Configuration Note

AudioCodes Professional Services – Interoperability Lab

Cisco Unified Communications Manager Ver.12 and Amazon Chime Voice Connector using AudioCodes Mediant™ SBC

Version 7.2

amazon Chime

CallManager

audiocodes

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1 Introduction

This Configuration Note describes how to set up the AudioCodes Session Border Controller (hereafter, referred to as *SBC*) for interworking between AWS Chime's Voice Connector and Cisco CUCM environment.

You can also use AudioCodes' SBC Wizard tool to automatically configure the 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 AWS Chime Partners who are responsible for installing and configuring AWS Chime's Voice Connector and Cisco CUCM for enabling VoIP calls using AudioCodes SBC.

1.2 About AudioCodes SBC Product Series

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

The 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 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 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 SBC Version

Table 2-1: AudioCodes SBC Version

SBC Vendor	AudioCodes
Models	▪ Mediant 500 Gateway & 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, VE and CE)
Software Version	7.20A.252.011
Protocol	▪ SIP/UDP or SIP/TCP or SIP/TLS (to the AWS Chime Voice Connector) ▪ SIP/TCP (to the Cisco CUCM)
Additional Notes	None

2.2 AWS Chime Voice Connector Version

Table 2-2: AWS Chime Version

Vendor/Service Provider	AWS Chime
SSW Model/Service	
Software Version	
Protocol	SIP
Additional Notes	None

2.3 IP-PBX Version

Table 2-3: IP-PBX Version

Vendor	Cisco
Model	CUCM
Software Version	12.0.1
Protocol	SIP
Additional Notes	None

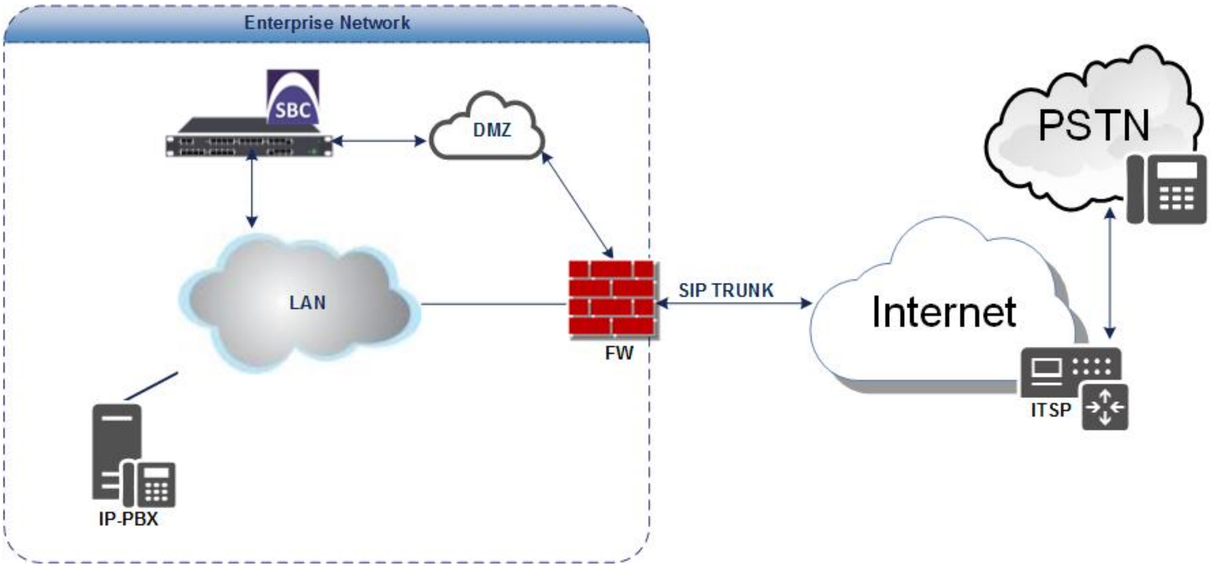
2.4 Interoperability Test Topology

The interoperability testing between AudioCodes SBC and AWS Chime Voice Connector with CUCM v12 was done using the following topology setup:

- Enterprise deployed with Cisco CUCM IP-PBX 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 AWS Chime's Voice Connector service.
- AudioCodes SBC is implemented to interconnect between the Enterprise LAN and the Voice Connector.
    - **Session:** Real-time voice session using the IP-based Session Initiation Protocol.
    - **Border:** IP-to-IP network border between IP-PBX network in the Enterprise LAN and AWS Chime's Voice Connector located in the public network.

The figure below illustrates this interoperability test topology:

Figure 2-1: Interoperability Test Topology between SBC and Cisco CUCM with AWS Chime Voice Connector



2.4.1 Environment Setup

The interoperability test topology includes the following environment setup:

Table 2-4: Environment Setup

Area	Setup
Network	▪ Cisco CUCM environment is located on the Enterprise's LAN ▪ AWS Chime Voice Connector is located on the WAN
Signaling Transcoding	▪ Cisco CUCM operates with SIP-over-TCP transport type ▪ AWS Chime Voice Connector operates with SIP-over-UDP or SIP-over-TCP or SIP-over-TLS transport types
Codecs Transcoding	▪ Cisco CUCM supports G.711A-law and G.711U-law coders ▪ AWS Chime Voice Connector supports G.711U-law coder
Media Transcoding	▪ Cisco CUCM operates with RTP media type ▪ AWS Chime Voice Connector operates with RTP or SRTP media types

2.4.2 Known Limitations

The following limitation was observed in the interoperability tests done for the AudioCodes SBC interworking between Cisco CUCM v.12 and AWS Chime's Voice Connector:

- For inbound calling from AWS Chime Voice Connector to an IP-PBX, where the phone number being called isn't assigned an origination route, a busy/announcement should be heard however is not heard.

Amazon Chime Team is working on fixing this issue in the next release.

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3 Configuring Cisco CUCM

This section describes how to configure the Cisco Unified Communications Manager.

