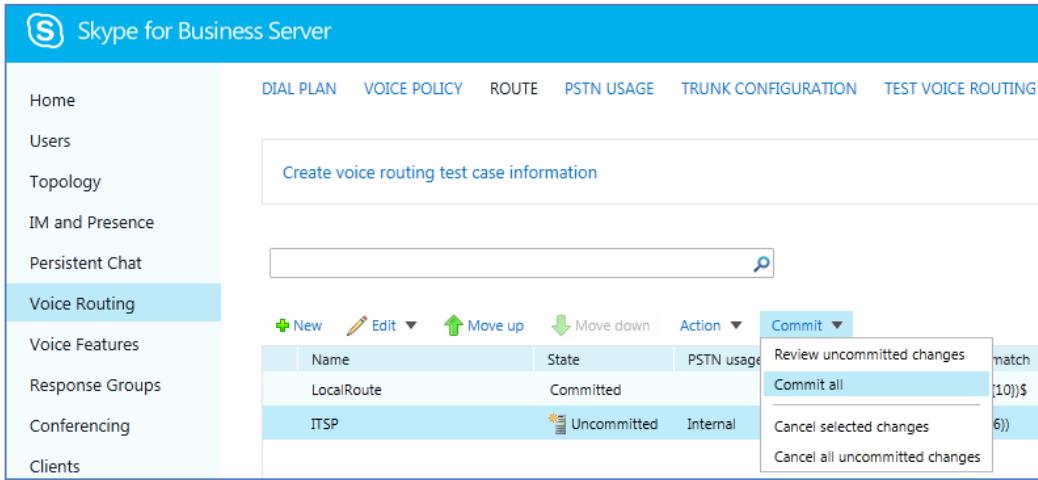


The screenshot shows the Skype for Business Server interface with the 'Voice Routing' tab selected. A table displays two voice routes: 'LocalRoute' (Committed) and 'ITSP' (Uncommitted). The 'ITSP' route has a pattern to match '^(\+1[0-9]{10})\$'. The 'Action' dropdown menu is open, showing options: 'Review uncommitted changes', 'Commit all', 'Cancel selected changes', and 'Cancel all uncommitted changes'. The 'Commit all' option is highlighted.

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

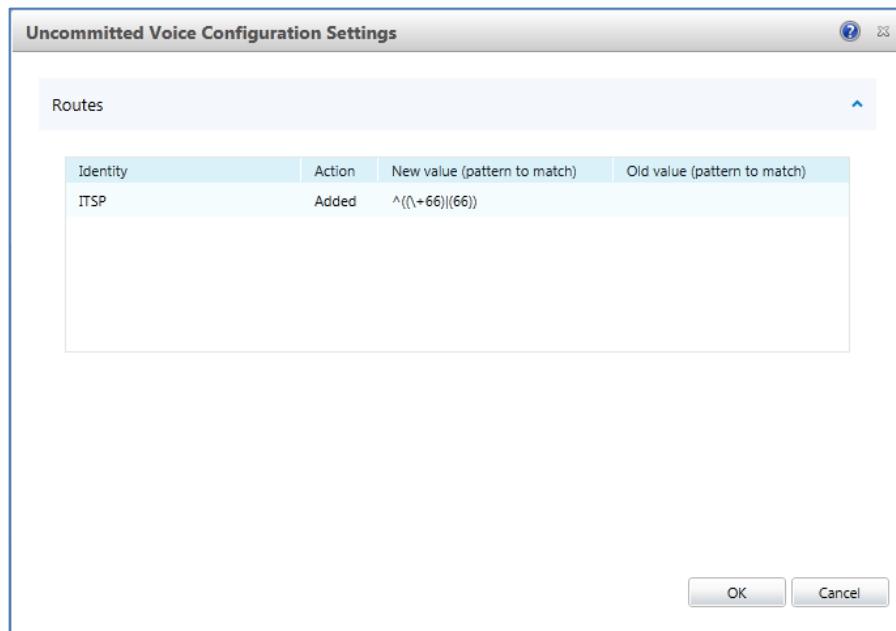


The screenshot shows the Skype for Business Server interface with the 'Voice Routing' tab selected. A table displays two voice routes: 'LocalRoute' (Committed) and 'ITSP' (Uncommitted). The 'Action' dropdown menu is open, showing options: 'Review uncommitted changes', 'Commit all', 'Cancel selected changes', and 'Cancel all uncommitted changes'. The 'Commit all' option is highlighted.

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

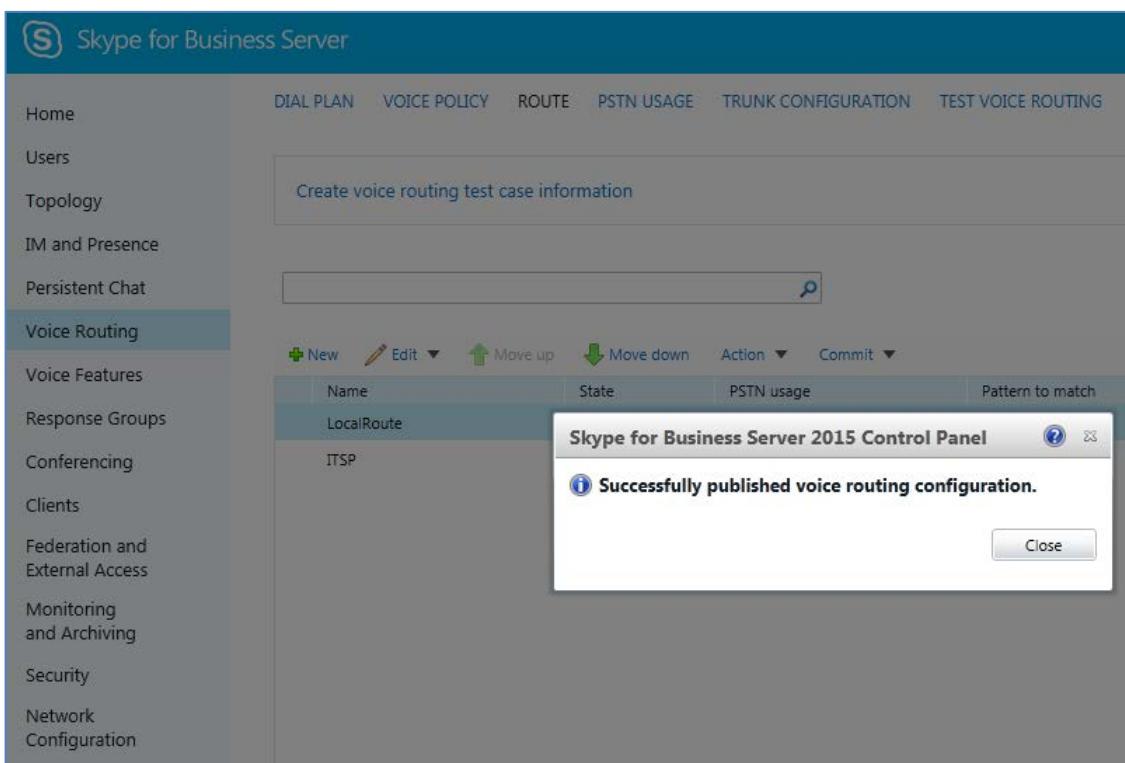
The Uncommitted Voice Configuration Settings page appears:

Figure 3-25: Uncommitted Voice Configuration Settings



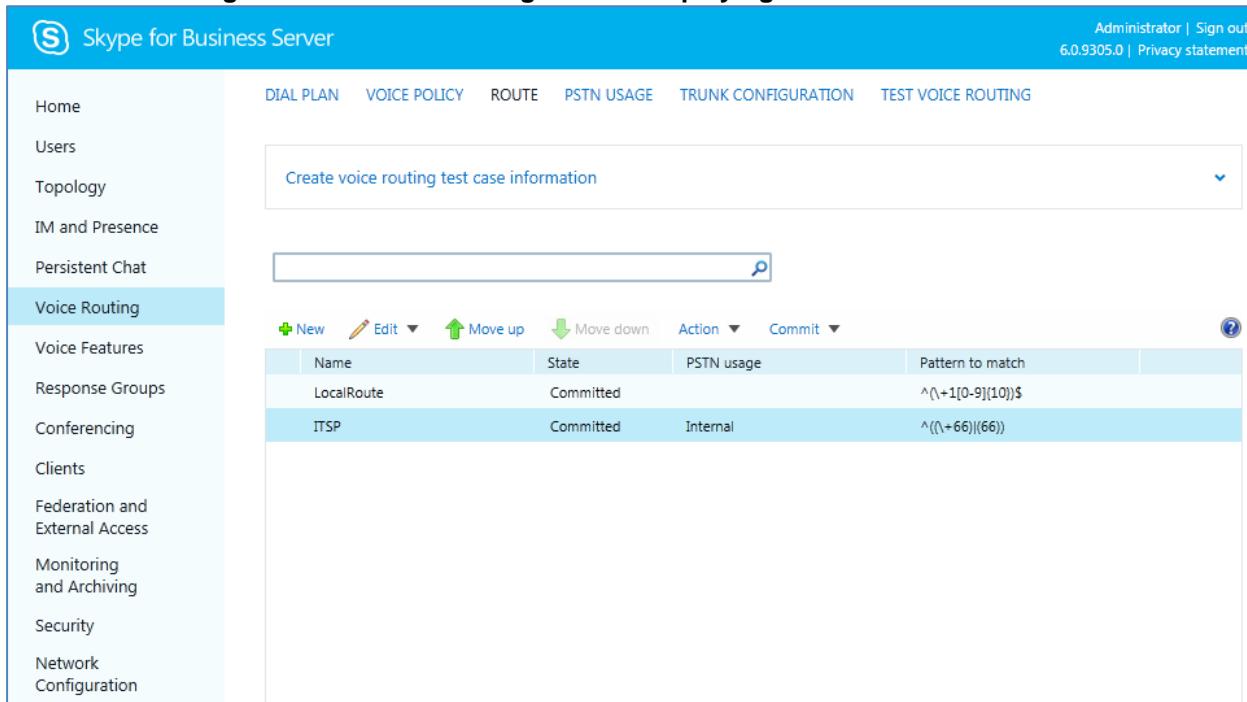
13. Click **Commit**; a message is displayed confirming a successful voice routing configuration, as shown below:

Figure 3-26: Confirmation of Successful Voice Routing Configuration



- 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



The screenshot shows the 'Voice Routing' tab selected in the left navigation menu. The main area displays a table of committed routes:

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

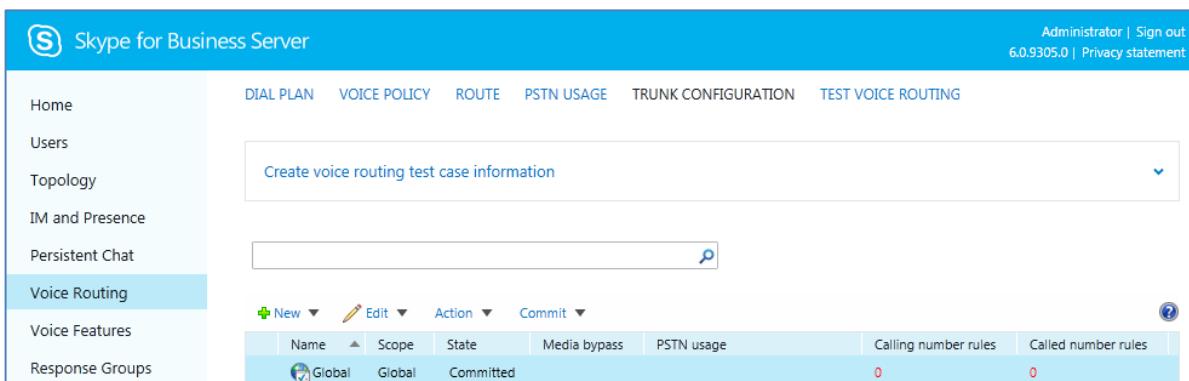
- 15.** For ITSPs that implement a call identifier, continue with the following steps:



Note: The SIP History-Info header provides a method to verify the identity (ID) of the call forwarder (i.e., the Skype for Business user number). This ID is required by BroadCloud 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 on page 45).

- 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.

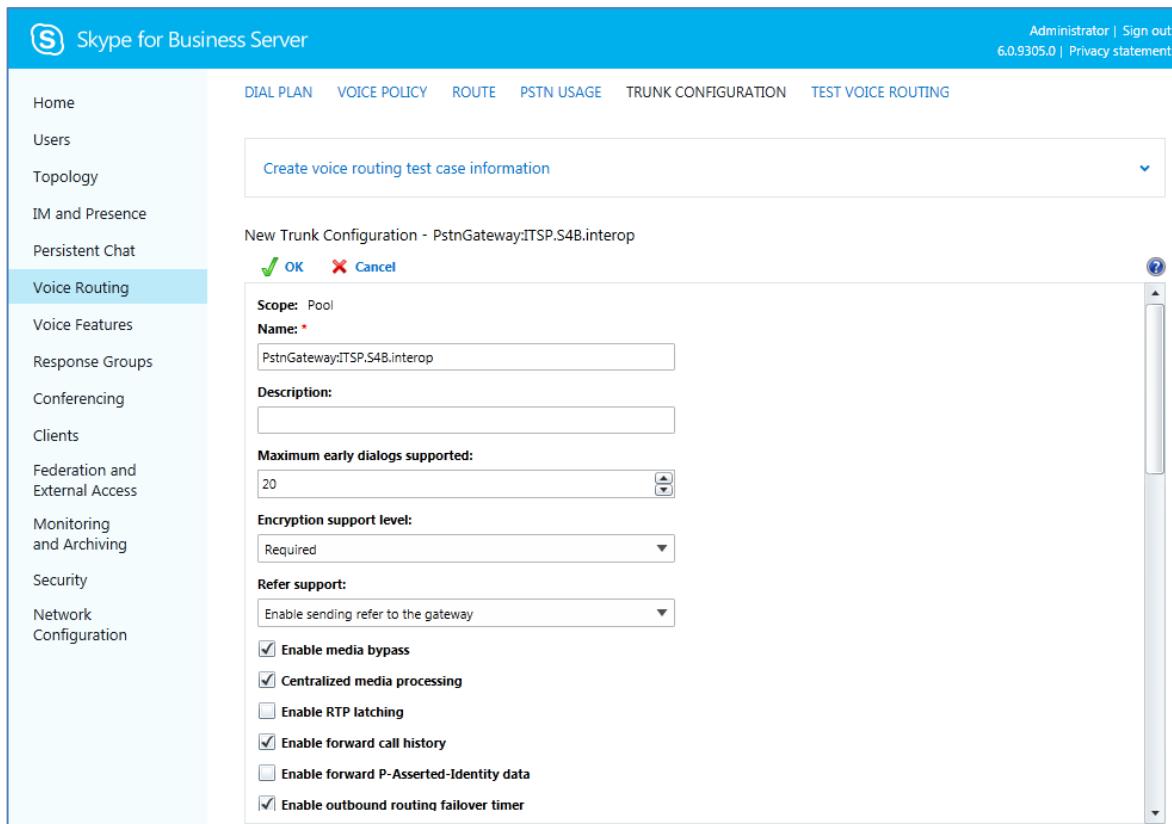
Figure 3-28: Voice Routing Screen – Trunk Configuration Tab



The screenshot shows the 'Trunk Configuration' tab selected in the left navigation menu. The main area displays a table of trunk configurations:

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:



- c. Select the **Enable forward call history** check box, and then click **OK**.
- d. Repeat Steps 11 through 13 to commit your settings.

16. Use the following command on the Skype for Business Server Management Shell after reconfiguration to verify correct values:

■ **Get-CsTrunkConfiguration**

```

Identity : 
Service:PstnGateway:ITSP.S4B.interop
OutboundTranslationRulesList : 
SipResponseCodeTranslationRulesList : {}
OutboundCallingNumberTranslationRulesList : {}
PstnUsages : {}
Description : 
ConcentratedTopology : True
EnableBypass : True
EnableMobileTrunkSupport : False
EnableReferSupport : True
EnableSessionTimer : True
EnableSignalBoost : False
MaxEarlyDialogs : 20
RemovePlusFromUri : False
RTCPActiveCalls : True
RTCPCallsOnHold : True
SRTPMode : Required
EnablePIDFLOSupport : False
EnableRTPLatching : False
EnableOnlineVoice : False
ForwardCallHistory : True

```

Enable3pccRefer	:	False
ForwardPAI	:	False
EnableFastFailoverTimer	:	True
EnableLocationRestriction	:	False
NetworkSiteID	:	

4 Configuring AudioCodes E-SBC

This chapter provides step-by-step procedures on how to configure AudioCodes E-SBC for interworking between Microsoft Skype for Business Server 2015 and the BroadCloud SIP Trunk. These configuration procedures are based on the interoperability test topology described in Section 2.4 on page 10, and includes the following main areas:

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

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

Notes:

- For implementing Microsoft Skype for Business and BroadCloud SIP Trunk based on the configuration described in this section, AudioCodes E-SBC must be installed with a License Key that includes the following software features:

- ✓ Microsoft
- ✓ SBC
- ✓ Security
- ✓ DSP
- ✓ RTP
- ✓ SIP



For more information about the License Key, contact your AudioCodes sales representative.

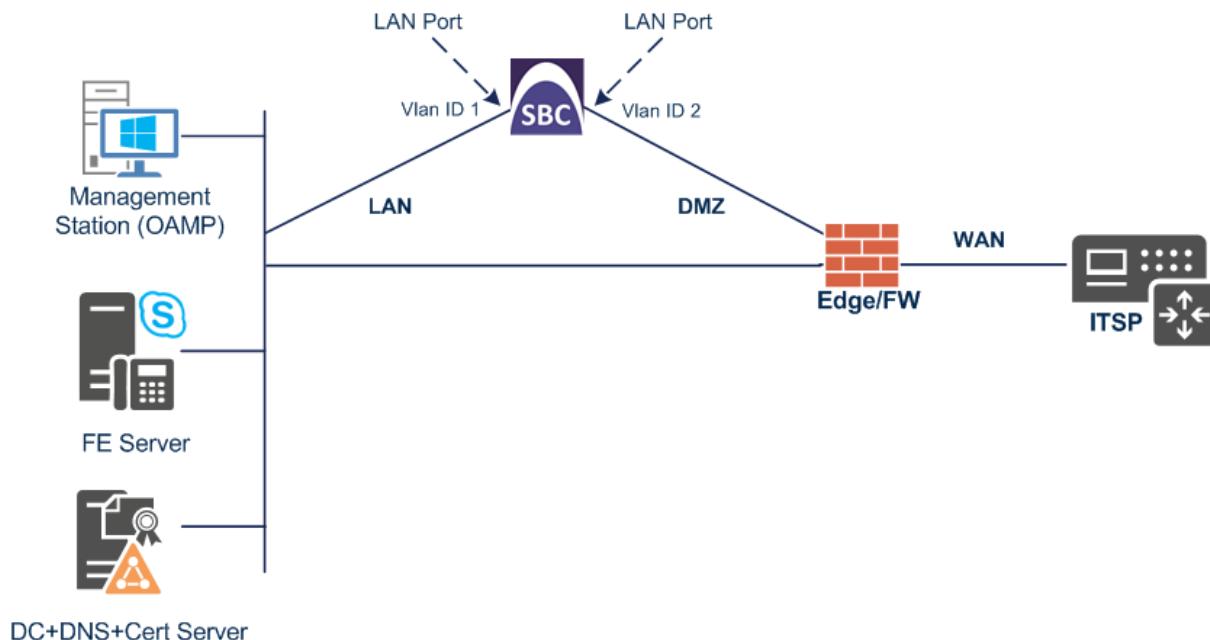
- The scope of this interoperability test and document does **not** cover all security aspects for configuring this topology. Comprehensive security measures should be implemented per your organization's security policies. For security recommendations on AudioCodes' products, refer to the *Recommended Security Guidelines* document.

