Configuration Note

AudioCodes Professional Services – Interoperability Lab

Mediant™ E-SBC for Mediant CCE Appliance and ITSP SIP Trunk

Version 7.2

Skype for Business

Microsoft Partner

Gold Communications

HD VoIP Sounds Better

AudioCodes

Table of Contents

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Notice

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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 ITSP's SIP Trunk and AudioCodes Mediant CCE Appliance.

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

1.1 Intended Audience

The document is intended for engineers, or AudioCodes and ITSP Partners who are responsible for installing and configuring ITSP's SIP Trunk and Microsoft's Skype for Business CCE 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 800B Gateway & E-SBC ▪ Mediant Software SBC (VE)
Software Version	SIP_7.20A or later
Protocol	▪ SIP/UDP (to the ITSP SIP Trunk) ▪ SIP/TCP or TLS (to the S4B Mediation Server)
Additional Notes	The Mediant CCE appliance is delivered with M800 SBC or SSBC in HP server.

2.2 Mediant CCE Appliance Version

Table 2-2: Microsoft Skype for Business CCE Version

Vendor	AudioCodes
Model	Mediant CCE Appliance
Software Version	Release 2.0.2
Protocol	SIP
Additional Notes	None

2.3 Microsoft Skype for Business Cloud Connector Edition Version

Table 2-3: Microsoft Skype for Business CCE Version

Vendor	Microsoft
Model	Cloud Connector Edition
Software Version	Release 1.4.2

2.4 Deploying the SBC

2.4.1 Example Environment

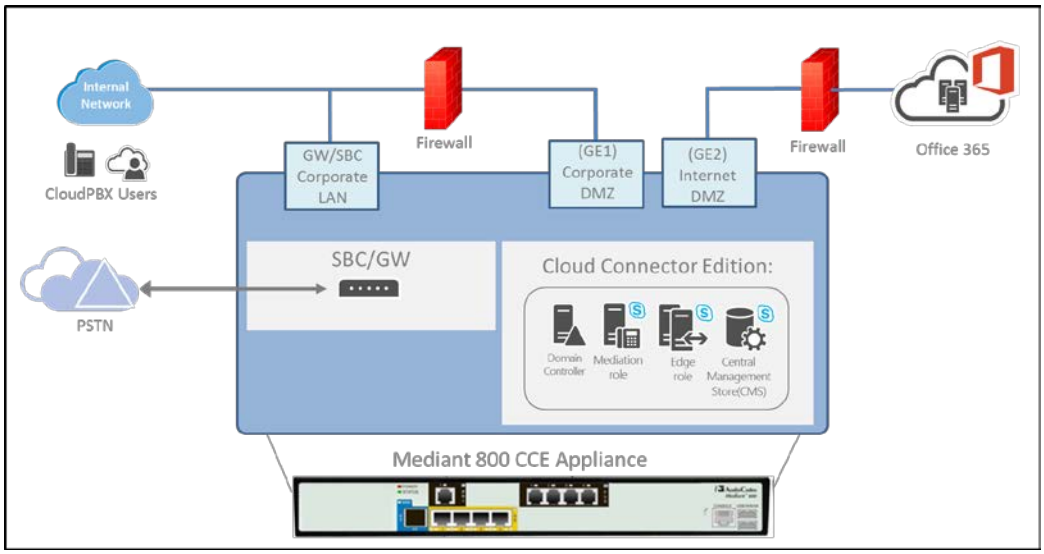
The example scenario below is referred to throughout this document in order to show how to deploy the SBC.

In the example environment:

- Enterprise deployed the Mediant CCE Appliance in its network for enhanced communication within the Cloud PBX.
- Enterprise wishes to offer its employees to connect the Enterprise to the Local PSTN network using ITSP's SIP Trunking service.
- AudioCodes E-SBC is implemented to interconnect between the CCE and the SIP Trunk.
  - **Session:** Real-time voice session using the IP-based Session Initiation Protocol (SIP).
  - **Border:** IP-to-IP network border between Mediant CCE network in the Enterprise LAN and ITSP's SIP Trunk located in the public network.

The figure below illustrates this interoperability test topology:

Figure 2-1: Example Environment Topology between E-SBC and Mediant CCE with ITSP SIP Trunk



2.4.2 Environment Setup

The interoperability test topology includes the following environment setup:

Table 2-4: Environment Setup

Area	Setup
Network	▪ CCE Mediation is located on the Corporate DMZ ▪ ITSP SIP Trunk is located on the Internet DMZ
Signaling Transcoding	▪ CCE operates with SIP-over-TLS transport type ▪ ITSP SIP Trunk operates with SIP-over-UDP transport type
Codecs Transcoding	▪ CCE supports G.711A-law and G.711U-law coders ▪ ITSP SIP Trunk supports G.711A-law, G.711U-law, and G.729 coder
Media Transcoding	▪ CCE operates with SRTP media type ▪ ITSP SIP Trunk operates with RTP media type

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3 Configuring Skype for Business CCE

This chapter describes how to configure Mediant CCE Appliance to assign certificate to the AudioCodes E-SBC.

Note: Other Settings on Cloud Connector Edition (CCE) are beyond the scope of this document. Refer to *LTRT-28086 Mediant Appliance for Microsoft Skype for Business CCE Installation Manual Ver. 2.0.2*.

3.1 Setting the SBC Certificate on the CCE

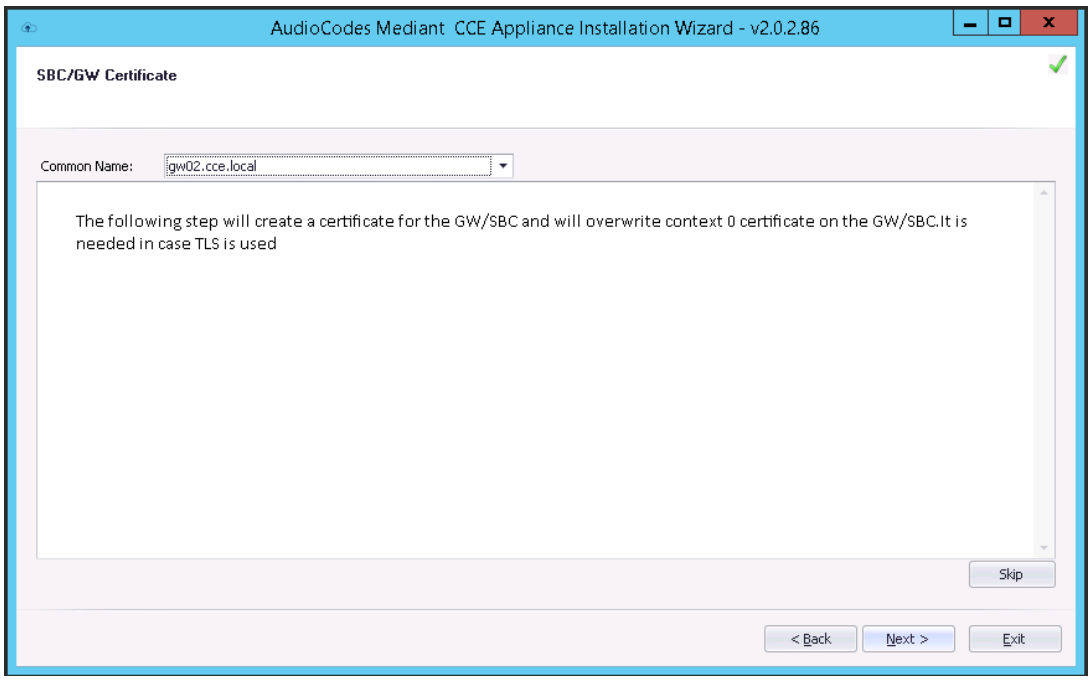
This procedure describes how to set the collocated SBC certificate. This certificate is required in case TLS is used to secure the connection between the Gateway/SBC and the CCE Mediation server.

The new certificate will be signed by the CCE internal CA automatically and be uploaded to the Gateway/SBC under Context index 0 (note: the new certificate will override any other certificate in context 0).

➢ **To set the SBC certificate:**

1. Select the correct SBC FQDN from the Common Name field. as shown below:

Figure 3-1: setting the SBC certificate



2. Click the **Next** button to continue.

Note: To deploy the certificate, ensure that the CCE has access to the SBC OSN IP (i.e., 169.254.100.1). The SBC is set with the parameter *OSNINTERNALVLAN=1*.

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4 Configuring AudioCodes E-SBC

This chapter provides step-by-step procedures on how to configure AudioCodes E-SBC for interworking between Mediant CCE Appliance and the ITSP SIP Trunk. These configuration procedures are based on the interoperability test topology described in Section 2.4.

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 ITSP 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**

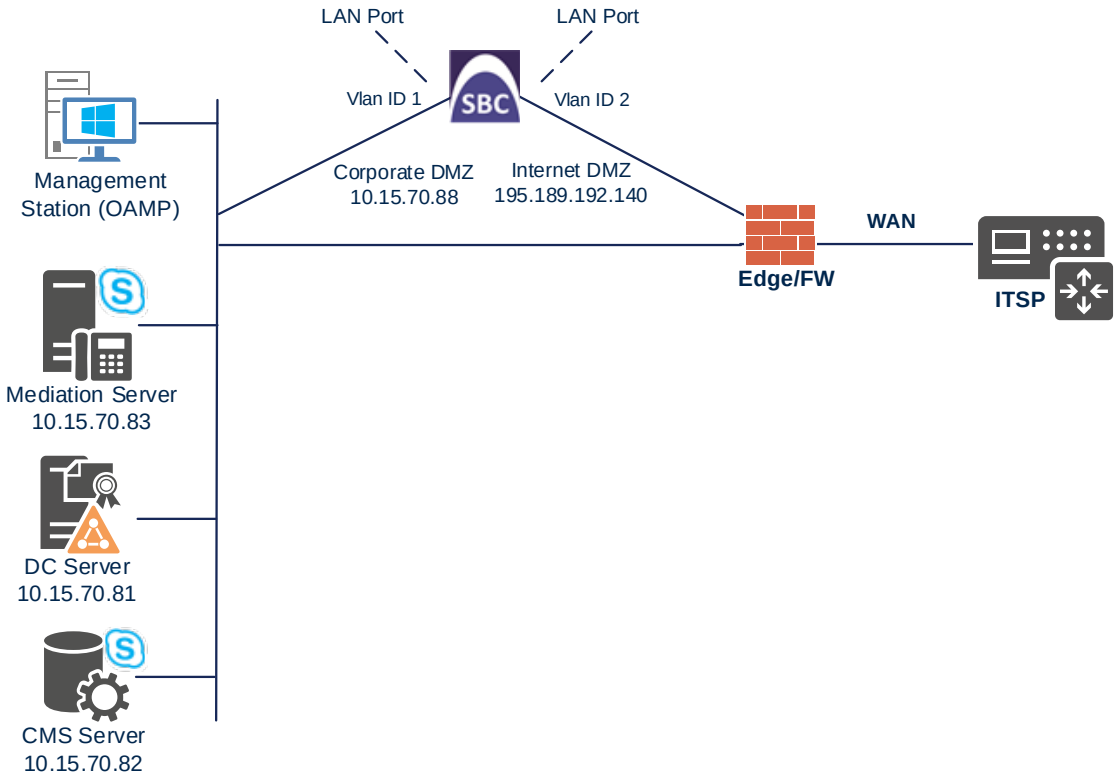
  For more information about the License Key, contact your AudioCodes sales representative.
- The scope of this interoperability test and document does **not** cover all security aspects for connecting the SIP Trunk to the Microsoft Skype for Business environment. Comprehensive security measures should be implemented per your organization's security policies. For security recommendations on AudioCodes' products, refer to the *Recommended Security Guidelines* document.