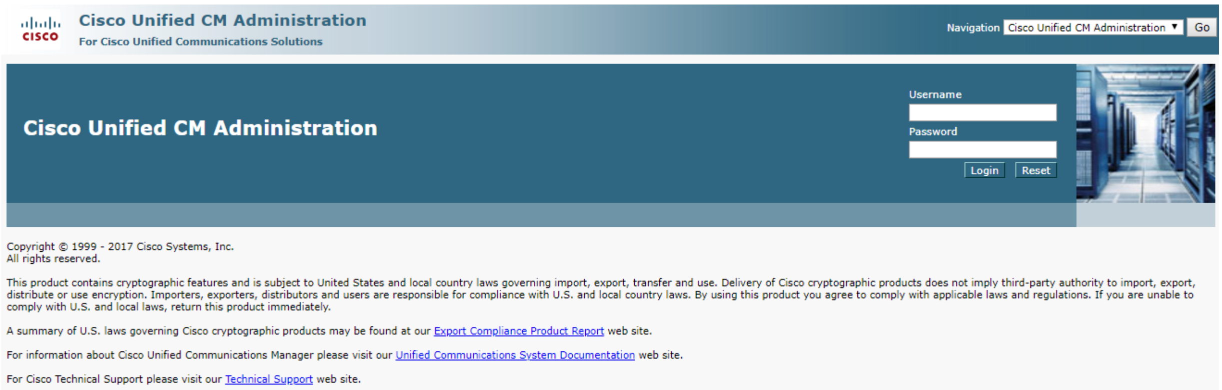
3.1 Log in to Cisco Unified Communications Manager

The procedure below describes how to log in to the Cisco CUCM Administration interface.

➢ **To log in to the Cisco Unified CM Administration interface:**

1. Log in to the Cisco Unified CM Administration by entering the IP address of the Cisco Unified Communications Manager (CUCM) in the Web browser address field.

Figure 3-1: Cisco Unified CM Administration



2. In the 'Username' field, enter the user name.
3. In the 'Password' field, enter the password.
4. Click **Login**.

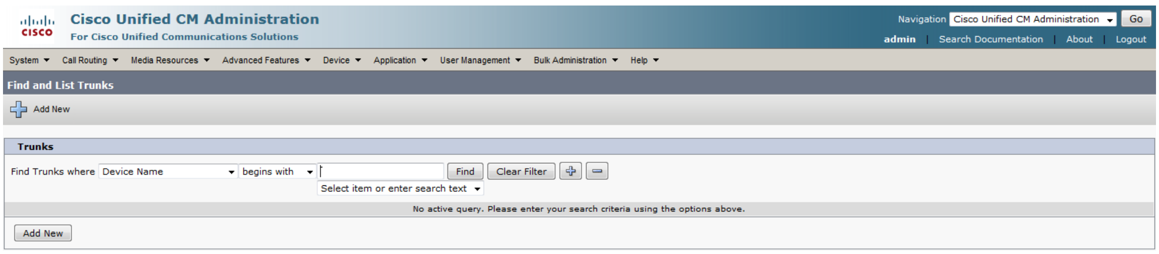
3.2 Create a New Trunk

This section describes how to create a new trunk.

➢ **To create a new trunk:**

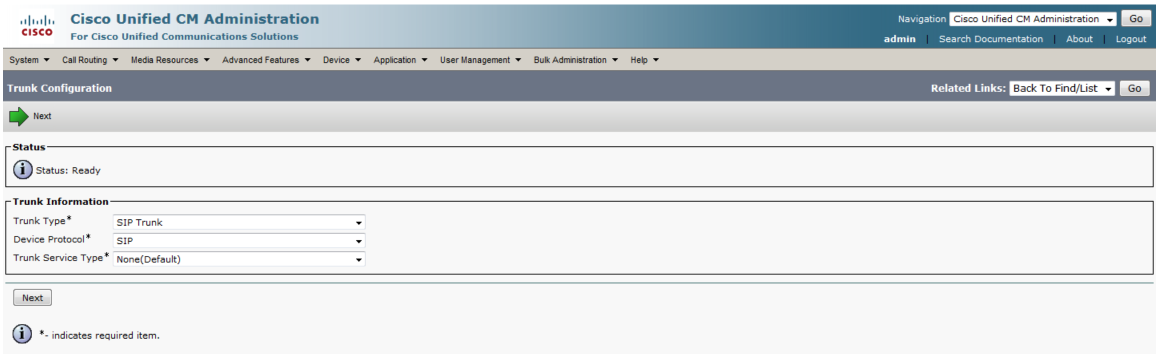
1. From the **Device** menu drop-down list, select **Trunk**.
2. Click **Add New**.

Figure 3-2: Trunk page



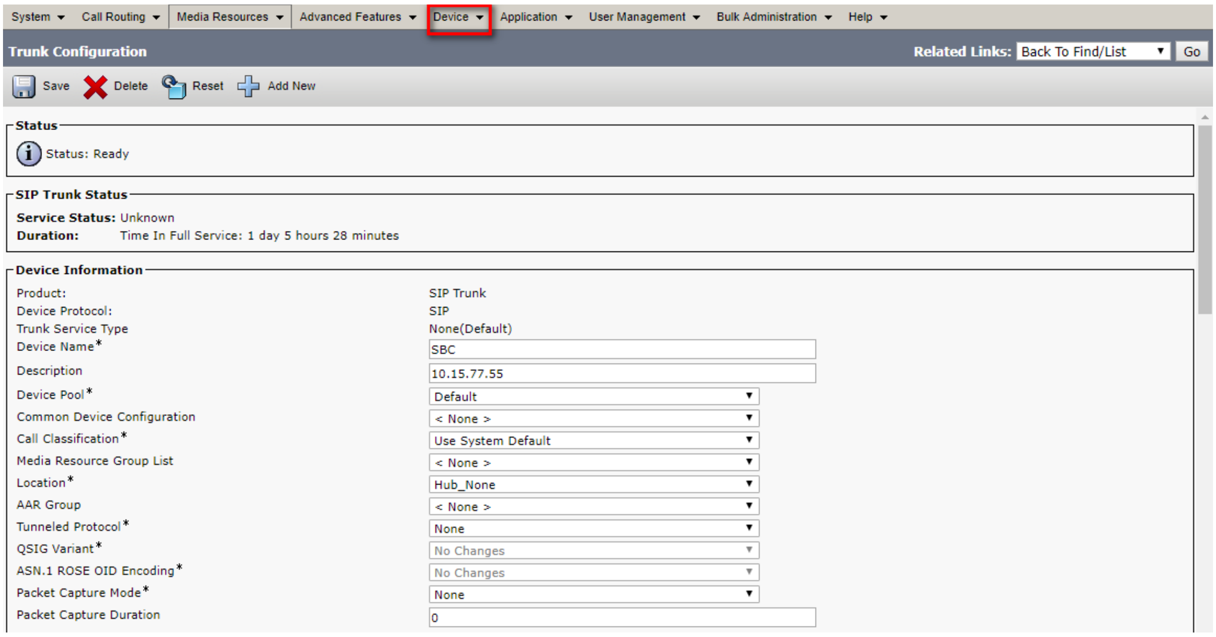
3. Select Trunk Type – **SIP Trunk**.
4. Click **Next**.