4.1 Step 1: IP Network Interfaces Configuration

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

- E-SBC interfaces with the following IP entities:
 - Skype for Business servers, located on the LAN
 - BroadCloud 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 DMZ 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)
 - DMZ (VLAN ID 2)

Figure 4-1: Network Interfaces in Interoperability Test Topology



4.1.1 Step 1a: Configure VLANs

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

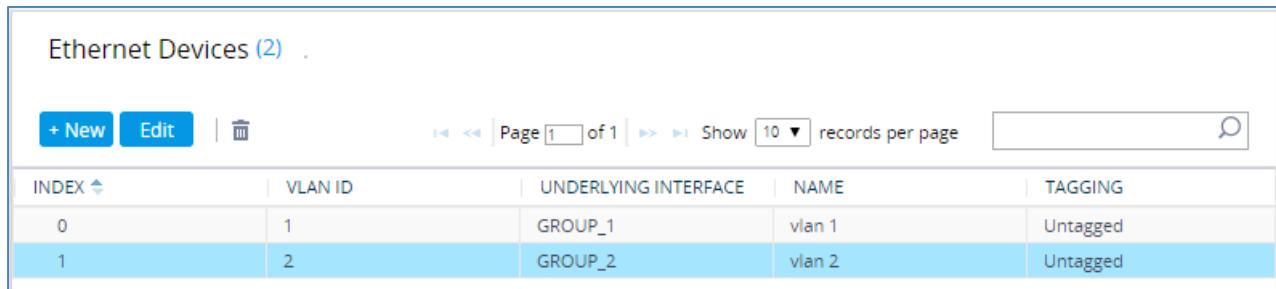
- LAN VoIP (assigned the name "LAN_IF")
- WAN VoIP (assigned the name "WAN_IF")

➤ **To configure the VLANs:**

1. Open the Ethernet Device table (**Setup** menu > **IP Network** tab > **Core Entities** folder > **Ethernet Devices**).
2. There will be one existing row for VLAN ID 1 and underlying interface GROUP_1.
3. Add another VLAN ID 2 for the WAN side as follows:

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

Figure 4-2: Configured VLAN IDs in Ethernet Device



The screenshot shows a table titled 'Ethernet Devices (2)' with the following data:

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 describes how to configure the IP network interfaces for each of the following interfaces:

- LAN VoIP (assigned the name "LAN_IF")
- WAN VoIP (assigned the name "WAN_IF")

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

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

Parameter	Value
Name	LAN_IF (arbitrary descriptive name)
Ethernet Device	vlan 1
IP Address	10.15.17.77 (LAN IP address of E-SBC)
Prefix Length	16 (subnet mask in bits for 255.255.0.0)
Default Gateway	10.15.0.1
Primary DNS	10.15.27.1

3. Add a network interface for the WAN side:

a. Click **New**.

b. Configure the interface as follows:

Parameter	Value
Name	WAN_IF
Application Type	Media + Control
Ethernet Device	vlan 2
IP Address	195.189.192.157 (DMZ IP address of E-SBC)
Prefix Length	25 (subnet mask in bits for 255.255.255.128)
Default Gateway	195.189.192.129 (router's IP address)
Primary DNS	80.179.52.100
Secondary DNS	80.179.55.100

4. Click **Apply**.

The configured IP network interfaces are shown below:

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

IP Interfaces (2) .									
				Page 1 of 1		Show 10 records per page			
INDEX	NAME	APPLICATION TYPE	INTERFACE MODE	IP ADDRESS	PREFIX LENGTH	DEFAULT GATEWAY	PRIMARY DNS	SECONDARY DNS	ETHERNET DEVICE
0	LAN_IF	OAMP + Media +	IPv4 Manual	10.15.17.77	16	10.15.0.1	10.15.27.1	0.0.0.0	vlan 1
1	WAN_IF	Media + Control	IPv4 Manual	195.189.192.157	25	195.189.192.129	80.179.52.100	80.179.55.100	vlan 2

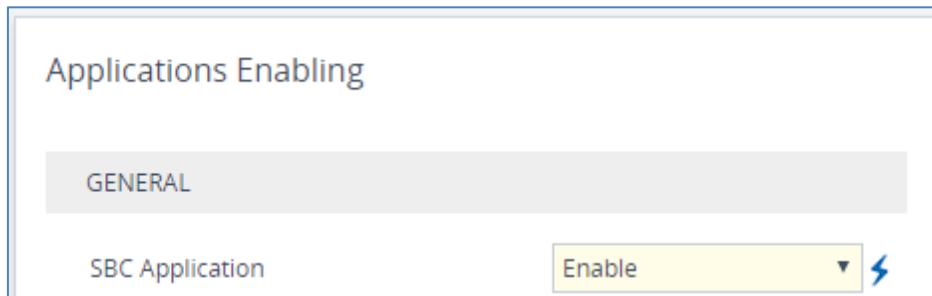
4.2 Step 2: Enable the SBC Application

This step describes how to enable the SBC application.

➤ **To enable the SBC application:**

1. Open the Applications Enabling page (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **Applications Enabling**).

Figure 4-4: Enabling SBC Application



2. From the 'SBC Application' drop-down list, select **Enable**.
3. Click **Apply**.
4. Reset the E-SBC with a burn to flash for this setting to take effect (see Section [4.17](#) on page [86](#)).

4.3 Step 3: Configure Media Realms

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

➤ **To configure Media Realms:**

1. Open the Media Realms table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **Media Realms**).
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
Name	MRLan (descriptive name)
IPv4 Interface Name	LAN_IF
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

Media Realms [MRLan]

GENERAL		QUALITY OF EXPERIENCE	
Index	0	QoE Profile	-- View
Name	MRLan	Bandwidth Profile	-- View
Topology Location	Down		
IPv4 Interface Name	#0 [LAN_IF] View		
Port Range Start	6000		
Number Of Media Session Legs	100		
Port Range End	6999		
Default Media Realm	No		
<input type="button" value="Cancel"/> <input type="button" value="APPLY"/>			

3. Configure a Media Realm for WAN traffic:

Parameter	Value
Index	1
Name	MRWan (arbitrary name)
Topology Location	Up
IPv4 Interface Name	WAN_IF
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

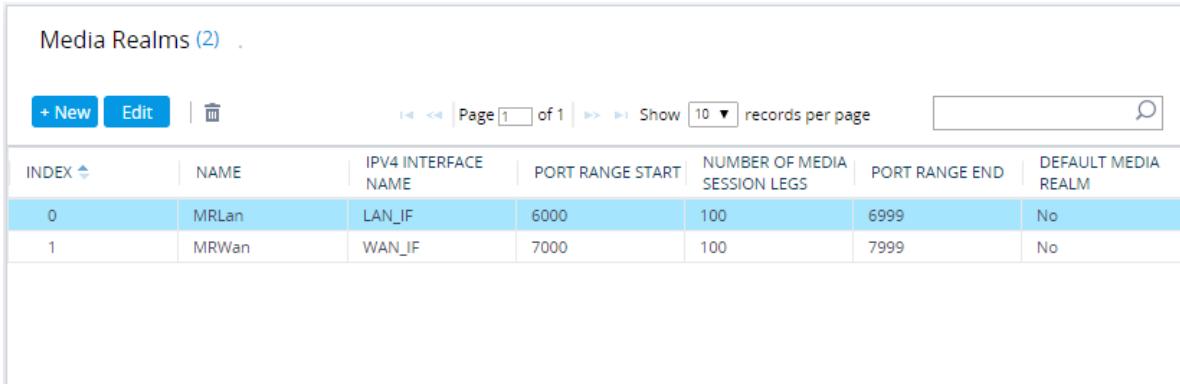
Media Realms [MRWan]

GENERAL		QUALITY OF EXPERIENCE	
Index	1	QoE Profile	-- View
Name	MRWan	Bandwidth Profile	-- View
Topology Location	Up		
IPv4 Interface Name	#1 [WAN_IF] View		
Port Range Start	7000		
Number Of Media Session Legs	100		
Port Range End	7999		
Default Media Realm	No		

Cancel **APPLY**

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	LAN_IF	6000	100	6999	No
1	MRWan	WAN_IF	7000	100	7999	No

4.4 Step 4: Configure SIP Signaling Interfaces

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

➤ **To configure SIP Interfaces:**

1. Open the SIP Interfaces table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **SIP Interfaces**).
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
Name	SIPInterface_LAN (see note at the end of this section)
Network Interface	LAN_IF
Application Type	SBC
UDP Port	0
TCP Port	0
TLS Port	5067 (see note below)
Media Realm	MRLan



Note: The TLS port parameter must be identically configured in the Skype for Business Topology Builder (see Section 3.1 on page 13).

3. Configure a SIP Interface for the WAN:

Parameter	Value
Index	1
Name	SIPInterface_WAN
Network Interface	WAN_IF
Application Type	SBC
UDP Port	0
TCP Port	0
TLS Port	5061
Media Realm	MRWan

The configured SIP Interfaces are shown in the figure below:

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

SIP Interfaces (2)									
		+ New Edit 		Page <input type="text" value="1"/> of 1		Show <input type="button" value="10"/> records per page		<input type="text"/> 	
INDEX	NAME	SRD	NETWORK INTERFACE	APPLICATION TYPE	UDP PORT	TCP PORT	TLS PORT	ENCAPSULATING PROTOCOL	MEDIA REALM
0	SIPInterface_LAN	 DefaultSRD (#)	Voice	SBC	0	0	5067	No encapsulation	MRLan
1	SIPInterface_WAN	 DefaultSRD (#)	WANSP	SBC	0	0	5061	No encapsulation	MRWan



Note: Current software releases uses the string **names** of the configuration entities (e.g., SIP Interface, Proxy Sets, and IP Groups). Therefore, it is recommended to configure each configuration entity with meaningful names for easy identification.

4.5 Step 5: Configure Proxy Sets

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

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

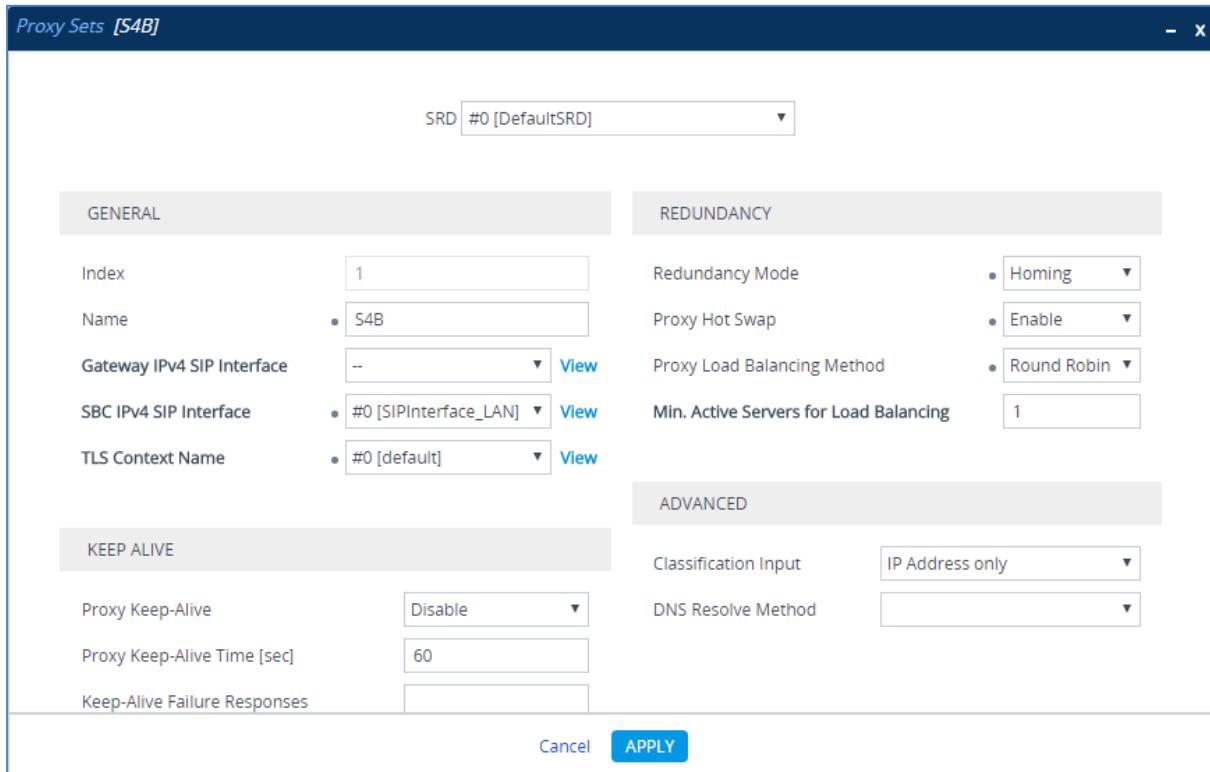
- Microsoft Skype for Business Server 2015
- BroadCloud SIP Trunk

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

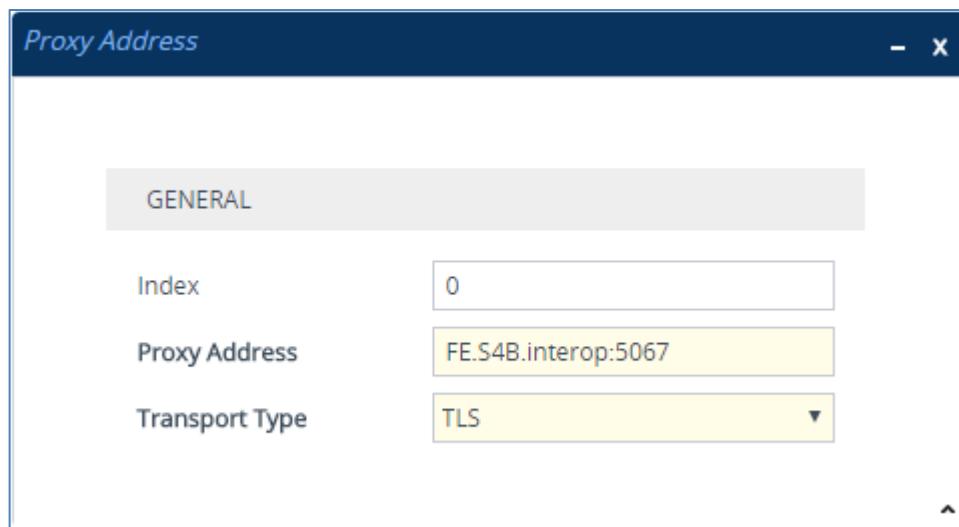
➤ **To configure Proxy Sets:**

1. Open the Proxy Sets table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder >**Proxy Sets**).
2. Add a Proxy Set for the Skype for Business Server 2015 as shown below:

Parameter	Value
Index	1
Name	S4B
SBC IPv4 SIP Interface	SIPInterface_LAN
TLS Context Name	default
Proxy Keep-Alive	Using Options
Redundancy Mode	Homing
Proxy Hot Swap	Enable
Proxy Load Balancing Method	Round Robin

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


- a. Select the index row of the Proxy Set that you added, and then click the **Proxy Address** link located below the table; the Proxy Address table opens.
- b. Click **New**; the following dialog box appears:

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


- c. Configure the address of the Proxy Set according to the parameters described in the table below.

d. Click **Apply**.

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

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

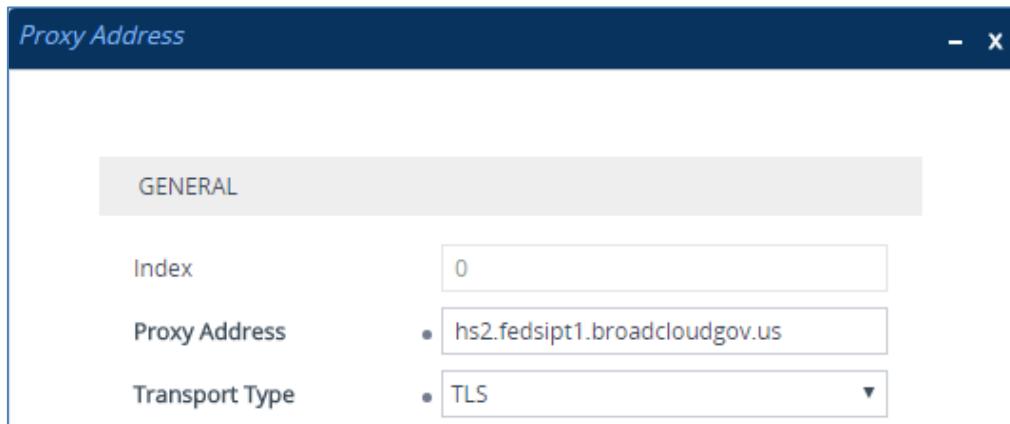
Parameter	Value
Index	2
Name	BroadCloud
SBC IPv4 SIP Interface	SIPInterface_WAN
TLS Context Name	default
Proxy Keep-Alive	Using Options
Redundancy Mode	Homing
Proxy Hot Swap	Enable
DNS Resolve Method	SRV

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

- a. Select the index row of the Proxy Set that you added, and then click the **Proxy Address** link located below the table; the Proxy Address table opens.

- b. Click **New**; the following dialog box appears:

Figure 4-12: Configuring Proxy Address for BroadCloud SIP Trunk



- c. Configure the address of the Proxy Set according to the parameters described in the table below.

- d. Click **Apply**.

Parameter	Value
Index	0
Proxy Address	hs2.fedsipt1.broadcloudgov.us (IP address / FQDN and destination port)
Transport Type	TLS

The configured Proxy Sets are shown in the figure below:

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

Proxy Sets (2) .								
Actions		List View						
INDEX	NAME	SRD	GATEWAY IPV4 SIP INTERFACE	SBC IPV4 SIP INTERFACE	PROXY KEEP-ALIVE TIME [SEC]	REDUNDANCY MODE	PROXY HOT SWAP	
1	S4B	DefaultSRD (#0)	--	SIPInterface_LAN	60	Homing	Enable	
2	BroadCloud	DefaultSRD (#0)	--	SIPInterface_WAN	60	Homing	Enable	

4.6 Step 6: Configure Coders

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

Note that the Coder Group ID for this entity will be assigned to its corresponding IP Profile in the next step.

