Configuration Note

AudioCodes Professional Services – Interoperability Lab

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

Version 7.2

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Abbreviations and Terminology

Each abbreviation, unless widely used, is spelled out in full when first used.

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1 Introduction

This Configuration Note describes how to set up AudioCodes Enterprise Session Border Controller (hereafter, referred to as *E-SBC*) for interworking between Virgin Media'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 https://www.audiocodes.com/partners/sbc-interoperability-list.

1.1 Intended Audience

The document is intended for engineers, or AudioCodes and Virgin Media Partners who are responsible for installing and configuring Virgin Media's SIP Trunk and Microsoft's Skype for Business Server 2015 for enabling VoIP calls using AudioCodes E-SBC.

1.2 About AudioCodes E-SBC Product Series

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

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

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

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

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2 Component Information

2.1 AudioCodes E-SBC Version

Table 2-1: AudioCodes E-SBC Version

SBC Vendor	AudioCodes
Models	▪ Mediant 500 E-SBC ▪ Mediant 500L Gateway & E-SBC ▪ Mediant 800B Gateway & E-SBC ▪ Mediant 1000B Gateway & E-SBC ▪ Mediant 2600 E-SBC ▪ Mediant 4000 SBC ▪ Mediant 4000B SBC ▪ Mediant 9000 SBC ▪ Mediant Software SBC (SE and VE)
Software Version	7.20A.200.055
Protocol	▪ SIP/UDP (to Virgin Media SIP Trunk) ▪ SIP/TCP or TLS (to the S4B FE Server)
Additional Notes	None

2.2 Virgin Media SIP Trunking Version

Table 2-2: Virgin Media Version

Vendor/Service Provider	Virgin Media
SSW Model/Service	GENBAND
Software Version	C20
Protocol	SIP
Additional Notes	None

2.3 Microsoft Skype for Business Server 2015 Version

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

Vendor	Microsoft
Model	Skype for Business
Software Version	Release 2015 6.0.9319.259
Protocol	SIP
Additional Notes	None

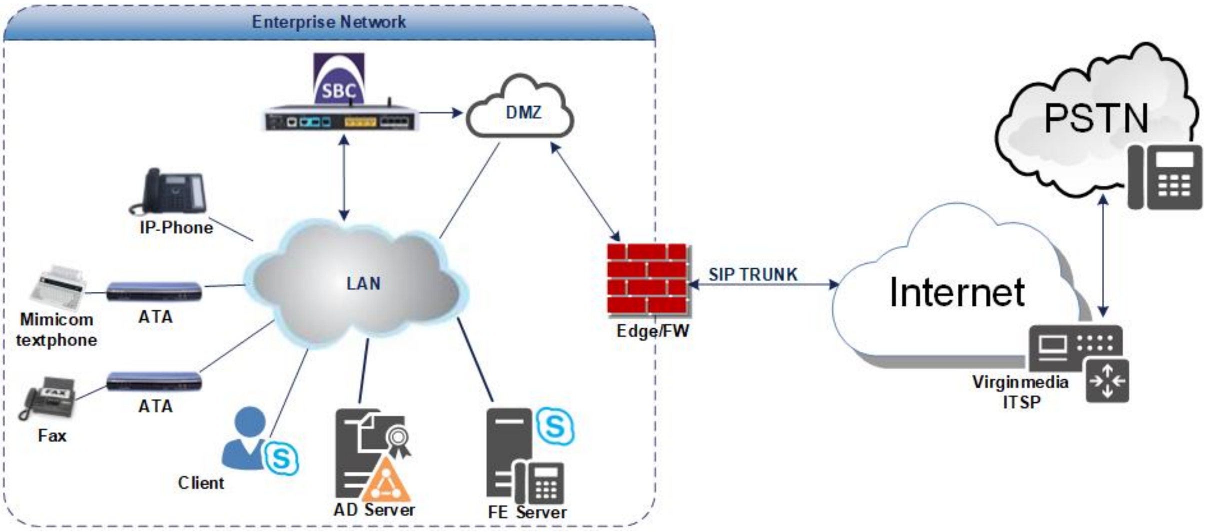
2.4 Interoperability Test Topology

The interoperability testing between AudioCodes E-SBC and Virgin Media 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 Virgin Media'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 Virgin Media'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 Virgin Media 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 ▪ Virgin Media SIP Trunk is located on the WAN
Signaling Transcoding	▪ Microsoft Skype for Business Server 2015 operates with SIP-over-TLS transport type ▪ Virgin Media SIP Trunk operates with SIP-over-UDP transport type
Codecs Transcoding	▪ Microsoft Skype for Business Server 2015 supports G.711A-law and G.711U-law coders ▪ Virgin Media SIP Trunk supports both G711 A-Law and G711 U-law. The testing was only conducted using G711 A-law as this is Virgin Media SIP Trunk preferred codec for interoperability testing
Media Transcoding	▪ Microsoft Skype for Business Server 2015 operates with SRTP media type ▪ Virgin Media SIP Trunk operates with RTP media type

2.4.2 Known Limitations

There were no limitations observed in the interoperability tests done for the AudioCodes E-SBC interworking between Microsoft Skype for Business Server 2015 and Virgin Media 's SIP Trunk.

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3 Configuring Skype for Business Server 2015

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

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

Note: For the intention of sending a proper address string for call establishment, it was agreed that a full E.164 number format be preceded by a leading '+' symbol. Therefore, you should ensure you configure Microsoft Skype for Business Server 2015 dial plan to work in full E.164 number format.

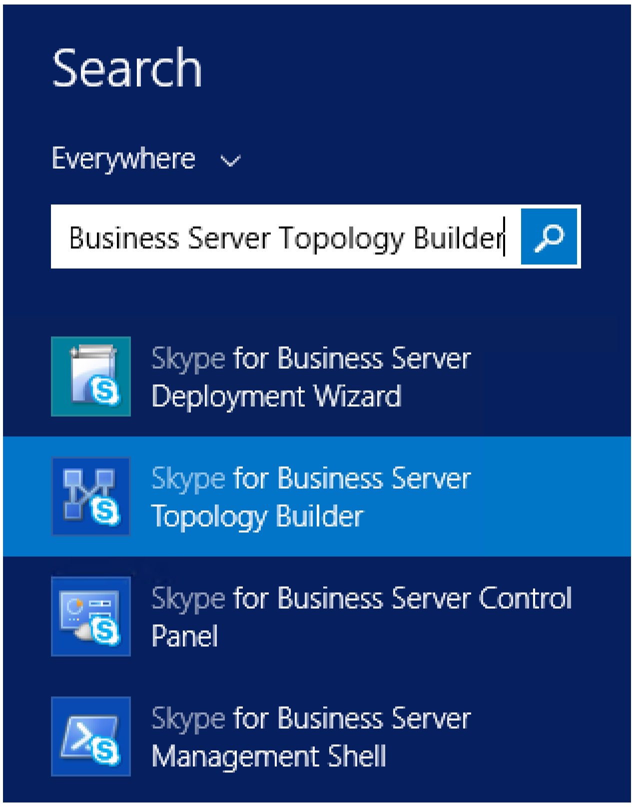
3.1 Configuring the E-SBC as an IP / PSTN Gateway

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:**

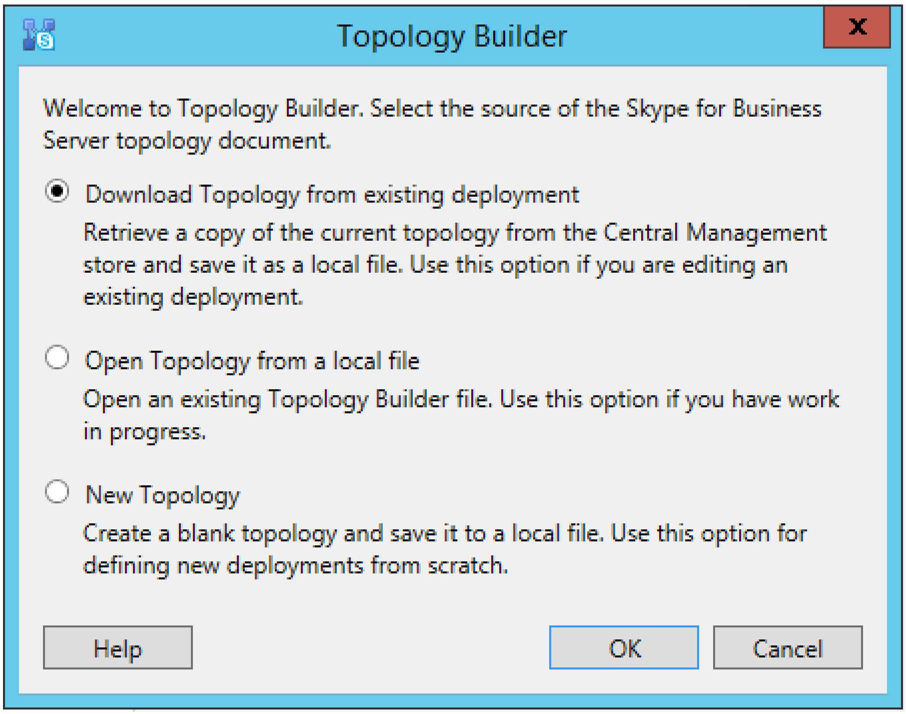
1. On the server where the Topology Builder is installed, start the Skype for Business Server 2015 Topology Builder (Windows **Start** menu > search for **Skype for Business Server Topology Builder**), as shown below:

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



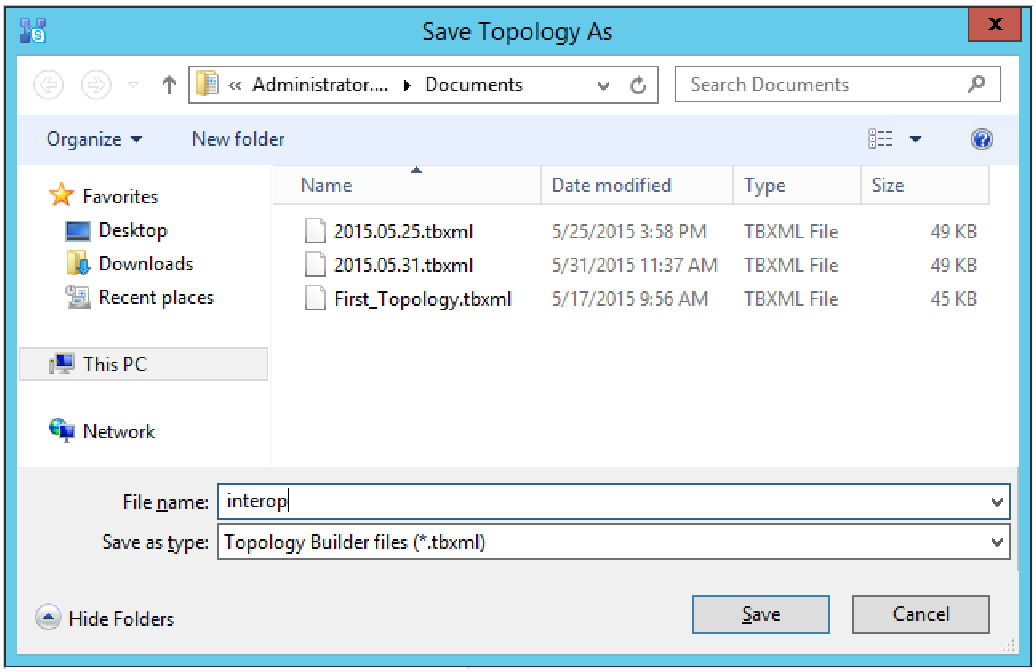
The following is displayed:

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:

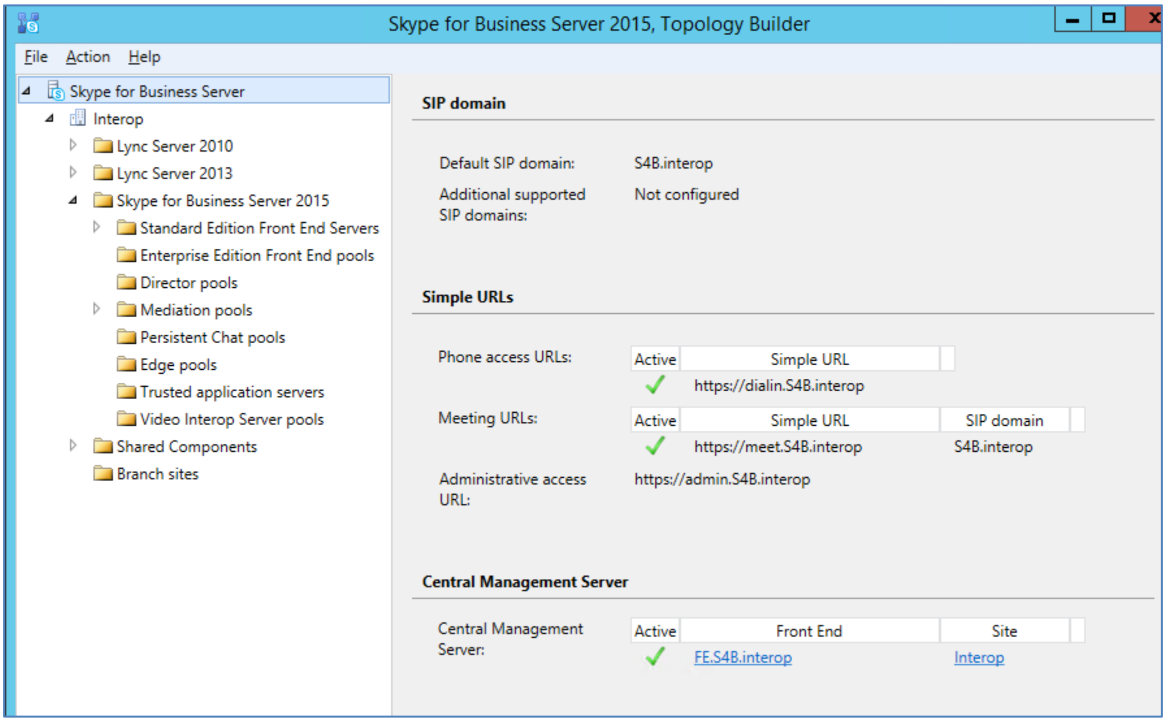
Figure 3-3: Save Topology Dialog Box



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.

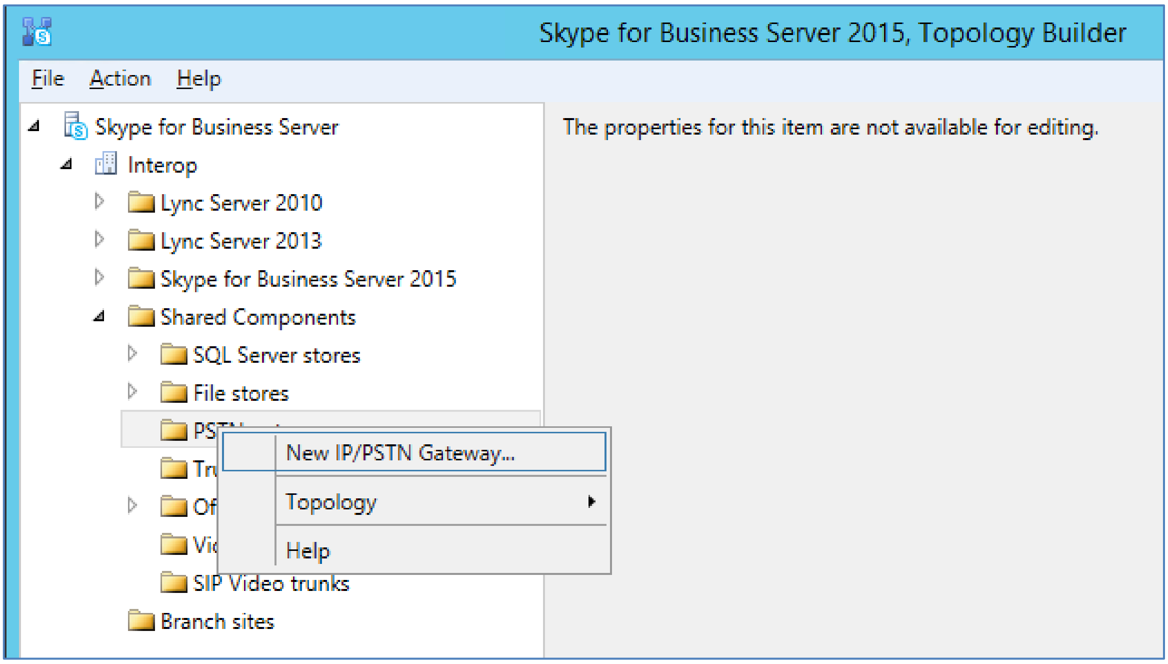
The Topology Builder screen with the downloaded Topology is displayed:

Figure 3-4: Downloaded Topology



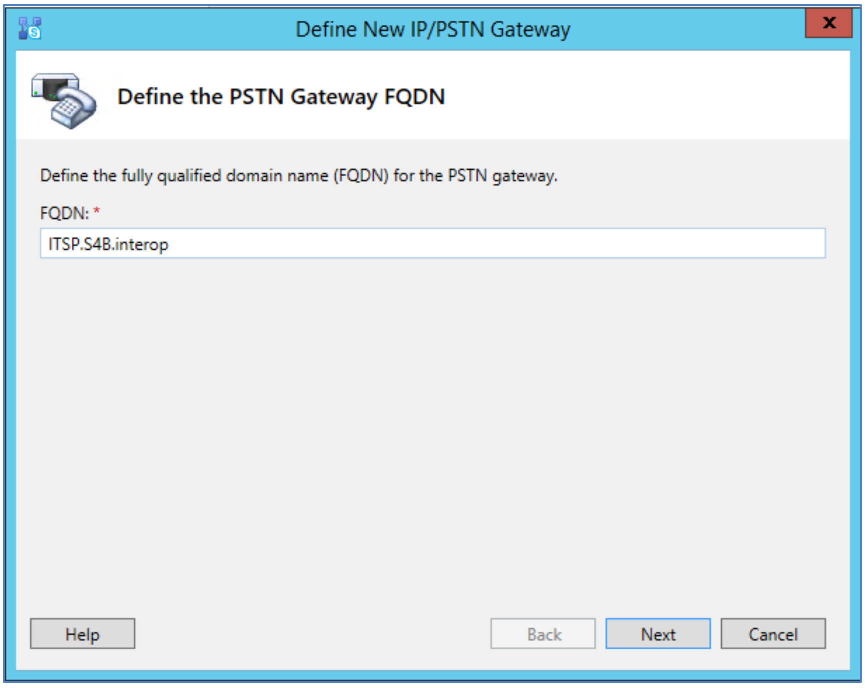
4. Under the **Shared Components** node, right-click the **PSTN gateways** node, and then from the shortcut menu, choose **New IP/PSTN Gateway**, as shown below:

Figure 3-5: Choosing New IP/PSTN Gateway



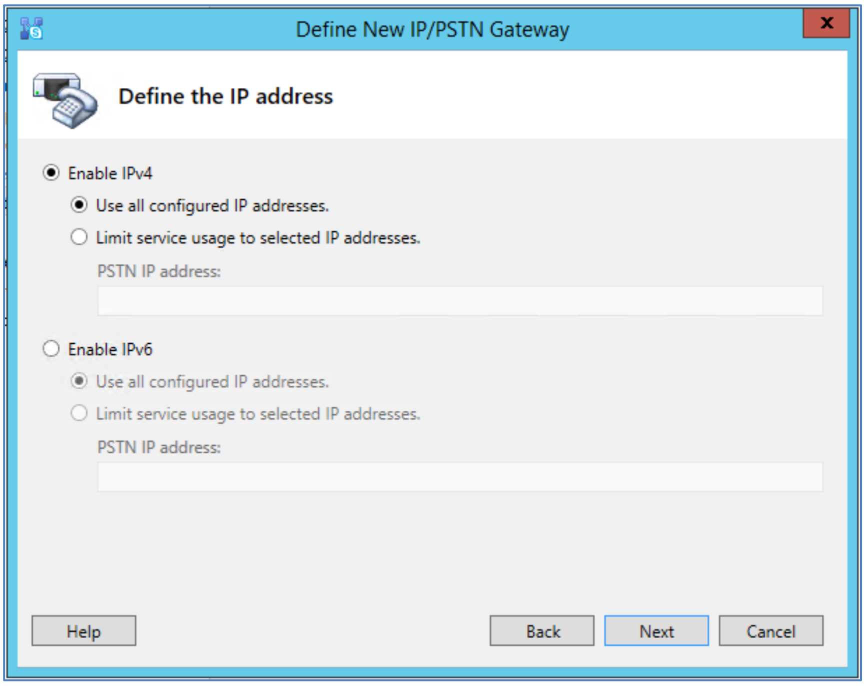
The following is displayed:

Figure 3-6: Define the PSTN Gateway FQDN



5. Enter the Fully Qualified Domain Name (FQDN) of the E-SBC (e.g., **ITSP.S4B.interop**). This FQDN should be equivalent to the configured Subject Name (CN) in the TLS Certificate Context (see Section 4.8.3 on page 55).
6. Click **Next**; the following is displayed:

Figure 3-7: Define the IP Address



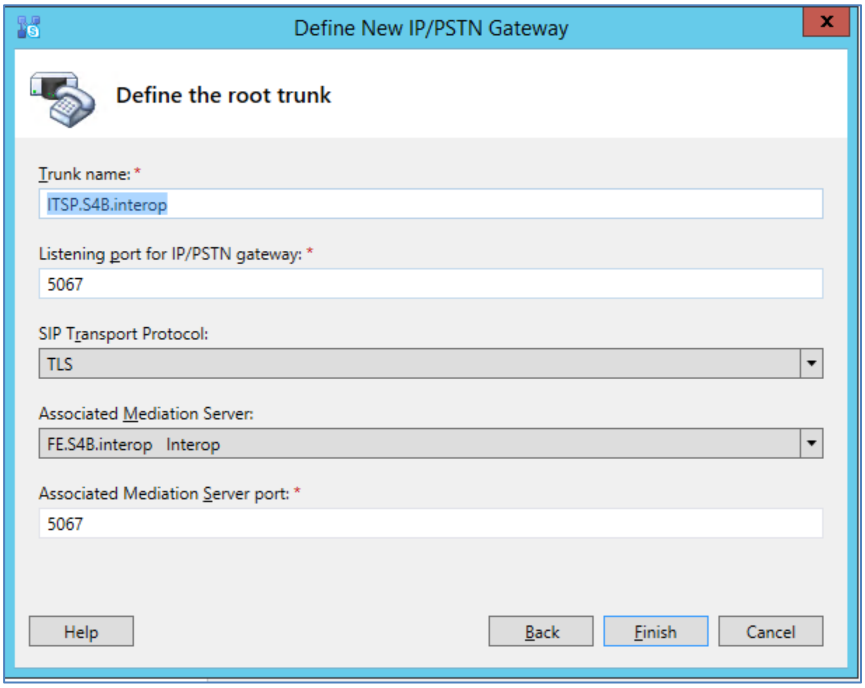
7. Define the listening mode (IPv4 or IPv6) of the IP address of your new PSTN gateway, and then click **Next**.

8. Define a *root trunk* for the PSTN gateway. A trunk is a logical connection between the Mediation Server and a gateway uniquely identified by the following combination: Mediation Server FQDN, Mediation Server listening port (TLS or TCP), gateway IP and FQDN, and gateway listening port.

Notes:

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

Figure 3-8: Define the Root Trunk



a. In the 'Listening Port for IP/PSTN Gateway' field, enter the listening port that the E-SBC will use for SIP messages from the Mediation Server that will be associated with the root trunk of the PSTN gateway (e.g., **5067**). This parameter is later configured in the SIP Interface table (see Section 4.2 on page 35).

b. In the 'SIP Transport Protocol' field, select the transport type (e.g., **TLS**) that the trunk uses. This parameter is later configured in the SIP Interface table (see Section 4.2 on page 35).

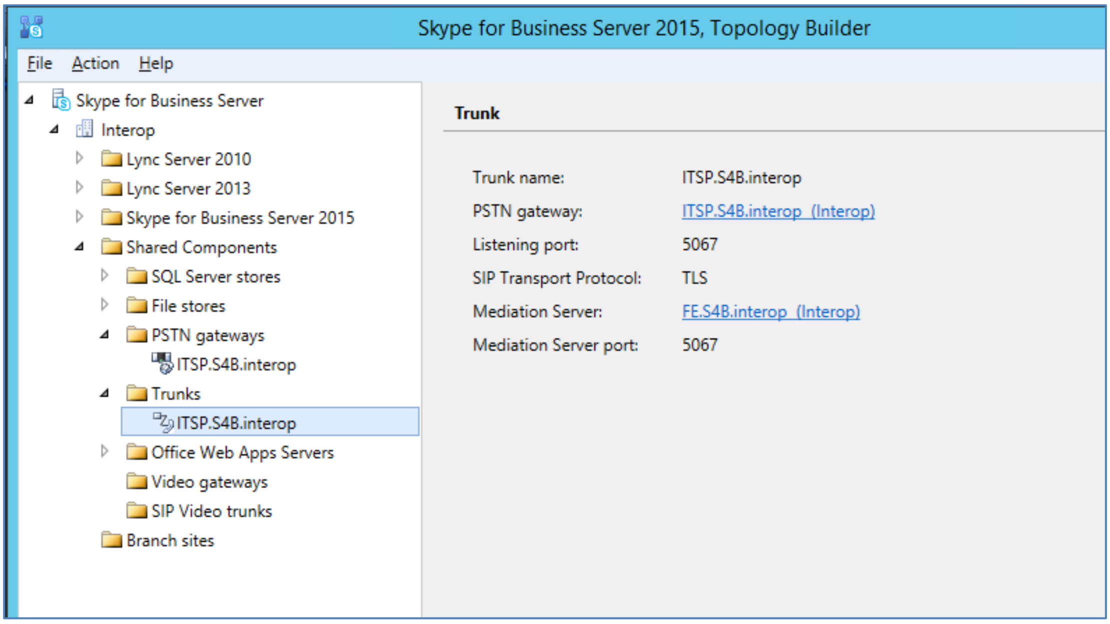
c. In the 'Associated Mediation Server' field, select the Mediation Server pool to associate with the root trunk of this PSTN gateway.

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**.

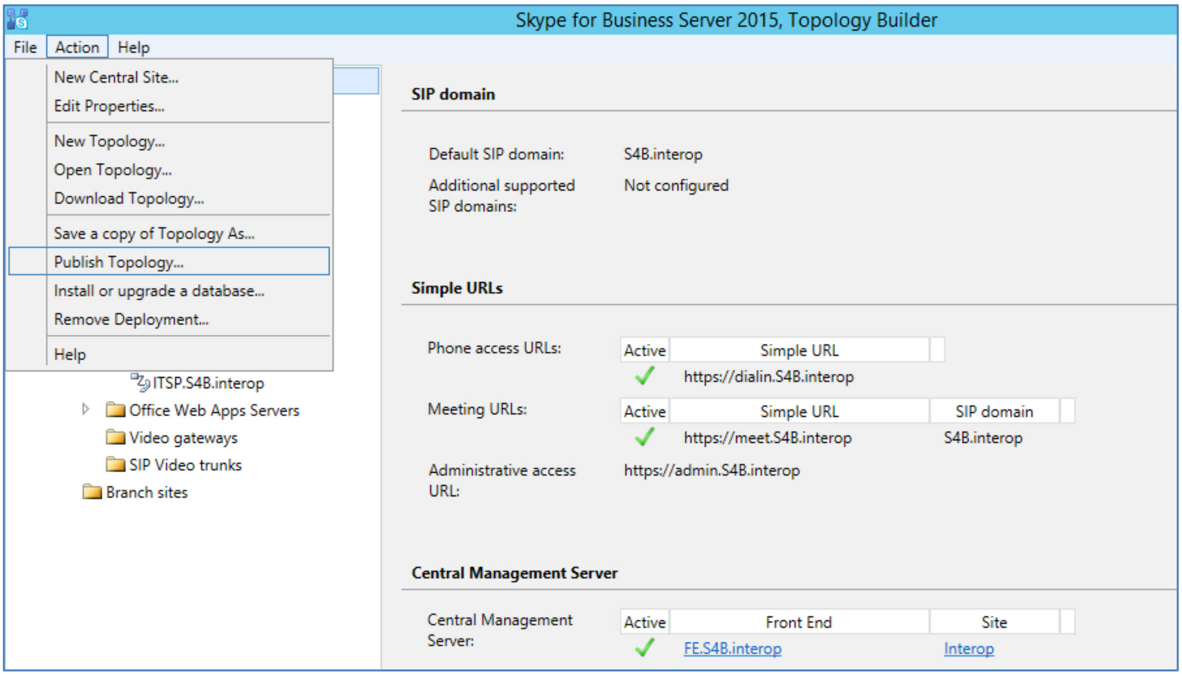
The E-SBC is added as a PSTN gateway, and a trunk is created as shown below:

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



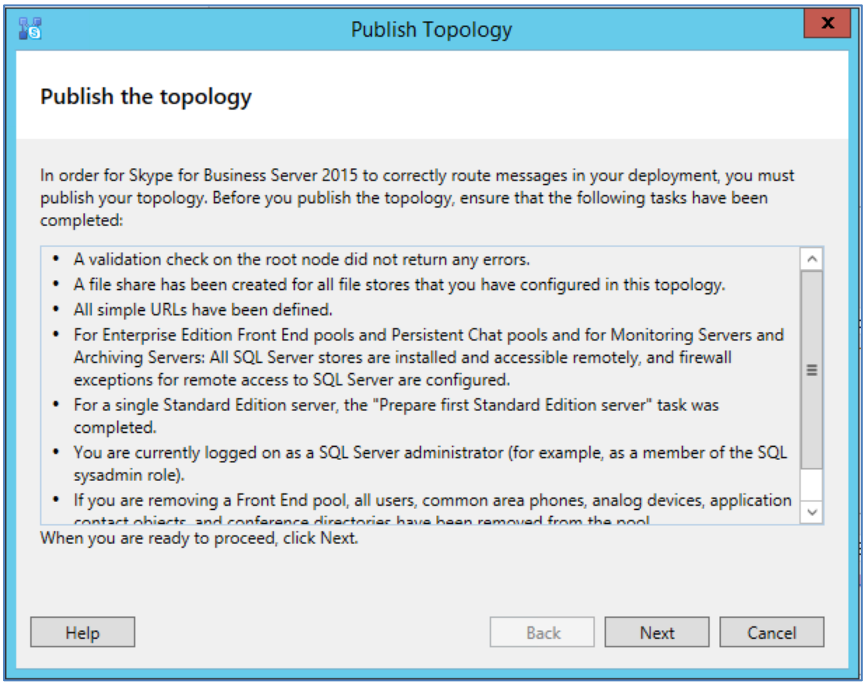
9. Publish the Topology: In the main tree, select the root node **Skype for Business Server**, and then from the **Action** menu, choose **Publish Topology**, as shown below:

Figure 3-10: Choosing Publish Topology



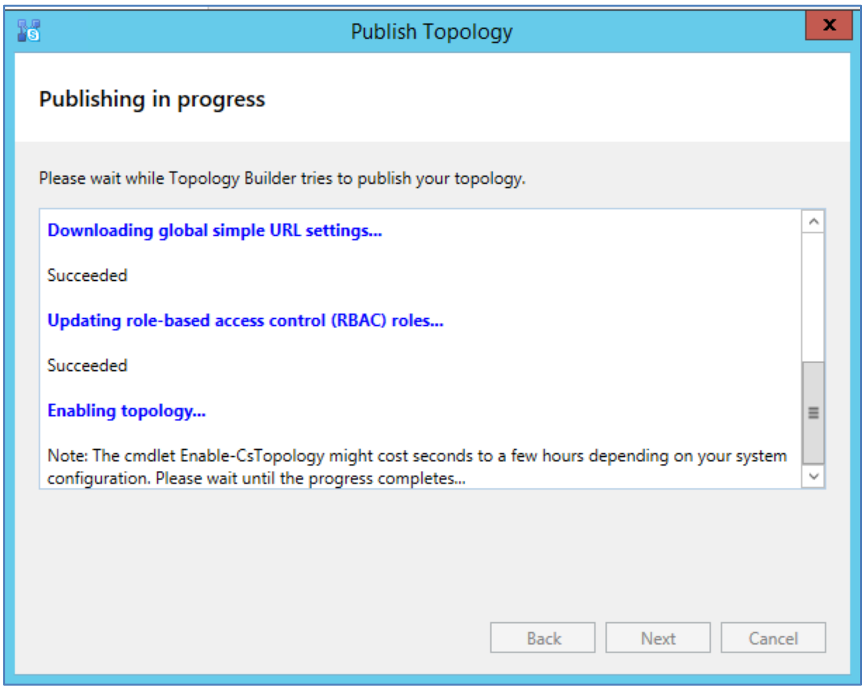
The following is displayed:

Figure 3-11: Publish the Topology



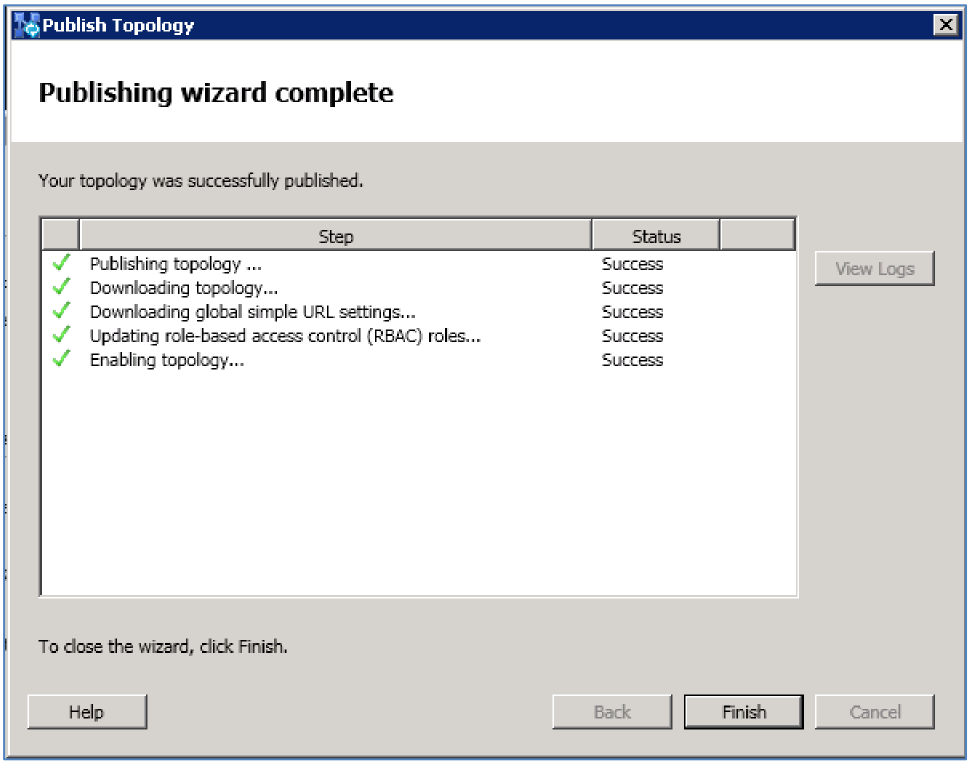
10. Click **Next**; the Topology Builder starts to publish your topology, as shown below:

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**.

