

Microsoft® Skype for Business Server 2015 and DTAG SIP Trunk using AudioCodes Mediant™ MSBR E-SBC

Version 6.8



Table of Contents

1	Introduction	7
1.1	Intended Audience	7
1.2	About AudioCodes E-SBC Product Series.....	7
2	Component Information.....	9
2.1	AudioCodes MSBR E-SBC Version.....	9
2.2	DTAG SIP Trunking Version.....	9
2.3	Microsoft Skype for Business Server 2015 Version	9
2.4	Interoperability Test Topology	10
2.4.1	Environment Setup	11
2.4.2	Known Limitations.....	11
3	Configuring Skype for Business Server 2015.....	13
3.1	Configuring the E-SBC as an IP / PSTN Gateway	13
3.2	Configuring the "Route" on Skype for Business Server 2015.....	21
4	Configuring AudioCodes E-SBC.....	31
4.1	Step 1: IP Network Interfaces Configuration	32
4.1.1	Step 1a: Configure Network Interface	33
4.2	Step 2: Enable the SBC Application	34
4.3	Step 3: Signaling Routing Domains Configuration	35
4.3.1	Step 3a: Configure Media Realms.....	35
4.3.2	Step 3b: Configure SRDs	37
4.3.3	Step 3c: Configure SIP Signaling Interfaces	38
4.4	Step 4: Configure Proxy Sets	40
4.5	Step 5: Configure IP Groups.....	43
4.6	Step 5: Configure IP Profiles	45
4.7	Step 7: Configure Coders	52
4.8	Step 8: SIP TLS Connection Configuration	54
4.8.1	Step 8a: Configure the NTP Server Address.....	54
4.8.2	Step 8b: Configure the TLS version	54
4.8.3	Step 8c: Configure a Certificate.....	56
4.9	Step 9: Configure SRTP	61
4.10	Step 10: Configure IP-to-IP Call Routing Rules	62
4.11	Step 11: Configure IP-to-IP Manipulation Rules.....	68
4.12	Step 12: Configure Message Manipulation Rules	70
4.13	Step 13: Configure Registration Accounts	82
4.14	Step 14: Miscellaneous Configuration.....	83
4.14.1	Step 14a: Configure Call Forking Mode	83
4.14.2	Step 14b: Configure DNS Query Type	84
4.14.3	Step 14c: Loading Prerecorded Tones File	85
4.14.4	Step 14d: Configure RTP Port for T.38 Fax	86
4.15	Step 15: Reset the E-SBC	87
A	AudioCodes INI File	89
B	AudioCodes MSBR Data Configuration	97

This page is intentionally left blank.

Notice

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

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

© Copyright 2018 AudioCodes Ltd. All rights reserved.

This document is subject to change without notice.

Date Published: July-17-2018

Trademarks

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

WEEE EU Directive

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

Customer Support

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

Document Revision Record

LTRT	Description
12590	Initial document release for Version 6.8.

Documentation Feedback

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

This page is intentionally left blank.

1 Introduction

This Configuration Note describes how to set up AudioCodes Enterprise Session Border Controller (hereafter, referred to as *E-SBC*) for interworking between DTAG'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 DTAG Partners who are responsible for installing and configuring DTAG's SIP Trunk and Microsoft's Skype for Business Server 2015 for enabling VoIP calls using AudioCodes E-SBC.

1.2 About AudioCodes E-SBC Product Series

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

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

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

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

This page is intentionally left blank.

2 Component Information

2.1 AudioCodes MSBR E-SBC Version

Table 2-1: AudioCodes MSBR E-SBC Version

SBC Vendor	AudioCodes
Models	▪ Mediant 500L MSBR & E-SBC ▪ Mediant 500 MSBR & E-SBC ▪ Mediant 800 MSBR & E-SBC
Software Version	SIP_6.80A.300.009
Protocol	▪ SIP/UDP (to the DTAG SIP Trunk) ▪ SIP/TCP or TLS (to the S4B FE Server)
Additional Notes	None

2.2 DTAG SIP Trunking Version

Table 2-2: DTAG Version

Vendor/Service Provider	IBM / DTAG
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 6.0.9319.0
Protocol	SIP
Additional Notes	None

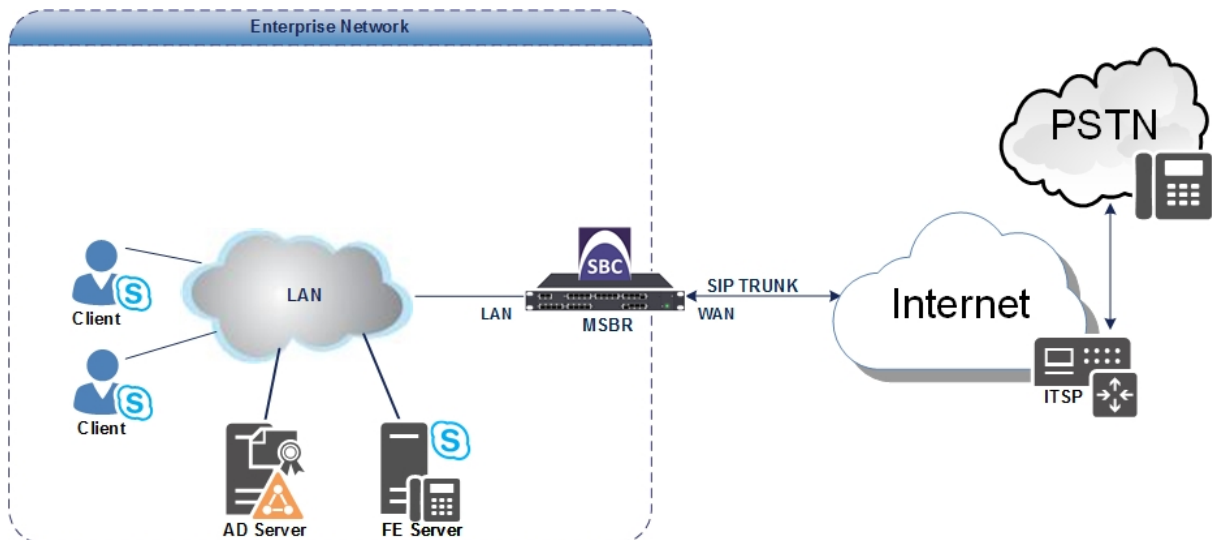
2.4 Interoperability Test Topology

The interoperability testing between AudioCodes E-SBC and DTAG 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 DTAG'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 DTAG'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 DTAG 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 - DTAG SIP Trunk is located on the WAN
Signaling Transcoding	- Microsoft Skype for Business Server 2015 operates with SIP-over-TLS transport type - DTAG SIP Trunk operates with SIP-over-TCP transport type
Codecs Transcoding	Both, Microsoft Skype for Business Server 2015 and DTAG SIP Trunk, supports G.711A-law and G.711U-law coders
Media Transcoding	- Microsoft Skype for Business Server 2015 operates with SRTP media type - DTAG SIP Trunk operates with RTP media type

2.4.2 Known Limitations

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

- When performing the Hold scenario without Music on Hold (MoH), (when a Microsoft Skype for Business Server 2015 user sends a Re-INVITE with an **inactive** SDP RTP Mode (i.e., a=inactive)), the DTAG responds with an "RTP Port 0 in the 200 OK" message. This is incorrect and not according to RFC 3264, and causes the media to be closed by the E-SBC. To deal with this limitation, set **Remote Hold Format** as 'Send Only' in the DTAG IP Profile (see Section 4.6 on page 45).
- If the Microsoft Skype for Business Server 2015 sends one of the following error responses:
 - 503 Service Unavailable
 - 603 Decline
 DTAG SIP Trunk still sends re-INVITES and does not disconnect the call.
 To disconnect the call, a message manipulation rule is used to replace the above error response with the '486 Busy Here' response (see Section 4.12 on page 70).
- With Incoming calls from the DTAG SIP Trunk, the SIP Record-Route Header is represented as **FQDN**. To resolve the IP address for any response, the DNS Query Type should be configured as "SRV" (see Section 4.14.2 on page 84).
- In Call Forwarding scenarios, when a Skype for Business user forwards a call to a PSTN user, RTP packets need to be sent to open a pinhole in the firewall. To overcome this problem with the first incoming RTP packet in this scenario, instead of generating Ringback Tone as Call Progress Tone (CPT), which requires DSP we decided to use a Prerecorded Tones (PRT) file for ringback tones.

This page is intentionally left blank.

3 Configuring Skype for Business Server 2015

This chapter describes how to configure Microsoft Skype for Business Server 2015 to operate with AudioCodes E-SBC.



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

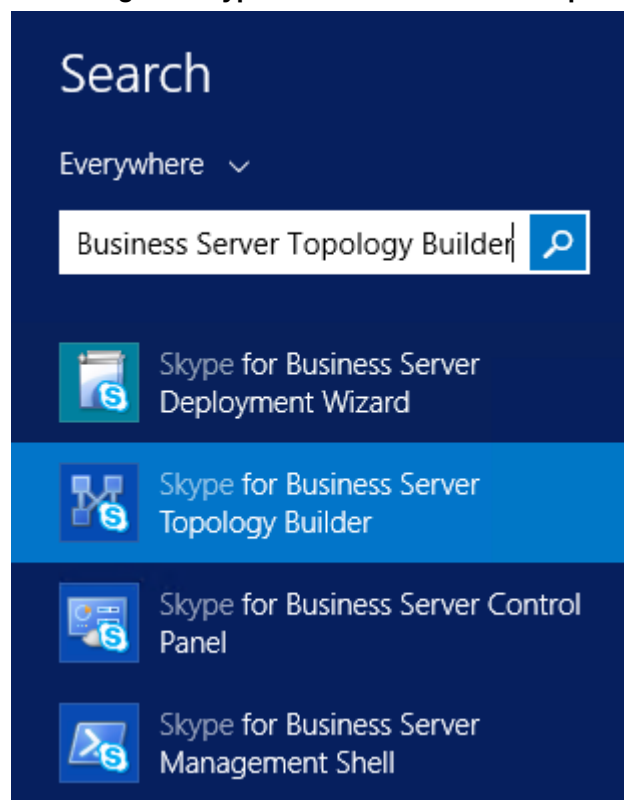
3.1 Configuring the E-SBC as an IP / PSTN Gateway

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

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

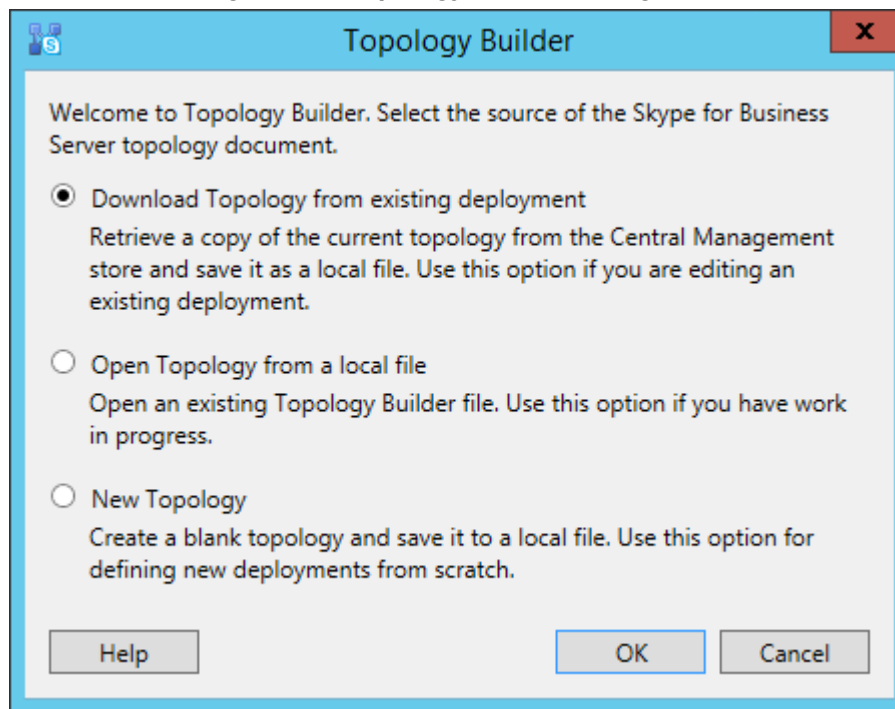
1. On the server where the Topology Builder is installed, start the Skype for Business Server 2015 Topology Builder (Windows **Start** menu > search for **Skype for Business Server Topology Builder**), as shown below:

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



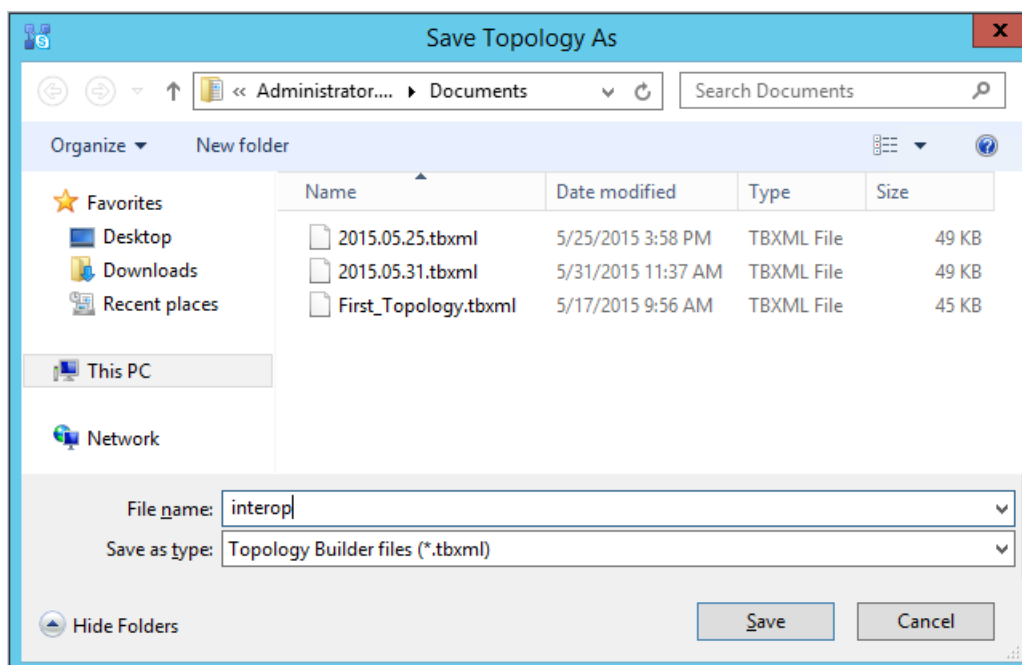
The following is displayed:

Figure 3-2: Topology Builder Dialog Box



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

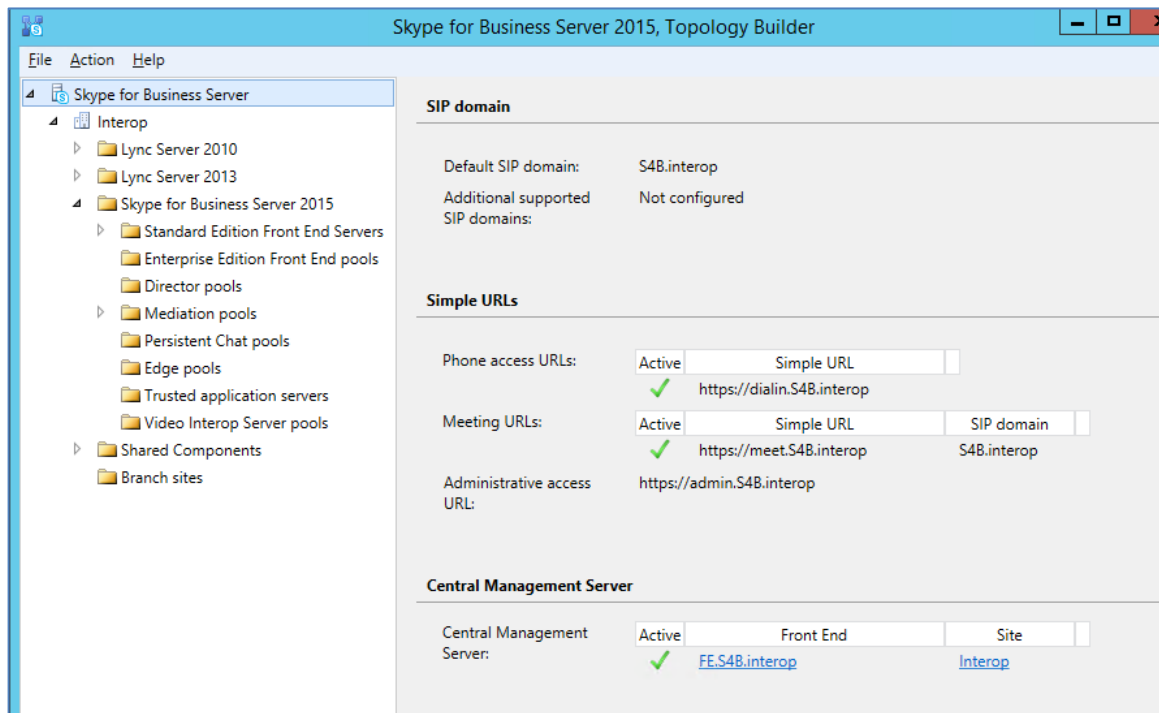
Figure 3-3: Save Topology Dialog Box



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

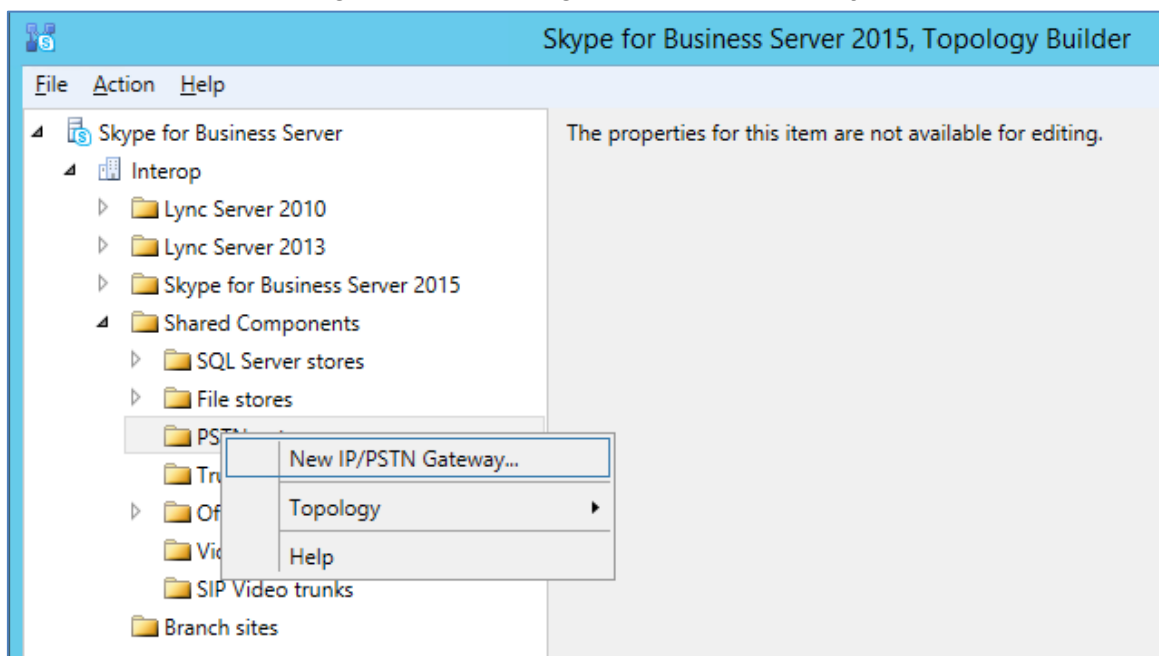
The Topology Builder screen with the downloaded Topology is displayed:

Figure 3-4: Downloaded Topology



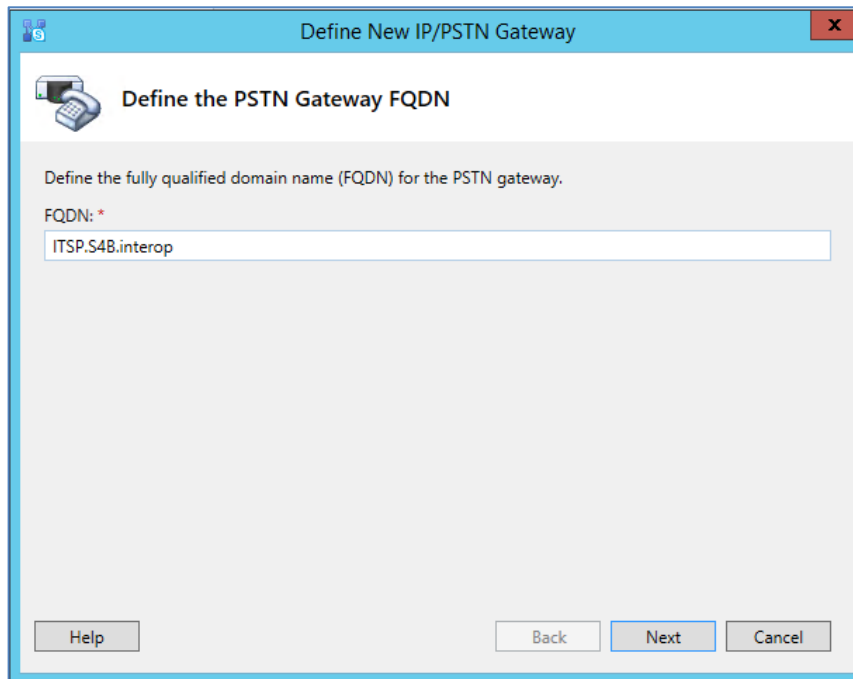
- Under the **Shared Components** node, right-click the **PSTN gateways** node, and then from the shortcut menu, choose **New IP/PSTN Gateway**, as shown below:

Figure 3-5: Choosing New IP/PSTN Gateway



The following is displayed:

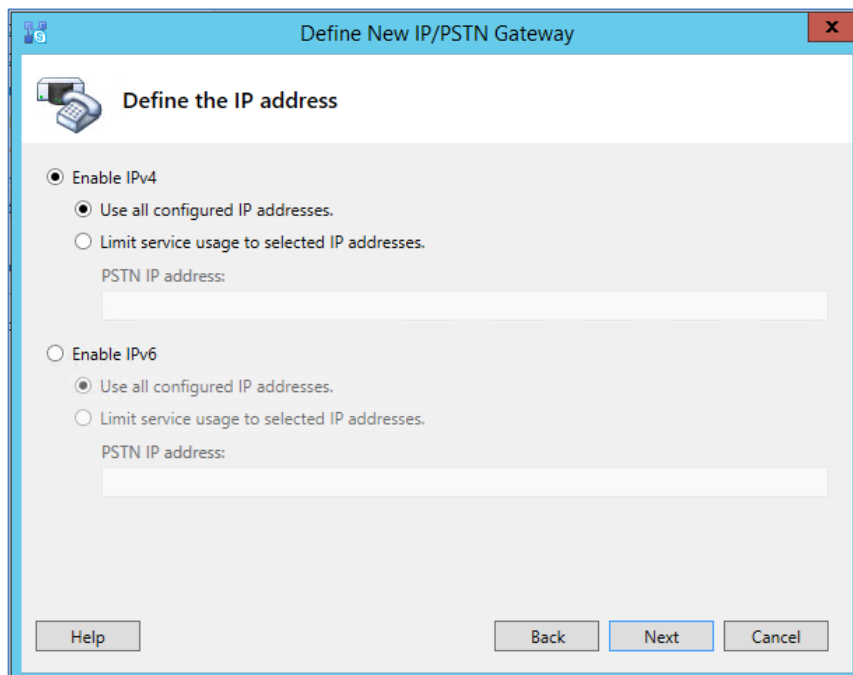
Figure 3-6: Define the PSTN Gateway FQDN



The screenshot shows a dialog box titled "Define New IP/PSTN Gateway". Inside, there is a section titled "Define the PSTN Gateway FQDN" with a telephone icon. Below the title, it says "Define the fully qualified domain name (FQDN) for the PSTN gateway." There is a text input 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 4.8.3 on page 56).
6. Click **Next**; the following is displayed:

Figure 3-7: Define the IP Address



The screenshot shows the same dialog box, but now the section is titled "Define the IP address" with a telephone icon. It has two main options: "Enable IPv4" and "Enable IPv6". Each option has two sub-options: "Use all configured IP addresses." (selected with a radio button) and "Limit service usage to selected IP addresses." (unselected). Below each sub-option is a text input field labeled "PSTN IP address:". 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:

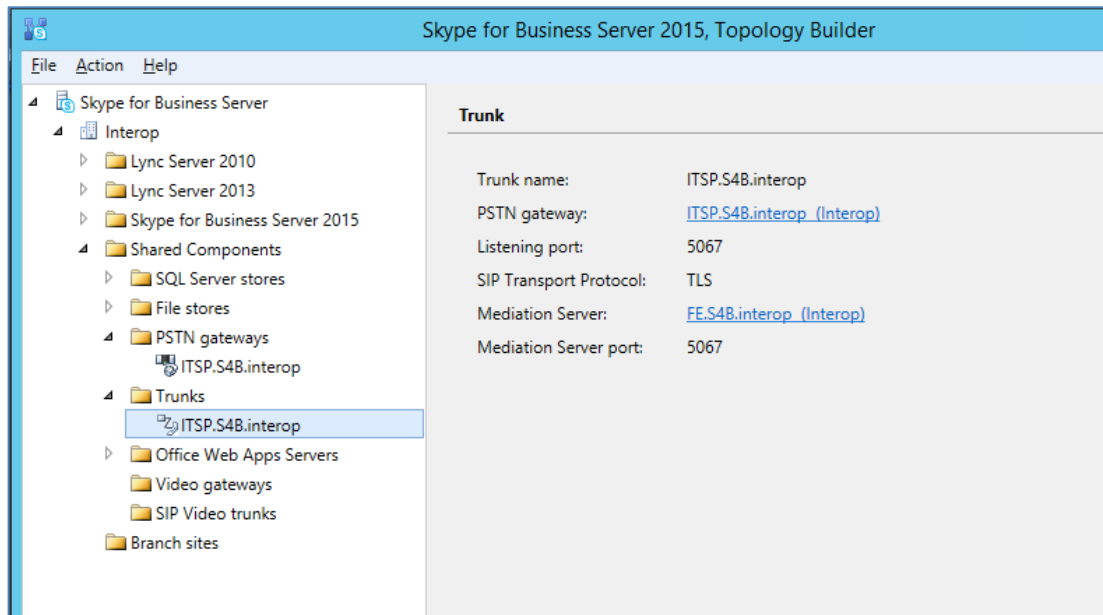
- When defining a PSTN gateway in Topology Builder, you must define a root trunk to successfully add the PSTN gateway to your topology.
- The root trunk cannot be removed until the associated PSTN gateway is removed.

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**).
- b. In the 'SIP Transport Protocol' field, select the transport type (e.g., **TLS**) that the trunk uses.
- c. In the 'Associated Mediation Server' field, select the Mediation Server pool to associate with the root trunk of this PSTN gateway.
- d. In the 'Associated Mediation Server Port' field, enter the listening port that the Mediation Server will use for SIP messages from the SBC (e.g., **5067**).
- e. Click **Finish**.

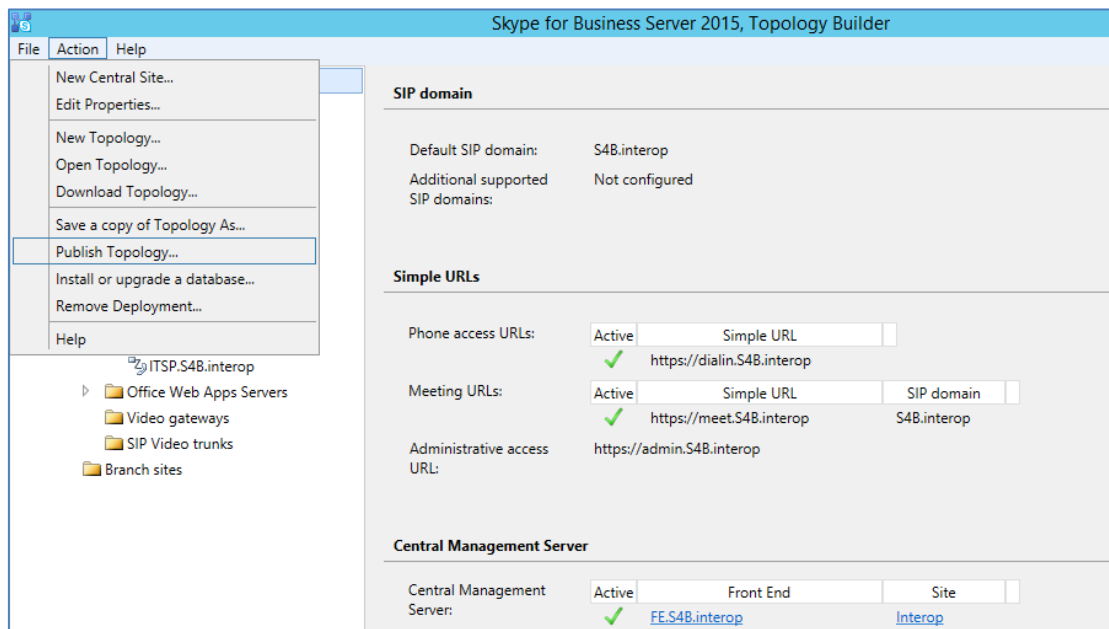
The E-SBC is added as a PSTN gateway, and a trunk is created as shown below:

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



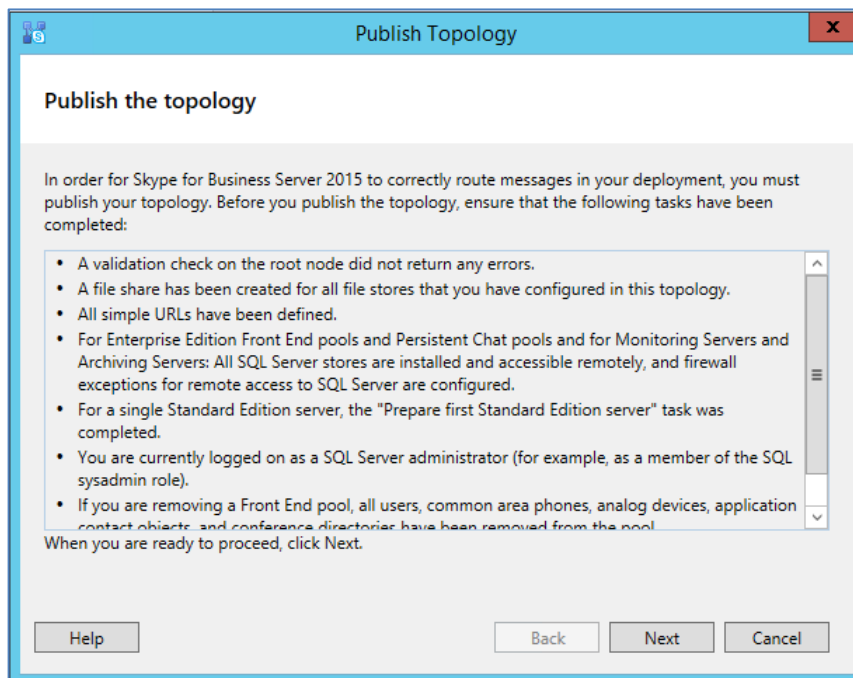
9. Publish the Topology: In the main tree, select the root node **Skype for Business Server**, and then from the **Action** menu, choose **Publish Topology**, as shown below:

Figure 3-10: Choosing Publish Topology



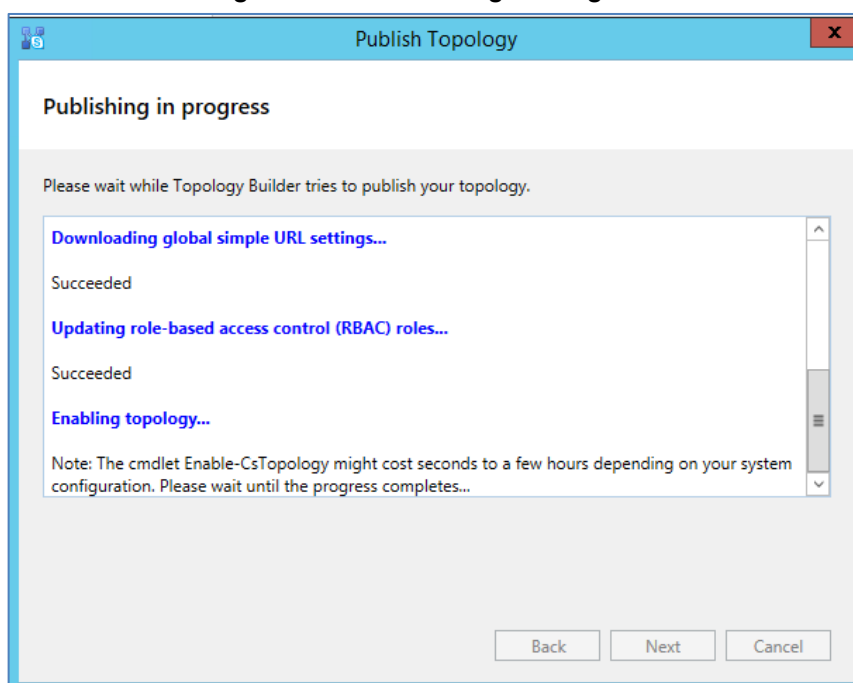
The following is displayed:

Figure 3-11: Publish the Topology



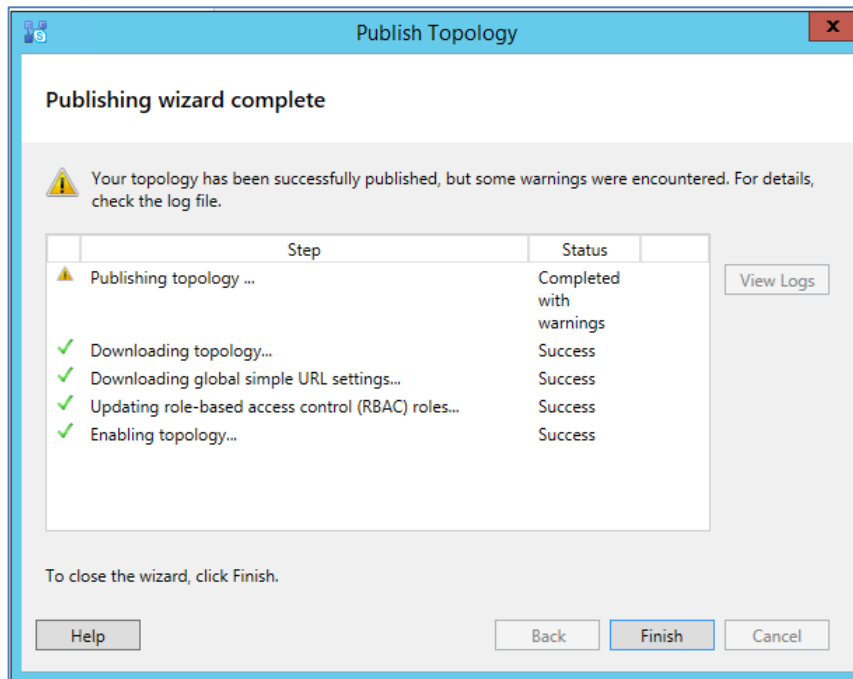
10. Click **Next**; the Topology Builder starts to publish your topology, as shown below:

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**.