➤ **To configure coders:**

1. Open the Coder Groups table (**Setup** menu > **Signaling & Media** tab > **Coders & Profiles** folder > **Coder Groups**).
2. Configure a Coder Group for Skype for Business Server 2015:

Parameter	Value
Coder Group Name	AudioCodersGroups_0
Coder Name	<ul style="list-style-type: none"> ▪ G.711 U-law ▪ G.711 A-law
Silence Suppression	Enable (for both coders)

Figure 4-14: Configuring Coder Group for Skype for Business Server 2015

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 BroadCloud SIP Trunk:

Parameter	Value
Coder Group Name	AudioCodersGroups_1
Coder Name	<ul style="list-style-type: none"> ▪ G.729 ▪ G.711 A-law ▪ G.711 U-law

Figure 4-15: Configuring Coder Group for BroadCloud SIP Trunk

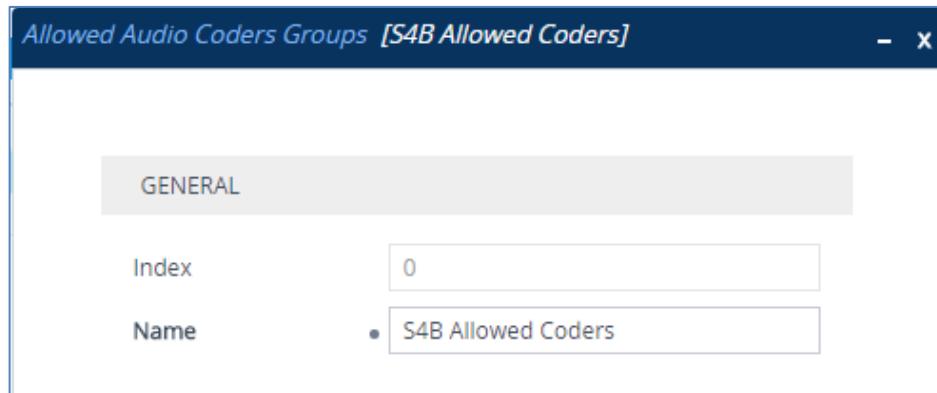
Coder Name	Packetization Time	Rate	Payload Type	Silence Suppression	Coder Specific
G.729	20	64	8	Disabled	
G.711A-law	20	64	8	Disabled	
G.711U-law	20	64	0	Disabled	

The procedure below describes how to configure an Allowed Coders Group to ensure that voice sent to the BroadCloud SIP Trunk uses the G.729 coder whenever possible. Note that this Allowed Coders Group ID will be assigned to the IP Profile belonging to the BroadCloud SIP Trunk in the next step.

➤ **To set a preferred coder for the Skype for Business Server 2015:**

1. Open the Allowed Audio Coders Groups table (**Setup** menu > **Signaling & Media** tab > **Coders & Profiles** folder > **Allowed Audio Coders Groups**).
2. Click **New** and configure a name for the Allowed Audio Coders Group Skype for Business Server 2015.

Figure 4-16: Configuring Allowed Coders Group for Skype for Business Server 2015



3. Click **Apply**.
4. Select the new row that you configured, and then click the **Allowed Audio Coders** link located below the table; the Allowed Audio Coders table opens.
5. Click **New** and configure an Allowed Coders as follows:

Parameter	Value
Index	0
Coder	G.711 A-law
Index	1
Coder	G.711 U-law

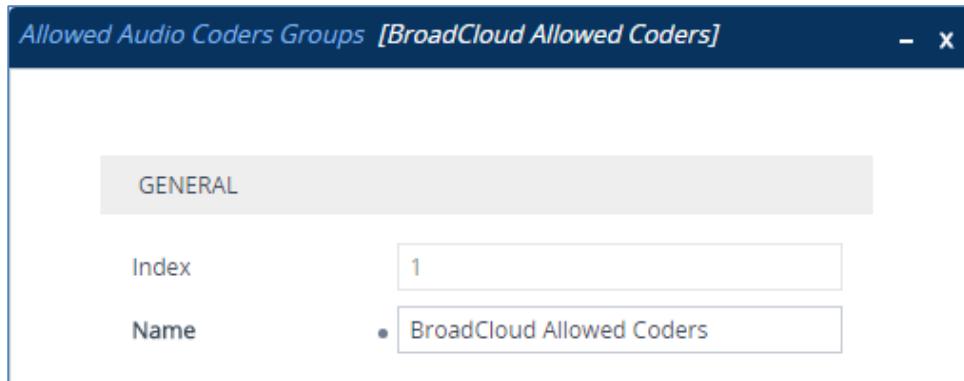
Figure 4-17: Configuring Allowed Coders the Skype for Business Server 2015

INDEX	CODER	USER-DEFINED CODER
0	G.711A-law	
1	G.711U-law	

➤ To set a preferred coder for the BroadCloud SIP Trunk:

1. Open the Allowed Audio Coders Groups table (Setup menu > **Signaling & Media** tab > **Coders & Profiles** folder > **Allowed Audio Coders Groups**).
2. Click **New** and configure a name for the Allowed Audio Coders Group for BroadCloud SIP Trunk.

Figure 4-18: Configuring Allowed Coders Group for BroadCloud SIP Trunk



3. Click **Apply**.
4. Select the new row that you configured, and then click the **Allowed Audio Coders** link located below the table; the Allowed Audio Coders table opens.
5. Click **New** and configure an Allowed Coders as follows:

Parameter	Value
Index	0
Coder	G.729
Index	1
Coder	G.711 A-law
Index	2
Coder	G.711 U-law

Figure 4-19: Configuring Allowed Coders for BroadCloud SIP Trunk

INDEX	CODER	USER-DEFINED CODER
0	G.729	
1	G.711A-law	
2	G.711U-law	

6. Open the Media Settings page (**Setup** menu > **Signaling & Media** tab > **Media** folder > **Media Settings**).

Figure 4-20: SBC Preferences Mode

Media Settings

<div style="border-bottom: 1px solid #ccc; padding-bottom: 5px; margin-bottom: 5px;"> GENERAL </div> <table border="0" style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 30%; vertical-align: top;"> NAT Traversal </td> <td style="width: 40%; text-align: center;"> <input type="button" value="Disable NAT"/> </td> <td style="width: 30%; text-align: right;"> New RTP Stream Packets </td> <td style="text-align: right;">3</td> </tr> <tr> <td>Enable Continuity Tones</td> <td style="text-align: center;"> <input type="button" value="Disable"/> </td> <td>New RTCP Stream Packets</td> <td style="text-align: right;">3</td> </tr> <tr> <td>Inbound Media Latch Mode</td> <td style="text-align: center;"> <input type="button" value="Dynamic"/> </td> <td>New SRTP Stream Packets</td> <td style="text-align: right;">3</td> </tr> <tr> <td>Number of Media Channels</td> <td style="text-align: center;"> <input type="button" value="0"/> </td> <td>New SRTCP Stream Packets</td> <td style="text-align: right;">3</td> </tr> <tr> <td>Enforce Media Order</td> <td style="text-align: center;"> <input type="button" value="Disable"/> </td> <td>Timeout To Relatch RTP (msec)</td> <td style="text-align: right;">200</td> </tr> <tr> <td>SDP Session Owner</td> <td style="text-align: center;"> <input type="button" value="AudiocodesGW"/> </td> <td>Timeout To Relatch SRTP (msec)</td> <td style="text-align: right;">200</td> </tr> </table> <div style="border-bottom: 1px solid #ccc; padding-bottom: 5px; margin-bottom: 5px;"> SBC SETTINGS </div> <table border="0" style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 30%; vertical-align: top;"> Preferences Mode </td> <td style="width: 40%; text-align: center;"> <input type="button" value="Include Extensions"/> </td> <td style="width: 30%; text-align: right;"> <input type="button" value="←"/> </td> </tr> <tr> <td>Enforce Media Order</td> <td style="text-align: center;"> <input type="button" value="Disable"/> </td> <td>Timeout To Relatch Silence (msec)</td> <td style="text-align: right;">10000</td> </tr> <tr> <td></td> <td></td> <td>Timeout To Relatch RTCP (msec)</td> <td style="text-align: right;">10000</td> </tr> </table> <div style="border-bottom: 1px solid #ccc; padding-bottom: 5px; margin-bottom: 5px;"> GATEWAY SETTINGS </div> <table border="0" style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 30%; vertical-align: top;"> Enable Early Media </td> <td style="width: 40%; text-align: center;"> <input type="button" value="Disable"/> </td> <td style="width: 30%; text-align: right;"> </td> </tr> <tr> <td>Multiple Packetization Time Format</td> <td style="text-align: center;"> <input type="button" value="None"/> </td> <td></td> </tr> </table>	NAT Traversal	<input type="button" value="Disable NAT"/>	New RTP Stream Packets	3	Enable Continuity Tones	<input type="button" value="Disable"/>	New RTCP Stream Packets	3	Inbound Media Latch Mode	<input type="button" value="Dynamic"/>	New SRTP Stream Packets	3	Number of Media Channels	<input type="button" value="0"/>	New SRTCP Stream Packets	3	Enforce Media Order	<input type="button" value="Disable"/>	Timeout To Relatch RTP (msec)	200	SDP Session Owner	<input type="button" value="AudiocodesGW"/>	Timeout To Relatch SRTP (msec)	200	Preferences Mode	<input type="button" value="Include Extensions"/>	<input type="button" value="←"/>	Enforce Media Order	<input type="button" value="Disable"/>	Timeout To Relatch Silence (msec)	10000			Timeout To Relatch RTCP (msec)	10000	Enable Early Media	<input type="button" value="Disable"/>		Multiple Packetization Time Format	<input type="button" value="None"/>		<div style="border-bottom: 1px solid #ccc; padding-bottom: 5px; margin-bottom: 5px;"> ROBUSTNESS </div> <table border="0" style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 30%; vertical-align: top;"> </td> <td style="width: 40%; text-align: center;"> </td> <td style="width: 30%; text-align: right;"> </td> </tr> <tr> <td>New RTP Stream Packets</td> <td style="text-align: center;">3</td> <td></td> </tr> <tr> <td>New RTCP Stream Packets</td> <td style="text-align: center;">3</td> <td></td> </tr> <tr> <td>New SRTP Stream Packets</td> <td style="text-align: center;">3</td> <td></td> </tr> <tr> <td>New SRTCP Stream Packets</td> <td style="text-align: center;">3</td> <td></td> </tr> <tr> <td>Timeout To Relatch RTP (msec)</td> <td style="text-align: center;">200</td> <td></td> </tr> <tr> <td>Timeout To Relatch SRTP (msec)</td> <td style="text-align: center;">200</td> <td></td> </tr> <tr> <td>Timeout To Relatch Silence (msec)</td> <td style="text-align: center;">10000</td> <td></td> </tr> <tr> <td>Timeout To Relatch RTCP (msec)</td> <td style="text-align: center;">10000</td> <td></td> </tr> </table>				New RTP Stream Packets	3		New RTCP Stream Packets	3		New SRTP Stream Packets	3		New SRTCP Stream Packets	3		Timeout To Relatch RTP (msec)	200		Timeout To Relatch SRTP (msec)	200		Timeout To Relatch Silence (msec)	10000		Timeout To Relatch RTCP (msec)	10000	
NAT Traversal	<input type="button" value="Disable NAT"/>	New RTP Stream Packets	3																																																																		
Enable Continuity Tones	<input type="button" value="Disable"/>	New RTCP Stream Packets	3																																																																		
Inbound Media Latch Mode	<input type="button" value="Dynamic"/>	New SRTP Stream Packets	3																																																																		
Number of Media Channels	<input type="button" value="0"/>	New SRTCP Stream Packets	3																																																																		
Enforce Media Order	<input type="button" value="Disable"/>	Timeout To Relatch RTP (msec)	200																																																																		
SDP Session Owner	<input type="button" value="AudiocodesGW"/>	Timeout To Relatch SRTP (msec)	200																																																																		
Preferences Mode	<input type="button" value="Include Extensions"/>	<input type="button" value="←"/>																																																																			
Enforce Media Order	<input type="button" value="Disable"/>	Timeout To Relatch Silence (msec)	10000																																																																		
		Timeout To Relatch RTCP (msec)	10000																																																																		
Enable Early Media	<input type="button" value="Disable"/>																																																																				
Multiple Packetization Time Format	<input type="button" value="None"/>																																																																				
New RTP Stream Packets	3																																																																				
New RTCP Stream Packets	3																																																																				
New SRTP Stream Packets	3																																																																				
New SRTCP Stream Packets	3																																																																				
Timeout To Relatch RTP (msec)	200																																																																				
Timeout To Relatch SRTP (msec)	200																																																																				
Timeout To Relatch Silence (msec)	10000																																																																				
Timeout To Relatch RTCP (msec)	10000																																																																				

7. From the '**Preferences Mode**' drop-down list, select **Include Extensions**.
8. Click **Apply**.

4.7 Step 7: Configure IP Profiles

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

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

- Microsoft Skype for Business Server 2015 – to operate in secure mode using SRTP and SIP over TLS
- BroadCloud SIP trunk – to operate in secure mode using SRTP and SIP over TLS

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

1. Open the IP Profiles table (**Setup** menu > **Signaling & Media** tab > **Coders & Profiles** folder > **IP Profiles**).
2. Click **New**, and then configure the parameters as follows:

Parameter	Value
General	
Index	1
Name	S4B
Media Security	
SBC Media Security Mode	SRTP
Symmetric MKI	Enable
MKI Size	1
Enforce MKI Size	Enforce
Reset SRTP State Upon Re-key	Enable
Generate SRTP Keys Mode:	Always
SBC Early Media	
Remote Early Media RTP Detection Mode	By Media (required, as Skype for Business Server 2015 does not send RTP immediately to remote side when it sends a SIP 18x response)
SBC Media	
Extension Coders Group	AudioCodersGroups_0
Allowed Audio Coders	S4B Allowed Coders
RFC 2833 Mode	Extend
RFC 2833 DTMF Payload Type	101
SBC Signaling	
Remote Update Support	Supported Only After Connect
Remote re-INVITE Support	Supported Only With SDP
Remote Delayed Offer Support	Not Supported
SBC Forward and Transfer	

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)
SBC Hold	
Remote Hold Format	Inactive
Media	
Broken Connection Mode	Ignore

Figure 4-21: Configuring IP Profile for Skype for Business Server 2015

IP Profiles [S4B]

GENERAL		SBC SIGNALING	
Index	1	PRACK Mode	Transparent
Name	S4B	P-Asserted-Identity Header Mode	As Is
Created by Routing Server	No	Diversion Header Mode	As Is
MEDIA SECURITY		History-Info Header Mode	
SBC Media Security Mode	SRTP	Session Expires Mode	Transparent
Gateway Media Security Mode	Preferable	Remote Update Support	Supported Only After Conn
Symmetric MKI	Enable	Remote re-INVITE	Supported only with SDP
MKI Size	1	Remote Delayed Offer Support	Not Supported
SBC Enforce MKI Size	Enforce	Remote Representation Mode	According to Operation Mo
SBC Media Security Method	SDES	Keep Incoming Via Headers	According to Operation Mo
		Keep Incoming Routing Headers	According to Operation Mo
		Keep User-Agent Header	According to Operation Mo
		Cancel	APPLY

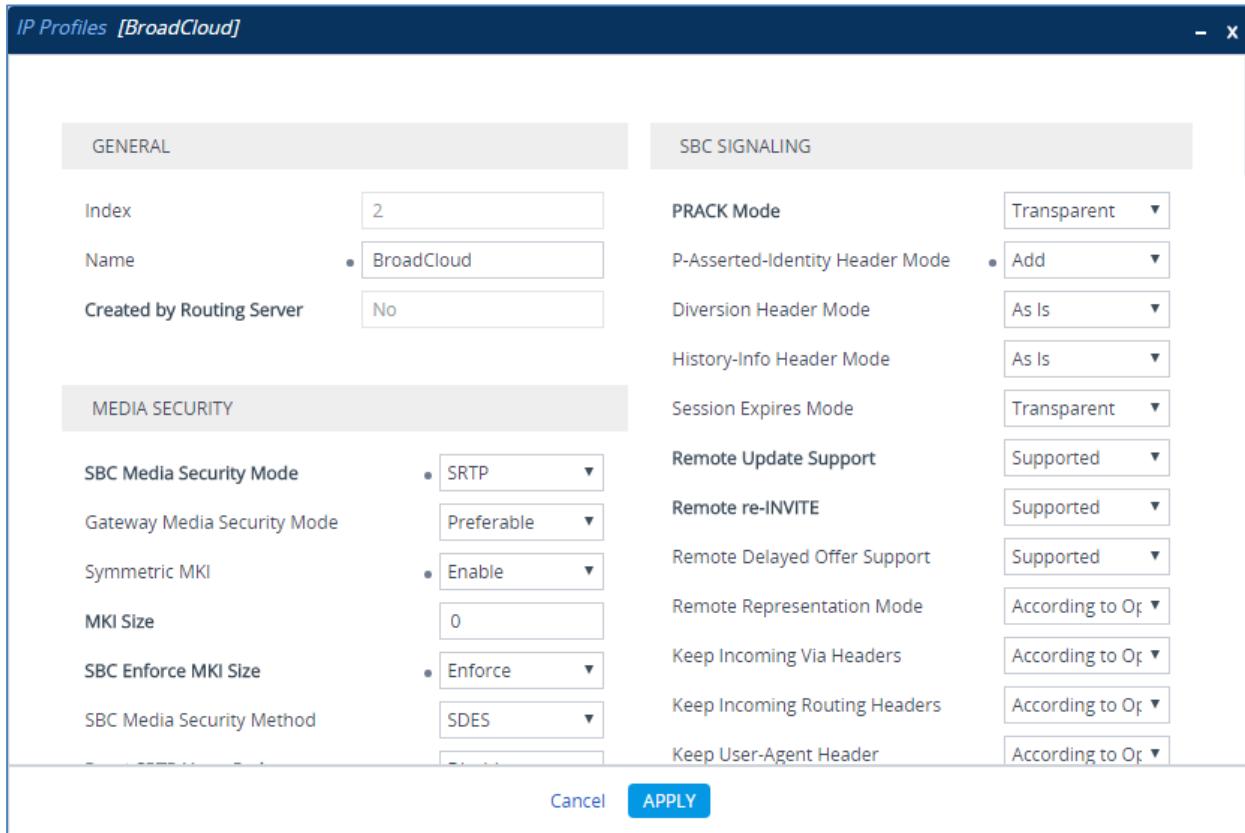
3. Click **Apply**.