Figure 3-3: Create Trunk Page



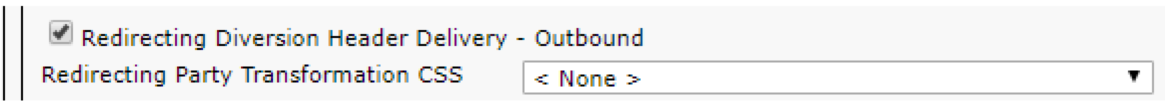
5. In the **Device Name** field, enter a unique SIP Trunk name and optionally provide a description.
6. From the **Device Pool** drop-down list, select a device pool.

Figure 3-4: SIP Trunk Settings Page



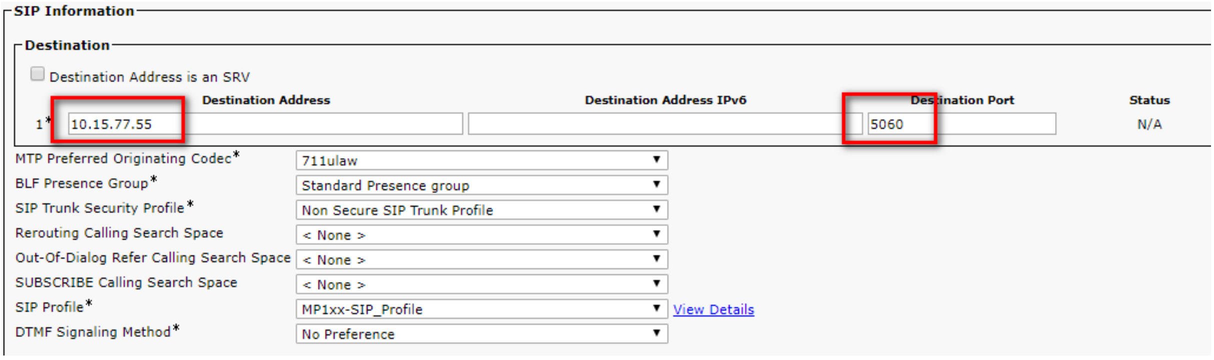
7. Select the 'Redirecting Diversion Header Delivery – Outbound' check box.

Figure 3-5: Redirecting Diversion Header Delivery



8. Enter the Destination Address and Destination Port of the AudioCodes SBC.

Figure 3-6: SIP Information Section



9. From the **SIP Trunk Security** drop-down list, select a profile.
10. From the **SIP Profile** drop-down list, select a profile.
11. Click **Save**.

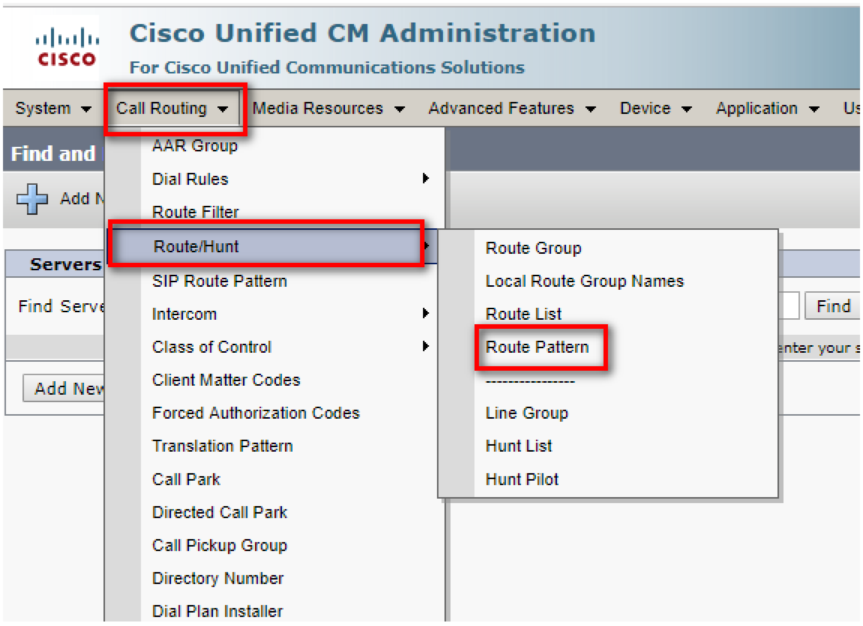
3.3 Create a New Route Pattern

This section describes how to create a new route pattern.