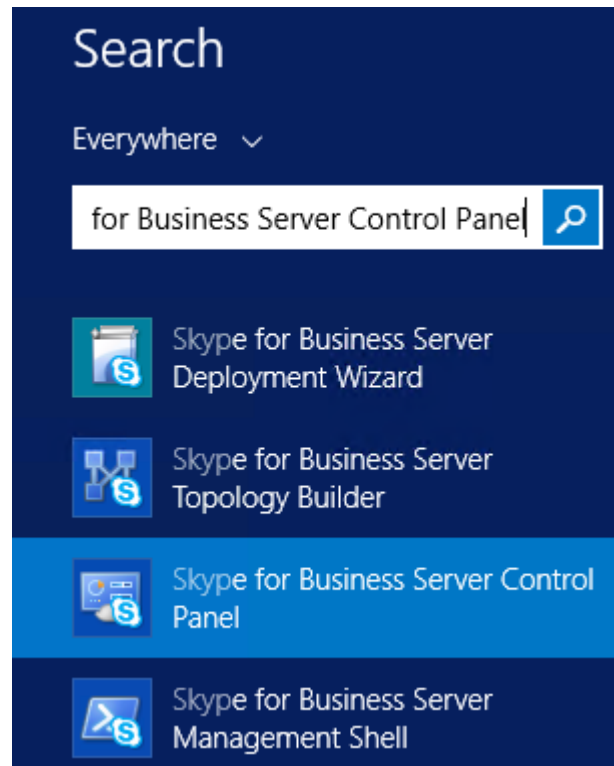
3.2 Configuring the "Route" on Skype for Business Server 2015

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:**

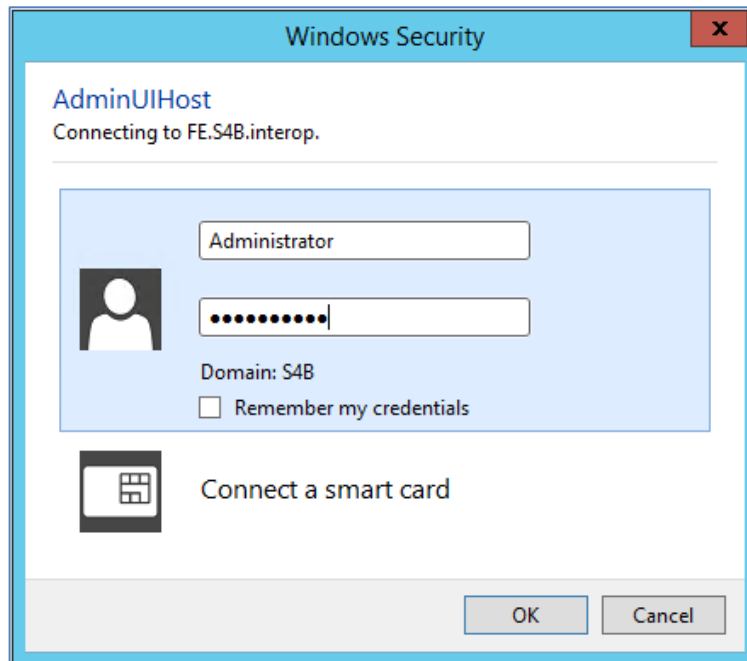
1. Start the Microsoft Skype for Business Server 2015 Control Panel (**Start** > search for **Microsoft Skype for Business Server Control Panel**), as shown below:

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



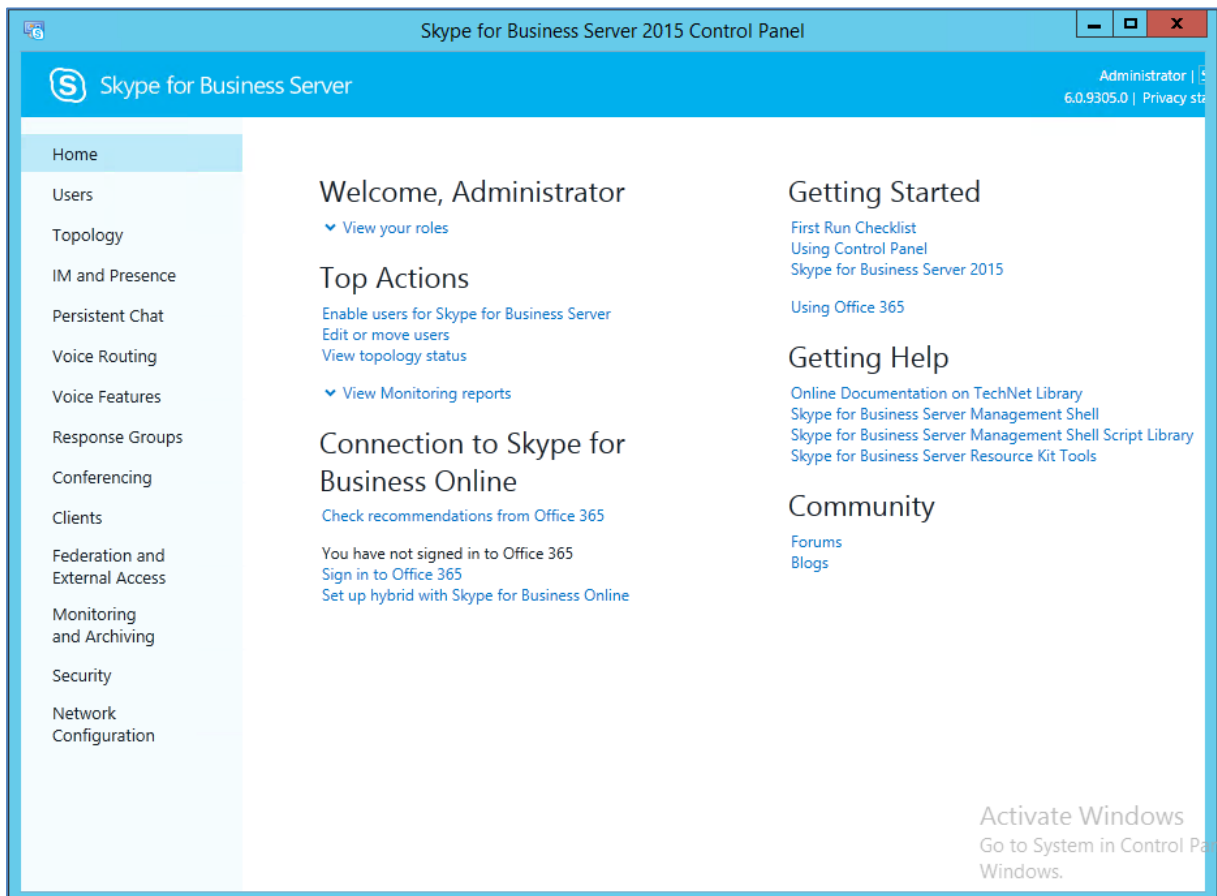
2. You are prompted to enter your login credentials.

Figure 3-15: Skype for Business Server Credentials



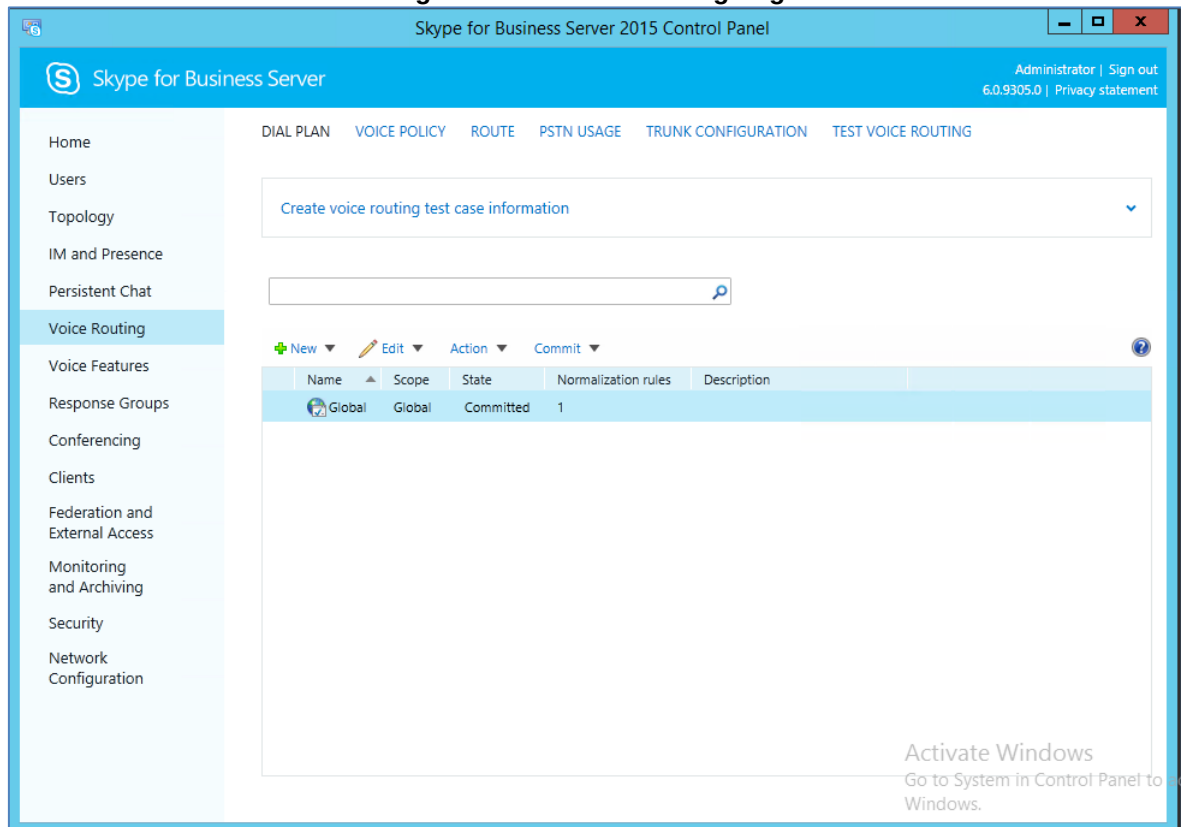
3. Enter your domain username and password, and then click **OK**; the Microsoft Skype for Business Server 2015 Control Panel is displayed.

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



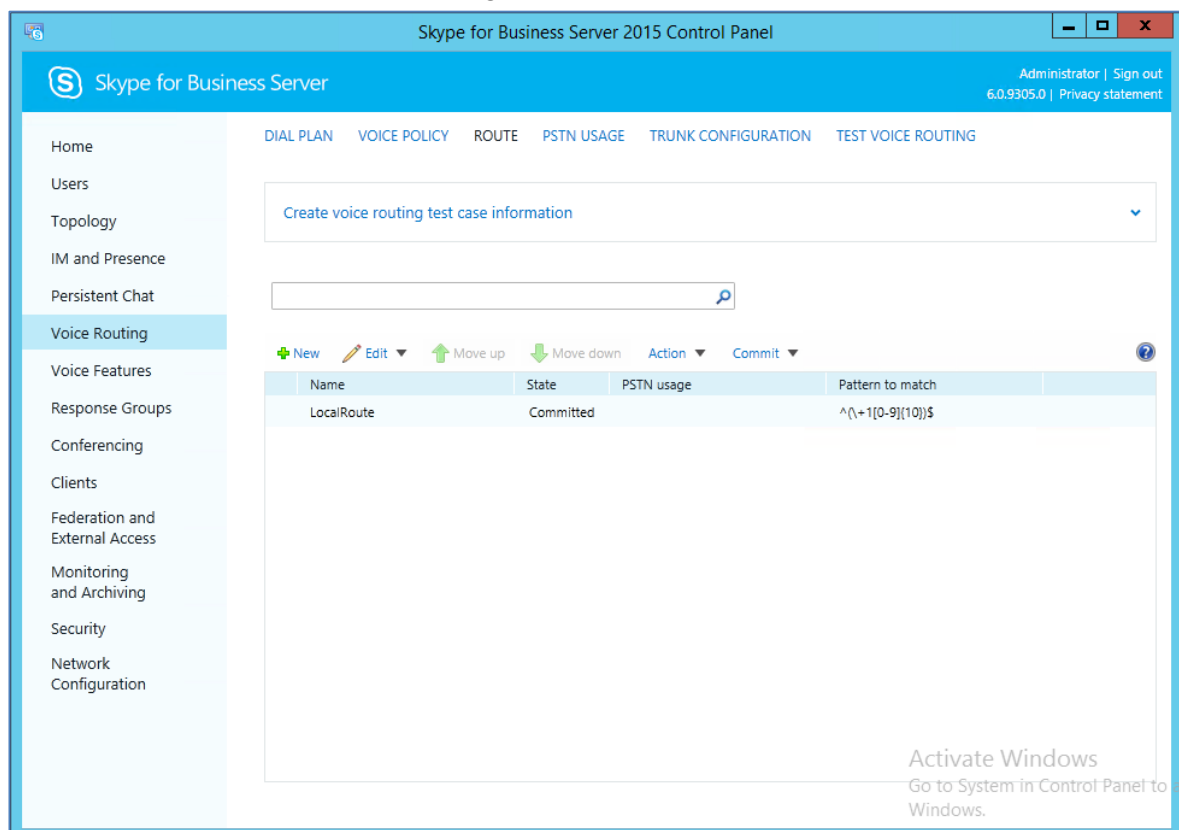
4. In the left navigation pane, select **Voice Routing**.

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

Match this pattern: *

Associated trunks

Select Trunk

Service	Site
PstnGateway:ITSP.S4B.interop	Interop

OK Cancel

- 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:
 - a. 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

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((66))

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

Figure 3-24: Committing Voice Routes

Skype for Business Server

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

Create voice routing test case information

Search

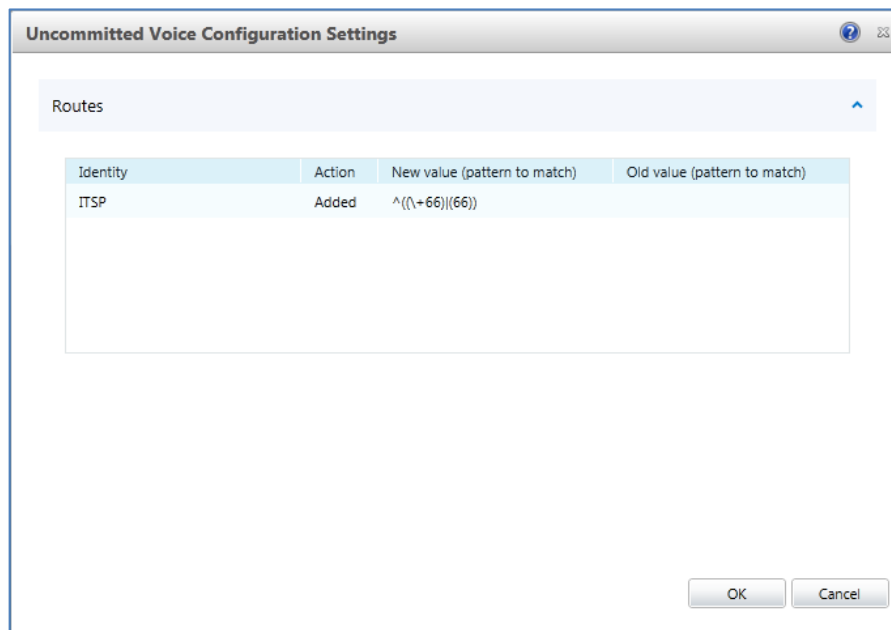
+ New Edit Move up Move down Action Commit

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

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

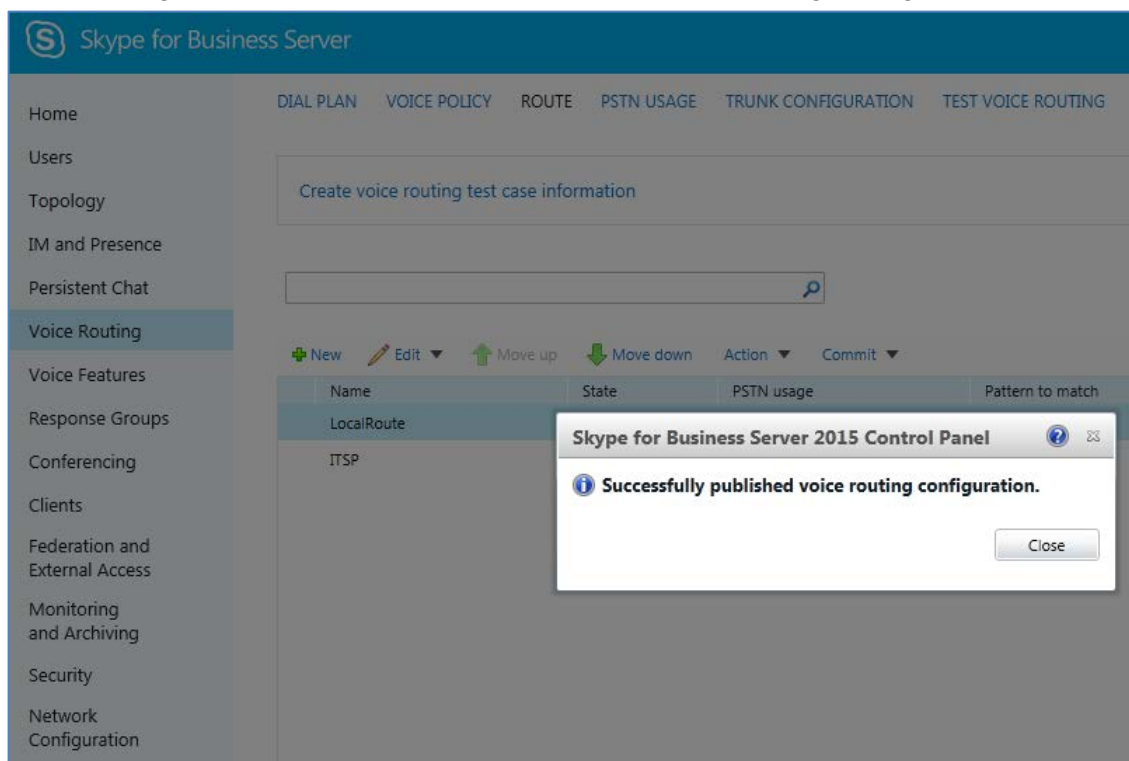
The Uncommitted Voice Configuration Settings page appears:

Figure 3-25: Uncommitted Voice Configuration Settings



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 DTAG 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 4.5 on page 43).

- 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 administration console. The left-hand navigation pane is expanded to 'Voice Routing'. The main content area displays the 'New Trunk Configuration' page for the entity 'PstnGateway:ITSP.S4B.interop'. At the top of the main area, there is a blue bar with the Skype for Business Server logo and the text 'Administrator | Sign out 6.0.9305.0 | Privacy statement'. Below this, there are tabs for 'DIAL PLAN', 'VOICE POLICY', 'ROUTE', 'PSTN USAGE', 'TRUNK CONFIGURATION', and 'TEST VOICE ROUTING'. The 'TRUNK CONFIGURATION' tab is active. Below the tabs, there is a search bar with the text 'Create voice routing test case information'. Below the search bar, there is a section for 'New Trunk Configuration - PstnGateway:ITSP.S4B.interop'. This section contains a green checkmark and the text 'OK' and a red X and the text 'Cancel'. Below this, there is a 'Scope' field with the value 'Pool'. Below the 'Scope' field, there is a 'Name' field with the value 'PstnGateway:ITSP.S4B.interop'. Below the 'Name' field, there is a 'Description' field. Below the 'Description' field, there is a 'Maximum early dialogs supported' field with the value '20'. Below the 'Maximum early dialogs supported' field, there is an 'Encryption support level' dropdown menu with the value 'Required'. Below the 'Encryption support level' dropdown menu, there is a 'Refer support' dropdown menu with the value 'Enable sending refer to the gateway'. Below the 'Refer support' dropdown menu, there 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 to 13 to commit your settings.

This page is intentionally left blank.

4 Configuring AudioCodes E-SBC

This chapter provides step-by-step procedures on how to configure AudioCodes MSBR E-SBC for interworking between Microsoft Skype for Business Server 2015 and the DTAG 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 MSBR WAN interface - DTAG 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*).

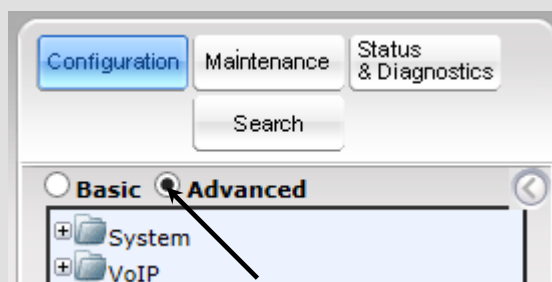
Notes:

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

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

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

- The scope of this interoperability test and document does **not** cover all security aspects for connecting the SIP Trunk to the Microsoft Skype for Business environment. Comprehensive security measures should be implemented per your organization's security policies. For security recommendations on AudioCodes' products, refer to the *Recommended Security Guidelines* document.
- Before you begin configuring the E-SBC, ensure that the E-SBC's Web interface Navigation tree is in Advanced-menu display mode. To do this, select the Advanced option, as shown below:



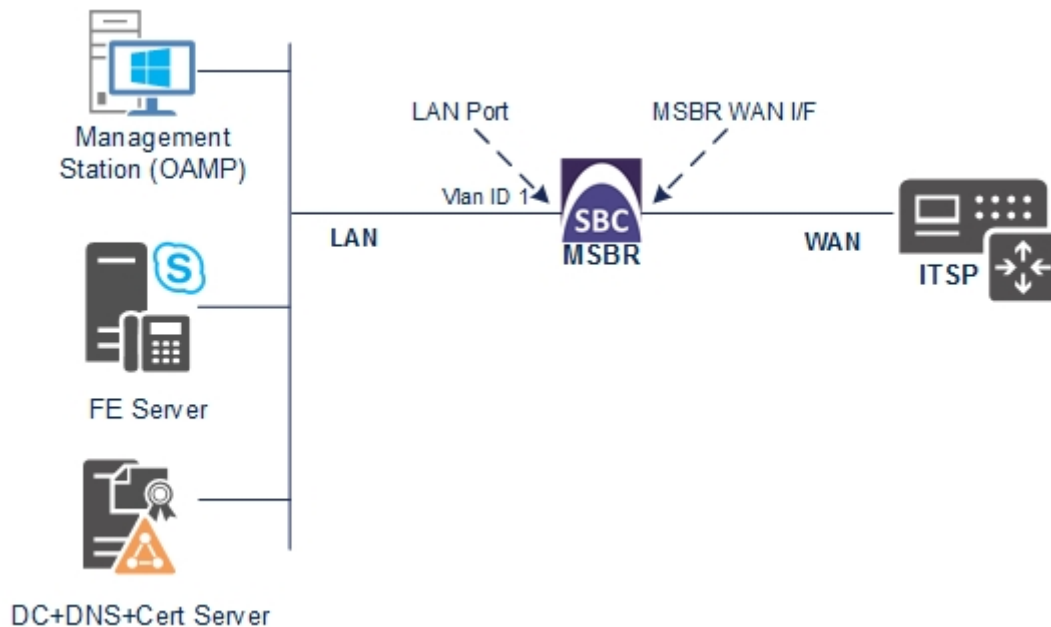
- When the E-SBC is reset, the Navigation tree reverts to Basic-menu display.