➤ **To configure an IP Profile for the BroadCloud SIP Trunk:**

1. Click **New**, and then configure the parameters as follows:

Parameter	Value
General	
Index	2
Name	BroadCloud
Media Security	
SBC Media Security Mode	SRTP
Symmetric MKI	Enable
SBC Enforce MKI Size	Enforce
SBC Remove Crypto Lifetime in SDP	Yes
SBC Early Media	
Remote Can Play Ringback	No (required, as Skype for Business Server 2015 does not provide a ringback tone for incoming calls)
SBC Media	
Extension Coders Group	AudioCodersGroups_1
Allowed Audio Coders	BroadCloud Allowed Coders
Allowed Coders Mode	Restriction and Preference (lists Allowed Coders first and then original coders in received SDP offer)
SBC Signaling	
P-Asserted-Identity Header Mode	Add (required for anonymous calls)
SBC Forward and Transfer	
Remote REFER Mode	Handle Locally (required, as Skype for Business Server 2015 does not support receipt of SIP REFER)
Play RBT To Transferee	Yes
Remote 3xx Mode	Handle Locally
Media	
Broken Connection Mode	Ignore

Figure 4-22: Configuring IP Profile for BroadCloud SIP Trunk



GENERAL		SBC SIGNALING	
Index	2	PRACK Mode	Transparent ▾
Name	BroadCloud	P-Asserted-Identity Header Mode	Add ▾
Created by Routing Server	No	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 ▾			
Remote Representation Mode According to Opt. ▾			
Keep Incoming Via Headers According to Opt. ▾			
Keep Incoming Routing Headers According to Opt. ▾			
Keep User-Agent Header According to Opt. ▾			

MEDIA SECURITY	
SBC Media Security Mode	SRTP ▾
Gateway Media Security Mode	Preferable ▾
Symmetric MKI	Enable ▾
MKI Size	0
SBC Enforce MKI Size	Enforce ▾
SBC Media Security Method	SDES ▾

Cancel **APPLY**

2. Click **Apply**.

4.8 Step 8: Configure IP Groups

This step describes how to configure IP Groups. The IP Group represents an IP entity on the network with which the E-SBC communicates. This can be a server (e.g., IP PBX or ITSP) or it can be a group of users (e.g., LAN IP phones). For servers, the IP Group is typically used to define the server's IP address by associating it with a Proxy Set. Once IP Groups are configured, they are used to configure IP-to-IP routing rules for denoting source and destination of the call.

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

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

➤ To configure IP Groups:

1. Open the IP Groups table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **IP Groups**).
2. Add an IP Group for the Skype for Business Server 2015:

Parameter	Value
Index	1
Name	S4B
Type	Server
Proxy Set	S4B
IP Profile	S4B
Media Realm	MRLan
SIP Group Name	interop.adpt-tech.com (according to ITSP requirement)

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

Parameter	Value
Index	2
Name	BroadCloud
Topology Location	Up
Type	Server
Proxy Set	BroadCloud
IP Profile	BroadCloud
Media Realm	MRWan
SIP Group Name	interop.adpt-tech.com (according to ITSP requirement)

The configured IP Groups are shown in the figure below:

Figure 4-23: Configured IP Groups in IP Group Table

IP Groups (2)

+ New	Edit	Delete	Page 1 of 1 Show 10 records per page	Search							
INDEX	NAME	SRD	TYPE	SBC OPERATION MODE	PROXY SET	IP PROFILE	MEDIA REALM	SIP GROUP NAME	CLASSIFY BY PROXY SET	INBOUND MESSAGE MANIPULATION SET	OUTBOUND MESSAGE MANIPULATION SET
1	S4B	Defaults	Server	Not Configured	S4B	S4B	MRLan	interop.adpt	Enable	-1	-1
2	BroadCloud	Defaults	Server	Not Configured	BroadCloud	BroadCloud	MRWan	interop.adpt	Enable	-1	4

4.9 Step 9: SIP TLS Connection Configuration

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

4.9.1 Step 9a: Configure the NTP Server Address

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

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

1. Open the Time & Date page (**Setup** menu > **Administration** tab > **Time & Date**).
2. In the 'Primary NTP Server Address' field, enter the IP address of the NTP server (e.g., **10.15.27.1**).

Figure 4-24: Configuring NTP Server Address

NTP SERVER	
Primary NTP Server Address (IP or FQDN)	• <input type="text" value="10.15.27.1"/>
Secondary NTP Server Address (IP or FQDN)	<input type="text"/>
NTP Update Interval	Hours: <input type="text" value="24"/> Minutes: <input type="text" value="0"/>
NTP Authentication Key Identifier	<input type="text" value="0"/>
NTP Authentication Secret Key	<input type="text"/>

3. Click **Apply**.

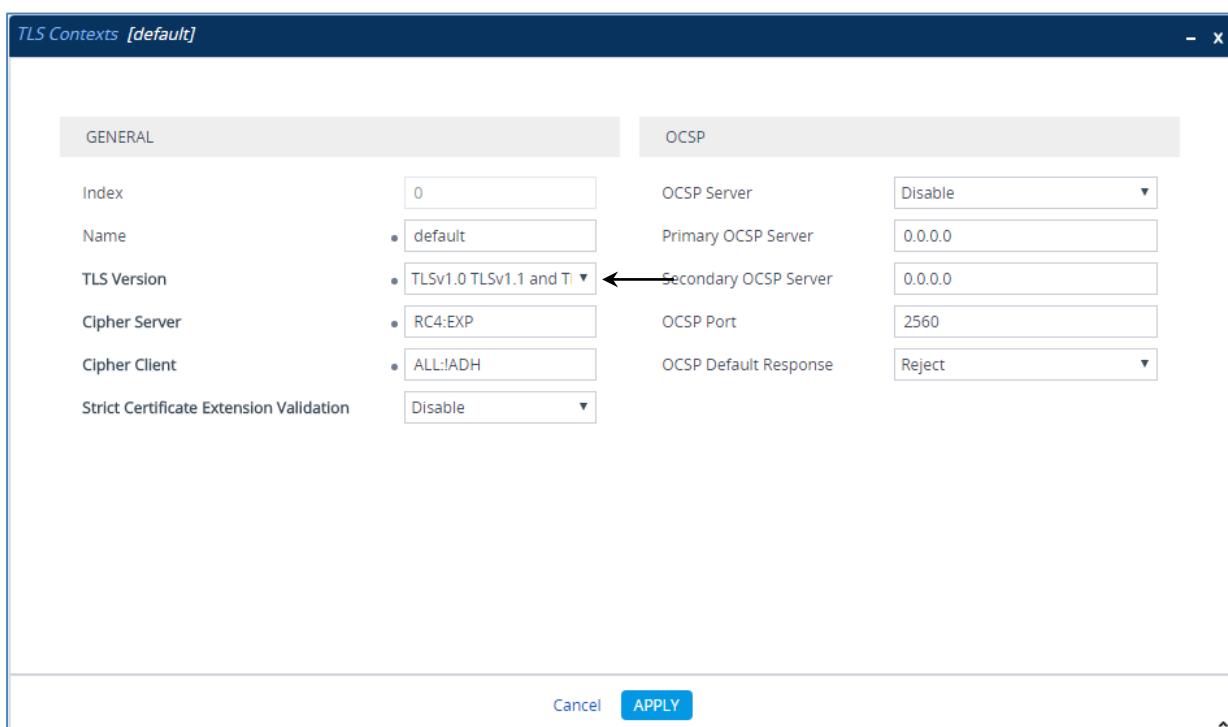
4.9.2 Step 9b: Configure the TLS version

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

➤ **To configure the TLS version:**

1. Open the TLS Contexts table (**Setup** menu > **IP Network** tab > **Security** folder > **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. From the '**TLS Version**' drop-down list, select '**TLSv1.0 TLSv1.1 and TLSv1.2**'

Figure 4-25: Configuring TLS version



GENERAL				OCSP	
Index	0	OCSP Server	Disable		
Name	default	Primary OCSP Server	0.0.0.0		
TLS Version	TLSv1.0 TLSv1.1 and TLSv1.2	Secondary OCSP Server	0.0.0.0		
Cipher Server	RC4:EXP	OCSP Port	2560		
Cipher Client	ALL:!ADH	OCSP Default Response	Reject		
Strict Certificate Extension Validation	Disable				

4. Click **Apply**.

4.9.3 Step 9c: Configure a Certificate for Operation with Microsoft Skype for Business Server 2015

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

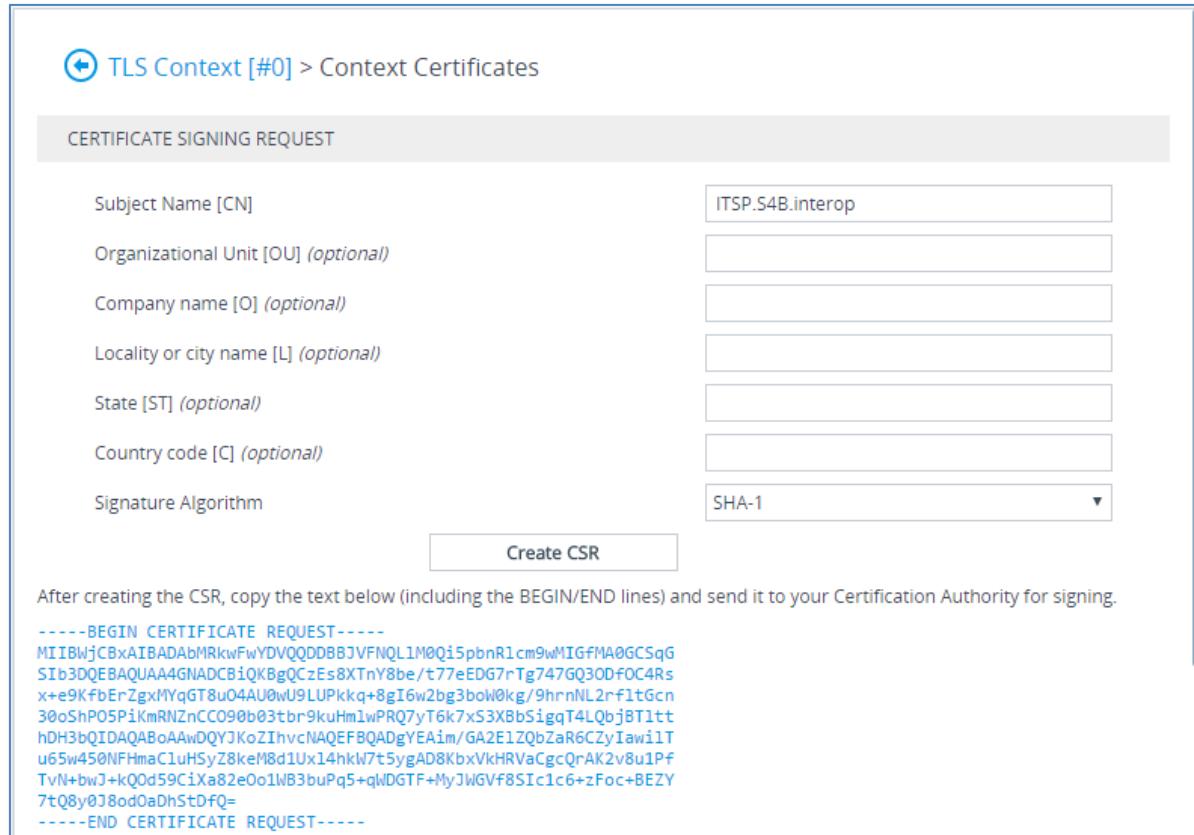


Note: The Subject Name (CN) field parameter should be identically configured in the DNS Active Directory and Topology Builder (see Section 3.1 on page 13).

➤ **To configure a certificate:**

1. Open the TLS Contexts page (**Setup** menu > **IP Network** tab > **Security** folder > **TLS Contexts**).
2. In the TLS Contexts page, select the required TLS Context index row, and then click the **Change Certificate** link located below the table; 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.
 - c. Click the **Create CSR** button; a textual certificate signing request is displayed in the area below the button:

Figure 4-26: Certificate Signing Request – Creating CSR

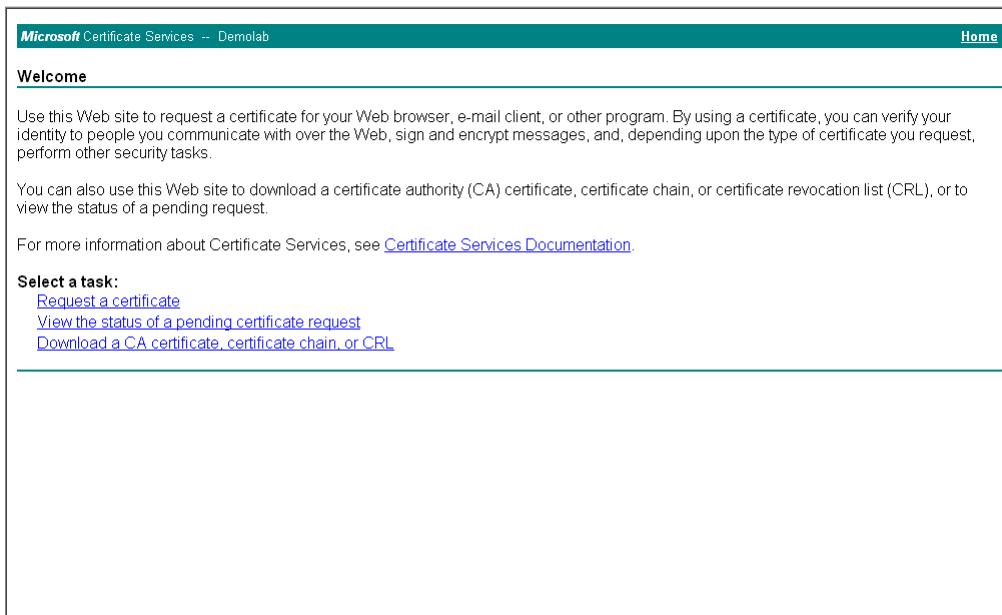


After creating the CSR, copy the text below (including the BEGIN/END lines) and send it to your Certification Authority for signing.

```
-----BEGIN CERTIFICATE REQUEST-----
MIIBWjCBxAIBADabMRkwFwYDVQQDBBJVFNL1M0Qi5pbmR1cm9wMIGfMA0GCSqG
SIb3DQEBAQAA4GNADCBiQKBgQCzEs8XTnY8be/t77eEDG7rTg747GQ30fOC4Rs
x-e9KfbErZgxIYqGT8u04AU0wU9LUPlkkq+8gI6w2bg3bw0kg/9hrnNL2rf1tGcn
3oShP05PiKmRNZnCC090b03tbr9kuHmlwPRQ7yT6k7xS3XBbSigqT4LQbjBT1tt
hDH3bQIDAQABAAwDQYIKoZIhvCNAQEF8AQDgYEAIm/GA2E1ZQbZaR6CZyIaw1lT
u65w450NFHmaCluHSyZ8keM8d1Ux14hkW7t5ygAD8KbxVKhRVaCgcQrAK2v8u1PF
TvN+bwJ+kQd59C1xa82e0oIW83buPq5+qlMDGF+MyJWGVFB8IC1c6+zFoc+BEZY
7tQ8y0J8od0aDhStDfQ=
-----END CERTIFICATE REQUEST-----
```

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

Figure 4-27: Microsoft Certificate Services Web Page



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.

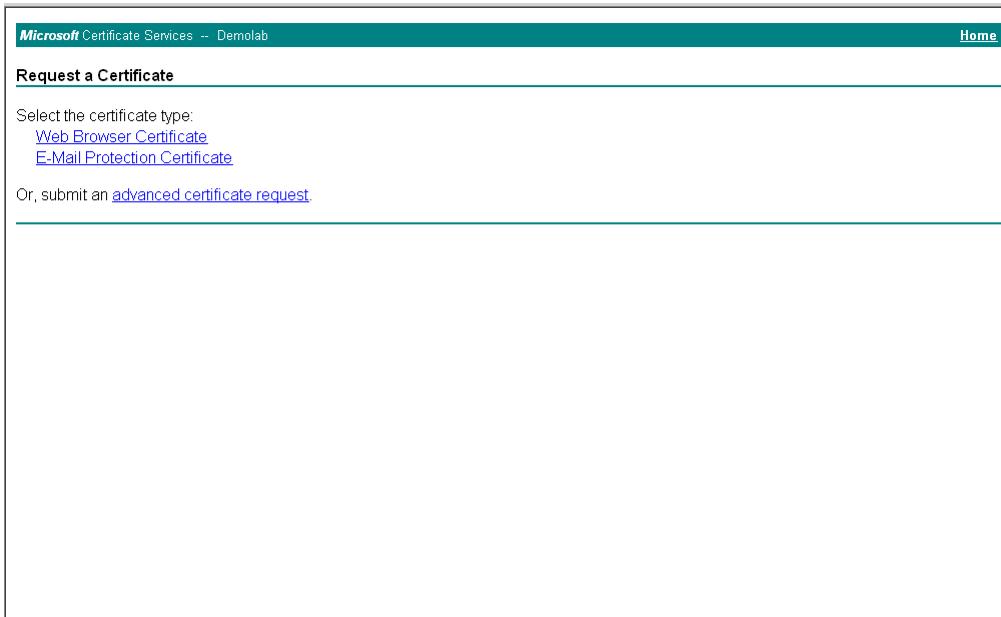
You can also use this Web site to download a certificate authority (CA) certificate, certificate chain, or certificate revocation list (CRL), or to view the status of a pending request.

For more information about Certificate Services, see [Certificate Services Documentation](#).