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, scenario exemplified in this document employs the following deployment method:

- E-SBC interfaces with the following IP entities:
  - CCE Mediation server, located on the Corporate DMZ
  - ITSP SIP Trunk, located on the Internet DMZ
- Physical connection: The type of physical connection to the Corporate depends on the method used to connect to the Enterprise's network. In this example, E-SBC connects to the Corporate DMZ and Internet DMZ using dedicated LAN ports (i.e., two ports and two network cables are used).
- E-SBC also uses two logical network interfaces:
  - Corporate DMZ (VLAN ID 1)
  - Internet 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:

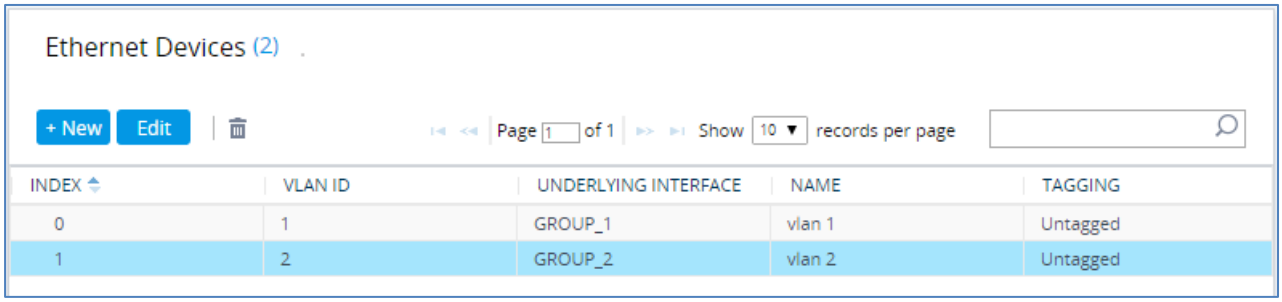
- Corporate DMZ VoIP (assigned the name "vlan 1")
- Internet DMZ VoIP (assigned the name "vlan 2")

➢ **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 IP Network Interfaces for LAN and WAN

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

- Corporate DMZ VoIP (assigned the name "LAN_IF")
- Internet DMZ VoIP (assigned the name "WAN_IF")

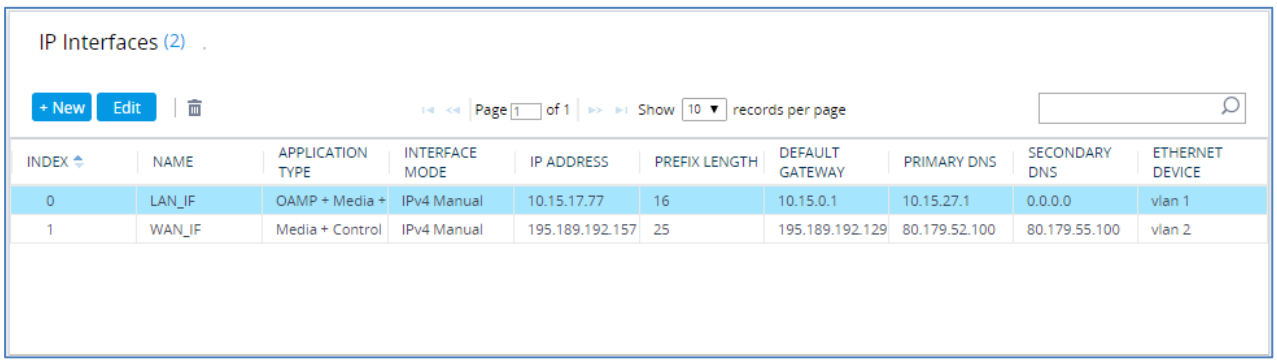
➢ **To configure the IP network interfaces:**

1. Open the IP Interfaces table (**Setup** menu > **IP Network** tab > **Core Entities** folder > **IP Interfaces**).
2. Configure the interface as follows:

IP Interface	Specific Configuration						
	Name	**Application Type**	**IP Address**	**Prefix Length**	**Default Gateway**	**Primary DNS**	**Ethernet Device**
Interfacing with Corporate DMZ	**LAN_IF**	**OAMP + Media + Control**	**10.15.70.88**	**16**	**10.15.0.1**	**10.15.28.1**	**vlan 1**
Interfacing with Internet DMZ	**WAN_IF**	**Media + Control**	**195.189.192.141**	**25**	**195.189.192.129**	**8.8.8.8**	**vlan 2**

3. Click **Apply**. The configured IP network interfaces are shown below:

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



INDEX	NAME	APPLICATION TYPE	INTERFACE MODE	IP ADDRESS	PREFIX LENGTH	DEFAULT GATEWAY	PRIMARY DNS	SECONDARY DNS	ETHERNET DEVICE
0	LAN_IF	OAMP + Media +	IPv4 Manual	10.15.17.77	16	10.15.0.1	10.15.27.1	0.0.0.0	vlan 1
1	WAN_IF	Media + Control	IPv4 Manual	195.189.192.157	25	195.189.192.129	80.179.52.100	80.179.55.100	vlan 2

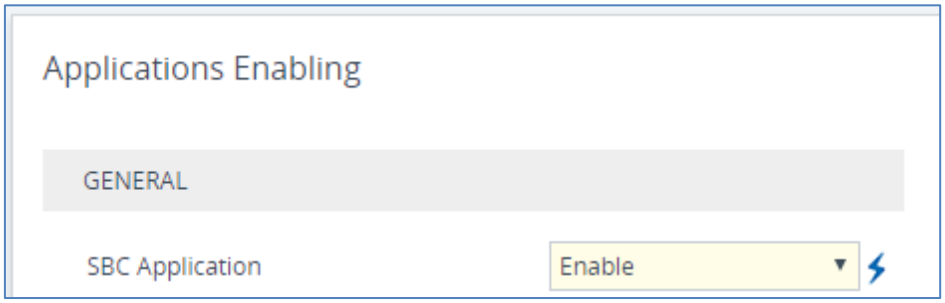
4.2 Step 2: Enable the SBC Application

This step describes how to enable the SBC application.

➢ **To enable the SBC application:**

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

Figure 4-4: Enabling SBC Application



2. From the 'SBC Application' drop-down list, select **Enable**.
3. Click **Apply**.
4. Reset the E-SBC with a burn to flash for this setting to take effect (see Section 4.17 on page 54).

4.3 Step 3: Configure Media Realms

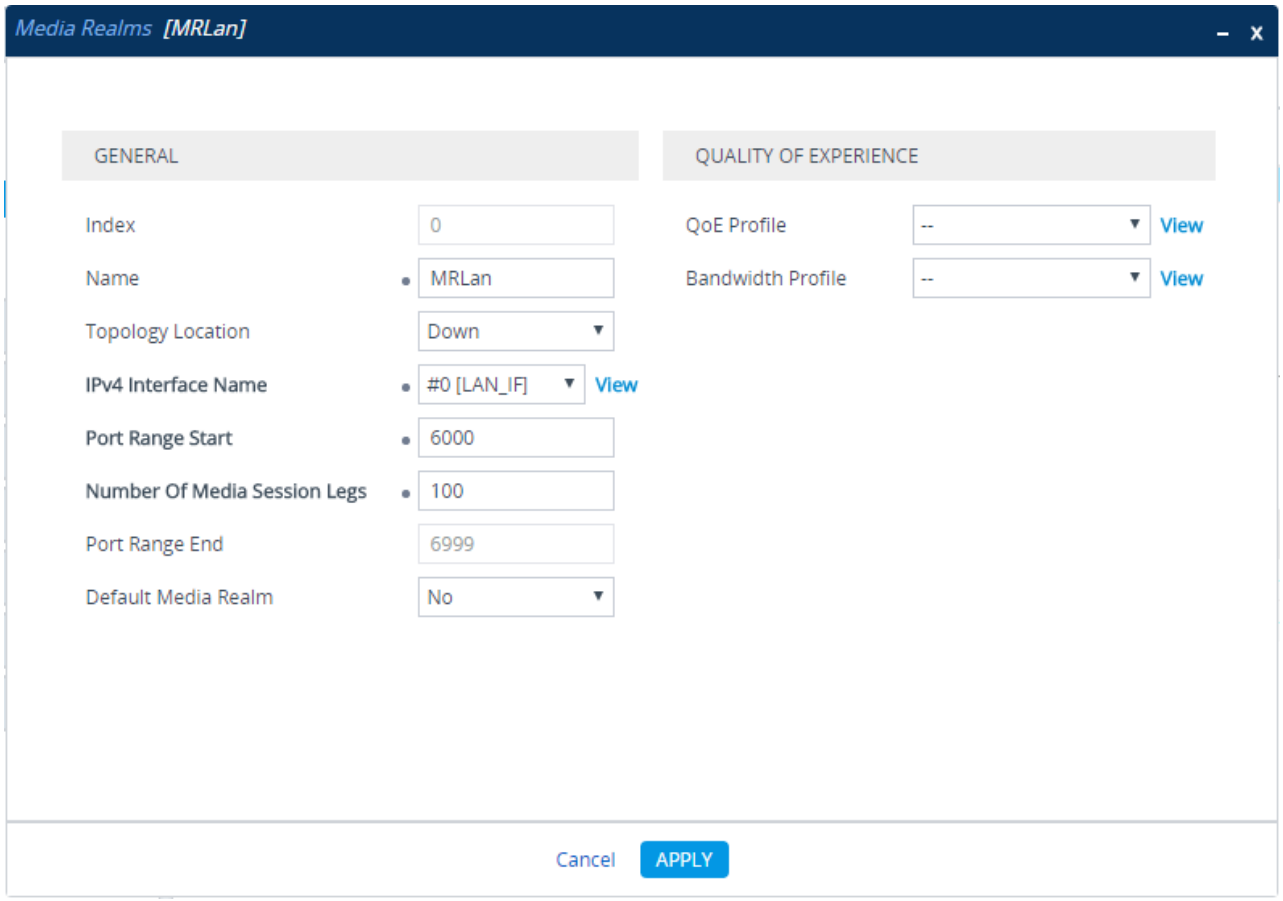
This step describes how to configure Media Realms. The simplest configuration is to create two Media Realms - one for internal (LAN) traffic and one for external (WAN) traffic.

➢ **To configure Media Realms:**

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

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

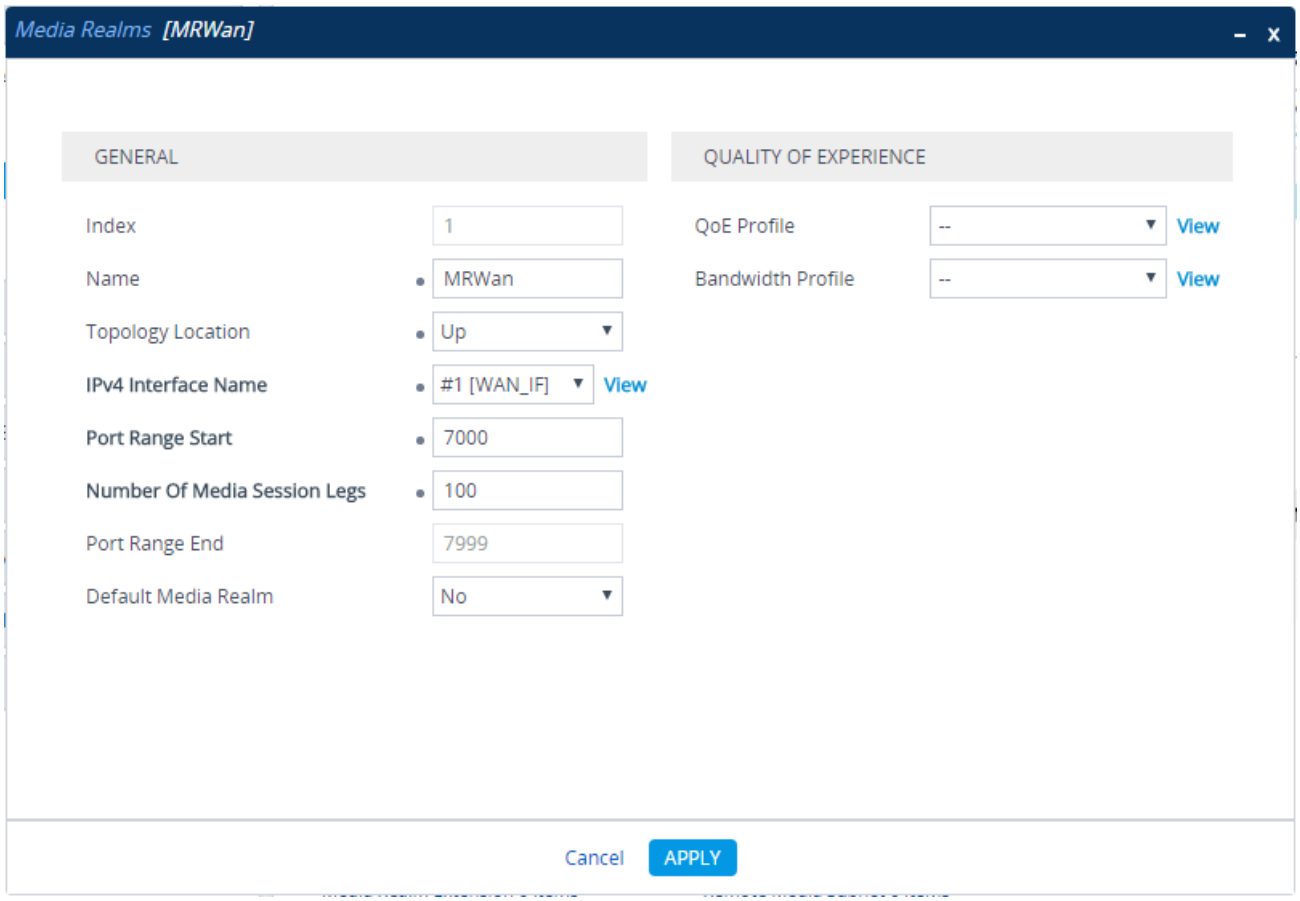
Figure 4-5: Configuring Media Realm for LAN



3. Configure a Media Realm for WAN traffic:

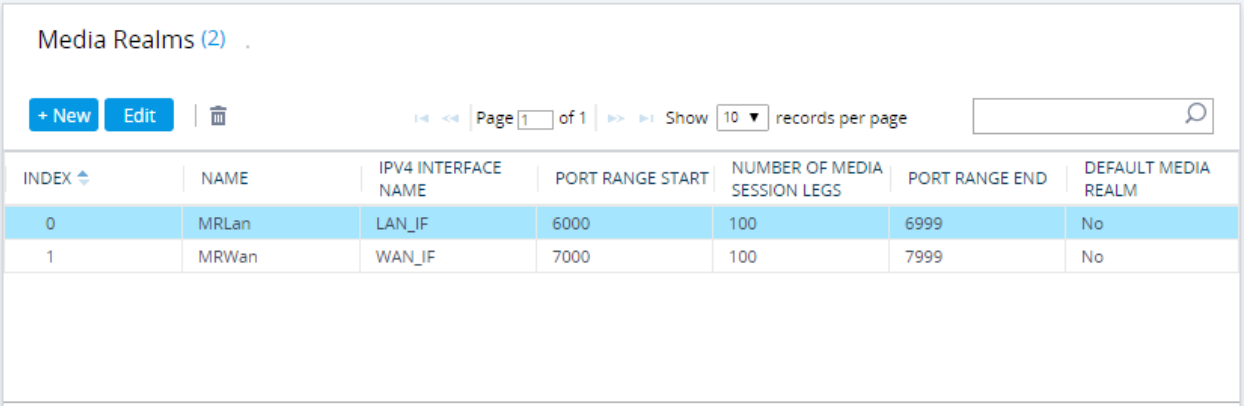
Parameter	Value
Index	**1**
Name	**MRWan** (arbitrary name)
Topology Location	**Up**
IPv4 Interface Name	**WAN_IF**
Port Range Start	**7000** (represents lowest UDP port number used for media on WAN)
Number of Media Session Legs	**100** (media sessions assigned with port range)

Figure 4-6: Configuring Media Realm for WAN



The configured Media Realms are shown in the figure below:

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



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

4.4 Step 4: Configure SIP Signaling Interfaces

This step describes how to configure SIP Interfaces. In the example scenario, an internal and external SIP Interface must be configured for the E-SBC.

