

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

Version 7.2



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1 Introduction

This Configuration Note describes how to set up AudioCodes Enterprise Session Border Controller (hereafter, referred to as *E-SBC*) for interworking between 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.audiocodes.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.

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2 Component Information

2.1 AudioCodes E-SBC Version

Table 2-1: AudioCodes E-SBC Version

SBC Vendor	AudioCodes
Models	▪ Mediant 500 E-SBC ▪ Mediant 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	▪ 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

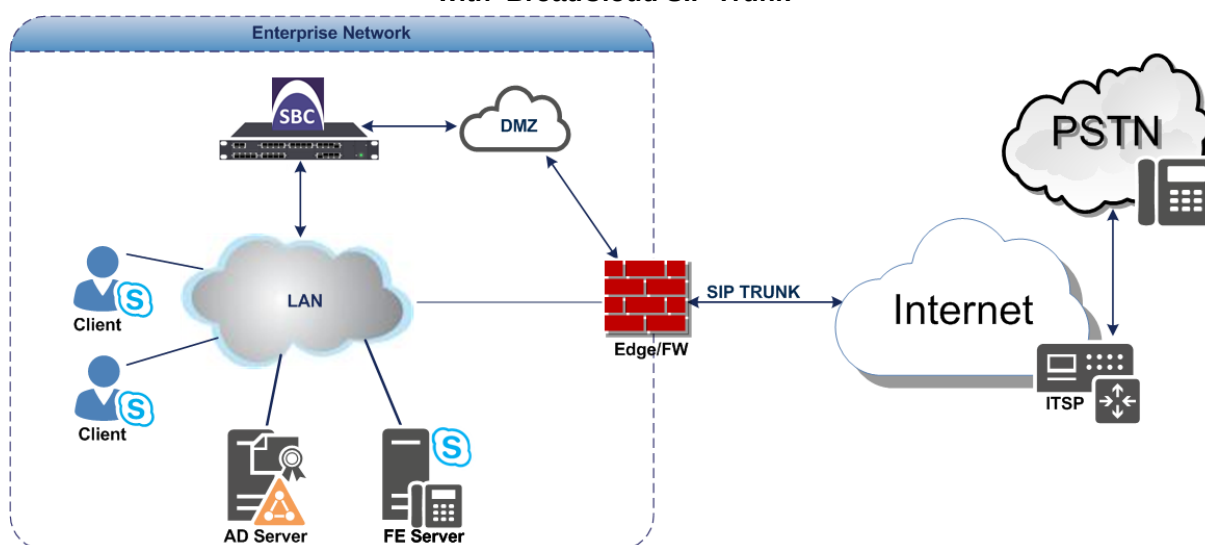
2.4 Interoperability Test Topology

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	- Microsoft Skype for Business Server 2015 environment is located on the Enterprise's LAN - BroadCloud SIP Trunk is located on the WAN
Signaling Transcoding	Both, Microsoft Skype for Business Server 2015 and BroadCloud SIP Trunk, operates with SIP-over-TLS transport type
Codecs Transcoding	- 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	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).

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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.



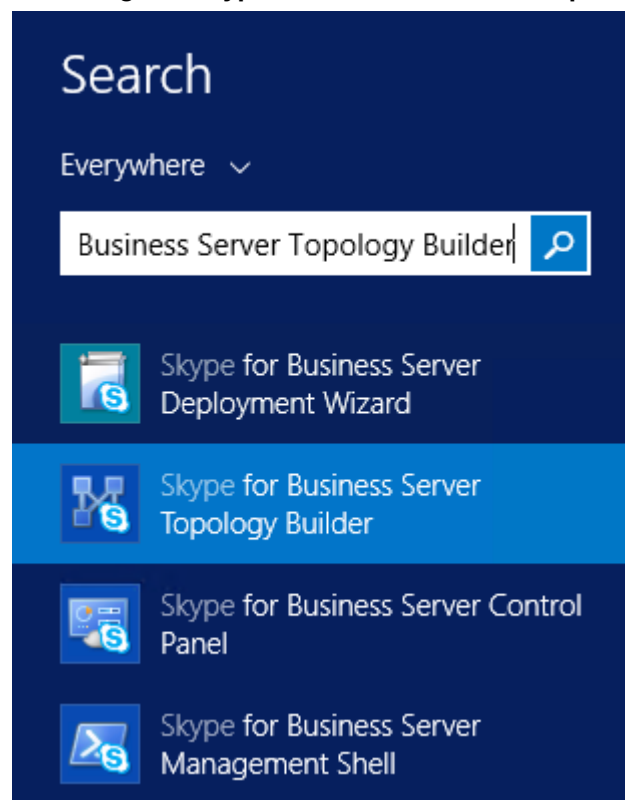
Note: Dial plans, voice policies, and PSTN usages are also necessary for Enterprise voice deployment; however, they are beyond the scope of this document.

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.

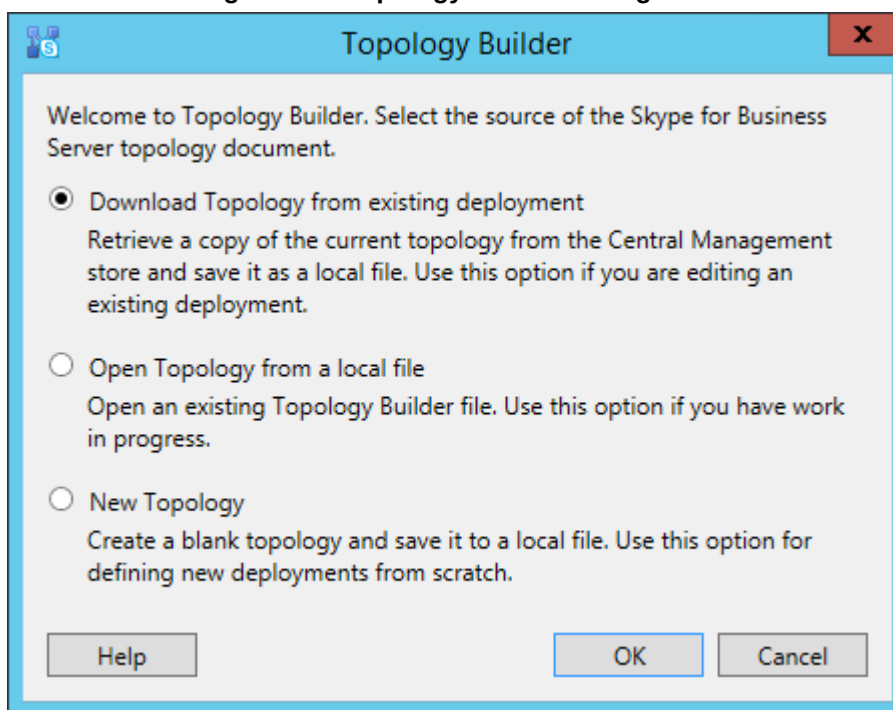
- **To configure E-SBC as IP/PSTN Gateway and associate it with Mediation Server:**
- 1. On the server where the Topology Builder is installed, start the Skype for Business Server 2015 Topology Builder (Windows **Start** menu > search for **Skype for Business Server Topology Builder**), as shown below:

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



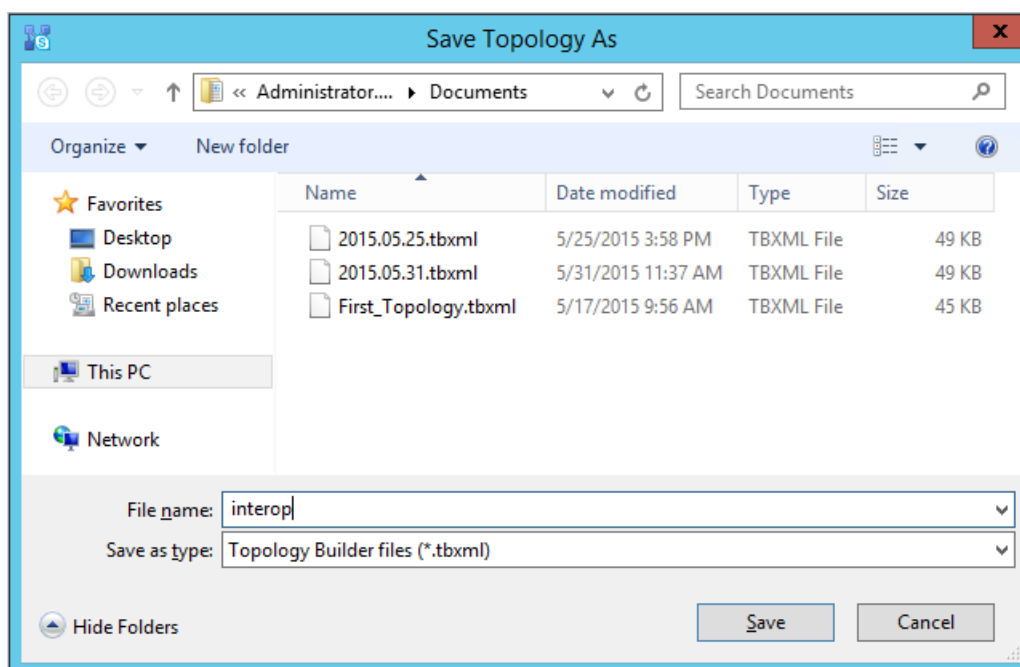
The following is displayed:

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:

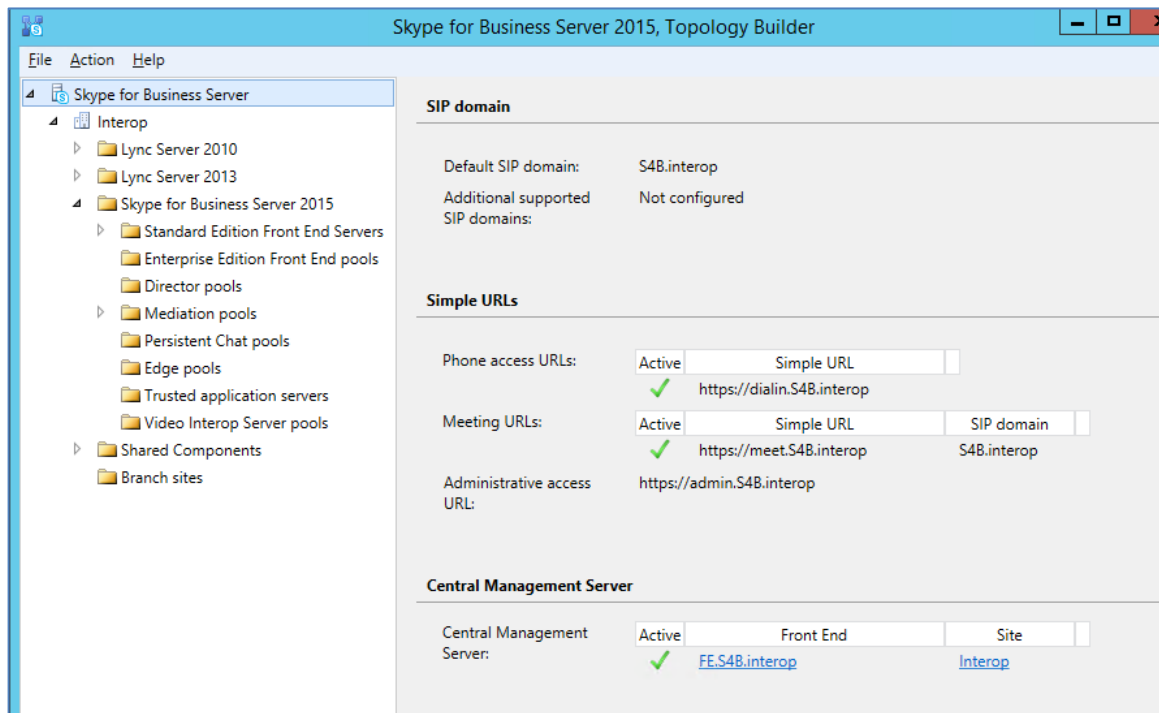
Figure 3-3: Save Topology Dialog Box



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.

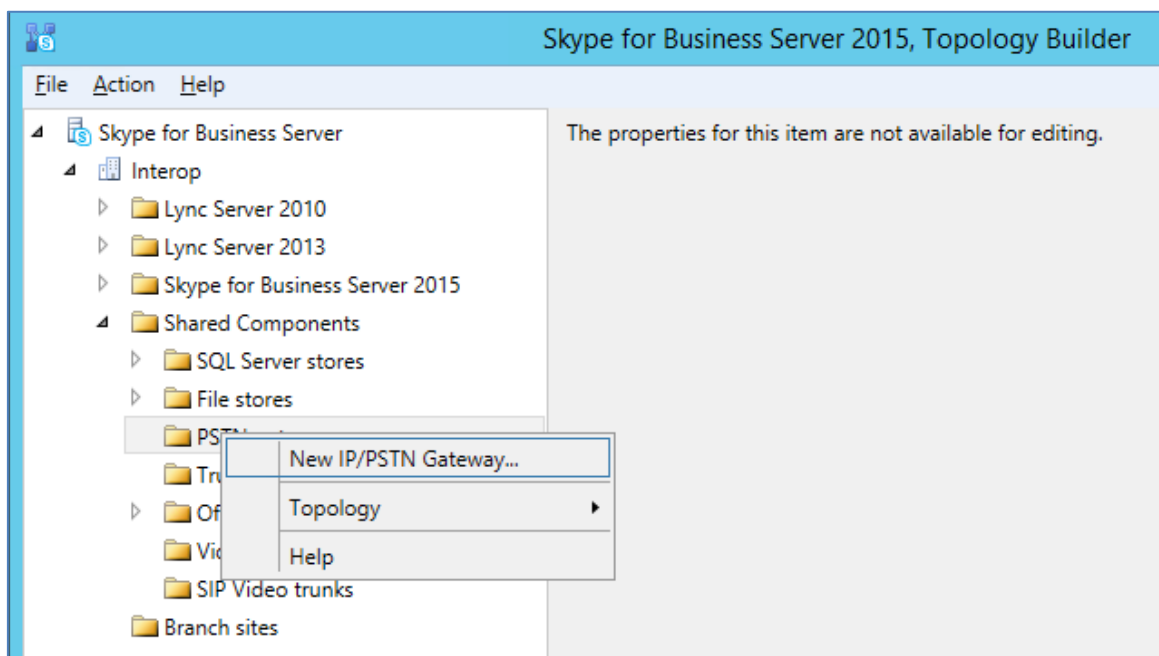
The Topology Builder screen with the downloaded Topology is displayed:

Figure 3-4: Downloaded Topology



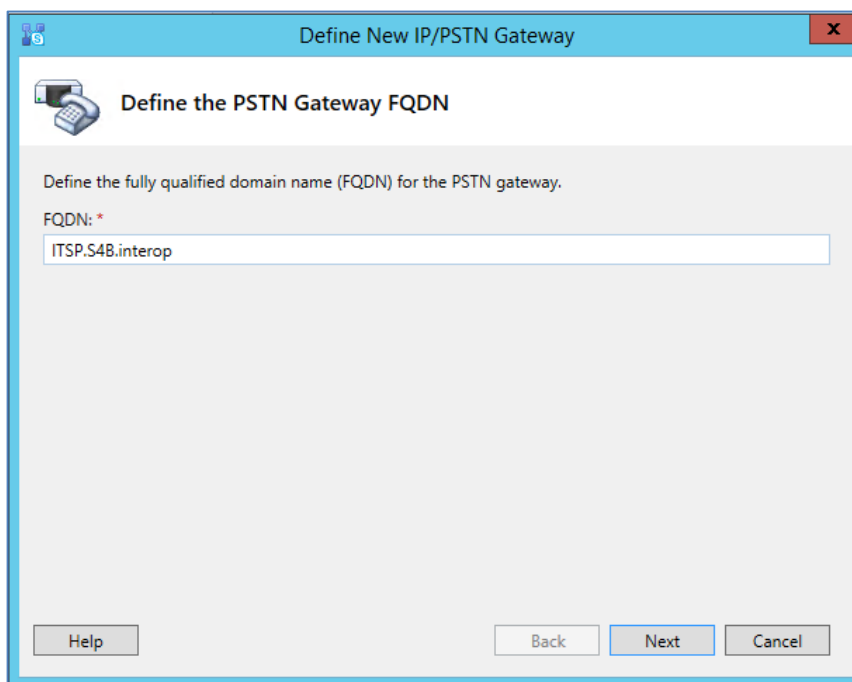
- Under the **Shared Components** node, right-click the **PSTN gateways** node, and then from the shortcut menu, choose **New IP/PSTN Gateway**, as shown below:

Figure 3-5: Choosing New IP/PSTN Gateway



The following is displayed:

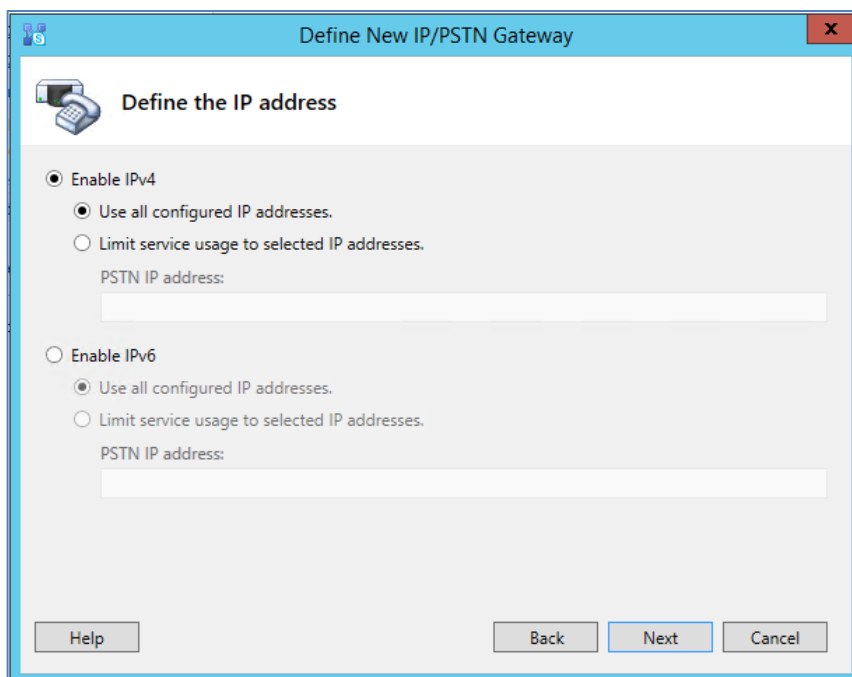
Figure 3-6: Define the PSTN Gateway FQDN



The screenshot shows a dialog box titled "Define New IP/PSTN Gateway" with a close button (X) in the top right corner. The main heading is "Define the PSTN Gateway FQDN" with a telephone icon. Below the heading, it says "Define the fully qualified domain name (FQDN) for the PSTN gateway." There is a text field labeled "FQDN: *" containing the text "ITSP.S4B.interop". At the bottom, there are three buttons: "Help", "Back", and "Next" (which is highlighted in blue), and a "Cancel" button.