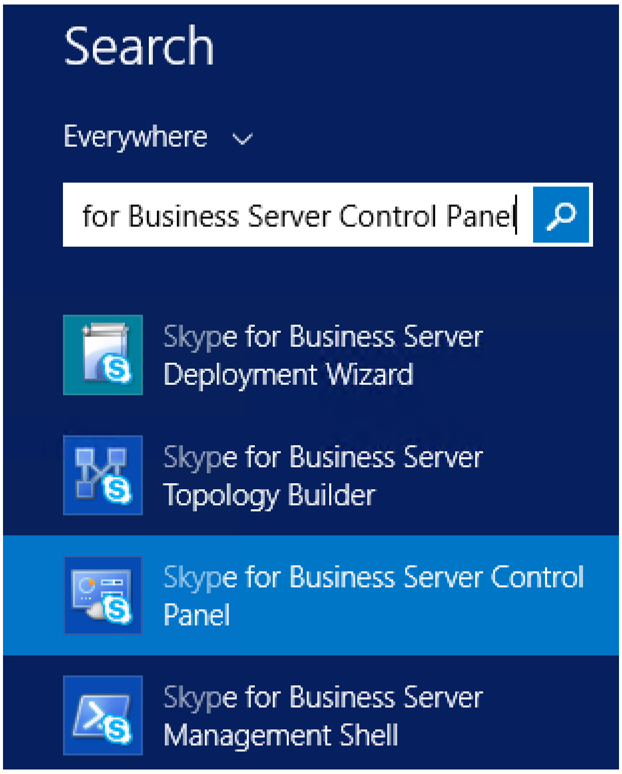
3.2 Configuring the "Route" on Skype for Business Server 2015

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:**

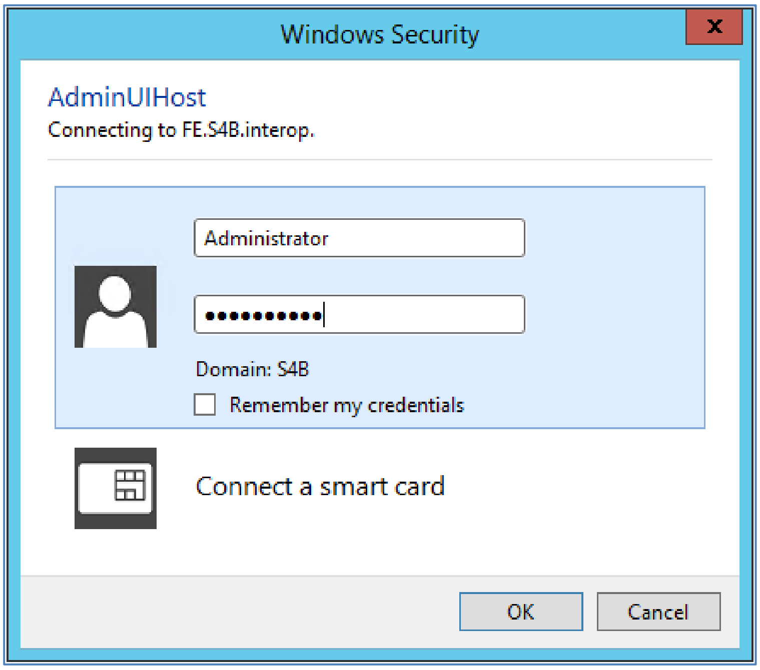
1. Start the Microsoft Skype for Business Server 2015 Control Panel (**Start** > search for **Microsoft Skype for Business Server Control Panel**), as shown below:

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



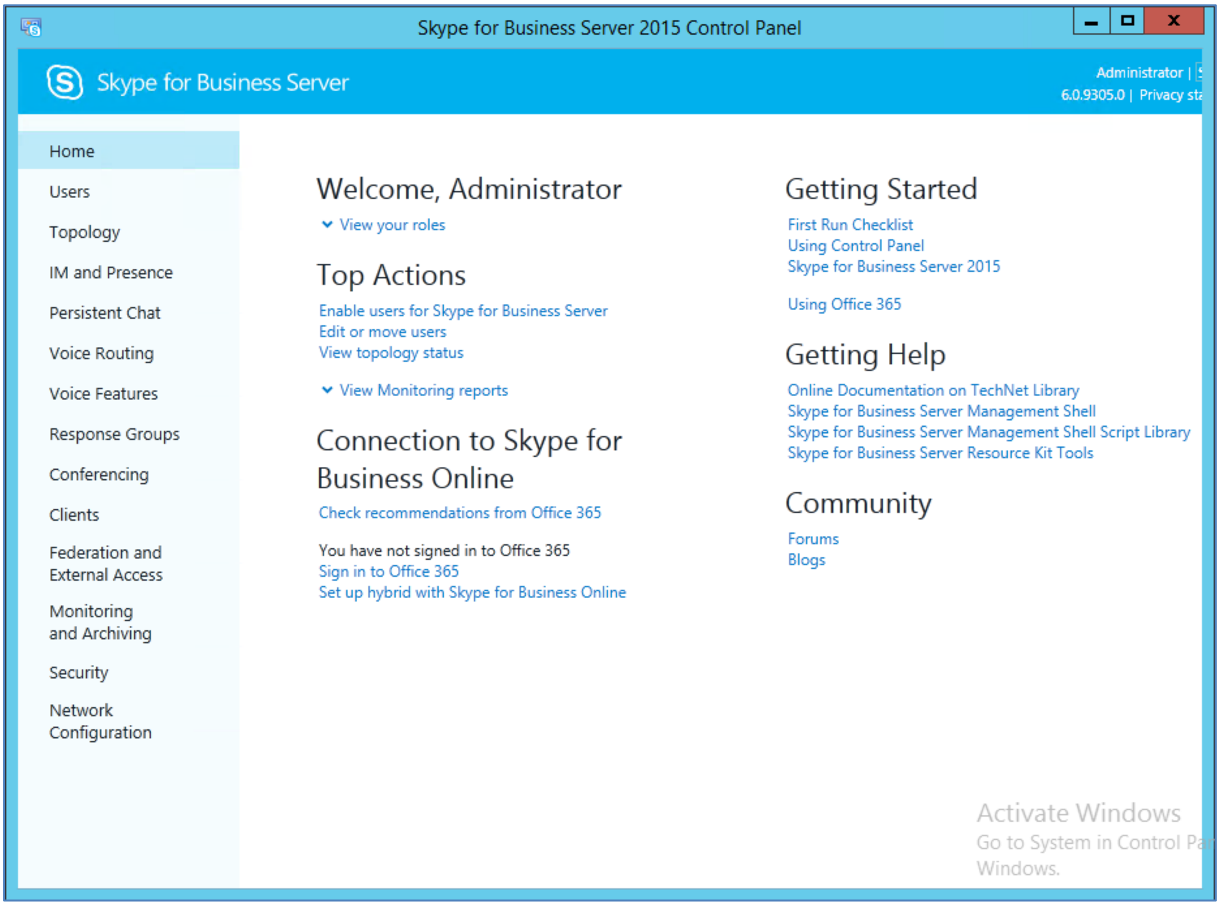
2. You are prompted to enter your login credentials:

Figure 3-15: Skype for Business Server Credentials



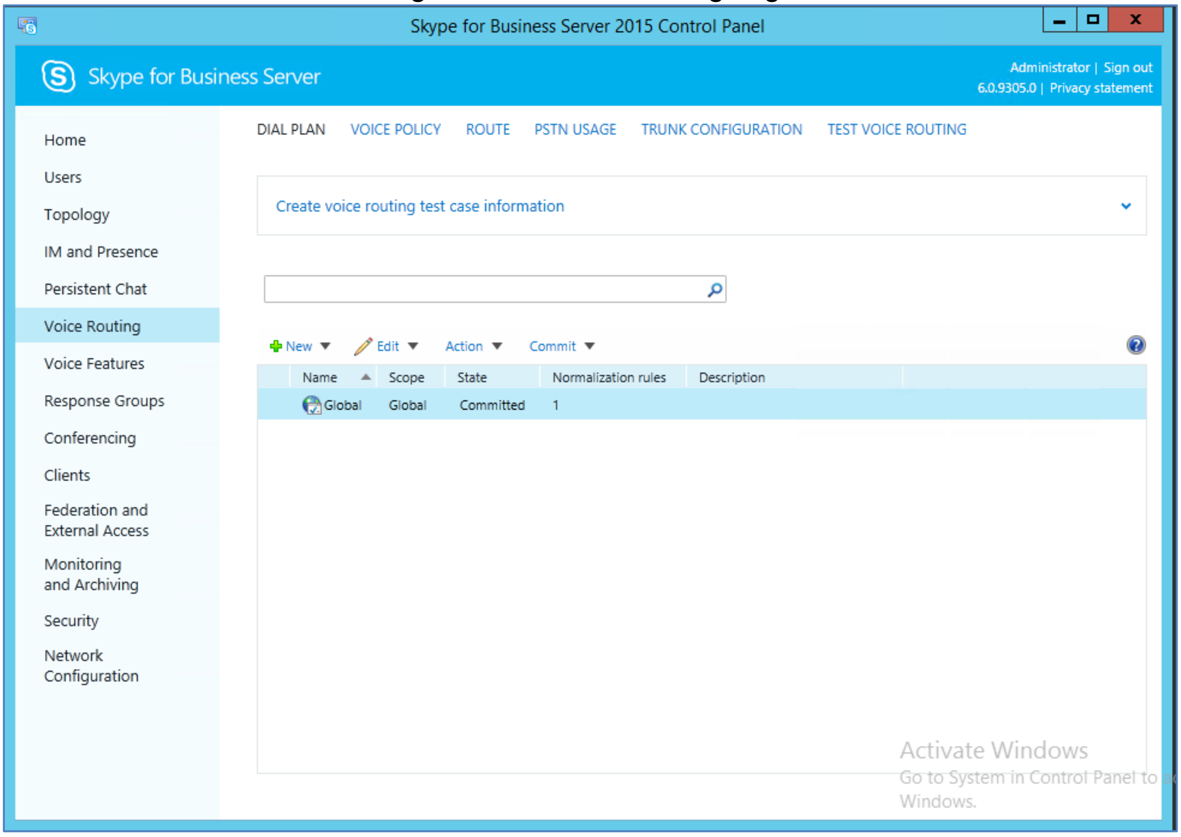
3. Enter your domain username and password, and then click **OK**; the Microsoft Skype for Business Server 2015 Control Panel is displayed:

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



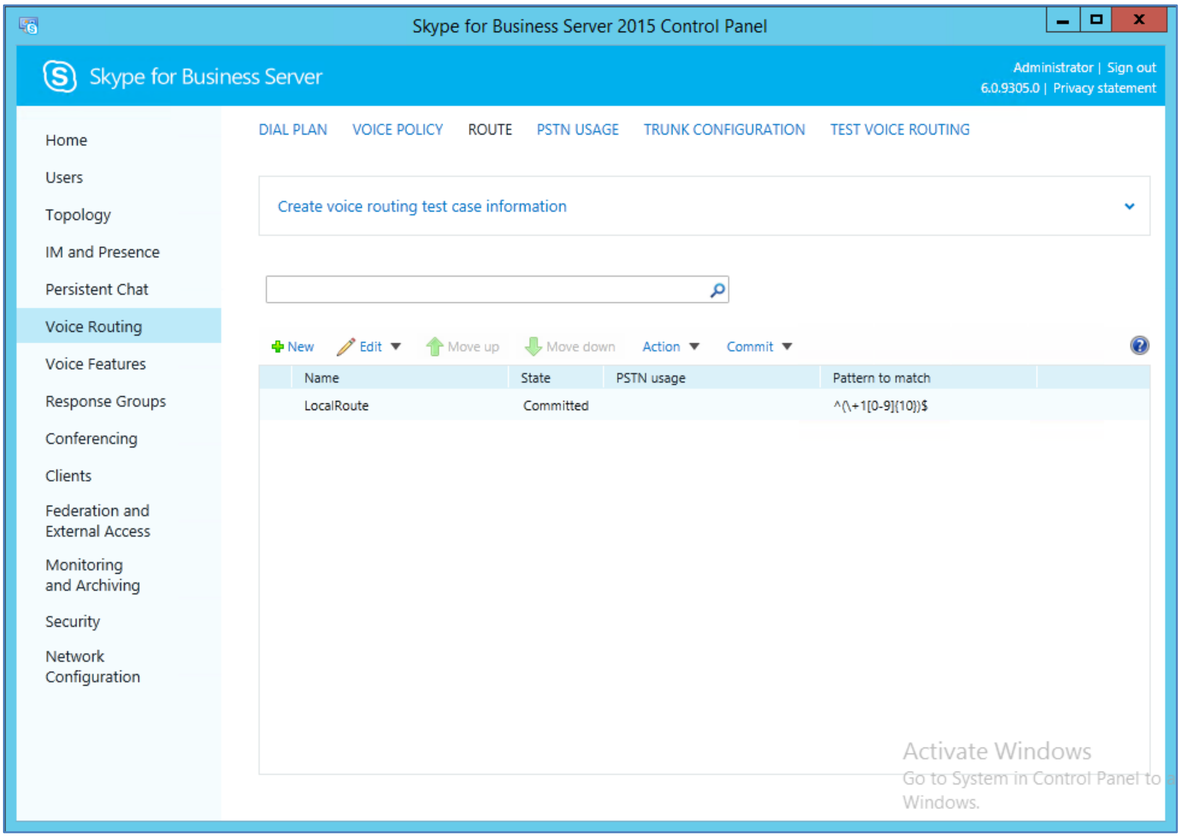
4. In the left navigation pane, select **Voice Routing**.

Figure 3-17: Voice Routing Page



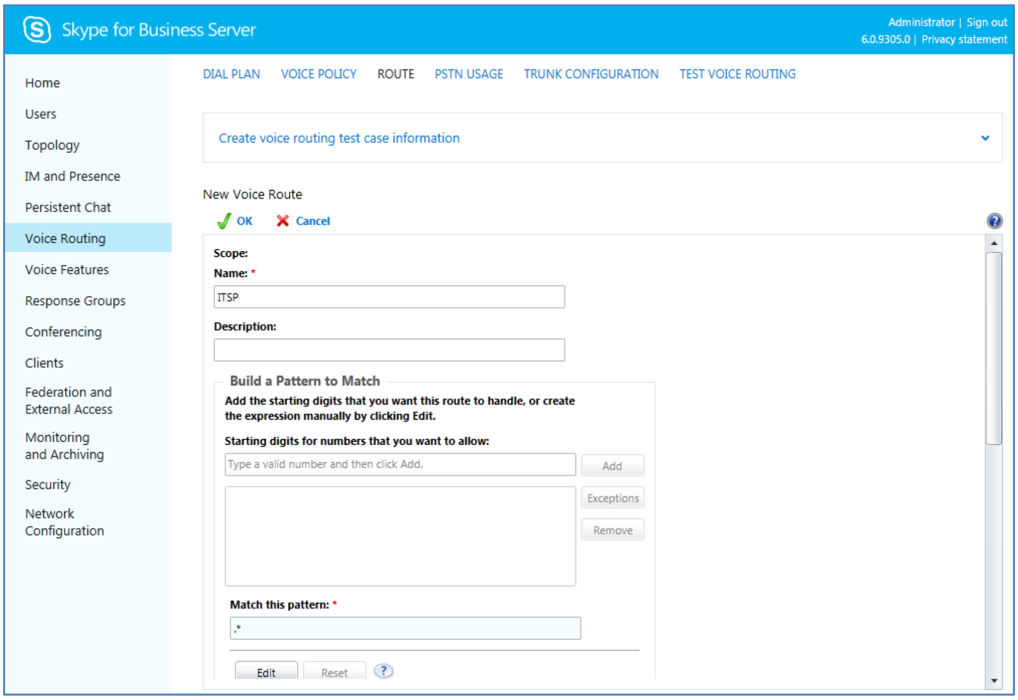
5. On the Voice Routing page, select the **Route** tab.

Figure 3-18: Route Tab



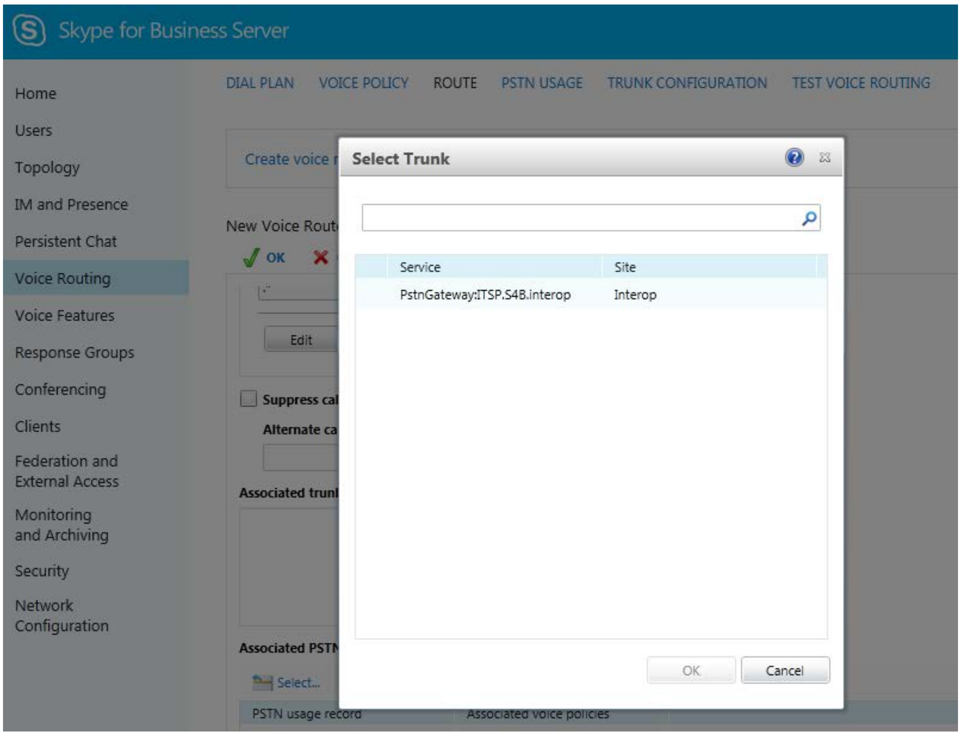
6. Click **New**; the New Voice Route page appears:

Figure 3-19: Adding New Voice Route



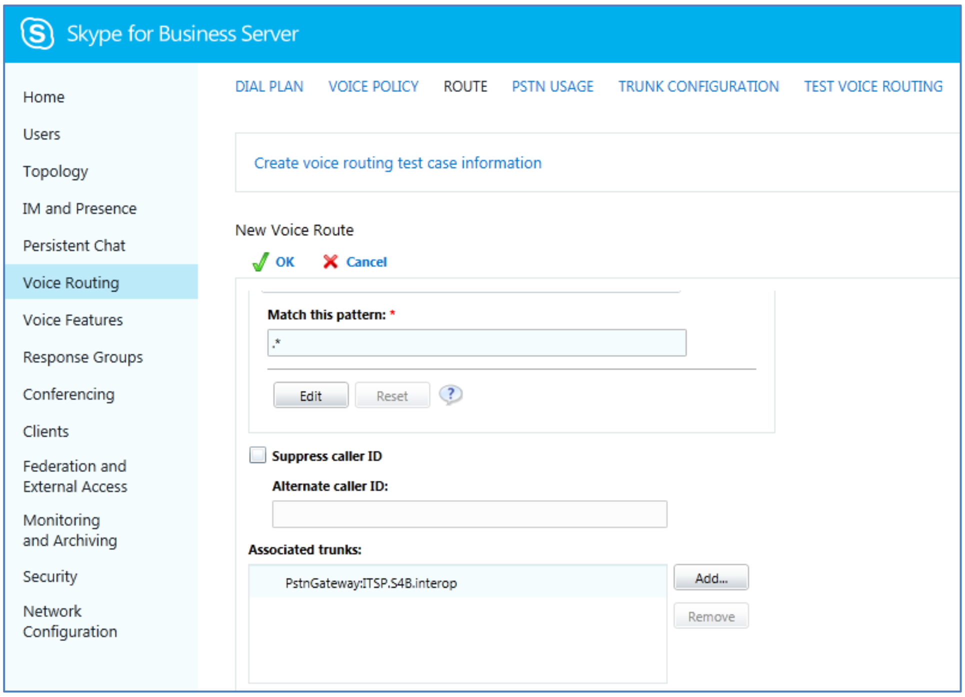
7. In the 'Name' field, enter a name for this route (e.g., **ITSP**).
8. In the 'Starting digits for numbers that you want to allow' field, enter the starting digits you want this route to handle (e.g., * to match all numbers), and then click **Add**.
9. Associate the route with the E-SBC Trunk that you created:
    a. Under the 'Associated Trunks' group, click **Add**; a list of all the deployed gateways is displayed:

Figure 3-20: List of Deployed Trunks



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

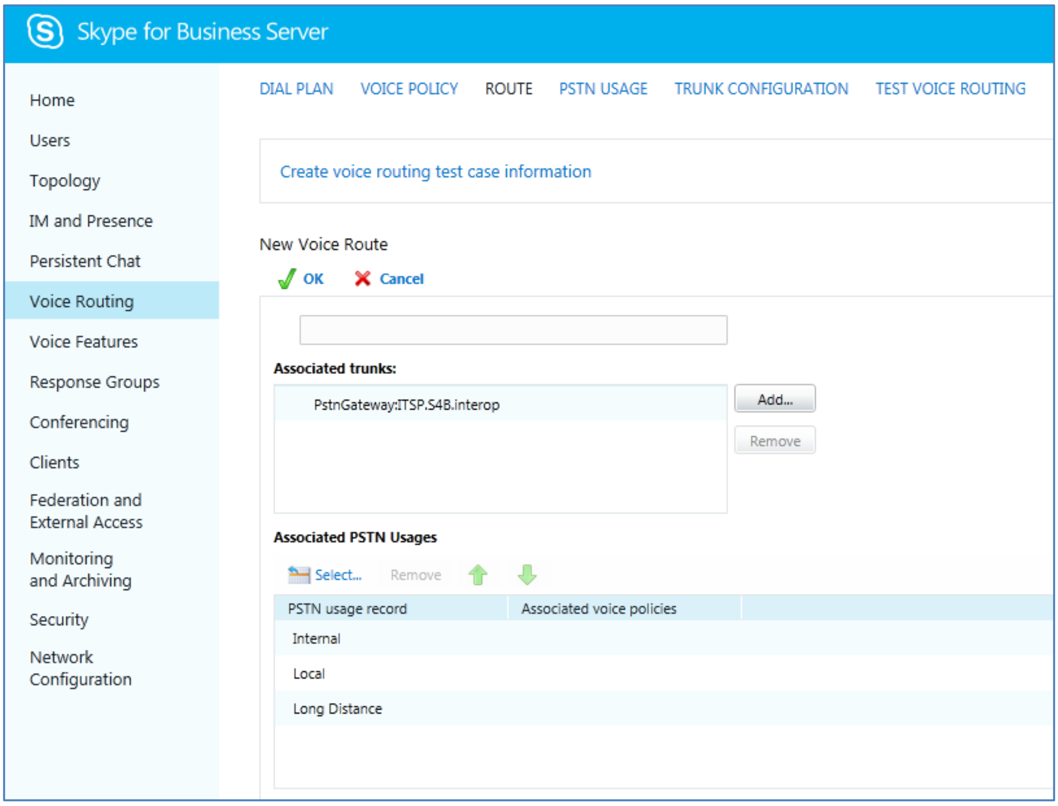
Figure 3-21: Selected E-SBC Trunk



10. Associate a PSTN Usage to this route:

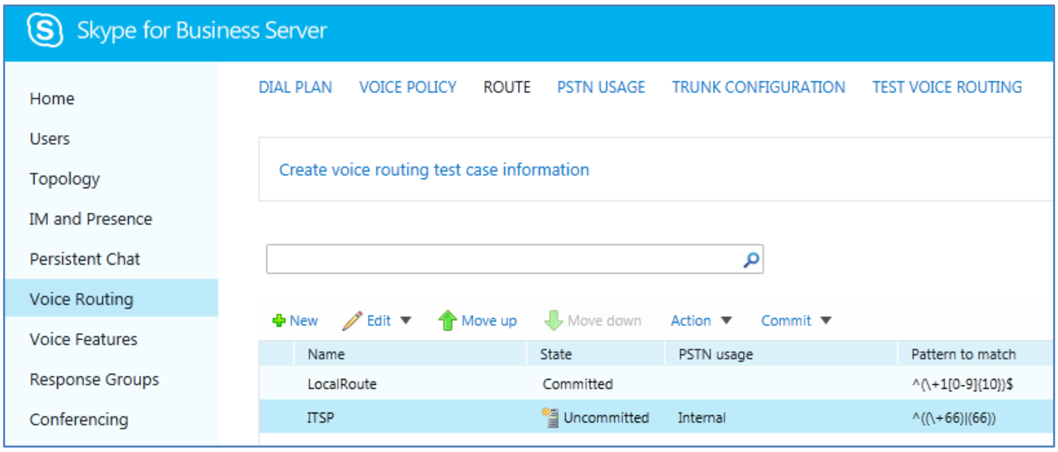
a. Under the 'Associated PSTN Usages' group, click **Select** and then add the associated PSTN Usage.

Figure 3-22: Associating PSTN Usage to Route



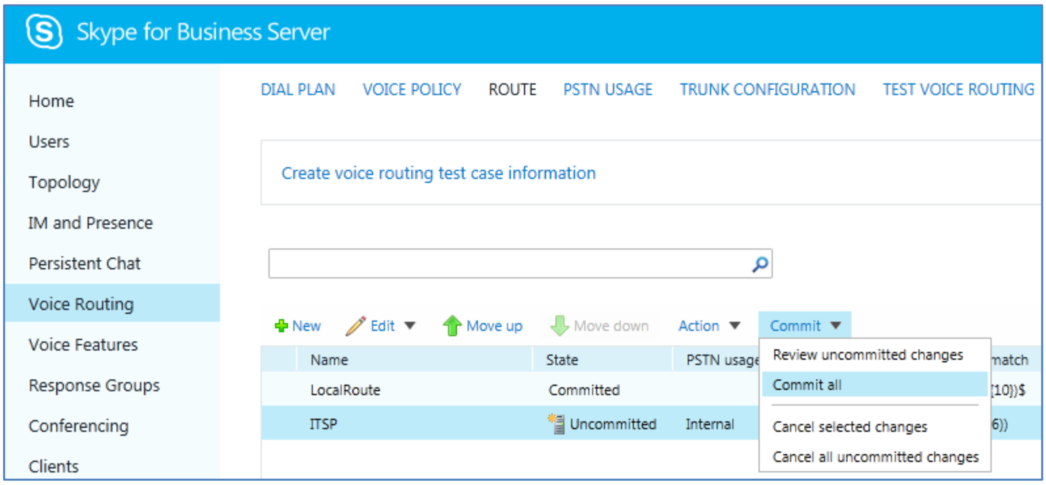
11. Click **OK** (located on the top of the New Voice Route page); the New Voice Route (Uncommitted) is displayed:

Figure 3-23: Confirmation of New Voice Route



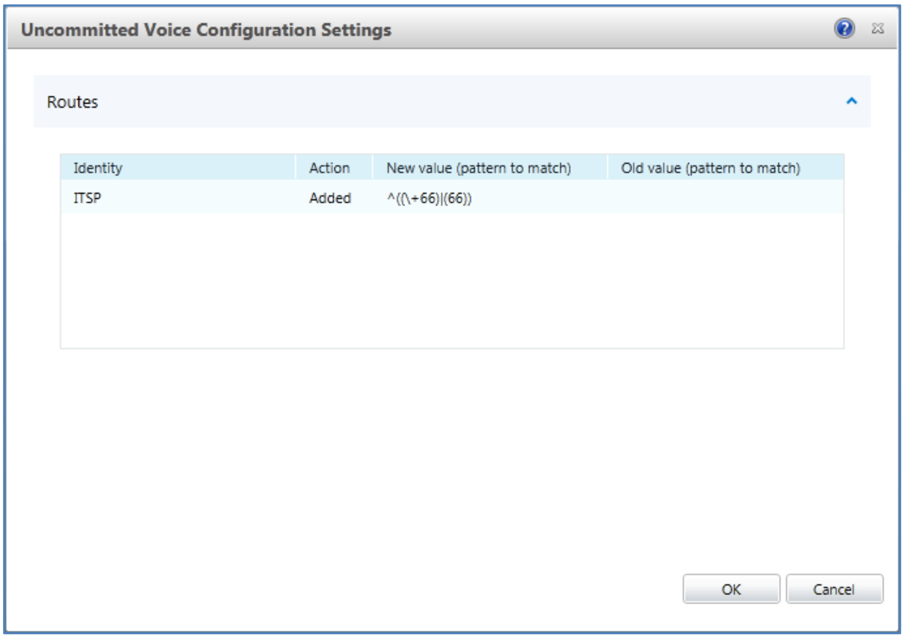
12. From the **Commit** drop-down list, choose **Commit all**, as shown below:

Figure 3-24: Committing Voice Routes



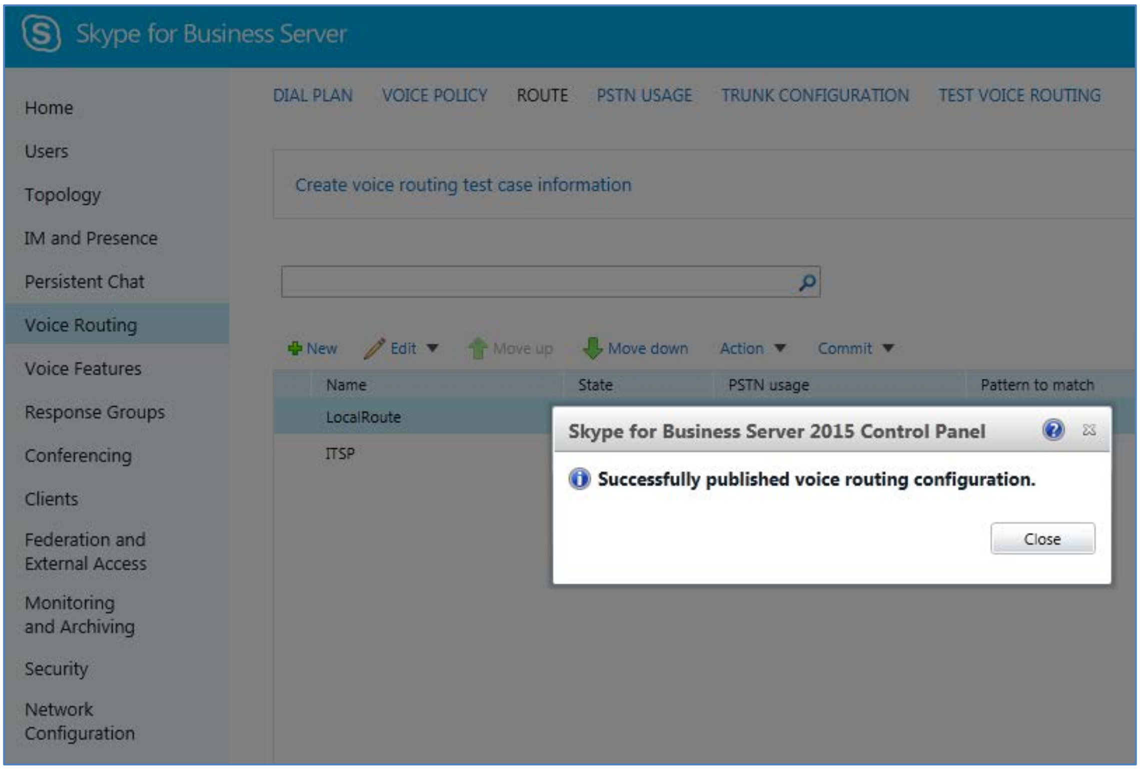
The Uncommitted Voice Configuration Settings page appears:

Figure 3-25: Uncommitted Voice Configuration Settings



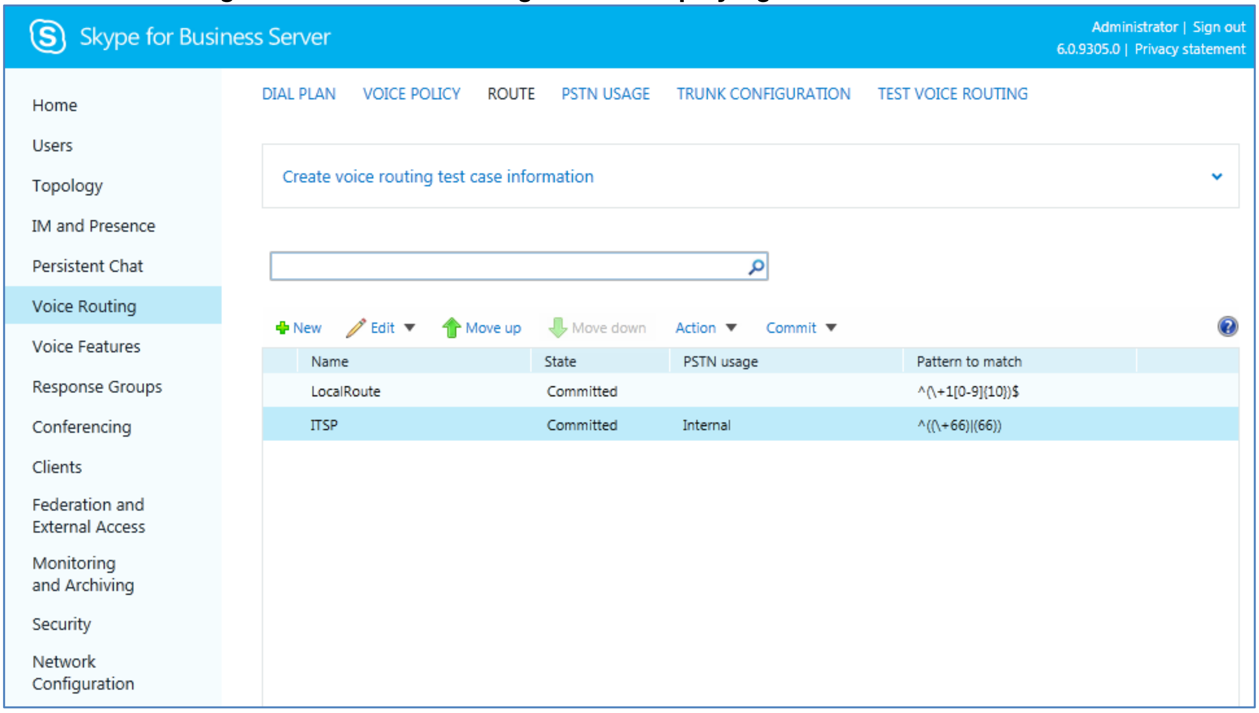
13. Click **Commit**; a message is displayed confirming a successful voice routing configuration, as shown below:

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

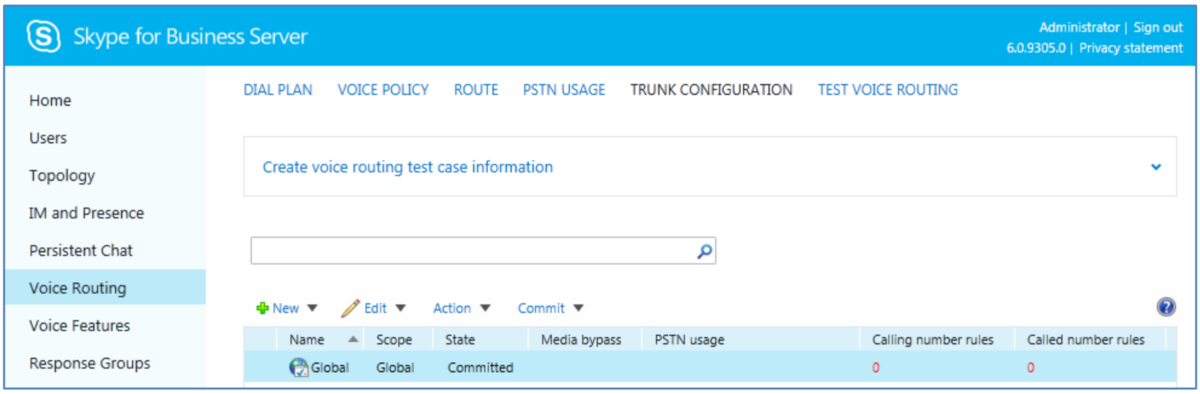


15. For ITSPs that implement a call identifier, continue with the following steps:

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 Virgin Media 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 47).