4.1 Step 1: IP Network Interfaces Configuration

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

- E-SBC interfaces with the following IP entities:
 - Skype for Business servers, located on the LAN
 - DTAG SIP Trunk, located on the WAN
- 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 using dedicated LAN port and to the WAN using MSBR's VDSL WAN interface.

Figure 4-1: Network Interfaces in Interoperability Test Topology



4.1.1 Step 1a: Configure Network Interface

This step describes how to configure the IP network interface for LAN VoIP interface (assigned the name "Voice"). Configuration of WAN data interface depends on physical interface, that's why it is out of the scope of this document.

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

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

Parameter	Value
IP Address	10.15.17.10 (LAN IP address of E-SBC)
Prefix Length	16 (subnet mask in bits for 255.255.0.0)
Default Gateway	10.15.17.34 (MSBR Data vlan 1 IP address)
Interface Name	Voice (arbitrary descriptive name)
Primary DNS Server IP Address	10.15.27.1
Underlying Device	vlan 1

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

The configured IP network interface is shown below:

Figure 4-2: Configured Network Interface in IP Interfaces Table

Interface Table									
Add +			Edit ↗			Delete 🗑			Show/Hide 📄
Index	Application Type	Interface Mode	IP Address	Prefix Length	Default Gateway	Interface Name	Primary DNS	Secondary DNS	Underlying Device
0	OAMP + Media	IPv4 Manual	10.15.17.10	16	10.15.17.34	Voice	10.15.27.1	0.0.0.0	vlan 1
⏪ ⏩ Page 1 of 1 Show 10 records per page View 1 - 1 of 1									

4.2 Step 2: Enable the SBC Application

This step describes how to enable the SBC application.

➤ **To enable the SBC application:**

1. Open the Applications Enabling page (**Configuration** tab > **VoIP** menu > **Applications Enabling** > **Applications Enabling**).

Figure 4-3: Enabling SBC Application

⚡ SAS Application	Disable
⚡ SBC Application	Enable
⚡ IP to IP Application	Disable

2. From the 'SBC Application' drop-down list, select **Enable**.
3. Click **Submit**.
4. Reset the E-SBC with a burn to flash for this setting to take effect (see Section 4.15 on page 87).

4.3 Step 3: Signaling Routing Domains Configuration

This step describes how to configure Signaling Routing Domains (SRD). The SRD represents a logical VoIP network. Each logical or physical connection requires an SRD, for example, if the E-SBC interfaces with both the LAN and WAN, a different SRD would be required for each one.

The SRD is composed of the following:

- **Media Realm:** Defines a UDP port range for RTP/SRTP (media) traffic on a specific logical IP network interface of the E-SBC.
- **SIP Interface:** Defines a listening port and type (UDP, TCP, or TLS) for SIP signaling traffic on a specific logical IP network interface of the E-SBC.

4.3.1 Step 3a: Configure Media Realms

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

➤ **To configure Media Realms:**

1. Open the Media Realm Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **Media Realm Table**).
2. Modify the existing Media Realm for LAN traffic:

Parameter	Value
Index	0
Media Realm Name	MRLan (descriptive name)
IPv4 Interface Name	Voice
Port Range Start	6000 (represents lowest UDP port number used for media on LAN)
Number of Media Session Legs	100 (media sessions assigned with port range)

Figure 4-4: Configuring Media Realm for LAN

The screenshot shows the 'Edit Record #0' dialog box with the following fields and values:

- Index: 0
- Media Realm Name: MRLan
- IPv4 Interface Name: Voice
- IPv6 Interface Name: None
- Port Range Start: 6000
- Number Of Media Session Legs: 100
- Port Range End: -1
- Default Media Realm: No
- QoS Profile: None
- BW Profile: None

At the bottom, there are 'Submit' and 'Cancel' buttons.

3. Configure a Media Realm for WAN traffic:

Parameter	Value
Index	1
Media Realm Name	MRWan (arbitrary name)
IPv4 Interface Name	WAN (a reserved word for MSBR WAN I/F)
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 4-5: Configuring Media Realm for WAN

The screenshot shows the 'Edit Record #1' dialog box. It contains the following fields and values:

- Index: 1
- Media Realm Name: MRWan
- IPv4 Interface Name: WAN
- IPv6 Interface Name: None
- Port Range Start: 7000
- Number Of Media Session Legs: 100
- Port Range End: 7990
- Default Media Realm: No
- QoS Profile: None
- BW Profile: None

Buttons at the bottom: Submit, Cancel.

The configured Media Realms are shown in the figure below:

Figure 4-6: Configured Media Realms in Media Realm Table

Media Realm Table			
Add +			
Index	Media Realm Name	IPv4 Interface Name	IPv6 Interface Name
0	MRLan	Voice	None
1	MRWan	WAN	None
Page 1 of 1 Show 10 records per page View 1 - 2 of 2			

4.3.2 Step 3b: Configure SRDs

This step describes how to configure the SRDs.

➤ **To configure SRDs:**

1. Open the SRD Settings page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **SRD Table**).
2. Configure an SRD for the E-SBC's internal interface (toward Skype for Business):

Parameter	Value
Index	0
Name	SRDLan (descriptive name for SRD)
Media Realm Name	MRLan (associates SRD with Media Realm)

Figure 4-7: Configuring LAN SRD

→ Index: 0

→ Name: SRDLan

→ Media Realm Name: MRLan

Media Anchoring: Enable

Block Unregistered Users: NO

Max. Number of Registered Users: -1

Enable Un-Authenticated Registrations: Enable

Submit Cancel

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

Parameter	Value
Index	1
Name	SRDWan
Media Realm Name	MRWan

Figure 4-8: Configuring WAN SRD

→ Index: 1

→ Name: SRDWan

→ Media Realm Name: MRWan

Media Anchoring: Enable

Block Unregistered Users: NO

Max. Number of Registered Users: -1

Enable Un-Authenticated Registrations: Enable

Submit Cancel

The configured SRDs are shown in the figure below:

Figure 4-9: Configured SRDs in SRD Table

SRD Table			
Add +			
Index	Name	Media Realm Name	Media Anchoring
0	SRDLan	MRLan	Enable
1	SRDWan	MRWan	Enable
Page 1 of 1 Show 10 records per page View 1 - 2 of 2			

4.3.3 Step 3c: Configure SIP Signaling Interfaces

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

➤ **To configure SIP Interfaces:**

1. Open the SIP Interface Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **SIP Interface Table**).
2. Configure a SIP interface for the LAN:

Parameter	Value
Index	0
Interface Name	S4B (arbitrary descriptive name)
Network Interface	Voice
Application Type	SBC
TLS Port	5067
TCP and UDP	0
SRD	0

3. Configure a SIP interface for the WAN:

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

The configured SIP Interfaces are shown in the figure below:

Figure 4-10: Configured SIP Interfaces in SIP Interface Table

SIP Interface Table							
Add +							
Index	SIP Interface Name	Network Interface	Application Type	UDP Port	TCP Port	TLS Port	SRD
0	S4B	Voice	SBC	0	0	5067	0
1	DTAG	WAN	SBC	0	5060	0	1
Page 1 of 1 Show 10 records per page View 1 - 2 of 2							



Note: The TLS port parameter (for S4B SIP Interface) must be identically configured in the Skype for Business Topology Builder (see Section 3.1 on page 13).

4.4 Step 4: Configure Proxy Sets

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

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

- Microsoft Skype for Business Server 2015
- DTAG 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 page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **Proxy Sets Table**).
2. Add a Proxy Set for the Skype for Business Server 2015 as shown below:

Parameter	Value
Proxy Set ID	1
Proxy Address	FE.S4B.interop:5067 (Skype for Business Server 2015 IP address / FQDN and destination port)
Transport Type	TLS
Proxy Name	S4B (arbitrary descriptive name)
Enable Proxy Keep Alive	Using Options
Proxy Load Balancing Method	Round Robin
Is Proxy Hot Swap	Yes
Proxy Redundancy Mode	Homing
SRD Index	0

Figure 4-11: Configuring Proxy Set for Microsoft Skype for Business Server 2015

Proxy Set ID: 1

	Proxy Address	Transport Type
1	FE.S4B.interop:5067	TLS
2		
3		
4		
5		
6		
7		
8		
9		
10		

Proxy Name: S4B

Enable Proxy Keep Alive: Using Options

Proxy Keep Alive Time: 60

KeepAlive Failure responses:

DNS Resolve Method: Not Configured

Proxy Load Balancing Method: Round Robin

Is Proxy Hot Swap: Yes

Proxy Redundancy Mode: Homing

SRD Index: 0

Classification Input: IP only

TLS Context Index: -1

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

Parameter	Value
Proxy Set ID	2
Proxy Address	reg.sip-trunk.telekom.de (DTAG IP address / FQDN)
Transport Type	TCP
Proxy Name	DTAG (arbitrary descriptive name)
Enable Proxy Keep Alive	Using Options
DNS Resolve Method	SRV
Is Proxy Hot Swap	Yes
Proxy Redundancy Mode	Homing
SRD Index	1

Figure 4-12: Configuring Proxy Set for DTAG SIP Trunk

The screenshot displays the configuration interface for a Proxy Set. At the top, a dropdown menu for 'Proxy Set ID' is set to '2'. Below this is a table with 10 rows for proxy addresses and transport types. The first row is populated with 'register-test.sip-trunk.telekom.de' and 'TCP'. The remaining rows are empty. Below the table, a series of configuration options are listed, each with a corresponding value or dropdown selection. Arrows point to specific fields in the interface.

	Proxy Address	Transport Type
1	register-test.sip-trunk.telekom.de	TCP
2		
3		
4		
5		
6		
7		
8		
9		
10		

Proxy Name	DTAG
Enable Proxy Keep Alive	Using Options
Proxy Keep Alive Time	60
KeepAlive Failure responses	
DNS Resolve Method	SRV
Proxy Load Balancing Method	Disable
Is Proxy Hot Swap	Yes
Proxy Redundancy Mode	Homing
SRD Index	1
Classification Input	IP only
TLS Context Index	-1

4.5 Step 5: Configure IP Groups

This step describes how to configure IP Groups. The IP Group represents an IP entity on the network with which the E-SBC communicates. This can be a server (e.g., IP PBX or ITSP) or it can be a group of users (e.g., LAN IP phones). For servers, the IP Group is typically used to define the server's IP address by associating it with a Proxy Set. A typical deployment consists of multiple IP Groups associated with the same SRD. For example, you can have two LAN IP PBXs sharing the same SRD, and two ITSPs / SIP Trunks sharing the same SRD. Once IP Groups are configured, they are used to configure IP-to-IP routing rules for denoting source and destination of the call.

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

- Skype for Business Server 2015 (Mediation Server)
- DTAG SIP Trunk

➤ To configure IP Groups:

1. Open the IP Group Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **IP Group Table**).
2. Add an IP Group for the Skype for Business Server 2015 as shown below:

Parameter	Value
Index	1
Type	Server
Description	S4B (arbitrary descriptive name)
Proxy Set ID	1
SIP Group Name	sip-trunk.telekom.de (according to ITSP requirement)
SRD	0
Media Realm Name	MRLan
IP Profile ID	1

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

Parameter	Value
Index	2
Type	Server
Description	DTAG (arbitrary descriptive name)
Proxy Set ID	2
SIP Group Name	sip-trunk.telekom.de (according to ITSP requirement)
SRD	1
Media Realm Name	MRWan
IP Profile ID	2

The configured IP Groups are shown in the figure below:

Figure 4-13: Configured IP Groups in IP Group Table

IP Group Table								
Add +								
Index	Type	Description	Proxy Set ID	SIP Group Name	Contact User	SIP Re-Routing Mode	Always Use Route Table	SRD
1	Server	S4B	1	sip-trunk.telekor			No	0
2	Server	DTAG	2	sip-trunk.telekor			No	1
Page 1 of 1 Show 10 records per page View 1 - 2 of 2								

4.6 Step 5: 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 TLS
- DTAG SIP trunk - to operate in non-secure mode using RTP and UDP

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

1. Open the IP Profile Settings page (**Configuration** tab > **VoIP** > **Coders and Profiles** > **IP Profile Settings**).
2. Click **Add**.
3. Click the **Common** tab, and then configure the parameters as follows:

Parameter	Value
Index	1
Profile Name	S4B
Symmetric MKI	Enable
MKI Size	1
Reset SRTP State Upon Re-key	Enable
Generate SRTP keys mode	Always

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

Common		GW	SBC
→ Index		1	
→ Profile Name		S4B	
	Profile Preference	1	
	Dynamic Jitter Buffer Minimum Delay [msec]	10	
	Dynamic Jitter Buffer Optimization Factor	10	
	RTP IP DiffServ	46	
	Signaling DiffServ	40	
	Silence Suppression	Disable	
	RTP Redundancy Depth	0	
	Echo Canceler	Line	
	Broken Connection Mode	Ignore	
	Input Gain (-32 to 31 dB)	0	
	Voice Volume (-32 to 31 dB)	0	
	Media IP Version Preference	Only IPv4	
→ Symmetric MKI		Enable	
→ MKI Size		1	
→ Reset SRTP Upon Re-key		Enable	
→ Generate SRTP keys mode		Always	
	Jitter Buffer Max Delay [msec]	300	
		Submit	Cancel

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

Parameter	Value
SBC Media Security Behavior	SRTP
RFC 2833 Behavior	Extend
PRACK Mode	Optional
Remote Update Support	Supported Only After Connect
Remote re-INVITE	Supported Only With SDP
Remote Delayed Offer Support	Not Supported
Remote REFER Behavior	Handle Locally (required, as Skype for Business Server 2015 does not support receipt of SIP REFER)
Remote 3xx Behavior	Handle Locally (required, as Skype for Business Server 2015 does not support receipt of SIP 3xx responses)
Enforce MKI Size	Enforce
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)
Remote Can Play Ringback	No
RTCP Mode	Generate Always

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

Common GW SBC	
Index	1
Extension Coders Group ID	None
Transcoding Mode	Only If Required
Allowed Media Types	
Allowed Coders Group ID	None
Allowed Video Coders Group ID	None
Allowed Coders Mode	Restriction
SBC Media Security Behavior	SRTP
RFC 2833 Behavior	Extend
Alternative DTMF Method	As Is
P-Asserted-Identity	As Is
Diversion Mode	As Is
History-Info Mode	As Is
Fax Coders Group ID	None
Fax Behavior	As Is
Fax Offer Mode	All coders
Fax Answer Mode	Single coder
PRACK Mode	Optional
Session Expires Mode	Supported
Remote Update Support	Supported Only Aft
Remote re-INVITE	Supported only with
Remote Delayed Offer Support	Not Supported
Remote REFER Behavior	Handle Locally
Remote 3xx Behavior	Handle Locally
Remote Multiple 18x	Supported
Remote Early Media Response Type	Transparent
Remote Early Media	Supported
Enforce MKI Size	Enforce
Remote Early Media RTP Detection Mode	By Media
Remote RFC 3960 Gateway Model Support	Not Supported
Remote Can Play Ringback	No
RFC 2833 DTMF Payload Type	0
User Registration Time	0
Reliable Held Tone Source	Yes
Play Held Tone	No
Remote Hold Format	Transparent
Remote Replaces Behavior	Standard
SDP Ptime Answer	Remote Answer
Preferred PTime	0
Use Silence Suppression	Transparent
RTP Redundancy Behavior	AS IS
Play RBT To Transferee	No
RTCP Mode	Generate Always
Jitter Compensation	Disable
Remote Renegotiate on Fax Detection	Transparent
Remote Multiple Answers Mode	Disabled
Keep VIA Headers	Not Configured
Keep User-Agent Header	Not Configured
User Behind NAT UDP Registration Time	-1
User Behind NAT TCP Registration Time	-1
Adapt RFC2833 BW to Voice coder BW	Disabled
Submit Cancel	

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

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

Parameter	Value
Index	2
Profile Name	DTAG

Figure 4-16: Configuring IP Profile for DTAG SIP Trunk – Common Tab

The screenshot shows a configuration window with three tabs: **Common**, **GW**, and **SBC**. The **Common** tab is selected. The configuration parameters are as follows:

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

At the bottom right, there are **Submit** and **Cancel** buttons.

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

Parameter	Value
Extension Coders Group ID	Coders Group 0
Allowed Coders Group ID	Coders Group 0
SBC Media Security Behavior	RTP
P-Asserted-Identity	Add (required for anonymous calls)
Remote REFER Behavior	Handle Locally (E-SBC handles / terminates incoming REFER requests instead of forwarding them to SIP Trunk)
Remote Hold Format	Send Only

Figure 4-17: Configuring IP Profile for DTAG SIP Trunk – SBC Tab

Common GW SBC		
Index	2	
→ Extension Coders Group ID	Coders Group 0 ▼	
Transcoding Mode	Only If Required ▼	
Allowed Media Types		
→ Allowed Coders Group ID	Coders Group 0 ▼	
Allowed Video Coders Group ID	None ▼	
Allowed Coders Mode	Restriction ▼	
→ SBC Media Security Behavior	RTP ▼	
RFC 2833 Behavior	As Is ▼	
Alternative DTMF Method	As Is ▼	
→ P-Asserted-Identity	Add ▼	
Diversion Mode	As Is ▼	
History-Info Mode	As Is ▼	
Fax Coders Group ID	None ▼	
Fax Behavior	As Is ▼	
Fax Offer Mode	All coders ▼	
Fax Answer Mode	Single coder ▼	
PRACK Mode	Transparent ▼	
Session Expires Mode	Transparent ▼	
Remote Update Support	Supported ▼	
Remote re-INVITE	Supported ▼	
Remote Delayed Offer Support	Supported ▼	
→ Remote REFER Behavior	Handle Locally ▼	
Remote 3xx Behavior	Transparent ▼	
Remote Multiple 18x	Supported ▼	
Remote Early Media Response Type	Transparent ▼	
Remote Early Media	Supported ▼	
Enforce MKI Size	Don't enforce ▼	
Remote Early Media RTP Detection Mode	By Signaling ▼	
Remote RFC 3960 Gateway Model Support	Not Supported ▼	
Remote Can Play Ringback	Yes ▼	
RFC 2833 DTMF Payload Type	0	
User Registration Time	0	
Reliable Held Tone Source	Yes ▼	
Play Held Tone	No ▼	
→ Remote Hold Format	Send Only ▼	
Remote Replaces Behavior	Standard ▼	
SDP Ptime Answer	Remote Answer ▼	
Preferred PTime	0	
Use Silence Suppression	Transparent ▼	
RTP Redundancy Behavior	AS IS ▼	
Play RBT To Transferee	No ▼	
RTCP Mode	Transparent ▼	
Jitter Compensation	Disable ▼	
Remote Renegotiate on Fax Detection	Transparent ▼	
Remote Multiple Answers Mode	Disabled ▼	
Keep VIA Headers	Not Configured ▼	
Keep User-Agent Header	Not Configured ▼	
User Behind NAT UDP Registration Time	-1	
User Behind NAT TCP Registration Time	-1	
Adapt RFC2833 BW to Voice coder BW	Disabled ▼	
Submit Cancel		

4.7 Step 7: Configure Coders

This step describes how to configure coders (termed *Coder Group*). As DTAG SIP Trunk supports the G.711 coders, you need to add a Coder Group with the G.711 coder for the DTAG SIP Trunk.