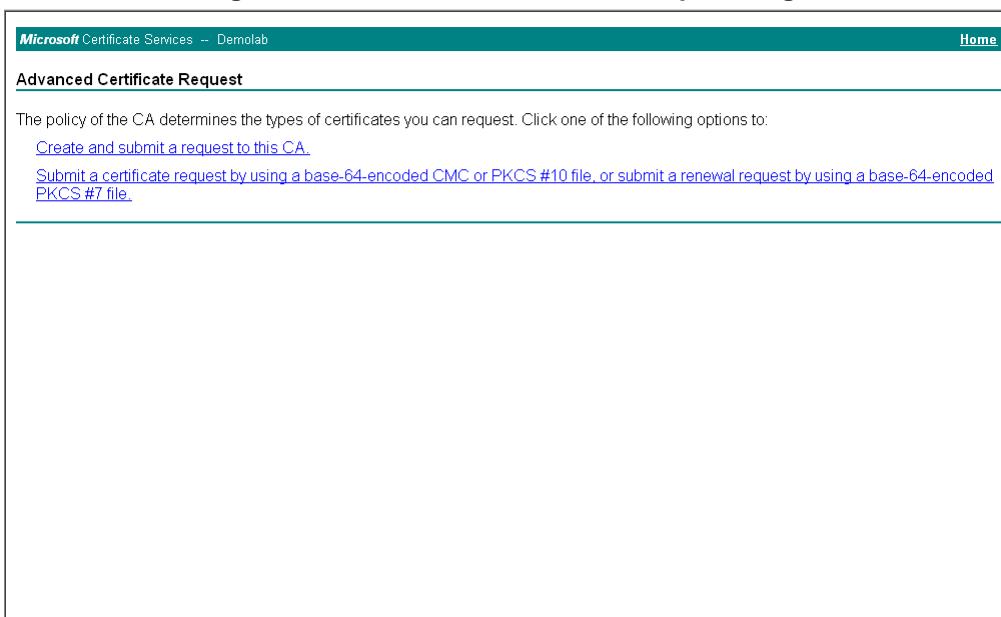
Select a task:

- [Request a certificate](#)
- [View the status of a pending certificate request](#)
- [Download a CA certificate, certificate chain, or CRL](#)

6. Click **Request a certificate**.

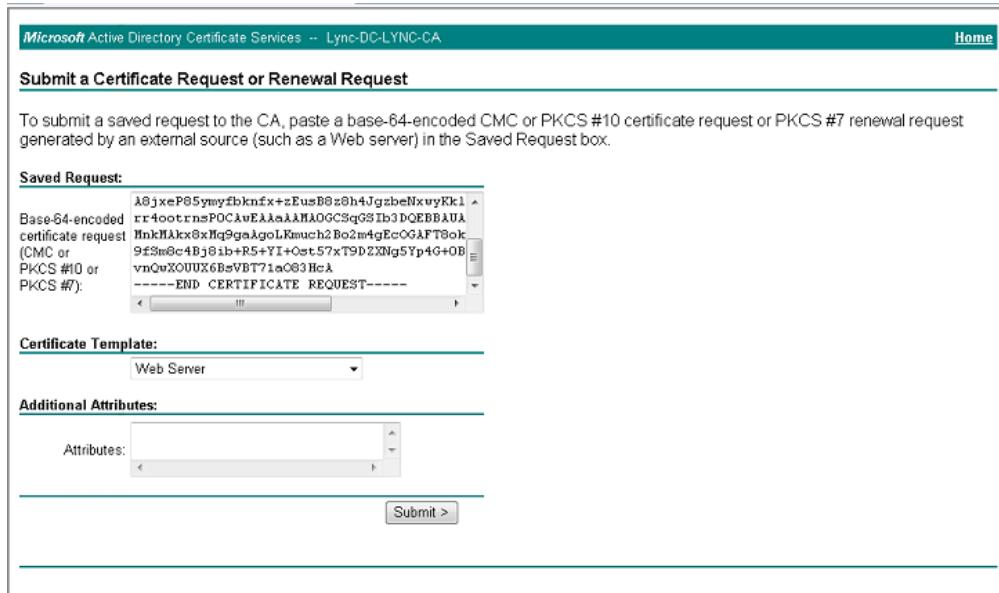
Figure 4-28: Request a Certificate Page

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

Figure 4-29: Advanced Certificate Request Page

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

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



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

Submit a Certificate Request or Renewal Request

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

Saved Request:

```

x8jxeP85ymyfbknfx+zEusB6z8h4JgzbeNxuyKk1
Base-64-encoded certificate request: rr4oootnsPOC4vxEAAkAAMAOGCSqGSIb3DQEBAU
(MMC or PKCS #10 or PKCS #7):
-----END CERTIFICATE REQUEST-----

```

Certificate Template:

Web Server

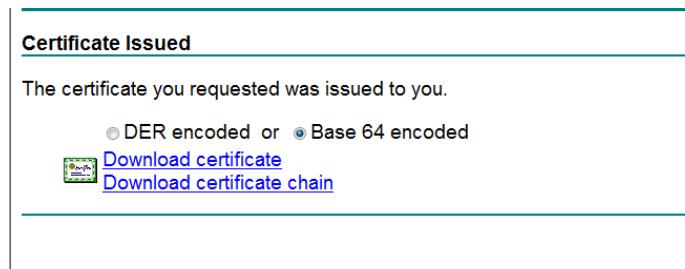
Additional Attributes:

Attributes:

Submit >

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

Figure 4-31: Certificate Issued Page



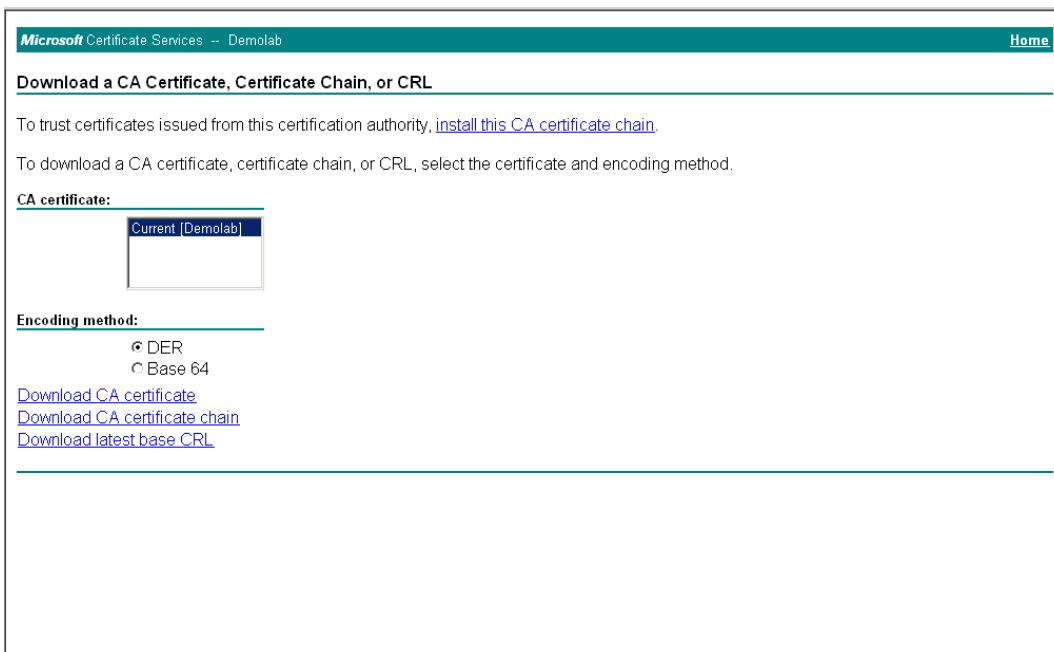
Certificate Issued

The certificate you requested was issued to you.

DER encoded or Base 64 encoded

[Download certificate](#) [Download certificate chain](#)

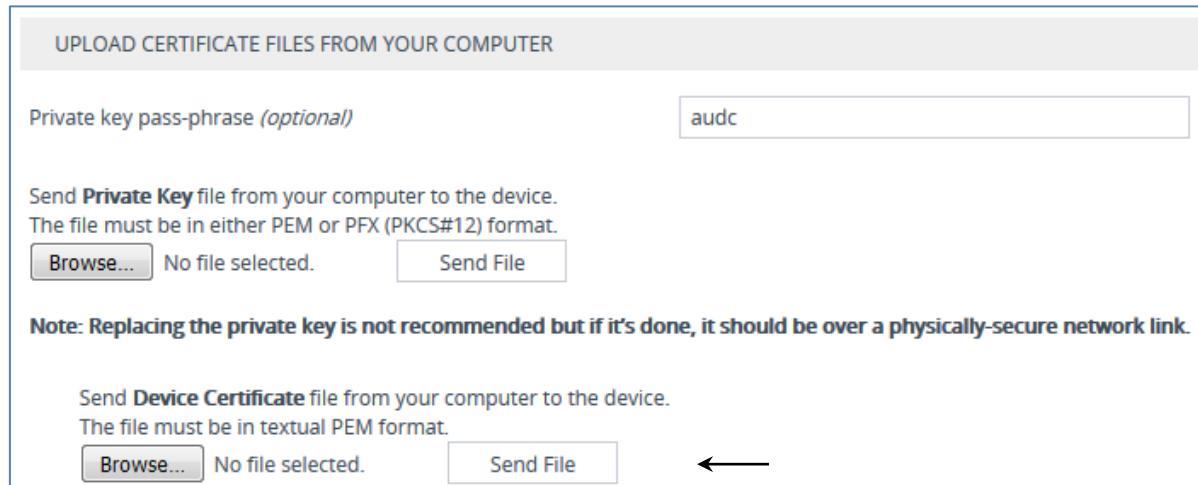
12. Select the **Base 64 encoded** option for encoding, and then click **Download certificate**.
13. Save the file as *gateway.cer* to a folder on your computer.
14. Click the **Home** button or navigate to the certificate server at <http://<Certificate Server>/CertSrv>.
15. Click **Download a CA certificate,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

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

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

- 19.** In the E-SBC's Web interface, return to the **TLS Contexts** page and do the following:
- In the TLS Contexts page, select the required TLS Context index row, and then click the **Change Certificate** link located below the table; the Context Certificates page appears.
 - 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 13, and then click **Send File** to upload the certificate to the E-SBC.

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



UPLOAD CERTIFICATE FILES FROM YOUR COMPUTER

Private key pass-phrase (*optional*)

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

No file selected.

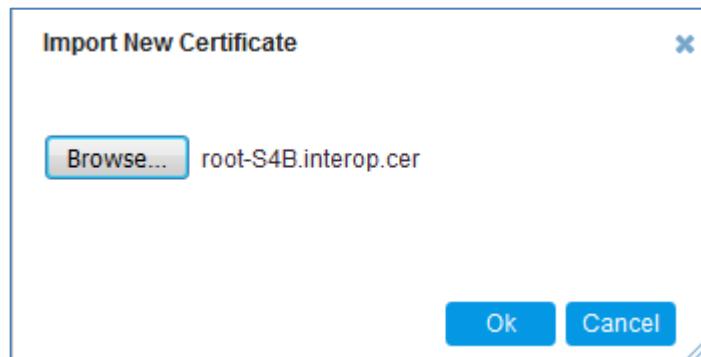
Note: Replacing the private key is not recommended but if it's done, it should be over a physically-secure network link.

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

No file selected. ←

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

Figure 4-34: Importing Root Certificate into Trusted Certificates Store



- Click **OK**; the certificate is loaded to the device and listed in the Trusted Certificates store.
- Reset the E-SBC with a burn to flash for your settings to take effect (see Section 4.17 on page 86).

4.9.4 Step 9d: Configure a Certificate for Operation with the BroadCloud SIP Trunk

This step describes how to load the BroadCloud Root Certificate as a Trusted Root Certificate. This certificate is used by the E-SBC to authenticate the connection with the BroadCloud SIP Trunk.

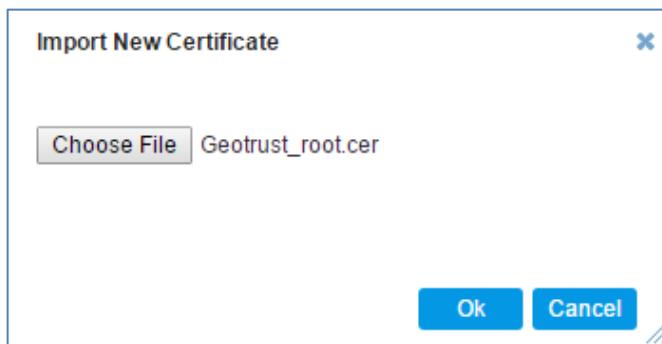
The procedure involves the following main steps:

- a. Obtaining a Trusted Root Certificate from the BroadCloud.
- b. Deploying the BroadCloud Root Certificate as Trusted Root Certificates on the E-SBC.

➤ **To load a certificate:**

1. Open the TLS Contexts page (**Setup** menu > **IP Network** tab > **Security** folder > **TLS Contexts**).
2. In the TLS Contexts page, select the required TLS Context index row (usually **default** index 0 will be used), and then click the **Trusted Root Certificates** link, located at the bottom of the TLS Contexts page; the Trusted Certificates page appears.
3. Click the **Import** button, and then select the certificate file to load.

Figure 4-35: Importing the BroadCloud Root Certificate into Trusted Certificates Store



4. Click **OK**; the certificate is loaded to the device and listed in the Trusted Certificates store.
5. Reset the E-SBC with a burn to flash for your settings to take effect (see Section 4.17 on page 86).

4.10 Step 10: Configure SRTP

This step describes how to configure media security. If you configure the Microsoft Mediation Server to use SRTP, you need to configure the E-SBC to operate in the same manner. Note that SRTP was enabled for Skype for Business Server 2015 when you configured an IP Profile for Skype for Business Server 2015 (see Section 4.6 on page 45).

➤ **To configure media security:**

1. Open the Media Security page (**Setup menu > Signaling & Media tab > Media folder > Media Security**).

Figure 4-36: Configuring SRTP

GENERAL		AUTHENTICATION & ENCRYPTION	
Media Security	→	Enable	Authentication On Transmitted RTP Packets Active
Media Security Behavior	Preferable	Encryption On Transmitted RTP Packets Active	
Offered SRTP Cipher Suites	All	Encryption On Transmitted RTCP Packets Active	
Aria Protocol Support	Disable	SRTP Tunneling Authentication for RTP Disable	
		SRTP Tunneling Authentication for RTCP Disable	
MASTER KEY IDENTIFIER		GATEWAY SETTINGS	
Master Key Identifier (MKI) Size	0	Enable Rekey After 181	Disable
Symmetric MKI	Disable		

2. From the 'Media Security' drop-down list, select **Enable** to enable SRTP.
3. Click **Apply**.
4. Reset the E-SBC with a burn to flash for your settings to take effect (see Section 4.17 on page 86).

4.11 Step 11: Configure Maximum IP Media Channels

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



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

- **To configure the maximum number of IP media channels:**
1. Open the Media Settings page (**Setup** menu > **Signaling & Media** tab > **Media** folder > **Media Settings**).

Figure 4-37: Configuring Number of Media Channels

The screenshot shows the 'Media Settings' configuration page. Under the 'GENERAL' tab, there are several settings:

- NAT Traversal:** Disable NAT (dropdown menu)
- Enable Continuity Tones:** Disable (dropdown menu with a lightning bolt icon)
- Inbound Media Latch Mode:** Dynamic (dropdown menu)
- Number of Media Channels:** 100 (radio button selected, with a lightning bolt icon)
- Enforce Media Order:** Disable (dropdown menu)
- SDP Session Owner:** AudiocodesGW (dropdown menu)

2. In the 'Number of Media Channels' field, enter the number of media channels according to your environments transcoding calls (e.g., **100**).
3. Click **Apply**.
4. Reset the E-SBC with a burn to flash for your settings to take effect (see Section 4.17 on page 86).

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

This step describes how to configure IP-to-IP call routing rules. These rules define the routes for forwarding SIP messages (e.g., INVITE) received from one IP entity to another. The E-SBC selects the rule whose configured input characteristics (e.g., IP Group) match those of the incoming SIP message. If the input characteristics do not match the first rule in the table, they are compared to the second rule, and so on, until a matching rule is located. If no rule is matched, the message is rejected. The routing rules use the configured IP Groups (as configured in Section 4.8 on page 44,) to denote the source and destination of the call.

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 BroadCloud SIP Trunk (DMZ):

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

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

1. Open the IP-to-IP Routing table (**Setup** menu > **Signaling & Media** tab > **SBC** folder > **Routing > IP-to-IP Routing**).
2. Configure a rule to terminate SIP OPTIONS messages received from the both LAN and DMZ:
 - a. Click **New**, and then configure the parameters as follows:

Parameter	Value
Index	0
Name	Terminate OPTIONS (arbitrary descriptive name)
Source IP Group	Any
Request Type	OPTIONS
Destination Type	Dest Address
Destination Address	internal

Figure 4-38: Configuring IP-to-IP Routing Rule for Terminating SIP OPTIONS

IP-to-IP Routing [Terminate OPTIONS]

GENERAL		ACTION	
Routing Policy #0 [Default_SBCRoutingPolicy]			
Index	0	Destination Type	• Dest Address
Name	• Terminate OPTIONS	Destination IP Group	--
Alternative Route Options	Route Row	Destination SIP Interface	--
MATCH		Destination Address	• internal
Source IP Group	Any	Destination Port	0
Request Type	• OPTIONS	Destination Transport Type	
Source Username Prefix	*	Call Setup Rules Set ID	-1
Source Host	*	Group Policy	Sequential
Source Tags		Cost Group	--
<input type="button" value="Cancel"/> <input type="button" value="APPLY"/>			

- b. Click **Apply**.

3. Configure a rule to route calls from Skype for Business Server 2015 to BroadCloud SIP Trunk:

- a. Click **New**, and then configure the parameters as follows:

Parameter	Value
Index	1
Route Name	S4B to BroadCloud (arbitrary descriptive name)
Source IP Group	S4B
Destination Type	IP Group
Destination IP Group	BroadCloud

Figure 4-39: Configuring IP-to-IP Routing Rule for S4B to BroadCloud

IP-to-IP Routing [S4B to ITSP]

Routing Policy #0 [Default_SBCRoutingPolicy] ▾

GENERAL		ACTION	
Index	1	Destination Type	IP Group ▾
Name	S4B to ITSP	Destination IP Group	#2 [BroadCloud] ▾ View
Alternative Route Options	Route Row	Destination SIP Interface	- ▾ View
MATCH		Destination Address	
Source IP Group	#1 [S4B] ▾ View	Destination Port	0
Request Type	All	Destination Transport Type	▪
Source Username Prefix	*	IP Group Set	- ▾ View
Source Host	*	Call Setup Rules Set ID	-1
Source Tag		Group Policy	Sequential ▾
<input type="button" value="Cancel"/> <input type="button" value="APPLY"/>			

- b. Click **Apply**.

4. Configure rule to route calls from BroadCloud SIP Trunk to Skype for Business Server 2015:

- a. Click **New**, and then configure the parameters as follows:

Parameter	Value
Index	2
Route Name	BroadCloud to S4B (arbitrary descriptive name)
Source IP Group	BroadCloud
Destination Type	IP Group
Destination IP Group	S4B

Figure 4-40: Configuring IP-to-IP Routing Rule for BroadCloud to S4B

The screenshot shows the 'IP-to-IP Routing [ITSP to S4B]' configuration dialog. The 'GENERAL' tab is selected, showing an Index of 2 and a Name of 'ITSP to S4B'. The 'ACTION' tab shows the Destination Type as 'IP Group' and the Destination IP Group as '#1 [S4B]'. The 'MATCH' tab shows the Source IP Group as '#2 [BroadCloud]'. Other match criteria like Request Type (All), Source Host (*), and Source Tag are also visible. At the bottom are 'Cancel' and 'APPLY' buttons.

- b. Click **Apply**.