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



The screenshot shows a dialog box titled "Define New IP/PSTN Gateway" with a close button (X) in the top right corner. The main heading is "Define the IP address" with a telephone icon. Below the heading, there are two radio button options: "Enable IPv4" (selected) and "Enable IPv6". Under "Enable IPv4", there are two sub-options: "Use all configured IP addresses." (selected) and "Limit service usage to selected IP addresses." Below these is a text field labeled "PSTN IP address:". The "Enable IPv6" section has the same sub-options and text field. At the bottom, there are three buttons: "Help", "Back", and "Next" (which is highlighted in blue), and a "Cancel" button.

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:

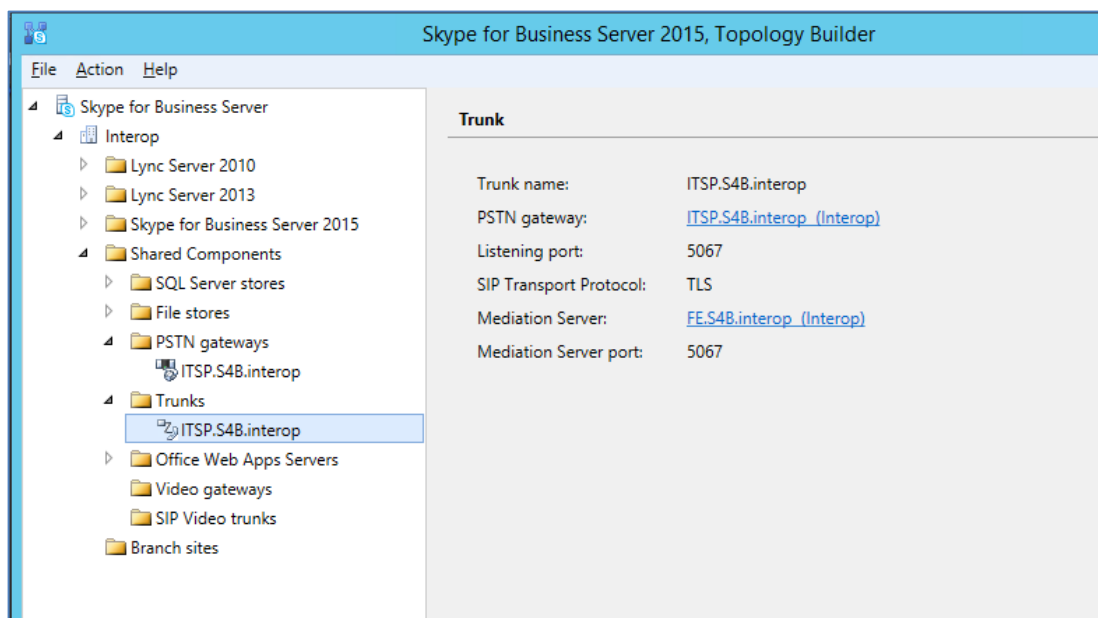
- When defining a PSTN gateway in Topology Builder, you must define a root trunk to successfully add the PSTN gateway to your topology.
- The root trunk cannot be removed until the associated PSTN gateway is removed.

Figure 3-8: Define the Root Trunk

- a. In the 'Listening Port for IP/PSTN Gateway' field, enter the listening port that the E-SBC will use for SIP messages from the Mediation Server that will be associated with the root trunk of the PSTN gateway (e.g., **5067**). This parameter is later configured in the SIP Interface table (see Section 4.3 on page 36).
- b. 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).
- c. In the 'Associated Mediation Server' field, select the Mediation Server pool to associate with the root trunk of this PSTN gateway.
- d. In the 'Associated Mediation Server Port' field, enter the listening port that the Mediation Server will use for SIP messages from the SBC (e.g., **5067**).
- e. Click **Finish**.

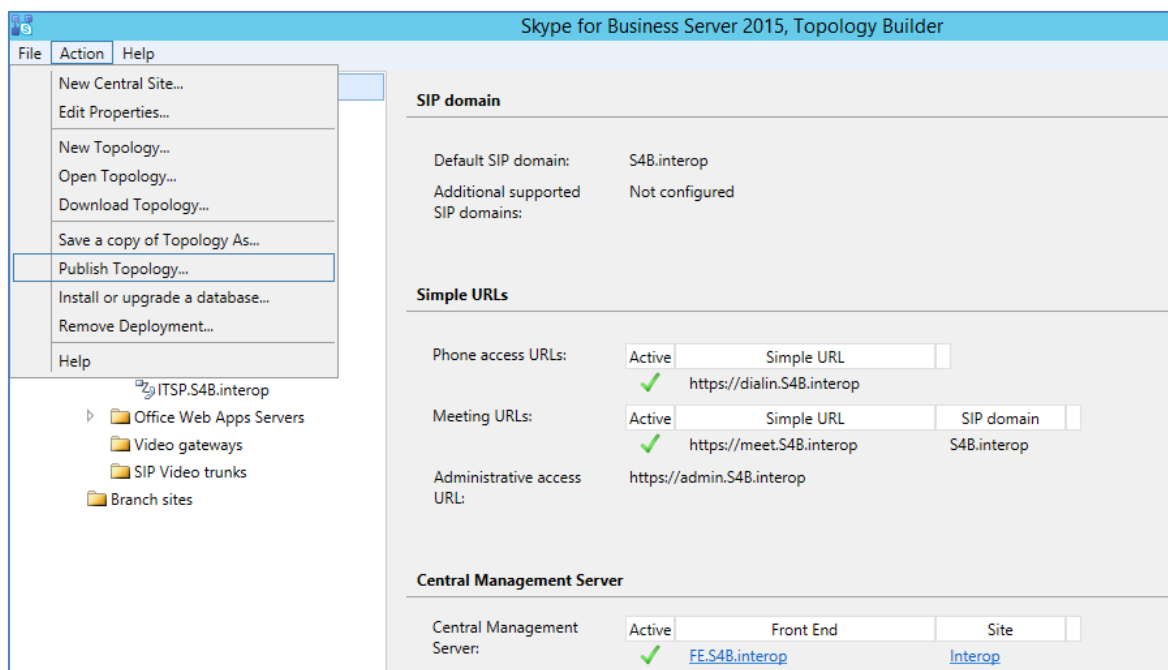
The E-SBC is added as a PSTN gateway, and a trunk is created as shown below:

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



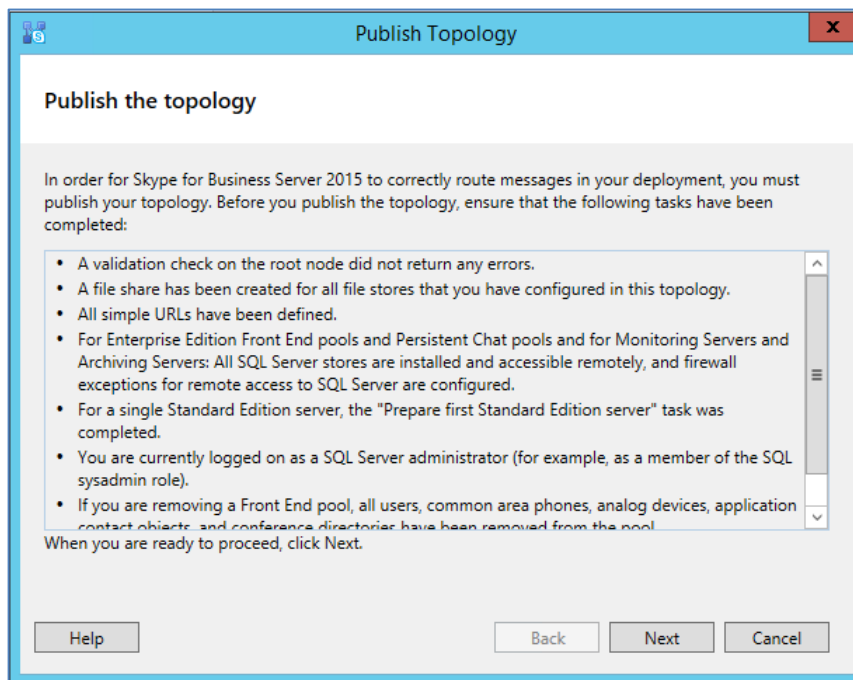
9. Publish the Topology: In the main tree, select the root node **Skype for Business Server**, and then from the **Action** menu, choose **Publish Topology**, as shown below:

Figure 3-10: Choosing Publish Topology



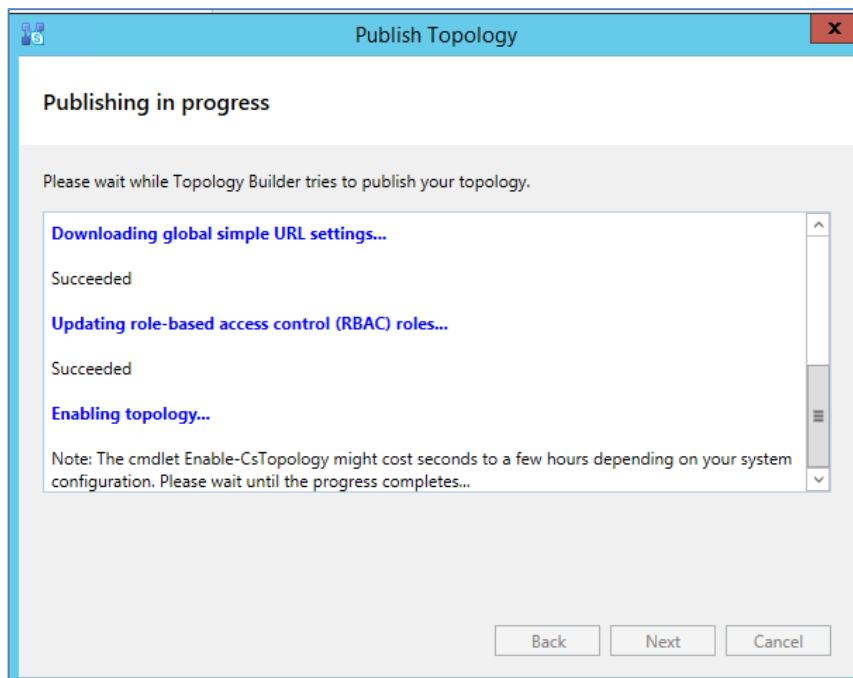
The following is displayed:

Figure 3-11: Publish the Topology



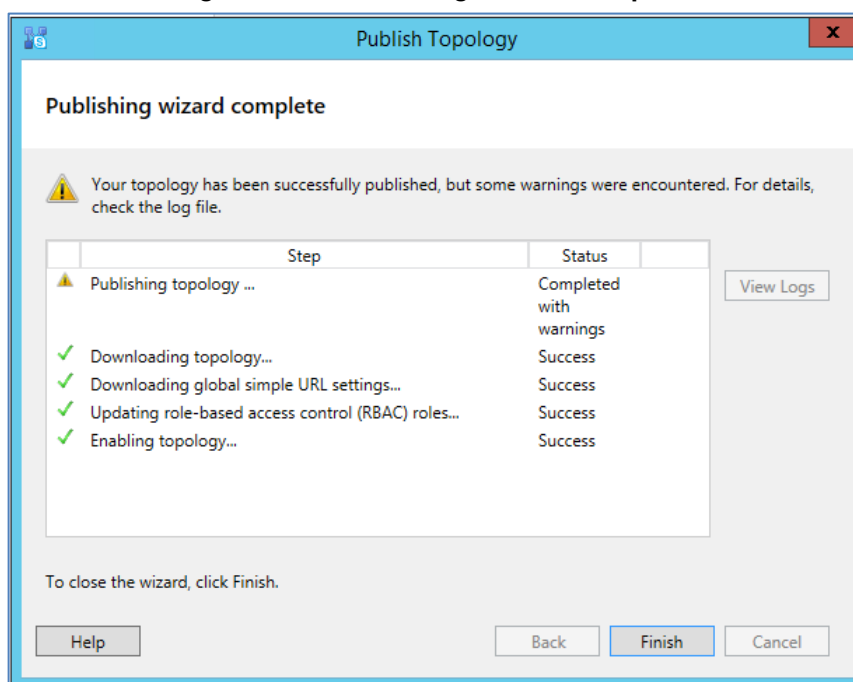
10. Click **Next**; the Topology Builder starts to publish your topology, as shown below:

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**.

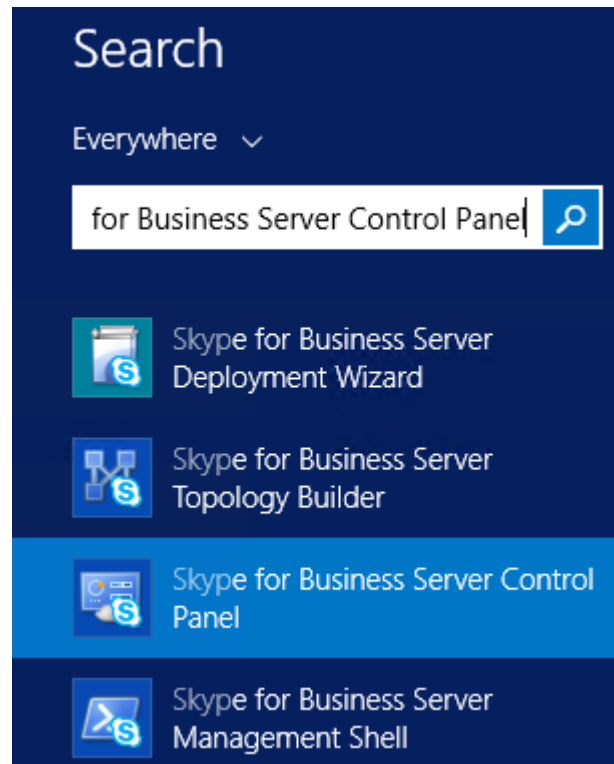
3.2 Configuring the "Route" on Skype for Business Server 2015

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:**

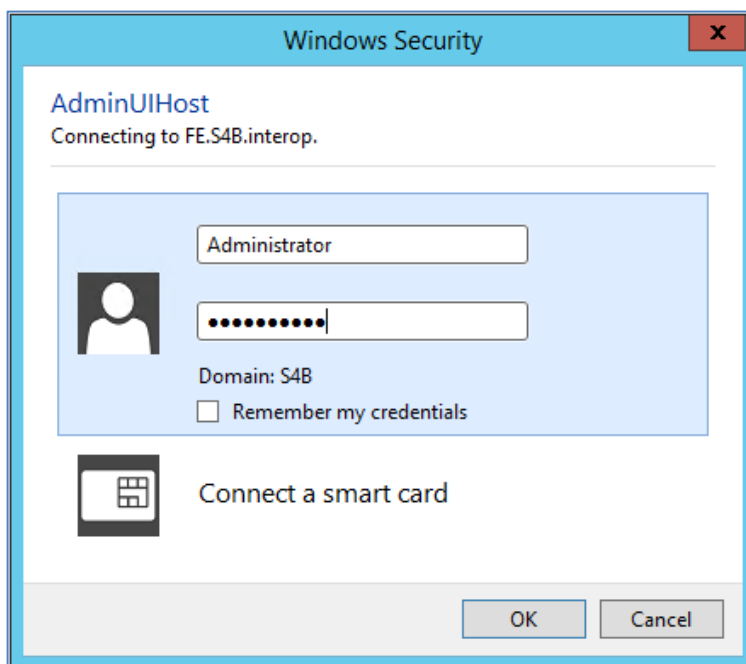
1. Start the Microsoft Skype for Business Server 2015 Control Panel (**Start** > search for **Microsoft Skype for Business Server Control Panel**), as shown below:

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



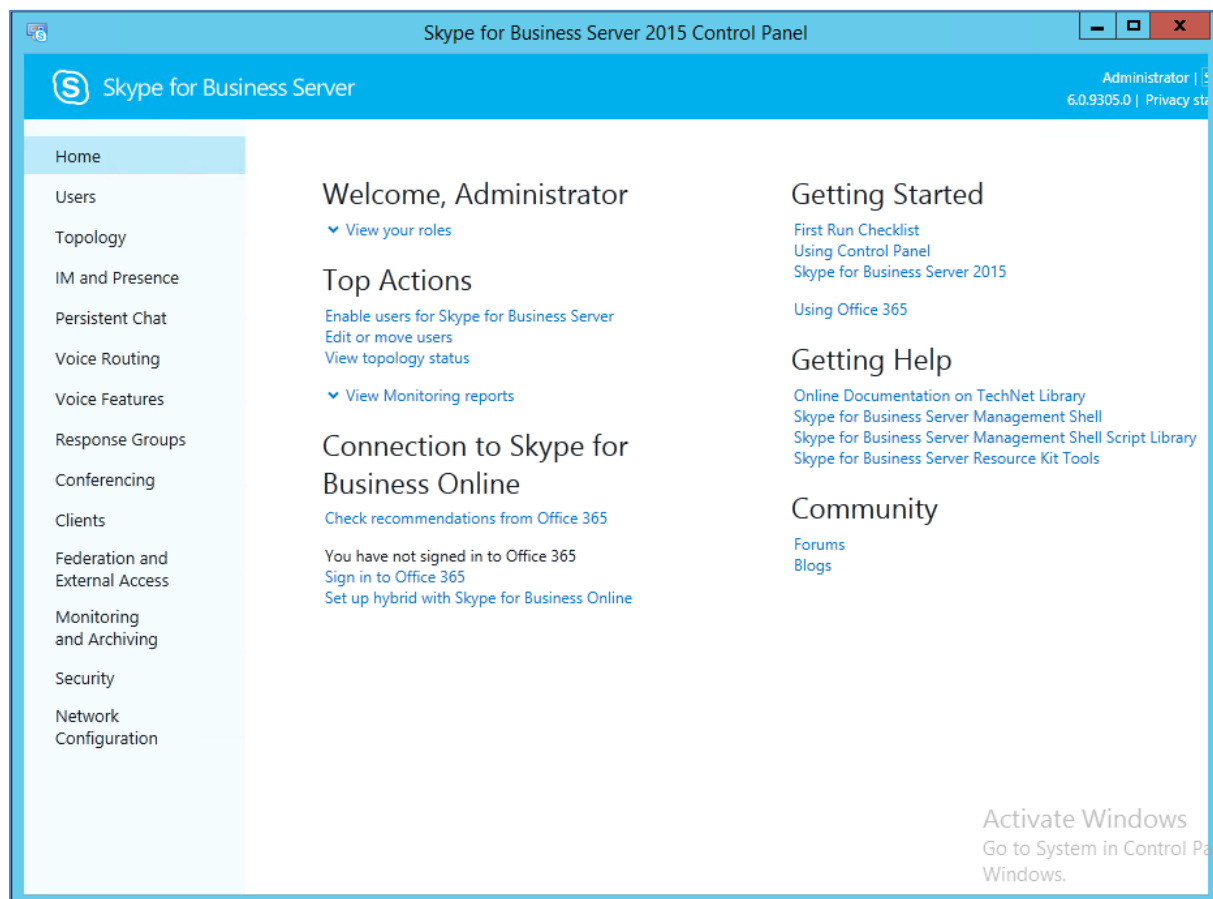
2. You are prompted to enter your login credentials:

Figure 3-15: Skype for Business Server Credentials



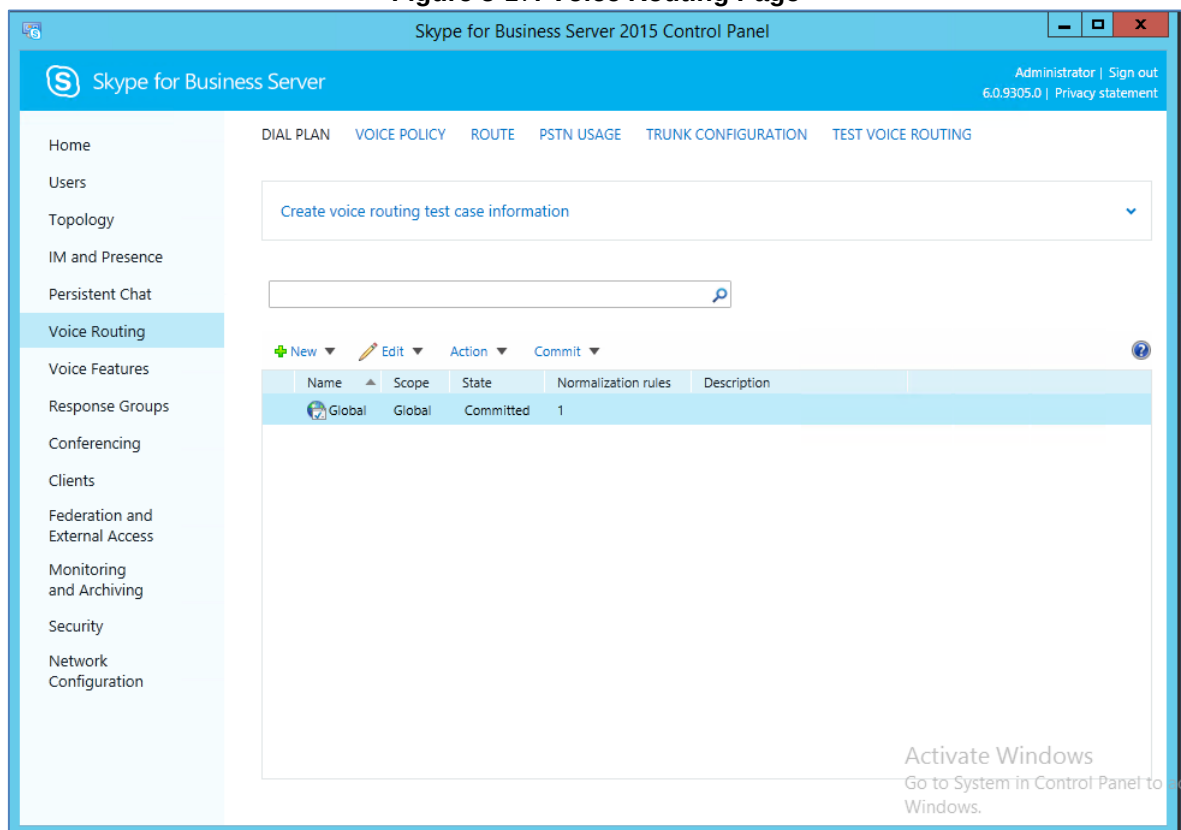
3. Enter your domain username and password, and then click **OK**; the Microsoft Skype for Business Server 2015 Control Panel is displayed:

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



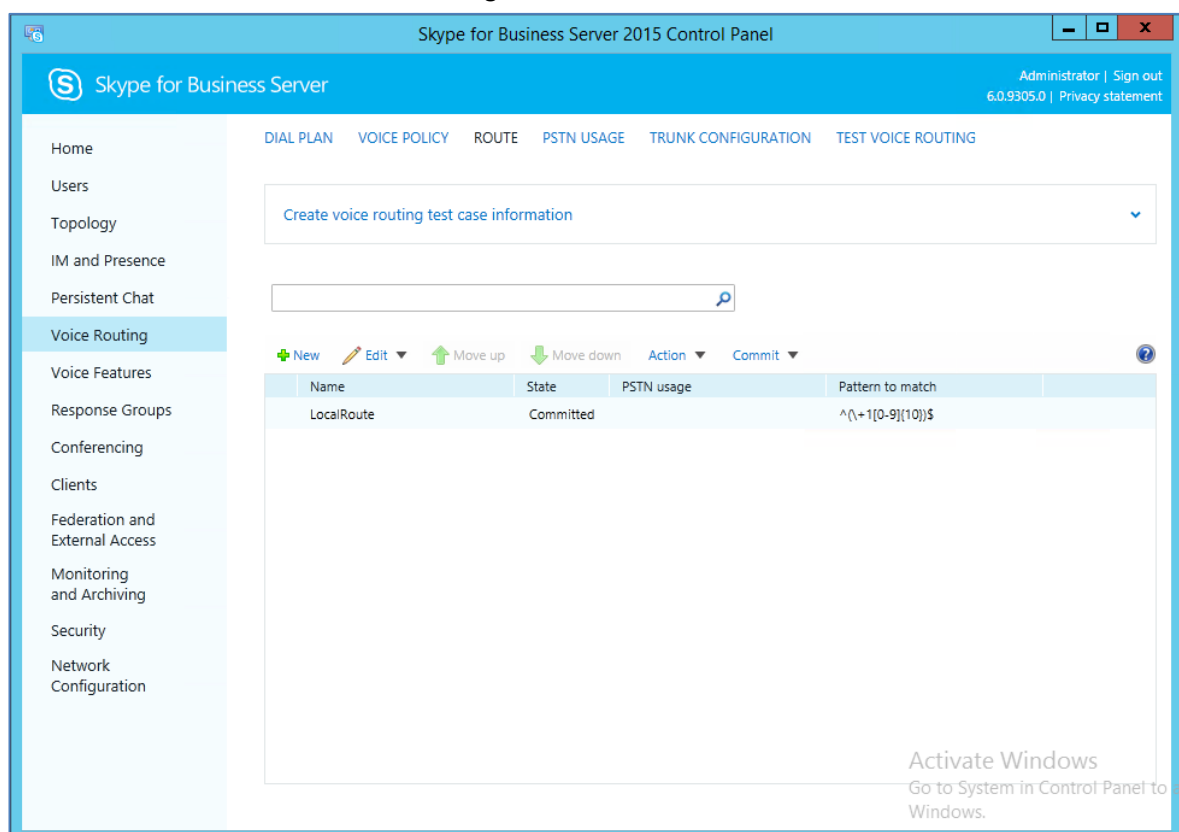
4. In the left navigation pane, select **Voice Routing**.

Figure 3-17: Voice Routing Page



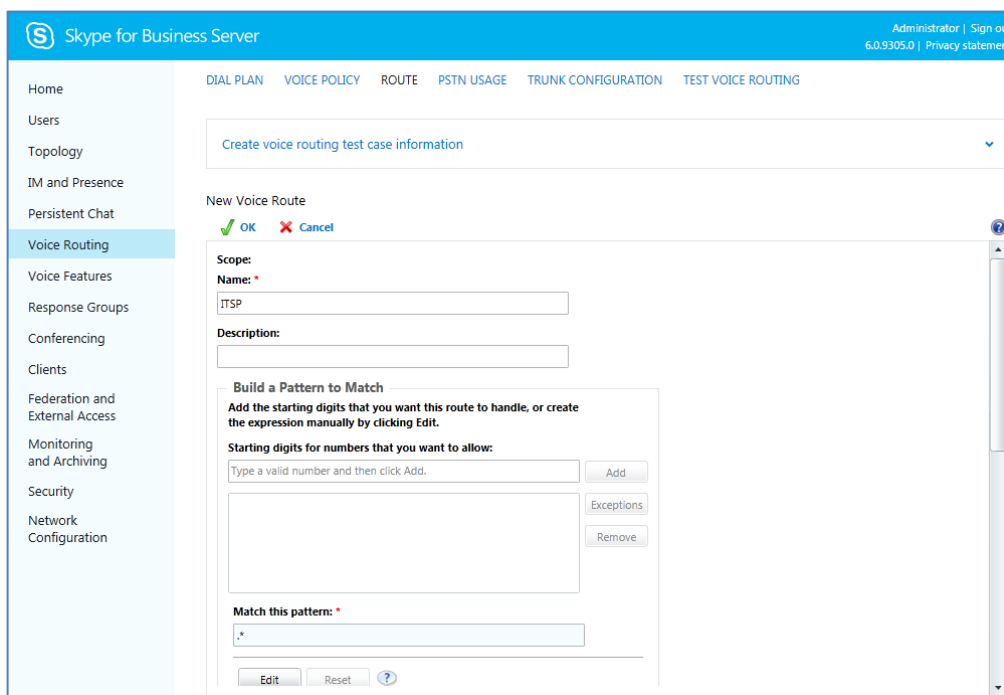
5. In the Voice Routing page, select the **Route** tab.

Figure 3-18: Route Tab



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

Figure 3-19: Adding New Voice Route



Skype for Business Server Administrator | Sign out 6.0.9305.0 | Privacy statement

Home DIAL PLAN VOICE POLICY ROUTE PSTN USAGE TRUNK CONFIGURATION TEST VOICE ROUTING

Home Users Topology IM and Presence Persistent Chat Voice Routing Voice Features Response Groups Conferencing Clients Federation and External Access Monitoring and Archiving Security Network Configuration

Create voice routing test case information

New Voice Route

OK Cancel

Scope:

Name: *

ITSP

Description:

Build a Pattern to Match

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

Starting digits for numbers that you want to allow:

Type a valid number and then click Add.

Add

Exceptions

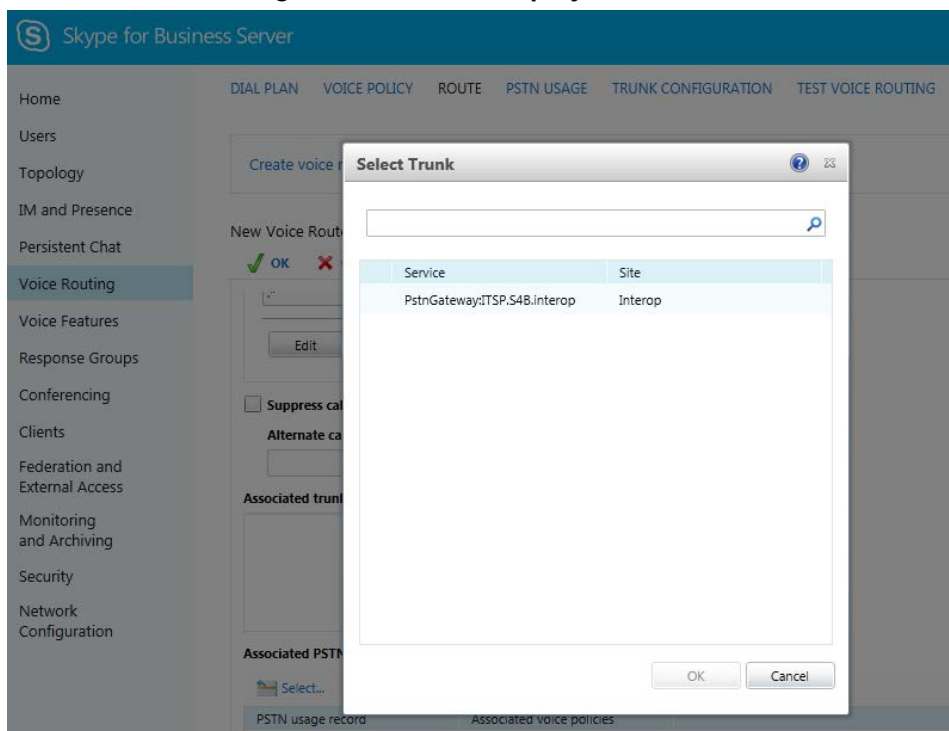
Remove

Match this pattern: *

Edit Reset ?

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



Skype for Business Server

Home DIAL PLAN VOICE POLICY ROUTE PSTN USAGE TRUNK CONFIGURATION TEST VOICE ROUTING

Home Users Topology IM and Presence Persistent Chat Voice Routing Voice Features Response Groups Conferencing Clients Federation and External Access Monitoring and Archiving Security Network Configuration

Create voice routing test case information

New Voice Route

OK Cancel

Scope:

Name: *

ITSP

Description:

Build a Pattern to Match

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

Starting digits for numbers that you want to allow:

Type a valid number and then click Add.

Add

Exceptions

Remove

Match this pattern: *

Edit Reset ?