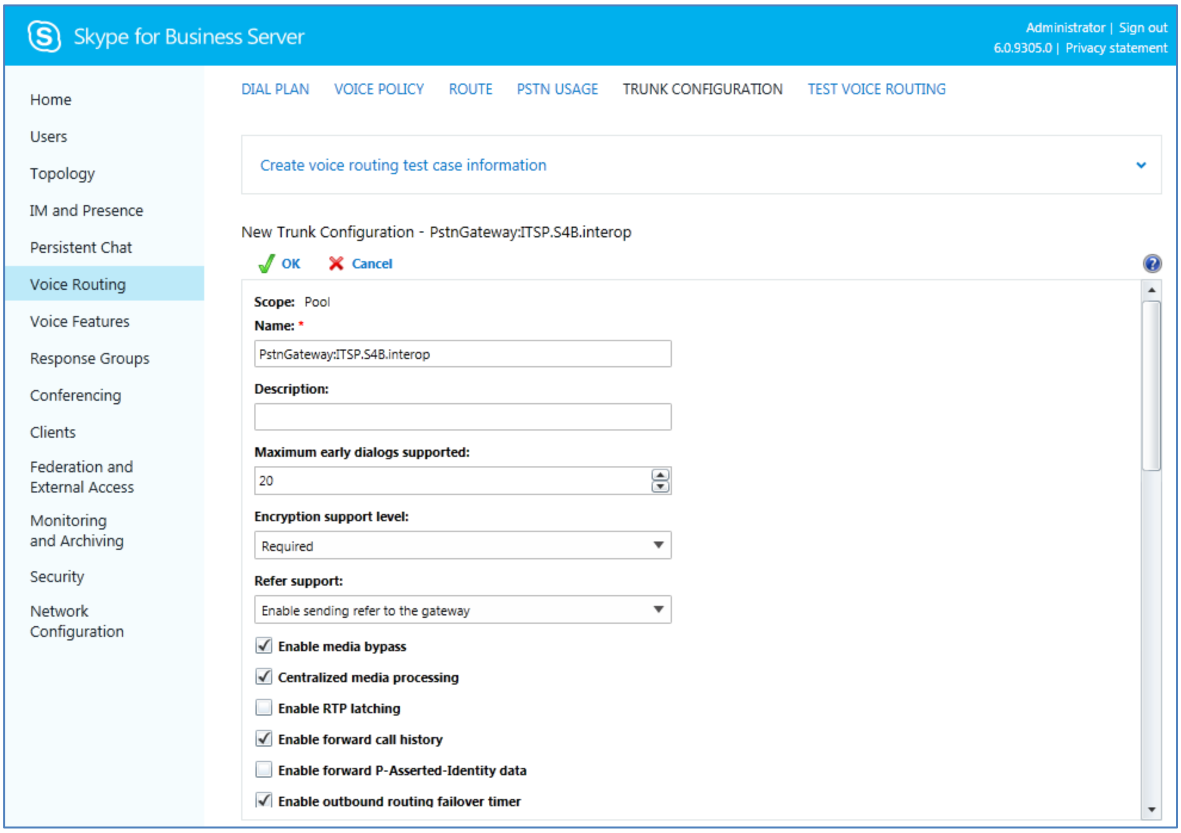
a. In the Voice Routing page, select the **Trunk Configuration** tab. Note that you can add and modify trunk configuration by site or by pool.

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



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

Figure 3-29:Edit Trunk Configuration Page



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

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

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

- **Get-CsTrunkConfiguration**

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

```
EnableRTPLatching                                : False
EnableOnlineVoice                                : False
ForwardCallHistory                               : True
Enable3pccRefer                                  : False
ForwardPAI                                       : False
EnableFastFailoverTimer                          : True
EnableLocationRestriction                        : False
NetworkSiteID                                    :
```

4 Configuring AudioCodes E-SBC

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

- E-SBC WAN interface - Virgin Media 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 Virgin Media SIP Trunk based on the configuration described in this section, AudioCodes E-SBC must be installed with a License Key that includes the following software features:
  - √ **Microsoft**
  - √ **SBC**
  - √ **Security**
  - √ **DSP**
  - √ **RTP**
  - √ **SIP**

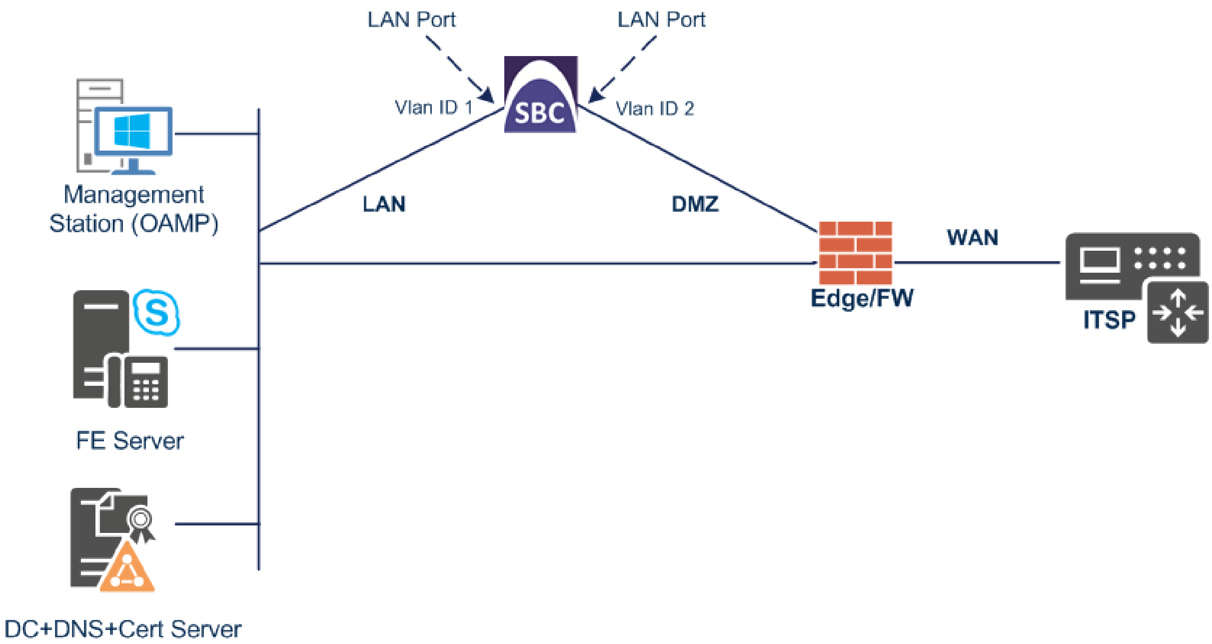
  For more information about the License Key, contact your AudioCodes sales representative.
- The scope of this interoperability test and document does **not** cover all security aspects for connecting the SIP Trunk to the Microsoft Skype for Business environment. Comprehensive security measures should be implemented per your organization's security policies. For security recommendations on AudioCodes' products, refer to the *Recommended Security Guidelines* document, which can be found at https://www.audiocodes.com/library/technical-documents.
- IP addresses used in this Configuration Note are for example purposes only and do not reflect the live environment.

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

Figure 4-1: Network Interfaces in Interoperability Test Topology



4.1.1 Step 1a: Configure VLANs

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

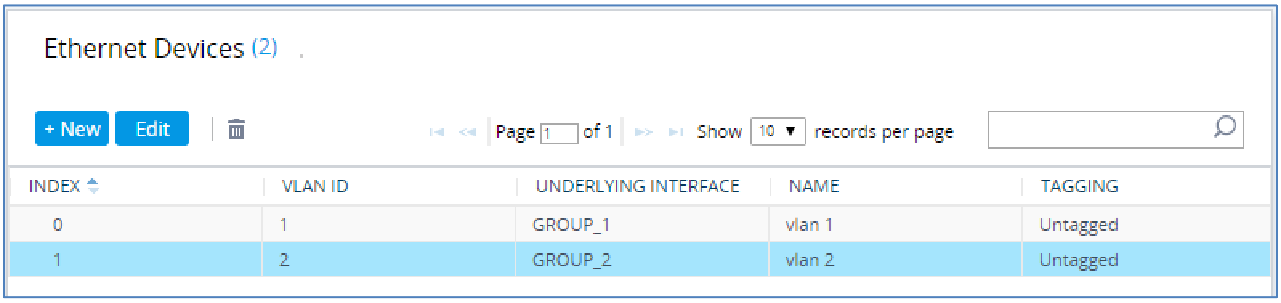
- LAN VoIP (assigned the name "LAN_IF")
- WAN VoIP (assigned the name "WAN_IF")

➢ **To configure the VLANs:**

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

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

Figure 4-2: Configured VLAN IDs in Ethernet Device



4.1.2 Step 1b: Configure Network Interfaces

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

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

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

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

b. Configure the interface as follows:

Parameter	Value
IP Address	**10.15.77.10** (LAN IP address of E-SBC)
Prefix Length	**16** (subnet mask in bits for 255.255.0.0)
Default Gateway	**10.15.0.1**
Interface Name	**LAN_IF** (arbitrary descriptive name)
Primary DNS Server IP Address	**10.15.27.1**
Underlying Device	**vlan 1**

3. Add a network interface for the WAN side:

   a. Click **New**.

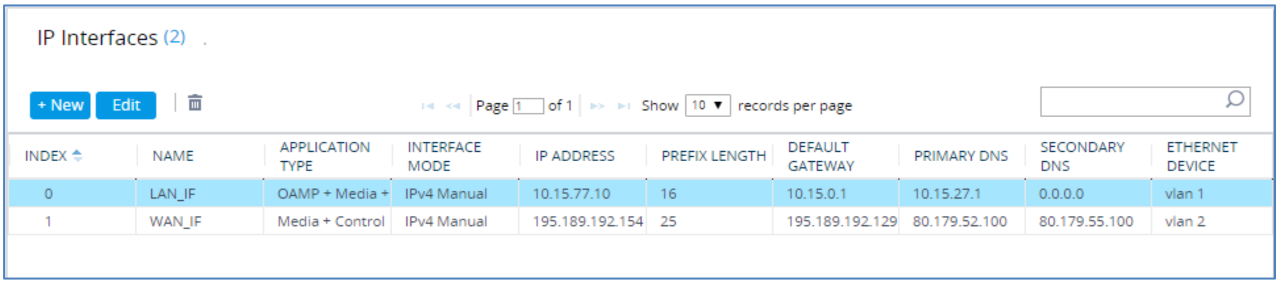
   b. Configure the interface as follows:

Parameter	Value
Application Type	**Media + Control**
IP Address	**195.189.192.154** (DMZ IP address of E-SBC)
Prefix Length	**25** (subnet mask in bits for 255.255.255.128)
Default Gateway	**195.189.192.129** (router's IP address)
Interface Name	**WAN_IF**
Primary DNS Server IP Address	**80.179.52.100**
Secondary DNS Server IP Address	**80.179.55.100**
Underlying Device	**vlan 2**

4. Click **Apply**.

The configured IP network interfaces are shown below:

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


IP Interfaces (2)

+ New Edit Page 1 of 1 Show 10 records per page

INDEX	NAME	APPLICATION TYPE	INTERFACE MODE	IP ADDRESS	PREFIX LENGTH	DEFAULT GATEWAY	PRIMARY DNS	SECONDARY DNS	ETHERNET DEVICE
0	LAN_IF	OAMP + Media +	IPv4 Manual	10.15.77.10	16	10.15.0.1	10.15.27.1	0.0.0.0	vlan 1
1	WAN_IF	Media + Control	IPv4 Manual	195.189.192.154	25	195.189.192.129	80.179.52.100	80.179.55.100	vlan 2

4.2 Step 2: Configure Media Realms

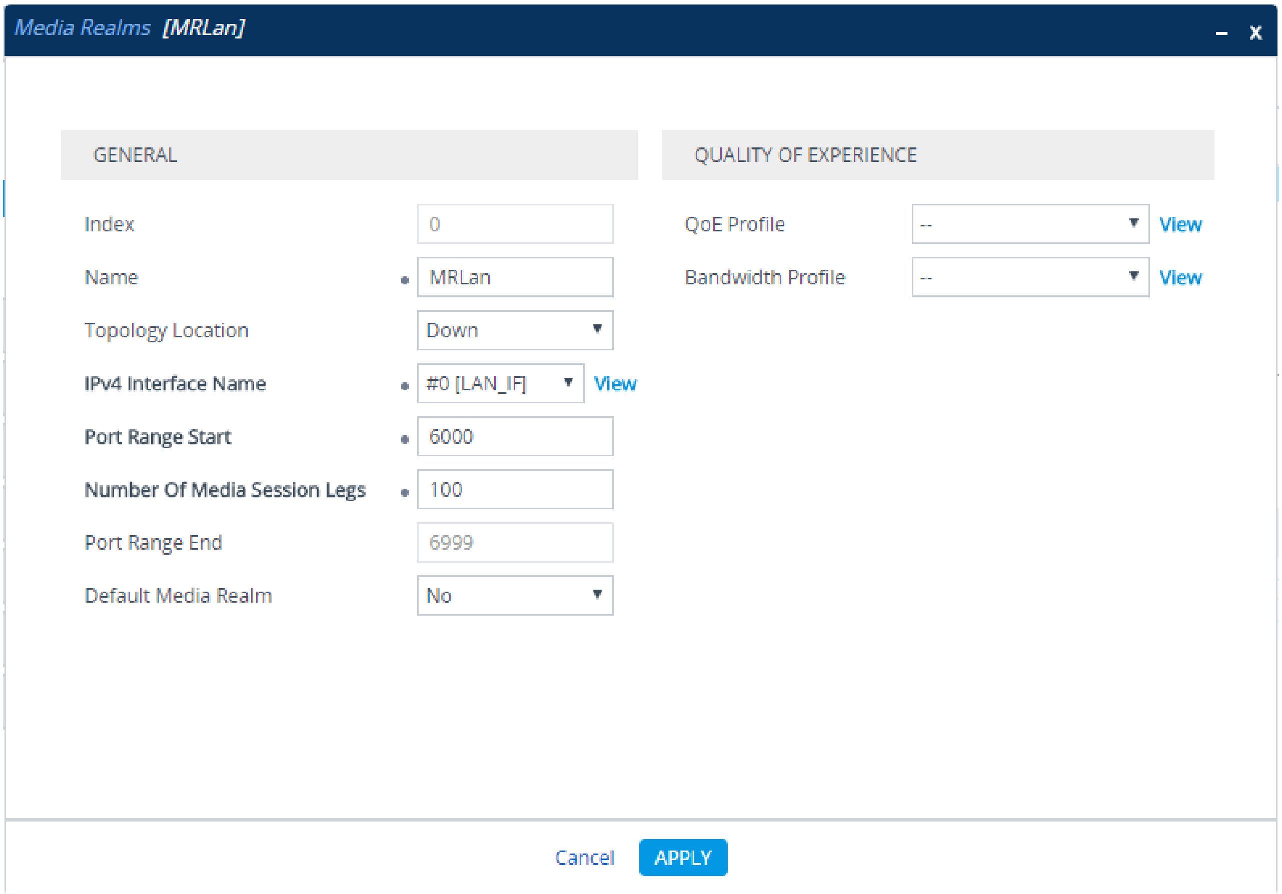
This step describes how to configure Media Realms. The simplest configuration is to create two Media Realms - one for internal (LAN) traffic and one for external (WAN) traffic.

➢ **To configure Media Realms:**

1. Open the Media Realms table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **Media Realms**).
2. Add a Media Realm for the LAN interface. You can use the default Media Realm (Index 0), but modify it as shown below:

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

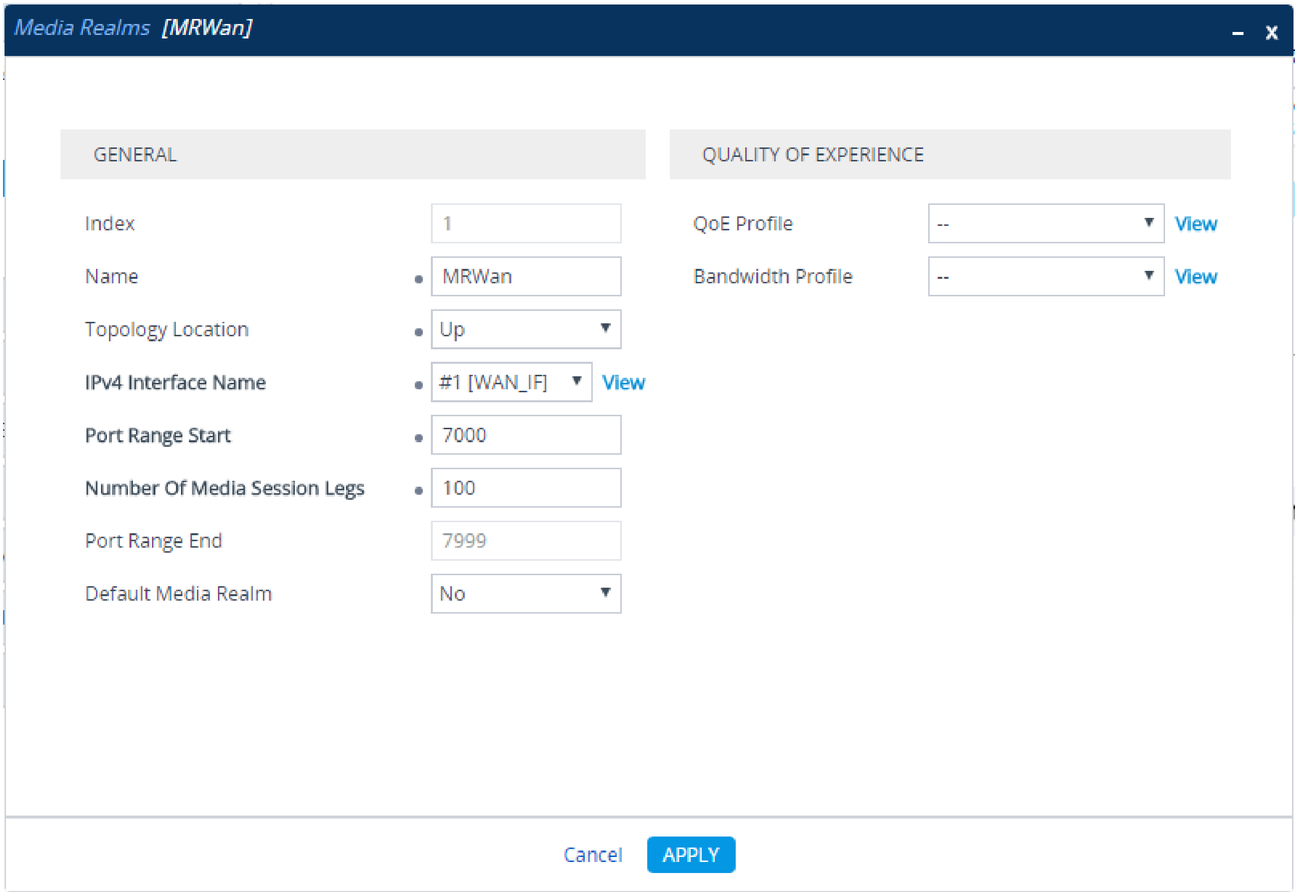
Figure 4-4: Configuring Media Realm for LAN



3. Click **New** to configure a Media Realm for WAN traffic:

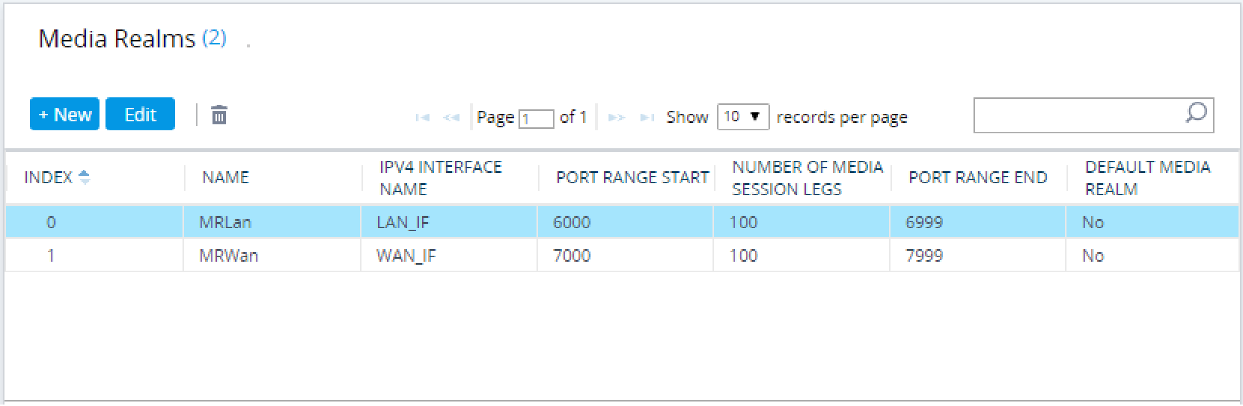
Parameter	Value
Index	**1**
Media Realm Name	**MRWan** (arbitrary name)
Topology Location	**Up**
IPv4 Interface Name	**WAN_IF**
Port Range Start	**7000** (represents lowest UDP port number used for media on WAN)
Number of Media Session Legs	**100** (media sessions assigned with port range)

Figure 4-5: Configuring Media Realm for WAN



The configured Media Realms are shown in the figure below:

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



INDEX	NAME	IPV4 INTERFACE NAME	PORT RANGE START	NUMBER OF MEDIA SESSION LEGS	PORT RANGE END	DEFAULT MEDIA REALM
0	MRLan	LAN_IF	6000	100	6999	No
1	MRWan	WAN_IF	7000	100	7999	No

4.3 Step 3: Configure SIP Signaling Interfaces

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

➢ **To configure SIP Interfaces:**

1. Open the SIP Interfaces table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **SIP Interfaces**).
2. Add a SIP Interface for the LAN interface. You can use the default SIP Interface (Index 0), but modify it as shown below:

Parameter	Value
Index	**0**
Interface Name	**S4B** (see note at the end of this section)
Network Interface	**LAN_IF**
Application Type	**SBC**
UDP Port (for supporting Fax ATA device)	**5060** (if required)
TCP	**0**
TLS Port	**5067** (see note below)
Media Realm	**MRLan**

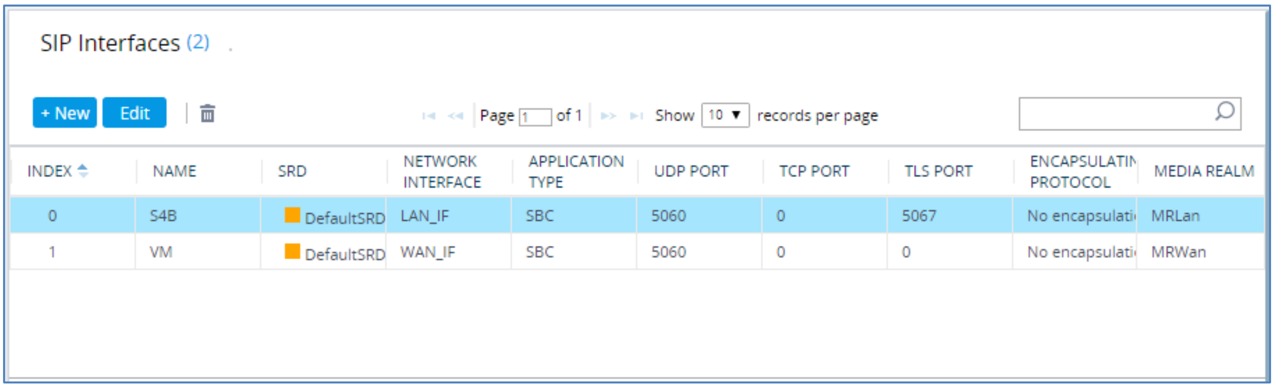
Note: The TLS port parameter must be identically configured in the Skype for Business Topology Builder (see Section 3.1 on page 13).

3. Click **New** to configure a SIP Interface for the WAN:

Parameter	Value
Index	**1**
Interface Name	**VM**
Network Interface	**WAN_IF**
Application Type	**SBC**
UDP Port	**5060**
TCP and TLS	**0**
Media Realm	**MRWan**

The configured SIP Interfaces are shown in the figure below:

Figure 4-7: Configured SIP Interfaces in SIP Interface Table



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

4.4 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, Proxy Sets need to be configured for the following IP entities:

- Microsoft Skype for Business Server 2015
- Virgin Media SIP Trunk A
- Virgin Media SIP Trunk B
- Fax supporting ATA device (optional)

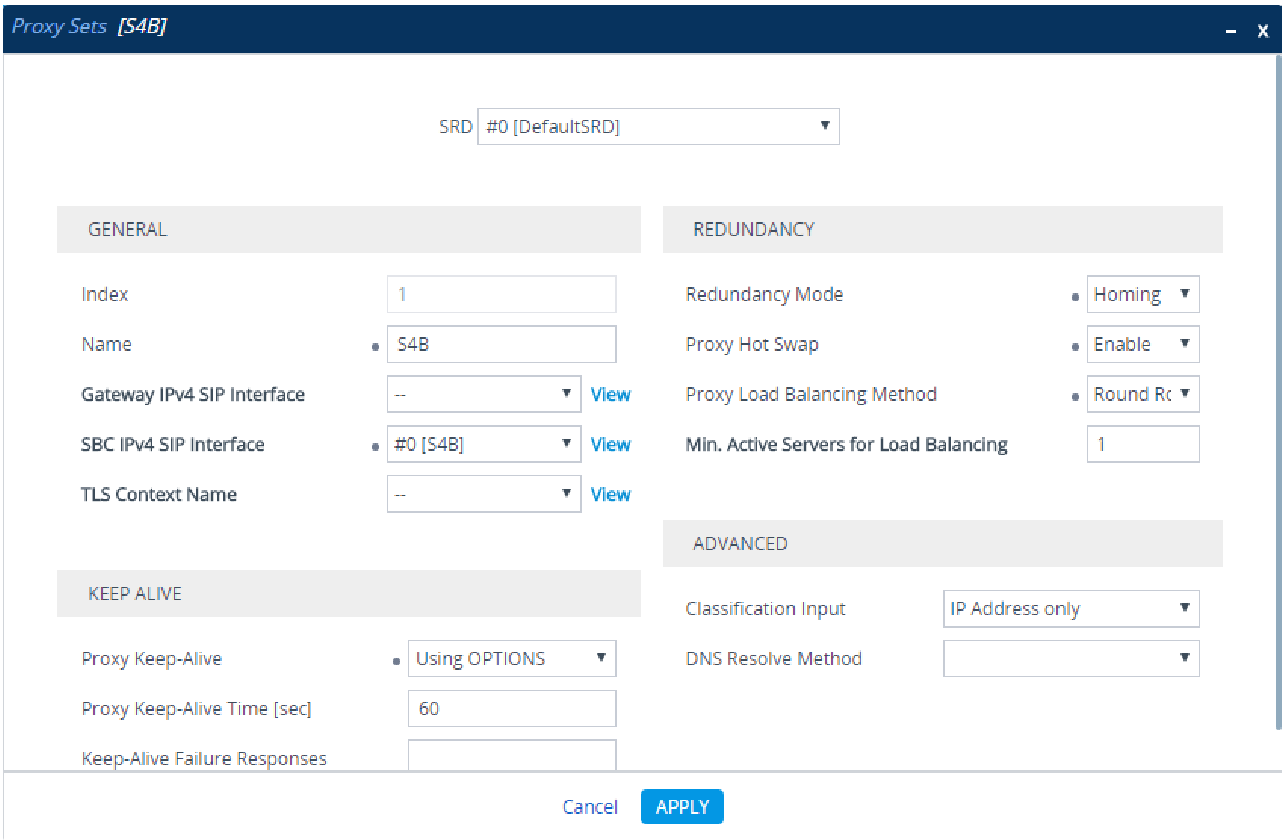
The Proxy Sets will be later applying to the VoIP network by assigning them to IP Groups.

➢ **To configure Proxy Sets:**

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

Parameter	Value
Proxy Set ID	**1**
Proxy Name	**S4B**
SBC IPv4 SIP Interface	**S4B**
Proxy Keep Alive	**Using Options**
Redundancy Mode	**Homing**
Proxy Hot Swap	**Enable**
Load Balancing Method	**Round Robin**

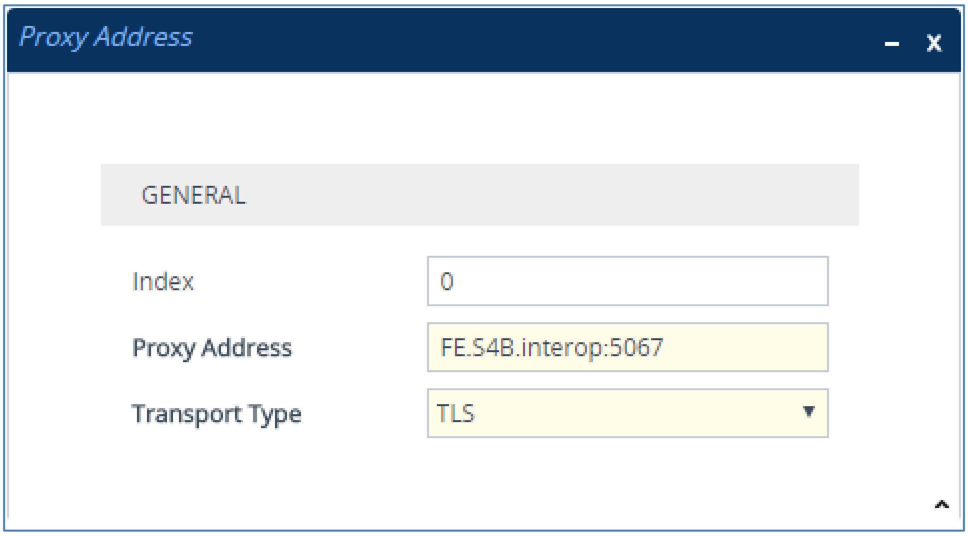
Figure 4-8: Configuring Proxy Set for Microsoft Skype for Business Server 2015



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

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

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



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

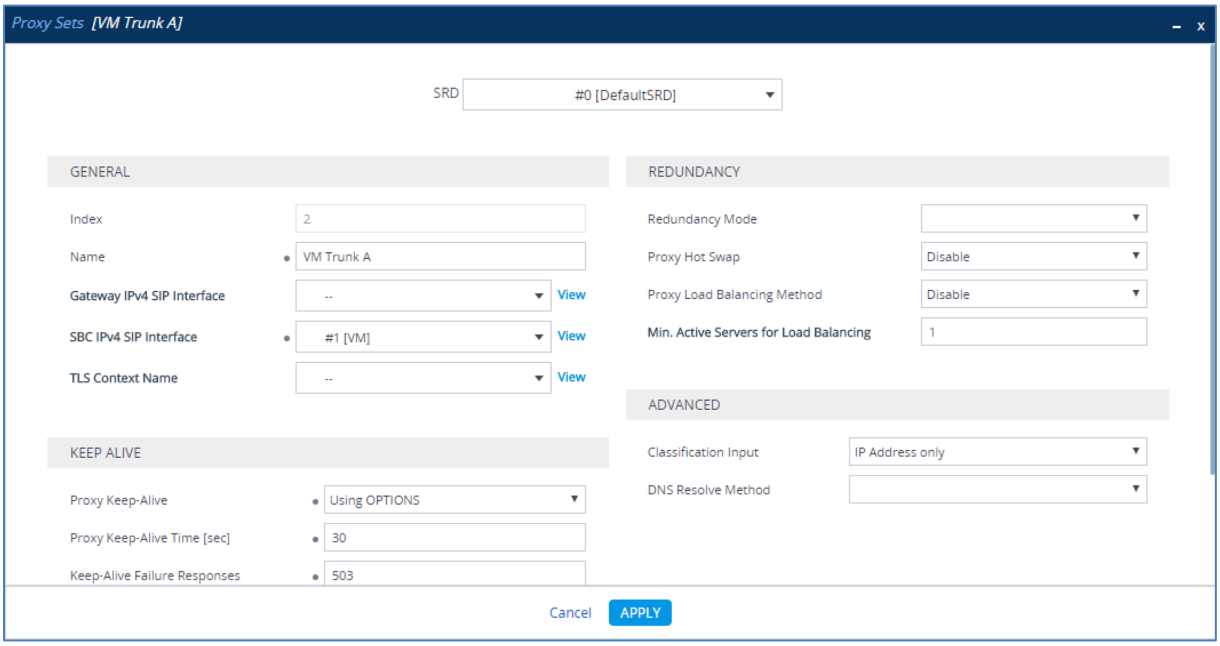
d. Click **Apply**.