The configured routing rules are shown in the figure below:

Figure 4-41: Configured IP-to-IP Routing Rules in IP-to-IP Routing Table

IP-to-IP Routing (3) .

+ New Edit Insert ↑ ↓ trash Page 1 of 1 Show 10 records per page Search

INDEX	NAME	ROUTING POLICY	ALTERNATIVE ROUTE OPTIONS	SOURCE IP GROUP	REQUEST TYPE	SOURCE USERNAME PREFIX	DESTINATNIC USERNAME PREFIX	DESTINATNIC TYPE	DESTINATNIC IP GROUP	DESTINATNIC SIP INTERFACE	DESTINATNIC ADDRESS
0	Terminate O	Default_SBC	Route Row	Any	OPTIONS	*	*	Dest Addres	--	--	internal
1	S4B to ITSP	Default_SBC	Route Row	S4B	All	*	*	IP Group	BroadCloud	--	
2	ITSP to S4B	Default_SBC	Route Row	BroadCloud	All	*	*	IP Group	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 describes how to configure IP-to-IP manipulation rules. These rules manipulate the SIP Request-URI user part (source or destination number). The manipulation rules use the configured IP Groups (as configured in Section 4.8 on page 44) to denote the source and destination of the call.



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

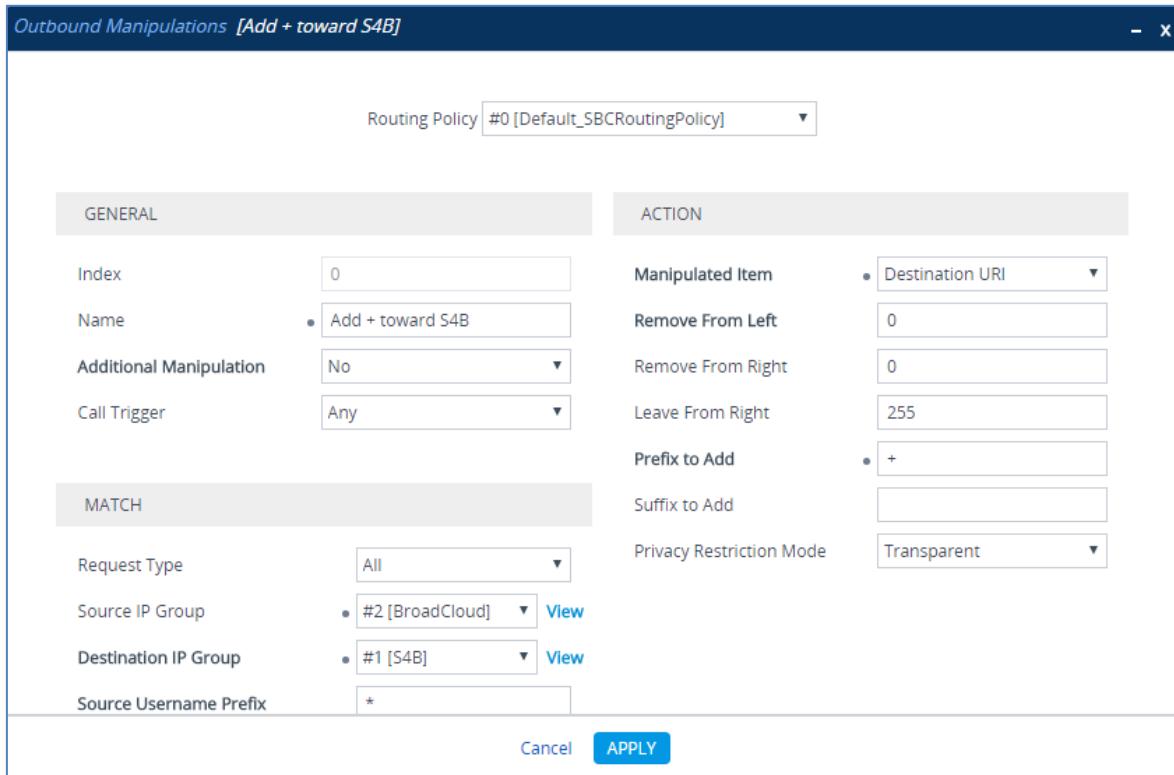
For example, for this interoperability test topology, a manipulation is configured to add the "+" (plus sign) to the destination number for calls from the BroadCloud SIP Trunk IP Group to the Skype for Business Server 2015 IP Group for any destination username prefix.

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

1. Open the Outbound Manipulations table (**Setup** menu > **Signaling & Media** tab > **SBC** folder > **Manipulation** > **Outbound Manipulations**).
2. Click **New**, and then configure the parameters as follows:

Parameter	Value
Index	0
Name	Add + toward S4B
Source IP Group	BroadCloud
Destination IP Group	S4B
Destination Username Prefix	* (asterisk sign)
Manipulated Item	Destination URI
Prefix to Add	+ (plus sign)

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



3. Click **Apply**.

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 BroadCloud SIP Trunk IP Group:

Figure 4-43: Example of Configured IP-to-IP Outbound Manipulation Rules

Outbound Manipulations (3)													
+ New Edit Insert <input type="button" value=""/> Page <input type="text" value="1"/> of 1 <input type="button" value=""/> Show <input type="text" value="10"/> records per page <input type="button" value=""/>													
INDEX	NAME	ROUTING POLICY	ADDITION MANIPUL	SOURCE IP GROUP	DESTINAT IP GROUP	SOURCE USERNAM PREFIX	DESTINAT USERNAN PREFIX	MANIPUL ITEM	REMOVE FROM LEFT	REMOVE FROM RIGHT	LEAVE FROM RIGHT	PREFIX TO ADD	SUFFIX TO ADD
0	Add + tow	Default_SE	No	BroadCloud	S4B	*	*	Destination	0	0	255	+	
1	Change +	Default_SE	No	S4B	BroadCloud	*	+	Destination	1	0	255	011	
2	Remove +	Default_SE	No	S4B	BroadCloud	+8	*	Source UF	1	0	255		

Rule Index	Description
1	Calls from BroadCloud IP Group to S4B IP Group with any <u>destination</u> number (*), add "+" to the prefix of the destination number.
2	Calls from S4B IP Group to BroadCloud IP Group with the prefix <u>destination</u> number "+", remove "+" from this prefix and add "011" to the prefix.
3	Calls from S4B IP Group to BroadCloud IP Group with <u>source</u> number prefix "+8", remove the "+" from this prefix.

4.14 Step 14: Configure Message Manipulation Rules

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

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

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

1. Open the Message Manipulations page (**Setup** menu > **Signaling & Media** tab > **Message Manipulation** folder > **Message Manipulations**).
2. Configure a new manipulation rule (Manipulation Set 4) for BroadCloud SIP Trunk. This rule applies to messages sent to the BroadCloud SIP Trunk IP Group in a Call Forward scenario. This adds a SIP Diversion Header with the value from the SIP From Header only if the SIP History-Info Header exists.

Parameter	Value
Index	0
Name	Call Forward
Manipulation Set ID	4
Message Type	invite
Condition	header.history-info exists
Action Subject	header.diversion
Action Type	Add
Action Value	header.from

Figure 4-44: Configuring SIP Message Manipulation Rule 0 (for BroadCloud SIP Trunk)

GENERAL		ACTION	
Index	0	Action Subject	header.diversion
Name	Call Forward	Action Type	Add
Manipulation Set ID	4	Action Value	header.from
Row Role	Use Current Condition		
MATCH			
Message Type	invite		
Condition	header.history-info exists		
<input type="button" value="Cancel"/> <input type="button" value="APPLY"/>			

3. Configure another manipulation rule (Manipulation Set 4) for BroadCloud SIP Trunk. This rule applies to messages sent to the BroadCloud SIP Trunk IP Group in a Call Forwarding scenario. This replaces the user part of the SIP From Header with the value from the SIP History-Info Header.

Parameter	Value
Index	1
Name	Call Forward
Manipulation Set ID	4
Message Type	invite
Condition	header.history-info.0 regex (<sip:(.).*)(@)(.*)
Action Subject	header.from.url.user
Action Type	Modify
Action Value	\$3

Figure 4-45: Configuring SIP Message Manipulation Rule 1 (for BroadCloud SIP Trunk)

Message Manipulations [Call Forward]

GENERAL
ACTION

Index	<input type="text" value="1"/>	Action Subject	<input type="text" value="header.diversion.url.user"/>
Name	<input checked="" type="radio"/> Call Forward	Action Type	<input checked="" type="radio"/> Modify
Manipulation Set ID	<input checked="" type="radio"/> 4	Action Value	<input type="text" value="\$3"/>
Row Role	Use Current Condition		

MATCH

Message Type	<input type="text" value="invite"/>	
Condition	<input type="text" value="header.history-info.0 regex (<sip:(.).*)(@)(.*)"/>	

Cancel
APPLY

4. If the manipulation rule Index 1 (above) is executed, then the following rule is also executed to remove the SIP History-Info Header.

Parameter	Value
Index	2
Name	Call Forward
Manipulation Set ID	4
Row Role	Use Previous Condition
Message Type	
Condition	
Action Subject	header.history-info
Action Type	Remove
Action Value	

Figure 4-46: Configuring SIP Message Manipulation Rule 2 (for BroadCloud SIP Trunk)

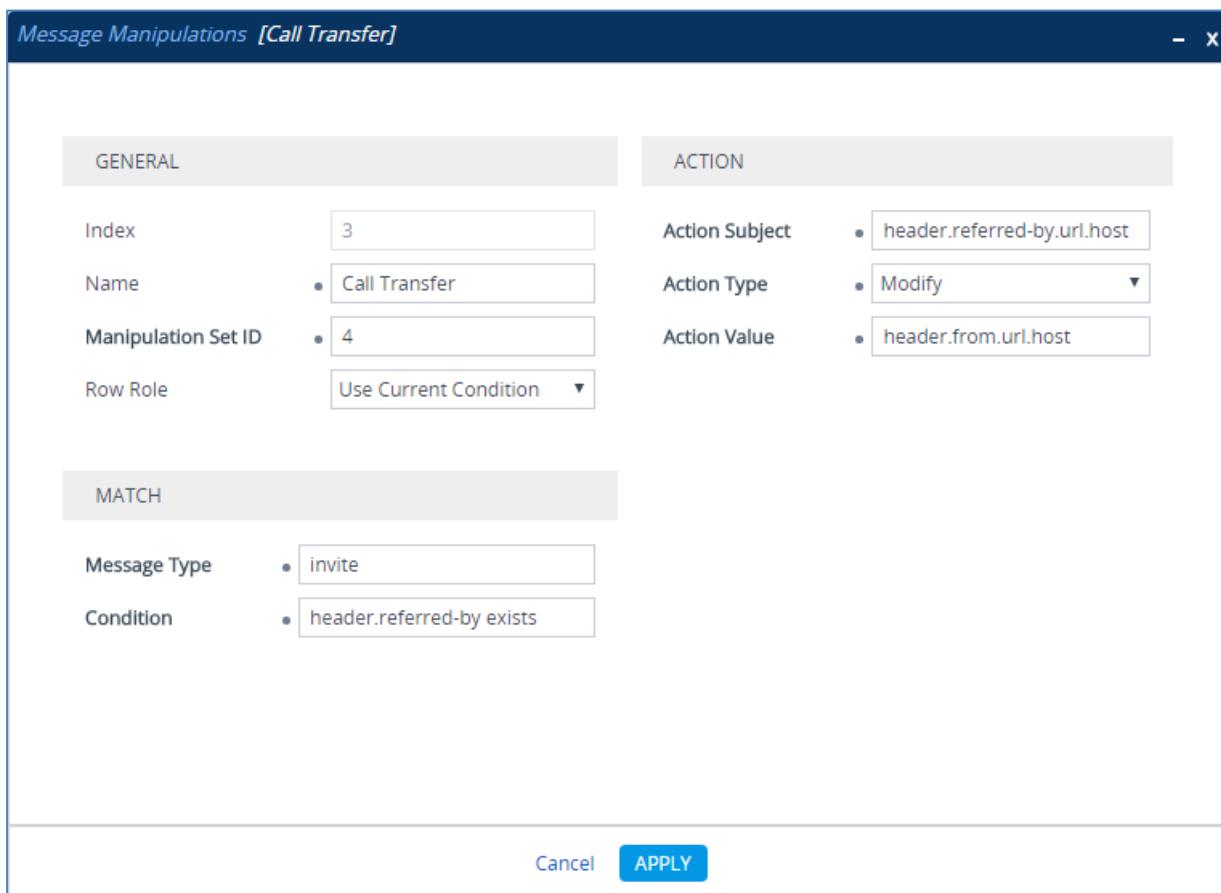
Message Manipulations [Call Forward]			
GENERAL		ACTION	
Index	2	Action Subject	header.history-info
Name	Call Forward	Action Type	Remove
Manipulation Set ID	4	Action Value	
Row Role	Use Previous Condition		
MATCH			
Message Type			
Condition			
<input type="button" value="Cancel"/> <input type="button" value="APPLY"/>			

5. Configure a manipulation rule (Manipulation Set 4) for BroadCloud SIP Trunk that applies to messages sent to the BroadCloud SIP Trunk IP Group in a Call Transfer scenario. This replaces the host part of the SIP Referred-By Header with the value from the SIP From Header.

Parameter	Value
Index	3
Name	Call Transfer
Manipulation Set ID	4
Message Type	invite
Condition	header.referred-by exists
Action Subject	header.referred-by.url.host
Action Type	Modify
Action Value	header.from.url.host

Figure 4-47: Configuring SIP Message Manipulation Rule 3 (for BroadCloud SIP Trunk)

Message Manipulations [Call Transfer]



The screenshot shows the 'Message Manipulations [Call Transfer]' configuration window. It has three tabs: GENERAL, ACTION, and MATCH. The GENERAL tab contains fields for Index (3), Name (Call Transfer), Manipulation Set ID (4), and Row Role (Use Current Condition). The ACTION tab contains fields for Action Subject (header.referred-by.url.host), Action Type (Modify), and Action Value (header.from.url.host). The MATCH tab contains fields for Message Type (invite) and Condition (header.referred-by exists). At the bottom are 'Cancel' and 'APPLY' buttons.

GENERAL		ACTION	
Index	3	Action Subject	• header.referred-by.url.host
Name	• Call Transfer	Action Type	• Modify ▾
Manipulation Set ID	• 4	Action Value	• header.from.url.host
Row Role	Use Current Condition ▾		

MATCH	
Message Type	• invite
Condition	• header.referred-by exists

Cancel	APPLY
--------	-------

6. Configure a manipulation rule (Manipulation Set 4) for BroadCloud SIP Trunk that applies to messages sent to the BroadCloud SIP Trunk IP Group in a Call Transfer scenario. This rule removes '+' prefix from the user part of the SIP Referred-By Header.

Parameter	Value
Index	4
Name	Call Transfer
Manipulation Set ID	4
Message Type	invite
Condition	header.referred-by exists
Action Subject	header.referred-by.url.user
Action Type	Remove Prefix
Action Value	'+'

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

Message Manipulations [Call Transfer]			
GENERAL		ACTION	
Index	4	Action Subject	header.referred-by.url.user
Name	• Call Transfer	Action Type	• Remove Prefix
Manipulation Set ID	• 4	Action Value	• '+'
Row Role	Use Current Condition		
MATCH			
Message Type	• invite		
Condition	• header.referred-by exists		
<input type="button" value="Cancel"/> <input type="button" value="APPLY"/>			

7. Configure a manipulation rule (Manipulation Set 4) for BroadCloud SIP Trunk that applies to messages sent to the BroadCloud SIP Trunk IP Group in a Call Transfer scenario. This replaces the user part of the SIP From Header with the value from the SIP Referred-By Header.

Parameter	Value
Index	5
Name	Call Transfer
Manipulation Set ID	4
Message Type	invite
Condition	header.referred-by exists
Action Subject	header.from.url.user
Action Type	Modify
Action Value	header.referred-by.url.user

Figure 4-49: Configuring SIP Message Manipulation Rule 5 (for BroadCloud SIP Trunk)

Message Manipulations [Call Transfer]

GENERAL
ACTION

Index	5	Action Subject	<input type="radio"/> header.from.url.user
Name	<input type="radio"/> Call Transfer	Action Type	<input type="radio"/> Modify
Manipulation Set ID	<input type="radio"/> 4	Action Value	<input type="radio"/> header.referred-by.url.user
Row Role	Use Current Condition		

MATCH

Message Type	<input type="radio"/> invite
Condition	<input type="radio"/> header.referred-by exists

8. Configure another manipulation rule (Manipulation Set 4) for BroadCloud SIP Trunk. This rule is applied to response messages sent to the BroadCloud SIP Trunk IP Group for Rejected Calls initiated by the Skype for Business Server 2015 IP Group. This replaces the method type '503' with the value '480', because BroadCloud SIP Trunk not recognizes '503' method type.

Parameter	Value
Index	6
Name	Reject Responses
Manipulation Set ID	4
Message Type	any.response
Condition	header.request-uri.methodtype=='503'
Action Subject	header.request-uri.methodtype
Action Type	Modify
Action Value	'480'

Figure 4-50: Configuring SIP Message Manipulation Rule 6 (for BroadCloud SIP Trunk)

GENERAL				ACTION	
Index	6	Action Subject	<input checked="" type="checkbox"/> header.request-uri.methodtype		
Name	Reject Responses	Action Type	<input checked="" type="checkbox"/> Modify		
Manipulation Set ID	4	Action Value	<input checked="" type="checkbox"/> '480'		
Row Role	Use Current Condition				
MATCH					
Message Type	any.response				
Condition	<input checked="" type="checkbox"/> header.request-uri.methodtype=='503'				
<input type="button" value="Cancel"/> <input type="button" value="APPLY"/>					

Figure 4-51: Example of Configured SIP Message Manipulation Rules

Message Manipulations (7)								
INDEX		NAME	MANIPULATION SET ID	MESSAGE TYPE	CONDITION	ACTION SUBJECT	ACTION TYPE	ACTION VALUE
0	Call Forward	4	invite	header.history-info.e	header.diversion	Add	header.from	Use Current Condition
1	Call Forward	4	invite	header.history-info.C	header.diversion.url	Modify	\$3	Use Current Condition
2	Call Forward	4			header.history-info	Remove		Use Previous Condition
3	Call Transfer	4	invite	header.referred-by.e	header.referred-by.u	Modify	header.from.url.host	Use Current Condition
4	Call Transfer	4	invite	header.referred-by.e	header.referred-by.u	Remove Prefix	'+'	Use Current Condition
5	Call Transfer	4	invite	header.referred-by.e	header.from.url.user	Modify	header.referred-by.u	Use Current Condition
6	Reject Responses	4	any.response	header.request-uri.n	header.request-uri.n	Modify	'480'	Use Current Condition

The table displayed below includes SIP message manipulation rules which are grouped together under Manipulation Set ID 4 and which are executed for messages sent to and from the BroadCloud SIP Trunk IP Group. These rules are specifically required to enable proper interworking between BroadCloud SIP Trunk and Skype for Business Server 2015. Refer to the *User's Manual* for further details concerning the full capabilities of header manipulation.