Select Trunk

Service	Site
PstnGateway:ITSP.S4B.interop	Interop

OK Cancel

Associated trunk

Associated PSTN

PSTN usage record Associated voice policies

- 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

Skype for Business Server

Home Users Topology IM and Presence Persistent Chat **Voice Routing** Voice Features Response Groups Conferencing Clients Federation and External Access Monitoring and Archiving Security Network Configuration

DIAL PLAN VOICE POLICY ROUTE PSTN USAGE TRUNK CONFIGURATION TEST VOICE ROUTING

Create voice routing test case information

New Voice Route

OK Cancel

Match this pattern: *

Edit Reset ?

☐ Suppress caller ID

Alternate caller ID:

Associated trunks:

PstnGateway:ITSP.S4B.interop Add... Remove

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

Skype for Business Server

Home Users Topology IM and Presence Persistent Chat **Voice Routing** Voice Features Response Groups Conferencing Clients Federation and External Access Monitoring and Archiving Security Network Configuration

DIAL PLAN VOICE POLICY ROUTE PSTN USAGE TRUNK CONFIGURATION TEST VOICE ROUTING

Create voice routing test case information

New Voice Route

OK Cancel

Associated trunks:

PstnGateway:ITSP.S4B.interop Add... Remove

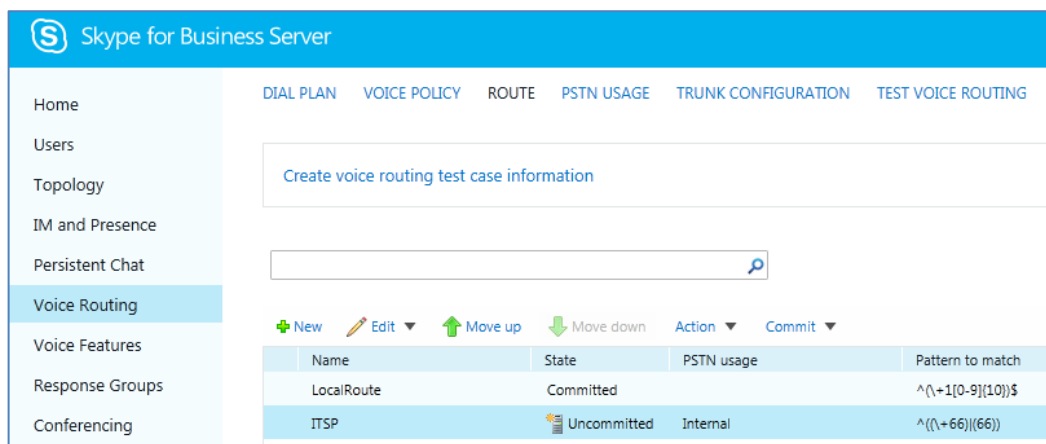
Associated PSTN Usages

Select... Remove ↑ ↓

PSTN usage record	Associated voice policies
Internal	
Local	
Long Distance	

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

Figure 3-23: Confirmation of New Voice Route



Skype for Business Server

Home Users Topology IM and Presence Persistent Chat **Voice Routing** Voice Features Response Groups Conferencing

DIAL PLAN VOICE POLICY **ROUTE** PSTN USAGE TRUNK CONFIGURATION TEST VOICE ROUTING

Create voice routing test case information

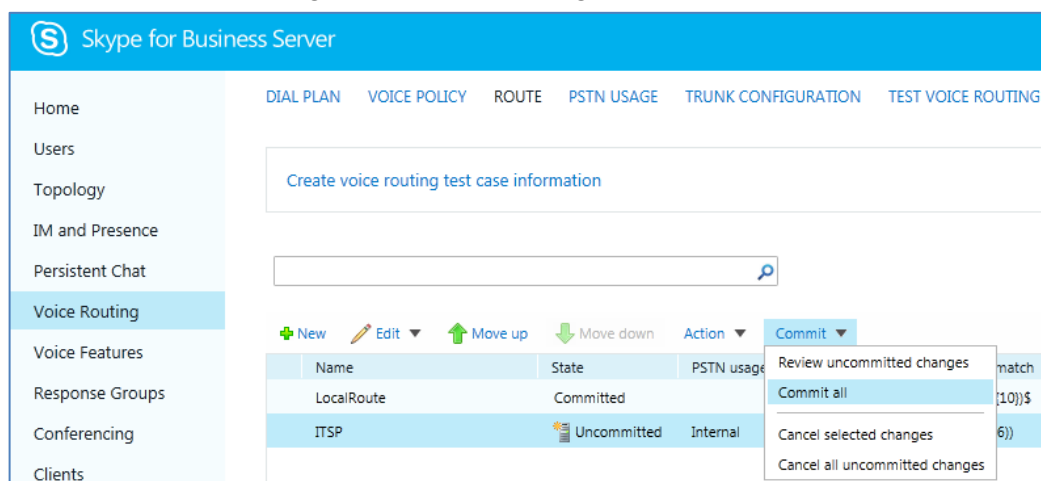
Search

+ New Edit Move up Move down Action Commit

Name	State	PSTN usage	Pattern to match
LocalRoute	Committed		^\+1[0-9]{10}\$
ITSP	Uncommitted	Internal	^\+66{6}\$

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

Figure 3-24: Committing Voice Routes



Skype for Business Server

Home Users Topology IM and Presence Persistent Chat **Voice Routing** Voice Features Response Groups Clients

DIAL PLAN VOICE POLICY **ROUTE** PSTN USAGE TRUNK CONFIGURATION TEST VOICE ROUTING

Create voice routing test case information

Search

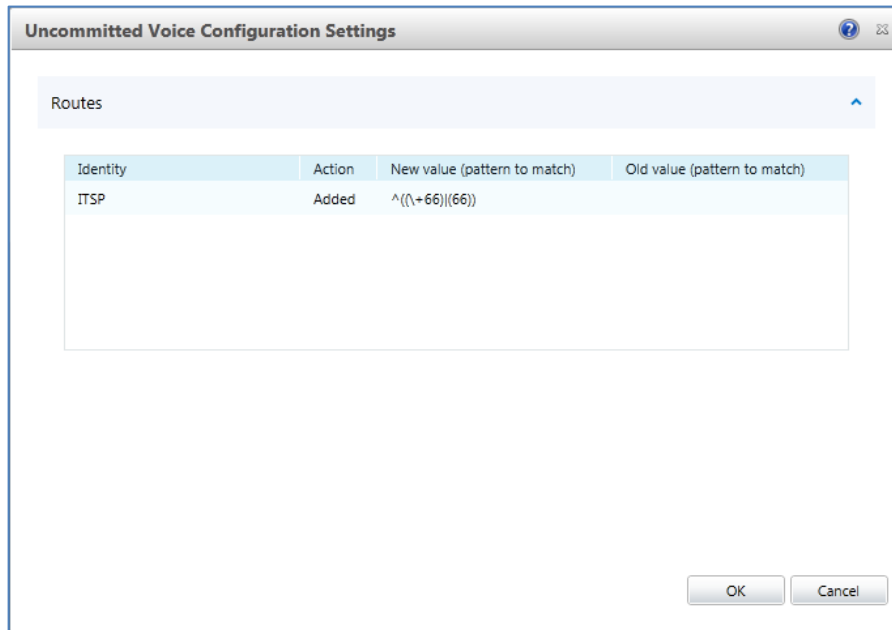
+ New Edit Move up Move down Action Commit

Name	State	PSTN usage	Pattern to match
LocalRoute	Committed		^\+1[0-9]{10}\$
ITSP	Uncommitted	Internal	^\+66{6}\$

Review uncommitted changes
Commit all
Cancel selected changes
Cancel all uncommitted changes

The Uncommitted Voice Configuration Settings page appears:

Figure 3-25: Uncommitted Voice Configuration Settings



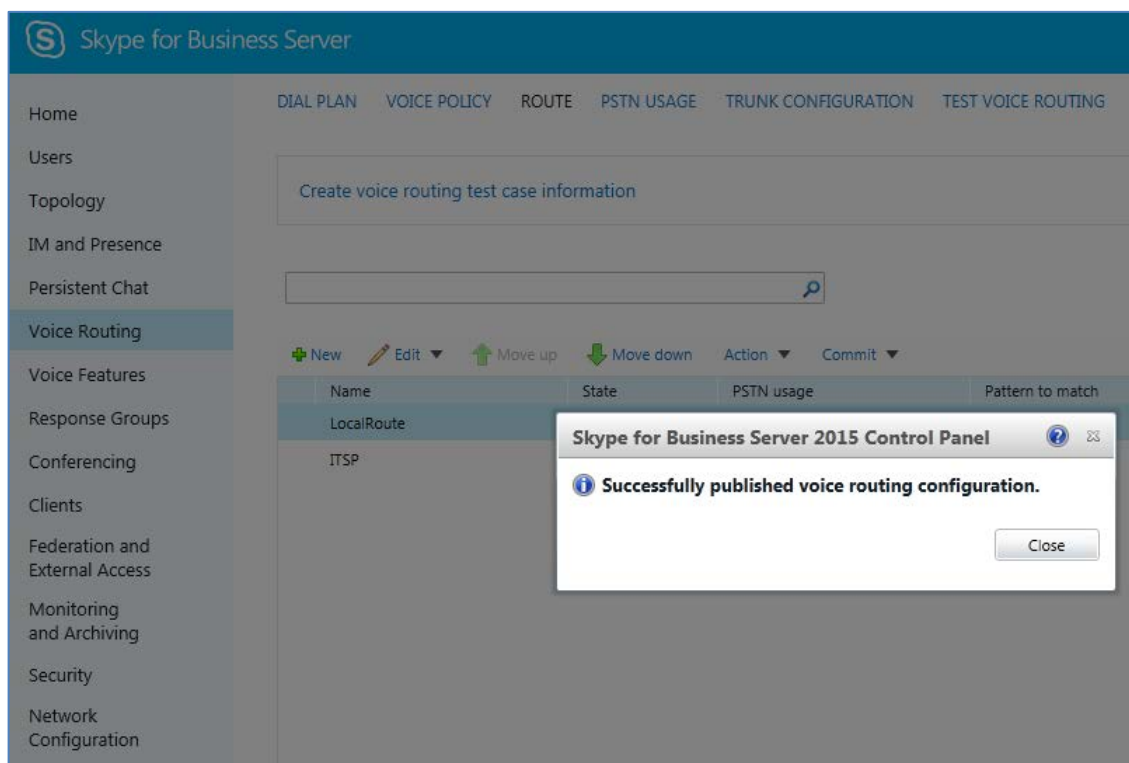
The dialog box titled "Uncommitted Voice Configuration Settings" contains a section labeled "Routes". Inside this section is a table with the following data:

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

At the bottom right of the dialog are "OK" and "Cancel" buttons.

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 Skype for Business Server interface. The left sidebar lists navigation options: Home, Users, Topology, IM and Presence, Persistent Chat, Voice Routing (selected), Voice Features, Response Groups, Conferencing, Clients, Federation and External Access, Monitoring and Archiving, Security, and Network Configuration. The top navigation bar includes tabs: DIAL PLAN, VOICE POLICY, ROUTE, PSTN USAGE, TRUNK CONFIGURATION, and TEST VOICE ROUTING. The main content 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 Skype for Business Server interface with the Trunk Configuration tab selected. The left sidebar is the same as in Figure 3-27. The top navigation bar includes tabs: DIAL PLAN, VOICE POLICY, ROUTE, PSTN USAGE, TRUNK CONFIGURATION (selected), and TEST VOICE ROUTING. The main content 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:

The screenshot shows the 'Skype for Business Server' management interface. The left sidebar lists various configuration areas, with 'Voice Routing' selected. The main pane displays the 'New Trunk Configuration' dialog for 'PstnGateway:ITSP.S4B.interop'. The dialog includes fields for 'Name' (PstnGateway:ITSP.S4B.interop), 'Description', 'Maximum early dialogs supported' (20), 'Encryption support level' (Required), and 'Refer support' (Enable sending refer to the gateway). Below these are several checkboxes: 'Enable media bypass' (checked), 'Centralized media processing' (checked), 'Enable RTP latching' (unchecked), 'Enable forward call history' (checked), 'Enable forward P-Asserted-Identity data' (unchecked), and 'Enable outbound routing failover timer' (checked).

- 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.

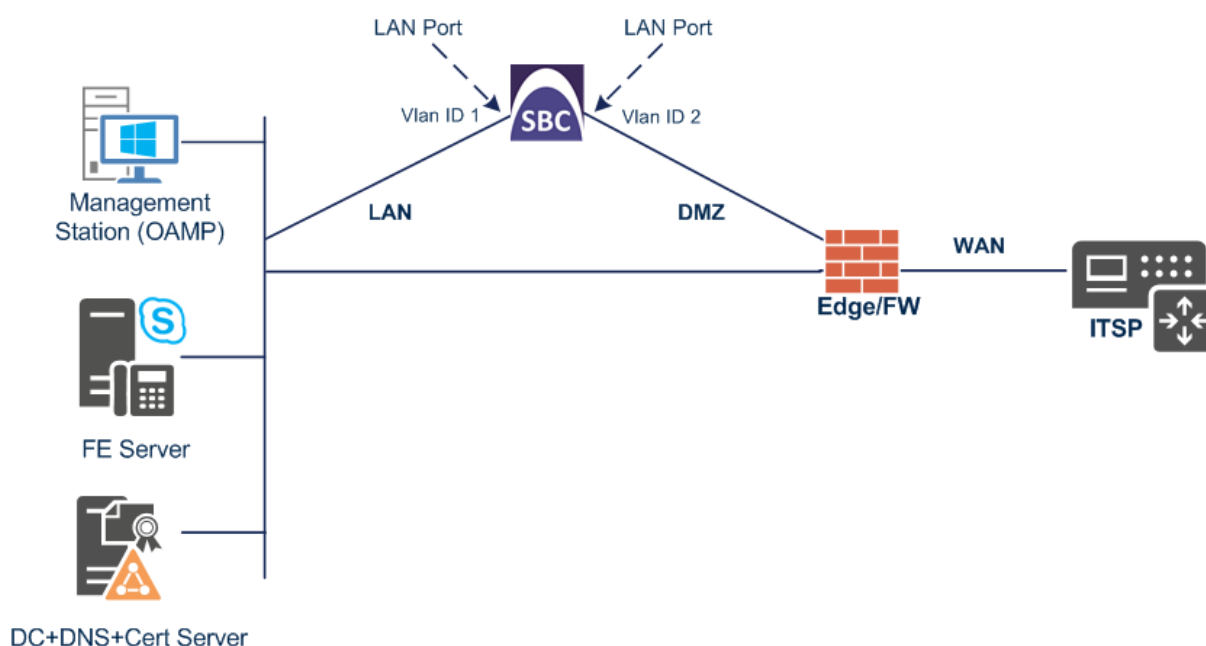
- The scope of this interoperability test and document does **not** cover all security aspects for configuring this topology. Comprehensive security measures should be implemented per your organization's security policies. For security recommendations on AudioCodes' products, refer to the *Recommended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

Ethernet Devices (2)				
+ New Edit 🗑️ Page 1 of 1 Show 10 records per page				
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:

- Click **New**.
- 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)									
+ New		Edit							
				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

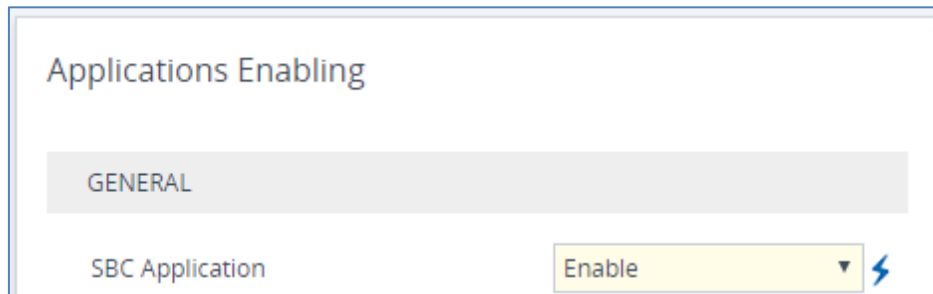
4.2 Step 2: Enable the SBC Application

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

Name

• MRLan

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

QoE Profile

-- View

Bandwidth Profile

-- View

Cancel

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

Index: 1

Name: MRWan

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

QUALITY OF EXPERIENCE

QoE Profile: -- [View](#)

Bandwidth Profile: -- [View](#)

Cancel **APPLY**

The configured Media Realms are shown in the figure below:

Figure 4-7: Configured Media Realms in Media Realm Table

Media Realms (2)

+ New

Edit

<<

>>

Page 1 of 1

Show 10 records per page

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 1 of 1

Show 10 records per page

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:

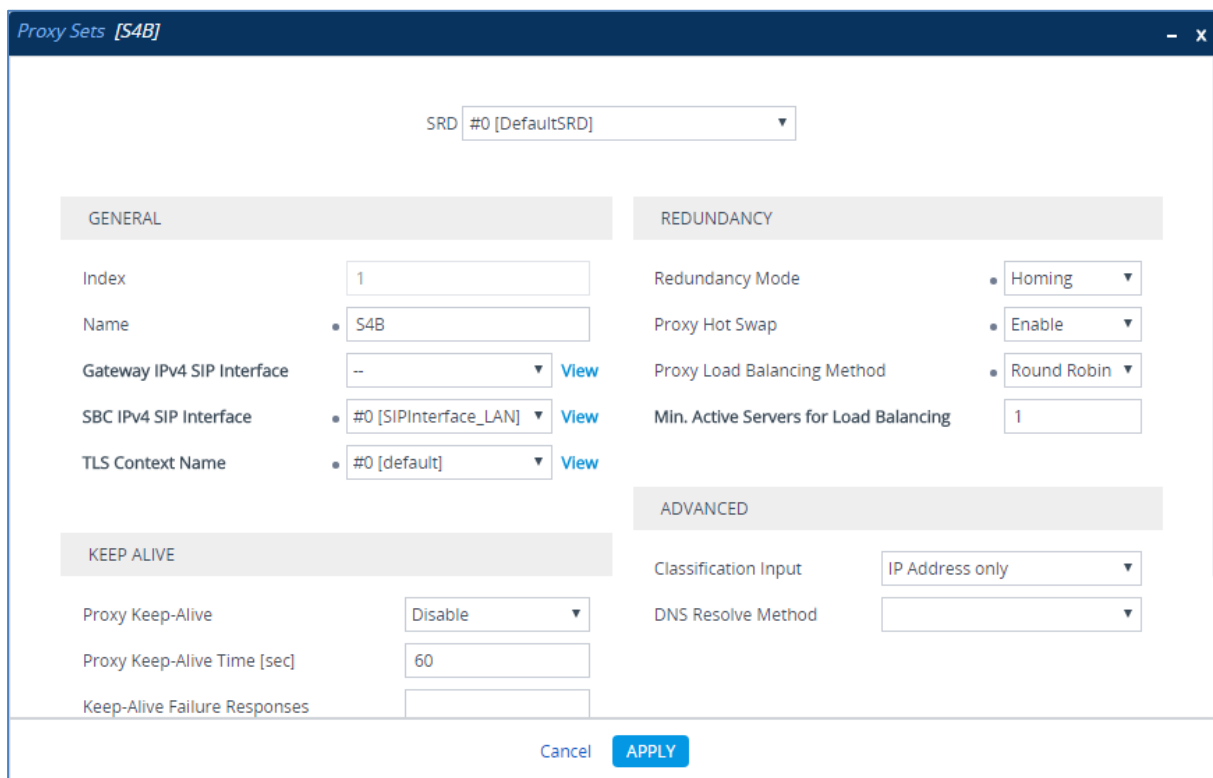
- Microsoft Skype for Business Server 2015
- BroadCloud SIP Trunk

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


Proxy Sets [S4B]

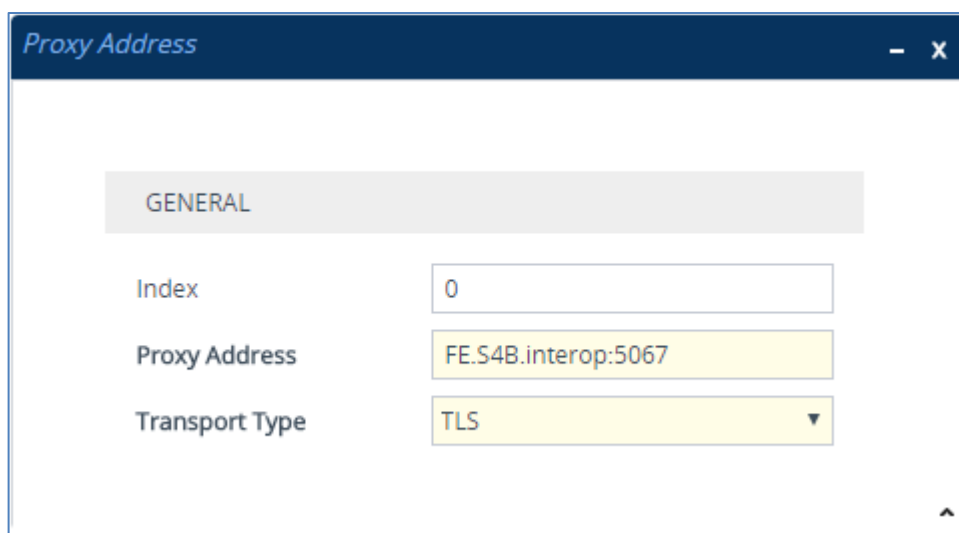
SRD #0 [DefaultSRD]

GENERAL		REDUNDANCY	
Index	1	Redundancy Mode	Homing
Name	S4B	Proxy Hot Swap	Enable
Gateway IPv4 SIP Interface	-- View	Proxy Load Balancing Method	Round Robin
SBC IPv4 SIP Interface	#0 [SIPInterface_LAN] View	Min. Active Servers for Load Balancing	1
TLS Context Name	#0 [default] View		

KEEP ALIVE		ADVANCED	
Proxy Keep-Alive	Disable	Classification Input	IP Address only
Proxy Keep-Alive Time [sec]	60	DNS Resolve Method	
Keep-Alive Failure Responses			

Cancel APPLY

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

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


Proxy Address

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

- 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

Proxy Sets [BroadCloud]

SRD: #0 [DefaultSRD]

GENERAL

Index: 2

Name: BroadCloud

Gateway IPv4 SIP Interface: -- [View](#)

SBC IPv4 SIP Interface: #1 [SIPInterface_WAN] [View](#)

TLS Context Name: #0 [default] [View](#)

REDUNDANCY

Redundancy Mode: Homing

Proxy Hot Swap: Enable

Proxy Load Balancing Method: Disable

Min. Active Servers for Load Balancing: 1

KEEP ALIVE

Proxy Keep-Alive: Using OPTIONS

Proxy Keep-Alive Time [sec]: 60

Keep-Alive Failure Responses:

ADVANCED

Classification Input: IP Address only

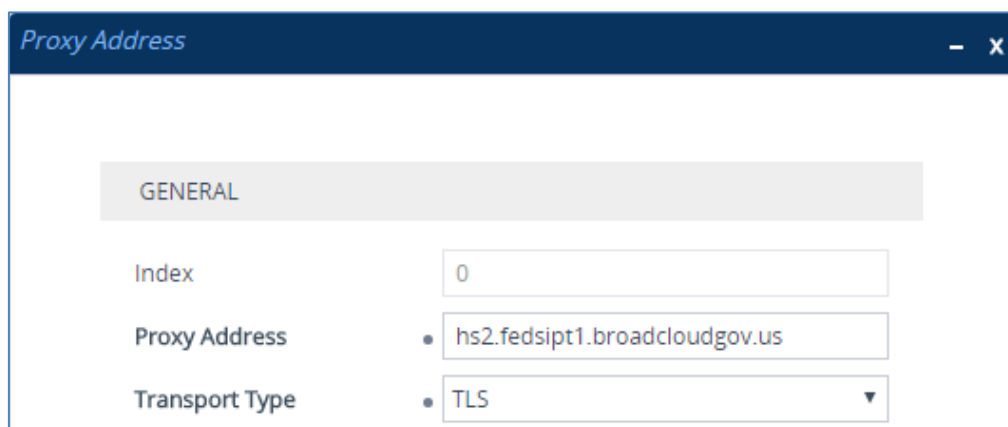
DNS Resolve Method: SRV

Cancel APPLY

- 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



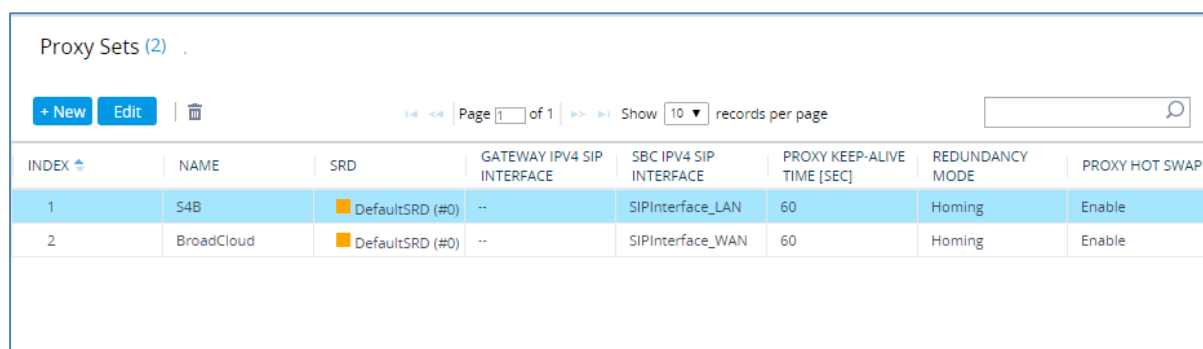
The dialog box titled "Proxy Address" has a "GENERAL" tab. It contains three fields: "Index" with the value "0", "Proxy Address" with the value "hs2.fedsipt1.broadcloudgov.us", and "Transport Type" with a dropdown menu showing "TLS".

- 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



The table shows two proxy sets. The first set is named "S4B" and the second is named "BroadCloud". Both sets are configured with "DefaultSRD (#0)" as the SRD, "SIPInterface_LAN" as the SBC IPv4 SIP Interface, and "SIPInterface_WAN" as the Gateway IPv4 SIP Interface. Both sets have a "Proxy Keep-Alive Time" of 60 seconds, "Homing" as the Redundancy Mode, and "Enable" as the Proxy Hot Swap.

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 assign 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	▪ 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 Groups

Coder Group Name: 0 : AudioCodersGroups_0 ▼ Delete Group

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	▪ G.729 ▪ G.711 A-law ▪ G.711 U-law

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

Coder Groups

Coder Group Name: 1 : AudioCodersGroups_1 ▼ Delete Group

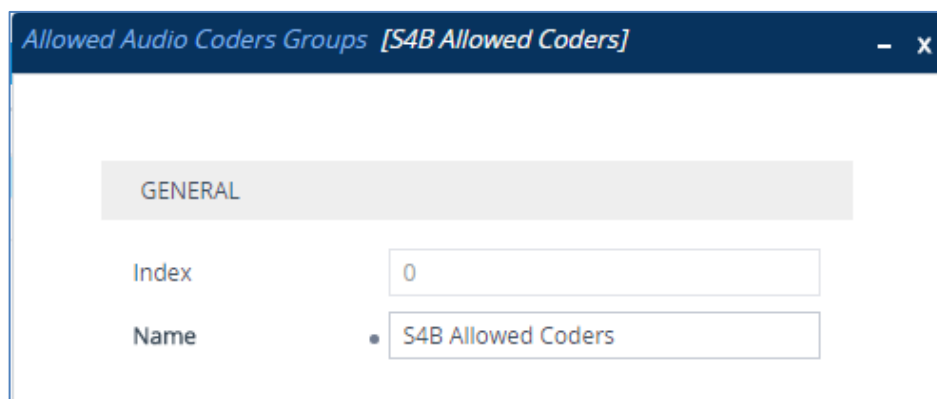
Coder Name	Packetization Time	Rate	Payload Type	Silence Suppression	Coder Specific
G.729 ▼	20 ▼	8 ▼	18	Disabled ▼	
G.711A-law ▼	20 ▼	64 ▼	8	Disabled ▼	
G.711U-law ▼	20 ▼	64 ▼	0	Disabled ▼	
▼	▼	▼		▼	

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 assign 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.

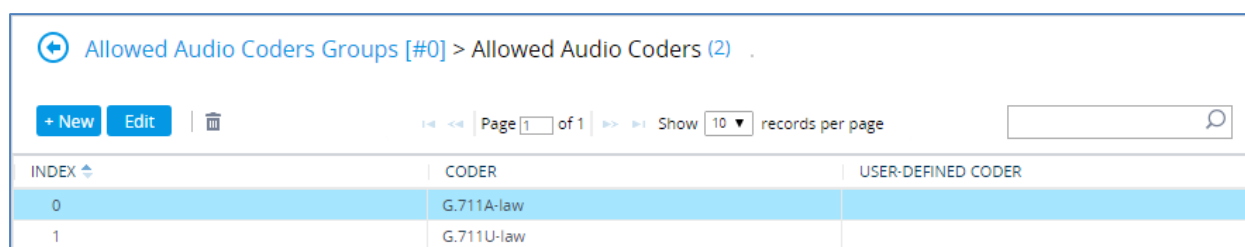
Figure 4-16: Configuring Allowed Coders Group for Skype for Business Server 2015



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

The screenshot shows a configuration window titled "Allowed Audio Coders Groups [BroadCloud Allowed Coders]". It has a "GENERAL" tab. The "Index" field is set to "1". The "Name" field is set to "BroadCloud Allowed Coders".

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

The screenshot shows the "Allowed Audio Coders Groups [#1] > Allowed Audio Coders (3)" page. It includes a table with 3 columns: INDEX, CODER, and USER-DEFINED CODER. The table contains 3 rows of data.

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

GENERAL		ROBUSTNESS	
NAT Traversal	Disable NAT	New RTP Stream Packets	3
Enable Continuity Tones	Disable	New RTCP Stream Packets	3
Inbound Media Latch Mode	Dynamic	New SRTP Stream Packets	3
Number of Media Channels	0	New SRTCP Stream Packets	3
Enforce Media Order	Disable	Timeout To Relatch RTP (msec)	200
SDP Session Owner	AudiocodesGW	Timeout To Relatch SRTP (msec)	200
		Timeout To Relatch Silence (msec)	10000
		Timeout To Relatch RTCP (msec)	10000
SBC SETTINGS			
Preferences Mode	• Include Extensions		
Enforce Media Order	Disable		
GATEWAY SETTINGS			
Enable Early Media	Disable		
Multiple Packetization Time Format	None		

Cancel
APPLY

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

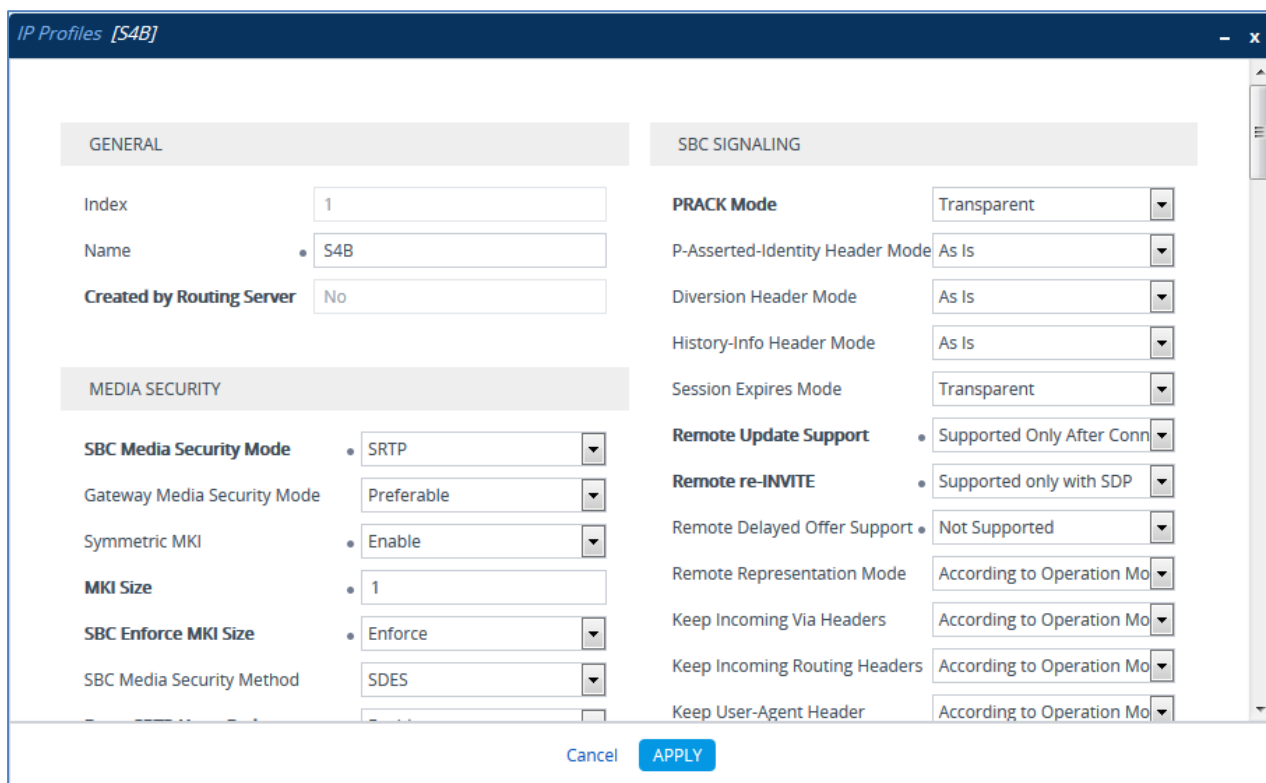
➤ **To configure IP Profile for the Skype for Business Server 2015:**

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

Index: 1

Name: S4B

Created by Routing Server: No

MEDIA SECURITY

SBC Media Security Mode: SRTP

Gateway Media Security Mode: Preferable

Symmetric MKI: Enable

MKI Size: 1

SBC Enforce MKI Size: Enforce

SBC Media Security Method: SDS

SBC SIGNALING

PRACK Mode: Transparent

P-Asserted-Identity Header Mode: As Is

Diversion Header Mode: As Is

History-Info Header Mode: As Is

Session Expires Mode: Transparent

Remote Update Support: Supported Only After Conn

Remote re-INVITE: Supported only with SDP

Remote Delayed Offer Support: Not Supported

Remote Representation Mode: According to Operation Mo

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

IP Profiles [BroadCloud]
— x

GENERAL

Index

Name •

Created by Routing Server

SBC SIGNALING

PRACK Mode

P-Asserted-Identity Header Mode •

Diversion Header Mode

History-Info Header Mode

Session Expires Mode

Remote Update Support

Remote re-INVITE

Remote Delayed Offer Support

Remote Representation Mode

Keep Incoming Via Headers

Keep Incoming Routing Headers

Keep User-Agent Header

MEDIA SECURITY

SBC Media Security Mode •

Gateway Media Security Mode

Symmetric MKI •

MKI Size

SBC Enforce MKI Size •

SBC Media Security Method

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

Page 1 of 1
Show 10 records per page

INDEX	NAME	SRD	TYPE	SBC OPERATION MODE	PROXY SET	IP PROFILE	MEDIA REALM	SIP GROUP NAME	CLASSIFY BY PROXY SET	INBOUND MESSAGE MANIPULAT SET	OUTBOUND MESSAGE MANIPULAT SET
1	S4B	Default	Server	Not Configu	S4B	S4B	MRLan	interop.adpt	Enable	-1	-1
2	BroadCloud	Default	Server	Not Configu	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.