Note that the Coder Group ID for this entity was assigned to its corresponding IP Profile in the previous step (see Section 4.5 on page 43).

➤ **To configure coders:**

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

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

Figure 4-18: Configuring Coder Group for DTAG SIP Trunk

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

The procedure below describes how to configure an Allowed Coders Group to ensure that voice sent to the DTAG SIP Trunk uses the G.711 coder only. Note that this Allowed Coders Group ID was assigned to the IP Profile belonging to the DTAG SIP Trunk (see Section 4.5 on page 43).

➤ **To set a preferred coder for the DTAG SIP Trunk:**

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

Parameter	Value
Allowed Audio Coders Group ID	0
Coder Name	G.711U-law
Coder Name	G.711A-law

Figure 4-19: Configuring Allowed Coders Group for DTAG SIP Trunk

Allowed Audio Coders Group				
Allowed Audio Coders Group ID	0			
<table border="1"> <thead> <tr> <th>Coder Name</th> </tr> </thead> <tbody> <tr> <td>G.711U-law</td> </tr> <tr> <td>G.711A-law</td> </tr> </tbody> </table>		Coder Name	G.711U-law	G.711A-law
Coder Name				
G.711U-law				
G.711A-law				

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

Figure 4-20: SBC Preferences Mode

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

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

4.8 Step 8: SIP TLS Connection Configuration

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

4.8.1 Step 8a: Configure the NTP Server Address

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

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

1. Open the Application Settings page (**Configuration** tab > **System** > **Time And Date**).
2. In the 'Primary NTP Server Address' field, enter the IP address of the NTP server (e.g., **10.15.25.1**).

Figure 4-21: Configuring NTP Server Address

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

3. Click **Submit**.

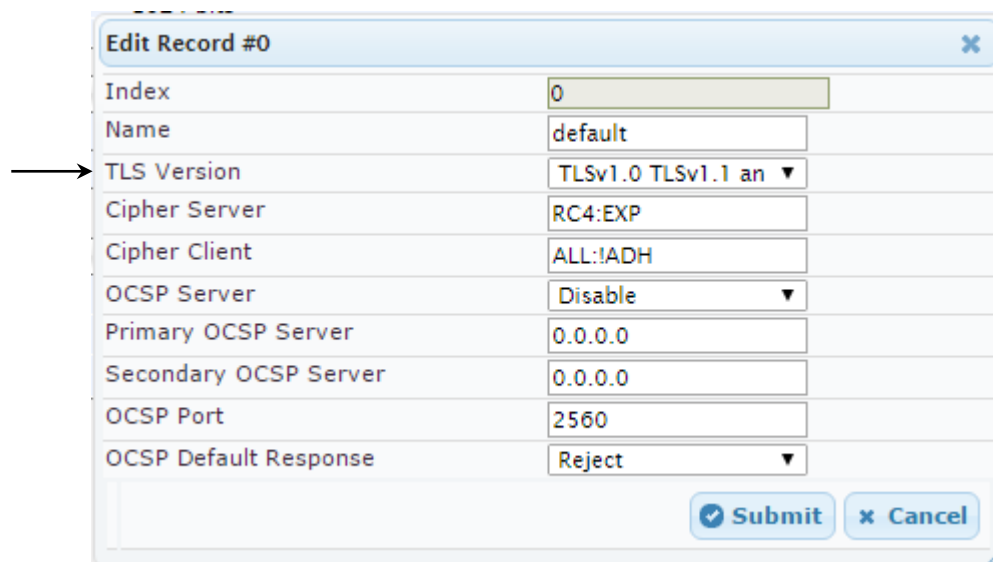
4.8.2 Step 8b: Configure the TLS version

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 page (**Configuration** tab > **System** menu > **TLS Contexts**).
2. In the TLS Contexts table, select the required TLS Context index row (usually default index 0 will be used), and then click 'Edit'.
3. From the 'TLS Version' drop-down list, select '**TLSv1.0 TLSv1.1 and TLSv1.2**'

Figure 4-22: Configuring TLS version



The screenshot shows a configuration window titled "Edit Record #0". It contains a table of settings for a TLS record. The "TLS Version" field is highlighted with a black arrow pointing to it from the left. The settings are as follows:

Field	Value
Index	0
Name	default
TLS Version	TLSv1.0 TLSv1.1 an ▼
Cipher Server	RC4:EXP
Cipher Client	ALL:!ADH
OCSP Server	Disable ▼
Primary OCSP Server	0.0.0.0
Secondary OCSP Server	0.0.0.0
OCSP Port	2560
OCSP Default Response	Reject ▼

At the bottom right of the dialog are two buttons: "Submit" (with a checkmark icon) and "Cancel" (with an 'x' icon).

4. Click **Submit**.

4.8.3 Step 8c: Configure a Certificate

This step describes how to exchange a certificate with Microsoft Certificate Authority (CA). The certificate is used by the E-SBC to authenticate the connection with 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 (**Configuration** tab > **System** menu > **TLS Contexts**).
2. In the TLS Contexts table, select the required TLS Context index row (usually default index 0 will be used), and then click the **TLS Context Certificates** button, located at the bottom of the TLS Contexts page; the Context Certificates page appears.
3. Under the **Certificate Signing Request** group, do the following:
 - a. In the 'Subject Name [CN]' field, enter the E-SBC FQDN name (e.g., **ITSP.S4B.interop**).
 - b. Fill in the rest of the request fields according to your security provider's instructions.
4. Click the **Create CSR** button; a textual certificate signing request is displayed in the area below the button:

Figure 4-23: Certificate Signing Request – Creating CSR

Certificate Signing Request

Subject Name [CN]	ITSP.S4B.interop
Organizational Unit [OU] (optional)	
Company name [O] (optional)	
Locality or city name [L] (optional)	
State [ST] (optional)	
Country code [C] (optional)	

Create CSR

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

```

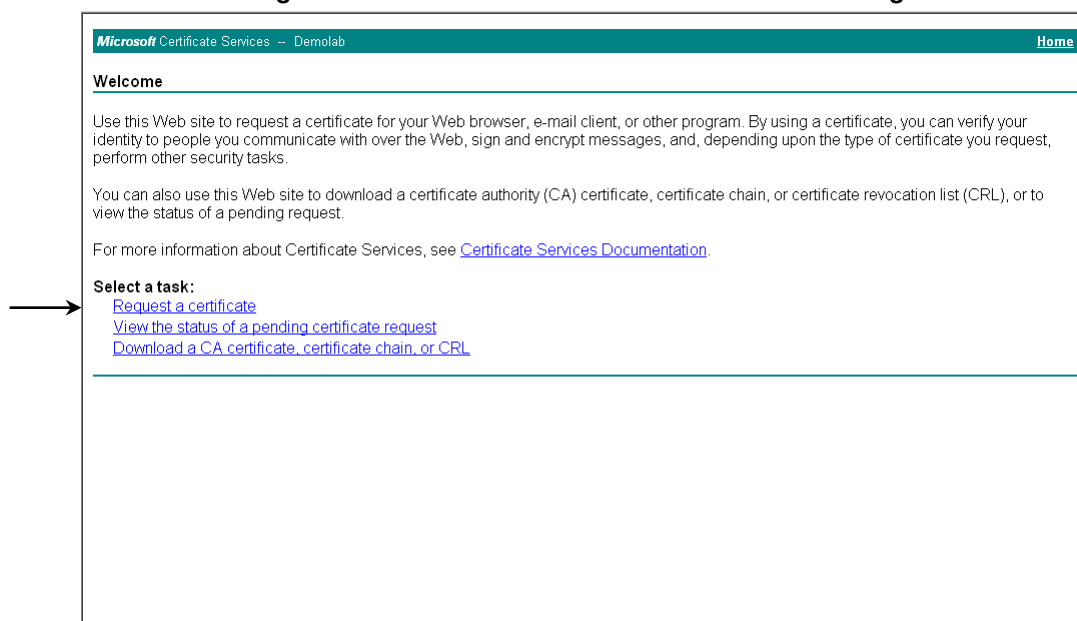
-----BEGIN CERTIFICATE REQUEST-----
MIIBWjCBxAIBADAbMRkwFwYDVQDDBBJVFNQLM0Q15pbmRlcm9wMIGfMA0GCSqG
SIb3DQEBAQUAA4GNADCBiQKBgQCzEs8XTnY8be/t77eEDG7rTg747GQ30DFOC4Rs
x+e9KfberZgxMYqGT8u04AU0wU9LUPkq+8gI6w2bg3bow0kg/9hrnNL2rf1tGcn
30oShP0SPikMRNznCC090b03tbr9kuHm1wPRQ7yT6k7xS3XBbSigqT4LQbjBT1tt
hDH3bQIDAQABoAAwDQYJKoZIhvcNAQEFBQADgYEAim/GA2ELzQbZaR6CZyIawilt
u65w450NFHmaCIuHsyZ8keM8d1Ux14hkw7t5ygAD8KbxVkhRvACgcQrAK2v8u1Pf
TVN+bwJ+kQ0d59CixA82e0o1WB3buPq5+qWDGTF+MyJWGVf8Sic1c6+zFoc+BEZY
7tQ8y0J8odo0aDhStdFQ=
-----END CERTIFICATE REQUEST-----
    
```



Note: The value entered in this field must be identical to the gateway name configured in the Topology Builder for Skype for Business Server 2015 (see Section 3.1 on page 13).

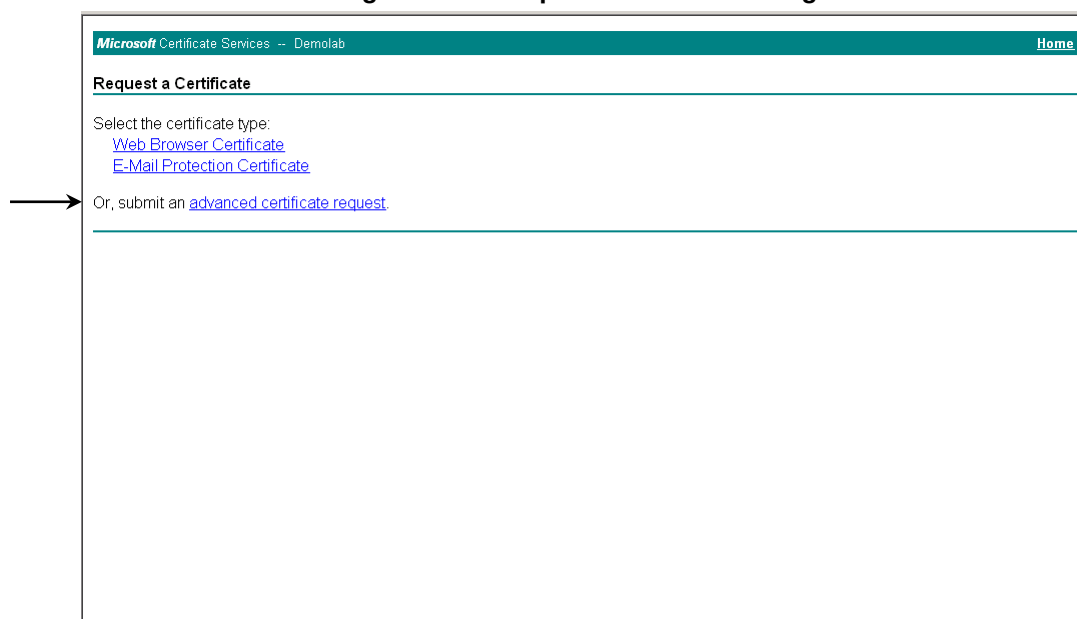
5. Copy the CSR from the line "**-----BEGIN CERTIFICATE REQUEST-----**" to a text file (such as Notepad), and then save it to a folder on your computer with the file name, *certreq.txt*.
6. Open a Web browser and navigate to the Microsoft Certificates Services Web site at <http://<certificate server>/CertSrv>.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

Figure 4-24: Microsoft Certificate Services Web Page



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

Figure 4-25: Request a Certificate Page



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

Figure 4-26: Advanced Certificate Request Page

Figure 4-27: Submit a Certificate Request or Renewal Request Page

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

Figure 4-28: Certificate Issued Page

13. Select the **Base 64 encoded** option for encoding, and then click **Download certificate**.

14. Save the file as *gateway.cer* to a folder on your computer.
15. Click the **Home** button or navigate to the certificate server at <http://<Certificate Server>/CertSrv>.
16. Click **Download a CA certificate, certificate chain, or CRL**.

Figure 4-29: Download a CA Certificate, Certificate Chain, or CRL Page

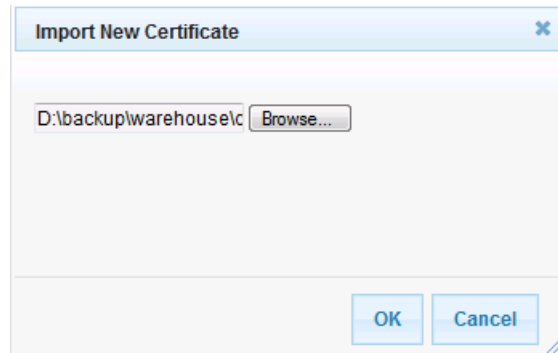
17. Under the 'Encoding method' group, select the **Base 64** option for encoding.
18. Click **Download CA certificate**.
19. Save the file as *certroot.cer* to a folder on your computer.
20. In the E-SBC's Web interface, return to the **TLS Contexts** page and do the following:
 - a. In the TLS Contexts table, select the required TLS Context index row (typically, the default TLS Context at Index 0 is used), and then click the **TLS Context Certificates**  button, located at the bottom of the TLS Contexts page; the Context Certificates page appears.
 - b. Scroll down to the **Upload certificates files from your computer** group, click the **Browse** button corresponding to the 'Send Device Certificate...' field, navigate to the *gateway.cer* certificate file that you saved on your computer in Step 14, and then click **Send File** to upload the certificate to the E-SBC.

Figure 4-30: Upload Device Certificate Files from your Computer Group

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

- d. In the TLS Contexts table, select the required TLS Context index row, and then click the **TLS Context Trusted-Roots Certificates**  button, located at the bottom of the TLS Contexts page; the Trusted Certificates page appears.
- e. Click the **Import** button, and then select the certificate file to load.

Figure 4-31: Importing Root Certificate into Trusted Certificates Store



21. Click **OK**; the certificate is loaded to the device and listed in the Trusted Certificates store.
22. Reset the E-SBC with a burn to flash for your settings to take effect (see Section 4.15 on page 87).

4.9 Step 9: Configure SRTP

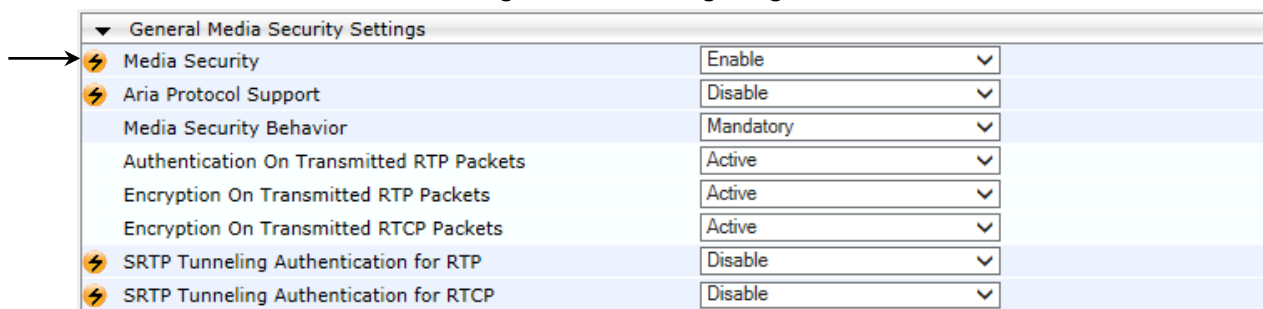
This step describes how to configure media security. If you configure the Microsoft Mediation Server to use SRTP, you need to configure the E-SBC to operate in the same manner. Note that SRTP was enabled for Skype for Business Server 2015 when you configured an IP Profile for Skype for Business Server 2015 (see Section 4.5 on page 43).

➤ **To configure media security:**

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

Parameter	Value
Media Security	Enable

Figure 4-32: Configuring SRTP



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

3. Click **Submit**.
4. Reset the E-SBC with a burn to flash for your settings to take effect (see Section 4.15 on page 87).

4.10 Step 10: Configure IP-to-IP Call Routing Rules

This step describes how to configure IP-to-IP call routing rules. These rules define the routes for forwarding SIP messages (e.g., INVITE) received from one IP entity to another. The E-SBC selects the rule whose configured input characteristics (e.g., IP Group) match those of the incoming SIP message. If the input characteristics do not match the first rule in the table, they are compared to the second rule, and so on, until a matching rule is located. If no rule is matched, the message is rejected. The routing rules use the configured IP Groups to denote the source and destination of the call. As configured in Section 4.5 on page 43, IP Group 1 represents Skype for Business Server 2015, and IP Group 2 represents DTAG SIP Trunk.