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

3. Configure a Proxy Set for the Virgin Media SIP Trunk A:

Parameter	Value
Proxy Set ID	**2**
Proxy Name	**VM Trunk A**
SBC IPv4 SIP Interface	**VM**
Proxy Keep-Alive	**Using Options**
Proxy Keep-Alive Time [sec]	**30**
Keep-Alive Failure Responses	**503**

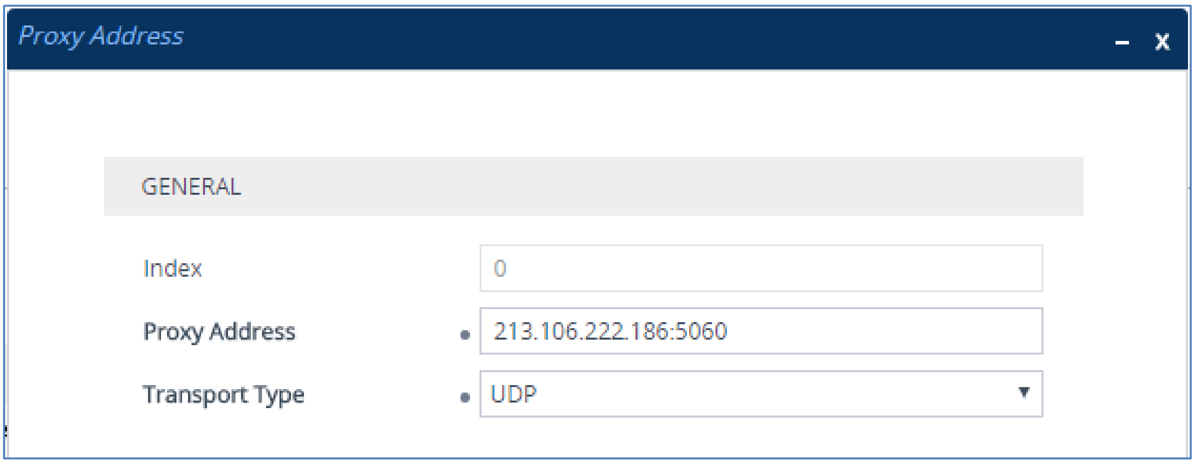
Figure 4-10: Configuring Proxy Set for Virgin Media SIP Trunk A



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

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

Figure 4-11: Configuring Proxy Address for Virgin Media SIP Trunk A



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

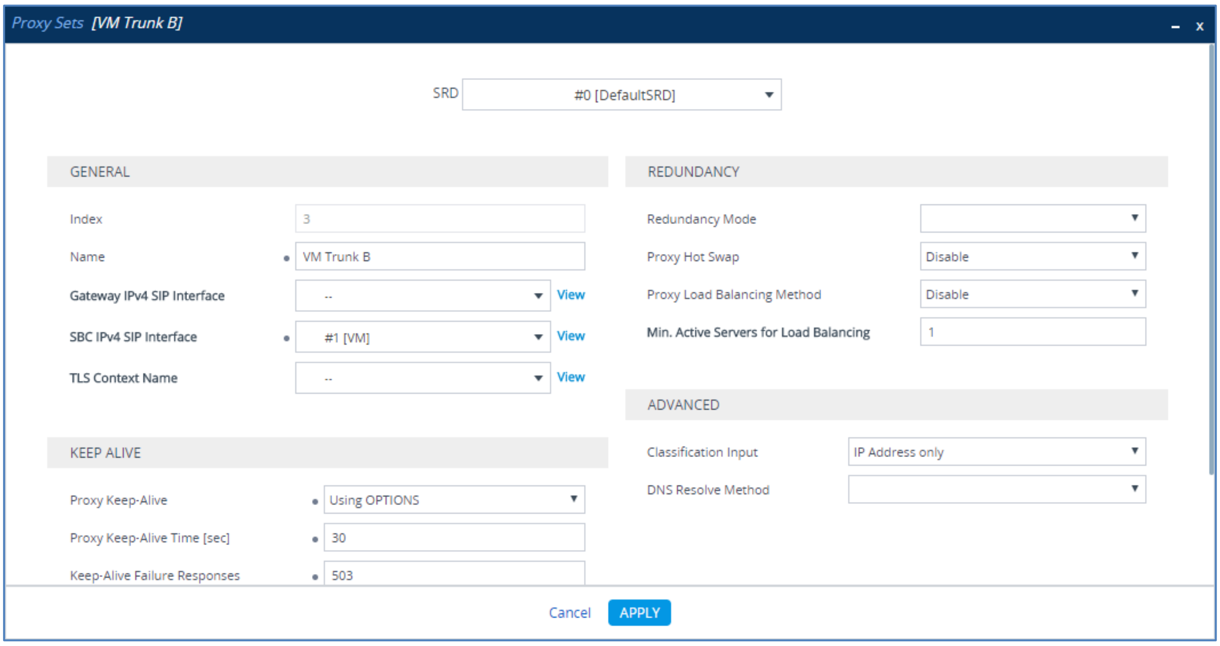
d. Click **Apply**.

Parameter	Value
Index	**0**
Proxy Address	**213.106.222.186:5060** (IP address / FQDN and destination port)
Transport Type	**UDP**

4. Configure a Proxy Set for the Virgin Media SIP Trunk B:

Parameter	Value
Proxy Set ID	**3**
Proxy Name	**VM Trunk B**
SBC IPv4 SIP Interface	**VM**
Proxy Keep-Alive	**Using Options**
Proxy Keep-Alive Time [sec]	**30**
Keep-Alive Failure Responses	**503**

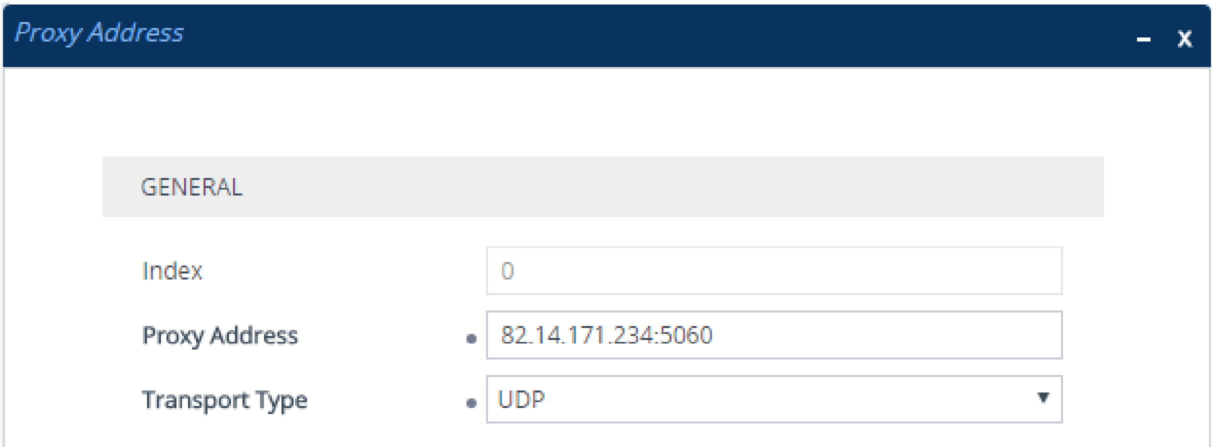
Figure 4-12: Configuring Proxy Set for Virgin Media SIP Trunk B



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

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

Figure 4-13: Configuring Proxy Address for Virgin Media SIP Trunk B



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

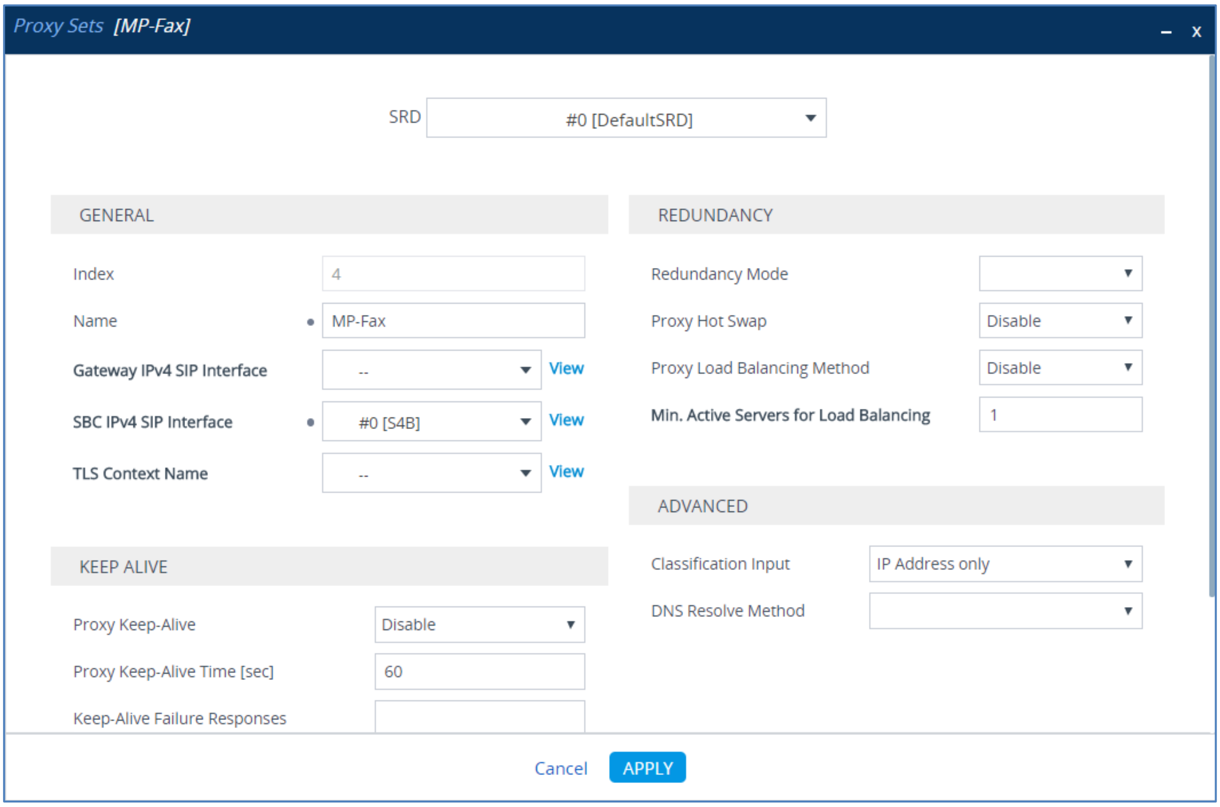
d. Click **Apply**.

Parameter	Value
Index	**0**
Proxy Address	**82.14.171.234:5060** (IP address / FQDN and destination port)
Transport Type	**UDP**

5. Configure a Proxy Set for Fax supporting ATA device (if required):

Parameter	Value
Proxy Set ID	**4**
Proxy Name	**MP-Fax**
SBC IPv4 SIP Interface	**S4B**

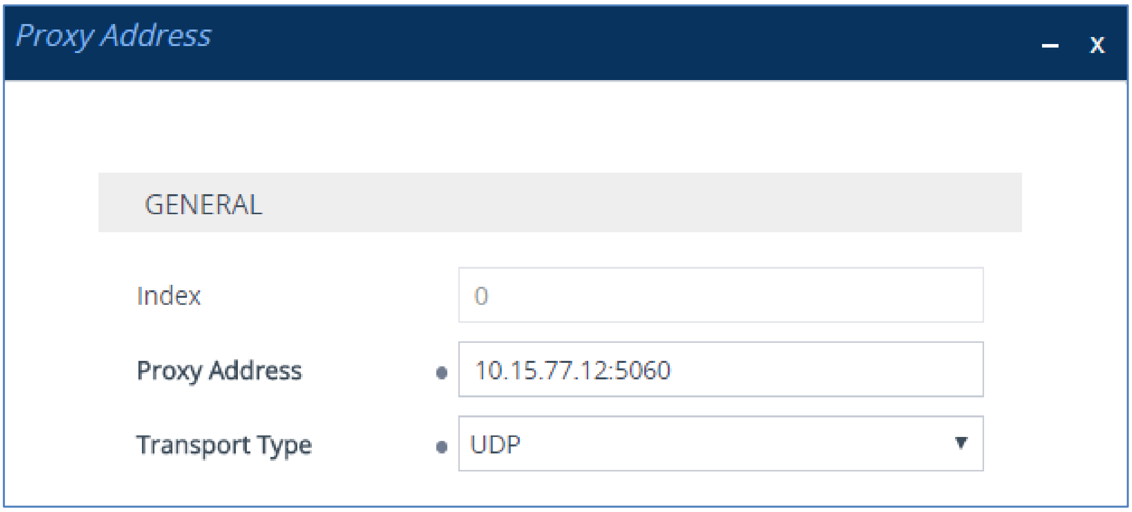
Figure 4-14: Configuring Proxy Set for Fax ATA device



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

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

Figure 4-15: Configuring Proxy Address for Fax ATA device



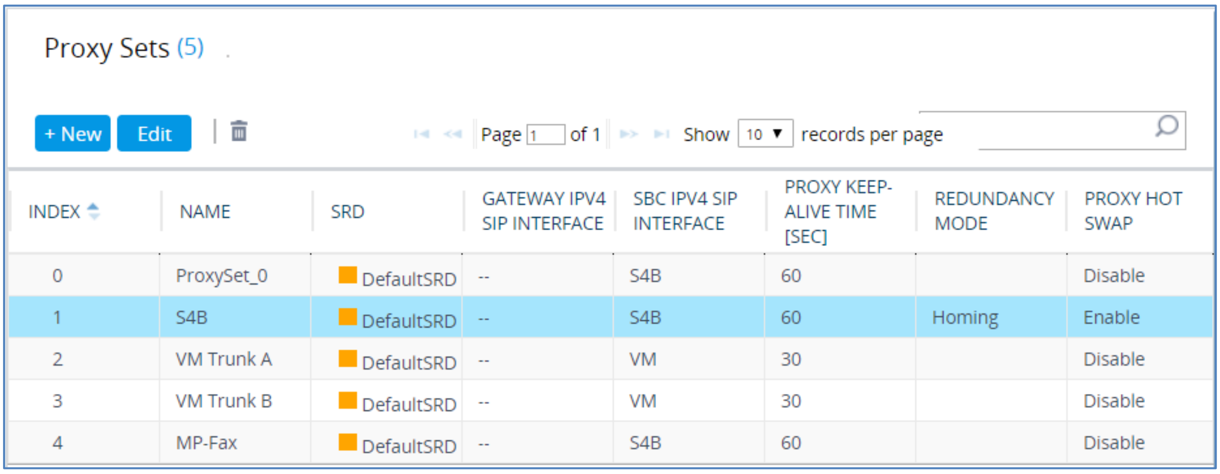
c. Configure the address of the Proxy Set according to the parameters described in the table below.

d. Click **Apply**.

Parameter	Value
Index	**0**
Proxy Address	**10.15.17.12:5060** (IP address / FQDN and destination port)
Transport Type	**UDP**

The configured Proxy Sets are shown in the figure below:

Figure 4-16: Configured Proxy Sets in Proxy Sets Table



Proxy Sets (5)

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

INDEX	NAME	SRD	GATEWAY IPV4 SIP INTERFACE	SBC IPV4 SIP INTERFACE	PROXY KEEP-ALIVE TIME [SEC]	REDUNDANCY MODE	PROXY HOT SWAP
0	ProxySet_0	DefaultSRD	--	S4B	60		Disable
1	S4B	DefaultSRD	--	S4B	60	Homing	Enable
2	VM Trunk A	DefaultSRD	--	VM	30		Disable
3	VM Trunk B	DefaultSRD	--	VM	30		Disable
4	MP-Fax	DefaultSRD	--	S4B	60		Disable

4.5 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:

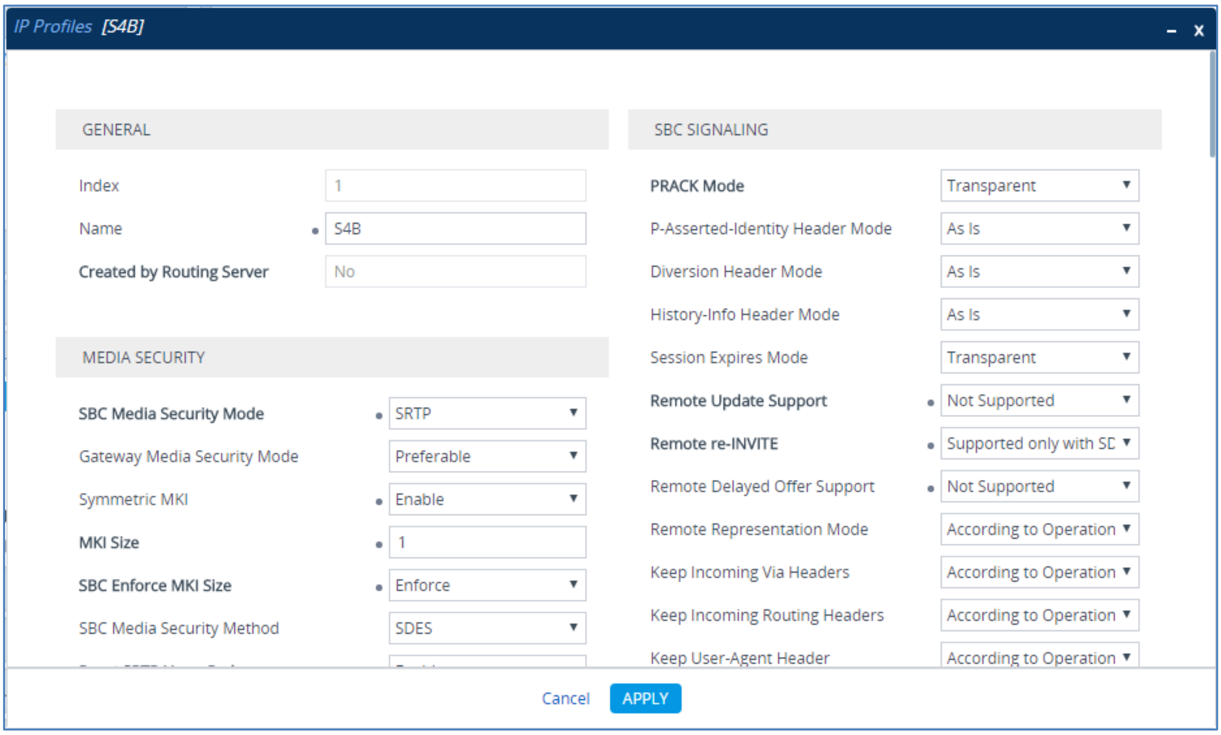
- Microsoft Skype for Business Server 2015 - to operate in secure mode using SRTP and TLS
- Virgin Media SIP trunk - to operate in non-secure mode using RTP and UDP
- Fax ATA device – to operate in non-secure mode using RTP and UDP

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

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

Parameter	Value
General	
Index	**1**
Name	**S4B**
Media Security	
SBC Media Security Mode	**SRTP**
Symmetric MKI	**Enable**
MKI Size	**1**
Enforce MKI Size	**Enforce**
Reset SRTP State Upon Re-key	**Enable**
Generate SRTP Keys Mode:	**Always**
SBC Early Media	
Remote Early Media RTP Detection Mode	**By Media** (required, as Skype for Business Server 2015 does not send RTP immediately to remote side when it sends a SIP 18x response)
SBC Signaling	
Remote Update Support	**Not Supported**
Remote re-INVITE Support	**Supported Only With SDP**
Remote Delayed Offer Support	**Not Supported**
SBC Forward and Transfer	
Remote REFER Mode	**Handle Locally** (required, as Skype for Business Server 2015 does not support receipt of SIP REFER)
Remote 3xx Mode	**Handle Locally** (required, as Skype for Business Server 2015 does not support receipt of SIP 3xx responses)

Figure 4-17: Configuring IP Profile for Skype for Business Server 2015



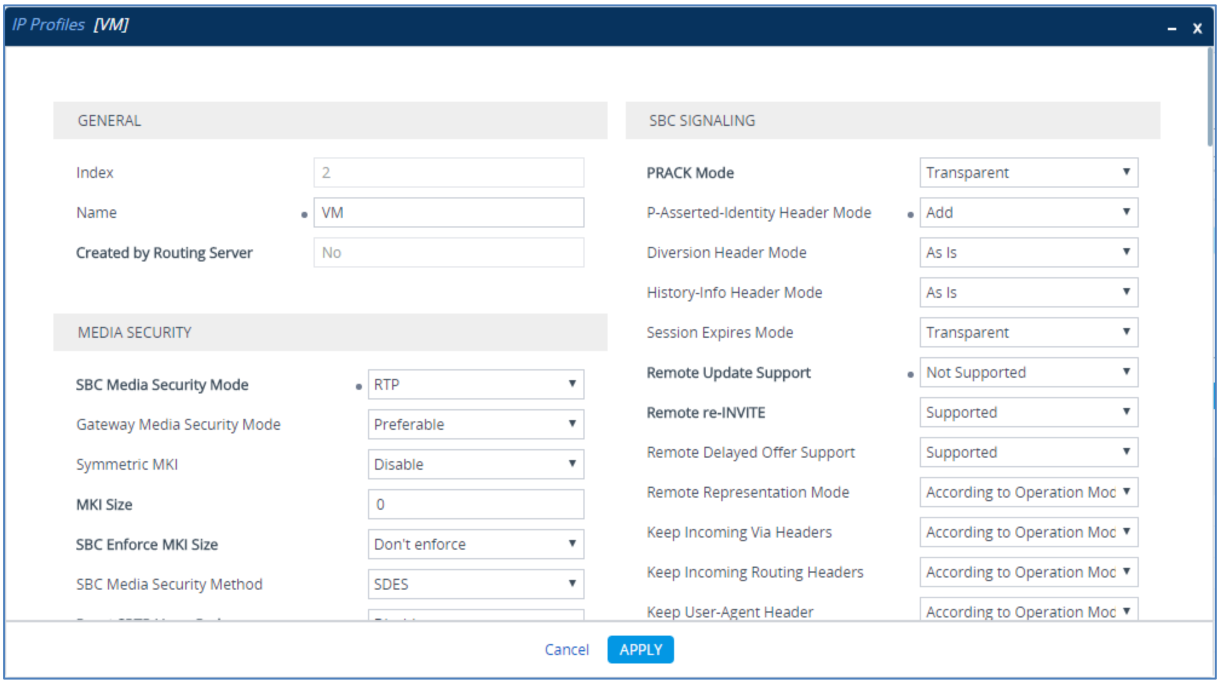
3. Click **Apply**.

➢ To configure an IP Profile for the Virgin Media SIP Trunk:

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

Parameter	Value
General	
Index	**2**
Name	**VM**
Media Security	
SBC Media Security Mode	**RTP**
SBC Media	
Allowed Audio Coders	**VM**
SBC Signaling	
Remote Update Support	**Not Supported**
P-Asserted-Identity Header Mode	**Add** (required for anonymous calls)
SBC Forward and Transfer	
Remote REFER Mode	**Handle Locally** (required, as Skype for Business Server 2015 does not support receipt of SIP REFER)
SBC Hold	
Remote Hold Format	**Send Only** (required, as Virgin Media work is in Send Only mode)

Figure 4-18: Configuring IP Profile for Virgin Media SIP Trunk



2. Click **Apply**.

4.6 Step 6: Configure IP Groups

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

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

- Skype for Business Server 2015 (Mediation Server) located on LAN
- Virgin Media SIP Trunk located on WAN
- Fax supporting ATA device located on LAN (if required)

➢ **To configure IP Groups:**

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

Parameter	Value
Index	**1**
Name	**S4B**
Topology Location	**Down**
Type	**Server**
Proxy Set	**S4B**
IP Profile	**S4B**
Media Realm	**MRLan**
SIP Group Name	**213.106.222.186** (according to ITSP requirement)
Classify By Proxy Set	**Enable**

3. Configure an IP Group for the Virgin Media SIP Trunk A:

Parameter	Value
Index	**2**
Name	**VM Trunk A**
Topology Location	**Up**
Type	**Server**
Proxy Set	**VM Trunk A**
IP Profile	**VM**
Media Realm	**MRWan**
SIP Group Name	**213.106.222.186** (according to ITSP requirement)

4. Configure an IP Group for the Virgin Media SIP Trunk B:

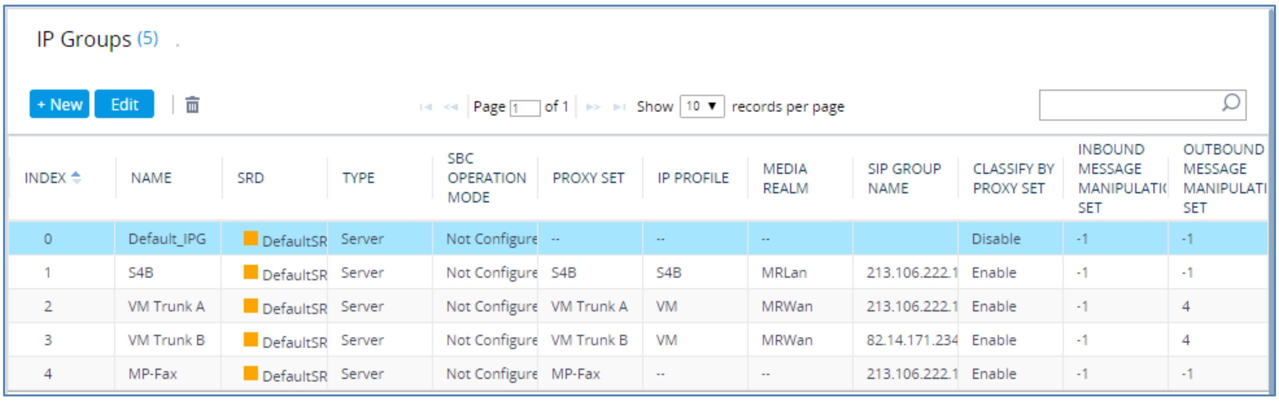
Parameter	Value
Index	**3**
Name	**VM Trunk B**
Topology Location	**Up**
Type	**Server**
Proxy Set	**VM Trunk B**
IP Profile	**VM**
Media Realm	**MRWan**
SIP Group Name	**82.14.171.234** (according to ITSP requirement)

5. Configure an IP Group for the Fax supporting ATA device:

Parameter	Value
Index	**3**
Name	**MP-Fax**
Type	**Server**
Proxy Set	**MP-Fax**
Media Realm	**MRLan**
SIP Group Name	(according to ITSP requirement)

The configured IP Groups are shown in the figure below:

Figure 4-19: Configured IP Groups in IP Group Table


IP Groups (5)

+ New Edit Page 1 of 1 Show 10 records per page

INDEX	NAME	SRD	TYPE	SBC OPERATION MODE	PROXY SET	IP PROFILE	MEDIA REALM	SIP GROUP NAME	CLASSIFY BY PROXY SET	INBOUND MESSAGE MANIPULATI(SET	OUTBOUND MESSAGE MANIPULATI SET
0	Default_IPG	DefaultSR	Server	Not Configure	--	--	--		Disable	-1	-1
1	S4B	DefaultSR	Server	Not Configure	S4B	S4B	MRLan	213.106.222.1	Enable	-1	-1
2	VM Trunk A	DefaultSR	Server	Not Configure	VM Trunk A	VM	MRWan	213.106.222.1	Enable	-1	4
3	VM Trunk B	DefaultSR	Server	Not Configure	VM Trunk B	VM	MRWan	82.14.171.234	Enable	-1	4
4	MP-Fax	DefaultSR	Server	Not Configure	MP-Fax	--	--	213.106.222.1	Enable	-1	-1

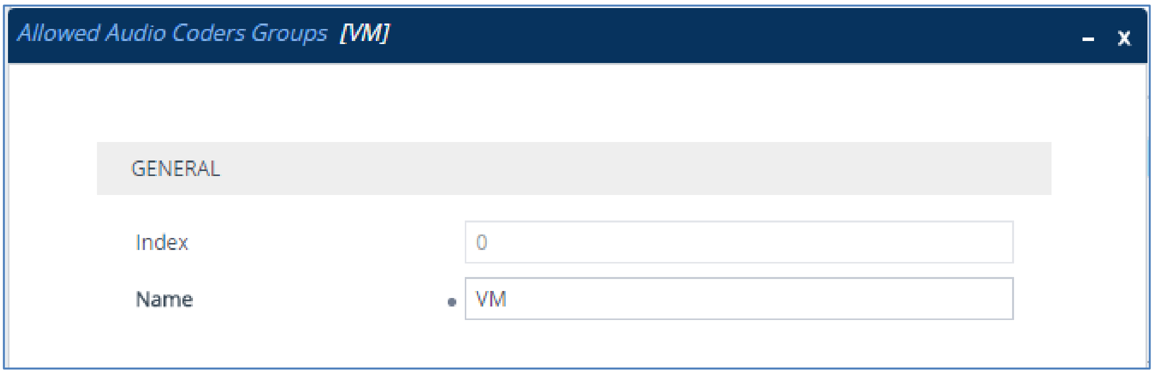
4.7 Step 7: Configure Coders

The procedure below describes how to configure an Allowed Coders Group to ensure that voice sent to the Virgin Media SIP Trunk uses the G.711A-law coder whenever possible. Note that this Allowed Coders Group ID was assigned to the IP Profile belonging to the Virgin Media SIP Trunk (see Section 4.5 on page 47).

➢ **To set a preferred coder for the Virgin Media SIP Trunk:**

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

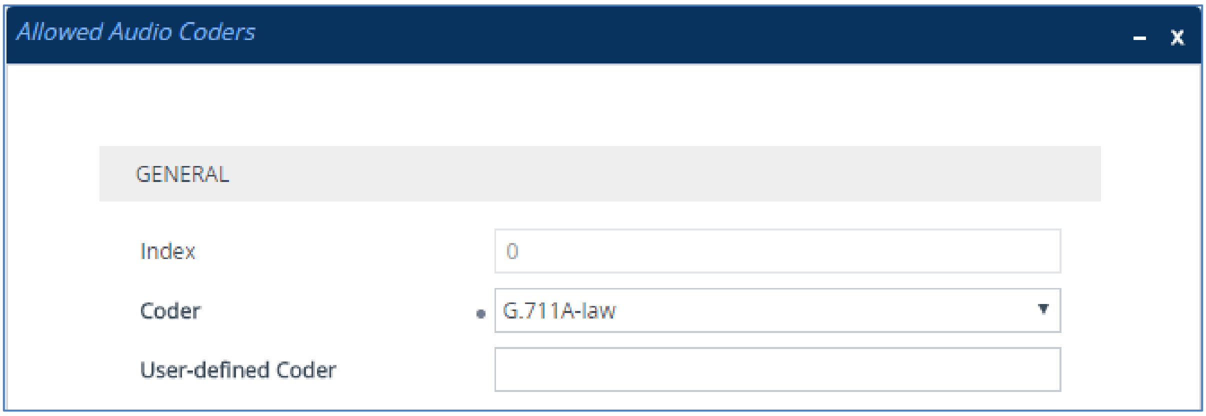
Figure 4-20: Configuring Allowed Coders Group for Virgin Media SIP Trunk



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

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

Figure 4-21: Configuring Allowed Coders for Virgin Media SIP Trunk



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.

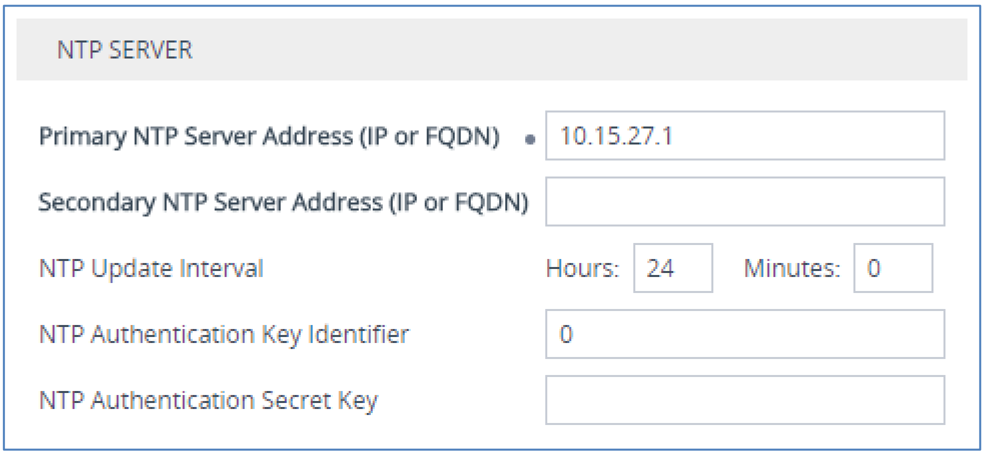
4.8.1 Step 8a: Configure the NTP Server Address

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

Figure 4-22: Configuring NTP Server Address



3. Click **Apply**.

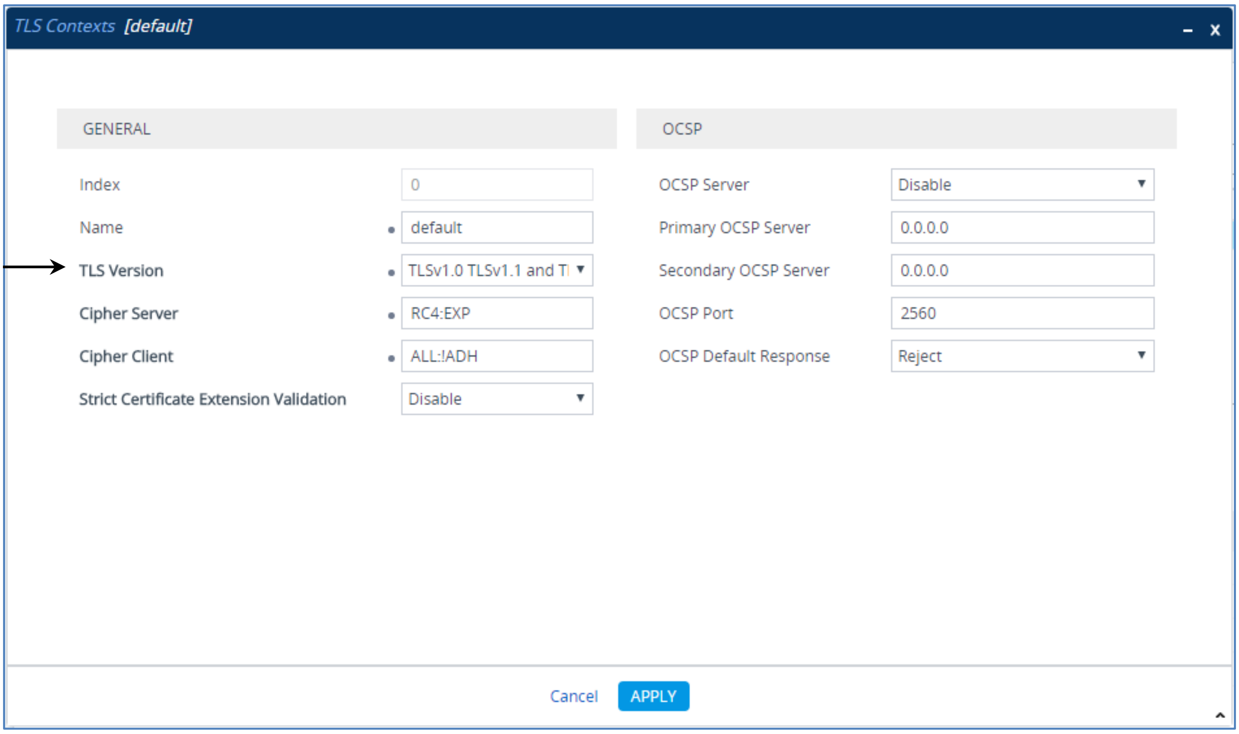
4.8.2 Step 8b: Configure the TLS version

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

Figure 4-23: Configuring TLS Version



4. Click **Apply**.

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.