Rule Index	Rule Description	Reason for Introducing Rule
0	This rule adds a SIP Diversion Header with the value from the SIP From Header only if the SIP History-Info Header exists.	For Call Forward scenarios, BroadCloud SIP Trunk needs the SIP Diversion Header. In order to do this, a SIP Diversion Header is added with the value from SIP From Header and the User part of the SIP Diversion Header is replaced with the value from History-Info Header.
1	This rule replaces the user part of the SIP From Header with the value from the SIP History-Info Header.	
2	If the manipulation rule Index 1 (above) is executed, then the following rule is also executed. It removes the SIP History-Info Header.	
3	This rule replaces the host part of the SIP Referred-By Header with the value from the SIP From Header.	For Call Transfer initiated by Skype for Business Server 2015, BroadCloud SIP Trunk needs to replace the host part of the SIP Referred-By Header with the value from the SIP From Header and user part of the From Header with the value from Referred-By Header.
4	This rule removes '+' prefix from the user part of the SIP Referred-By Header.	
5	This rule replaces the user part of the SIP From Header with the value from the SIP Referred-By Header.	
6	This rule replaces the method type '503' with the value '480', because BroadCloud SIP Trunk does not recognize the '503' method type.	BroadCloud SIP Trunk does not recognize the '503' method type and continues to send an INVITE message i.e. it tries to setup another call.

9. Assign Manipulation Set ID 4 to the BroadCloud SIP trunk IP Group:
- Open the IP Groups table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **IP Groups**).
 - Select the row of the BroadCloud SIP trunk IP Group, and then click **Edit**.
 - Set the 'Outbound Message Manipulation Set' field to 4.

Figure 4-52: Assigning Manipulation Set 4 to the BroadCloud SIP Trunk IP Group

The screenshot shows the 'IP Groups [BroadCloud]' configuration interface. The 'GENERAL' tab is selected. In the 'MESSAGE MANIPULATION' section, the 'Outbound Message Manipulation Set' dropdown is set to 4. The 'APPLY' button is visible at the bottom right.

Setting	Value
SRD	#0 [DefaultSRD]
Index	2
Name	BroadCloud
Topology Location	Up
Type	Server
Proxy Set	#2 [BroadCloud]
IP Profile	#2 [BroadCloud]
Media Realm	#1 [MRWan]
Contact User	
SIP Group Name	interop.adpt-tech.com
Created By Routing Server	No
Inbound Message Manipulation Set	-1
Outbound Message Manipulation Set	4
Message Manipulation User-Defined String 1	
Message Manipulation User-Defined String 2	

- Click **Apply**.

4.15 Step 15: Configure Registration Accounts

This step describes how to configure SIP registration accounts. This is required so that the E-SBC can register with the BroadCloud SIP Trunk on behalf of Skype for Business Server 2015. The BroadCloud SIP Trunk requires registration and authentication to provide service.

In the interoperability test topology, the Served IP Group is Skype for Business Server 2015 IP Group and the Serving IP Group is BroadCloud SIP Trunk IP Group.

➤ **To configure a registration account:**

1. Open the Accounts table (**Setup** menu > **Signaling & Media** tab > **SIP Definitions** folder > **Accounts**).
2. Click **New**.
3. Configure the account according to the provided information from , for example:

Parameter	Value
Served IP Group	S4B
Application Type	SBC
Serving IP Group	BroadCloud
Host Name	As provided by the SIP Trunk provider
Register	Regular
Contact User	1234567890 (trunk main line)
Username	As provided by the SIP Trunk provider
Password	As provided by the SIP Trunk provider

Figure 4-53: Configuring a SIP Registration Account

The screenshot shows a configuration interface for an SIP registration account. At the top, there is a dropdown menu labeled "Served IP Group" set to "#1 [S4B]". Below this, the interface is divided into two main sections: "GENERAL" and "CREDENTIALS".

GENERAL Section:

- Index: 0
- Served Trunk Group: -1
- Application Type: SBC
- Serving IP Group: #2 [BroadCloud] (with a "View" link)
- Host Name: interop.adpt-tech.com
- Register: Regular
- Contact User: 1234567890

CREDENTIALS Section:

- User Name: 1234567890
- Password: (empty field)

At the bottom of the interface are two buttons: "Cancel" and "APPLY".

4. Click **Apply**.

4.16 Step 16: Miscellaneous Configuration

This section describes miscellaneous E-SBC configuration.

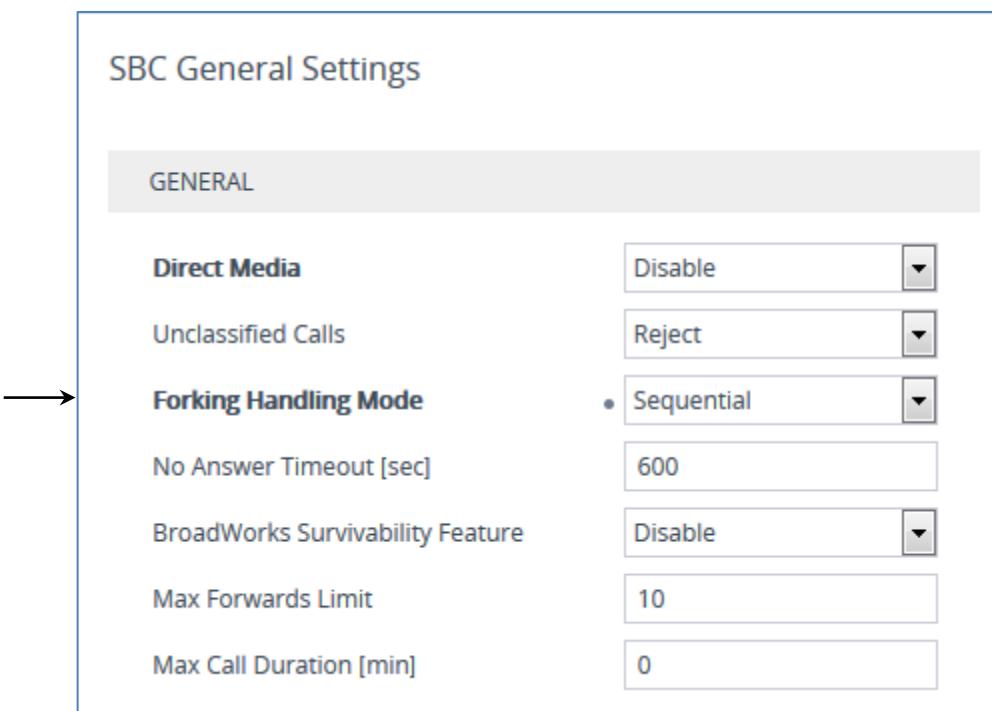
4.16.1 Step 16a: Configure Call Forking Mode

This step describes how to configure the E-SBC's handling of SIP 18x responses received for call forking of INVITE messages. For the interoperability test topology, if a SIP 18x response with SDP is received, the E-SBC opens a voice stream according to the received SDP. The E-SBC re-opens the stream according to subsequently received 18x responses with SDP or plays a ringback tone if 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 SBC General Settings page (**Setup** menu > **Signaling & Media** tab > **SBC** folder > **SBC General Settings**).
2. From the 'SBC Forking Handling Mode' drop-down list, select **Sequential**.

Figure 4-54: Configuring Forking Mode



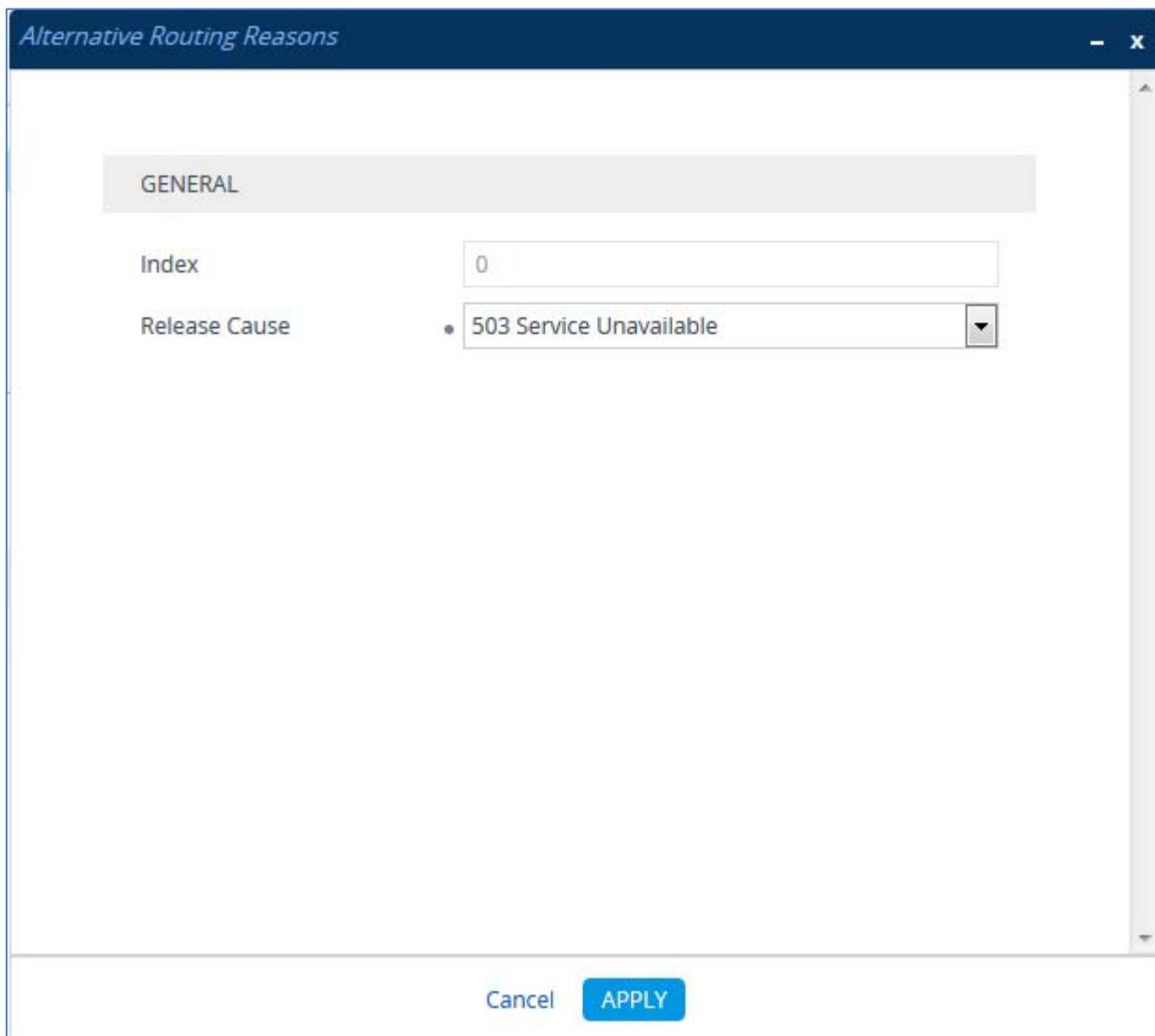
3. Click **Apply**.

4.16.2 Step 16b: Configure SBC Alternative Routing Reasons

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

- **To configure SIP reason codes for alternative IP routing:**
 1. Open the Alternative Routing Reasons table (**Setup** menu > **Signaling & Media** tab > **SBC** folder > **Routing** > **Alternative Reasons**).
 2. Click **New**.
 3. From the 'Release Cause' drop-down list, select **503 Service Unavailable**.

Figure 4-55: SBC Alternative Routing Reasons Table



4. Click **Apply**.

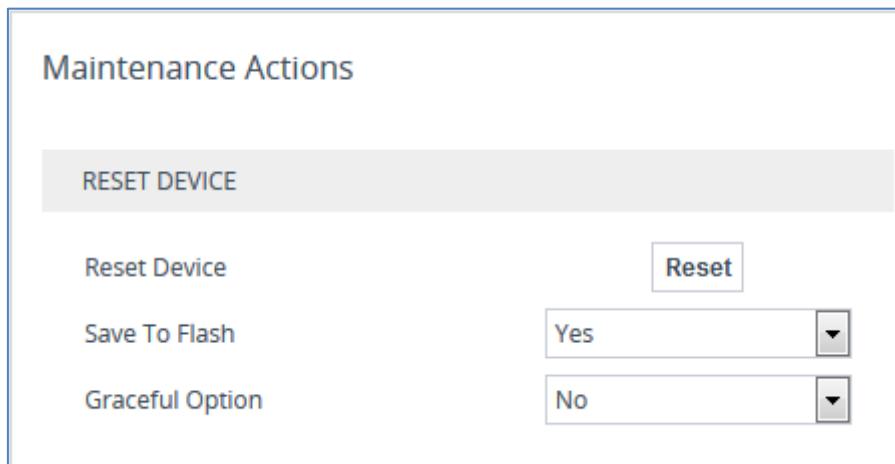
4.17 Step 17: Reset the E-SBC

After you have completed the configuration of the E-SBC described in this chapter, save ("burn") the configuration to the E-SBC's flash memory with a reset for the settings to take effect.

➤ **To reset the device through Web interface:**

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

Figure 4-56: Resetting the E-SBC



2. Ensure that the ' Save To Flash' field is set to **Yes** (default).
3. Click the **Reset** button; a confirmation message box appears, requesting you to confirm.
4. Click **OK** to confirm device reset.

A AudioCodes INI File

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



Note: To load or save an *ini* file, use the Configuration File page (**Setup** menu > **Administration** tab > **Maintenance** folder > **Configuration File**).

```

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


;Board: Mediant 800
;HW Board Type: 69  FK Board Type: 72
;Serial Number: 2265355
;Slot Number: 1
;Software Version: 7.20A.002
;DSP Software Version: 5014AE3_R => 720.25
;Board IP Address: 10.15.77.77
;Board Subnet Mask: 255.255.0.0
;Board Default Gateway: 10.15.0.1
;Ram size: 512M  Flash size: 64M  Core speed: 300Mhz
;Num of DSP Cores: 3  Num DSP Channels: 30
;Num of physical LAN ports: 12
;Profile: NONE
;;;;Key features:;Board Type: Mediant 800 ;PSTN FALLBACK Supported
;BRITrunks=4 ;E1Trunks=1 ;T1Trunks=1 ;FXSPorts=4 ;FXOPorts=0 ;Channel
Type: DspCh=30 IPMediaDspCh=30 ;HA ;DATA features: ;QOE features:
VoiceQualityMonitoring MediaEnhancement ;DSP Voice features: RTCP-XR
;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 OPUS_NB OPUS_WB ;Security: IPSEC MediaEncryption
StrongEncryption EncryptControlProtocol ;IP Media: Conf VXML CALEA
TrunkTesting ;Control Protocols: MGCP SIP SASurvivability SBC=250 MSFT
FEU=100 TestCall=100 ;Default features:;Coders: G711 G726;

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


[SYSTEM Params]

SyslogServerIP = 10.15.77.100
EnableSyslog = 1
;NTPServerIP_abs is hidden but has non-default value
NTPServerUTCOffset = 7200
;VpFileLastUpdateTime is hidden but has non-default value
TR069ACSPASSWORD = '$1$gQ=='
```

```
TR069CONNECTIONREQUESTPASSWORD = '$1$gQ=='
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

[WEB Params]

LogoWidth = '145'
UseProductName = 1
;HTTPSPkeyFileName is hidden but has non-default value

[SIP Params]

MEDIACHANNELS = 200
SIPDESTINATIONPORT = 8934
GWDEBUGLEVEL = 5
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_4_1", 1, 4, "User Port #0", "GROUP_1",
"Active";
PhysicalPortsTable 1 = "GE_4_2", 1, 4, "User Port #1", "GROUP_1",
"Redundant";
PhysicalPortsTable 2 = "GE_4_3", 1, 4, "User Port #2", "GROUP_2",
"Active";
PhysicalPortsTable 3 = "GE_4_4", 1, 4, "User Port #3", "GROUP_2",
"Redundant";
PhysicalPortsTable 4 = "FE_5_1", 0, 4, "User Port #4", "None", " ";
PhysicalPortsTable 5 = "FE_5_2", 0, 4, "User Port #5", "None", " ";
PhysicalPortsTable 6 = "FE_5_3", 0, 4, "User Port #6", "None", " ";
PhysicalPortsTable 7 = "FE_5_4", 0, 4, "User Port #7", "None", " ";
PhysicalPortsTable 8 = "FE_5_5", 0, 4, "User Port #8", "None", " ";
PhysicalPortsTable 9 = "FE_5_6", 0, 4, "User Port #9", "None", " ";
PhysicalPortsTable 10 = "FE_5_7", 0, 4, "User Port #10", "None", " ";
PhysicalPortsTable 11 = "FE_5_8", 0, 4, "User Port #11", "None", " ";

[ \PhysicalPortsTable ]

[ EtherGroupTable ]

FORMAT EtherGroupTable_Index = EtherGroupTable_Group,
EtherGroupTable_Mode, EtherGroupTable_Member1, EtherGroupTable_Member2;
EtherGroupTable 0 = "GROUP_1", 2, "GE_4_1", "GE_4_2";
EtherGroupTable 1 = "GROUP_2", 2, "GE_4_3", "GE_4_4";
EtherGroupTable 2 = "GROUP_3", 0, "", "";
EtherGroupTable 3 = "GROUP_4", 0, "", "";
EtherGroupTable 4 = "GROUP_5", 0, "", "";
EtherGroupTable 5 = "GROUP_6", 0, "", "";
EtherGroupTable 6 = "GROUP_7", 0, "", "";
EtherGroupTable 7 = "GROUP_8", 0, "", "";

```

```

EtherGroupTable 8 = "GROUP_9", 0, "", "";
EtherGroupTable 9 = "GROUP_10", 0, "", "";
EtherGroupTable 10 = "GROUP_11", 0, "", "";
EtherGroupTable 11 = "GROUP_12", 0, "", "";

[ \EtherGroupTable ]

[ DeviceTable ]

FORMAT DeviceTable_Index = DeviceTable_VlanID,
DeviceTable_UnderlyingInterface, DeviceTable_DeviceName,
DeviceTable_Tagging, DeviceTable_MTU;
DeviceTable 0 = 1, "GROUP_1", "vlan 1", 0, 1500;
DeviceTable 1 = 2, "GROUP_2", "vlan 2", 0, 1500;

[ \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, 10.15.77.77, 16, 10.15.0.1, "Voice",
10.15.27.1, , "vlan 1";
InterfaceTable 1 = 5, 10, 195.189.192.153, 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+eihzKSgvfDIzMDI1YGc0YWhub2h1P
GpUVwdVB1NSBgpRXV4=", 1, 0, 2, 15, 60, 200,
"62cabed25276f6d59432fcacf295a1346";
WebUsers 1 = "User",
"$1$fRwcHLO4tOHmvOKy7Oiys7m5vrbzpqfy0KL0r6v7q/iv/P35kpmUwcXBkZWYy5iaz8+Wm