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

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

➤ To configure IP-to-IP routing rules:

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

Parameter	Value
Index	0
Route Name	Terminate OPTIONS (arbitrary descriptive name)
Request Type	OPTIONS

Figure 4-33: Configuring IP-to-IP Routing Rule for Terminating SIP OPTIONS – Rule Tab

Rule	Action
Index	0
Route Name	Terminate OPTIONS
Source IP Group ID	-1
Source Username Prefix	*
Source Host	*
Destination Username Prefix	*
Destination Host	*
Request Type	OPTIONS
Message Condition	None
ReRoute IP Group ID	-1
Call Trigger	Any
Call Setup Rules Set ID	-1

Submit Cancel

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

Parameter	Value
Destination Type	Dest Address
Destination Address	internal

Figure 4-34: Configuring IP-to-IP Routing Rule for Terminating SIP OPTIONS – Action Tab

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

Submit Cancel

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

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

Figure 4-35: Configuring IP-to-IP Routing Rule for S4B to ITSP – Rule tab

Field	Value
Index	1
Route Name	S4B to ITSP
Source IP Group ID	1
Source Username Prefix	*
Source Host	*
Destination Username Prefix	*
Destination Host	*
Request Type	All
Message Condition	None
ReRoute IP Group ID	-1
Call Trigger	Any
Call Setup Rules Set ID	-1

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

Parameter	Value
Destination Type	IP Group
Destination IP Group ID	2
Destination SRD ID	1

Figure 4-36: Configuring IP-to-IP Routing Rule for S4B to ITSP – Action tab

The screenshot shows the 'Action' tab of a configuration window. The 'Rule' tab is also visible. The 'Action' tab contains the following fields:

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

Arrows point to the Destination Type, Destination IP Group ID, and Destination SRD ID fields. The 'Submit' and 'Cancel' buttons are at the bottom right.

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

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

Figure 4-37: Configuring IP-to-IP Routing Rule for ITSP to S4B – Rule tab

The screenshot shows a configuration window with two tabs: "Rule" (selected) and "Action". The "Rule" tab contains the following fields and values:

- Index:** 2 (indicated by an arrow)
- Route Name:** ITSP to S4B (indicated by an arrow)
- Source IP Group ID:** 2 (indicated by an arrow)
- Source Username Prefix:** *
- Source Host:** *
- Destination Username Prefix:** *
- Destination Host:** *
- Request Type:** All (dropdown menu)
- Message Condition:** None (dropdown menu)
- ReRoute IP Group ID:** -1
- Call Trigger:** Any (dropdown menu)
- Call Setup Rules Set ID:** -1

At the bottom right, there are "Submit" and "Cancel" buttons.

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

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

Figure 4-38: Configuring IP-to-IP Routing Rule for ITSP to S4B – Action tab

The screenshot shows the 'Action' tab of the IP-to-IP Routing Rule configuration. The fields are as follows:

Field	Value
Index	2
Destination Type	IP Group
Destination IP Group ID	1
Destination SRD ID	0
Destination Address	
Destination Port	0
Destination Transport Type	
Alternative Route Options	Route Row
Group Policy	None
Cost Group	None

Buttons: Submit, Cancel

The configured routing rules are shown in the figure below:

Figure 4-39: Configured IP-to-IP Routing Rules in IP-to-IP Routing Table

The screenshot shows the 'IP-to-IP Routing Table' with the following data:

Index	Route Name	Source Host	Destination Username Prefix	Destination Host	Message Condition	ReRoute IP Group ID	Call Trigger	Call Setup Rules Set ID	Destination Type	Destination SRD ID
0	Terminate Of	*	*	*	None	-1	Any	-1	Dest Address	None
1	S4B to ITSP	*	*	*	None	-1	Any	-1	IP Group	1
2	ITSP to S4B	*	*	*	None	-1	Any	-1	IP Group	0

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



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

4.11 Step 11: Configure IP-to-IP Manipulation Rules

This step describes how to configure IP-to-IP manipulation rules. These rules manipulate the source and / or destination number. The manipulation rules use the configured IP Groups to denote the source and destination of the call. As configured in Section 4.5 on page 43, IP Group 0 represents Skype for Business Server 2015, and IP Group 1 represents DTAG SIP Trunk.



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

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

➤ To configure a number manipulation rule:

1. Open the IP-to-IP Outbound Manipulation page (**Configuration** tab > **VoIP** menu > **SBC > Manipulations SBC > IP-to-IP Outbound**).
2. Click **Add**.
3. Click the **Rule** tab, and then configure the parameters as follows:

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

Figure 4-40: Configuring IP-to-IP Outbound Manipulation Rule – Rule Tab

The screenshot shows the 'Rule' tab of the configuration form. The fields and their values are as follows:

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

Buttons at the bottom: Submit, Cancel.

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

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

Figure 4-41: Configuring IP-to-IP Outbound Manipulation Rule - Action Tab

5. Click **Submit**.

The figure below shows an example of configured IP-to-IP outbound manipulation rules for calls between IP Group 1 (i.e., Skype for Business Server 2015) and IP Group 2 (i.e., DTAG SIP Trunk):

Figure 4-42: Example of Configured IP-to-IP Outbound Manipulation Rules

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

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

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

4.12 Step 12: Configure Message Manipulation Rules

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

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

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

1. Open the Message Manipulations page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **Msg Policy & Manipulation** > **Message Manipulations**).
2. Configure a new manipulation rule (Manipulation Set 2) for the Skype for Business Server 2015. This rule applies to messages sent to the Skype for Business Server 2015 IP Group. This replaces the user part of the SIP Request-URI Header with the value from the SIP To Header.

Parameter	Value
Index	0
Manipulation Name	Change RequestURI
Manipulation Set ID	2
Message Type	invite
Condition	
Action Subject	header.request-uri.url.user
Action Type	Modify
Action Value	header.to.url.user

Figure 4-43: Configuring SIP Message Manipulation Rule 0 (for Skype for Business)

Edit Record #0

Index: 0

Manipulation Name: Change RequestURI

Manipulation Set ID: 2

Message Type: invite

Condition:

Action Subject: header.request-uri.url.u

Action Type: Modify

Action Value: header.to.url.user

Row Role: Use Current Condit

Submit Cancel

3. Configure another manipulation rule (Manipulation Set 4) for DTAG SIP Trunk. This rule is applied to messages sent to the DTAG SIP Trunk IP Group for Call Forward initiated by the Skype for Business Server 2015 IP Group. This replaces the user part of the SIP From Header with the value from the SIP History-Info Header.

Parameter	Value
Index	1
Manipulation 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	\$2

Figure 4-44: Configuring SIP Message Manipulation Rule 1 (for DTAG SIP Trunk)

The screenshot shows a web-based configuration interface for SIP Message Manipulation Rules. The 'Edit Record #1' dialog box is open, displaying the following configuration details:

- Index:** 1
- Manipulation Name:** Call Forward
- Manipulation Set ID:** 4
- Message Type:** invite
- Condition:** header.history-info.0 re
- Action Subject:** header.from.url.user
- Action Type:** Modify
- Action Value:** \$2
- Row Role:** Use Current Condit

At the bottom right of the dialog are 'Submit' and 'Cancel' buttons. On the left side of the dialog, arrows point to each of the configuration fields.

4. If manipulation rule index 1 (above) is executed, then the following rule is also executed. This rule is applied to messages sent to the DTAG SIP Trunk IP Group for Call Forward initiated by the Skype for Business Server 2015 IP Group. This removes the SIP History-Info Header.

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

Figure 4-45: Configuring SIP Message Manipulation Rule 2 (for DTAG SIP Trunk)

The screenshot shows a web-based configuration interface for SIP message manipulation. The dialog box is titled 'Edit Record #2'. It contains several input fields and dropdown menus. Arrows on the left side of the image point to specific fields: Index, Manipulation Name, Manipulation Set ID, Action Subject, Action Type, and Row Role. The values entered in these fields are: Index: 2, Manipulation Name: Call Forward, Manipulation Set ID: 4, Action Subject: header.history-info, Action Type: Remove, and Row Role: Use Previous Condi. The dialog also includes a 'Submit' button and a 'Cancel' button at the bottom right.

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

5. If manipulation rule index 1 (above) is executed, then the following rule is also executed. This rule is applied to messages sent to the DTAG SIP Trunk IP Group for Call Forward initiated by the Skype for Business Server 2015 IP Group. This replaces the user part of the SIP P-Asserted-Identity Header with the value from the SIP From Header.

Parameter	Value
Index	3
Manipulation Name	Call Forward
Manipulation Set ID	4
Action Subject	header.p-asserted-identity.url.user
Action Type	Modify
Action Value	header.from.url.user
Row Role	Use Previous Condition

Figure 4-46: Configuring SIP Message Manipulation Rule 3 (for DTAG SIP Trunk)

Edit Record #3

→ Index: 3

→ Manipulation Name: Call Forward

→ Manipulation Set ID: 4

Message Type:

Condition:

→ Action Subject: header.p-asserted-iden

→ Action Type: Modify

→ Action Value: header.from.url.user

→ Row Role: Use Previous Condi

Submit Cancel

6. Configure another manipulation rule (Manipulation Set 4) for DTAG SIP Trunk. This rule is applied to messages sent to the DTAG SIP Trunk IP Group during Call Transfer initiated by the Skype for Business Server 2015 IP Group. This replaces the user part of the SIP From Header with the value from the SIP Referred-By Header.

Parameter	Value
Index	4
Manipulation 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 4-47: Configuring SIP Message Manipulation Rule 4 (for DTAG SIP Trunk)

Edit Record #4

→ Index: 4

→ Manipulation 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

Row Role: Use Current Condit

Submit Cancel

7. If manipulation rule index 4 (above) is executed, then the following rule is also executed. This rule is applied to messages sent to the DTAG SIP Trunk IP Group for Call Forward initiated by the Skype for Business Server 2015 IP Group. This replaces the user part of the SIP P-Asserted-Identity Header with the value from the SIP From Header.

Parameter	Value
Index	5
Manipulation Name	Call Transfer
Manipulation Set ID	4
Action Subject	header.p-asserted-identity.url.user
Action Type	Modify
Action Value	header.from.url.user
Row Role	Use Previous Condition

Figure 4-48: Configuring SIP Message Manipulation Rule 5 (for DTAG SIP Trunk)

Edit Record #5

Index	5
Manipulation Name	Call Transfer
Manipulation Set ID	4
Message Type	
Condition	
Action Subject	header.p-asserted-iden
Action Type	Modify
Action Value	header.from.url.user
Row Role	Use Previous Condi

Submit Cancel

8. If manipulation rule index 4 (above) is executed, then the following rule is also executed. This rule is applied to messages sent to the DTAG SIP Trunk IP Group for Call Forward initiated by the Skype for Business Server 2015 IP Group. This replaces the host part of the SIP Referred-By Header with the value "SIP Group Name", configured for the DTAG SIP Trunk IP Group.

Parameter	Value
Index	6
Manipulation Name	Call Transfer
Manipulation Set ID	4
Action Subject	header.referred-by.url.host
Action Type	Modify
Action Value	param.ipg.dst.host
Row Role	Use Previous Condition

Figure 4-49: Configuring SIP Message Manipulation Rule 6 (for DTAG SIP Trunk)

Edit Record #6

→ Index: 6

→ Manipulation Name: Call Transfer

→ Manipulation Set ID: 4

Message Type:

Condition:

→ Action Subject: header.referred-by.url.l

→ Action Type: Modify

→ Action Value: param.ipg.dst.host

→ Row Role: Use Previous Condi

Submit Cancel

9. Configure another manipulation rule (Manipulation Set 4) for DTAG SIP Trunk. This rule is applied to response messages sent to the DTAG SIP Trunk IP Group for Rejected Calls initiated by the Skype for Business Server 2015 IP Group. This replaces the method type '503' or '603' with the value '486', because DTAG SIP Trunk not recognizes '503' or '603' method types.

Parameter	Value
Index	7
Manipulation Name	Responses
Manipulation Set ID	4
Message Type	any.response
Condition	header.request-uri.methodtype=='503' OR header.request-uri.methodtype=='603'
Action Subject	header.request-uri.methodtype
Action Type	Modify
Action Value	'486'

Figure 4-50: Configuring SIP Message Manipulation Rule 7 (for DTAG SIP Trunk)

The screenshot shows a web-based configuration interface for SIP Message Manipulation Rule 7. The dialog box is titled "Edit Record #7" and contains the following fields:

- Index:** 7
- Manipulation Name:** Responses
- Manipulation Set ID:** 4
- Message Type:** any.response
- Condition:** header.request-uri.metl
- Action Subject:** header.request-uri.metl
- Action Type:** Modify (dropdown menu)
- Action Value:** '486'
- Row Role:** Use Current Condit (dropdown menu)

Arrows on the left side of the dialog point to the Index, Manipulation Name, Manipulation Set ID, Message Type, Condition, Action Subject, Action Type, and Action Value fields. At the bottom right, there are "Submit" and "Cancel" buttons.

Figure 4-51: Configured SIP Message Manipulation Rules

▼ Message Manipulations							
Add +		Insert +					
Index	Manipulation Name	Manipulation Set ID	Message Type	Condition	Action Subject	Action Type	Action Value
0	Change RequestURI	2	invite		header.request-uri.u	Modify	header.to.url.i
1	Call Forward	4	invite	header.history-info.(	header.from.url.use	Modify	\$2
2	Call Forward	4			header.history-info	Remove	
3	Call Forward	4			header.p-asserted-i	Modify	header.from.u
4	Call Transfer	4	invite	header.referred-by	header.from.url.use	Modify	header.referr
5	Call Transfer	4			header.p-asserted-i	Modify	header.from.u
6	Call Transfer	4			header.referred-by	Modify	param.ipg.dst
7	Responses	4	any.response	header.request-uri.r	header.request-uri.r	Modify	'486'
⏪ ⏩ Page 1 of 1 ⏪ ⏩ Show 10 records per page View 1 - 8 of 8							

The table displayed below includes SIP message manipulation rules which are grouped together under Manipulation Set IDs (Manipulation Set IDs 2 and 4) and which are executed for messages sent to and from the DTAG SIP Trunk IP Group as well as the Skype for Business Server 2015 IP Group. These rules are specifically required to enable proper interworking between DTAG 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 applies to messages sent to the Skype for Business Server 2015 IP Group. This replaces the user part of the SIP Request-URI Header with the value from the SIP To Header.	
1	This rule is applied to messages sent to the DTAG SIP Trunk IP Group for Call Forward initiated by the Skype for Business Server 2015 IP Group. This replaces the user part of the SIP From Header with the value from the SIP History-Info Header.	For Call Forward scenarios, DTAG SIP Trunk needs that User part in SIP From Header will be defined number. In order to do this, User part of the SIP From Header replaced with the value from History-Info Header.
2	If manipulation rule index 1 (above) is executed, then the following rule is also executed. This removes the SIP History-Info Header.	
3	If manipulation rule index 1 (above) is executed, then the following rule is also executed. This replaces the user part of the SIP P-Asserted-Identity Header with the value from the SIP From Header	
4	This rule is applied to messages sent to the DTAG SIP Trunk IP Group during Call Transfer initiated by the Skype for Business Server 2015 IP Group. This replaces the user part of the SIP From Header with the value from the SIP Referred-By Header.	For Call Transfer initiated by Skype for Business Server 2015, DTAG 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.
5	If manipulation rule index 4 (above) is executed, then the following rule is also executed. This replaces the user part of the SIP P-Asserted-Identity Header with the value from the SIP From Header.	
6	If manipulation rule index 4 (above) is executed, then the following rule is also executed. This replaces the host part of the SIP Referred-By Header with the value "SIP Group Name", configured for the DTAG SIP Trunk IP Group.	
7	This rule is applied to response messages sent to the DTAG SIP Trunk IP Group for Rejected Calls initiated by the Skype for Business Server 2015 IP Group. This replaces the method type '503' or '603' with the value '486'.	DTAG SIP Trunk not recognizes '503' or '603' method types.

10. Assign Manipulation Set IDs 1 and 2 to the Skype for Business 2015 IP Group:
 - a. Open the IP Group Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **IP Group Table**).
 - b. Select the row of the Skype for Business 2015 IP Group, and then click **Edit**.
 - c. Click the **SBC** tab.
 - d. Set the 'Outbound Message Manipulation Set' field to 2.

Figure 4-52: Assigning Manipulation Set to the Skype for Business 2015 IP Group

Common		GW		SBC	
Index	1				
Classify By Proxy Set	Enable				
Max. Number of Registered Users	-1				
Inbound Message Manipulation Set	-1				
Outbound Message Manipulation Set	2				
Registration Mode	User Initiates Regis				
Authentication Mode	User Authenticates				
Authentication Method List					
SBC Client Forking Mode	Sequential				
Source URI Input					
Destination URI Input					
Username	Admin				
Password	•				
Msg Man User Defined String1					
Msg Man User Defined String2					
SIP Connect	No				
Route Using Request URI Port	Disable				
Submit Cancel					

- e. Click **Submit**.

11. Assign Manipulation Set ID 4 to the DTAG SIP trunk IP Group:
 - a. Open the IP Group Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **IP Group Table**).
 - b. Select the row of the DTAG SIP trunk IP Group, and then click **Edit**.
 - c. Click the **SBC** tab.
 - d. Set the 'Outbound Message Manipulation Set' field to 4.

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

The screenshot shows the 'SBC' configuration tab for an IP Group. The 'Outbound Message Manipulation Set' field is highlighted with an arrow and set to the value '4'. Other fields include 'Index' (2), 'Classify By Proxy Set' (Enable), 'Max. Number of Registered Users' (-1), 'Inbound Message Manipulation Set' (-1), 'Registration Mode' (User Initiates Regis), 'Authentication Mode' (User Authenticates), 'Authentication Method List' (empty), 'SBC Client Forking Mode' (Sequential), 'Source URI Input' (empty), 'Destination URI Input' (empty), 'Username' (Admin), 'Password' (masked), 'Msg Man User Defined String1' (empty), 'Msg Man User Defined String2' (empty), 'SIP Connect' (No), and 'Route Using Request URI Port' (Disable). The 'Submit' and 'Cancel' buttons are at the bottom right.

Common		GW		SBC	
Index	2				
Classify By Proxy Set	Enable				
Max. Number of Registered Users	-1				
Inbound Message Manipulation Set	-1				
Outbound Message Manipulation Set	4				
Registration Mode	User Initiates Regis				
Authentication Mode	User Authenticates				
Authentication Method List					
SBC Client Forking Mode	Sequential				
Source URI Input					
Destination URI Input					
Username	Admin				
Password	•				
Msg Man User Defined String1					
Msg Man User Defined String2					
SIP Connect	No				
Route Using Request URI Port	Disable				
Submit Cancel					

- e. Click **Submit**.

4.13 Step 13: Configure Registration Accounts

This step describes how to configure SIP registration accounts. This is required so that the E-SBC can register with the DTAG SIP Trunk on behalf of Skype for Business Server 2015. The DTAG 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 1) and the Serving IP Group is DTAG SIP Trunk (IP Group 2).

➤ **To configure a registration account:**

1. Open the Account Table page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **Account Table**).
2. Click **Add**.
3. Configure the account according to the provided information from DTAG, for example:

Parameter	Value
Served IP Group	1 (Skype for Business Server 2015)
Serving IP Group	2 (DTAG SIP Trunk)
User Name	As provided by DTAG
Password	As provided by DTAG
Host Name	sip-trunk.telekom.de
Register	Regular
Contact User	+496987409358 (trunk main line)
Application Type	SBC

4. Click **Add**.

Figure 4-54: Configuring a SIP Registration Account

The screenshot shows a web-based configuration interface for a SIP Registration Account. The dialog box is titled 'Edit Record #0'. It contains the following fields and values:

- Index: 0
- Served Trunk Group: -1
- Served IP Group: 1
- Serving IP Group: 2
- User Name: audiocodes-vosip-reg1
- Password: (masked with dots)
- Host Name: sip-trunk.telekom.de
- Register: Regular (dropdown menu)
- Contact User: +496987409358
- Application Type: SBC (dropdown menu)

Arrows on the left side of the dialog point to the Served IP Group, Serving IP Group, User Name, Password, Host Name, Register, Contact User, and Application Type fields. At the bottom right, there are 'Submit' and 'Cancel' buttons.