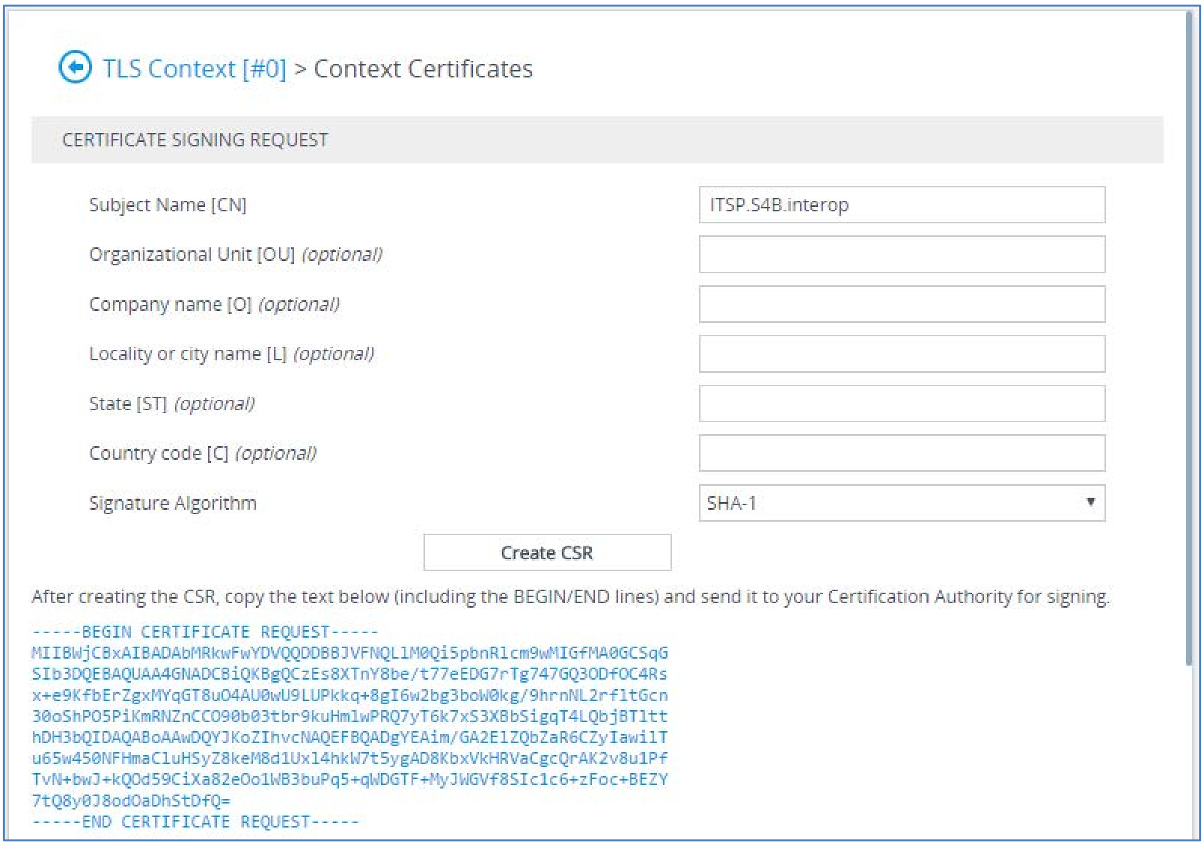
d. Deploying Device and Trusted Root Certificates on E-SBC.

> **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:**

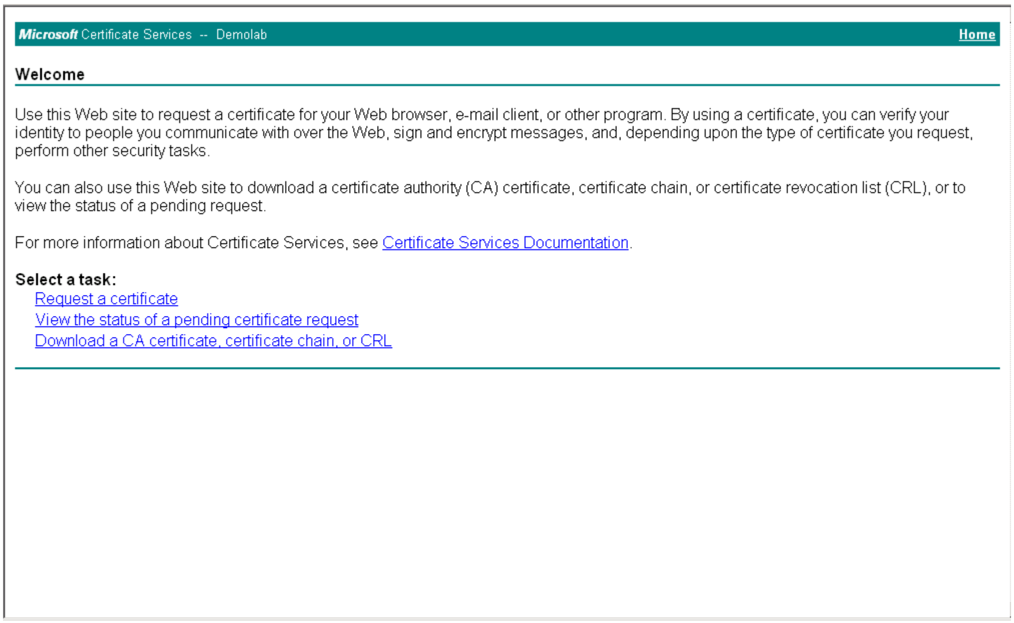
1. Open the TLS Contexts page (**Setup** menu > **IP Network** tab > **Security** folder > **TLS Contexts**).
2. In the TLS Contexts page, select the required TLS Context index row, and then click the **Change Certificate** link located below the table; the Context Certificates page appears.
3. Under the **Certificate Signing Request** group, do the following:
   a. In the 'Subject Name [CN]' field, enter the E-SBC FQDN name (e.g., **ITSP.S4B.interop**).
   b. Fill in the rest of the request fields according to your security provider's instructions.
   c. Click the **Create CSR** button; a textual certificate signing request is displayed in the area below the button:

Figure 4-24: Certificate Signing Request – Creating CSR



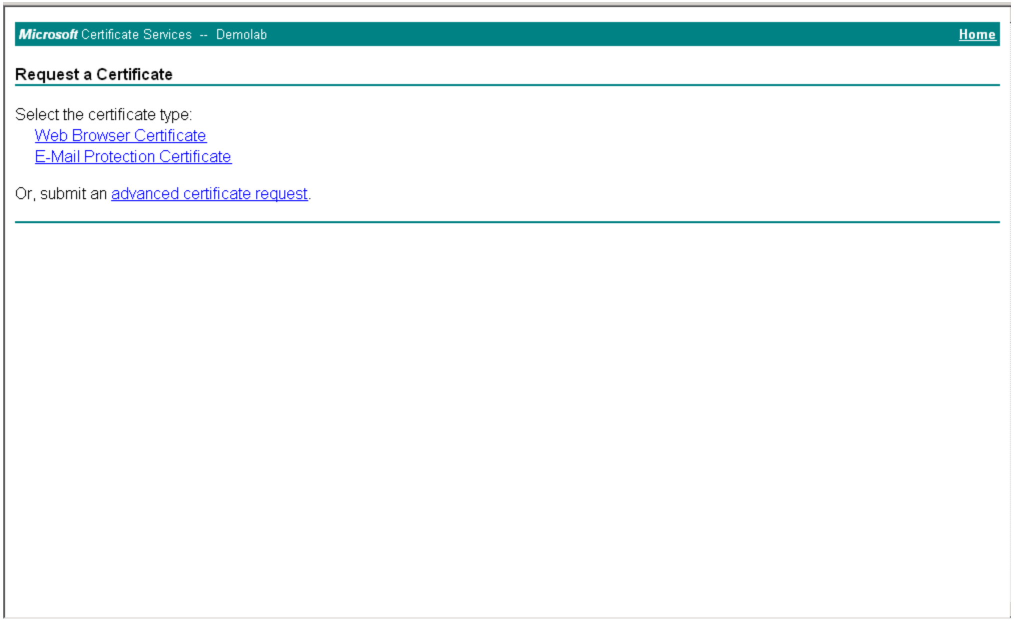
4. Copy the CSR from the line **"----BEGIN CERTIFICATE" to "END CERTIFICATE REQUEST----**" to a text file (such as Notepad), and then save it to a folder on your computer with the file name, *certreq.txt*.
5. Open a Web browser and navigate to the Microsoft Certificates Services Web site at http://<certificate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

Figure 4-25: Microsoft Certificate Services Web Page



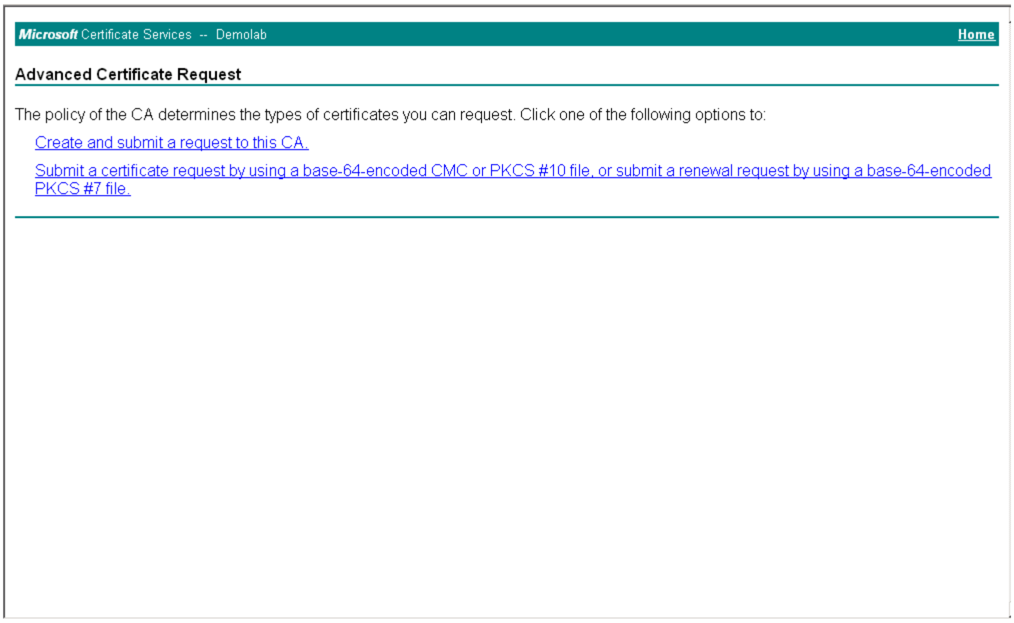
6. Click **Request a certificate**.

Figure 4-26: Request a Certificate Page



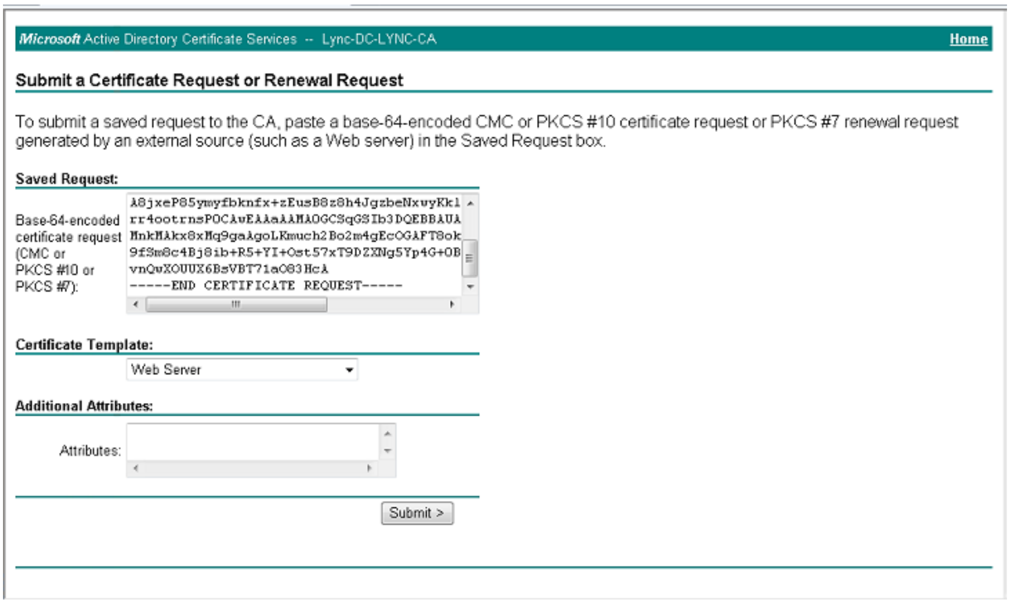
7. Click **advanced certificate request**, and then click **Next**.

Figure 4-27: Advanced Certificate Request Page



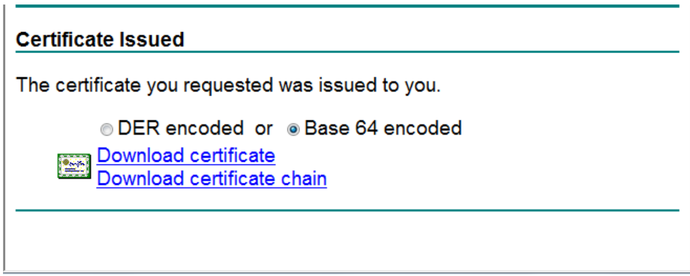
8. Click **Submit a certificate request ...**, and then click **Next**.

Figure 4-28: Submit a Certificate Request or Renewal Request Page



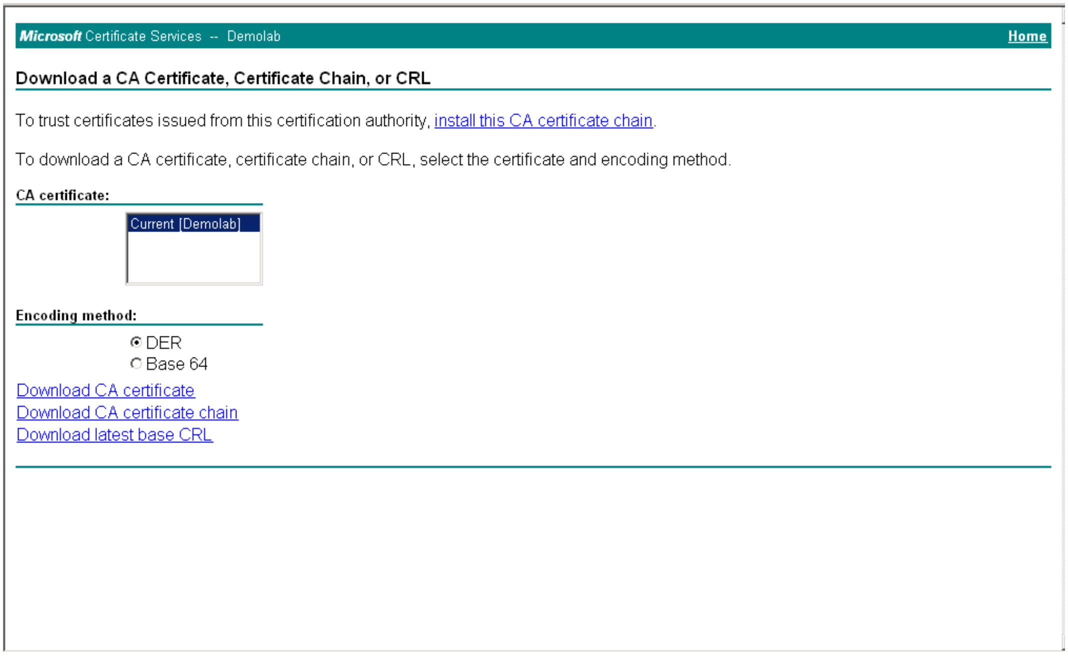
9. Open the *certreq.txt* file that you created and saved in Step 4, and then copy its contents to the 'Saved Request' field.
10. From the 'Certificate Template' drop-down list, select **Web Server**.
11. Click **Submit**.

Figure 4-29: Certificate Issued Page



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

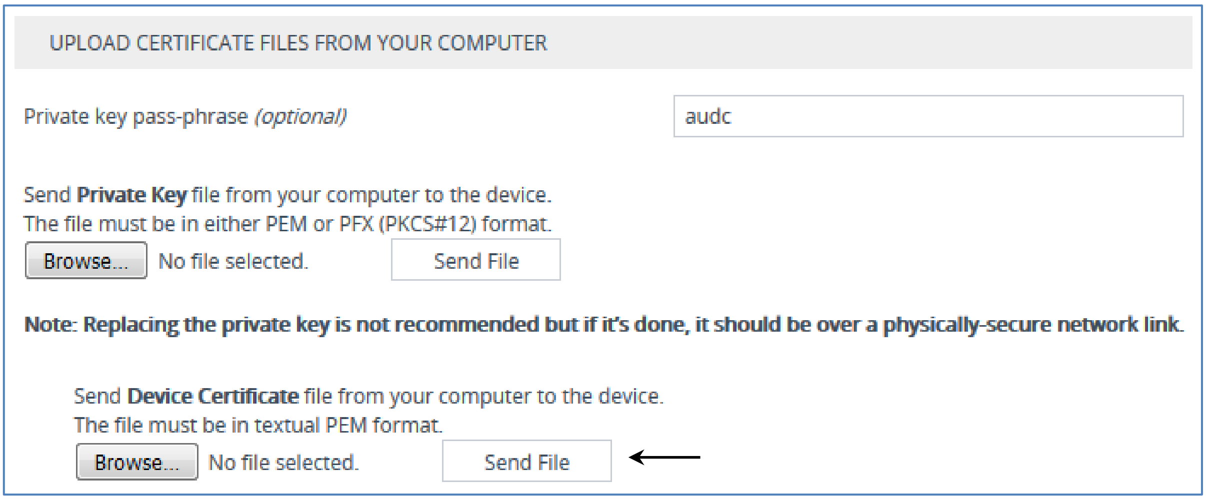
Figure 4-30: Download a CA Certificate, Certificate Chain, or CRL Page



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

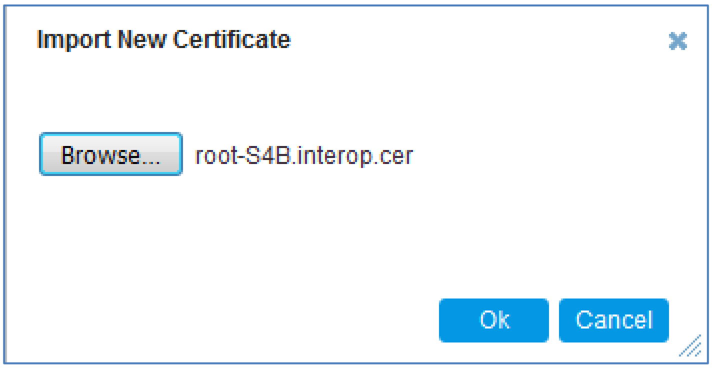
19. In the E-SBC's Web interface, return to the **TLS Contexts** page and do the following:
    a. In the TLS Contexts page, select the required TLS Context index row, and then click the **Change Certificate** link located below the table; the Context Certificates page appears.
    b. Scroll down to the **Upload certificates files from your computer** group, click the **Browse** button corresponding to the **'Send Device Certificate**...' field, navigate to the *gateway.cer* certificate file that you saved on your computer in Step 13, and then click **Send File** to upload the certificate to the E-SBC.

Figure 4-31: Upload Device Certificate Files from your Computer Group



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

Figure 4-32: 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 4.16 on page 89).

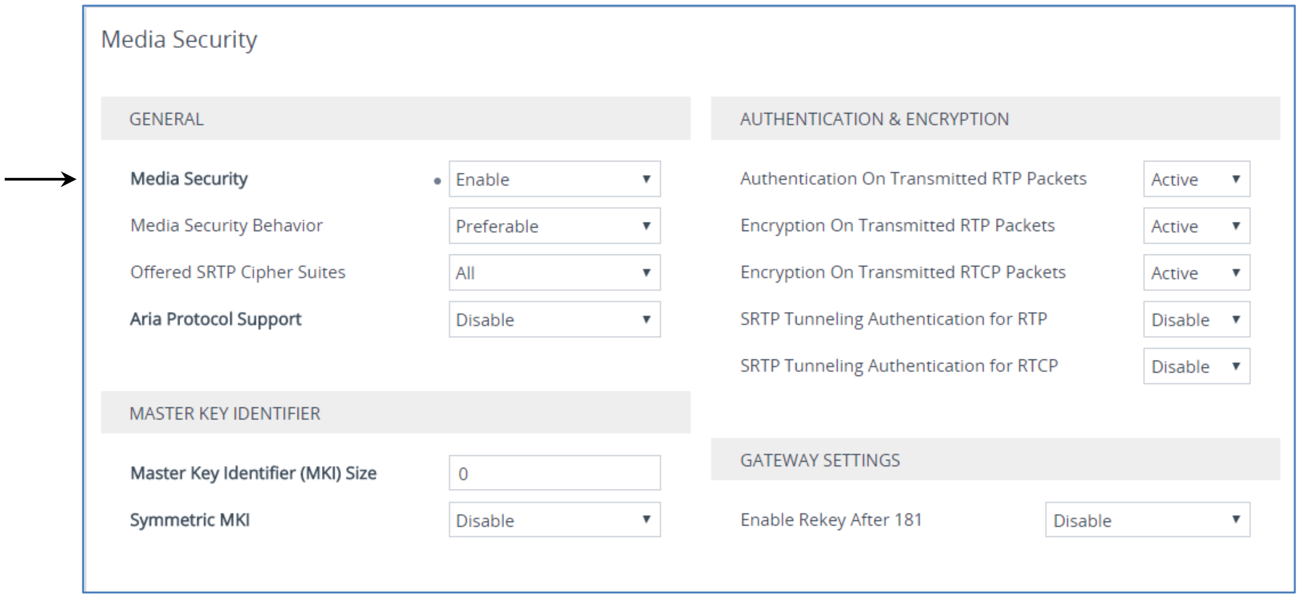
4.9 Step 9: Configure SRTP

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 47).

➢ **To configure media security:**

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

Figure 4-33: Configuring SRTP



2. From the 'Media Security' drop-down list, select **Enable** to enable SRTP.
3. Click **Apply**.
4. Reset the E-SBC with a burn to flash for your settings to take effect (see Section 4.16 on page 89).

4.10 Step 10: Configure Maximum IP Media Channels

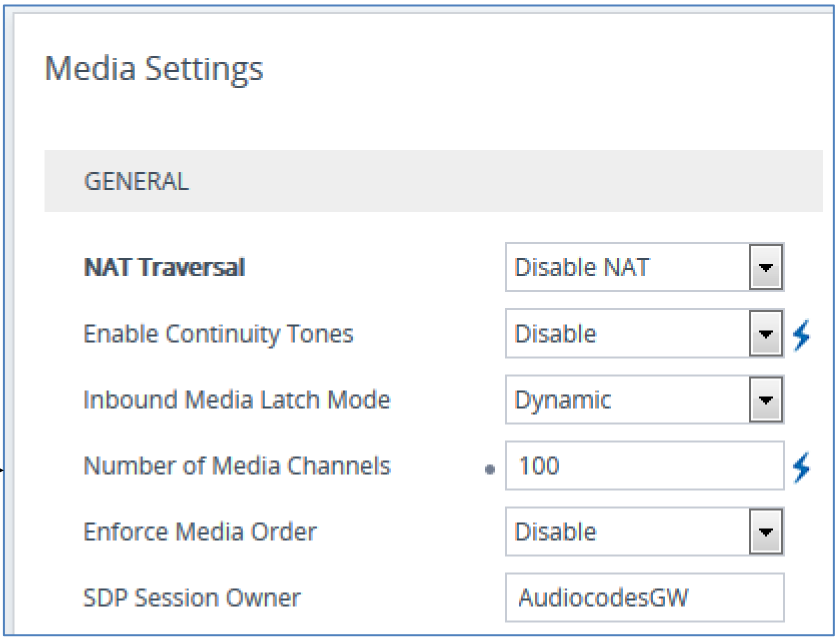
This step describes how to configure the maximum number of required IP media channels. The number of media channels represents the number of DSP channels that the E-SBC allocates to call sessions.

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

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

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

Figure 4-34: Configuring Number of Media Channels



2. In the 'Number of Media Channels' field, enter the number of media channels according to your environments transcoding calls (e.g., 100).
3. Click **Apply**.
4. Reset the E-SBC with a burn to flash for your settings to take effect (see Section 4.16 on page 89).

4.11 Step 11: Configure IP-to-IP Call Routing Rules

This step describes how to configure IP-to-IP call routing rules. These rules define the routes for forwarding SIP messages (e.g., INVITE) received from one IP entity to another. The E-SBC selects the rule whose configured input characteristics (e.g., IP Group) match those of the incoming SIP message. If the input characteristics do not match the first rule in the table, they are compared to the second rule, and so on, until a matching rule is located. If no rule is matched, the message is rejected. The routing rules use the configured IP Groups (as configured in Section 4.6 on page46) to denote the source and destination of the call.

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

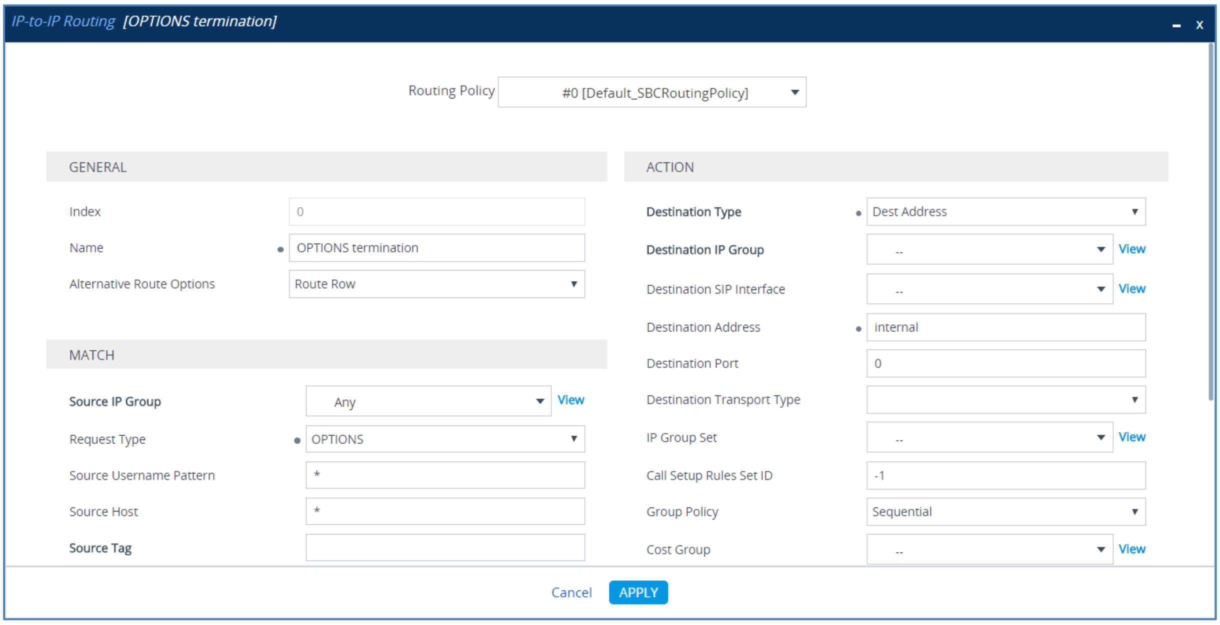
- Terminate SIP OPTIONS messages on the E-SBC that are received from the both LAN and WAN interfaces.
- Calls from Skype for Business Server 2015 to Virgin Media SIP Trunk A
- Calls from Skype for Business Server 2015 to Virgin Media SIP Trunk B as an alternative route if SIP Trunk A fails
- Calls from Fax / Minicom supporting ATA device to Virgin Media SIP Trunk A (if required)
- Calls from Fax / Minicom supporting ATA device to Virgin Media SIP Trunk B as an alternative route if SIP Trunk A fails (if required)
- Calls from Virgin Media SIP Trunk A or B to Fax / Minicom supporting ATA device (if required)
- Calls from Virgin Media SIP Trunk A or B to Skype for Business Server 2015

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

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

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

Figure 4-35: Configuring IP-to-IP Routing Rule for Terminating SIP OPTIONS



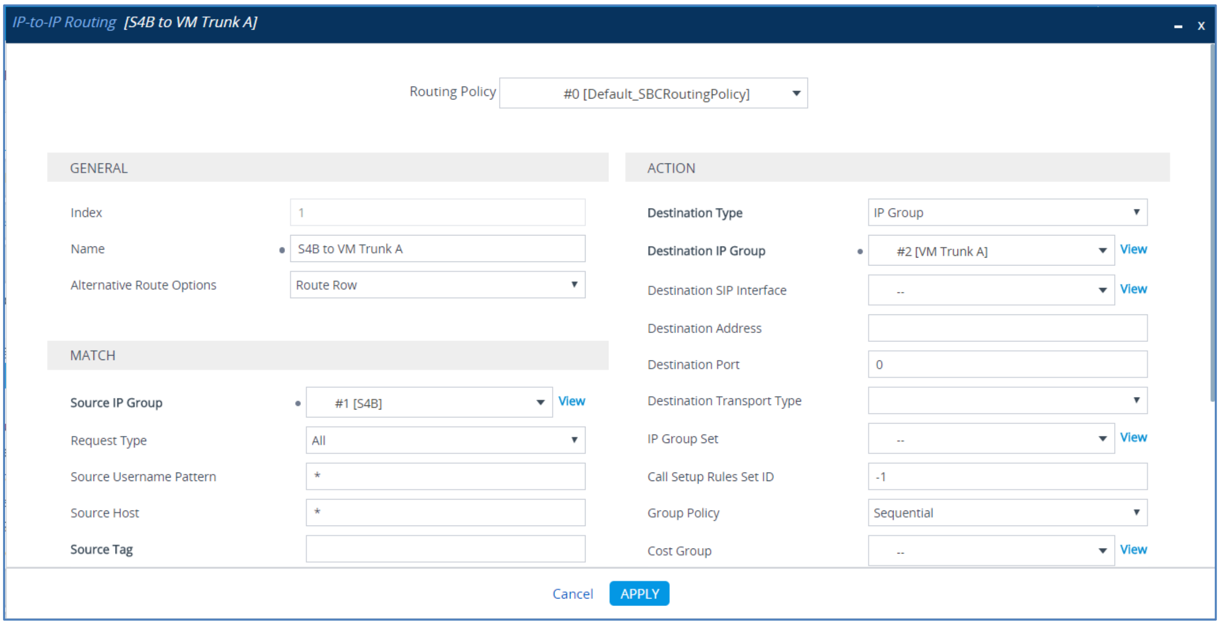
   b. Click **Apply**.

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

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

Parameter	Value
Index	**1**
Route Name	**S4B to VM Trunk A** (arbitrary descriptive name)
Source IP Group	**S4B**
Destination Type	**IP Group**
Destination IP Group	**VM Trunk A**

Figure 4-36: Configuring IP-to-IP Routing Rule for S4B to VM Trunk A



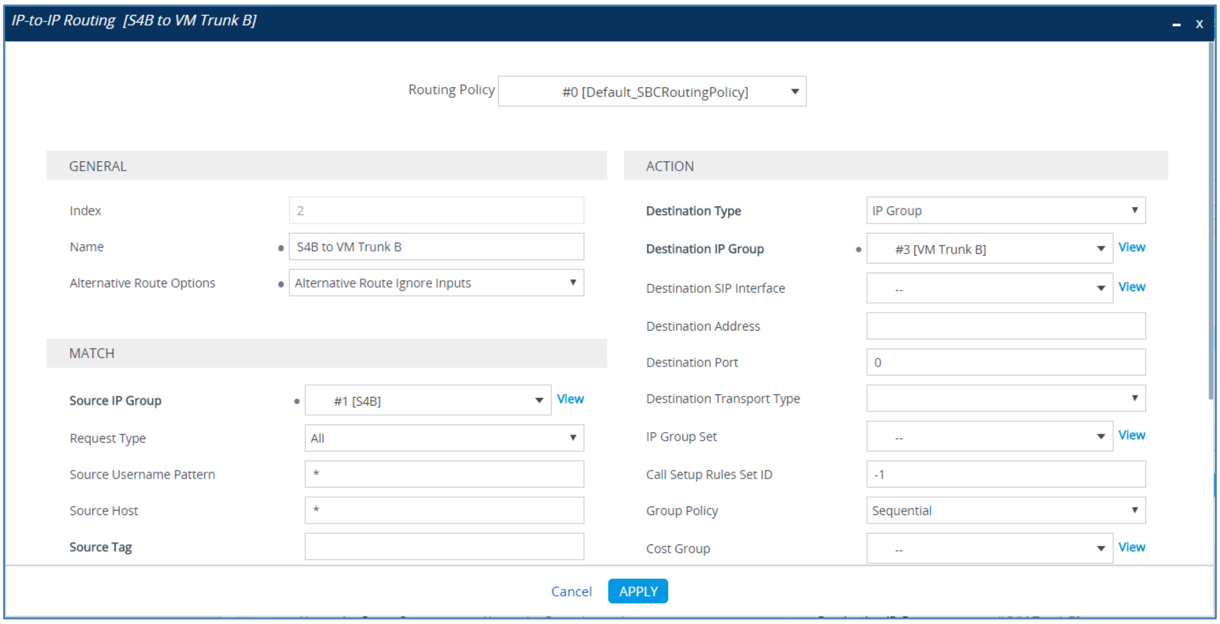
   b. Click **Apply**.