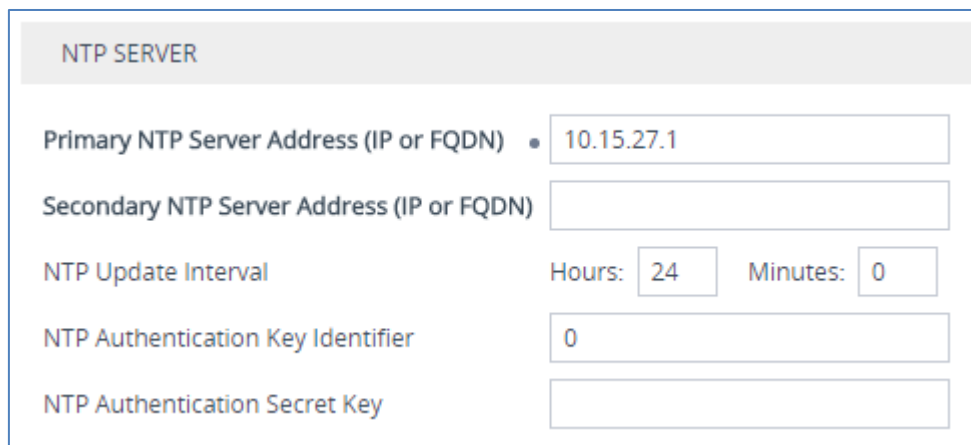
4.9.1 Step 9a: Configure the NTP Server Address

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



The screenshot shows the 'NTP SERVER' configuration page. It contains the following fields and controls:

- Primary NTP Server Address (IP or FQDN)**: A text input field containing '10.15.27.1'.
- Secondary NTP Server Address (IP or FQDN)**: An empty text input field.
- NTP Update Interval**: A section with two sub-fields: 'Hours' (set to 24) and 'Minutes' (set to 0).
- NTP Authentication Key Identifier**: A text input field containing '0'.
- NTP Authentication Secret Key**: An empty text input field.

3. Click **Apply**.

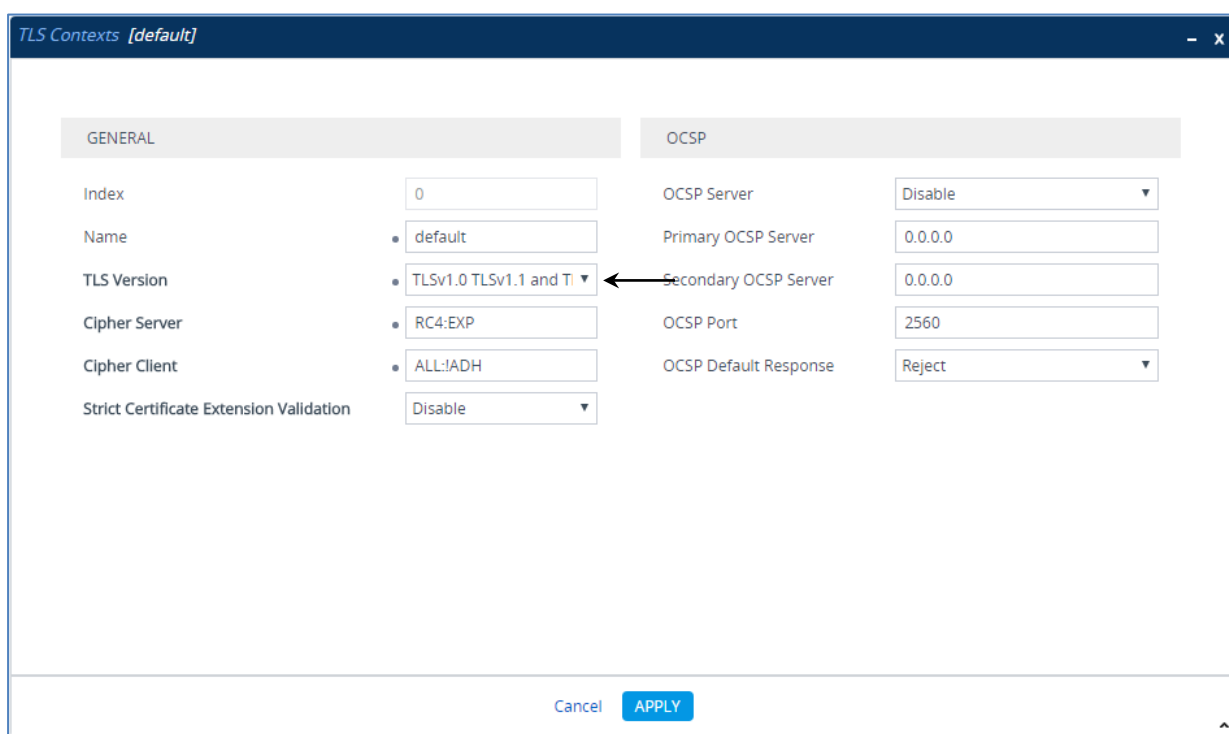
4.9.2 Step 9b: Configure the TLS version

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		

Cancel APPLY

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


TLS Context [#0] > Context Certificates

CERTIFICATE SIGNING REQUEST

Subject Name [CN]

ITSP.S4B.interop

Organizational Unit [OU] (optional)

Company name [O] (optional)

Locality or city name [L] (optional)

State [ST] (optional)

Country code [C] (optional)