➢ **To create new Route Pattern:**

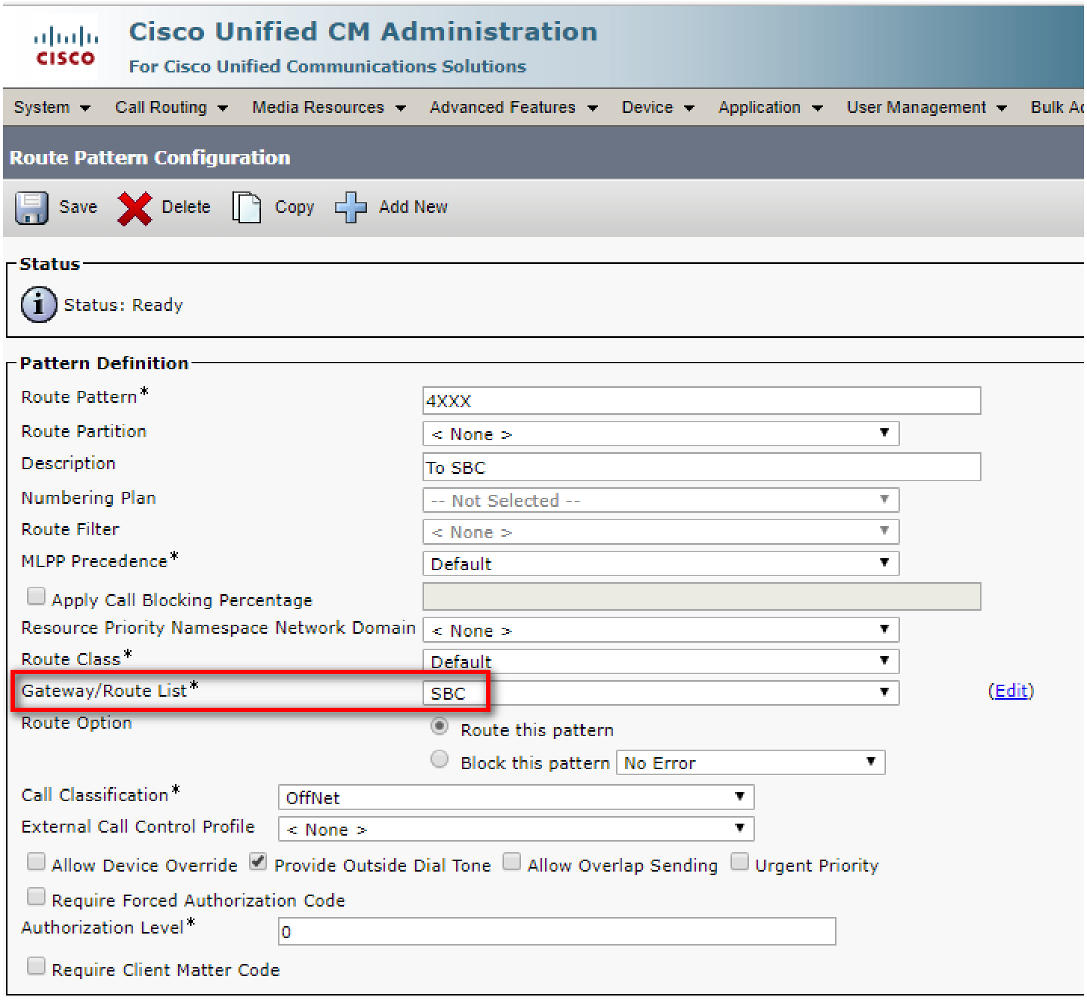
1. From the **Call Routing** menu drop-down list, go to the **Route/Hunt** menu and select **Route Pattern**.

Figure 3-7: Route Pattern page



2. Click **Add New**.
3. Enter a Route Pattern according to schema (optionally provide a description).
4. From the **Gateway/Route List** drop-down list, select the SIP Trunk device name.

Figure 3-8: Create Route Pattern Page



5. Click **Save**.

Figure 3-9: Added Route Pattern

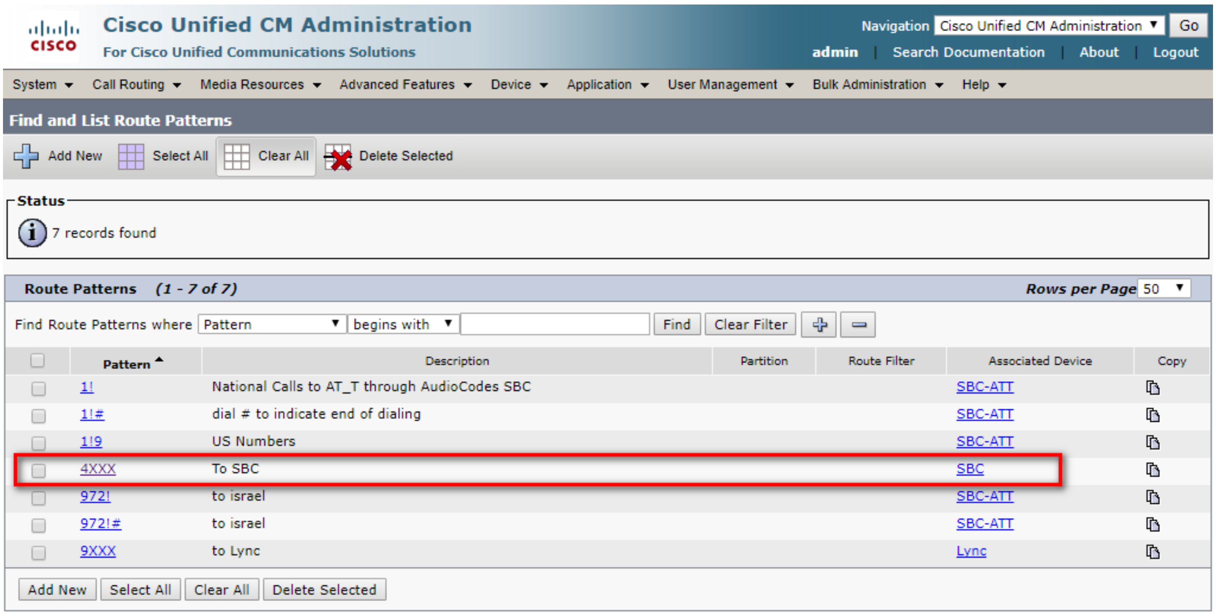
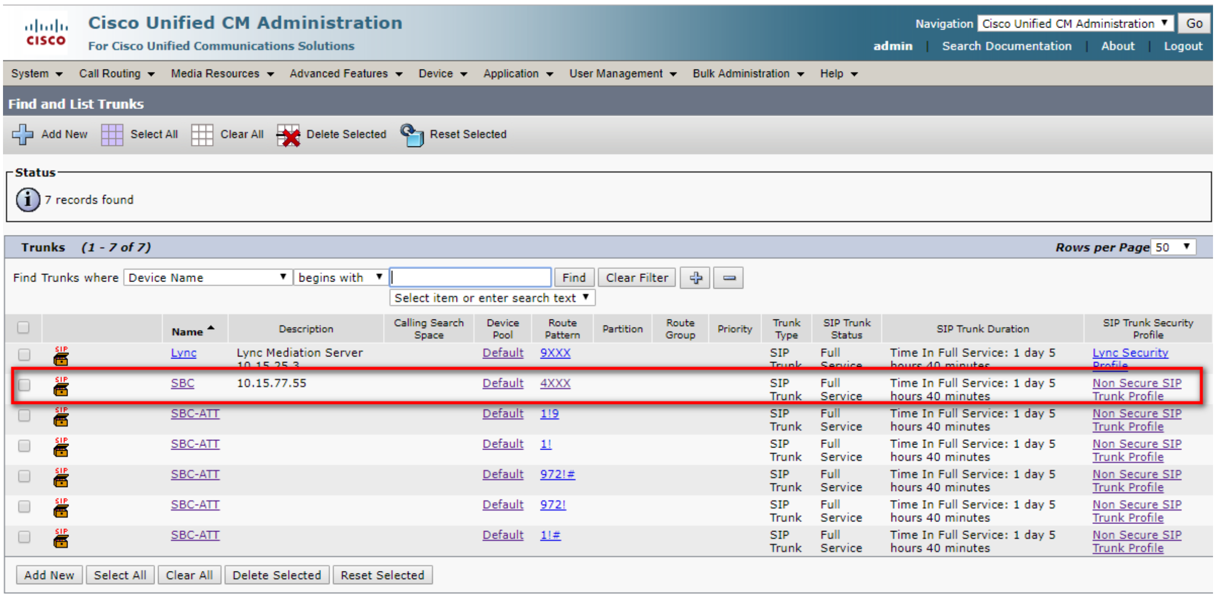


Figure 3-10: Added Trunk



Note: An '*' indicates a mandatory field.