```

```

NGBgoPXhdTRi4yDj94=", 3, 0, 2, 15, 60, 50,
"e124fc45691a62316416e055a60edb6f";

[ \WebUsers ]

[ TLSContexts ]

FORMAT TLSContexts_Index = TLSContexts_Name, TLSContexts_TLSVersion,
TLSContexts_DTLSVersion, TLSContexts_ServerCipherString,
TLSContexts_ClientCipherString, TLSContexts_RequireStrictCert,
TLSContexts_OcspEnable, TLSContexts_OcspServerPrimary,
TLSContexts_OcspServerSecondary, TLSContexts_OcspServerPort,
TLSContexts_OcspDefaultResponse, TLSContexts_DHKeySize;
TLSContexts 0 = "default", 7, 0, "RC4:EXP", "ALL:!ADH", 0, 0, 0.0.0.0,
0.0.0.0, 2560, 0, 1024;

[ \TLSContexts ]

[ AudioCodersGroups ]

FORMAT AudioCodersGroups_Index = AudioCodersGroups_Name;
AudioCodersGroups 0 = "AudioCodersGroups_0";
AudioCodersGroups 1 = "AudioCodersGroups_1";

[ \AudioCodersGroups ]

[ AllowedAudioCodersGroups ]

FORMAT AllowedAudioCodersGroups_Index = AllowedAudioCodersGroups_Name;
AllowedAudioCodersGroups 0 = "S4B Allowed Coders";
AllowedAudioCodersGroups 1 = "BroadCloud Allowed Coders";

[ \AllowedAudioCodersGroups ]

[ IpProfile ]

FORMAT IpProfile_Index = IpProfile_ProfileName, IpProfile_IpPreference,
IpProfile_CodersGroupName, IpProfile_IsFaxUsed,
IpProfile_JitterBufMinDelay, IpProfile_JitterBufOptFactor,
IpProfile_IPDiffServ, IpProfile_SigIPDiffServ, IpProfile_SCE,
IpProfile_RTPRedundancyDepth, 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_SBCExtensionCodersGroupName,
IpProfile_MediaIPVersionPreference, IpProfile_TranscodingMode,
IpProfile_SBCAllowedMediaTypes, IpProfile_SBCAllowedAudioCodersGroupName,
IpProfile_SBCAllowedVideoCodersGroupName, 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_SBCFaxCodersGroupName,
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_SBC RTP Redundancy Behavior,
IpProfile_SBCPlayRBTTToTransferee, IpProfile_SBC RTCP Mode,
IpProfile_SBCJitterCompensation,
IpProfile_SBCRemoteRenegotiateOnFaxDetection,
IpProfile_JitterBufMaxDelay,
IpProfile_SBCUserBehindUdpNATRegistrationTime,
IpProfile_SBCUserBehindTcpNATRegistrationTime,
IpProfile_SBCSDPHandleRTCPAttribute,
IpProfile_SBCRemoveCryptoLifetimeInSDP, IpProfile_SBCIceMode,
IpProfile_SBC RTP Mux, IpProfile_SBC Media Security Method,
IpProfile_SBCHandleXDetect, IpProfile_SBC RTCP Feedback,
IpProfile_SBCRemoteRepresentationMode, IpProfile_SBC Keep VIA Headers,
IpProfile_SBC Keep Routing Headers, IpProfile_SBC Keep User Agent Header,
IpProfile_SBCRemoteMultipleEarlyDialogs,
IpProfile_SBCRemoteMultipleAnswersMode, IpProfile_SBC Direct Media Tag,
IpProfile_SBC Adapt RFC2833 BW To Voice Coder BW,
IpProfile_CreatedByRoutingServer, IpProfile_SBC Fax Rerouting Mode,
IpProfile_SBC Max Call Duration, IpProfile_SBC Generate RTP,
IpProfile_SBC ISUP Body Handling, IpProfile_SBC ISUP Variant,
IpProfile_SBC Voice Quality Enhancement, IpProfile_SBC Max Opus BW;
IpProfile 1 = "S4B", 1, "AudioCodersGroups_0", 0, 10, 10, 46, 24, 0, 0,
0, 2, 0, 0, 0, -1, 1, 0, 0, -1, 0, 4, -1, 1, 1, 0, 0, "",
"AudioCodersGroups_0", 0, 0, "", "S4B Allowed Coders", "", 0, 1, 1, 0, 0,
0, 8, 300, 400, 0, 0, 0, "", 0, 0, 1, 3, 0, 1, 1, 0, 3, 2, 1, 0, 1, 1, 1,
1, 1, 0, 1, 0, 0, 101, 0, 1, 0, 1, 1, 0, 3, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0,
300, -1, -1, 0, 0, 0, 0, 0, 0, -1, -1, -1, 0, "", 0, 0, 0, 0,
0, 0, 0, 0, 0, 0;

IpProfile 2 = "BroadCloud", 1, "AudioCodersGroups_0", 0, 10, 10, 46, 24,
0, 0, 0, 2, 0, 0, 0, -1, 1, 0, 0, -1, 0, 4, -1, 1, 1, 0, 0, "",
"AudioCodersGroups_1", 0, 0, "", "BroadCloud Allowed Coders", "", 2, 1,
0, 0, 1, 0, 8, 300, 400, 0, 0, 0, "", 0, 0, 1, 3, 0, 2, 2, 1, 3, 2, 1, 0,
1, 1, 0, 1, 0, 0, 0, 0, 0, 0, 0, 1, 0, 0, 0, 0, 0, 0, 0, 0, 1, 0,
0, 0, 300, -1, -1, 0, 1, 0, 0, 0, 0, -1, -1, -1, 0, "", 0, 0,
0, 0, 0, 0, 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_TopoLocation;
CpMediaRealm 0 = "MRLan", "Voice", "", 6000, 250, 8499, 1, "", "", 0;

```

```

CpMediaRealm 1 = "MRWan", "WANSP", "", 6000, 250, 8499, 0, "", "", 1;

[ \CpMediaRealm ]

[ SBCRoutingPolicy ]

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

[ \SBCRoutingPolicy ]

[ SRD ]

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

[ \SRD ]

[ MessagePolicy ]

FORMAT MessagePolicy_Index = MessagePolicy_Name,
MessagePolicy_MaxMessageLength, MessagePolicy_MaxHeaderLength,
MessagePolicy_MaxBodyLength, MessagePolicy_MaxNumHeaders,
MessagePolicy_MaxNumBodies, MessagePolicy_SendRejection,
MessagePolicy_MethodList, MessagePolicy_MethodListType,
MessagePolicy_BodyList, MessagePolicy_BodyListType,
MessagePolicy_UseMaliciousSignatureDB;
MessagePolicy 0 = "Malicious Signature DB Protection", -1, -1, -1, -1, -1,
1, "", 0, "", 0, 1;

[ \MessagePolicy ]

[ SIPInterface ]

FORMAT SIPInterface_Index = SIPInterface_InterfaceName,
SIPInterface_NetworkInterface, SIPInterface_ApplicationType,
SIPInterface_UDPPort, SIPInterface_TCPPort, SIPInterface_TLSPort,
SIPInterface_SRDNName, 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_TopoLocation;
SIPInterface 0 = "SIPInterface_LAN", "Voice", 2, 0, 0, 5067,
"DefaultSRD", "", "default", -1, 0, 500, -1, 0, "MRLan", 0, -1, -1, -1,
0, 0;
SIPInterface 1 = "SIPInterface_WAN", "WANSP", 2, 0, 0, 5061,
"DefaultSRD", "", "default", -1, 0, 500, -1, 0, "MRWan", 0, -1, -1, -1,
0, 1;

```

```

[ \SIPInterface ]

[ ProxySet ]

FORMAT ProxySet_Index = ProxySet_ProxyName,
ProxySet_EnableProxyKeepAlive, ProxySet_ProxyKeepAliveTime,
ProxySet_ProxyLoadBalancingMethod, ProxySet_IsProxyHotSwap,
ProxySet_SRDNName, ProxySet_ClassificationInput, ProxySet_TLSContextName,
ProxySet_ProxyRedundancyMode, ProxySet_DNSResolveMethod,
ProxySet_KeepAliveFailureResp, ProxySet_GWIPv4SIPInterfaceName,
ProxySet_SBCIPv4SIPInterfaceName, ProxySet_GWIPv6SIPInterfaceName,
ProxySet_SBCIPv6SIPInterfaceName, ProxySet_MinActiveServersLB,
ProxySet_SuccessDetectionRetries, ProxySet_SuccessDetectionInterval,
ProxySet_FailureDetectionRetransmissions;
ProxySet 1 = "S4B", 1, 60, 1, 1, "DefaultSRD", 0, "default", 1, -1, "",
"", "SIPInterface_LAN", "", "", 1, 1, 10, -1;
ProxySet 2 = "BroadCloud", 1, 60, 0, 1, "DefaultSRD", 0, "default", 1, 1,
"", "", "SIPInterface_WAN", "", "", 1, 1, 10, -1;

[ \ProxySet ]

[ IPGroup ]

FORMAT IPGroup_Index = IPGroup_Type, IPGroup_Name, IPGroup_ProxySetName,
IPGroup_SIPGroupName, IPGroup_ContactUser, IPGroup_SipReRoutingMode,
IPGroup_AlwaysUseRouteTable, IPGroup_SRDNName, IPGroup_MediaRealm,
IPGroup_ClassifyByProxySet, IPGroup_ProfileName,
IPGroup_MaxNumOfRegUsers, IPGroup_InboundManSet, IPGroup_OutboundManSet,
IPGroup_RegistrationMode, IPGroup_AuthenticationMode, IPGroup_MethodList,
IPGroup_EnableSBCCClientForking, IPGroup_SourceUriInput,
IPGroup_DestUriInput, IPGroup_ContactName, IPGroup_Username,
IPGroup_Password, IPGroup_UUIFormat, IPGroup_QOEProfile,
IPGroup_BWProfile, IPGroup_AlwaysUseSourceAddr, IPGroup_MsgManUserDef1,
IPGroup_MsgManUserDef2, IPGroup_SIPConnect, IPGroup_SBCPSAPMode,
IPGroup_DTLSContext, IPGroup_CreatedByRoutingServer,
IPGroup_UsedByRoutingServer, IPGroup_SBCOperationMode,
IPGroup_SBCRouteUsingRequestURIPort, IPGroup_SBCKeepOriginalCallID,
IPGroup_TopologyLocation, IPGroup_SBCDialPlanName,
IPGroup_CallSetupRulesSetId;
IPGroup 1 = 0, "S4B", "S4B", "interop.adpt-tech.com", "", -1, 0,
"DefaultSRD", "MRLan", 1, "S4B", -1, -1, -1, 0, 0, "", 0, -1, -1, "",
"Admin", "$1$aCkNBwIC", 0, "", "", 0, "", "", 0, 0, "default", 0, 0, -1,
0, 0, 0, "", -1;
IPGroup 2 = 0, "BroadCloud", "BroadCloud", "interop.adpt-tech.com", "", -1,
0, "DefaultSRD", "MRWan", 1, "BroadCloud", -1, -1, 4, 0, 0, "", 0, -1,
-1, "", "Admin", "$1$aCkNBwIC", 0, "", "", 0, "", "", 0, 0, "default", 0,
0, -1, 0, 0, 1, "", -1;

[ \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 = "1", 0, "FE.S4B.interop:5067", 2;
ProxyIp 1 = "2", 0, "hs2.fedsipt1.broadcloudgov.us", 2;

[ \ProxyIp ]


[ Account ]

FORMAT Account_Index = Account_ServedTrunkGroup,
Account_ServedIPGroupName, Account_ServingIPGroupName, Account_Username,
Account_Password, Account_HostName, Account_Register,
Account_ContactUser, Account_ApplicationType;
Account 0 = -1, "S4B", "BroadCloud", "8325624857",
"$1$jt/69vP78vC0ruL8uA==", "interop.adpt-tech.com", 1, "8325624857", 2;

[ \Account ]


[ 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_DestTags,
IP2IPRouting_SrcTags, IP2IPRouting_IPGroupSetName;
IP2IPRouting 0 = "Terminate OPTIONS", "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, "BroadCloud", "", "", 0, -1, 0, 0,
",", ", ", ";
IP2IPRouting 2 = "ITSP to S4B", "Default_SBCRoutingPolicy", "BroadCloud",
"**", "**", "**", 0, "", "Any", 0, -1, 0, "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_DestTags, IPOutboundManipulation_SrcTags;
IPOutboundManipulation 0 = "Add + toward S4B",
"Default_SBCRoutingPolicy", 0, "BroadCloud", "S4B", "**", **, **, **,
**", "", 0, "Any", 0, 1, 0, 0, 255, "+", "", 0, "", "";
IPOutboundManipulation 1 = "Change + to 011", "Default_SBCRoutingPolicy",
0, "S4B", "BroadCloud", **, **, "+", **, **, **, "", 0, "Any", 0, 1, 1,
0, 255, "011", "", 0, "", "";
IPOutboundManipulation 2 = "Remove + from Source",
"Default_SBCRoutingPolicy", 0, "S4B", "BroadCloud", "+8", **, **, **,
**", "", 0, "Any", 0, 0, 1, 0, 255, "", "", 0, "", "";
[ \IPOutboundManipulation ]

[ MessageManipulations ]

FORMAT MessageManipulations_Index =
MessageManipulations_ManipulationName, MessageManipulations_ManSetID,
MessageManipulations_MessageType, MessageManipulations_Condition,
MessageManipulations_ActionSubject, MessageManipulations_ActionType,
MessageManipulations_ActionValue, MessageManipulations_RowRole;
MessageManipulations 0 = "Call Forward", 4, "invite", "header.history-
info exists", "header.diversion", 0, "header.from", 0;
MessageManipulations 1 = "Call Forward", 4, "invite", "header.history-
info.0 regex (<sip:(.)().*)@(.*)", "header.diversion.url.user", 2,
"$3", 0;
MessageManipulations 2 = "Call Forward", 4, "", "", "header.history-
info", 1, "", 1;
MessageManipulations 3 = "Call Transfer", 4, "invite", "header.referred-
by exists", "header.referred-by.url.host", 2, "header.from.url.host", 0;
MessageManipulations 4 = "Call Transfer", 4, "invite", "header.referred-
by exists", "header.referred-by.url.user", 6, "+", 0;
MessageManipulations 5 = "Call Transfer", 4, "invite", "header.referred-
by exists", "header.from.url.user", 2, "header.referred-by.url.user", 0;
MessageManipulations 6 = "Reject Responses", 4, "any.response",
"header.request-uri.methodtype=='503'", "header.request-uri.methodtype",
2, "'480'", 0;
[ \MessageManipulations ]

[ GwRoutingPolicy ]

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

[ MaliciousSignatureDB ]

FORMAT MaliciousSignatureDB_Index = MaliciousSignatureDB_Name,
MaliciousSignatureDB_Pattern;
MaliciousSignatureDB 0 = "SIPVicious", "Header.User-Agent.content prefix
'friendly-scanner'";
MaliciousSignatureDB 1 = "SIPScan", "Header.User-Agent.content prefix
'sip-scan'";
MaliciousSignatureDB 2 = "Smap", "Header.User-Agent.content prefix
'smap'";
MaliciousSignatureDB 3 = "Sipsak", "Header.User-Agent.content prefix
'sipsak'";
MaliciousSignatureDB 4 = "Sipcli", "Header.User-Agent.content prefix
'sipcli'";
MaliciousSignatureDB 5 = "Sivus", "Header.User-Agent.content prefix
'SIVuS'";
MaliciousSignatureDB 6 = "Gulp", "Header.User-Agent.content prefix
'Gulp'";
MaliciousSignatureDB 7 = "Sipv", "Header.User-Agent.content prefix
'sipv'";
MaliciousSignatureDB 8 = "Sundayddr Worm", "Header.User-Agent.content
prefix 'sundayddr'";
MaliciousSignatureDB 9 = "VaxIPUserAgent", "Header.User-Agent.content
prefix 'VaxIPUserAgent'";
MaliciousSignatureDB 10 = "VaxSIPUserAgent", "Header.User-Agent.content
prefix 'VaxSIPUserAgent'";
MaliciousSignatureDB 11 = "SipArmyKnife", "Header.User-Agent.content
prefix 'siparmyknife'";

[ \MaliciousSignatureDB ]

[ AllowedAudioCoders ]

FORMAT AllowedAudioCoders_Index =
AllowedAudioCoders_AllowedAudioCodersGroupName,
AllowedAudioCoders_AllowedAudioCodersIndex, AllowedAudioCoders_CoderID,
AllowedAudioCoders_UserDefineCoder;
AllowedAudioCoders 0 = "S4B Allowed Coders", 0, 1, "";
AllowedAudioCoders 1 = "S4B Allowed Coders", 1, 2, "";
AllowedAudioCoders 2 = "BroadCloud Allowed Coders", 0, 3, "";
AllowedAudioCoders 3 = "BroadCloud Allowed Coders", 1, 1, "";
AllowedAudioCoders 4 = "BroadCloud Allowed Coders", 2, 2, "";

[ \AllowedAudioCoders ]

```

```
[ AudioCoders ]  
  
FORMAT AudioCoders_Index = AudioCoders_AudioCodersGroupId,  
AudioCoders_AudioCodersIndex, AudioCoders_Name, AudioCoders_pTime,  
AudioCoders_rate, AudioCoders_PayloadType, AudioCoders_Sce,  
AudioCoders_CoderSpecific;  
AudioCoders 0 = "AudioCodersGroups_0", 0, 2, 2, 90, -1, 1, "";  
AudioCoders 1 = "AudioCodersGroups_0", 1, 1, 2, 90, -1, 1, "";  
AudioCoders 2 = "AudioCodersGroups_1", 0, 3, 2, 19, -1, 0, "";  
AudioCoders 3 = "AudioCodersGroups_1", 1, 1, 2, 90, -1, 0, "";  
AudioCoders 4 = "AudioCodersGroups_1", 2, 2, 2, 90, -1, 0, "";  
  
[ \AudioCoders ]
```

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/contact

Website: www.audioCodes.com

©2017 AudioCodes Ltd. All rights reserved. AudioCodes, AC, HD VoIP, HD VoIP Sounds Better, IPmedia, Mediant, MediaPack, What's Inside Matters, OSN, SmartTAP, User Management Pack, VMAS, VoIPerfect, VoIPerfectHD, Your Gateway To VoIP, 3GX, VocaNom, AudioCodes One Voice and CloudBond 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.

Document #: LTTRT-12381