Signature Algorithm

SHA-1 ▼

Create 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-----
MIIBWjCBxAlBADAAbMRkwFwYDVQDDBBJVFNQ1M0Q15pbmR1cm9wMIGfMA0GCSqG
SIb3DQEBAQUAA4GNADCBiQKBgQCzEs8XTnY8be/t77eEDG7rTg747GQ3ODfOC4Rs
x+e9KfbErZgxMYqGT8u04AU0wU9LUPkkq+8gI6w2bg3boW0kg/9hrnNL2rf1tGcn
30oShP05PiKmRNZnCC090b03tbr9kuHmlwPRQ7yT6k7xS3X8bSigqT4LQbJBT1tt
hDH3bQIDAQABAAwDQYJKoZIhvcNAQEFBQADgYEAim/GA2E1ZQbZaR6CZyIawilT
u65w450NFHmaC1uHsyZ8keM8d1Ux14hkw7t5ygAD8KbxVkhRvACgcQrAK2v8u1Pf
TvN+bwJ+kQ0d59CiXa82e0o1wB3buPq5+qWdGTF+MyJWGVf8SiC1c6+zFoc+BEZY
7tQ8y0J8od0aDhStDfQ=
-----END CERTIFICATE REQUEST-----

```

- Copy the CSR from the line "**-----BEGIN CERTIFICATE REQUEST-----**" 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*.
- Open a Web browser and navigate to the Microsoft Certificates Services Web site at <http://<certificate server>/CertSrv>.

Figure 4-27: Microsoft Certificate Services Web Page

Microsoft Certificate Services -- Demolab

Home

Welcome

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

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

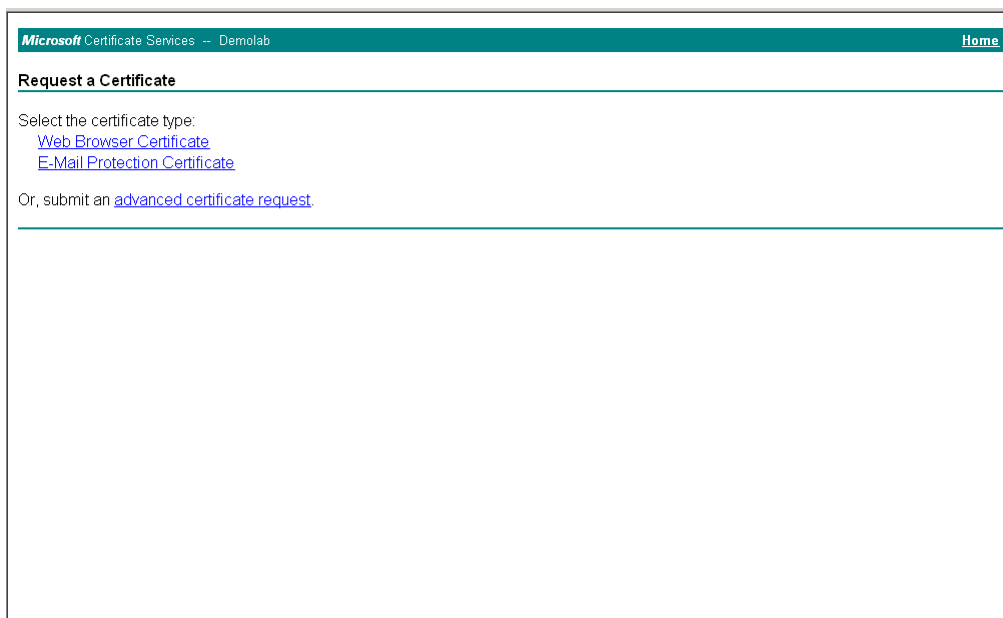
For more information about Certificate Services, see [Certificate Services Documentation](#).

Select a task:

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

- Click **Request a certificate**.

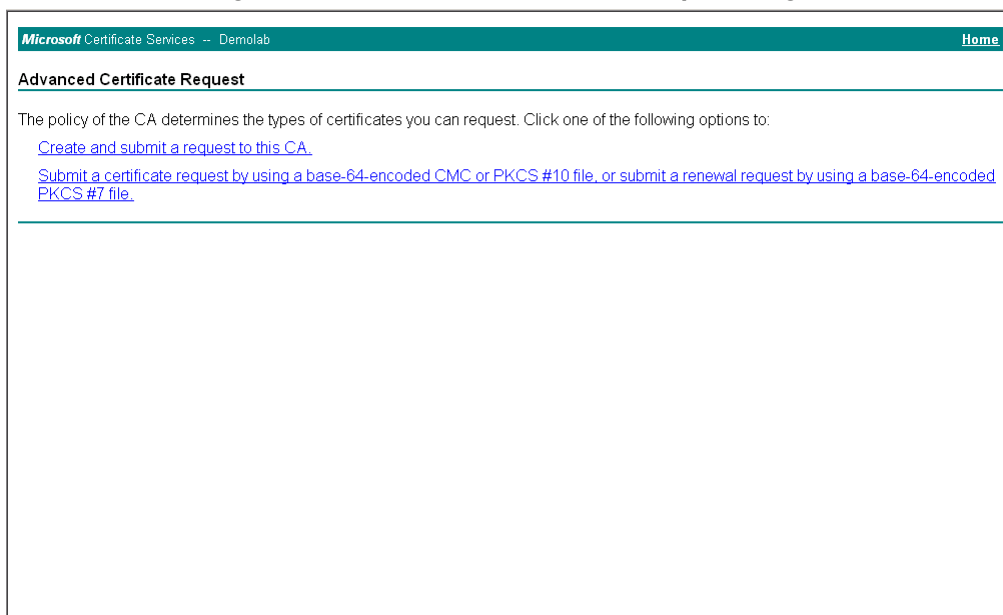
Figure 4-28: Request a Certificate Page



The screenshot shows a web browser window with the title bar 'Microsoft Certificate Services -- Demolab'. The address bar shows 'Home'. The main content area is titled 'Request a Certificate'. It contains the text 'Select the certificate type:' followed by two links: 'Web Browser Certificate' and 'E-Mail Protection Certificate'. Below this, it says 'Or, submit an [advanced certificate request](#)'.

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

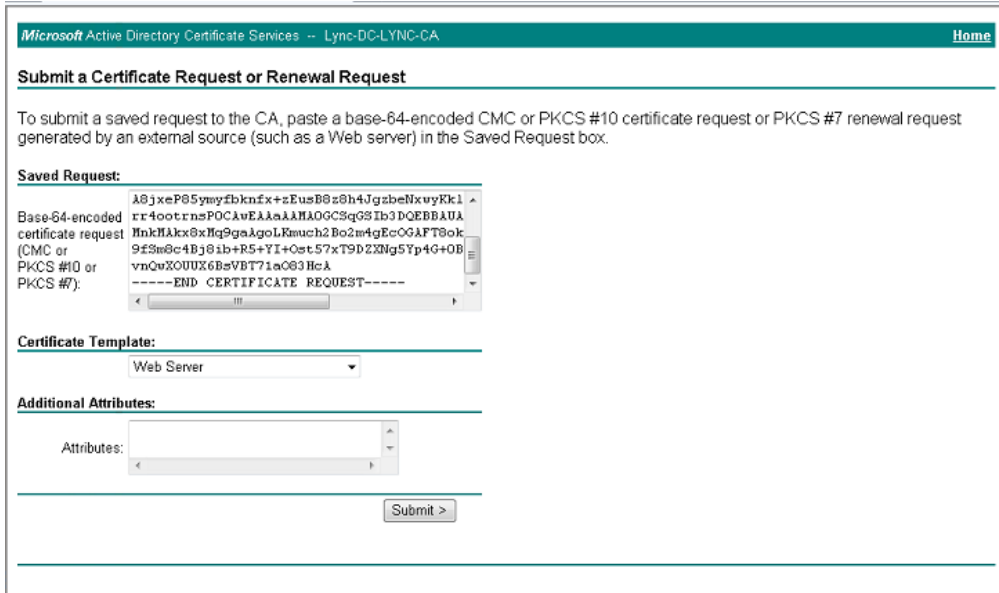
Figure 4-29: Advanced Certificate Request Page



The screenshot shows a web browser window with the title bar 'Microsoft Certificate Services -- Demolab'. The address bar shows 'Home'. The main content area is titled 'Advanced Certificate Request'. It contains the text 'The policy of the CA determines the types of certificates you can request. Click one of the following options to:' followed by two links: 'Create and submit a request to this CA.' and 'Submit a certificate request by using a base-64-encoded CMC or PKCS #10 file, or submit a renewal request by using a base-64-encoded PKCS #7 file.'

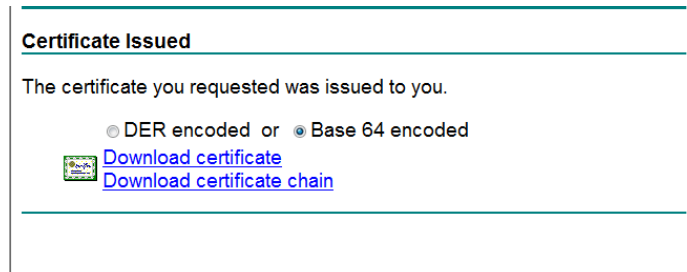
8. Click **Submit a certificate request ...**, and then click **Next**.

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



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

Figure 4-31: Certificate Issued Page



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

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

Microsoft Certificate Services -- Demolab [Home](#)

Download a CA Certificate, Certificate Chain, or CRL

To trust certificates issued from this certification authority, [install this CA certificate chain](#).

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

CA certificate:

Current [Demolab]

Encoding method:

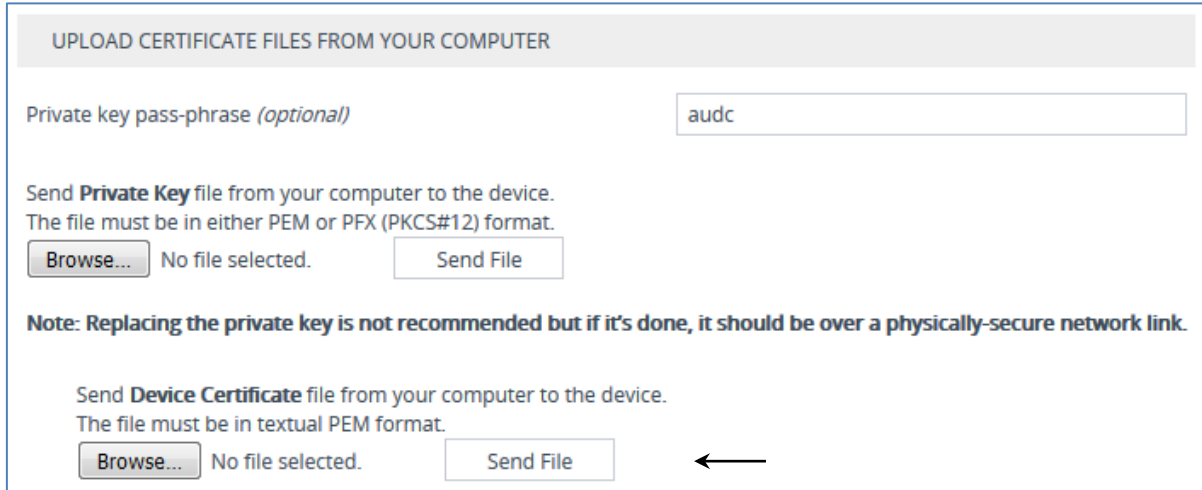
☒ DER
☐ Base 64

[Download CA certificate](#)
[Download CA certificate chain](#)
[Download latest base CRL](#)

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:
 - a. 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.
 - b. Scroll down to the **Upload certificates files from your computer** group, click the **Browse** button corresponding to the 'Send Device Certificate...' field, navigate to the *gateway.cer* certificate file that you saved on your computer in Step 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.

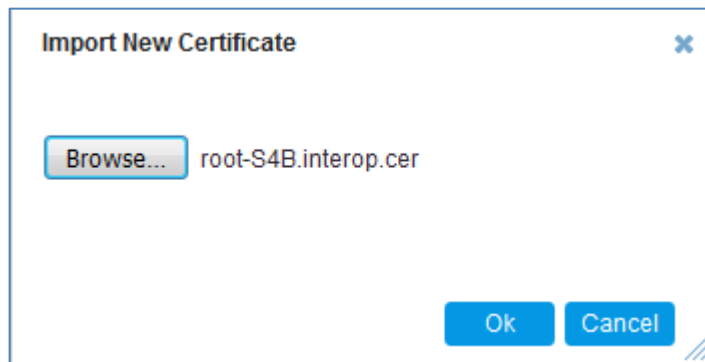
Note: Replacing the private key is not recommended but if it's done, it should be over a physically-secure network link.

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.
 - a. 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.
 - b. Click the **Import** button, and then select the certificate file to load.

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



Import New Certificate

root-S4B.interop.cer

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

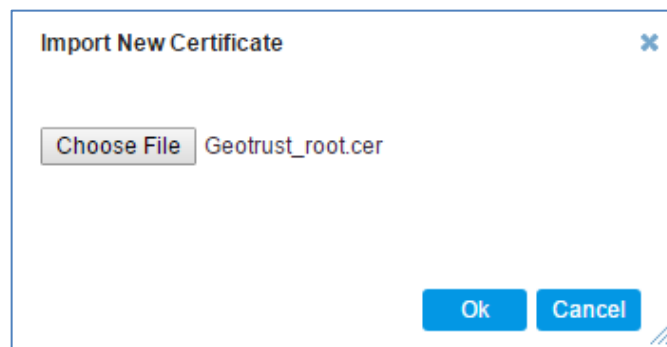
The procedure involves the following main steps:

- 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).

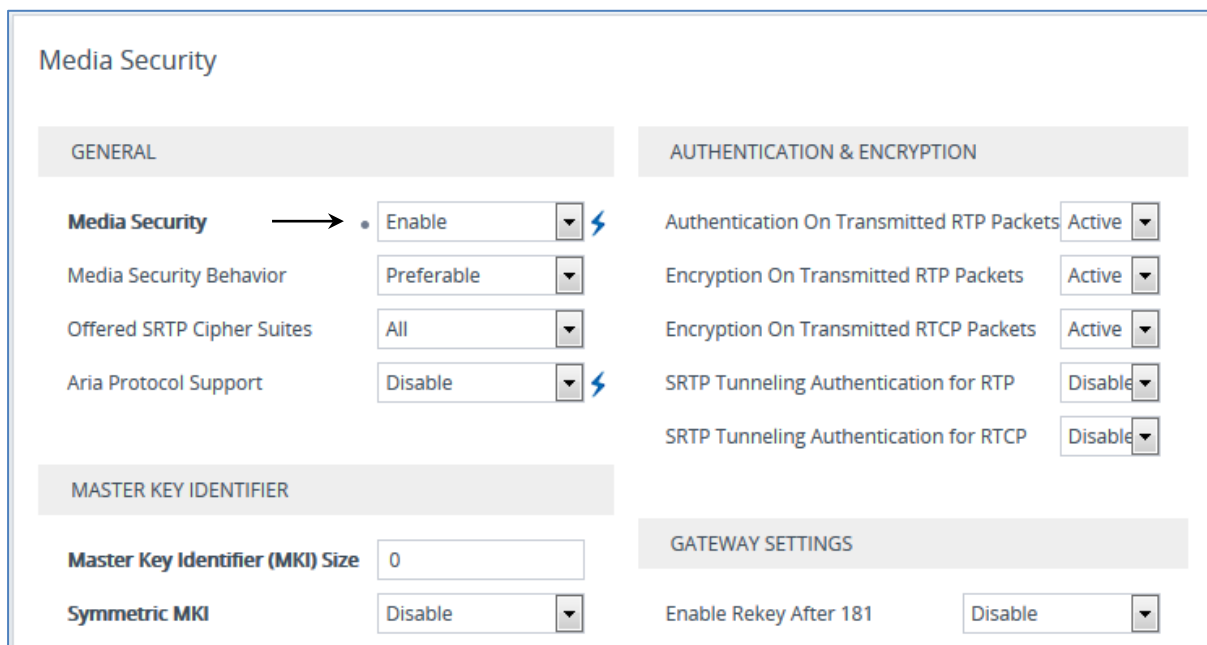
4.10 Step 10: Configure SRTP

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



Media Security	
GENERAL	
Media Security	Enable
Media Security Behavior	Preferable
Offered SRTP Cipher Suites	All
Aria Protocol Support	Disable
MASTER KEY IDENTIFIER	
Master Key Identifier (MKI) Size	0
Symmetric MKI	Disable
AUTHENTICATION & ENCRYPTION	
Authentication On Transmitted RTP Packets	Active
Encryption On Transmitted RTP Packets	Active
Encryption On Transmitted RTCP Packets	Active
SRTP Tunneling Authentication for RTP	Disable
SRTP Tunneling Authentication for RTCP	Disable
GATEWAY SETTINGS	
Enable Rekey After 181	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' page with the 'GENERAL' tab active. The settings are as follows:

Setting	Value
NAT Traversal	Disable NAT
Enable Continuity Tones	Disable
Inbound Media Latch Mode	Dynamic
Number of Media Channels	100
Enforce Media Order	Disable
SDP Session Owner	AudiocodesGW

A lightning bolt icon is next to the 'Number of Media Channels' field, and an arrow points to it from the right.

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

The screenshot shows the 'IP-to-IP Routing [Terminate OPTIONS]' configuration window. At the top, the 'Routing Policy' is set to '#0 [Default_SBCRoutingPolicy]'. The window is divided into two main sections: 'GENERAL' and 'ACTION'.

GENERAL Section:

- Index:** 0
- Name:** Terminate OPTIONS
- Alternative Route Options:** Route Row
- MATCH Section:**
 - Source IP Group:** Any
 - Request Type:** OPTIONS
 - Source Username Prefix:** *
 - Source Host:** *
 - Source Tags:** (empty)

ACTION Section:

- Destination Type:** Dest Address
- Destination IP Group:** --
- Destination SIP Interface:** --
- Destination Address:** internal
- Destination Port:** 0
- Destination Transport Type:** (empty)
- Call Setup Rules Set ID:** -1
- Group Policy:** Sequential
- Cost Group:** --

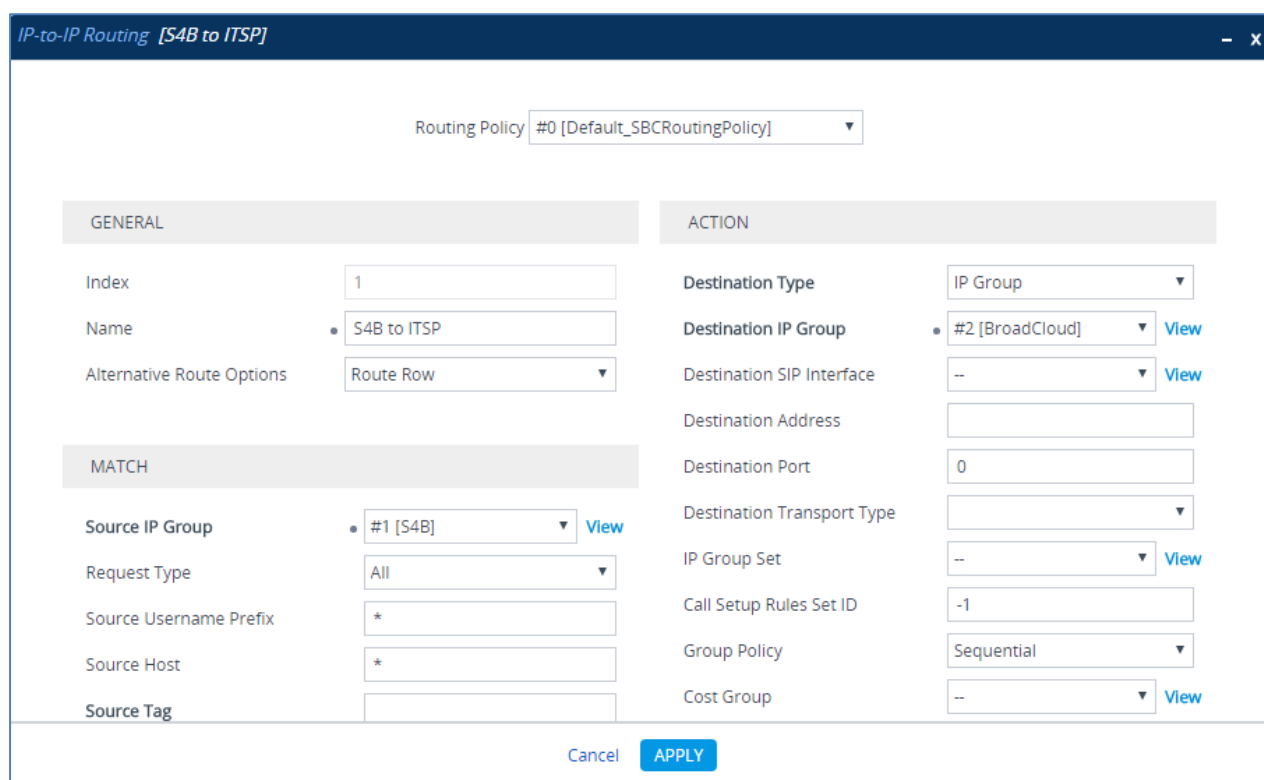
At the bottom of the window, there are 'Cancel' and 'APPLY' buttons.

- 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

Index: 1

Name: S4B to ITSP

Alternative Route Options: Route Row

MATCH

Source IP Group: #1 [S4B] View

Request Type: All

Source Username Prefix: *

Source Host: *

Source Tag:

ACTION

Destination Type: IP Group

Destination IP Group: #2 [BroadCloud] View

Destination SIP Interface: -- View

Destination Address:

Destination Port: 0

Destination Transport Type:

IP Group Set: -- View

Call Setup Rules Set ID: -1

Group Policy: Sequential

Cost Group: -- View

Cancel 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

IP-to-IP Routing [ITSP to S4B]

Routing Policy: #0 [Default_SBCRoutingPolicy]

GENERAL	ACTION
Index: 2	Destination Type: IP Group
Name: ITSP to S4B	Destination IP Group: #1 [S4B] View
Alternative Route Options: Route Row	Destination SIP Interface: -- View
	Destination Address:
	Destination Port: 0
	Destination Transport Type:
	IP Group Set: -- View
	Call Setup Rules Set ID: -1
	Group Policy: Sequential
	Cost Group: -- View

Cancel [APPLY](#)

- b. Click **Apply**.

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



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

4.13 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

Outbound Manipulations [Add + toward S4B]

Routing Policy #0 [Default_SBCRoutingPolicy]

GENERAL

Index 0
Name Add + toward S4B
Additional Manipulation No
Call Trigger Any

ACTION

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

MATCH

Request Type All
Source IP Group #2 [BroadCloud] View
Destination IP Group #1 [S4B] View
Source Username Prefix *

Cancel APPLY

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

Page 1 of 1 Show 10 records per page

INDEX	NAME	ROUTING POLICY	ADDITION MANIPUL	SOURCE IP GROUP	DESTINAT IP GROUP	SOURCE USERNAN 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_S	No	BroadClou	S4B	*	*	Destinatio	0	0	255	+	
1	Change +	Default_S	No	S4B	BroadClou	*	+	Destinatio	1	0	255	011	
2	Remove +	Default_S	No	S4B	BroadClou	+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)

Message Manipulations [Call Forward]

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		

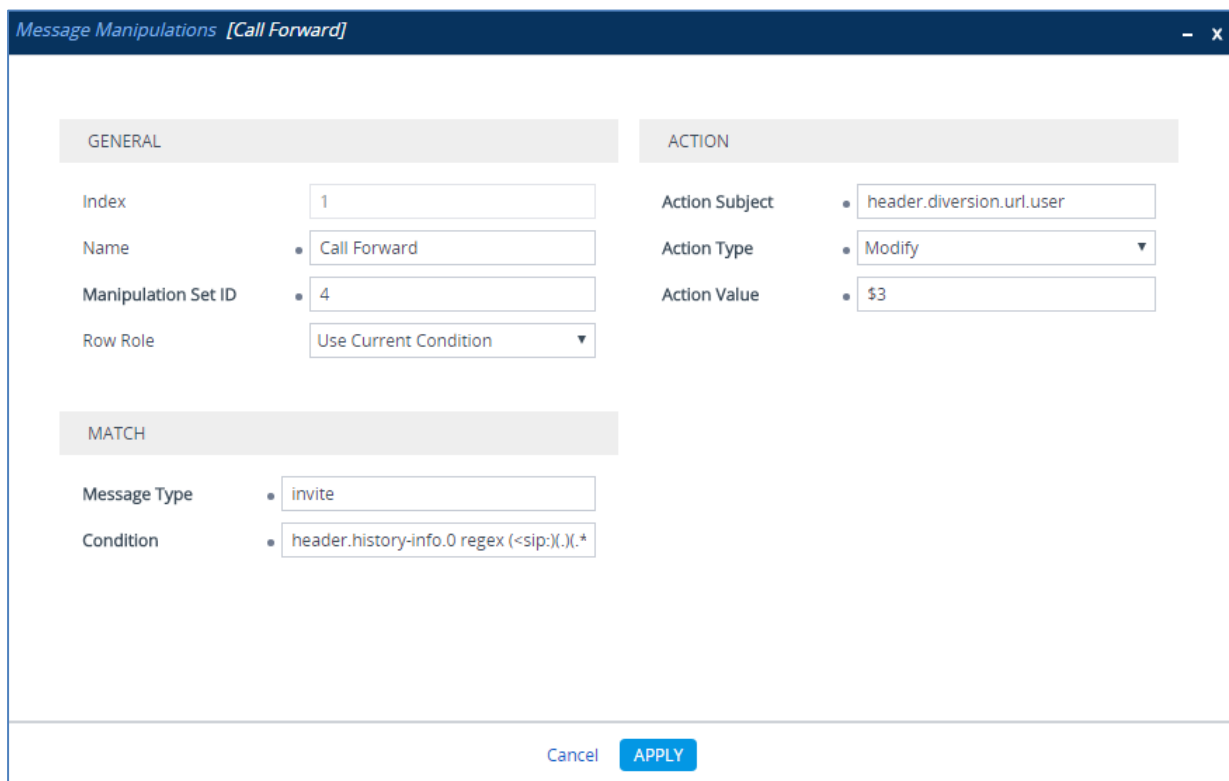
MATCH	
Message Type	invite
Condition	header.history-info exists

Cancel 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)



The screenshot shows the 'Message Manipulations' configuration window for a rule named 'Call Forward'. The window is divided into three main sections: GENERAL, ACTION, and MATCH.

GENERAL Section:

- Index: 1
- Name: Call Forward
- Manipulation Set ID: 4
- Row Role: Use Current Condition

ACTION Section:

- Action Subject: header.diversion.url.user
- Action Type: Modify
- Action Value: \$3

MATCH Section:

- Message Type: invite
- Condition: header.history-info.0 regex (<sip:)(.)(.*)(@)(.*)

At the bottom of the window, there are 'Cancel' and 'APPLY' buttons.