4.14 Step 14: Miscellaneous Configuration

This section describes miscellaneous E-SBC configuration.

4.14.1 Step 14a: 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 General Settings page (**Configuration** tab > **VoIP** menu > **SBC** > **General Settings**).
2. From the 'SBC Forking Handling Mode' drop-down list, select **Sequential**.

Figure 4-55: Configuring Forking Mode

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

3. Click **Submit**.

4.14.2 Step 14b: Configure DNS Query Type

This step describes how to configure SRV DNS Query Type in the E-SBC. In the incoming calls from the DTAG SIP Trunk, SIP Record-Route Header represented as FQDN. In order to resolve IP address for any responses, DNS Query Type should be configured as SRV.

➤ **To configure DNS Query Type:**

1. Open the Proxy & Registration page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **Proxy & Registration**).
2. From the 'DNS Query Type' drop-down list, select **SRV**.
3. From the 'Proxy DNS Query Type' drop-down list, select **SRV**.

Figure 4-56: Configuring DNS Query Type

Proxy & Registration	
Use Default Proxy	No
Proxy Name	
Redundancy Mode	Parking
Proxy IP List Refresh Time	60
Enable Fallback to Routing Table	Disable
Prefer Routing Table	No
Always Use Proxy	Disable
Redundant Routing Mode	Routing Table
SIP ReRouting Mode	Standard Mode
Enable Registration	Disable
Registration Time	360
Re-registration Timing [%]	50
Registration Retry Time	30
Registration Time Threshold	0
Re-register On INVITE Failure	Disable
ReRegister On Connection Failure	Disable
Gateway Name	sip-trunk.telekom.de
Gateway Registration Name	
DNS Query Type	SRV
Proxy DNS Query Type	SRV
Subscription Mode	Per Endpoint
Number of RTX Before Hot-Swap	3
Use Gateway Name for OPTIONS	Yes

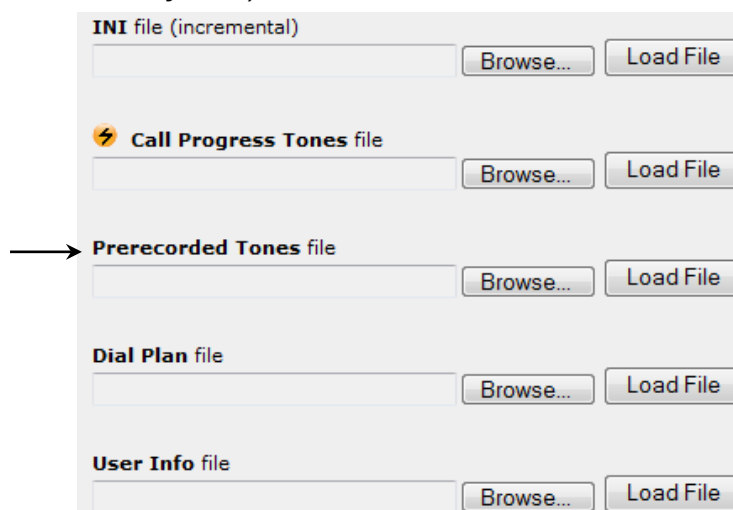
4. Click **Submit**.

4.14.3 Step 14c: Loading Prerecorded Tones File

This step describes how to load prerecorded tones file in order to overcome problem with first incoming RTP packet in call forwarding scenario, when Skype for Business user forward call to PSTN user. In this scenario, instead of generating Ringback Tone as Call Progress Tone (CPT), which requires DSP we decided to use Prerecorded Tones (PRT) file for ringback tones.

➤ To load PRT file to the device using the Web interface:

1. Open the Load Auxiliary Files page (**Maintenance** tab > **Software Update** menu > **Load Auxiliary Files**).



Note: The appearance of certain file load fields depends on the installed Software License Key.

2. Click the **Browse** button corresponding to the **Prerecorded Tones** file type that you want to load, navigate to the folder in which the file is located, and then click **Open**; the name and path of the file appear in the field next to the **Browse** button.
3. Click the **Load File** button corresponding to the file you want to load.
4. Save the loaded auxiliary files to flash memory.

4.14.4 Step 14d: Configure RTP Port for T.38 Fax

This step describes how to configure E-SBC to use the same RTP port for T.38 Fax for incoming fax.

➤ **To configure use RTP port for T.38 fax:**

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

Figure 4-57: Configuring SBC Session Refreshing Policy in AdminPage

Parameter Name: Enter Value:

Output Window

```
Parameter Name: T38USERTPPORT
Parameter New Value: 1
Parameter Description:T.38 packets will be received on RTP port
```

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

Parameter	Value
T38UseRTPPort	1

5. Click the **Apply New Value** button for each field.

4.15 Step 15: Reset the E-SBC

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

➤ **To save the configuration to flash memory:**

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

Figure 4-58: Resetting the E-SBC

The screenshot shows a web interface for resetting the E-SBC. It is divided into three main sections: 'Reset Configuration', 'LOCK / UNLOCK', and 'Save Configuration'. In the 'Reset Configuration' section, there is a 'Reset Board' button, a 'Burn To FLASH' dropdown menu set to 'Yes', and a 'Graceful Option' dropdown menu set to 'No'. In the 'LOCK / UNLOCK' section, there is a 'Lock' button, a 'Graceful Option' dropdown menu set to 'No', and a 'Gateway Operational State' label showing 'UNLOCKED'. In the 'Save Configuration' section, there is a 'Burn To FLASH' button labeled 'BURN'. An arrow points to the 'Burn To FLASH' dropdown menu in the 'Reset Configuration' section.

2. Ensure that the 'Burn to FLASH' field is set to **Yes** (default).
3. Click the **Reset** button.

This page is intentionally left blank.

A AudioCodes INI File

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



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

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

;Board: Mediant 500L - MSBR
;HW Board Type: 69  FK Board Type: 84
;Serial Number: 5817015
;Slot Number: 1
;Software Version: 6.80A.300.009
;DSP Software Version: 5011AE3_R => 680.31
;Board IP Address: 10.15.17.33
;Board Subnet Mask: 255.255.0.0
;Board Default Gateway: 10.15.17.34
;Ram size: 369M  Flash size: 64M  Core speed: 300Mhz
;Num of DSP Cores: 1  Num DSP Channels: 26
;Num of physical LAN ports: 4
;Profile: NONE
;;Key features;;Board Type: 84 ;Security: IPSEC MediaEncryption
StrongEncryption EncryptControlProtocol ;DSP Voice features: RTCP-XR
;DATA features: Routing FireWall&VPN WAN BGP Advanced-Routing 3G Shdsl-
Pairs=1 WIFI-COUNTRY-CODE=0x318 FTTX-WAN T1E1-Wan-Trunks=2 ;Channel Type:
DspCh=50 IPMediaDspCh=50 ;HA ;IP Media: VXML ;QOE features:
VoiceQualityMonitoring MediaEnhancement ;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 ;FXSPorts=4
;FXOPorts=4 ;Control Protocols: MGCP SIP SASurvivability SBC=60 MSFT
FEU=100 TestCall=100 ;Default features;;Coders: G711 G726;

;----- HW components-----
;
; Slot # : Module type : # of ports
;-----
;      2 : FXS          : 4
;      3 : FX0          : 4
;-----

[SYSTEM Params]

SyslogServerIP = 10.15.17.100
EnableSyslog = 1
;NTPServerIP_abs is hidden but has non-default value
TelnetServerIdleDisconnect = 1000
;VpFileLastUpdateTime is hidden but has non-default value
NTPServerIP = '10.15.27.1'
LdapSearchServerMethod = 0
```

```

Tr069TLSContext = 0
;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]

PrerecordedTonesFileName = 'RingbackTone-Guitar-U-law.dat'
ENABLEMEDIASECURITY = 1
SRTPTxPacketMKISize = 1

[WEB Params]

SharedSecret = '$1$woS2sLC0opqIjoKZng== '
UserProductName = 'Mediant 500L - MSBR'
LogoWidth = '145'
UseProductName = 1
HTTPSCipherString = 'RC4:EXP'

[SIP Params]

REGISTRATIONTIME = 360
GWDEBUGLEVEL = 5

```

```

;ISPRACKREQUIRED is hidden but has non-default value
SIPGATEWAYNAME = 'sip-trunk.telekom.de'
T38USERTPPPORT = 1
USEGATEWAYNAMEFOROPTIONS = 1
;ENABLEPROXYSRVQUERY is hidden but has non-default value
;ENABLESRVQUERY is hidden but has non-default value
DNSQUERYTYPE = 1
PROXYDNSQUERYTYPE = 1
ENABLESBCAPPLICATION = 1
MSLDAPPRIMARYKEY = 'telephoneNumber'
SBCPREFERENCESENABLEMODE = 1
SBCFORKINGHANDLINGMODE = 1
ENERGYDETECTORCMD = 587202560
ANSWERDETECTORCMD = 10486144

[SCTP Params]

[IPsec Params]

[Audio Staging Params]

[SNMP Params]

[ DeviceTable ]

FORMAT DeviceTable_Index = DeviceTable_VlanID,
DeviceTable_UnderlyingInterface, DeviceTable_DeviceName;
DeviceTable 0 = 1, "", "vlan 1";

[ \DeviceTable ]

[ InterfaceTable ]

FORMAT InterfaceTable_Index = InterfaceTable_ApplicationTypes,
InterfaceTable_InterfaceMode, InterfaceTable_IPAddress,
InterfaceTable_PrefixLength, InterfaceTable_Gateway,
InterfaceTable_InterfaceName, InterfaceTable_PrimaryDNSServerIPAddress,
InterfaceTable_SecondaryDNSServerIPAddress,
InterfaceTable_UnderlyingDevice;
InterfaceTable 0 = 6, 10, 10.15.17.33, 16, 10.15.17.34, "Voice",
10.15.27.1, 0.0.0.0, "vlan 1";

[ \InterfaceTable ]

[ DspTemplates ]

;
; *** TABLE DspTemplates ***
; This table contains hidden elements and will not be exposed.
; This table exists on board and will be saved during restarts.
;

```

```
[ \DspTemplates ]

[ CpMediaRealm ]

FORMAT CpMediaRealm_Index = CpMediaRealm_MediaRealmName,
CpMediaRealm_IPv4IF, CpMediaRealm_IPv6IF, CpMediaRealm_PortRangeStart,
CpMediaRealm_MediaSessionLeg, CpMediaRealm_PortRangeEnd,
CpMediaRealm_IsDefault, CpMediaRealm_QoeProfile, CpMediaRealm_BWProfile;
CpMediaRealm 0 = "MRLan", "Voice", "", 6000, 100, 6990, 1, "", "";
CpMediaRealm 1 = "MRWan", "WAN", "", 7000, 100, 7990, 0, "", "";

[ \CpMediaRealm ]

[ SRD ]

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

[ \SRD ]

[ ProxyIp ]

FORMAT ProxyIp_Index = ProxyIp_IpAddress, ProxyIp_TransportType,
ProxyIp_ProxySetId;
ProxyIp 0 = "FE.S4B.interop:5067", 2, 1;
ProxyIp 1 = "reg.sip-trunk.telekom.de", 1, 2;

[ \ProxyIp ]

[ IpProfile ]

FORMAT IpProfile_Index = IpProfile_ProfileName, IpProfile_IpPreference,
IpProfile_CodersGroupID, IpProfile_IsFaxUsed,
IpProfile_JitterBufMinDelay, IpProfile_JitterBufOptFactor,
IpProfile_IPDiffServ, IpProfile_SigIPDiffServ, IpProfile_SCE,
IpProfile_RTPRedundancyDepth, IpProfile_RemoteBaseUDPPort,
IpProfile_CNGmode, IpProfile_VxxTransportType, IpProfile_NSEMode,
IpProfile_IsDTMFUsed, IpProfile_PlayRBTone2IP,
IpProfile_EnableEarlyMedia, IpProfile_ProgressIndicator2IP,
IpProfile_EnableEchoCanceller, IpProfile_CopyDest2RedirectNumber,
IpProfile_MediaSecurityBehaviour, IpProfile_CallLimit,
IpProfile_DisconnectOnBrokenConnection, IpProfile_FirstTxDtmfOption,
IpProfile_SecondTxDtmfOption, IpProfile_RxDTMFOption,
IpProfile_EnableHold, IpProfile_InputGain, IpProfile_VoiceVolume,
IpProfile_AddIEInSetup, IpProfile_SBCExtensionCodersGroupID,
IpProfile_MediaIPVersionPreference, IpProfile_TranscodingMode,
IpProfile_SBCAllowedMediaTypes, IpProfile_SBCAllowedCodersGroupID,
IpProfile_SBCAllowedVideoCodersGroupID, IpProfile_SBCAllowedCodersMode,
IpProfile_SBCMediaSecurityBehaviour, IpProfile_SBCRFC2833Behavior,
IpProfile_SBCAlternativeDTMFMethod, IpProfile_SBCAssertIdentity,
IpProfile_AMDSensitivityParameterSuit, IpProfile_AMDSensitivityLevel,
IpProfile_AMDMaxGreetingTime, IpProfile_AMDMaxPostSilenceGreetingTime,
IpProfile_SBCDiversionsMode, IpProfile_SBCHistoryInfoMode,
IpProfile_EnableQSIGTunneling, IpProfile_SBCFaxCodersGroupID,
```

```

IpProfile_SBCFaxBehavior, IpProfile_SBCFaxOfferMode,
IpProfile_SBCFaxAnswerMode, IpProfile_SbcPrackMode,
IpProfile_SBCSessionExpiresMode, IpProfile_SBCRemoteUpdateSupport,
IpProfile_SBCRemoteReinviteSupport,
IpProfile_SBCRemoteDelayedOfferSupport, IpProfile_SBCRemoteReferBehavior,
IpProfile_SBCRemote3xxBehavior, IpProfile_SBCRemoteMultiple18xSupport,
IpProfile_SBCRemoteEarlyMediaResponseType,
IpProfile_SBCRemoteEarlyMediaSupport, IpProfile_EnableSymmetricMKI,
IpProfile_MKISize, IpProfile_SBCEnforceMKISize,
IpProfile_SBCRemoteEarlyMediaRTP, IpProfile_SBCRemoteSupportsRFC3960,
IpProfile_SBCRemoteCanPlayRingback, IpProfile_EnableEarly183,
IpProfile_EarlyAnswerTimeout, IpProfile_SBC2833DTMFPayloadType,
IpProfile_SBCUserRegistrationTime, IpProfile_ResetSRTPStateUponRekey,
IpProfile_AmdMode, IpProfile_SBCReliableHeldToneSource,
IpProfile_GenerateSRTPKeys, IpProfile_SBCPlayHeldTone,
IpProfile_SBCRemoteHoldFormat, IpProfile_SBCRemoteReplacesBehavior,
IpProfile_SBCSDPPtimeAnswer, IpProfile_SBCPreferredPTIME,
IpProfile_SBCUseSilenceSupp, IpProfile_SBCRTPRedundancyBehavior,
IpProfile_SBCPlayRBTToTransferee, IpProfile_SBCRTCPMode,
IpProfile_SBCJitterCompensation,
IpProfile_SBCRemoteRenegotiateOnFaxDetection,
IpProfile_JitterBufMaxDelay, IpProfile_SBCRemoteMultipleAnswersMode,
IpProfile_SBCKeepVIAHeaders, IpProfile_SBCKeepUserAgentHeader,
IpProfile_SBCUserBehindUdpNATRegistrationTime,
IpProfile_SBCUserBehindTcpNATRegistrationTime,
IpProfile_SBCAdaptRFC2833BWToVoiceCoderBW;
IpProfile 1 = "S4B", 1, 0, 0, 10, 10, 46, 40, 0, 0, 0, 0, 2, 0, 0, 0, 0,
-1, 1, 0, 0, -1, 0, 4, -1, 1, 1, 0, 0, "", -1, 0, 0, "", -1, -1, 0, 1, 1,
0, 0, 0, 8, 300, 400, 0, 0, 0, -1, 0, 0, 1, 1, 3, 1, 1, 0, 3, 2, 1, 0, 1,
1, 1, 1, 1, 0, 0, 0, 0, 0, 0, 1, 0, 1, 1, 0, 0, 0, 0, 0, 0, 0, 0, 1, 0,
0, 300, 0, -1, -1, -1, -1, 0;
IpProfile 2 = "DTAG", 1, 0, 0, 10, 10, 46, 40, 0, 0, 0, 0, 2, 0, 0, 0, 0,
-1, 1, 0, 0, -1, 0, 4, -1, 1, 1, 0, 0, "", 0, 0, 0, "", 0, -1, 0, 2, 0,
0, 1, 0, 8, 300, 400, 0, 0, 0, -1, 0, 0, 1, 3, 0, 2, 2, 1, 3, 0, 1, 0, 1,
0, 0, 0, 0, 0, 1, 0, 0, 0, 0, 0, 0, 1, 0, 0, 1, 0, 0, 0, 0, 0, 0, 0,
0, 300, 0, -1, -1, -1, -1, 0;

[ \IpProfile ]

[ ProxySet ]

FORMAT ProxySet_Index = ProxySet_ProxyName,
ProxySet_EnableProxyKeepAlive, ProxySet_ProxyKeepAliveTime,
ProxySet_ProxyLoadBalancingMethod, ProxySet_IsProxyHotSwap, ProxySet_SRD,
ProxySet_ClassificationInput, ProxySet_TLSContext,
ProxySet_ProxyRedundancyMode, ProxySet_DNSResolveMethod,
ProxySet_KeepAliveFailureResp;
ProxySet 0 = "", 0, 60, 0, 0, 0, 0, 0, "-1", -1, -1, "";
ProxySet 1 = "S4B", 1, 60, 1, 1, 0, 0, 0, "-1", 1, -1, "";
ProxySet 2 = "DTAG", 1, 60, 0, 1, 1, 0, 0, "-1", 1, 1, "";

[ \ProxySet ]

[ IPGroup ]

FORMAT IPGroup_Index = IPGroup_Type, IPGroup_Description,
IPGroup_ProxySetId, IPGroup_SIPGroupName, IPGroup_ContactUser,
IPGroup_EnableSurvivability, IPGroup_ServingIPGroup,
IPGroup_SipReRoutingMode, IPGroup_AlwaysUseRouteTable,
IPGroup_RoutingMode, IPGroup_SRD, IPGroup_MediaRealm,
IPGroup_ClassifyByProxySet, IPGroup_ProfileId, IPGroup_MaxNumOfRegUsers,
IPGroup_InboundManSet, IPGroup_OutboundManSet, IPGroup_RegistrationMode,

```

```

IPGroup_AuthenticationMode, IPGroup_MethodList,
IPGroup_EnableSBCCClientForking, IPGroup_SourceUriInput,
IPGroup_DestUriInput, IPGroup_ContactName, IPGroup_Username,
IPGroup_Password, IPGroup_UUIFormat, IPGroup_QOEProfile,
IPGroup_BWProfile, IPGroup_MediaEnhancementProfile,
IPGroup_AlwaysUseSourceAddr, IPGroup_MsgManUserDef1,
IPGroup_MsgManUserDef2, IPGroup_SIPConnect,
IPGroup_SBCRouteUsingRequestURIPort;