➢ **To configure SIP Interfaces:**

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

Parameter	Value
Index	**0**
Name	**S4B** (see note at the end of this section)
Network Interface	**LAN_IF**
Application Type	**SBC**
UDP and TCP	**0**
TLS Port	**5067** (see note below)
Media Realm	**MRLan**

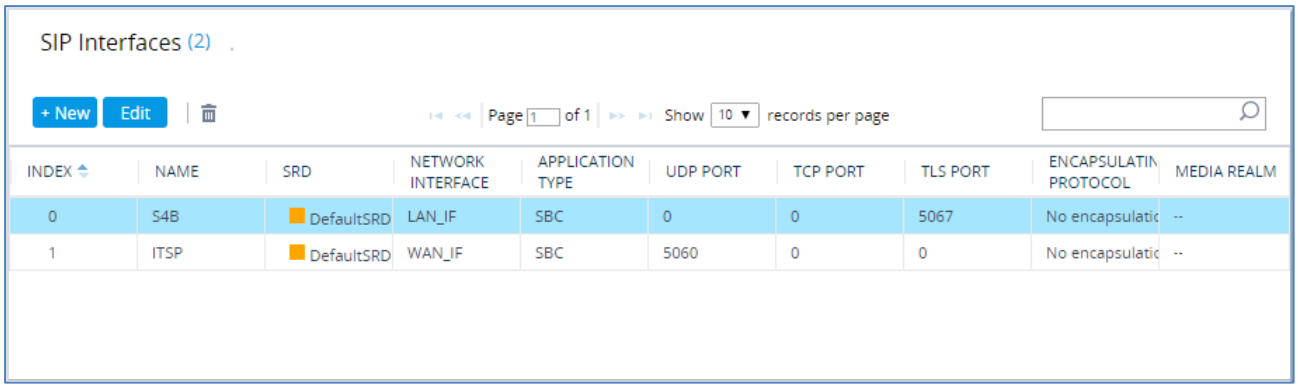
Note: The TLS port parameter must be identically as configured during the Mediant CCE installation using the CCE Install Wizard.

3. Configure a SIP Interface for the WAN:

Parameter	Value
Index	**1**
Name	**ITSP**
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 4-8: Configured SIP Interfaces in SIP Interface Table



INDEX	NAME	SRD	NETWORK INTERFACE	APPLICATION TYPE	UDP PORT	TCP PORT	TLS PORT	ENCAPSULATIN PROTOCOL	MEDIA REALM
0	S4B	DefaultSRD	LAN_IF	SBC	0	0	5067	No encapsulatic	--
1	ITSP	DefaultSRD	WAN_IF	SBC	5060	0	0	No encapsulatic	--

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

4.5 Step 5: Configure Proxy Sets

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

In the example scenario, two Proxy Sets need to be configured for the following IP entities:

- Mediant CCE
- ITSP SIP Trunk

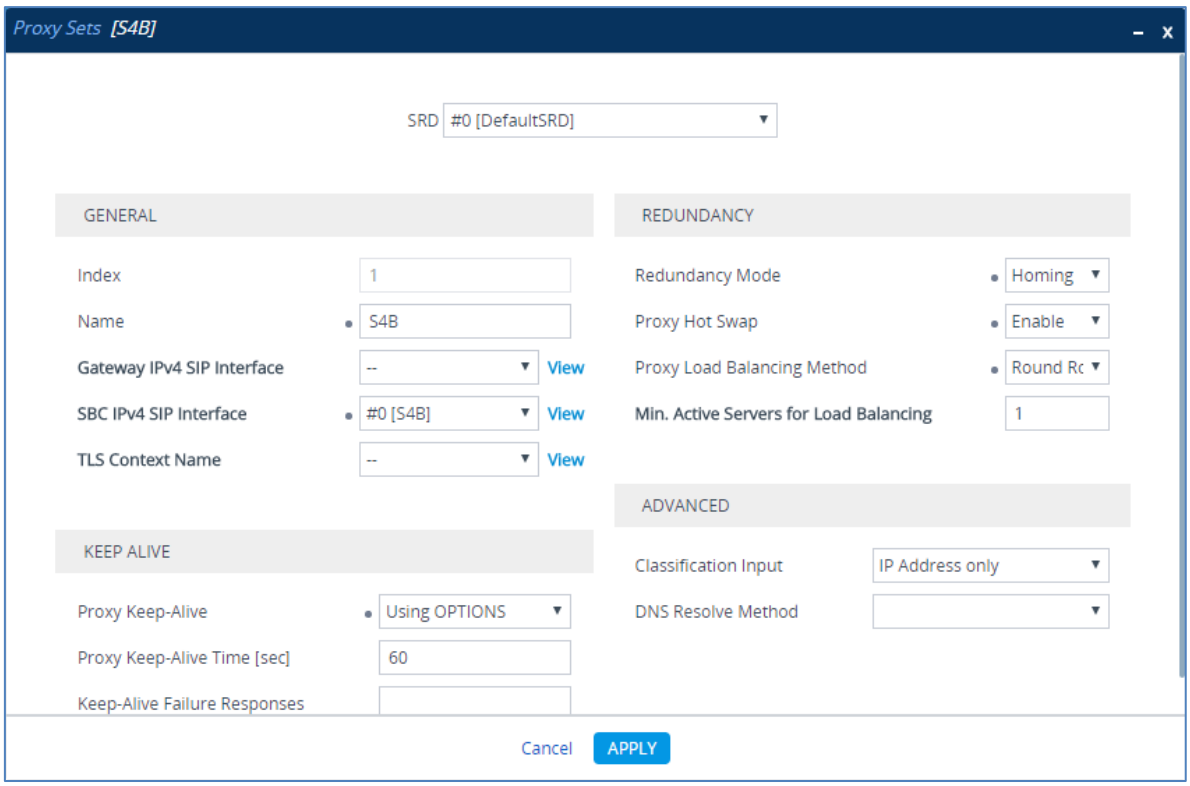
The Proxy Sets will be later applied to the VoIP network by assigning them to IP Groups.

➢ **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 Mediant CCE as shown below:

Parameter	Value
Index	**1**
Name	**S4B**
SBC IPv4 SIP Interface	**S4B**
Proxy Keep-Alive	**Using Options**
Proxy Hot Swap	**Enable**
Load Balancing Method	**Round Robin**

Figure 4-9: Configuring Proxy Set for CCE



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

Parameter	Value
Index	**0**
Proxy Address	**10.15.70.83:5067** (Primary CCE IP address / FQDN and destination port)
Transport Type	**TLS**

d. If there is additional CCE on the Site, Configure the parameters as described in the table below:

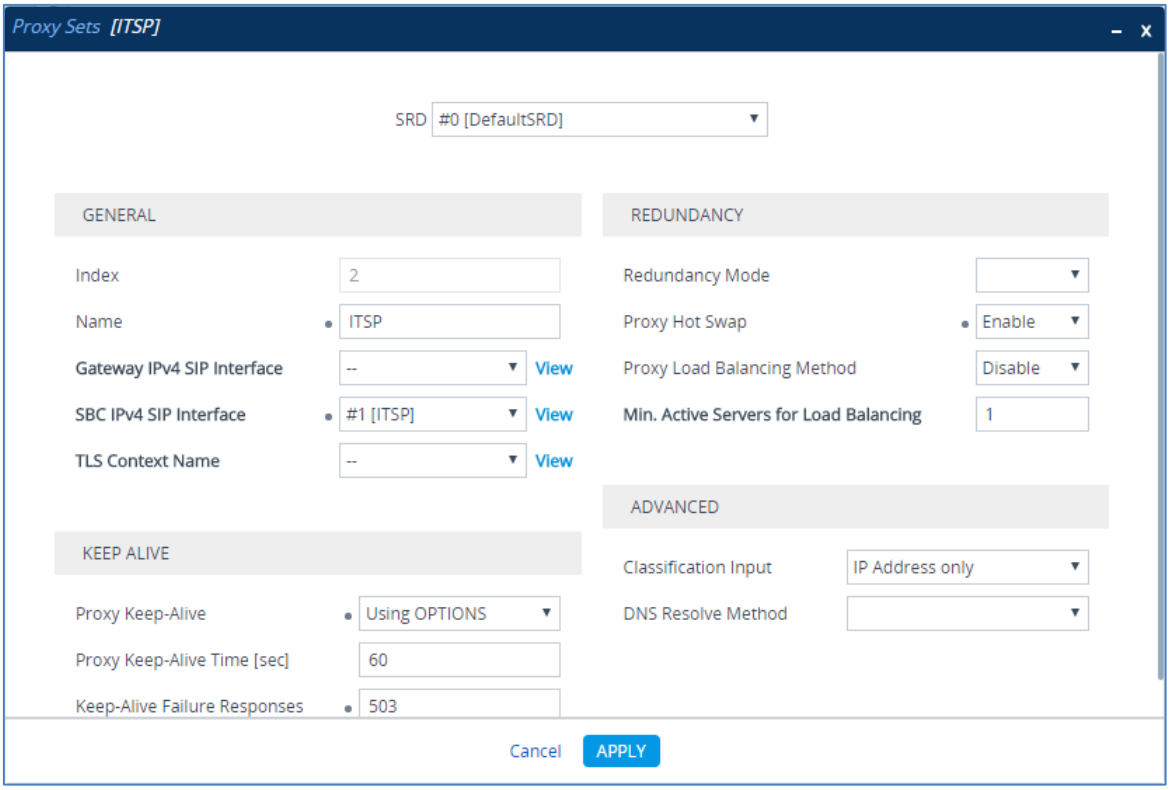
Parameter	Value
Index	**1**
Proxy Address	**10.15.70.93:5067** (Secondary CCE IP address / FQDN and destination port)
Transport Type	**TLS**

e. Click **Apply**.

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

Parameter	Value
Index	**2**
Name	**ITSP**
SBC IPv4 SIP Interface	**ITSP**
Proxy Keep-Alive	**Using Options**
Keep-Alive Failure responses	**503** (If this is received in response to a keep-alive message using SIP OPTIONS, the SBC considers the proxy as down and tries the next proxy.)
Proxy Hot Swap	**Enable**

Figure 4-10: Configuring Proxy Set for ITSP SIP Trunk



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

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

Figure 4-11: Configuring Proxy Address for ITSP SIP Trunk

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

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

d. Click **Apply**.

4.6 Step 6: Configure Coders

This step describes how to configure coders (termed *Coder Group*). As Skype for Business CCE supports the G.711 coder while the network connection to ITSP SIP Trunk may restrict operation with a lower bandwidth coder such as G.729, you need to add a Coder Group with the G.729 coder for the ITSP SIP Trunk.

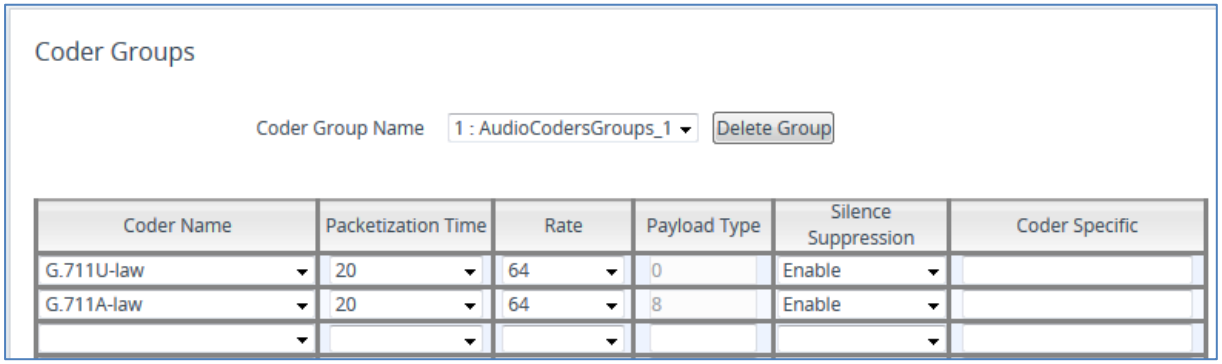
Note that the Coder Group ID for this entity will be assign to its corresponding IP Profile in the next step.