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4 Configuring Amazon Chime Voice Connector

To configure Amazon Chime Voice Connector please refer to the following link:
https://docs.aws.amazon.com/chime/latest/ag/voice-connectors.html

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5 Configuring AudioCodes SBC

This chapter provides step-by-step procedures on how to configure AudioCodes SBC for interworking between Cisco CUCM and the AWS Chime Voice Connector. These configuration procedures are based on the interoperability test topology described in Section 2.4 on page 10, and include the following main areas:

- SBC WAN interface - AWS Chime Voice Connector environment
- SBC LAN interface - CUCM environment

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

Notes:

- For implementing CUCM and AWS Chime Voice Connector based on the configuration described in this section, AudioCodes SBC must be installed with a License Key that includes the following software features:
  - √ **SBC**
  - √ **Security**
  - √ **DSP**
  - √ **RTP**
  - √ **SIP**

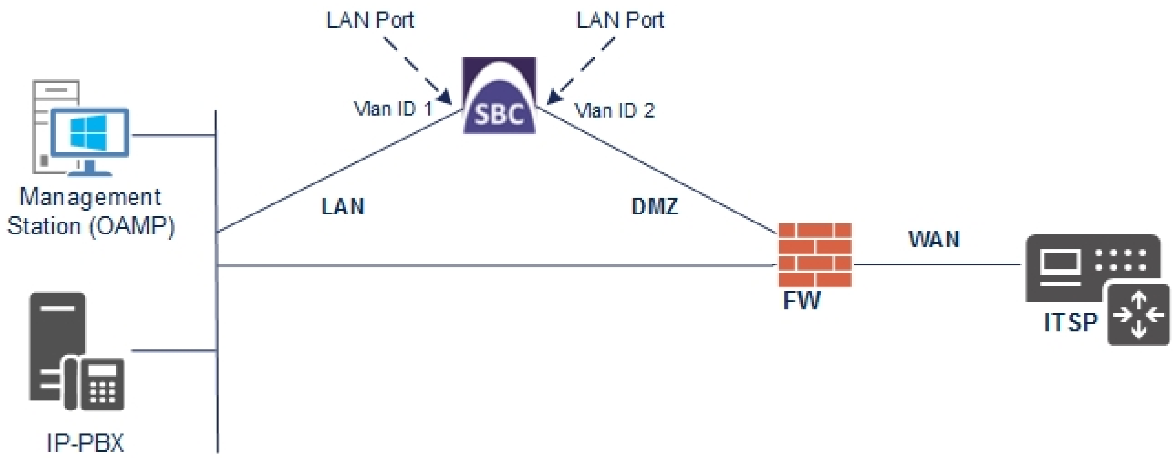
  For more information about the License Key, contact your AudioCodes sales representative.
- The scope of this interoperability test and document does **not** cover all security aspects for configuring this topology. Comprehensive security measures should be implemented per your organization's security policies. For security recommendations on AudioCodes' products, refer to the *Recommended Security Guidelines* document, which can be found at AudioCodes web site

5.1 Step 1: IP Network Interfaces Configuration

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

- SBC interfaces with the following IP entities:
  - Cisco CUCM, located on the LAN
  - AWS Chime Voice Connector, located on the WAN
- 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, SBC connects to the LAN and DMZ using dedicated LAN ports (i.e., two ports and two network cables are used).
- SBC also uses two logical network interfaces:
  - LAN (VLAN ID 1)
  - DMZ (VLAN ID 2)

Figure 5-1: Network Interfaces in Interoperability Test Topology



5.1.1 Step 1a: Configure VLANs

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

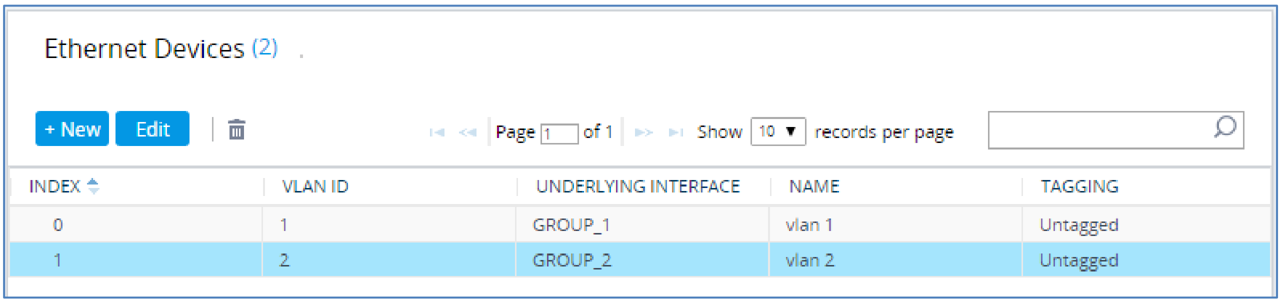
- LAN VoIP (assigned the name "LAN_IF")
- WAN VoIP (assigned the name "WAN_IF")

➢ **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 5-2: Configured VLAN IDs in Ethernet Device