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

Index2

Name• Call Forward

Manipulation Set ID• 4

Row Role• Use Previous Condition ▼

ACTION

Action Subject• header.history-info

Action Type• Remove ▼

Action Value

MATCH

Message Type

Condition

Cancel

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]

GENERAL

Index

3

Name

Call Transfer

Manipulation Set ID

4

Row Role

Use Current Condition ▼

ACTION

Action Subject

header.referred-by.url.host

Action Type

Modify ▼

Action Value

header.from.url.host

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

Index
Name •
Manipulation Set ID •
Row Role

ACTION

Action Subject •
Action Type •
Action Value •

MATCH

Message Type •
Condition •

Cancel

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

Index
Name •
Manipulation Set ID •
Row Role

ACTION

Action Subject •
Action Type •
Action Value •

MATCH

Message Type •
Condition •

Cancel

APPLY

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)

Message Manipulations [Reject Responses]

GENERAL

Index: 6

Name: Reject Responses

Manipulation Set ID: 4

Row Role: Use Current Condition

ACTION

Action Subject: header.request-uri.methodtype

Action Type: Modify

Action Value: '480'

MATCH

Message Type: any.response

Condition: header.request-uri.methodtype=='503'

Cancel APPLY

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

Message Manipulations (7)

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

INDEX	NAME	MANIPULATION SET ID	MESSAGE TYPE	CONDITION	ACTION SUBJECT	ACTION TYPE	ACTION VALUE	ROW ROLE
0	Call Forward	4	invite	header.history-info	header.diversion	Add	header.from	Use Current Condition
1	Call Forward	4	invite	header.history-info	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	header.referred-by	Modify	header.from.url.host	Use Current Condition
4	Call Transfer	4	invite	header.referred-by	header.referred-by	Remove Prefix	'+'	Use Current Condition
5	Call Transfer	4	invite	header.referred-by	header.from.url.user	Modify	header.referred-by	Use Current Condition
6	Reject Responses	4	any.response	header.request-uri	header.request-uri	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:
 - a. Open the IP Groups table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **IP Groups**).
 - b. Select the row of the BroadCloud SIP trunk IP Group, and then click **Edit**.
 - c. Set the 'Outbound Message Manipulation Set' field to **4**.

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

IP Groups [BroadCloud]

SRD #0 [DefaultSRD]

GENERAL		QUALITY OF EXPERIENCE	
Index	2	QoE Profile	-- View
Name	BroadCloud	Bandwidth Profile	-- View
Topology Location	Up		
Type	Server		
Proxy Set	#2 [BroadCloud] View		
IP Profile	#2 [BroadCloud] View		
Media Realm	#1 [MRWan] View		
Contact User			
SIP Group Name	interop.adpt-tech.com		
Created By Routing Server	No		

MESSAGE MANIPULATION

Inbound Message Manipulation Set -1

Outbound Message Manipulation Set 4

Message Manipulation User-Defined String 1

Message Manipulation User-Defined String 2

SBC REGISTRATION AND AUTHENTICATION

Cancel APPLY

- d. 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 web interface titled "Accounts" with a close button (X) in the top right corner. Below the title bar, there is a "Served IP Group" dropdown menu set to "#1 [S4B]". The main configuration area is divided into two tabs: "GENERAL" and "CREDENTIALS".

GENERAL Tab:

- 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 Tab:

- User Name: 1234567890
- Password: *

At the bottom of the window, there are "Cancel" and "APPLY" buttons.

4. Click **Apply**.

4.16 Step 16: Miscellaneous Configuration

This section describes miscellaneous E-SBC configuration.

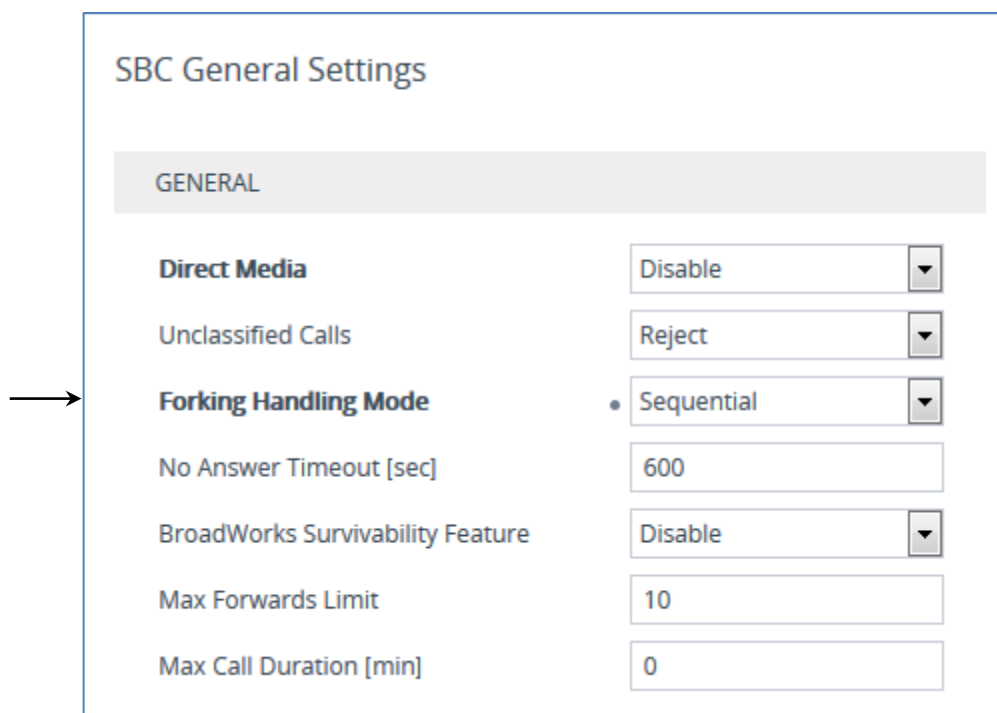
4.16.1 Step 16a: Configure Call Forking Mode

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



The screenshot shows the 'SBC General Settings' window. A horizontal arrow points to the 'Forking Handling Mode' dropdown menu, which is currently set to 'Sequential'. The other settings visible are: Direct Media (Disable), Unclassified Calls (Reject), No Answer Timeout [sec] (600), BroadWorks Survivability Feature (Disable), Max Forwards Limit (10), and Max Call Duration [min] (0).

SBC General Settings	
GENERAL	
Direct Media	Disable
Unclassified Calls	Reject
Forking Handling Mode	Sequential
No Answer Timeout [sec]	600
BroadWorks Survivability Feature	Disable
Max Forwards Limit	10
Max Call Duration [min]	0

3. Click **Apply**.

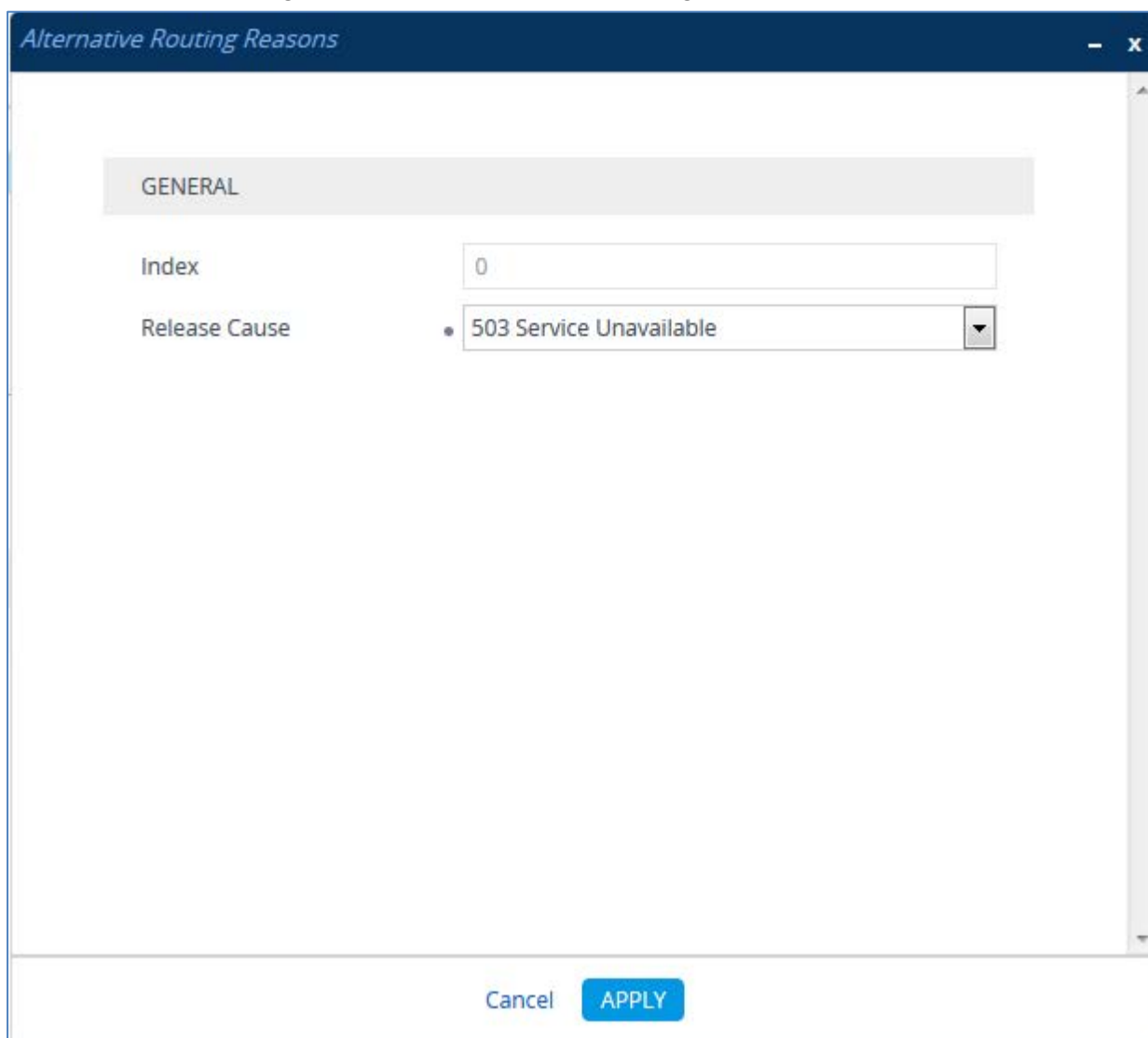
4.16.2 Step 16b: Configure SBC Alternative Routing Reasons

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



The screenshot shows a configuration window titled "Alternative Routing Reasons". It features a "GENERAL" tab. Under this tab, there are two configuration fields: "Index" with a text input field containing the value "0", and "Release Cause" with a dropdown menu currently displaying "503 Service Unavailable". At the bottom of the window, there are two buttons: "Cancel" and "APPLY".

4. Click **Apply**.

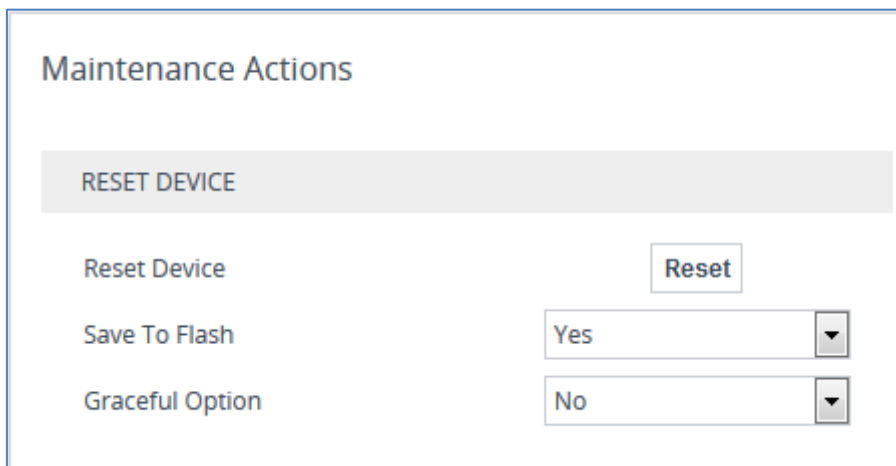
4.17 Step 17: Reset the E-SBC

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



The screenshot shows the 'Maintenance Actions' web page. At the top, there is a section titled 'RESET DEVICE' in a light gray box. Below this, there are three rows of controls:

RESET DEVICE	
Reset Device	Reset
Save To Flash	Yes ▼
Graceful Option	No ▼

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+eihzKSgyfDIzMDI1YGc0YWhub2h1P
GpUVwdVBlnSBgpRXV4=", 1, 0, 2, 15, 60, 200,
"62cabed25276f6d59432fcac295a1346";
WebUsers 1 = "User",
"$1$FrWcHL04tOHmvOKy70iys7m5vrbzpqfyoKL0r6v7q/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,

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IpProfile_SBCDiversioMode, 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_SBCRTPRedundancyBehavior,
IpProfile_SBCPlayRBTToTransferee, IpProfile_SBCRTCPMode,
IpProfile_SBCJitterCompensation,
IpProfile_SBCRemoteRenegotiateOnFaxDetection,
IpProfile_JitterBufMaxDelay,
IpProfile_SBCUserBehindUdpNATRegistrationTime,
IpProfile_SBCUserBehindTcpNATRegistrationTime,
IpProfile_SBCSDPHandleRTCPAttribute,
IpProfile_SBCRemoveCryptoLifetimeInSDP, IpProfile_SBCIceMode,
IpProfile_SBCRTCPMux, IpProfile_SBCMediaSecurityMethod,
IpProfile_SBCHandleXDetect, IpProfile_SBCRTCPFeedback,
IpProfile_SBCRemoteRepresentationMode, IpProfile_SBCKeepVIAHeaders,
IpProfile_SBCKeepRoutingHeaders, IpProfile_SBCKeepUserAgentHeader,
IpProfile_SBCRemoteMultipleEarlyDialogs,
IpProfile_SBCRemoteMultipleAnswersMode, IpProfile_SBCDirectMediaTag,
IpProfile_SBCAdaptRFC2833BWToVoiceCoderBW,
IpProfile_CreatedByRoutingServer, IpProfile_SBCFaxReroutingMode,
IpProfile_SBCMaxCallDuration, IpProfile_SBCGenerateRTP,
IpProfile_SBCISUPBodyHandling, IpProfile_SBCISUPVariant,
IpProfile_SBCVoiceQualityEnhancement, IpProfile_SBCMaxOpusBW;
IpProfile 1 = "S4B", 1, "AudioCodersGroups_0", 0, 10, 10, 46, 24, 0, 0,
0, 2, 0, 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,
300, -1, -1, 0, 0, 0, 0, 0, 0, 0, -1, -1, -1, -1, -1, 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, 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, 0, 1, 0, 0, 0, 0, 0, 0, 0, 1, 0,
0, 0, 300, -1, -1, 0, 1, 0, 0, 0, 0, 0, 0, -1, -1, -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_TopologyLocation;
CpMediaRealm 0 = "MRLan", "Voice", "", 6000, 250, 8499, 1, "", "", 0;

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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_SRDName, SIPInterface_MessagePolicyName,
SIPInterface_TLSContext, SIPInterface_TLSMutualAuthentication,
SIPInterface_TCPKeepaliveEnable,
SIPInterface_ClassificationFailureResponseType,
SIPInterface_PreClassificationManSet, SIPInterface_EncapsulatingProtocol,
SIPInterface_MediaRealm, SIPInterface_SBCDirectMedia,
SIPInterface_BlockUnRegUsers, SIPInterface_MaxNumOfRegUsers,
SIPInterface_EnableUnAuthenticatedRegistrations,
SIPInterface_UsedByRoutingServer, SIPInterface_TopologyLocation;
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;

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[ \SIPInterface ]

[ ProxySet ]

FORMAT ProxySet_Index = ProxySet_ProxyName,
ProxySet_EnableProxyKeepAlive, ProxySet_ProxyKeepAliveTime,
ProxySet_ProxyLoadBalancingMethod, ProxySet_IsProxyHotSwap,
ProxySet_SRDName, ProxySet_ClassificationInput, ProxySet_TLSContextName,
ProxySet_ProxyRedundancyMode, ProxySet_DNSResolveMethod,
ProxySet_KeepAliveFailureResp, ProxySet_GWIPv4SIPInterfaceName,
ProxySet_SBCIPv4SIPInterfaceName, ProxySet_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_SRDName, 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 ]

[ SBCCAlternativeRoutingReasons ]

FORMAT SBCCAlternativeRoutingReasons_Index =
SBCCAlternativeRoutingReasons_ReleaseCause;
SBCCAlternativeRoutingReasons 0 = 503;

[ \SBCCAlternativeRoutingReasons ]
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[ 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,

```



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[ 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 ]

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```
[ AudioCoders ]
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```
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 ]
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