➢ **To configure coders:**

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

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

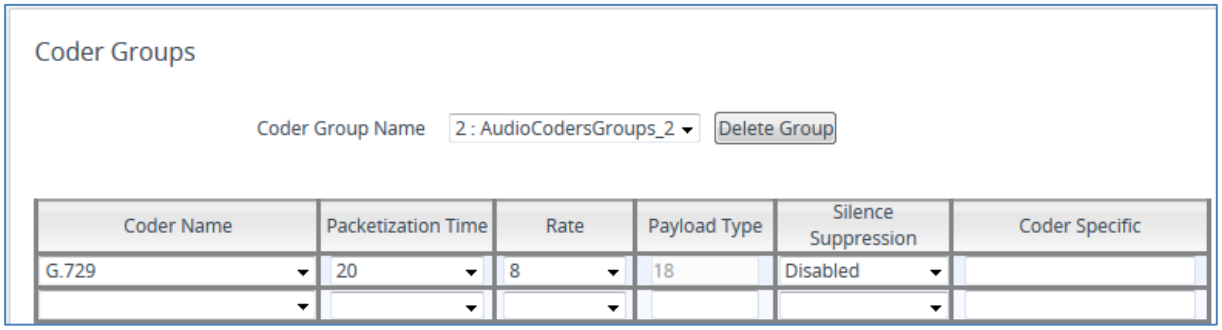
Figure 4-12: Configuring Coder Group for CCE



3. Configure a Coder Group for ITSP SIP Trunk:

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

Figure 4-13: Configuring Coder Group for ITSP SIP Trunk

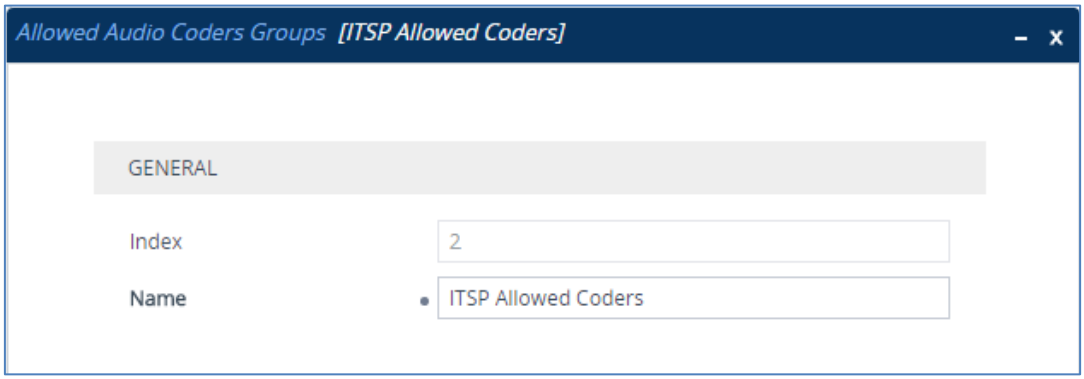


The procedure below describes how to configure an Allowed Coders Group to ensure that voice sent to the ITSP SIP Trunk uses the G.729 coder whenever possible. Note that this Allowed Coders Group ID will be assign to the IP Profile belonging to the ITSP SIP Trunk Profile in the next step.

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

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

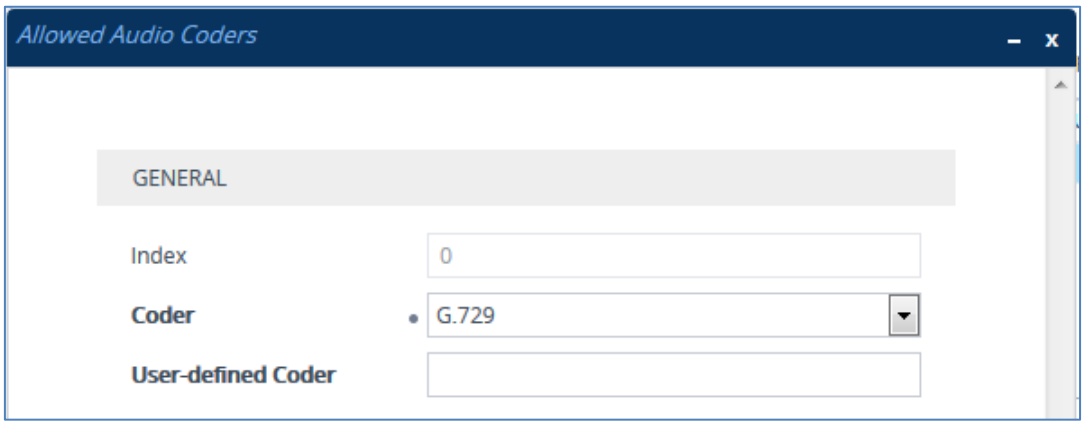
Figure 4-14: Configuring Allowed Coders Group for ITSP SIP Trunk



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

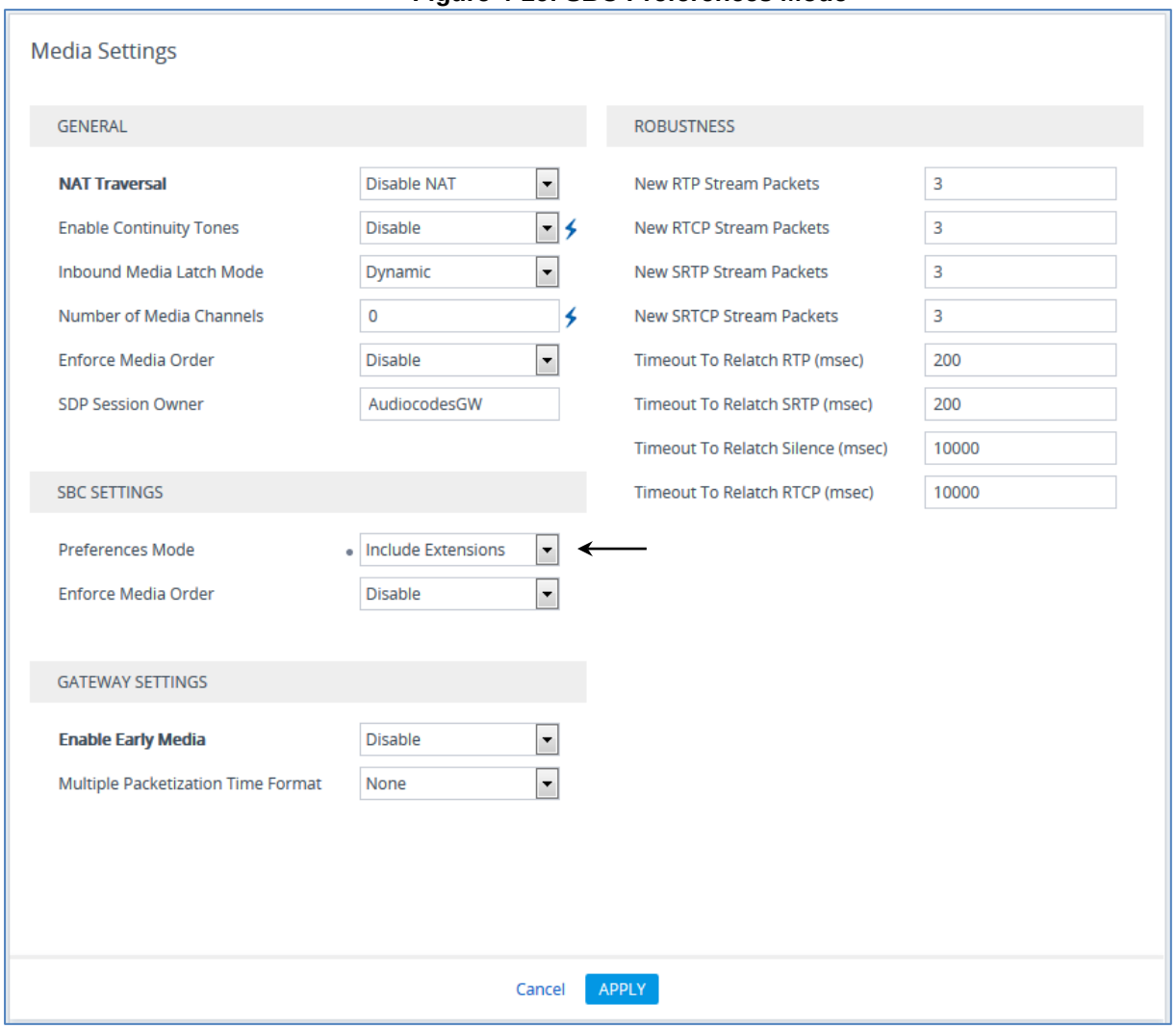
Parameter	Value
Index	**0**
Coder	**G.729**

Figure 4-15: Configuring Allowed Coders for ITSP SIP Trunk



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

Figure 4-16: SBC Preferences Mode



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

4.7 Step 7: Configure IP Profiles

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

In the example scenario, IP Profiles need to be configured for the following IP entities:

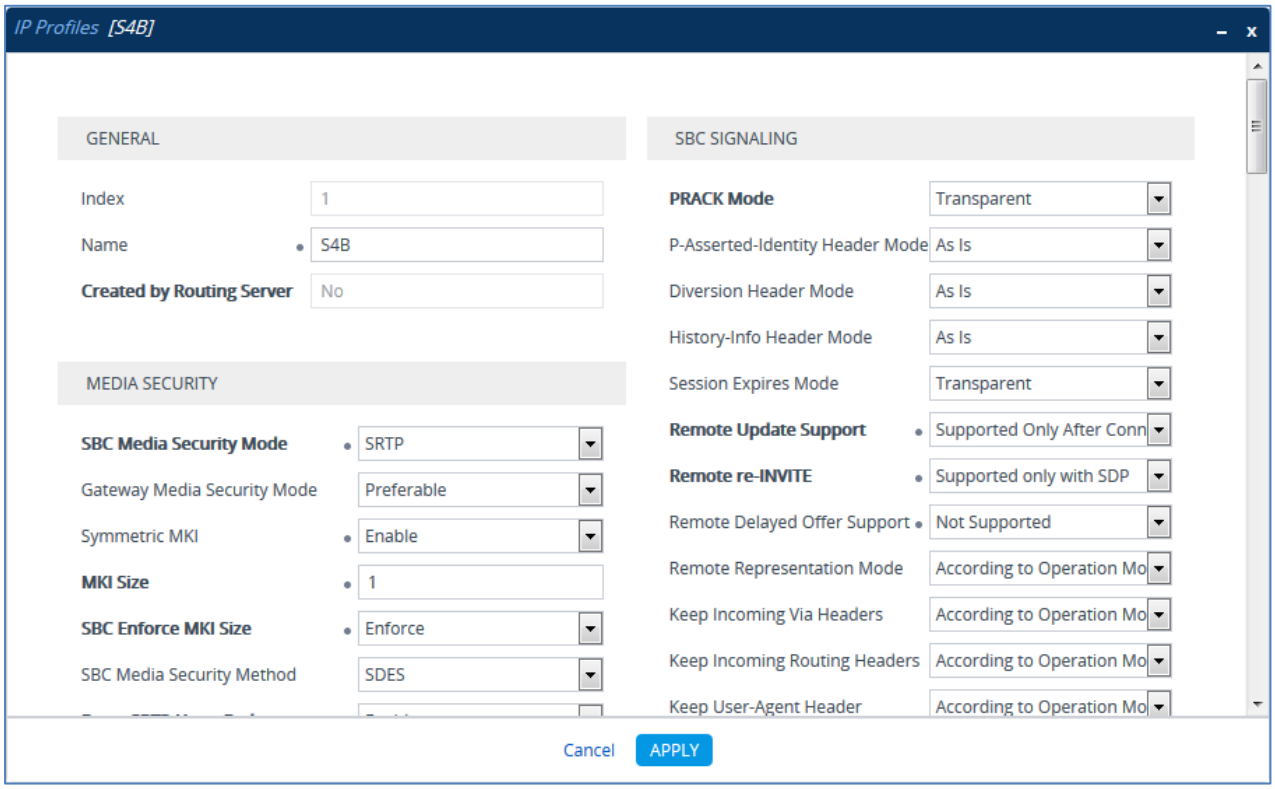
- Microsoft Skype for Business CCE - to operate in secure mode using SRTP and TLS
- ITSP SIP trunk - to operate in non-secure mode using RTP and UDP

➢ **To configure IP Profile for the CCE:**

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 CCE does not send RTP immediately to remote side when it sends a SIP 18x response)
SBC Media	
Extension Coders Group	**AudioCodersGroups_1**
SBC Signaling	
Remote Update Support	**Supported Only After Connect**
Remote re-INVITE Support	**Supported Only With SDP**
Remote Delayed Offer Support	**Not Supported**
SBC Forward and Transfer	
Remote REFER Mode	**Handle Locally** (required, as CCE does not support receipt of SIP REFER)
Remote 3xx Mode	**Handle Locally** (required, as CCE does not support receipt of SIP 3xx responses)