4. Configure a rule to route calls from Skype for Business Server 2015 to Virgin Media SIP Trunk B as alternative routing if the connection with SIP Trunk A fails:

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

Parameter	Value
Index	**2**
Route Name	**S4B to VM Trunk B** (arbitrary descriptive name)
Alternative Route Options	**Alternative Route Ignore Inputs**
Source IP Group	**S4B**
Destination Type	**IP Group**
Destination IP Group	**VM Trunk B**

Figure 4-37: Configuring IP-to-IP Routing Rule for S4B to VM Trunk B



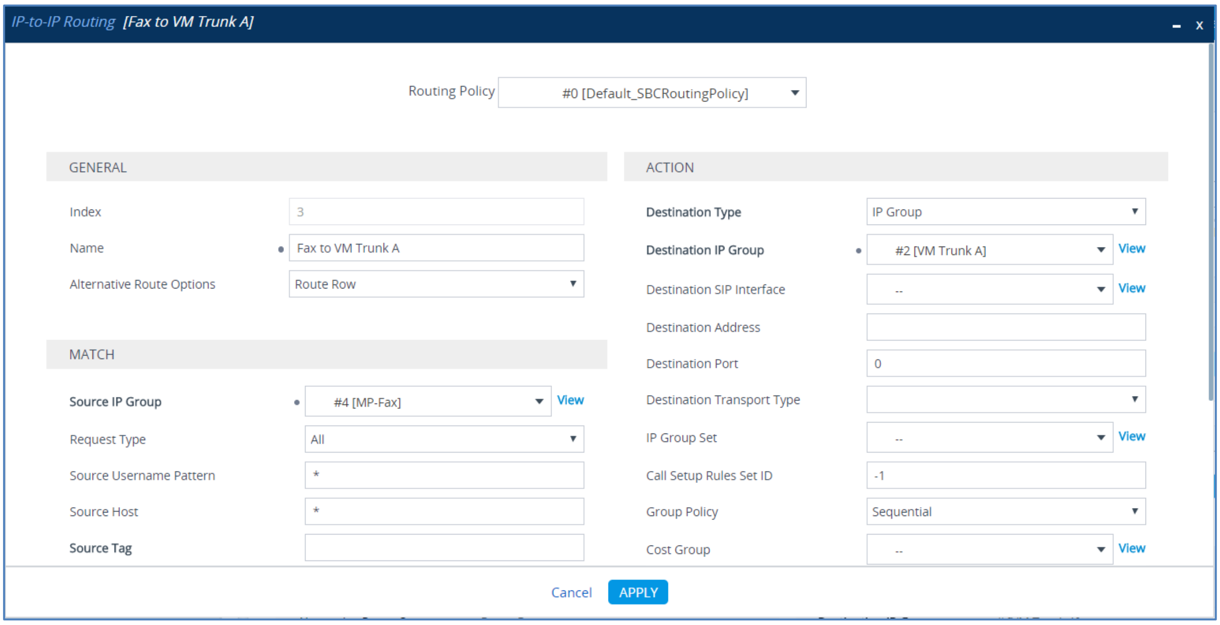
   b. Click **Apply**.

5. Configure a rule to route calls from Fax / Minicom supporting ATA device to Virgin Media SIP Trunk A:

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

Parameter	Value
Index	**3**
Route Name	**Fax to VM Trunk A** (arbitrary descriptive name)
Source IP Group	**MP-Fax**
Destination Type	**IP Group**
Destination IP Group	**VM Trunk A**

Figure 4-38: Configuring IP-to-IP Routing Rule for Fax / Minicom to VM Trunk A



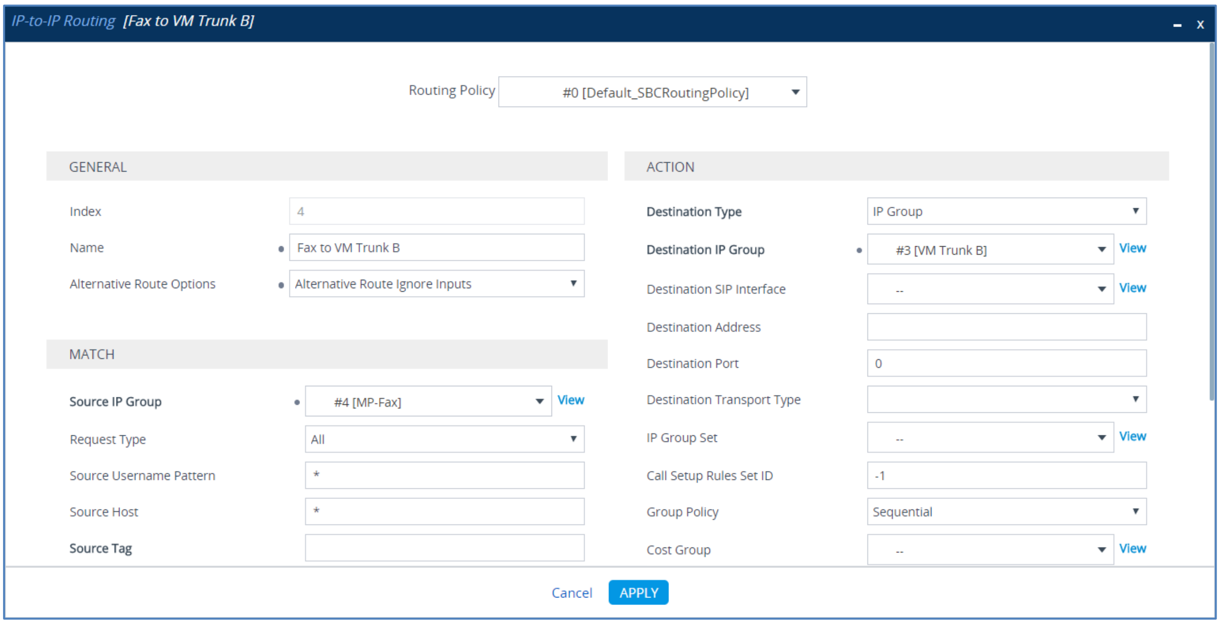
   b. Click **Apply**.

6. Configure a rule to route calls from the Fax / Minicom supporting the ATA device to Virgin Media SIP Trunk B, as alternative routing if the connection with the SIP Trunk A fails:

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

Parameter	Value
Index	**4**
Route Name	**Fax to VM Trunk B** (arbitrary descriptive name)
Alternative Route Options	**Alternative Route Ignore Inputs**
Source IP Group	**MP-Fax**
Destination Type	**IP Group**
Destination IP Group	**VM Trunk B**

Figure 4-39: Configuring IP-to-IP Routing Rule for Fax / Minicom to VM Trunk B



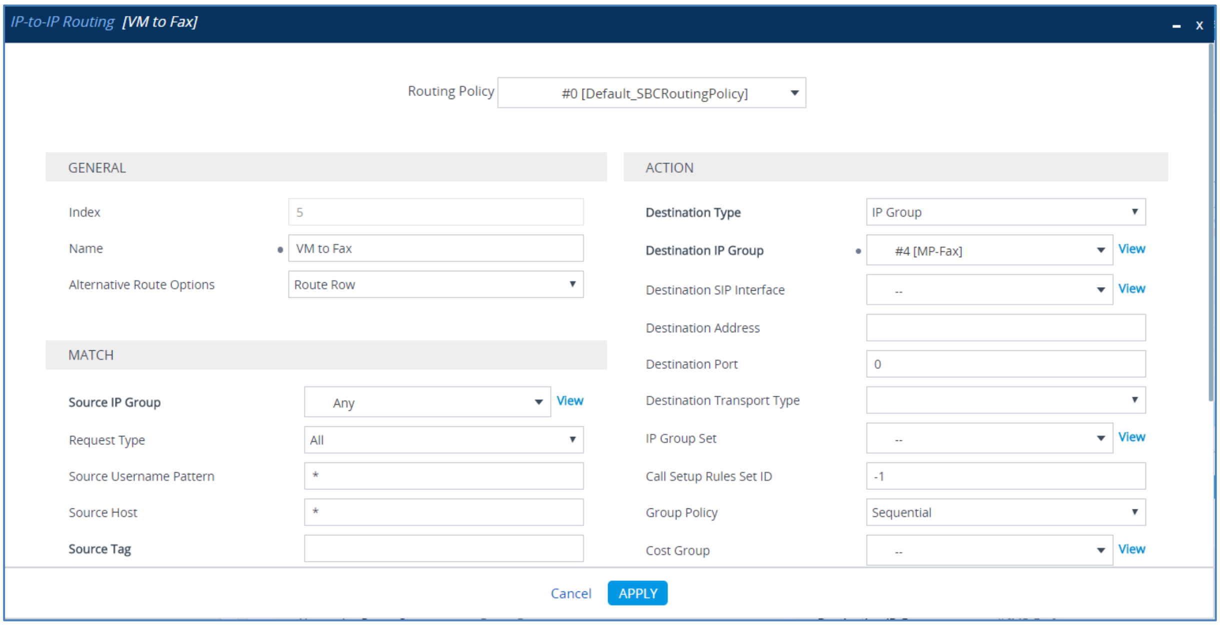
   b. Click **Apply**.

7. Configure a rule to route calls from Virgin Media SIP Trunk A or B to the Fax /Minicom ATA Device:

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

Parameter	Value
Index	**5**
Route Name	**VM to Fax** (arbitrary descriptive name)
Source IP Group	**Any**
Destination Username Prefix	**+441183374147** (dedicated fax line)
Destination Type	**IP Group**
Destination IP Group	**MP-Fax**

Figure 4-40: Configuring IP-to-IP Routing Rule for VM Trunk to Fax / Minicom ATA Device



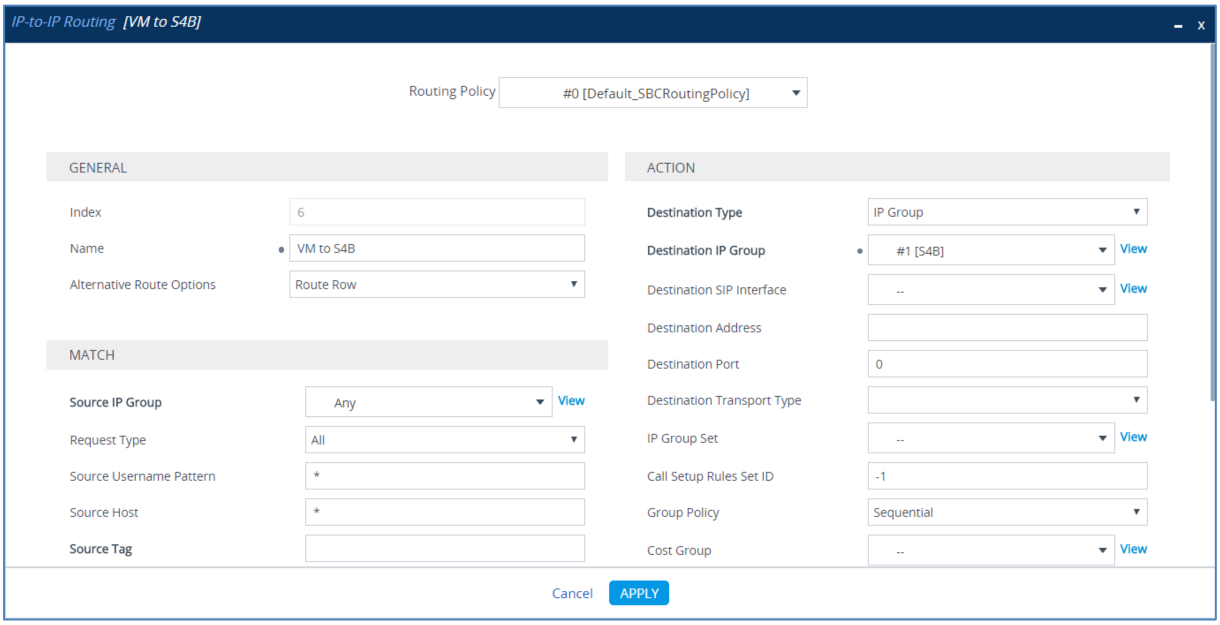
   b. Click **Apply**.

8. Configure a rule to route calls from Virgin Media SIP Trunk A or B to Skype for Business Server 2015:

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

Parameter	Value
Index	**6**
Route Name	**VM to S4B** (arbitrary descriptive name)
Source IP Group	**Any**
Destination Type	**IP Group**
Destination IP Group	**S4B**

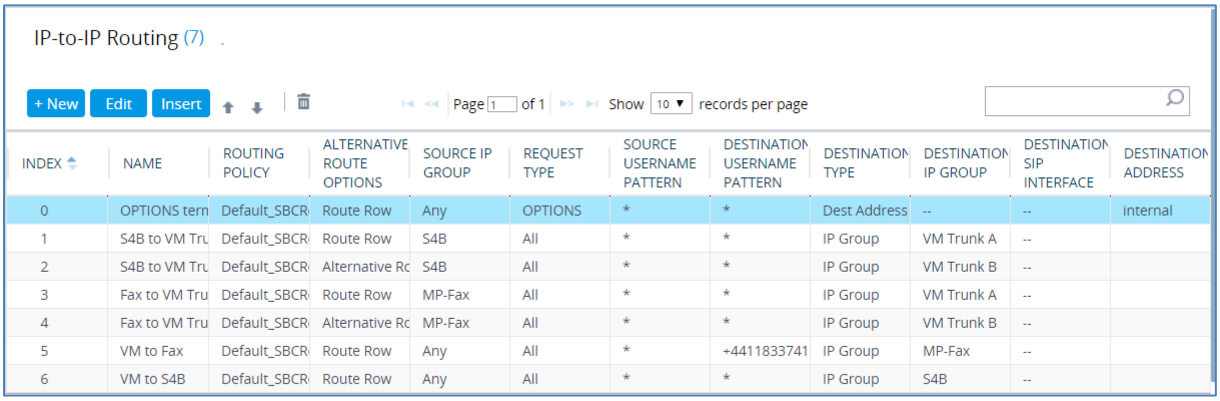
Figure 4-41: Configuring IP-to-IP Routing Rule for VM to S4B



   b. Click **Apply.**

The configured routing rules are shown in the figure below:

Figure 4-42: Configured IP-to-IP Routing Rules in IP-to-IP Routing Table



IP-to-IP Routing (7)

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

INDEX	NAME	ROUTING POLICY	ALTERNATIVE ROUTE OPTIONS	SOURCE IP GROUP	REQUEST TYPE	SOURCE USERNAME PATTERN	DESTINATION USERNAME PATTERN	DESTINATION TYPE	DESTINATION IP GROUP	DESTINATION SIP INTERFACE	DESTINATION ADDRESS
0	OPTIONS tern	Default_SBCR	Route Row	Any	OPTIONS	*	*	Dest Address	--	--	internal
1	S4B to VM Tru	Default_SBCR	Route Row	S4B	All	*	*	IP Group	VM Trunk A	--	
2	S4B to VM Tru	Default_SBCR	Alternative Rc	S4B	All	*	*	IP Group	VM Trunk B	--	
3	Fax to VM Tru	Default_SBCR	Route Row	MP-Fax	All	*	*	IP Group	VM Trunk A	--	
4	Fax to VM Tru	Default_SBCR	Alternative Rc	MP-Fax	All	*	*	IP Group	VM Trunk B	--	
5	VM to Fax	Default_SBCR	Route Row	Any	All	*	+4411833741	IP Group	MP-Fax	--	
6	VM to S4B	Default_SBCR	Route Row	Any	All	*	*	IP Group	S4B	--	

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

4.12 Step 12: Configure IP-to-IP Manipulation Rules

This step describes how to configure IP-to-IP manipulation rules. These rules manipulate the SIP Request-URI user part (source or destination number). The manipulation rules use the configured IP Groups (as configured in Section 4.6 on page 46) to denote the source and destination of the call.

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

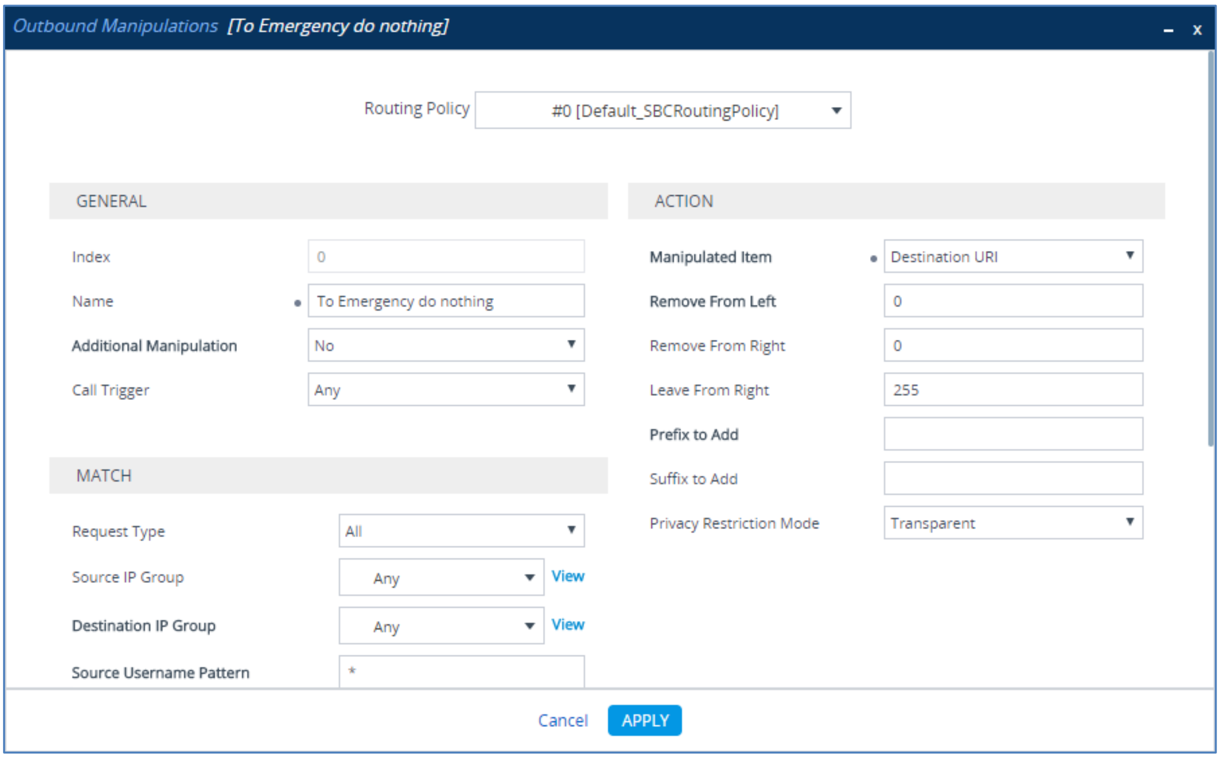
For example, for this interoperability test topology, a manipulation is configured to strip the "+" (plus sign) from the destination number for Emergency calls to the Virgin Media SIP Trunk IP Group if the plus sign exists and to not perform any action for all other emergency calls.

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

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

Parameter	Value
Index	**0**
Name	**To Emergency do nothing**
Source IP Group	**Any**
Destination IP Group	**Any**
Destination Username Pattern	**[999,112,18000]**
Manipulated Item	**Destination URI**

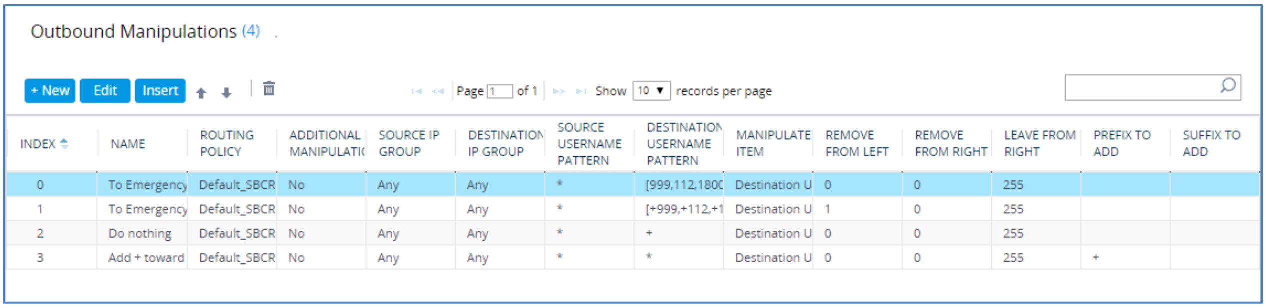
Figure 4-43: Configuring IP-to-IP Outbound Manipulation Rule



3. Click **Apply**.

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

Figure 4-44: Example of Configured IP-to-IP Outbound Manipulation Rules



Rule Index	Description
0	Calls from any (S4B or MP Fax) IP Group with destination number 999 or 112 or 18000, do not perform any action for the destination number.
1	Calls from any (S4B or MP Fax) IP Group with destination number +999 or +112 or +18000. Remove "+" from this numbers.
2	Calls from any (S4B or MP Fax) IP Group with the prefix destination number "+", do not perform any action for the destination number.
3	Calls from any (S4B or MP Fax) IP Group with any destination number (*), add "+" prefix to the destination number.

4.13 Step 13: Configure Message Manipulation Rules

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

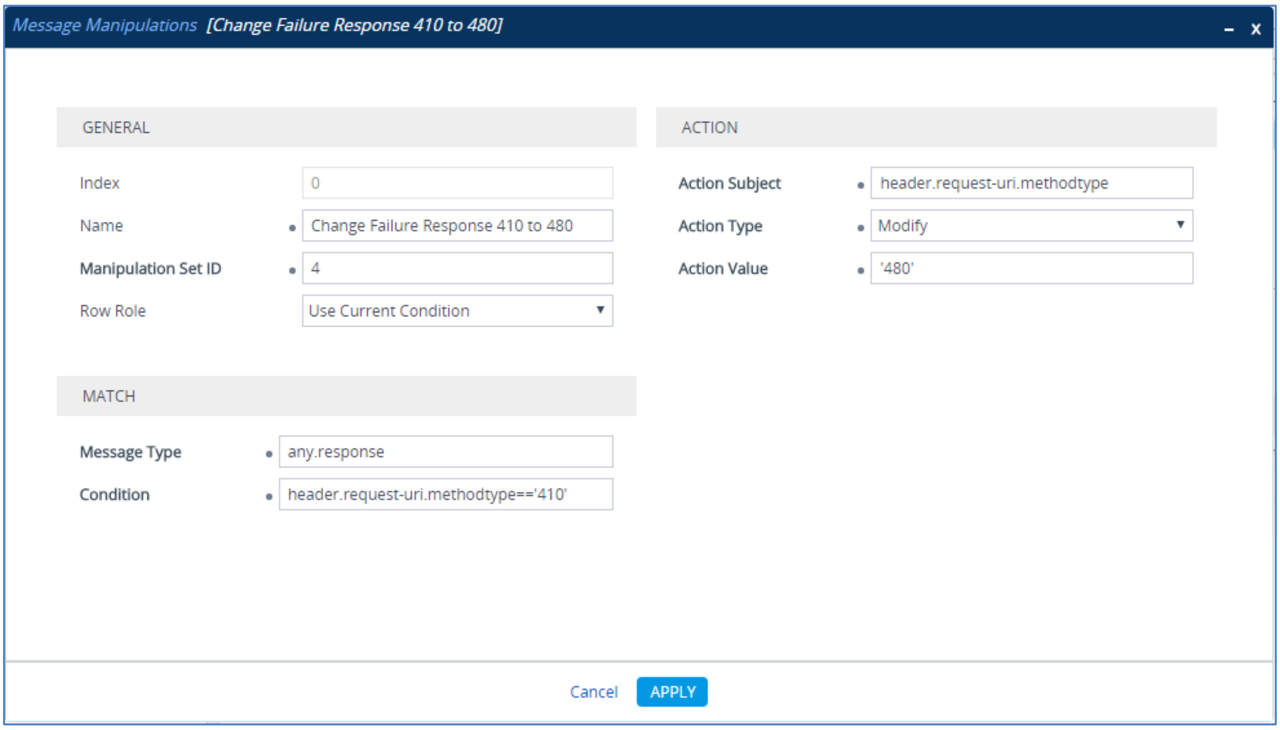
Once you have configured the SIP message manipulation rules, you need to assign them to the relevant IP Group (in the IP Group table) and determine whether they must be applied to inbound or outbound messages.

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

1. Open the Message Manipulations page (**Setup** menu > **Signaling & Media** tab > **Message Manipulation** folder > **Message Manipulations**).
2. Configure a new manipulation rule (Manipulation Set 4) for Virgin Media SIP Trunk. This rule applies to response messages sent to Virgin Media SIP Trunk A or B and consist method type '410 Gone'. This replaces the method type '410' with the value '480', according to Virgin Media request.

Parameter	Value
Index	**0**
Name	**Change Failure Response 410 to 480**
Manipulation Set ID	**4**
Message Type	**any.response**
Condition	**header.request-uri.methodtype=='410'**
Action Subject	**header.request-uri.methodtype**
Action Type	**Modify**
Action Value	**'480'**

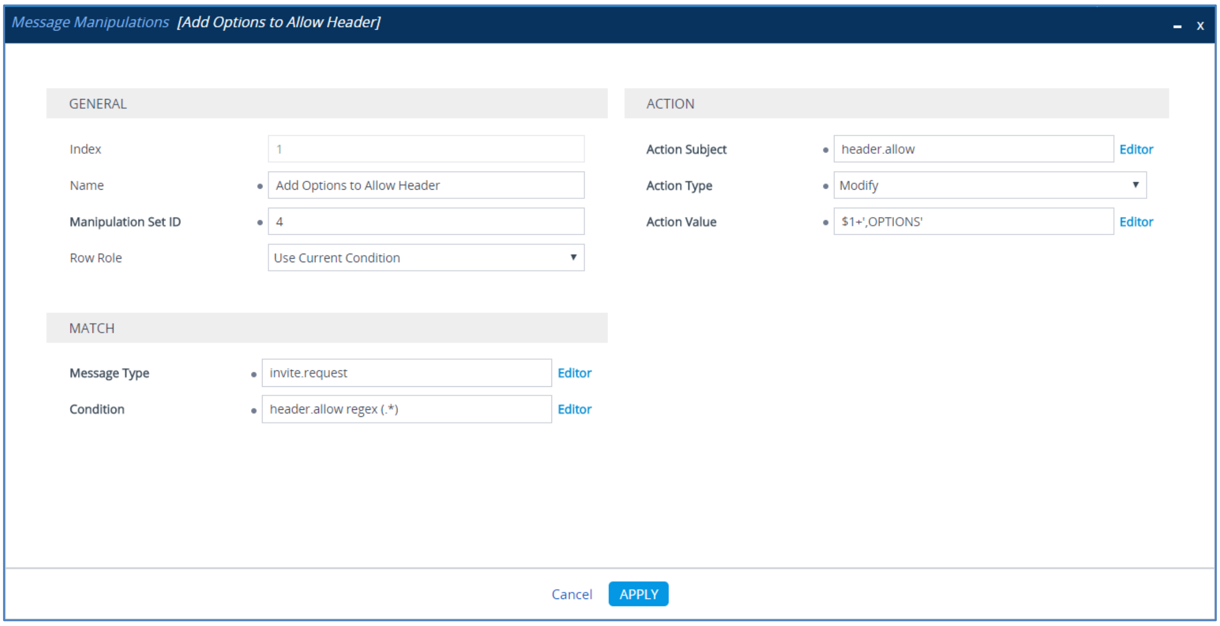
Figure 4-45: Configuring SIP Message Manipulation Rule 0 (for Virgin Media SIP Trunk)



3. Configure another manipulation rule (Manipulation Set 4) for Virgin Media SIP Trunk. This rule is applied on SIP INVITE request messages sent to the Virgin Media SIP Trunk IP Group. This add OPTIONS method to the SIP Allow header.

Parameter	Value
Index	**1**
Name	**Add Options to Allow Header**
Manipulation Set ID	**4**
Message Type	**invite.request**
Condition	**header.allow regex (.*)**
Action Subject	**header.allow regex**
Action Type	**Modify**
Action Value	**$1+',OPTIONS'**

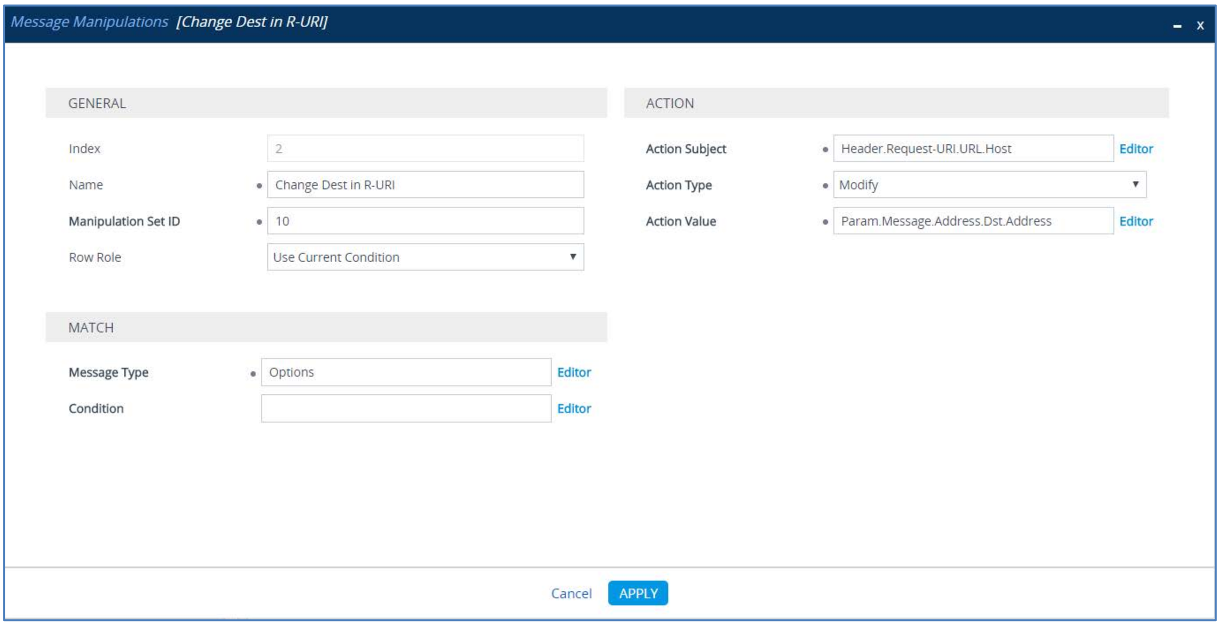
Figure 4-46: Configuring SIP Message Manipulation Rule 1 (for Virgin Media SIP Trunk)



4. Configure another manipulation rule (Manipulation Set 10) for Virgin Media SIP Trunk. This rule is applied on SIP OPTIONS messages sent to the Virgin Media SIP Trunk IP Group. This replaces the host part of the SIP Request-Uri header with the destination address.

Parameter	Value
Index	**2**
Name	**Change Dest in R-URI**
Manipulation Set ID	**10**
Message Type	**Options**
Action Subject	**Header.Request-URI.URL.Host**
Action Type	**Modify**
Action Value	**Param.Message.Address.Dst.Address**

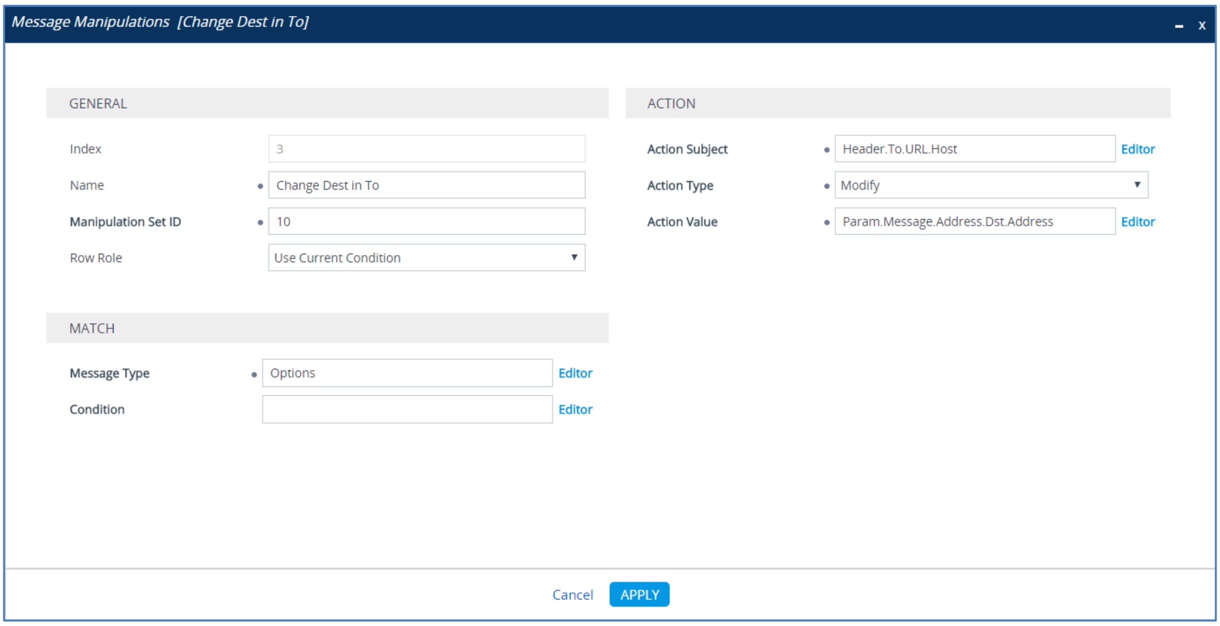
Figure 4-47: Configuring SIP Message Manipulation Rule 2 (for Virgin Media SIP Trunk)



5. Configure another manipulation rule (Manipulation Set 10) for Virgin Media SIP Trunk. This rule is applied on SIP OPTIONS messages sent to the Virgin Media SIP Trunk IP Group. This replaces the host part of the SIP To header with the destination address.

Parameter	Value
Index	**3**
Name	**Change Dest in To**
Manipulation Set ID	**10**
Message Type	**Options**
Action Subject	**Header.To.URL.Host**
Action Type	**Modify**
Action Value	**Param.Message.Address.Dst.Address**

Figure 4-48: Configuring SIP Message Manipulation Rule 3 (for Virgin Media SIP Trunk)



6. Configure another manipulation rule (Manipulation Set 4) for Virgin Media SIP Trunk. This rule is applied on all SIP request messages with SIP P-Asserted-Identity header, sent to the Virgin Media SIP Trunk IP Group. This replaces the user part of the SIP P-Asserted-Identity header with the pre-defined 'pilot' number.

Note: Adapt the pre-defined 'pilot' number according to your environment dial plan.