5.1.2 Step 1b: Configure Network Interfaces

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

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

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

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

Parameter	Value

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

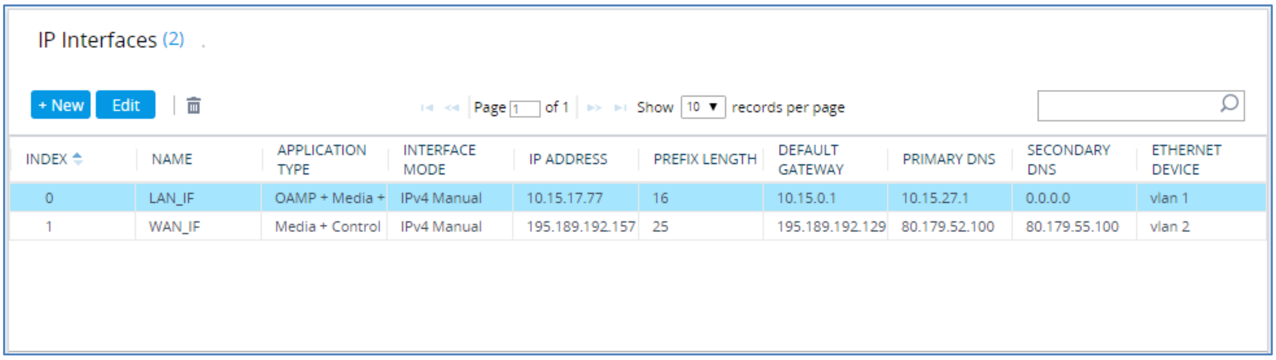
3. Add a network interface for the WAN side:
   a. Click **New**.
   b. Configure the interface as follows:

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

4. Click **Apply**.

The configured IP network interfaces are shown below:

Figure 5-3: Configured Network Interfaces in IP Interfaces Table


IP Interfaces (2)

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

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

5.2 Step 2: Configure Media Realms

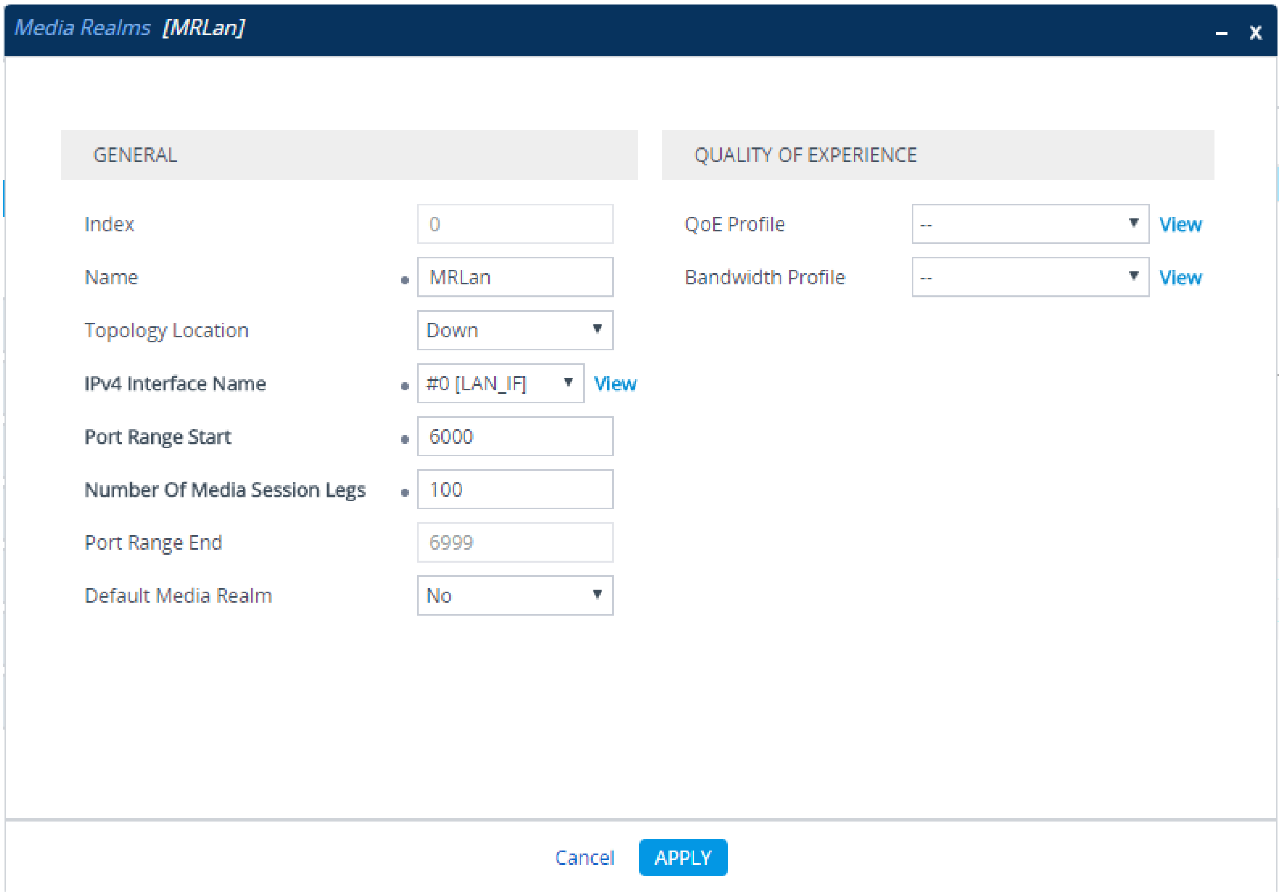
This step describes how to configure Media Realms. The simplest configuration is to create two Media Realms - one for internal (LAN) traffic and one for external (WAN) traffic.

➢ **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**
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)