Figure 4-17: Configuring IP Profile for CCE



3. Click **Apply**.

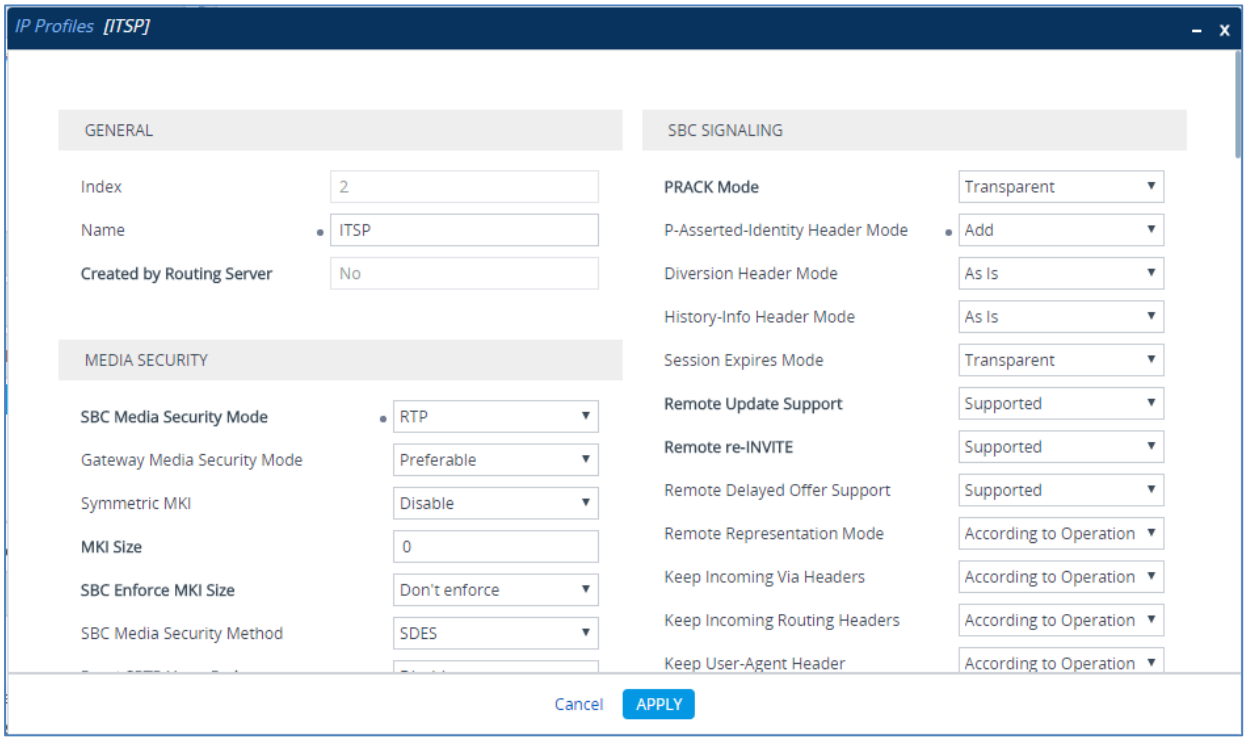
➢ **To configure an IP Profile for the ITSP SIP Trunk:**

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

Parameter	Value
General	
Index	**2**
Name	**ITSP**
Media Security	
SBC Media Security Mode	**RTP**
SBC Early Media	
Remote Can Play Ringback	**No** (required, as CCE does not provide a ringback tone for incoming calls)
SBC Media	
Extension Coders Group	**AudioCodersGroups_2**
Allowed Audio Coders	**ITSP Allowed Coders**
Allowed Coders Mode	**Preference** (lists Allowed Coders first and then original coders in received SDP offer)
SBC Signaling	
P-Asserted-Identity Header Mode	**Add** (required for anonymous calls)

Note: The SIP Trunk's IP Profile depends on the SIP Trunk behavior. Refer to the explanations of the IP Profile parameters in the *SBC User's Manual* in order to configure the profile according to SIP Trunk behavior.

Figure 4-18: Configuring IP Profile for ITSP SIP Trunk



2. Click **Apply**.

4.8 Step 8: Configure IP Groups

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

In the example scenario, IP Groups must be configured for the following IP entities:

- CCE (Mediation Server) located on LAN
- ITSP SIP Trunk located on WAN

➢ **To configure IP Groups:**

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

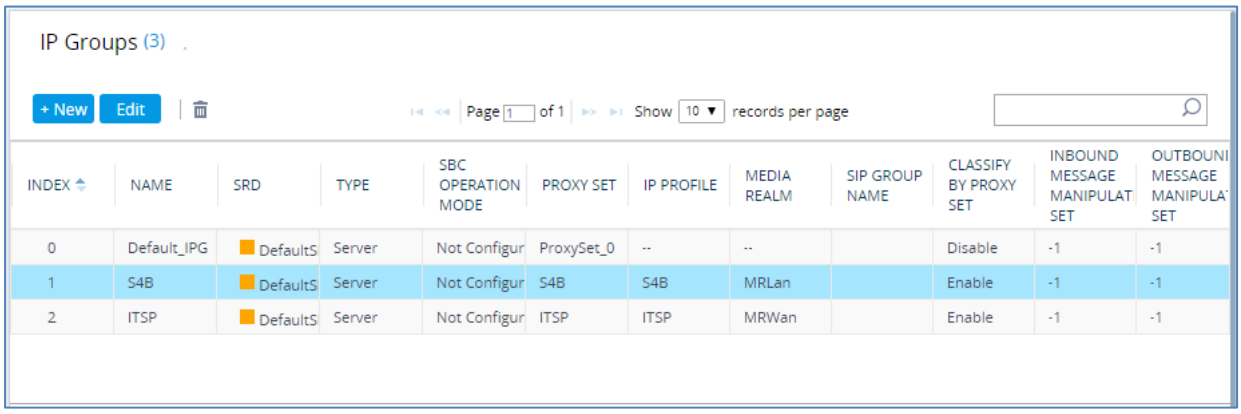
Parameter	Value
Index	**1**
Name	**S4B**
Type	**Server**
Proxy Set	**S4B**
IP Profile	**S4B**
Media Realm	**MRLan**
SIP Group Name	(according to ITSP requirement)

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

Parameter	Value
Index	**2**
Name	**ITSP**
Topology Location	**Up**
Type	**Server**
Proxy Set	**ITSP**
IP Profile	**ITSP**
Media Realm	**MRWan**
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 (3)

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

INDEX	NAME	SRD	TYPE	SBC OPERATION MODE	PROXY SET	IP PROFILE	MEDIA REALM	SIP GROUP NAME	CLASSIFY BY PROXY SET	INBOUND MESSAGE MANIPULAT SET	OUTBOUNI MESSAGE MANIPULA SET
0	Default_IPG	DefaultS	Server	Not Configur	ProxySet_0	--	--		Disable	-1	-1
1	S4B	DefaultS	Server	Not Configur	S4B	S4B	MRLan		Enable	-1	-1
2	ITSP	DefaultS	Server	Not Configur	ITSP	ITSP	MRWan		Enable	-1	-1

4.9 Step 9: SIP TLS Connection Configuration

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

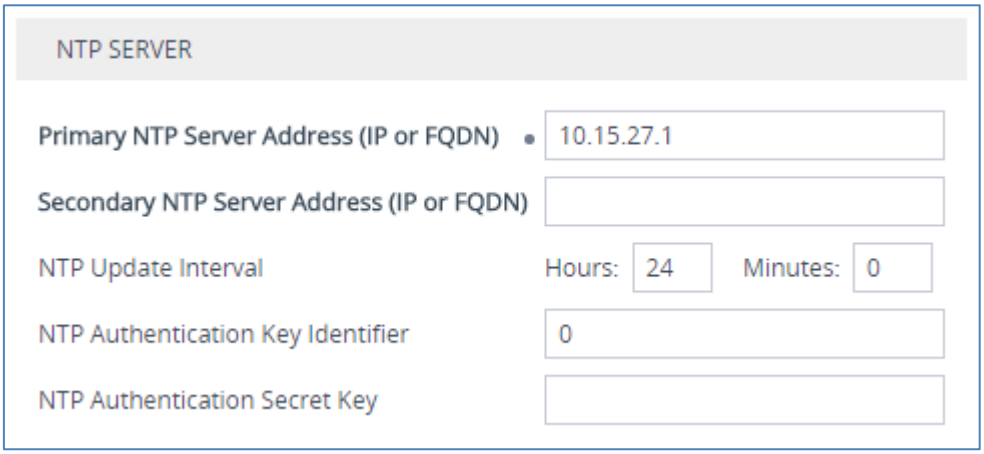
4.9.1 Step 9a: Configure the NTP Server Address

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

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

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

Figure 4-20: Configuring NTP Server Address



3. Click **Apply**.

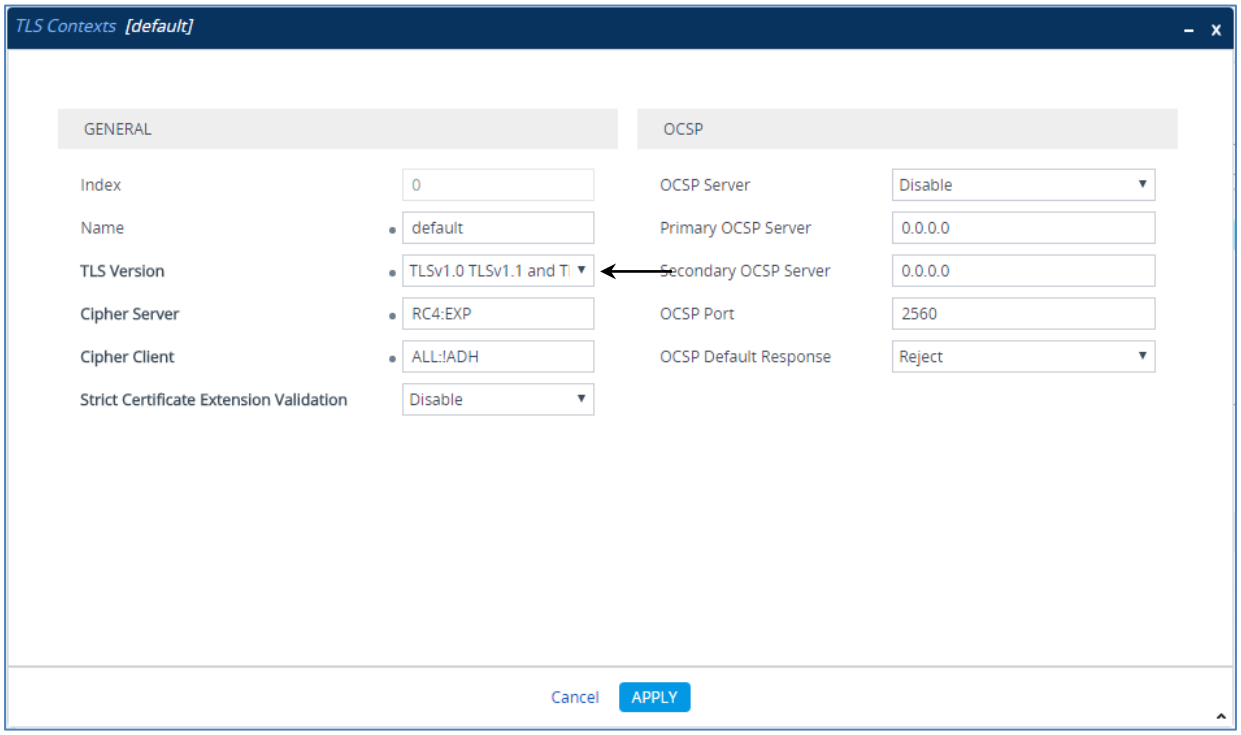
4.9.2 Step 9b: Configure the TLS version

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

➢ **To configure the TLS version:**

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

Figure 4-21: Configuring TLS version



4. Click **Apply**.

4.9.3 Step 9c: Configure a Certificate

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

Note: The CCE Wizard supports applying Certificates to the SBC (refer to *LTRT-28086 Mediant Appliance for Microsoft Skype for Business CCE Installation Manual Ver. 2.0.2* Section "Set the GW/SBC Certificate").

- Reset the E-SBC with a burn to flash for your settings to take effect (see Section 4.17 on page 54).

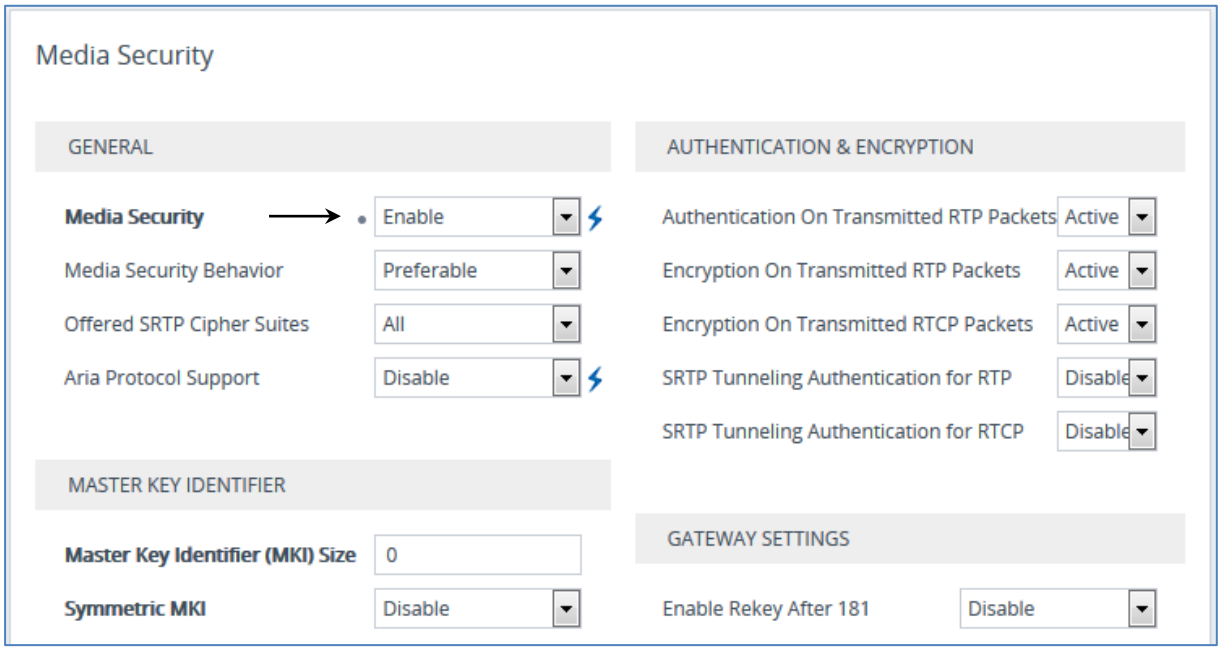
4.10 Step 10: Configure SRTP

This step describes how to configure media security. If you configure the Mediation Server to use SRTP, you need to configure the E-SBC to operate in the same manner. Note that SRTP was enabled for the CCE when you configured an IP Profile for Skype for Business CCE (see Section 4.6 on page 28).