Parameter	Value
Index	**4**
Name	**For test 27**
Manipulation Set ID	**4**
Message Type	**Any.Request**
Condition	**Header.P-Asserted-Identity exists**
Action Subject	**Header.P-Asserted-Identity.URL.User**
Action Type	**Modify**
Action Value	**'+441183374142'**

Figure 4-49: Configuring SIP Message Manipulation Rule 4 (for Virgin Media SIP Trunk)

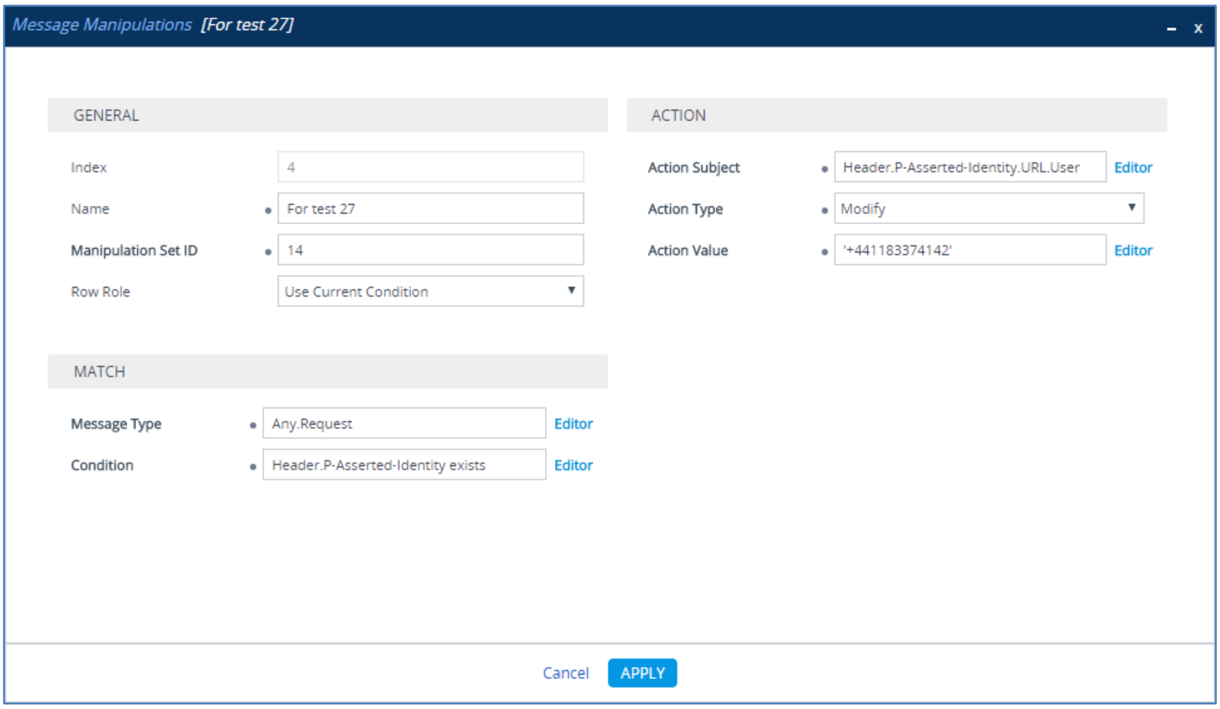
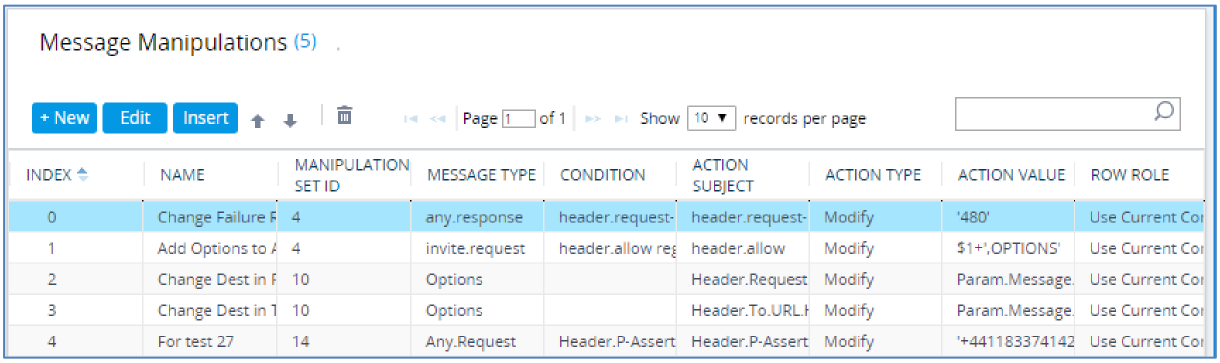


Figure 4-50: Example of Configured SIP Message Manipulation Rules


Message Manipulations (5)

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

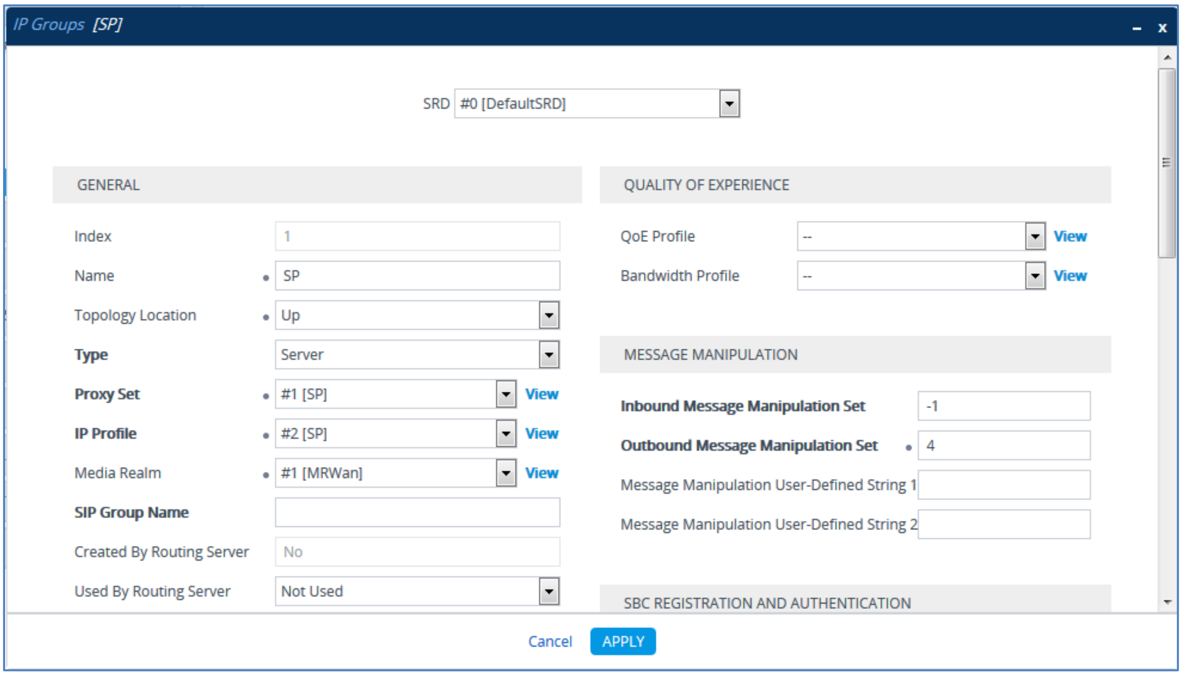
INDEX	NAME	MANIPULATION SET ID	MESSAGE TYPE	CONDITION	ACTION SUBJECT	ACTION TYPE	ACTION VALUE	ROW ROLE
0	Change Failure F	4	any.response	header.request-	header.request-	Modify	'480'	Use Current Coi
1	Add Options to /	4	invite.request	header.allow reg	header.allow	Modify	$1+',OPTIONS'	Use Current Coi
2	Change Dest in F	10	Options		Header.Request	Modify	Param.Message.	Use Current Coi
3	Change Dest in 1	10	Options		Header.To.URL.H	Modify	Param.Message.	Use Current Coi
4	For test 27	14	Any.Request	Header.P-Assert	Header.P-Assert	Modify	'+441183374142	Use Current Coi

The table below includes a SIP message manipulation rule which is executed for messages sent to Virgin Media SIP Trunk IP Groups. This rule is specifically required to enable proper interworking between Virgin Media SIP Trunk and Skype for Business Server 2015. Refer to the *User's Manual* for further details regarding the full capabilities of header manipulation.

Rule Index	Rule Description	Reason for Introducing Rule
0	This rule applies to response messages sent to Virgin Media SIP Trunk A or B and consist method type '410 Gone'. This replaces the method type '410' with the value '480', according to the Virgin Media request.	Virgin Media request to change '410' method type to '480'.
1	This rule is applied on SIP INVITE request messages sent to the Virgin Media SIP Trunk IP Group. This adds the OPTIONS method to the SIP Allow header.	According to Virgin Media request, SIP Allow Header of Invite messages should contain OPTIONS method.
2	This rule is applied on SIP OPTIONS messages sent to the Virgin Media SIP Trunk IP Group. This replaces the host part of the SIP Request-Uri header with the destination address.	Virgin Media request that SIP OPTIONS messages from the SBC sent with destination address of appropriated trunk (Trunk A or Trunk B) in the Request-Uri and To headers.
3	This rule is applied on SIP OPTIONS messages sent to the Virgin Media SIP Trunk IP Group. This replaces the host part of the SIP To header with the destination address.	
4	This rule is applied on all SIP request messages with SIP P-Asserted-Identity header, sent to the Virgin Media SIP Trunk IP Group. This replaces the user part of the SIP P-Asserted-Identity header with the pre-defined 'pilot' number.	According to Virgin Media request, the customer should have the ability to configure a pre-defined number which will be displayed as CLI instead of a real caller number.

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

Figure 4-51: Assigning Manipulation Set 4 to the Virgin Media SIP Trunk IP Group



   d. Click **Apply**.

4.14 Step 14: Configure Registration Accounts

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

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

➢ **To configure a registration account:**

1. Open the Accounts table (**Setup** menu > **Signaling & Media** tab > **SIP Definitions** folder > **Accounts**).
2. Click **New**.
3. Configure the account according to the provided information from Virgin Media, for S4B, serving by VM Trunk A:

Parameter	Value
Served IP Group	**S4B**
Application Type	**SBC**
Serving IP Group	**VM Trunk A**
Register	**No**
Username	As provided by the SIP Trunk provider
Password	As provided by the SIP Trunk provider

4. Click **Apply**.
5. Repeat for S4B, serving by Virgin Media Trunk B:

Parameter	Value
Served IP Group	**S4B**
Application Type	**SBC**
Serving IP Group	**VM Trunk B**
Register	**No**
Username	As provided by the SIP Trunk provider
Password	As provided by the SIP Trunk provider

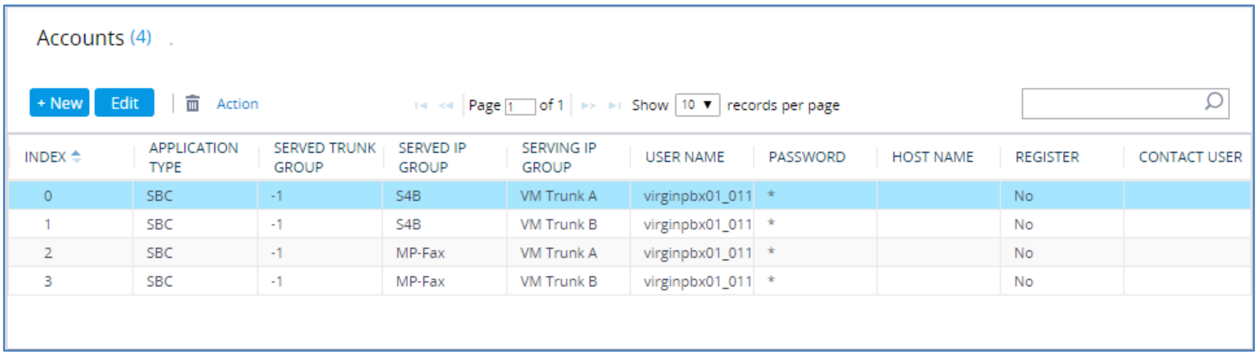
6. Click **Apply**.
7. Repeat the same for Fax / Minicom ATA Device, serving by VM Trunk A:

Parameter	Value
Served IP Group	**MP-Fax**
Application Type	**SBC**
Serving IP Group	**VM Trunk A**
Register	**No**
Username	As provided by the SIP Trunk provider
Password	As provided by the SIP Trunk provider

8. Click **Apply**.
9. Repeat the same for Fax / Minicom ATA Device, serving by VM Trunk B:

Parameter	Value
Served IP Group	**MP-Fax**
Application Type	**SBC**
Serving IP Group	**VM Trunk B**
Register	**No**
Username	As provided by the SIP Trunk provider
Password	As provided by the SIP Trunk provider

Figure 4-52: Configuring a SIP Registration Account


Accounts (4)

+ New Edit Action Page 1 of 1 Show 10 records per page

INDEX	APPLICATION TYPE	SERVED TRUNK GROUP	SERVED IP GROUP	SERVING IP GROUP	USER NAME	PASSWORD	HOST NAME	REGISTER	CONTACT USER
0	SBC	-1	S4B	VM Trunk A	virginpbx01_011	*		No	
1	SBC	-1	S4B	VM Trunk B	virginpbx01_011	*		No	
2	SBC	-1	MP-Fax	VM Trunk A	virginpbx01_011	*		No	
3	SBC	-1	MP-Fax	VM Trunk B	virginpbx01_011	*		No	

10. Click **Apply**.

4.15 Step 15: Miscellaneous Configuration

This section describes miscellaneous E-SBC configuration.

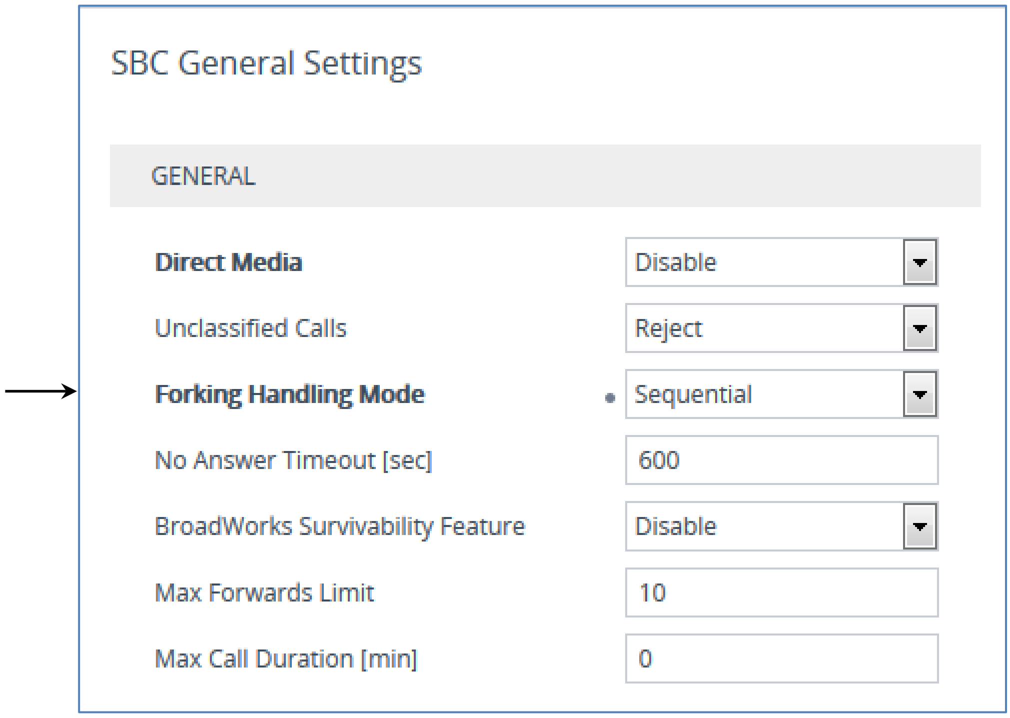
4.15.1 Step 15a: Configure Call Forking Mode

This step describes how to configure the E-SBC's handling of SIP 18x responses received for call forking of INVITE messages. For the interoperability test topology, if a SIP 18x response with SDP is received, the E-SBC opens a voice stream according to the received SDP. The E-SBC re-opens the stream according to subsequently received 18x responses with SDP or plays a ringback tone if a 180 response without SDP is received. It is mandatory to set this field for the Skype for Business Server 2015 environment.

➢ **To configure call forking:**

1. Open the SBC General Settings page (**Setup** menu > **Signaling & Media** tab > **SBC** folder > **SBC General Settings**).
2. From the 'SBC Forking Handling Mode' drop-down list, select **Sequential**.

Figure 4-53: Configuring Forking Mode



3. Click **Apply**.

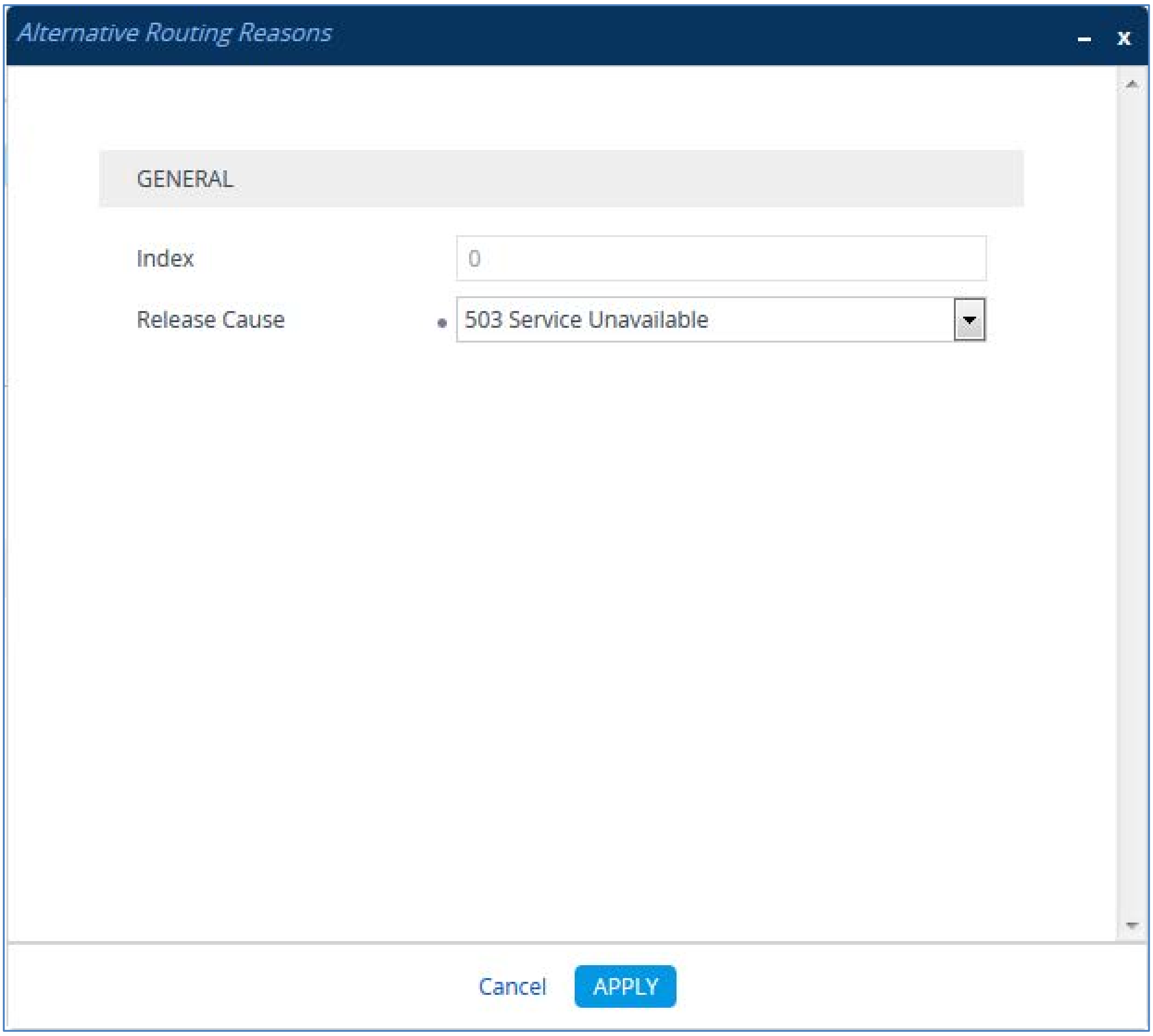
4.15.2 Step 15b: Configure SBC Alternative Routing Reasons

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

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

1. Open the Alternative Routing Reasons table (**Setup** menu > **Signaling & Media** tab > **SBC** folder > **Routing** > **Alternative Reasons**).
2. Click **New**.
3. From the 'Release Cause' drop-down list, select **503 Service Unavailable**.

Figure 4-54: SBC Alternative Routing Reasons Table



4. Click **Apply**.

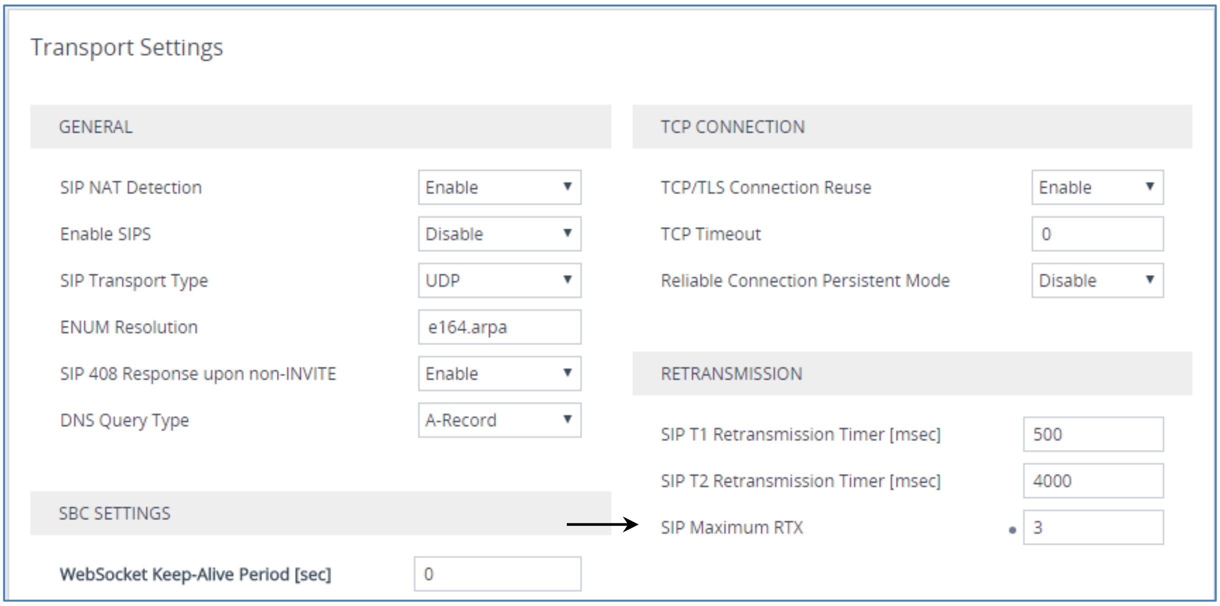
4.15.3 Step 15c: Configure SBC Max Retransmission Time

This step describes how to configure the E-SBC's maximum retransmission attempts. In this case, E-SBC attempts to locate an alternative route for the call after three attempts.

➢ **To configure SIP SBC Max Retransmission Time:**

1. Open the Transport Settings (**Setup** menu > **Signaling & Media** tab > **SIP Definitions** folder > **Transport Settings**).
2. Click **New**.
3. In the 'SIP Maximum RTX' field, enter **3** (retransmission will be stopped after 3 attempts).

Figure 4-55: SBC Max Retransmission Time



4. Click **Apply**.

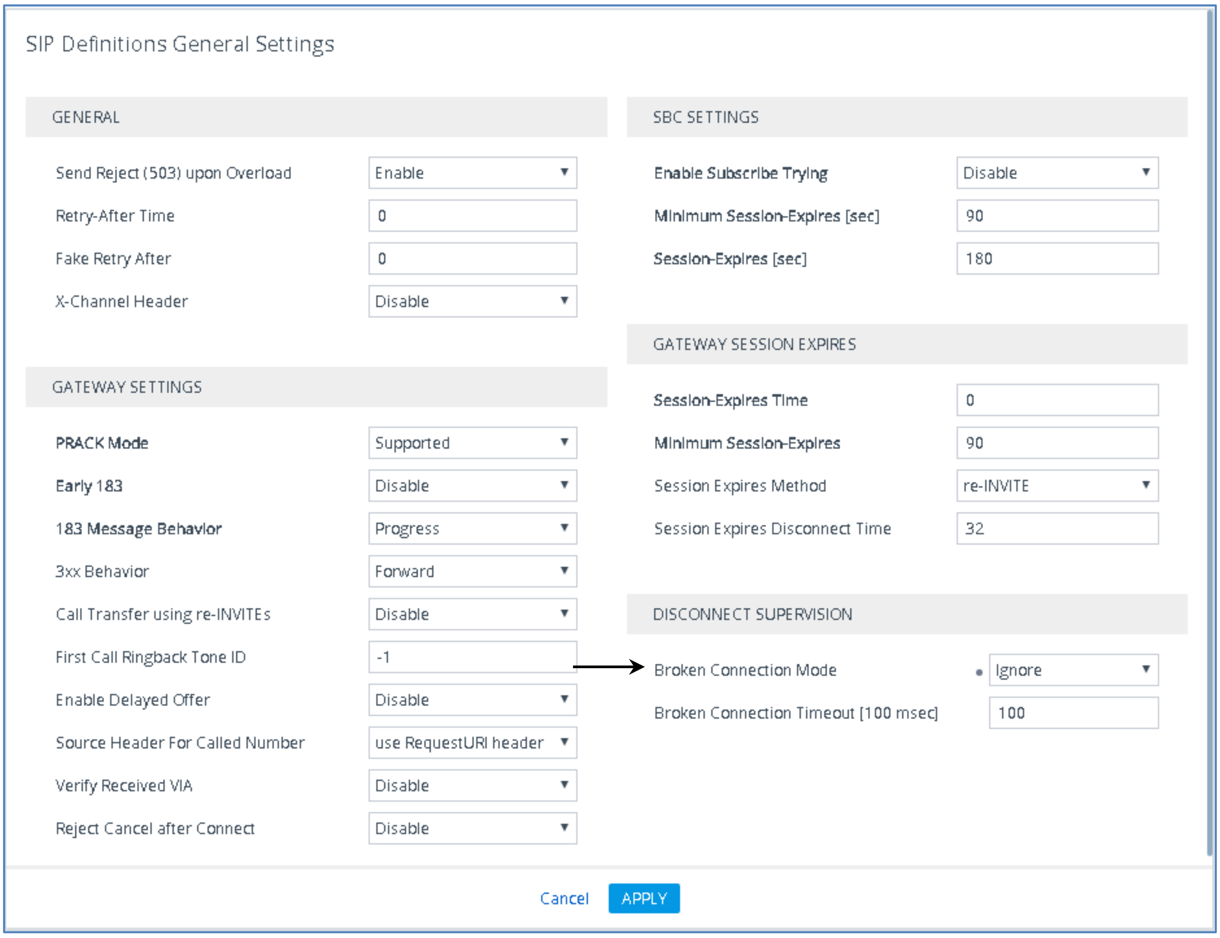
4.15.4 Step 15d: Configure Broken Connection Behavior

This step describes how to configure the E-SBC to ignore broken connection. This is needed for proper behavior during fax transmission.

➢ **To configure SIP Broken Connection Behavior:**

1. Open the SIP Definitions General Settings (**Setup** menu > **Signaling & Media** tab > **SIP Definitions** folder > **SIP Definitions General Settings**).
2. Click **New**.
3. From the 'Broken Connection Mode' drop-down list, select **Ignore**.

Figure 4-56: SBC Broken Connection Behavior



4. Click **Apply**.

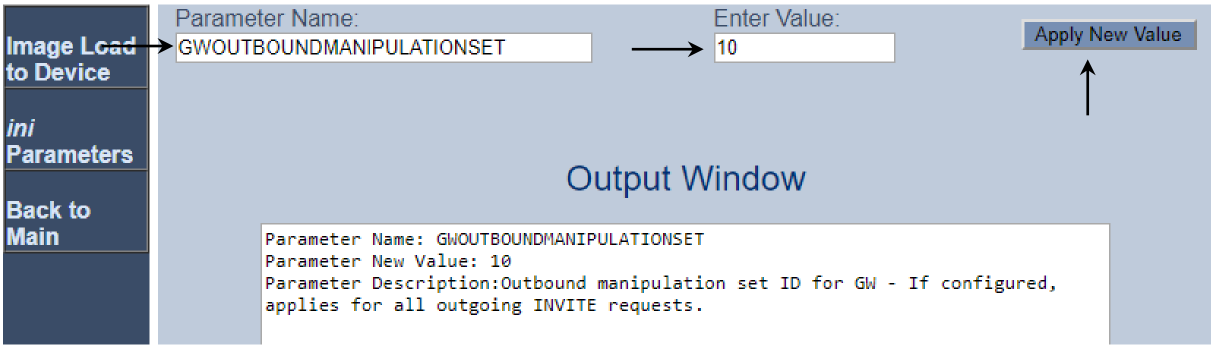
4.15.5 Step 15e: Configuration Needed for Manipulating SIP OPTIONS

This step describes how to configure the E-SBC's string name in SIP OPTIONS Keep-alive messages (host part of the Request-URI and To SIP headers).

- **To configure the Gateway Outbound Manipulation Set:**

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 GW Outbound Manipulation Set via AdminPage



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

Parameter	Value
GWOUTBOUNDMANIPULATIONSET	10

5. Click the **Apply New Value** button for each field.
6. Click on **Back to Main**. On the main page don't forget to save the configuration.

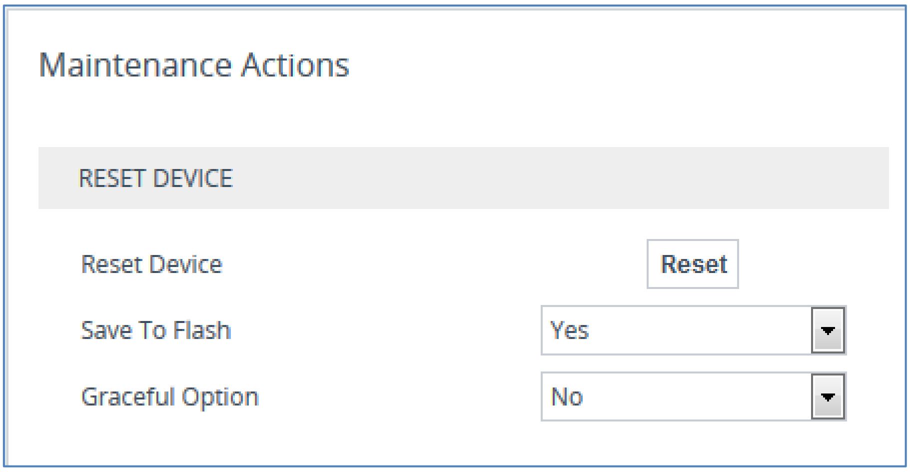
4.16 Step 16: Reset the E-SBC

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

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

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

Figure 4-58: Resetting the E-SBC



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

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

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

[SYSTEM Params]

SyslogServerIP = 10.10.10.10
EnableSyslog = 1
NTPServerUTCOffset = 10800
NTPServerIP = '10.15.27.1'

[BSP Params]

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

[Analog Params]


[ControlProtocols Params]

AdminStateLockControl = 0


[Voice Engine Params]

BrokenConnectionEventTimeout = 1000
ENABLEMEDIASECURITY = 1
CallProgressTonesFilename = 'usa_tones_13.dat'

[WEB Params]

UseProductName = 1
FaviconCurrentVersion = 2

[SIP Params]

MEDIACHANNELS = 100
GWDEBUGLEVEL = 5
SIPMAXRTX = 3
ENABLESBCAPPLICATION = 1
MSLDAPPRIMARYKEY = 'telephoneNumber'
GWOUTBOUNDMANIPULATIONSET = 10
```

```
SBCFORKINGHANDLINGMODE = 1
ENERGYDETECTORCMD = 587202560
ANSWERDETECTORCMD = 10486144

[IPsec Params]


[SNMP Params]


[ DeviceTable ]

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

[ \DeviceTable ]


[ InterfaceTable ]

FORMAT InterfaceTable_Index = InterfaceTable_ApplicationTypes,
InterfaceTable_InterfaceMode, InterfaceTable_IPAddress,
InterfaceTable_PrefixLength, InterfaceTable_Gateway,
InterfaceTable_InterfaceName, InterfaceTable_PrimaryDNSServerIPAddress,
InterfaceTable_SecondaryDNSServerIPAddress,
InterfaceTable_UnderlyingDevice;
InterfaceTable 0 = 6, 10, 10.15.77.10, 16, 10.15.0.1, "LAN_IF",
10.15.27.1, 0.0.0.0, "vlan 1";
InterfaceTable 1 = 5, 10, 195.189.192.154, 25, 195.189.192.129, "WAN_IF",
80.179.52.100, 80.179.55.100, "vlan 2";

[ \InterfaceTable ]


[ WebUsers ]

FORMAT WebUsers_Index = WebUsers_Username, WebUsers_Password,
WebUsers_Status, WebUsers_PwAgeInterval, WebUsers_SessionLimit,
WebUsers_CliSessionLimit, WebUsers_SessionTimeout, WebUsers_BlockTime,
WebUsers_UserLevel, WebUsers_PwNonce, WebUsers_SSHPublicKey;
WebUsers 0 = "Admin",
"$1$bgtdFkgQREJNFRNJHUhDGRtPTuPju+bhteC1ubG4vby9t7fy9fb1oqfyoKmt+KP5/qz9m
ZSTlpyUkpDNzMudz54=", 1, 0, 5, -1, 15, 60, 200,
"e4064f90b5b26631d46fbcdb79f2b7a0", ".fc";
WebUsers 1 = "User",
"$1$Cj46OmhtN3ElJiolcSQnfXh4Ii5+Jn4ZRBQRHR0fHx4bTB9ITE8aVgRQVQUGAAEPXVkCD
w0GWSEgIHN0dHB2LHE=", 1, 0, 5, -1, 15, 60, 50,
"c26a27dd91a886b99de5e81b9a736232", "";

[ \WebUsers ]


[ TLSContexts ]

FORMAT TLSContexts_Index = TLSContexts_Name, TLSContexts_TLSVersion,
TLSContexts_DTLSVersion, TLSContexts_ServerCipherString,
TLSContexts_ClientCipherString, TLSContexts_RequireStrictCert,
```

```
TLSContexts_OcspEnable, TLSContexts_OcspServerPrimary,
TLSContexts_OcspServerSecondary, TLSContexts_OcspServerPort,
TLSContexts_OcspDefaultResponse, TLSContexts_DHKeySize;
TLSContexts 0 = "default", 7, 0, "RC4:EXP", "ALL:!ADH", 0, 0, 0.0.0.0,
0.0.0.0, 2560, 0, 1024;

[ \TLSContexts ]


[ AudioCodersGroups ]

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

[ \AudioCodersGroups ]


[ AllowedAudioCodersGroups ]

FORMAT AllowedAudioCodersGroups_Index = AllowedAudioCodersGroups_Name;
AllowedAudioCodersGroups 0 = "VM";