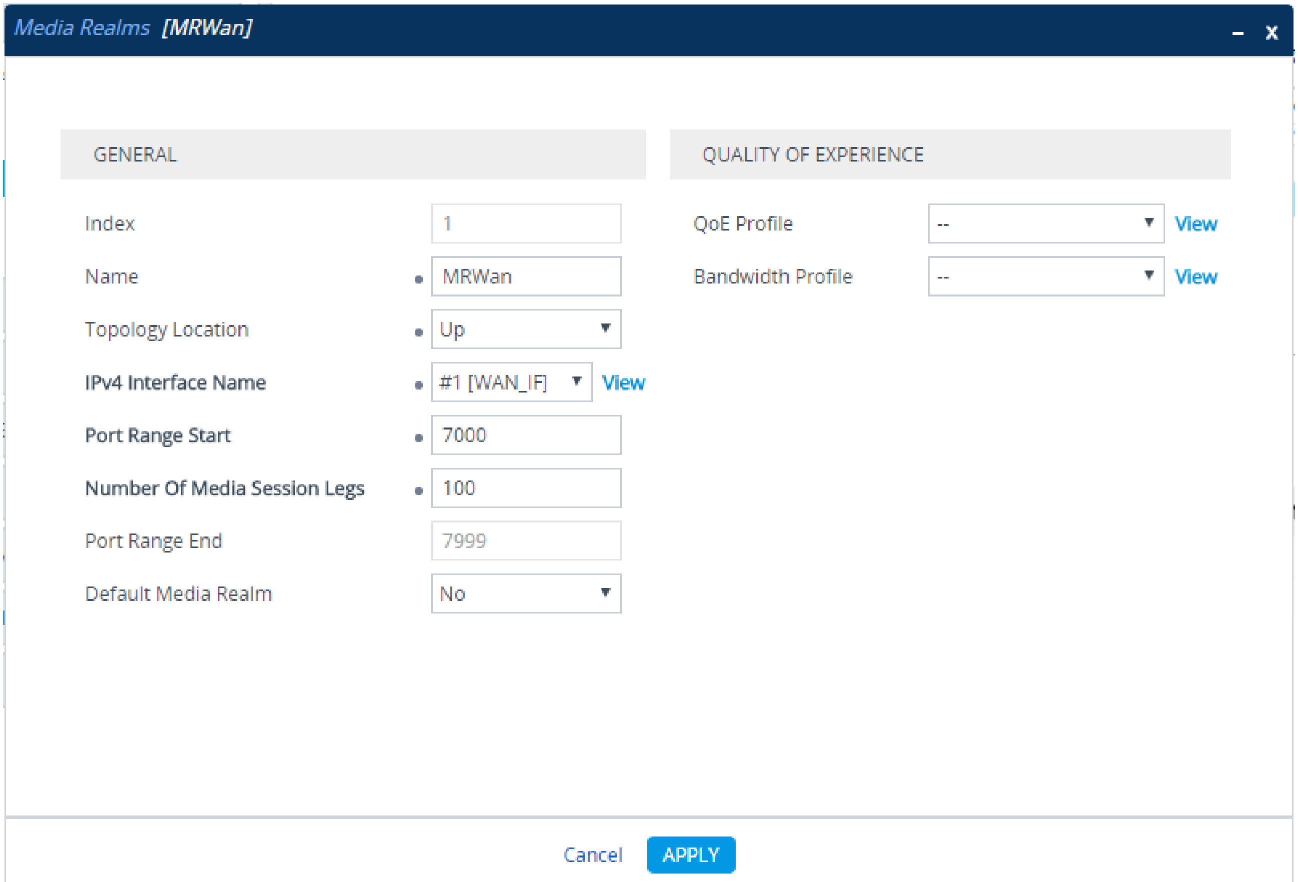
Figure 5-4: Configuring Media Realm for LAN



3. Configure a Media Realm for WAN traffic:

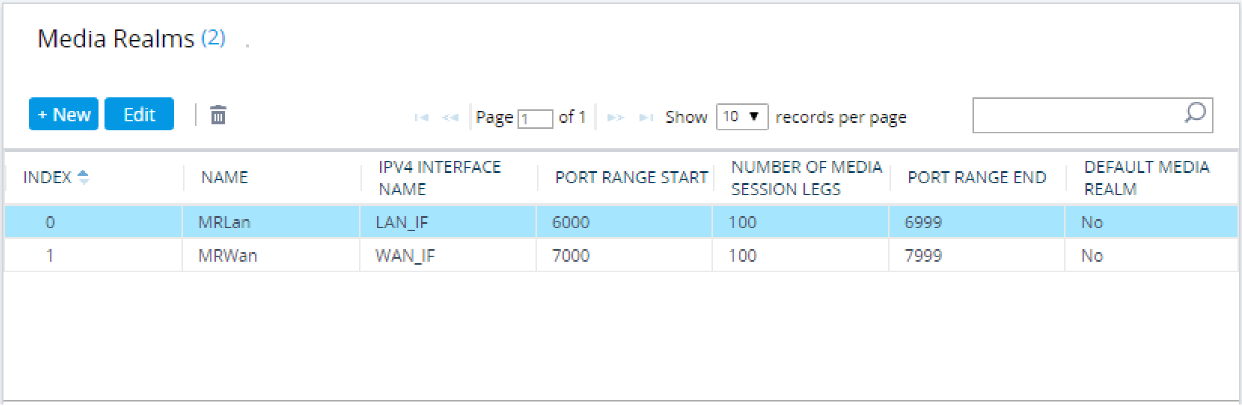
Parameter	Value
Index	**1**
Name	**MRWan** (arbitrary name)
Topology Location	**Up**
IPv4 Interface Name	**WAN_IF**
Port Range Start	**7000** (represents lowest UDP port number used for media on WAN)
Number of Media Session Legs	**100** (media sessions assigned with port range)

Figure 5-5: Configuring Media Realm for WAN



The configured Media Realms are shown in the figure below:

Figure 5-6: Configured Media Realms in Media Realm Table



5.3 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 SBC.

➢ **To configure SIP Interfaces:**

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

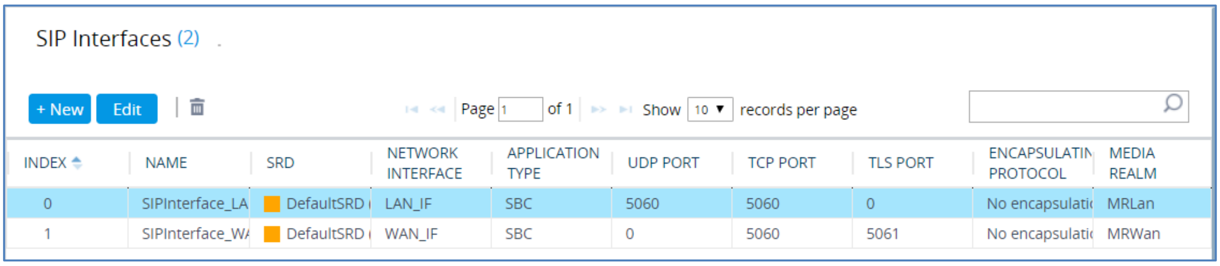
Parameter	Value
Index	**0**
Name	**SIPInterface_LAN** (see note at the end of this section)
Network Interface	**LAN_IF**
Application Type	**SBC**
UDP and TCP Ports	**5060**
TLS Port	**0**
Media Realm	**MRLan**

3. Configure a SIP Interface for the WAN:

Parameter	Value
Index	**1**
Name	**SIPInterface_WAN**
Network Interface	**WAN_IF**
Application Type	**SBC**
UDP Port	**5060**
TCP Port	**0**
TLS Port	**5061**
Media Realm	**MRWan**