➢ **To configure media security:**

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

Figure 4-22: 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 4.17 on page 54).

4.11 Step 11: Configure Maximum IP Media Channels

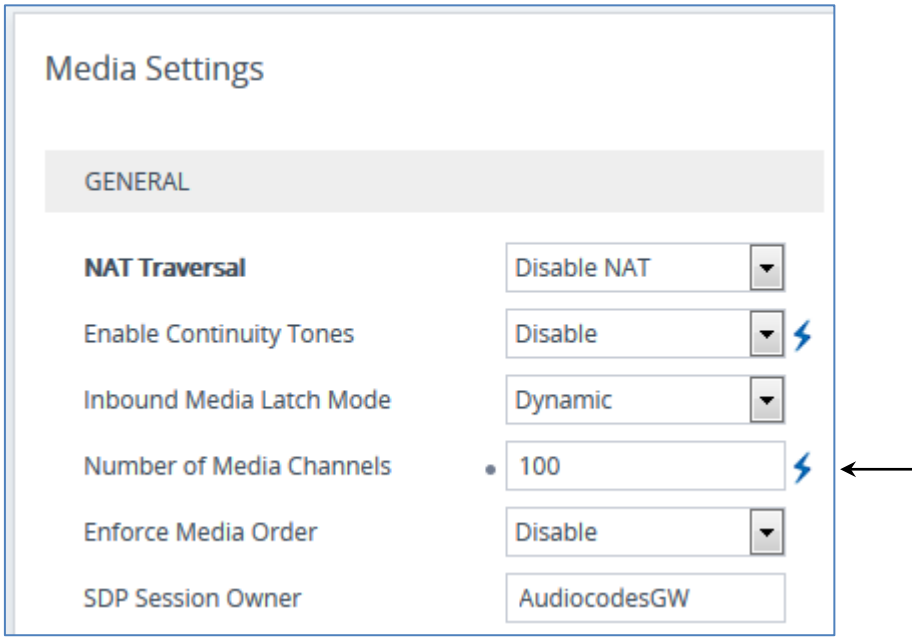
This step describes how to configure the maximum number of required IP media channels. The number of media channels represents the number of DSP channels that the E-SBC allocates to call sessions.

Note: This step is mandatory **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 4-23: 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., 10**0**).
3. Click **Apply**.
4. Reset the E-SBC with a burn to flash for your settings to take effect (see Section 4.17 on page 54).

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

This step describes how to configure IP-to-IP call routing rules. These rules define the routes for forwarding SIP messages (e.g., INVITE) received from one IP entity to another. The E-SBC selects the rule whose configured input characteristics (e.g., IP Group) match those of the incoming SIP message. If the input characteristics do not match the first rule in the table, they are compared to the second rule, and so on, until a matching rule is located. If no rule is matched, the message is rejected. The routing rules use the configured IP Groups (as configured in Section 4.8 on page 35,) to denote the source and destination of the call.

In the example scenario, the following IP-to-IP routing rules need to be configured to route calls between Skype for Business CCE (LAN) and ITSP SIP Trunk (DMZ):

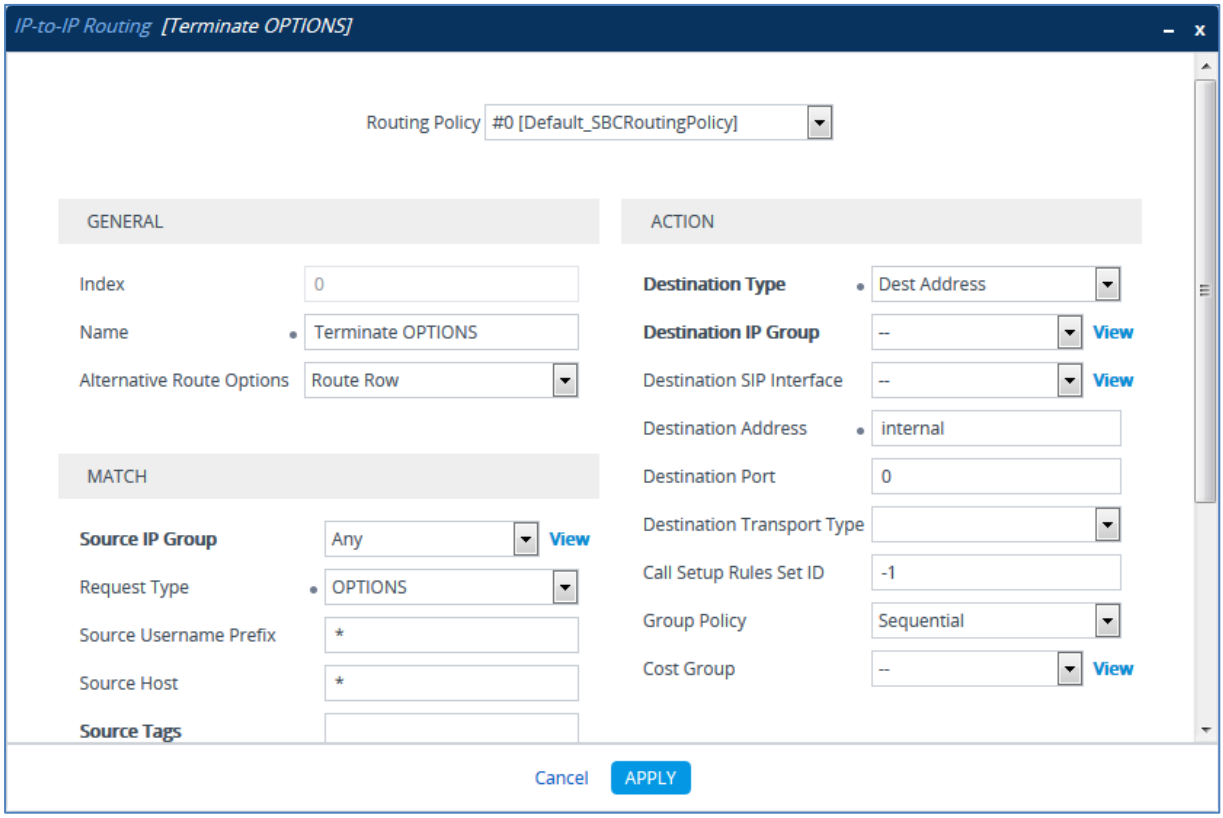
- Terminate SIP OPTIONS messages on the E-SBC that are received from the both LAN and DMZ
- Calls from CCE to ITSP SIP Trunk
- Calls from ITSP SIP Trunk to CCE

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

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

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

Figure 4-24: Configuring IP-to-IP Routing Rule for Terminating SIP OPTIONS



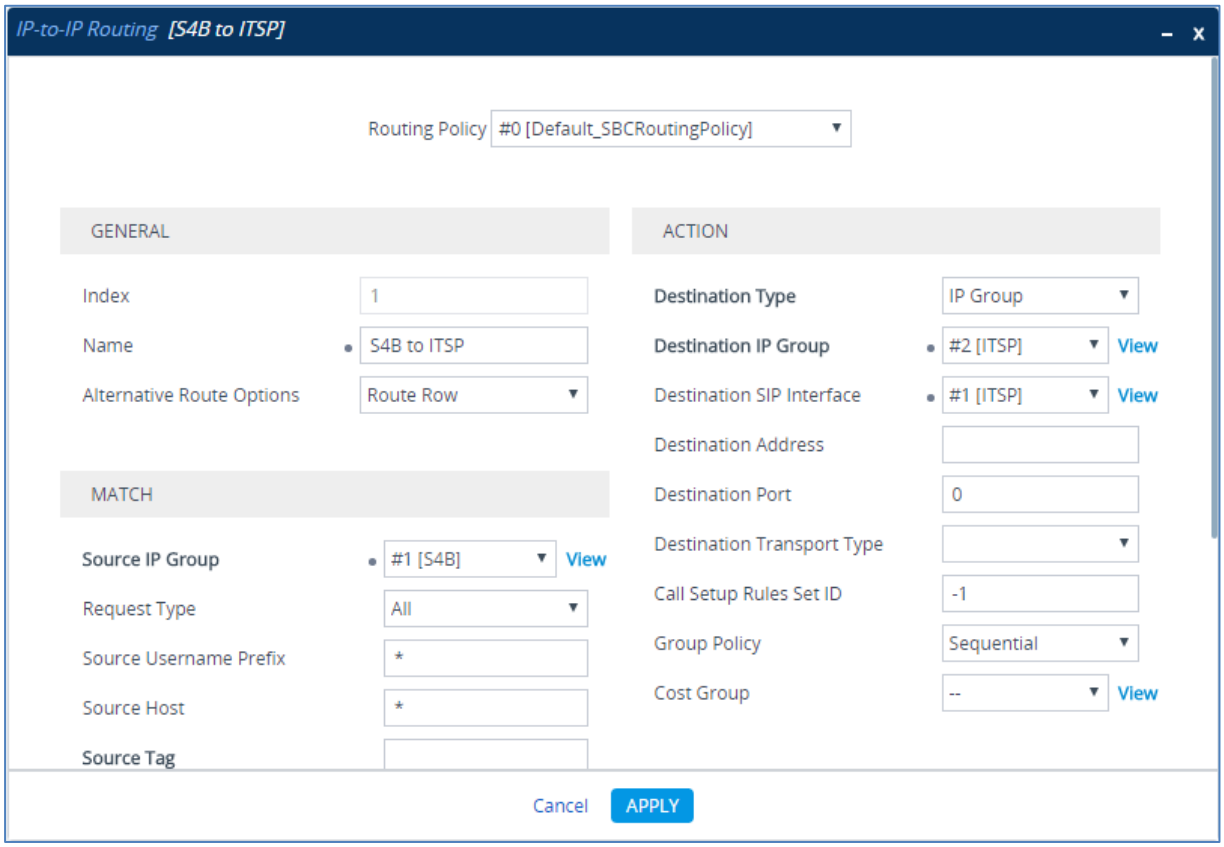
b. Click **Apply**.

3. Configure a rule to route calls from CCE to ITSP SIP Trunk:

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

Parameter	Value
Index	**1**
Name	**S4B to ITSP** (arbitrary descriptive name)
Source IP Group	**S4B**
Destination Type	**IP Group**
Destination IP Group	**ITSP**
Destination SIP Interface	**ITSP**

Figure 4-25: Configuring IP-to-IP Routing Rule for S4B to ITSP

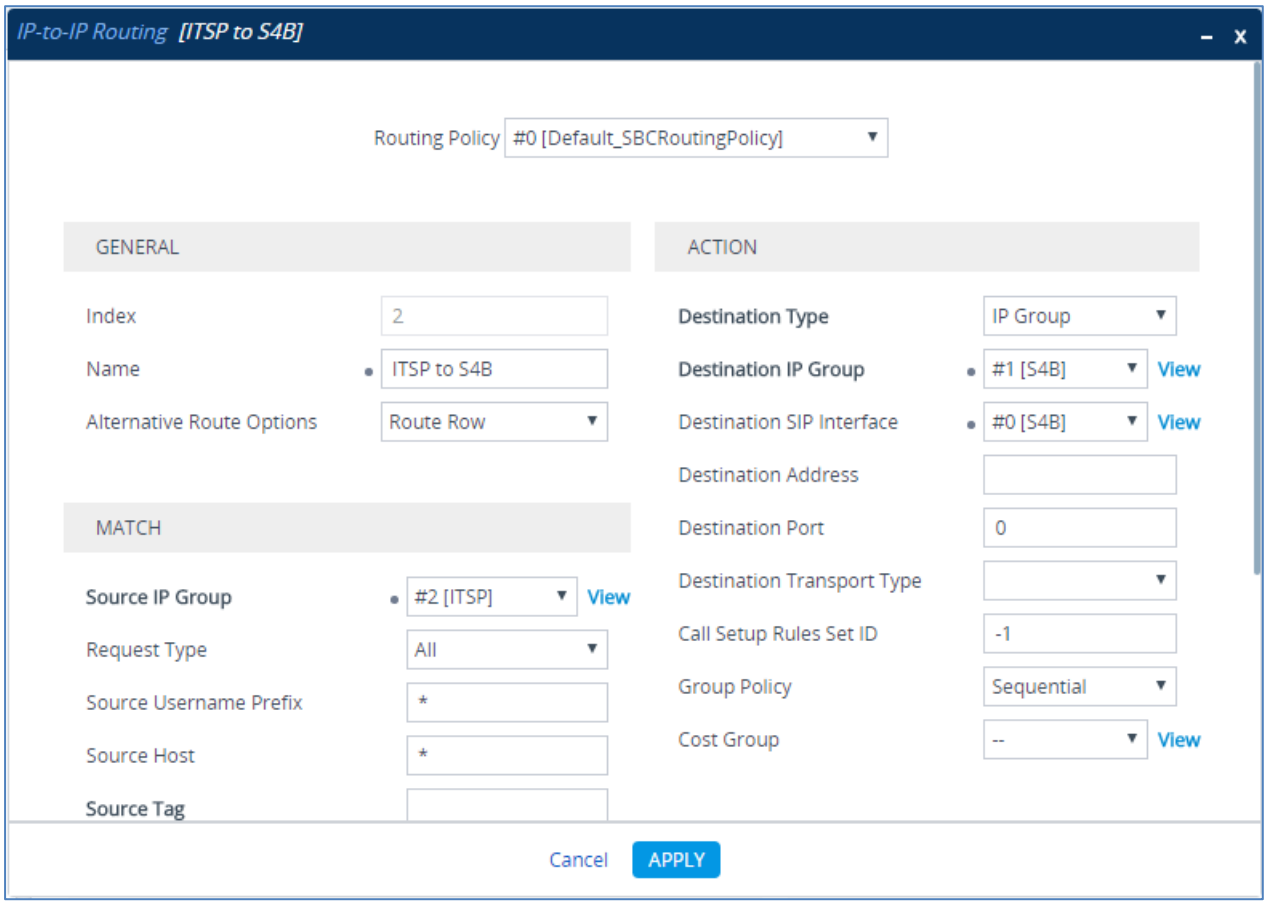


b. Click **Apply**.