[ \AllowedAudioCodersGroups ]


[ IpProfile ]

FORMAT IpProfile_Index = IpProfile_ProfileName, IpProfile_IpPreference,
IpProfile_CodersGroupName, IpProfile_IsFaxUsed,
IpProfile_JitterBufMinDelay, IpProfile_JitterBufOptFactor,
IpProfile_IPDiffServ, IpProfile_SigIPDiffServ,
IpProfile_RTPRedundancyDepth, IpProfile_CNGmode,
IpProfile_VxxTransportType, IpProfile_NSEMode, IpProfile_IsDTMFUsed,
IpProfile_PlayRBTone2IP, IpProfile_EnableEarlyMedia,
IpProfile_ProgressIndicator2IP, IpProfile_EnableEchoCanceller,
IpProfile_CopyDest2RedirectNumber, IpProfile_MediaSecurityBehaviour,
IpProfile_CallLimit, IpProfile_DisconnectOnBrokenConnection,
IpProfile_FirstTxDtmfOption, IpProfile_SecondTxDtmfOption,
IpProfile_RxDTMFOption, IpProfile_EnableHold, IpProfile_InputGain,
IpProfile_VoiceVolume, IpProfile_AddIEInSetup,
IpProfile_SBCExtensionCodersGroupName,
IpProfile_MediaIPVersionPreference, IpProfile_TranscodingMode,
IpProfile_SBCAllowedMediaTypes, IpProfile_SBCAllowedAudioCodersGroupName,
IpProfile_SBCAllowedVideoCodersGroupName, IpProfile_SBCAllowedCodersMode,
IpProfile_SBCMediaSecurityBehaviour, IpProfile_SBCRFC2833Behavior,
IpProfile_SBCAlternativeDTMFMethod, IpProfile_SBCSendMultipleDTMFMethods,
IpProfile_SBCAssertIdentity, IpProfile_AMDSensitivityParameterSuit,
IpProfile_AMDSensitivityLevel, IpProfile_AMDMaxGreetingTime,
IpProfile_AMDMaxPostSilenceGreetingTime, IpProfile_SBCDiversionMode,
IpProfile_SBCHistoryInfoMode, IpProfile_EnableQSIGTunneling,
IpProfile_SBCFaxCodersGroupName, IpProfile_SBCFaxBehavior,
IpProfile_SBCFaxOfferMode, IpProfile_SBCFaxAnswerMode,
IpProfile_SbcPrackMode, IpProfile_SBCSessionExpiresMode,
IpProfile_SBCRemoteUpdateSupport, IpProfile_SBCRemoteReinviteSupport,
IpProfile_SBCRemoteDelayedOfferSupport, IpProfile_SBCRemoteReferBehavior,
IpProfile_SBCRemote3xxBehavior, IpProfile_SBCRemoteMultiple18xSupport,
IpProfile_SBCRemoteEarlyMediaResponseType,
IpProfile_SBCRemoteEarlyMediaSupport, IpProfile_EnableSymmetricMKI,
IpProfile_MKISize, IpProfile_SBCEnforceMKISize,
IpProfile_SBCRemoteEarlyMediaRTP, IpProfile_SBCRemoteSupportsRFC3960,
IpProfile_SBCRemoteCanPlayRingback, IpProfile_EnableEarly183,
IpProfile_EarlyAnswerTimeout, IpProfile_SBC2833DTMFPayloadType,
IpProfile_SBCUserRegistrationTime, IpProfile_ResetSRTPStateUponRekey,
```

```
IpProfile_AmdMode, IpProfile_SBCReliableHeldToneSource,
IpProfile_GenerateSRTPKeys, IpProfile_SBCPlayHeldTone,
IpProfile_SBCRemoteHoldFormat, IpProfile_SBCRemoteReplacesBehavior,
IpProfile_SBCSDPPtimeAnswer, IpProfile_SBCPreferredPTime,
IpProfile_SBCUseSilenceSupp, IpProfile_SBCRTPRedundancyBehavior,
IpProfile_SBCPlayRBTToTransferee, IpProfile_SBCRTCPMode,
IpProfile_SBCJitterCompensation,
IpProfile_SBCRemoteRenegotiateOnFaxDetection,
IpProfile_JitterBufMaxDelay,
IpProfile_SBCUserBehindUdpNATRegistrationTime,
IpProfile_SBCUserBehindTcpNATRegistrationTime,
IpProfile_SBCSDPHandleRTCPAttribute,
IpProfile_SBCRemoveCryptoLifetimeInSDP, IpProfile_SBCIceMode,
IpProfile_SBCRTCPMux, IpProfile_SBCMediaSecurityMethod,
IpProfile_SBCHandleXDetect, IpProfile_SBCRTCPFeedback,
IpProfile_SBCRemoteRepresentationMode, IpProfile_SBCKeepVIAHeaders,
IpProfile_SBCKeepRoutingHeaders, IpProfile_SBCKeepUserAgentHeader,
IpProfile_SBCRemoteMultipleEarlyDialogs,
IpProfile_SBCRemoteMultipleAnswersMode, IpProfile_SBCDirectMediaTag,
IpProfile_SBCAdaptRFC2833BWToVoiceCoderBW,
IpProfile_CreatedByRoutingServer, IpProfile_SBCFaxReroutingMode,
IpProfile_SBCMaxCallDuration, IpProfile_SBCGenerateRTP,
IpProfile_SBCISUPBodyHandling, IpProfile_SBCISUPVariant,
IpProfile_SBCVoiceQualityEnhancement, IpProfile_SBCMaxOpusBW,
IpProfile_SBCEnhancedPlc, IpProfile_LocalRingbackTone,
IpProfile_LocalHeldTone, IpProfile_SBCGenerateNoOp;
IpProfile 1 = "S4B", 1, "AudioCodersGroups_0", 0, 10, 10, 46, 24, 0, 0,
2, 0, 0, 0, 0, -1, 1, 0, 0, -1, 1, 4, -1, 1, 1, 0, 0, "", "", 0, 0,
"audio", "", "", 0, 1, 0, 0, 0, 0, 0, 8, 300, 400, 0, 0, 0, "", 0, 0, 1,
3, 0, 0, 1, 0, 3, 2, 1, 0, 1, 1, 1, 1, 1, 0, 1, 0, 0, 0, 0, 1, 0, 1, 1,
0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 300, -1, -1, 0, 0, 0, 0, 0, 0, 0, -1, -
1, -1, -1, -1, 0, "", 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, -1, -1, 0;
IpProfile 2 = "VM", 1, "AudioCodersGroups_0", 0, 10, 10, 46, 24, 0, 0, 2,
0, 0, 0, 0, -1, 1, 0, 0, -1, 1, 4, -1, 1, 1, 0, 0, "", "", 0, 0, "",
"VM", "", 0, 2, 0, 0, 0, 1, 0, 8, 300, 400, 0, 0, 0, "", 0, 0, 1, 3, 0,
0, 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, 0, 300, -1, -1, 0, 0, 0, 0, 0, 0, 0, -1, -1, -1,
-1, -1, 0, "", 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, -1, -1, 0;

[ \IpProfile ]


[ CpMediaRealm ]

FORMAT CpMediaRealm_Index = CpMediaRealm_MediaRealmName,
CpMediaRealm_IPv4IF, CpMediaRealm_IPv6IF, CpMediaRealm_RemoteIPv4IF,
CpMediaRealm_RemoteIPv6IF, CpMediaRealm_PortRangeStart,
CpMediaRealm_MediaSessionLeg, CpMediaRealm_PortRangeEnd,
CpMediaRealm_IsDefault, CpMediaRealm_QoeProfile, CpMediaRealm_BWProfile,
CpMediaRealm_TopologyLocation;
CpMediaRealm 0 = "MRLan", "LAN_IF", "", "", "", 6000, 100, 6999, 0, "",
"", 0;
CpMediaRealm 1 = "MRWan", "WAN_IF", "", "", "", 7000, 100, 7999, 0, "",
"", 1;

[ \CpMediaRealm ]


[ SBCRoutingPolicy ]

FORMAT SBCRoutingPolicy_Index = SBCRoutingPolicy_Name,
SBCRoutingPolicy_LCREnable, SBCRoutingPolicy_LCRAverageCallLength,
SBCRoutingPolicy_LCRDefaultCost, SBCRoutingPolicy_LdapServerGroupName;
```

```
SBCRoutingPolicy 0 = "Default_SBCRoutingPolicy", 0, 1, 0, "";

[ \SBCRoutingPolicy ]


[ SRD ]

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

[ \SRD ]


[ MessagePolicy ]

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

[ \MessagePolicy ]


[ SIPInterface ]

FORMAT SIPInterface_Index = SIPInterface_InterfaceName,
SIPInterface_NetworkInterface, SIPInterface_ApplicationType,
SIPInterface_UDPPort, SIPInterface_TCPPort, SIPInterface_TLSPort,
SIPInterface_AdditionalUDPPorts, SIPInterface_SRDName,
SIPInterface_MessagePolicyName, SIPInterface_TLSContext,
SIPInterface_TLSMutualAuthentication, SIPInterface_TCPKeepaliveEnable,
SIPInterface_ClassificationFailureResponseType,
SIPInterface_PreClassificationManSet, SIPInterface_EncapsulatingProtocol,
SIPInterface_MediaRealm, SIPInterface_SBCDirectMedia,
SIPInterface_BlockUnRegUsers, SIPInterface_MaxNumOfRegUsers,
SIPInterface_EnableUnAuthenticatedRegistrations,
SIPInterface_UsedByRoutingServer, SIPInterface_TopologyLocation,
SIPInterface_PreParsingManSetName, SIPInterface_AdmissionProfile;
SIPInterface 0 = "S4B", "LAN_IF", 2, 5060, 0, 5067, "", "DefaultSRD", "",
"default", -1, 0, 500, -1, 0, "MRLan", 0, -1, -1, -1, 0, 0, "", "";
SIPInterface 1 = "VM", "WAN_IF", 2, 5060, 0, 0, "", "DefaultSRD", "",
"default", -1, 0, 500, -1, 0, "MRWan", 0, -1, -1, -1, 0, 1, "", "";

[ \SIPInterface ]


[ ProxySet ]

FORMAT ProxySet_Index = ProxySet_ProxyName,
ProxySet_EnableProxyKeepAlive, ProxySet_ProxyKeepAliveTime,
ProxySet_ProxyLoadBalancingMethod, ProxySet_IsProxyHotSwap,
ProxySet_SRDName, ProxySet_ClassificationInput, ProxySet_TLSContextName,
```

```
ProxySet_ProxyRedundancyMode, ProxySet_DNSResolveMethod,
ProxySet_KeepAliveFailureResp, ProxySet_GWIPv4SIPInterfaceName,
ProxySet_SBCIPv4SIPInterfaceName, ProxySet_GWIPv6SIPInterfaceName,
ProxySet_SBCIPv6SIPInterfaceName, ProxySet_MinActiveServersLB,
ProxySet_SuccessDetectionRetries, ProxySet_SuccessDetectionInterval,
ProxySet_FailureDetectionRetransmissions;
ProxySet 0 = "ProxySet_0", 0, 60, 0, 0, "DefaultSRD", 0, "", -1, -1, "",
"", "S4B", "", "", 1, 1, 10, -1;
ProxySet 1 = "S4B", 1, 60, 1, 1, "DefaultSRD", 0, "", 1, -1, "", "",
"S4B", "", "", 1, 1, 10, -1;
ProxySet 2 = "VM Trunk A", 1, 30, 0, 0, "DefaultSRD", 0, "", -1, -1,
"503", "", "VM", "", "", 1, 1, 10, -1;
ProxySet 3 = "VM Trunk B", 1, 30, 0, 0, "DefaultSRD", 0, "", -1, -1,
"503", "", "VM", "", "", 1, 1, 10, -1;
ProxySet 4 = "MP-Fax", 0, 60, 0, 0, "DefaultSRD", 0, "", -1, -1, "", "",
"S4B", "", "", 1, 1, 10, -1;

[ \ProxySet ]


[ IPGroup ]

FORMAT IPGroup_Index = IPGroup_Type, IPGroup_Name, IPGroup_ProxySetName,
IPGroup_SIPGroupName, IPGroup_ContactUser, IPGroup_SipReRoutingMode,
IPGroup_AlwaysUseRouteTable, IPGroup_SRDName, IPGroup_MediaRealm,
IPGroup_ClassifyByProxySet, IPGroup_ProfileName,
IPGroup_MaxNumOfRegUsers, IPGroup_InboundManSet, IPGroup_OutboundManSet,
IPGroup_RegistrationMode, IPGroup_AuthenticationMode, IPGroup_MethodList,
IPGroup_EnableSBCClientForking, IPGroup_SourceUriInput,
IPGroup_DestUriInput, IPGroup_ContactName, IPGroup_Username,
IPGroup_Password, IPGroup_UUIFormat, IPGroup_QOEProfile,
IPGroup_BWProfile, IPGroup_AlwaysUseSourceAddr, IPGroup_MsgManUserDef1,
IPGroup_MsgManUserDef2, IPGroup_SIPConnect, IPGroup_SBCPSAPMode,
IPGroup_DTLSContext, IPGroup_CreatedByRoutingServer,
IPGroup_UsedByRoutingServer, IPGroup_SBCOperationMode,
IPGroup_SBCRouteUsingRequestURIPort, IPGroup_SBCKeepOriginalCallID,
IPGroup_TopologyLocation, IPGroup_SBCDialPlanName,
IPGroup_CallSetupRulesSetId, IPGroup_Tags, IPGroup_SBCUserStickiness,
IPGroup_UserUDPPortAssignment, IPGroup_AdmissionProfile;
IPGroup 0 = 0, "Default_IPG", "", "", "", -1, 0, "DefaultSRD", "", 0, "",
-1, -1, -1, 0, 0, "", 0, -1, -1, "", "Admin", "$1$aCkNBwIC", 0, "", "",
0, "", "", 0, 0, "default", 0, 0, -1, 0, 0, 0, "", -1, "", 0, 0, "";
IPGroup 1 = 0, "S4B", "S4B", "213.106.222.186", "", -1, 0, "DefaultSRD",
"MRLan", 1, "S4B", -1, -1, -1, 0, 0, "", 0, -1, -1, "", "Admin",
"$1$aCkNBwIC", 0, "", "", 0, "", "", 0, 0, "default", 0, 0, -1, 0, 0, 0,
"", -1, "", 0, 0, "";
IPGroup 2 = 0, "VM Trunk A", "VM Trunk A", "213.106.222.186", "", -1, 0,
"DefaultSRD", "MRWan", 1, "VM", -1, -1, 4, 0, 0, "", 0, -1, -1, "",
"Admin", "$1$aCkNBwIC", 0, "", "", 0, "", "", 0, 0, "default", 0, 0, -1,
0, 0, 1, "", -1, "", 0, 0, "";
IPGroup 3 = 0, "VM Trunk B", "VM Trunk B", "82.14.171.234", "", -1, 0,
"DefaultSRD", "MRWan", 1, "VM", -1, -1, 4, 0, 0, "", 0, -1, -1, "",
"Admin", "$1$aCkNBwIC", 0, "", "", 0, "", "", 0, 0, "default", 0, 0, -1,
0, 0, 1, "", -1, "", 0, 0, "";
IPGroup 4 = 0, "MP-Fax", "MP-Fax", "213.106.222.186", "", -1, 0,
"DefaultSRD", "", 1, "", -1, -1, -1, 0, 0, "", 0, -1, -1, "", "Admin",
"$1$aCkNBwIC", 0, "", "", 0, "", "", 0, 0, "default", 0, 0, -1, 0, 0, 0,
"", -1, "", 0, 0, "";

[ \IPGroup ]


[ SBCAlternativeRoutingReasons ]
```

```
FORMAT SBCAlternativeRoutingReasons_Index =
SBCAlternativeRoutingReasons_ReleaseCause;
SBCAlternativeRoutingReasons 0 = 503;

[ \SBCAlternativeRoutingReasons ]


[ ProxyIp ]

FORMAT ProxyIp_Index = ProxyIp_ProxySetId, ProxyIp_ProxyIpIndex,
ProxyIp_IpAddress, ProxyIp_TransportType;
ProxyIp 0 = "1", 0, "FE.S4B.interop:5067", 2;
ProxyIp 1 = "2", 0, "213.106.222.186:5060", 0;
ProxyIp 2 = "3", 0, "82.14.171.234:5060", 0;
ProxyIp 3 = "4", 0, "10.15.77.12:5060", 0;

[ \ProxyIp ]


[ Account ]

FORMAT Account_Index = Account_ServedTrunkGroup,
Account_ServedIPGroupName, Account_ServingIPGroupName, Account_Username,
Account_Password, Account_HostName, Account_ContactUser,
Account_Register, Account_RegistrarStickiness,
Account_RegistrarSearchMode, Account_RegEventPackageSubscription,
Account_ApplicationType, Account_RegByServedIPG,
Account_UDPPortAssignment;
Account 0 = -1, "S4B", "VM Trunk A", "virginpbx01_01183374140",
"$1$WWsfbG0bGmtTJCEmIiFeUSssK18q", "", "", 0, 0, 0, 0, 2, 0, 0;
Account 1 = -1, "S4B", "VM Trunk B", "virginpbx01_01183374140",
"$1$WWsfbG0bGmtTJCEmIiFeUSssK18q", "", "", 0, 0, 0, 0, 2, 0, 0;
Account 2 = -1, "MP-Fax", "VM Trunk A", "virginpbx01_01183374140",
"$1$WWsfbG0bGmtTJCEmIiFeUSssK18q", "", "", 0, 0, 0, 0, 2, 0, 0;
Account 3 = -1, "MP-Fax", "VM Trunk B", "virginpbx01_01183374140",
"$1$WWsfbG0bGmtTJCEmIiFeUSssK18q", "", "", 0, 0, 0, 0, 2, 0, 0;

[ \Account ]


[ IP2IPRouting ]

FORMAT IP2IPRouting_Index = IP2IPRouting_RouteName,
IP2IPRouting_RoutingPolicyName, IP2IPRouting_SrcIPGroupName,
IP2IPRouting_SrcUsernamePrefix, IP2IPRouting_SrcHost,
IP2IPRouting_DestUsernamePrefix, IP2IPRouting_DestHost,
IP2IPRouting_RequestType, IP2IPRouting_MessageConditionName,
IP2IPRouting_ReRouteIPGroupName, IP2IPRouting_Trigger,
IP2IPRouting_CallSetupRulesSetId, IP2IPRouting_DestType,
IP2IPRouting_DestIPGroupName, IP2IPRouting_DestSIPInterfaceName,
IP2IPRouting_DestAddress, IP2IPRouting_DestPort,
IP2IPRouting_DestTransportType, IP2IPRouting_AltRouteOptions,
IP2IPRouting_GroupPolicy, IP2IPRouting_CostGroup, IP2IPRouting_DestTags,
IP2IPRouting_SrcTags, IP2IPRouting_IPGroupSetName,
IP2IPRouting_RoutingTagName, IP2IPRouting_InternalAction;
IP2IPRouting 0 = "OPTIONS termination", "Default_SBCRoutingPolicy",
"Any", "*", "*", "*", "*", 6, "", "Any", 0, -1, 1, "", "", "internal", 0,
-1, 0, 0, "", "", "", "", "default", "";
```

```
IP2IPRouting 1 = "S4B to VM Trunk A", "Default_SBCRoutingPolicy", "S4B",
"*", "*", "*", "*", 0, "", "Any", 0, -1, 0, "VM Trunk A", "", "", 0, -1,
0, 0, "", "", "", "", "default", "";
IP2IPRouting 2 = "S4B to VM Trunk B", "Default_SBCRoutingPolicy", "S4B",
"*", "*", "*", "*", 0, "", "Any", 0, -1, 0, "VM Trunk B", "", "", 0, -1,
1, 0, "", "", "", "", "default", "";
IP2IPRouting 3 = "Fax to VM Trunk A", "Default_SBCRoutingPolicy", "MP-
Fax", "*", "*", "*", "*", 0, "", "Any", 0, -1, 0, "VM Trunk A", "", "",
0, -1, 0, 0, "", "", "", "", "default", "";
IP2IPRouting 4 = "Fax to VM Trunk B", "Default_SBCRoutingPolicy", "MP-
Fax", "*", "*", "*", "*", 0, "", "Any", 0, -1, 0, "VM Trunk B", "", "",
0, -1, 1, 0, "", "", "", "", "default", "";
IP2IPRouting 5 = "VM to Fax", "Default_SBCRoutingPolicy", "Any", "*",
"*", "+441183374147", "*", 0, "", "Any", 0, -1, 0, "MP-Fax", "", "", 0, -
1, 0, 0, "", "", "", "", "default", "";
IP2IPRouting 6 = "VM to S4B", "Default_SBCRoutingPolicy", "Any", "*",
"*", "*", "*", 0, "", "Any", 0, -1, 0, "S4B", "", "", 0, -1, 0, 0, "",
"", "", "", "default", "";

[ \IP2IPRouting ]


[ IPOutboundManipulation ]

FORMAT IPOutboundManipulation_Index =
IPOutboundManipulation_ManipulationName,
IPOutboundManipulation_RoutingPolicyName,
IPOutboundManipulation_IsAdditionalManipulation,
IPOutboundManipulation_SrcIPGroupName,
IPOutboundManipulation_DestIPGroupName,
IPOutboundManipulation_SrcUsernamePrefix, IPOutboundManipulation_SrcHost,
IPOutboundManipulation_DestUsernamePrefix,
IPOutboundManipulation_DestHost,
IPOutboundManipulation_CallingNamePrefix,
IPOutboundManipulation_MessageConditionName,
IPOutboundManipulation_RequestType,
IPOutboundManipulation_ReRouteIPGroupName,
IPOutboundManipulation_Trigger, IPOutboundManipulation_ManipulatedURI,
IPOutboundManipulation_RemoveFromLeft,
IPOutboundManipulation_RemoveFromRight,
IPOutboundManipulation_LeaveFromRight, IPOutboundManipulation_Prefix2Add,
IPOutboundManipulation_Suffix2Add,
IPOutboundManipulation_PrivacyRestrictionMode,
IPOutboundManipulation_DestTags, IPOutboundManipulation_SrcTags;
IPOutboundManipulation 0 = "To Emergency do nothing",
"Default_SBCRoutingPolicy", 0, "Any", "Any", "*", "*", "[999,112,18000]",
"*", "*", "", 0, "Any", 0, 1, 0, 0, 255, "", "", 0, "", "";
IPOutboundManipulation 1 = "To Emergency strip +",
"Default_SBCRoutingPolicy", 0, "Any", "Any", "*", "*",
"[+999,+112,+18000]", "*", "*", "", 0, "Any", 0, 1, 1, 0, 255, "", "", 0,
"", "";
IPOutboundManipulation 2 = "Do nothing", "Default_SBCRoutingPolicy", 0,
"Any", "Any", "*", "*", "+", "*", "*", "", 0, "Any", 0, 1, 0, 0, 255, "",
"", 0, "", "";
IPOutboundManipulation 3 = "Add + toward VM", "Default_SBCRoutingPolicy",
0, "Any", "Any", "*", "*", "*", "*", "*", "", 0, "Any", 0, 1, 0, 0, 255,
"+", "", 0, "", "";

[ \IPOutboundManipulation ]


[ MessageManipulations ]
```

```
FORMAT MessageManipulations_Index = 
MessageManipulations_ManipulationName, MessageManipulations_ManSetID, 
MessageManipulations_MessageType, MessageManipulations_Condition, 
MessageManipulations_ActionSubject, MessageManipulations_ActionType, 
MessageManipulations_ActionValue, MessageManipulations_RowRole;
MessageManipulations 0 = "Change Failure Response 410 to 480", 4, 
"any.response", "header.request-uri.methodtype=='410'", "header.request-
uri.methodtype", 2, "'480'", 0;
MessageManipulations 1 = "Add Options to Allow Header", 4, 
"invite.request", "header.allow regex (.*)", "header.allow", 2, 
"$1+',OPTIONS'", 0;
MessageManipulations 2 = "Change Dest in R-URI", 10, "Options", "", 
"Header.Request-URI.URL.Host", 2, "Param.Message.Address.Dst.Address", 0;
MessageManipulations 3 = "Change Dest in To", 10, "Options", "", 
"Header.To.URL.Host", 2, "Param.Message.Address.Dst.Address", 0;
MessageManipulations 4 = "For test 27", 14, "Any.Request", "Header.P-
Asserted-Identity exists", "Header.P-Asserted-Identity.URL.User", 2, 
"'+441183374142'", 0;

[ \MessageManipulations ]


[ GwRoutingPolicy ]

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

[ \GwRoutingPolicy ]


[ ResourcePriorityNetworkDomains ]

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

[ \ResourcePriorityNetworkDomains ]


[ MaliciousSignatureDB ]

FORMAT MaliciousSignatureDB_Index = MaliciousSignatureDB_Name, 
MaliciousSignatureDB_Pattern;
MaliciousSignatureDB 0 = "SIPVicious", "Header.User-Agent.content prefix 
'friendly-scanner'";
MaliciousSignatureDB 1 = "SIPScan", "Header.User-Agent.content prefix 
'sip-scan'";
MaliciousSignatureDB 2 = "Smap", "Header.User-Agent.content prefix 
'smap'";
MaliciousSignatureDB 3 = "Sipsak", "Header.User-Agent.content prefix 
'sipsak'";
MaliciousSignatureDB 4 = "Sipcli", "Header.User-Agent.content prefix 
'sipcli'";
```

```
MaliciousSignatureDB 5 = "Sivus", "Header.User-Agent.content prefix
'SIVuS'";
MaliciousSignatureDB 6 = "Gulp", "Header.User-Agent.content prefix
'Gulp'";
MaliciousSignatureDB 7 = "Sipv", "Header.User-Agent.content prefix
'sipv'";
MaliciousSignatureDB 8 = "Sundayddr Worm", "Header.User-Agent.content
prefix 'sundayddr'";
MaliciousSignatureDB 9 = "VaxIPUserAgent", "Header.User-Agent.content
prefix 'VaxIPUserAgent'";
MaliciousSignatureDB 10 = "VaxSIPUserAgent", "Header.User-Agent.content
prefix 'VaxSIPUserAgent'";
MaliciousSignatureDB 11 = "SipArmyKnife", "Header.User-Agent.content
prefix 'siparmyknife'";

[ \MaliciousSignatureDB ]


[ AllowedAudioCoders ]

FORMAT AllowedAudioCoders_Index =
AllowedAudioCoders_AllowedAudioCodersGroupName,
AllowedAudioCoders_AllowedAudioCodersIndex, AllowedAudioCoders_CoderID,
AllowedAudioCoders_UserDefineCoder;
AllowedAudioCoders 0 = "VM", 0, 1, "";

[ \AllowedAudioCoders ]


[ AudioCoders ]

FORMAT AudioCoders_Index = AudioCoders_AudioCodersGroupId,
AudioCoders_AudioCodersIndex, AudioCoders_Name, AudioCoders_pTime,
AudioCoders_rate, AudioCoders_PayloadType, AudioCoders_Sce,
AudioCoders_CoderSpecific;
AudioCoders 0 = "AudioCodersGroups_0", 0, 1, 2, 90, -1, 0, "";

[ \AudioCoders ]
```

B Configuring Analog Devices (ATAs) for Fax Support

This section describes how to configure the analog device entity to route its calls to the AudioCodes Media Gateway for supporting faxes or Minicoms. The analog device entity must be configured to send all calls to the AudioCodes SBC.

Note: The configuration described in this section is for ATA devices configured for AudioCodes MP-11x series.

B.1 Step 1: Configure the Endpoint Phone Number Table

The 'Endpoint Phone Number Table' page allows you to activate the MP-11x ports (endpoints) by defining telephone numbers. The configuration below uses the example of ATA destination phone number "+441183374143" (IP address 10.15.17.12) with all routing directed to the SBC device (10.15.17.10).

➢ **To configure the Endpoint Phone Number table:**

1. Open the Endpoint Phone Number Table page (**Configuration** tab > **VoIP** menu > **GW and IP to IP submenu** > **Hunt Group** sub-menu > **Endpoint Phone Number**).

Figure B-1: Endpoint Phone Number Table Page

Endpoint Phone Number Table

	Channel(s)	Phone Number	Hunt Group ID	Tel Profile ID
1	1	+441183374143		0
2				
3				

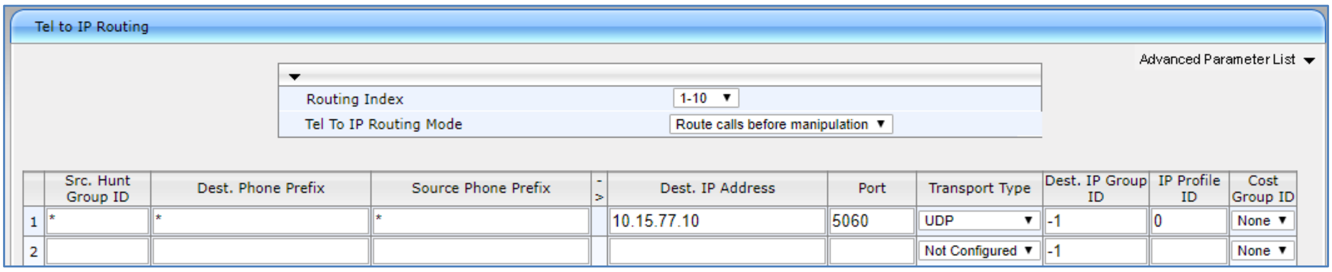
B.2 Step 2: Configure Tel to IP Routing Table

This step describes how to configure the Tel-to-IP routing rules to ensure that the MP-11x device sends all calls to the AudioCodes central E-SBC device.

➢ To configure the Tel to IP Routing table:

1. Open the Tel to IP Routing page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** sub-menu > **Routing** sub-menu > **Tel to IP Routing**).

Figure B-2: Tel to IP Routing Page



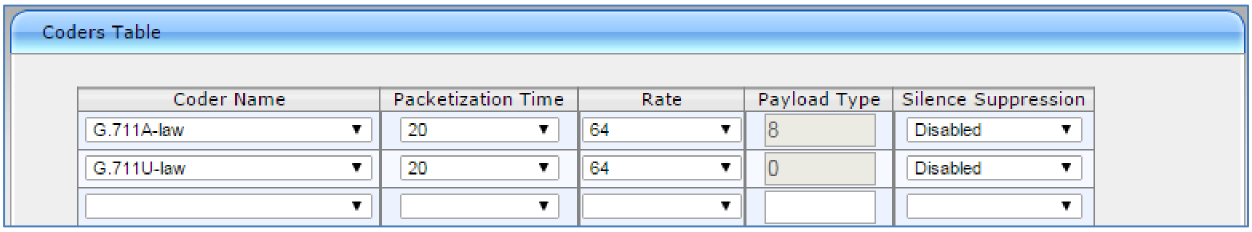
B.3 Step 3: Configure Coders Table

This step describes how to configure the coders for the MP-11x device.

➢ To configure MP-11x coders:

1. Open the Coders page (**Configuration** tab > **VoIP** menu > **Coders And Profiles** sub-menu > **Coders**).

Figure B-3: Coders Table Page



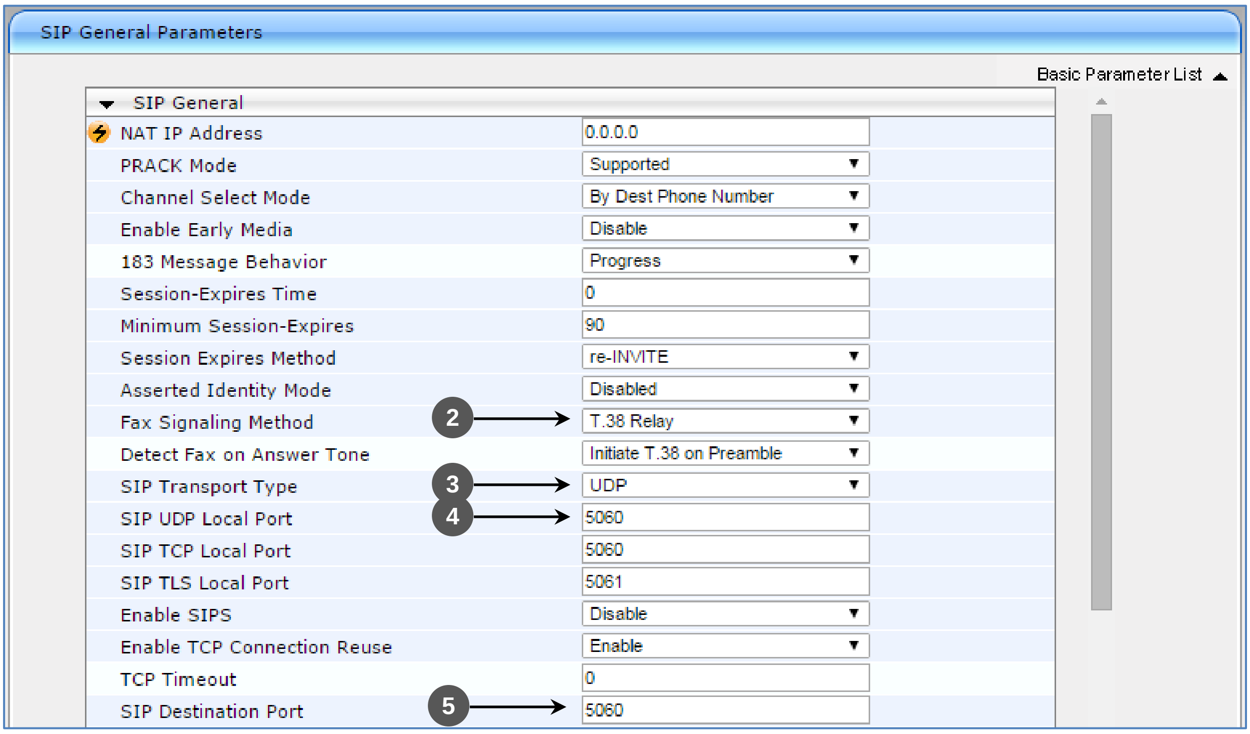
B.4 Step 4: Configure SIP UDP Transport Type and Fax Signaling Method

This step describes how to configure the fax signaling method for the MP-11x device.

➢ **To configure the fax signaling method:**

1. Open the SIP General Parameters page (**Configuration** tab > **VoIP** menu > **SIP Definitions** submenu > **General Parameters**).

Figure B-4: SIP General Parameters Page



2. From the ‘FAX Signaling Method’ drop-down list, select **G.711 Transport** for G.711 fax support and select **T.38 Relay** for T.38 fax support**.**
3. From the ‘SIP Transport Type’ drop-down list, select **UDP***.*
4. In the ‘SIP UDP Local Port’ field, enter **5060** (corresponding to the Central Gateway UDP transmitting port configuration).
5. In the ‘SIP Destination Port’, enter **5060** (corresponding to the Central Gateway UDP listening port configuration).

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