IPGroup 1 = 0, "S4B", 1, "sip-trunk.telekom.de", "", 0, -1, -1, 0, -1, 0,
"MRlan", 1, 1, -1, -1, 2, 0, 0, "", 0, -1, -1, "", "Admin",
"$1$aCkNBwIC", 0, "", "", "", 0, "", "", 0, 0;

IPGroup 2 = 0, "DTAG", 2, "sip-trunk.telekom.de", "", 0, -1, -1, 0, -1,
1, "MRWan", 1, 2, -1, -1, 4, 0, 0, "", 0, -1, -1, "", "Admin",
"$1$aCkNBwIC", 0, "", "", "", 0, "", "", 0, 0;

[ \IPGroup ]

[ Account ]

FORMAT Account_Index = Account_ServedTrunkGroup, Account_ServedIPGroup,
Account_ServingIPGroup, Account_Username, Account_Password,
Account_HostName, Account_Register, Account_ContactUser,
Account_ApplicationType;

Account 0 = -1, 1, 2, "audiocodes-vosip-reg1",
"$1$mdT4/f/4pqvL6eXFwZDE1A==", "sip-trunk.telekom.de", 1,
"+496987409358", 2;

[ \Account ]

[ IP2IPRouting ]

FORMAT IP2IPRouting_Index = IP2IPRouting_RouteName,
IP2IPRouting_SrcIPGroupID, IP2IPRouting_SrcUsernamePrefix,
IP2IPRouting_SrcHost, IP2IPRouting_DestUsernamePrefix,
IP2IPRouting_DestHost, IP2IPRouting_RequestType,
IP2IPRouting_MessageCondition, IP2IPRouting_ReRouteIPGroupID,
IP2IPRouting_Trigger, IP2IPRouting_CallSetupRulesSetId,
IP2IPRouting_DestType, IP2IPRouting_DestIPGroupID,
IP2IPRouting_DestSRDID, IP2IPRouting_DestAddress, IP2IPRouting_DestPort,
IP2IPRouting_DestTransportType, IP2IPRouting_AltRouteOptions,
IP2IPRouting_GroupPolicy, IP2IPRouting_CostGroup;

IP2IPRouting 0 = "Terminate OPTIONS", -1, "", "", "", "", 6, "", -1,
0, -1, 1, -1, "", "internal", 0, -1, 0, 0, "";
IP2IPRouting 1 = "S4B to ITSP", 1, "", "", "", "", 0, "", -1, 0, -1,
0, 2, "1", "", 0, -1, 0, 0, "";
IP2IPRouting 2 = "ITSP to S4B", 2, "", "", "", "", 0, "", -1, 0, -1,
0, 1, "0", "", 0, -1, 0, 0, "";

[ \IP2IPRouting ]

[ TLSContexts ]

FORMAT TLSContexts_Index = TLSContexts_Name, TLSContexts_TLSVersion,
TLSContexts_ServerCipherString, TLSContexts_ClientCipherString,
TLSContexts_OcspEnable, TLSContexts_OcspServerPrimary,
TLSContexts_OcspServerSecondary, TLSContexts_OcspServerPort,
TLSContexts_OcspDefaultResponse;

TLSContexts 0 = "default", 7, "RC4:EXP", "ALL:!ADH", 0, , , 2560, 0;

```

```
[ \TLSContexts ]

[ SIPInterface ]

FORMAT SIPInterface_Index = SIPInterface_InterfaceName,
SIPInterface_NetworkInterface, SIPInterface_ApplicationType,
SIPInterface_UDPPort, SIPInterface_TCPPort, SIPInterface_TLSPort,
SIPInterface_SRD, SIPInterface_MessagePolicy, SIPInterface_TLSContext,
SIPInterface_TLSMutualAuthentication, SIPInterface_TCPKeepaliveEnable,
SIPInterface_ClassificationFailureResponseType,
SIPInterface_PreClassificationManSet;
SIPInterface 0 = "S4B", "Voice", 2, 0, 0, 5067, 0, "", "", -1, 0, 500, -1;
SIPInterface 1 = "DTAG", "WAN", 2, 0, 5060, 0, 1, "", "", -1, 0, 500, -1;

[ \SIPInterface ]

[ CodersGroup0 ]

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

[ \CodersGroup0 ]

[ AllowedCodersGroup0 ]

FORMAT AllowedCodersGroup0_Index = AllowedCodersGroup0_Name;
AllowedCodersGroup0 0 = "g711Ulaw64k";
AllowedCodersGroup0 1 = "g711Alaw64k";

[ \AllowedCodersGroup0 ]

[ MessageManipulations ]

FORMAT MessageManipulations_Index =
MessageManipulations_ManipulationName, MessageManipulations_ManSetID,
MessageManipulations_MessageType, MessageManipulations_Condition,
MessageManipulations_ActionSubject, MessageManipulations_ActionType,
MessageManipulations_ActionValue, MessageManipulations_RowRole;
MessageManipulations 0 = "Change RequestURI", 2, "invite", "",
"header.request-uri.url.user", 2, "header.to.url.user", 0;
MessageManipulations 1 = "Call Forward", 4, "invite", "header.history-
info.0 regex (<sip:)(.*)"(.*)"(.*)", "header.from.url.user", 2, "$2", 0;
MessageManipulations 2 = "Call Forward", 4, "", "", "header.history-
info", 1, "", 1;
MessageManipulations 3 = "Call Forward", 4, "", "", "header.p-asserted-
identity.url.user", 2, "header.from.url.user", 1;
MessageManipulations 4 = "Call Transfer", 4, "invite", "header.referred-
by exists", "header.from.url.user", 2, "header.referred-by.url.user", 0;
MessageManipulations 5 = "Call Transfer", 4, "", "", "header.p-asserted-
identity.url.user", 2, "header.from.url.user", 1;
MessageManipulations 6 = "Call Transfer", 4, "", "", "header.referred-
by.url.host", 2, "param.ipg.dst.host", 1;
```

```
MessageManipulations 7 = "Responses", 4, "any.response", "header.request-  
uri.methodtype=='503' OR header.request-uri.methodtype=='603'",  
"header.request-uri.methodtype", 2, "'486'", 0;
```

```
[ \MessageManipulations ]
```

```
[ RoutingRuleGroups ]
```

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

```
[ \RoutingRuleGroups ]
```

```
[ 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 ]
```

B AudioCodes MSBR Data Configuration

Below shown example of AudioCodes MSBR data configuration file:

```
# Running Configuration Mediant 500L - MSBR

## VoIP Configuration

configure voip
  tls 0
    name default
    tls-version tls-v1.0_1.1_1.2
    ciphers-server "RC4:EXP"
    ciphers-client "ALL:!ADH"
    ocsp-server disable
    ocsp-port 2560
    ocsp-default-response reject
  exit
  appli-enabling
    enable-sbc on
    activate
  exit
  coders-and-profiles ip-profile 1
    profile-name "S4B"
    disconnect-on-broken-connection ignore
    sbc-media-security-behaviour srtp
    sbc-rfc2833-behavior extend
    sbc-prack-mode optional
    sbc-session-expires-mode supported
    sbc-rmt-update-supp supported-only-after-connect
    sbc-rmt-re-invite-supp supported-only-with-sdp
    sbc-rmt-delayed-offer not-supported
    sbc-rmt-refer-behavior handle-locally
    sbc-rmt-3xx-behavior handle-locally
    enable-symmetric-mki enable
    sbc-enforce-mki-size enforce
    sbc-rmt-early-media-rtp by-media
    sbc-rmt-can-play-ringback no
    early-answer-timeout 0
    reset-srtp-upon-re-key enable
    generate-srtp-keys always
    sbc-rtcp-mode generate-always
    activate
  exit
  coders-and-profiles ip-profile 2
    profile-name "DTAG"
    disconnect-on-broken-connection ignore
    sbc-ext-coders-group-id coders-group-0
    sbc-allowed-coders-group-id coders-group-0
    sbc-media-security-behaviour rtp
    sbc-assert-identity add
    sbc-rmt-refer-behavior handle-locally
    mki-size 0
    early-answer-timeout 0
```

```

reset-srtp-upon-re-key disable
generate-srtp-keys only-if-required
remote-hold-Format sendonly
activate
exit
coders-and-profiles coders-group-0 0
  name "g711Ulaw64k"
  p-time 20
  rate 0
  activate
exit
coders-and-profiles coders-group-0 1
  name "g711Alaw64k"
  p-time 20
  rate 0
  activate
exit
interface network-dev 0
  name "vlan 1"
  activate
exit
interface network-if 0
  ip-address 10.15.17.33
  gateway 10.15.17.34
  name "Voice"
  primary-dns 10.15.27.1
  underlying-dev "vlan 1"
  activate
exit
voip-network realm 0
  name "MRLan"
  ipv4if "Voice"
  port-range-start 6000
  session-leg 100
  port-range-end 6990
  is-default true
  activate
exit
voip-network realm 1
  name "MRWan"
  ipv4if "WAN"
  port-range-start 7000
  session-leg 100
  port-range-end 7990
  activate
exit
voip-network srd 0
  name "SRDLan"
  media-realm-name "MRLan"
  activate
exit
voip-network srd 1
  name "SRDWan"
  media-realm-name "MRWan"
  activate
exit

```

```
voip-network sip-interface 0
  interface-name "S4B"
  network-interface "Voice"
  application-type sbc
  udp-port 0
  tcp-port 0
  tls-port 5067
  activate
exit
voip-network sip-interface 1
  interface-name "DTAG"
  network-interface "WAN"
  application-type sbc
  udp-port 0
  tls-port 0
  srd 1
  activate
exit
voip-network proxy-set 0
  proxy-name ""
  activate
exit
voip-network proxy-set 1
  proxy-name "S4B"
  proxy-enable-keep-alive using-options
  proxy-load-balancing-method round-robin
  is-proxy-hot-swap yes
  proxy-redundancy-mode homing
  activate
exit
voip-network proxy-set 2
  proxy-name "DTAG"
  proxy-enable-keep-alive using-options
  is-proxy-hot-swap yes
  srd-id 1
  proxy-redundancy-mode homing
  dns-resolve-method srv
  activate
exit
voip-network ip-group 1
  description "S4B"
  proxy-set-id 1
  sip-group-name "sip-trunk.telekom.de"
  media-realm-name "MRLan"
  ip-profile-id 1
  outbound-mesg-manipulation-set 2
  username "Admin"
  password aCkNBwIC obscured
  activate
exit
voip-network ip-group 2
  description "DTAG"
  proxy-set-id 2
  sip-group-name "sip-trunk.telekom.de"
  srd 1
  media-realm-name "MRWan"
```

```

ip-profile-id 2
outbound-mesg-manipulation-set 4
username "Admin"
password aCkNBwIC obscured
activate
exit
gw digitalgw digital-gw-parameters
answer-detector-cmd 10486144
energy-detector-cmd 587202560
activate
exit
ldap
ldap-search-server-method sequentially
activate
exit
media udp-port-configuration
udp-port-spacing 10
activate
exit
media security
media-security-enable on
srtp-tx-packet-mKi-size 1
activate
exit
sbc routing ip2ip-routing 0
route-name "Terminate OPTIONS"
request-type options
dst-type dst-address
dst-address "internal"
activate
exit
sbc routing ip2ip-routing 1
route-name "S4B to ITSP"
src-ip-group-id 1
dst-ip-group-id 2
dst-srd-id "1"
activate
exit
sbc routing ip2ip-routing 2
route-name "ITSP to S4B"
src-ip-group-id 2
dst-ip-group-id 1
dst-srd-id "0"
activate
exit
sbc manipulations message-manipulations 0
manipulation-name "Change Request URI"
manipulation-set-id 2
message-type "invite"
action-subject "header.request-uri.url.user"
action-type modify
action-value "header.to.url.user"
activate
exit
sbc manipulations message-manipulations 1
manipulation-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 "$2"
activate
exit
sbc manipulations message-manipulations 2
manipulation-name "Call Forward"
manipulation-set-id 4
action-subject "header.history-info"
action-type remove
row-role use-previous-condition
activate
exit
sbc manipulations message-manipulations 3
manipulation-name "Call Forward"
manipulation-set-id 4
action-subject "header.p-asserted-identity.url.user"
action-type modify
action-value "header.from.url.user"
row-role use-previous-condition
activate
exit
sbc manipulations message-manipulations 4
manipulation-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"
activate
exit
sbc manipulations message-manipulations 5
manipulation-name "Call Transfer"
manipulation-set-id 4
action-subject "header.p-asserted-identity.url.user"
action-type modify
action-value "header.from.url.user"
row-role use-previous-condition
activate
exit
sbc manipulations message-manipulations 6
manipulation-name "Call Transfer"
manipulation-set-id 4
action-subject "header.referred-by.url.host"
action-type modify
action-value "param.ipg.dst.host"
row-role use-previous-condition
activate
exit
sbc manipulations message-manipulations 7
manipulation-name "Responses"
manipulation-set-id 4
```

```

message-type "any.response"
condition "header.request-uri.methodtype=='503' OR header.request-
uri.methodtype=='603'"
action-subject "header.request-uri.methodtype"
action-type modify
action-value "'486'"
activate
exit
sbc general-setting
sbc-forking-handling-mode sequential
sbc-preferences with-extensions
activate
exit
sbc allowed-coders-group group-0 0
name "g711Ulaw64k"
activate
exit
sbc allowed-coders-group group-0 1
name "g711Alaw64k"
activate
exit
activate
exit
sip-definition proxy-and-registration
dns-query srv
set gw-name "sip-trunk.telekom.de"
proxy-dns-query srv
registration-time 360
use-gw-name-for-opt enable
activate
exit
sip-definition general-settings
t38-use-rtp-port on
activate
exit
sip-definition advanced-settings
set ldap-primary-key "telephoneNumber"
activate
exit
sip-definition account 0
served-ip-group 1
serving-ip-group 2
user-name "audiocodes-vosip-reg1"
password mdT4/f/4pqvL6eXfwZDE1A== obscured
host-name "sip-trunk.telekom.de"
register reg
contact-user "+496987409358"
application-type sbc
activate
exit
tdm
pcm-law-select mulaw
activate
exit
voip-network proxy-ip 0
proxy-address "FE.S4B.interop:5067"

```

```
transport-type tls
proxy-set-id 1
activate
exit
voip-network proxy-ip 1
proxy-address "reg.sip-trunk.telekom.de"
transport-type tcp
proxy-set-id 2
activate
exit
exit

## System Configuration

configure system
cli-terminal
idle-timeout 10000
activate
exit
cwmpp
tls-context 0
activate
exit
logging
source voip
syslog on
debug-level detailed
syslog-ip 10.15.17.100
activate
exit
ntp
set primary-server "10.15.27.1"
activate
exit
radius
set shared-secret "$1$woS2sLC0opqIjoKZng== "
activate
exit
snmp
no activate-keep-alive-trap
activate
exit
web
set https-cipher-string "RC4:EXP"
activate
exit
hostname "Mediant 500L - MSBR"
configuration-version 0
exit

## Data Configuration

configure data
radio shutdown
interface GigabitEthernet 0/0
no ip address
```

```

mtu auto
desc "WAN Copper"
no ipv6 enable
speed auto
duplex auto
no service dhcp
ip dns server static
no shutdown
exit
interface Fiber 0/1
no ip address
mtu auto
desc "WAN Fiber"
no ipv6 enable
no service dhcp
ip dns server static
no shutdown
exit
interface dsl 0/2
#DSL configuration is automatic
#Termination cpe
mode vdsl
auto-switch-attempts vdsl 3 vdsl-v43 3 adsl 3
no shutdown
exit
interface EFM 0/2
no ip address
mtu auto
desc "VDSL"
no ipv6 enable
no service dhcp
ip dns server static
no shutdown
exit
interface FastEthernet 1/1
speed auto
duplex auto
switchport mode trunk
switchport trunk native vlan 1
no shutdown
exit
interface FastEthernet 1/2
speed auto
duplex auto
switchport mode trunk
switchport trunk native vlan 1
no shutdown
exit
interface FastEthernet 1/3
speed auto
duplex auto
switchport mode trunk
switchport trunk native vlan 1
no shutdown
exit
interface FastEthernet 1/4

```

```
speed auto
duplex auto
switchport mode trunk
switchport trunk native vlan 1
no shutdown
exit
interface VLAN 1
no ip address
bridge-group 1
mtu auto
desc "LAN switch VLAN 1"
no ipv6 enable
no service dhcp
ip dns server static
no link-state monitor
no shutdown
exit
interface BVI 1
ip address 10.15.17.34 255.255.0.0
mtu auto
desc "LAN Bridge"
no ipv6 enable
no service dhcp
ip dns server static
no napt
no firewall enable
no shutdown
exit
interface dot11radio 1
#This interface is DISABLED due to physical layer configuration
no ip address
bridge-group 1
mtu auto
desc "LAN Wireless 802.11n Access Point"
no ipv6 enable
no service dhcp
ssid MSBR
broadcast
security mode NONE
no security mac mode
mode ngb
channel width 40/20
channel auto
power 100
beacon dtim-period 1
beacon period 100
fragment threshold 2346
cts mode none
cts type cts
burst num 3
burst time 2
rts threshold 2346
wmm
shutdown
exit
interface pppoe 0
```

```

firewall enable
napt
mtu auto
ppp user acl121@014 obscured-pass VDVjYGRrYWxv
ppp authentication chap
ppp authentication ms-chap
ppp authentication ms-chap-v2
ppp authentication pap
ppp lcp-echo 6 5
no ppp compression
ip address auto
no ipv6 address
ip dns server auto
underlying EFM 0/2
no shutdown
exit
ip nat translation udp-timeout 120
ip nat translation tcp-timeout 3600
ip nat translation icmp-timeout 6
# Note: The following WAN ports are in use by system services,
#       conflicting rules should not be created:
#       Ports 82 - 82 --> TR069
#       Ports 7000 - 7990 --> RealmPortPool::MRWan
#       Ports 5060 - 5060 --> SIPLISTENING#1
# Note: The following NAT rules are in effect for system services,
#       conflicting rules should not be created:
#       RealmPortPool::MRWan: LAN ports 7000-7990 to WAN IP
62.219.46.190 ports 7000-7990, interface PPPoE 0
#       SIPLISTENING#1: LAN ports 5060-5060 to WAN IP 62.219.46.190
ports 5060-5060, interface PPPoE 0
ip route 0.0.0.0 0.0.0.0 PPPoE 0 1
ip domain name home
ip domain localhost msbr
pm sample-interval minute 5
pm sample-interval seconds 15
exit

```

This page is intentionally left blank.

International Headquarters

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

AudioCodes Inc.

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

Contact us: <https://www.audiocodes.com/corporate/offices-worldwide>

Website: <https://www.audiocodes.com/>

©2018 AudioCodes Ltd. All rights reserved. AudioCodes, AC, HD VoIP, HD VoIP Sounds Better, IPmedia, Mediant, MediaPack, What's Inside Matters, OSN, SmartTAP, User Management Pack, VMAS, VoIPerfect, VoIPerfectHD, Your Gateway To VoIP, 3GX, VocaNom, AudioCodes One Voice and CloudBond are trademarks or registered trademarks of AudioCodes Limited. All other products or trademarks are property of their respective owners. Product specifications are subject to change without notice.

Document #: LTRT-12590