4. Configure rule to route calls from ITSP SIP Trunk to CCE:
   a. Click **New**, and then configure the parameters as follows:

Parameter	Value
Index	**2**
Name	**ITSP to S4B** (arbitrary descriptive name)
Source IP Group	**ITSP**
Destination Type	**IP Group**
Destination IP Group	**S4B**
Destination SIP Interface	**S4B**

Figure 4-26: Configuring IP-to-IP Routing Rule for ITSP to S4B



   b. Click **Apply.**

The configured routing rules are shown in the figure below:

Figure 4-27: Configured IP-to-IP Routing Rules in IP-to-IP Routing Table

IP-to-IP Routing (3)

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

INDEX	NAME	ROUTING POLICY	ALTERNATIV ROUTE OPTIONS	SOURCE IP GROUP	REQUEST TYPE	SOURCE USERNAME PREFIX	DESTINATIO USERNAME PREFIX	DESTINATIO TYPE	DESTINATIO IP GROUP	DESTINATIO SIP INTERFACE	DESTINATIO ADDRESS
0	Terminate O	Default_SBCI	Route Row	Any	OPTIONS	*	*	Dest Addres	--	--	internal
1	S4B to ITSP	Default_SBCI	Route Row	S4B	All	*	*	IP Group	ITSP	ITSP	
2	ITSP to S4B	Default_SBCI	Route Row	ITSP	All	*	*	IP Group	S4B	S4B	

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

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

This step describes how to configure IP-to-IP manipulation rules. These rules manipulate the SIP Request-URI user part (source or destination number). The manipulation rules use the configured IP Groups (as configured in Section 4.8 on page 35) to denote the source and destination of the call.

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

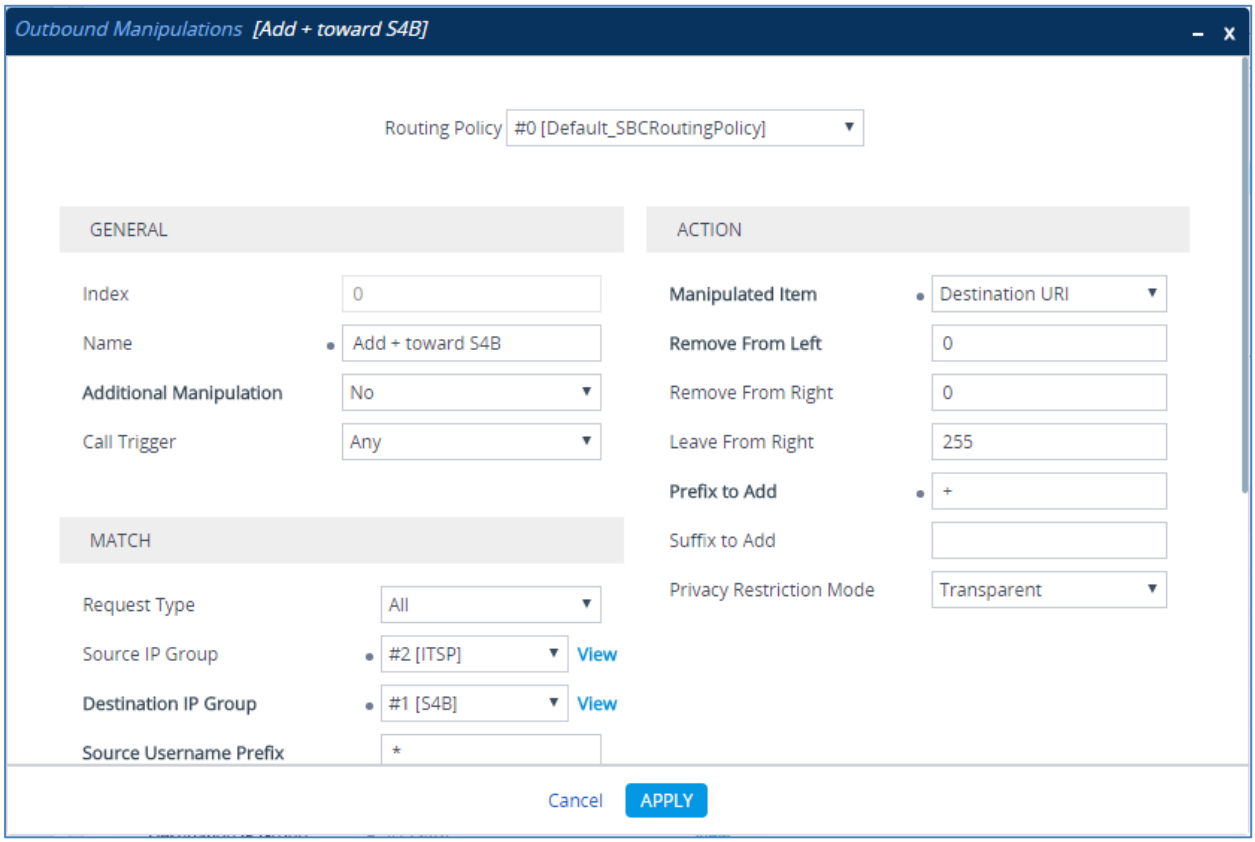
For this example scenario, a manipulation is configured to add the "+" (plus sign) to the destination number for calls from the ITSP SIP Trunk IP Group to the CCE IP Group for any destination username prefix.

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

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

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

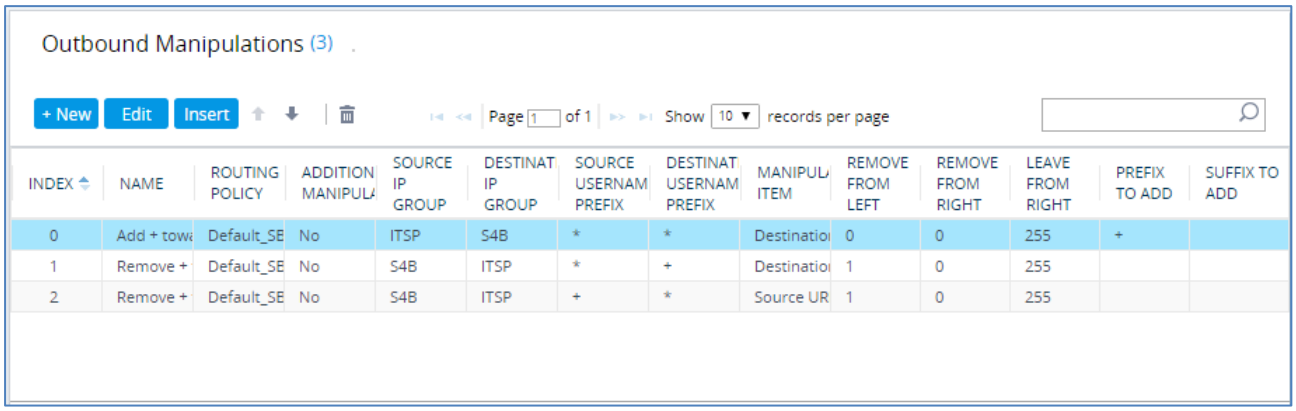
Figure 4-28: Configuring IP-to-IP Outbound Manipulation Rule



3. Click **Apply**.

The figure below shows an example of configured IP-to-IP outbound manipulation rules for calls between CCE IP Group and ITSP SIP Trunk IP Group:

Figure 4-29: Example of Configured IP-to-IP Outbound Manipulation Rules



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

4.14 Step 14: Configure Message Manipulation Rules

This step 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.

See an example below of a message manipulation rule configuration; use the *SBC User's Manual* for detailed instructions on how to configure message manipulation rules according to your requirements.

In the example scenario, the configured manipulation rule replaces the user part of the SIP From Header with the value from the SIP History-Info Header.

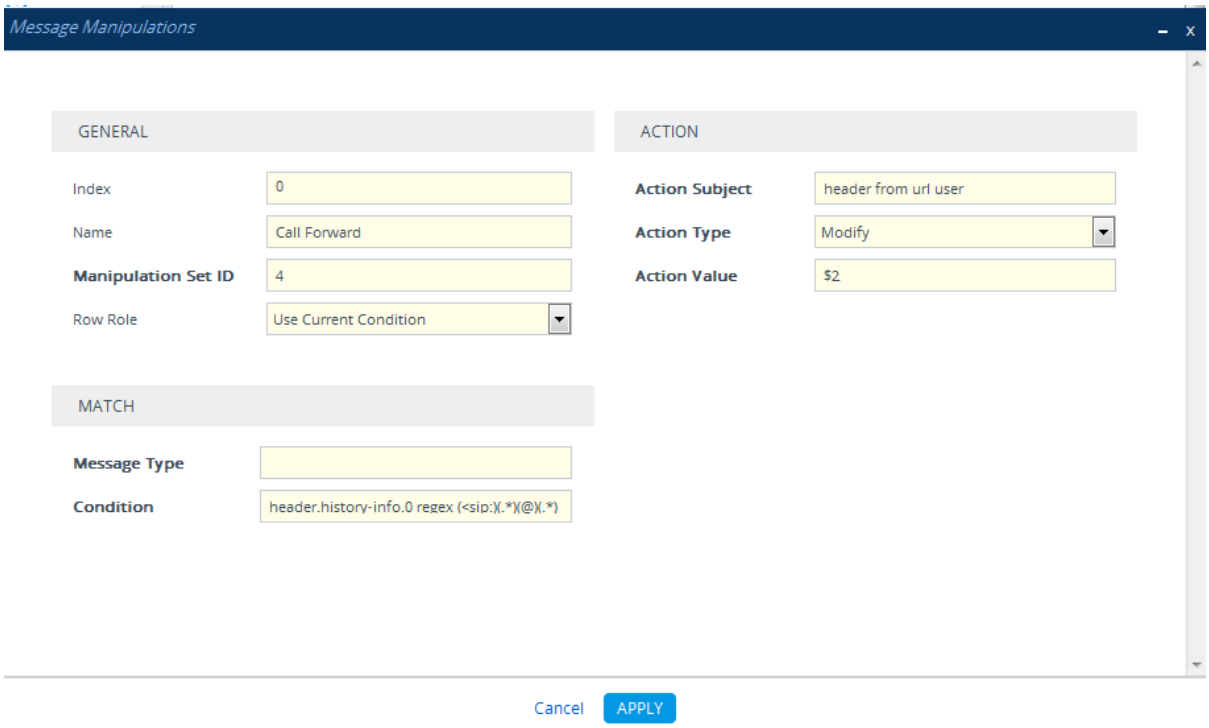
Note: The History-info header isn't set by default, in order to use the below MMS need to enable *ForwardCallHistory* parameter on the CCE Trunk Configuration.

➢ **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 ITSP SIP Trunk. This rule applies to messages sent to the ITSP SIP Trunk IP Group in a call forward scenario.

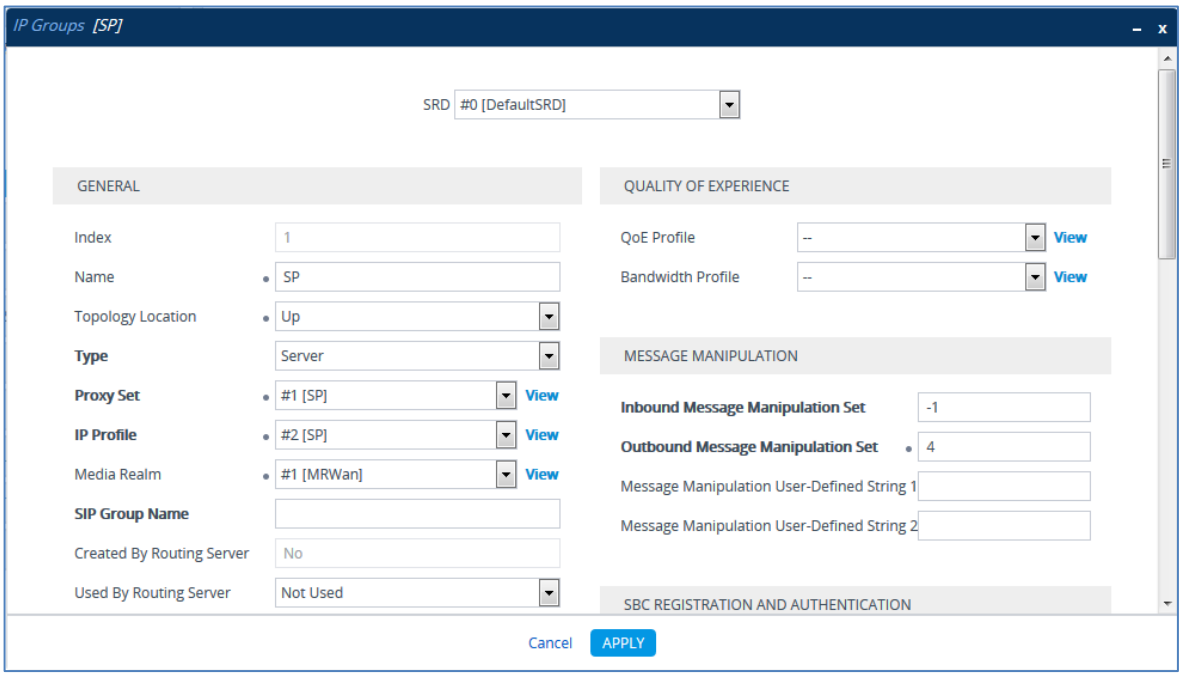
Parameter	Value
Index	**0**
Name	**Call Forward**
Manipulation Set ID	**4**
Message Type	**invite.request**
Condition	**header.history-info.0 regex (<sip:)(.*)(@)(.*)**
Action Subject	**header.from.url.user**
Action Type	**Modify**
Action Value	**$2**

Figure 4-30: Configuring SIP Message Manipulation Rule 0 (for ITSP SIP Trunk)



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

Figure 4-31: Assigning Manipulation Set 4 to the ITSP SIP Trunk IP Group



   d. Click **Apply**.

4.15 Step 15: Configure Registration Accounts

This step describes how to configure SIP registration accounts. This is required so that the E-SBC can register with the ITSP SIP Trunk on behalf of CCE. The ITSP SIP Trunk requires registration and authentication to provide service.

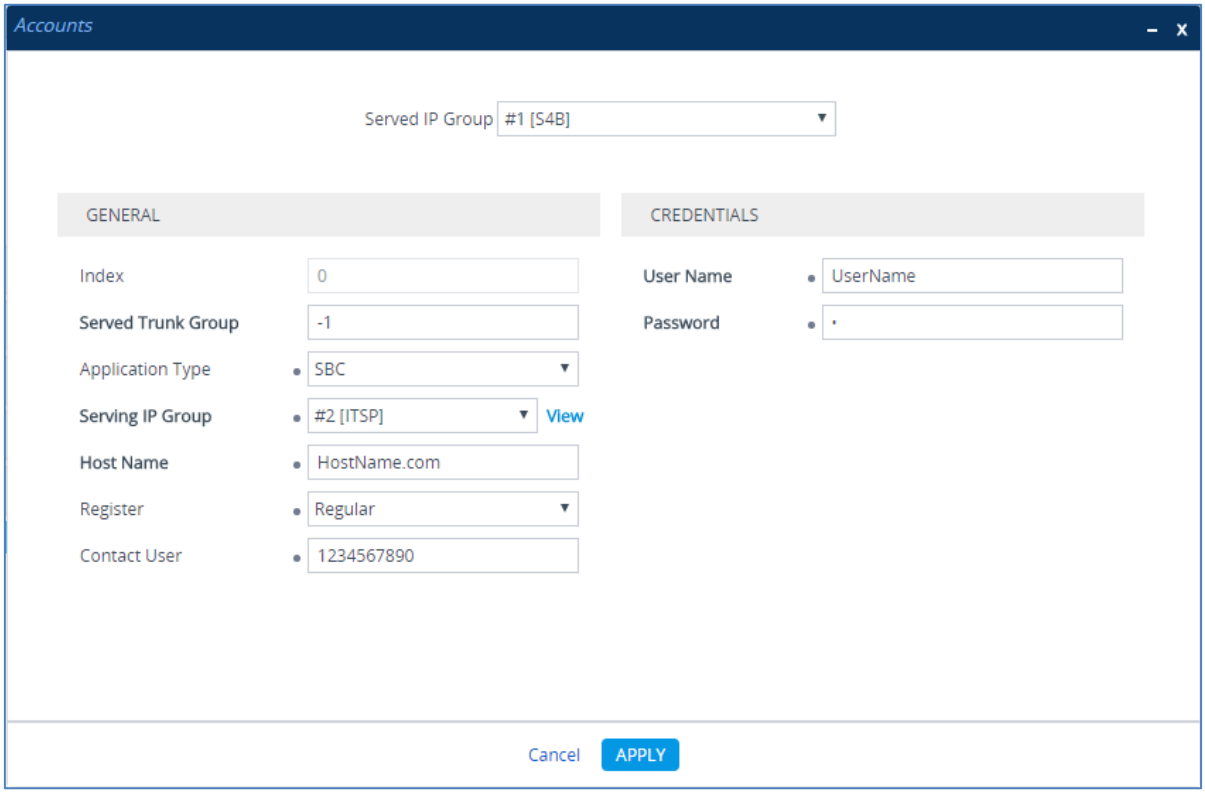
In the interoperability test topology, the Served IP Group is CCE IP Group and the Serving IP Group is ITSP SIP Trunk IP Group.

➢ **To configure a registration account:**

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

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

Figure 4-32: Configuring a SIP Registration Account



4. Click **Apply**.

4.16 Step 16: Miscellaneous Configuration

This section describes miscellaneous E-SBC configuration.

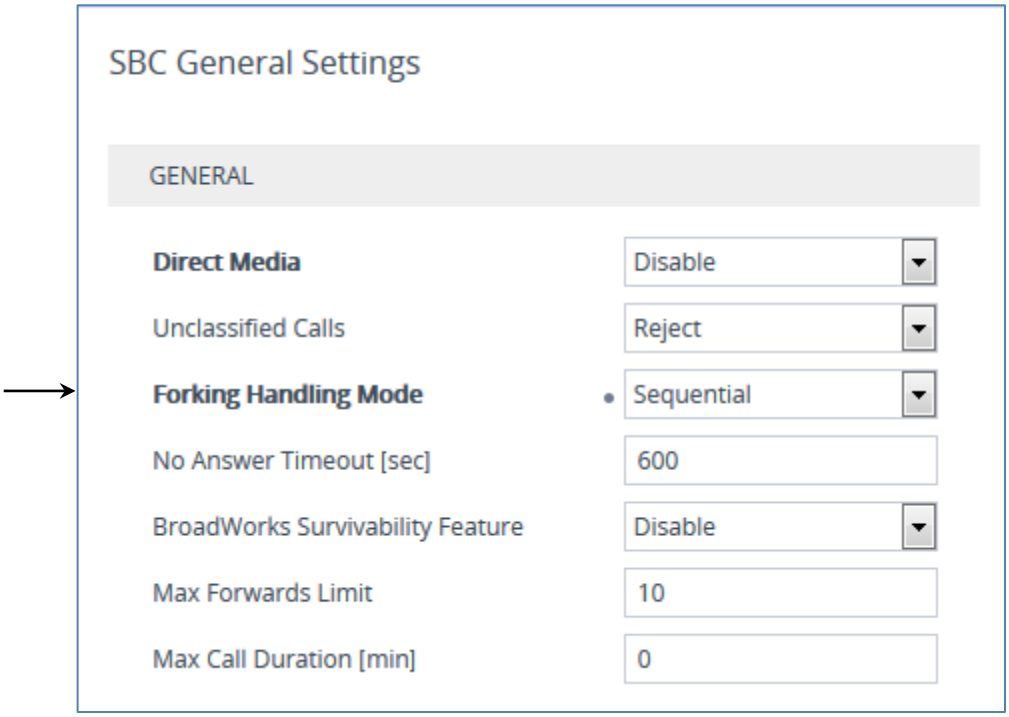
4.16.1 Step 16a: Configure Call Forking Mode

This step describes how to configure the E-SBC's handling of SIP 18x responses received for call forking of INVITE messages. In the example scenario, 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 CCE 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 4-33: Configuring Forking Mode



3. Click **Apply**.

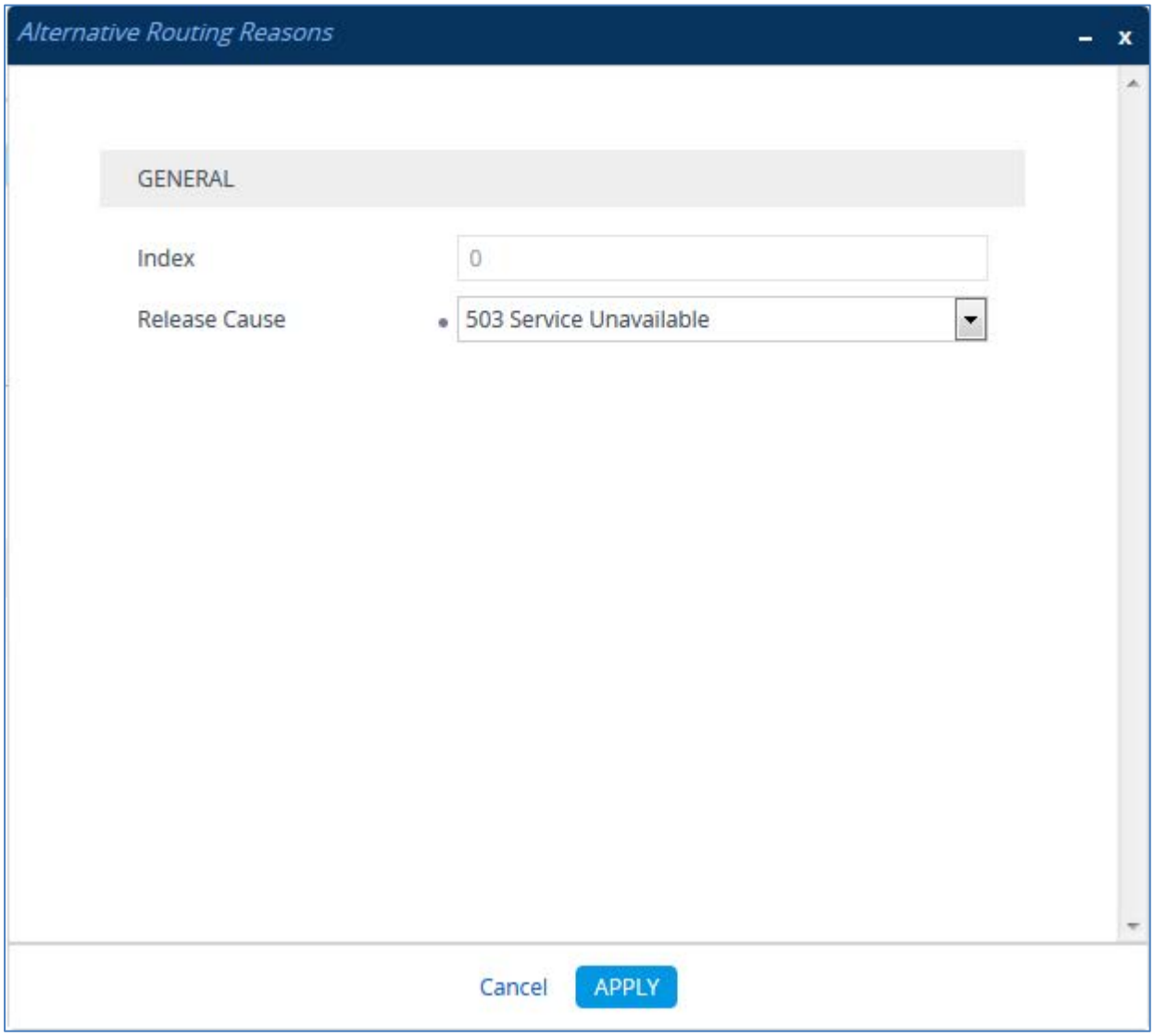
4.16.2 Step 16b: Configure SBC Alternative Routing Reasons

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

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

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

Figure 4-34: SBC Alternative Routing Reasons Table



4. Click **Apply**.

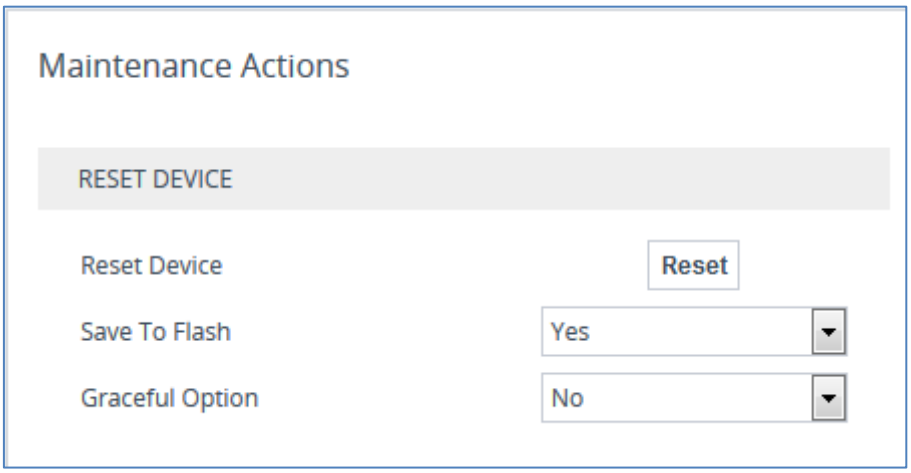
4.17 Step 17: Reset the E-SBC

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

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

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

Figure 4-35: 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.

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