The configured SIP Interfaces are shown in the figure below:

Figure 5-7: Configured SIP Interfaces in SIP Interface Table


SIP Interfaces (2)

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

INDEX	NAME	SRD	NETWORK INTERFACE	APPLICATION TYPE	UDP PORT	TCP PORT	TLS PORT	ENCAPSULATIN PROTOCOL	MEDIA REALM
0	SIPInterface_LA	DefaultSRD (	LAN_IF	SBC	5060	5060	0	No encapsulati	MRLan
1	SIPInterface_W/	DefaultSRD (	WAN_IF	SBC	0	5060	5061	No encapsulati	MRWan

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

5.4 Step 4: Configure Proxy Sets

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

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

- Cisco CUCM
- AWS Chime Voice Connector

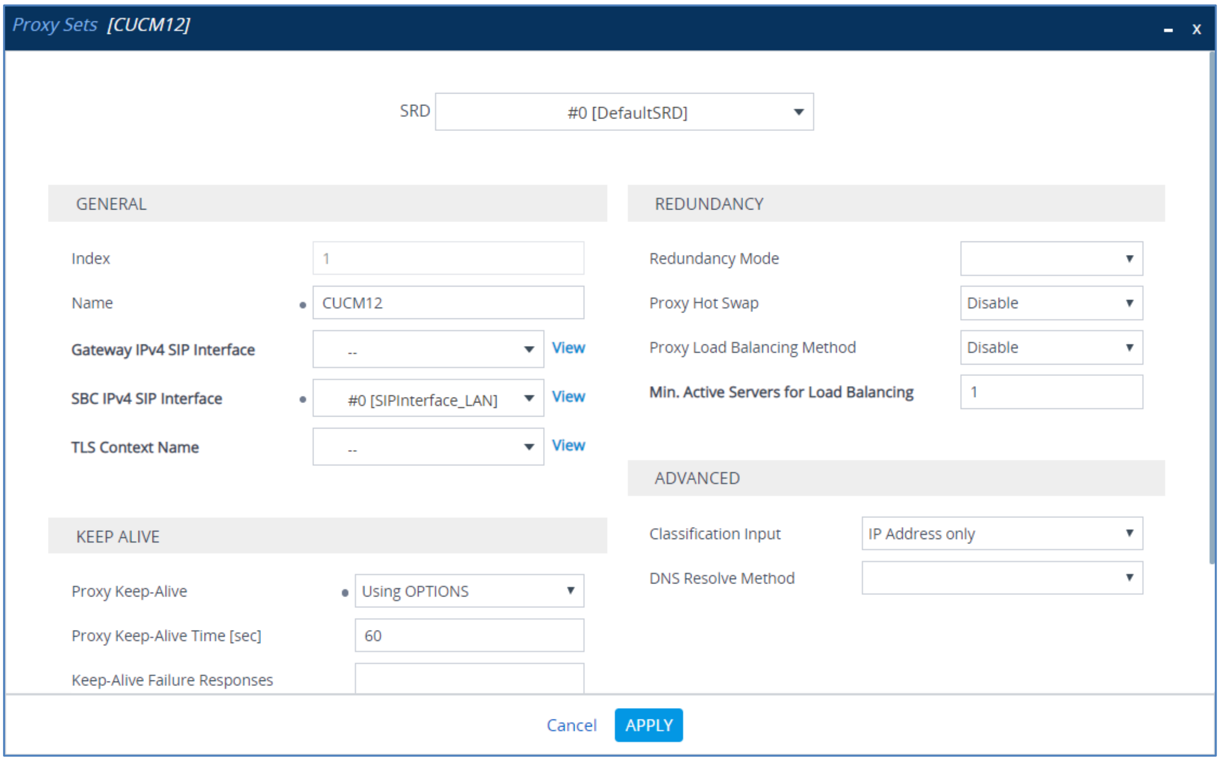
The Proxy Sets will be later applied to the VoIP network by assigning them to IP Groups.

➢ 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 Cisco CUCM as shown below:

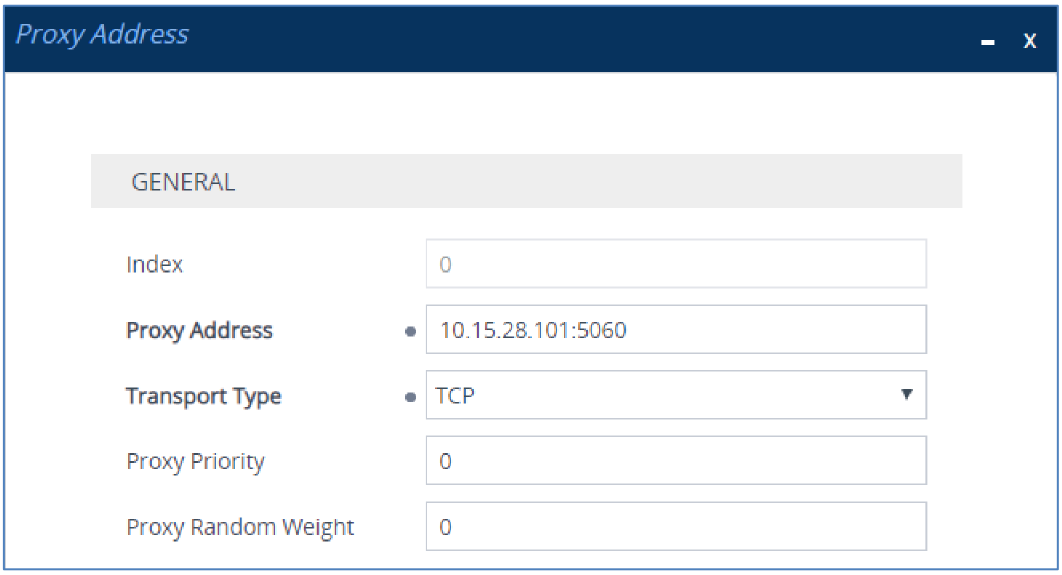
Parameter	Value
Index	**1**
Name	**CUCM12** (arbitrary name)
SBC IPv4 SIP Interface	**SIPInterface_LAN**
Proxy Keep-Alive	**Using Options**

Figure 5-8: Configuring Proxy Set for Cisco CUCM



   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 5-9: Configuring Proxy Address for Cisco CUCM



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

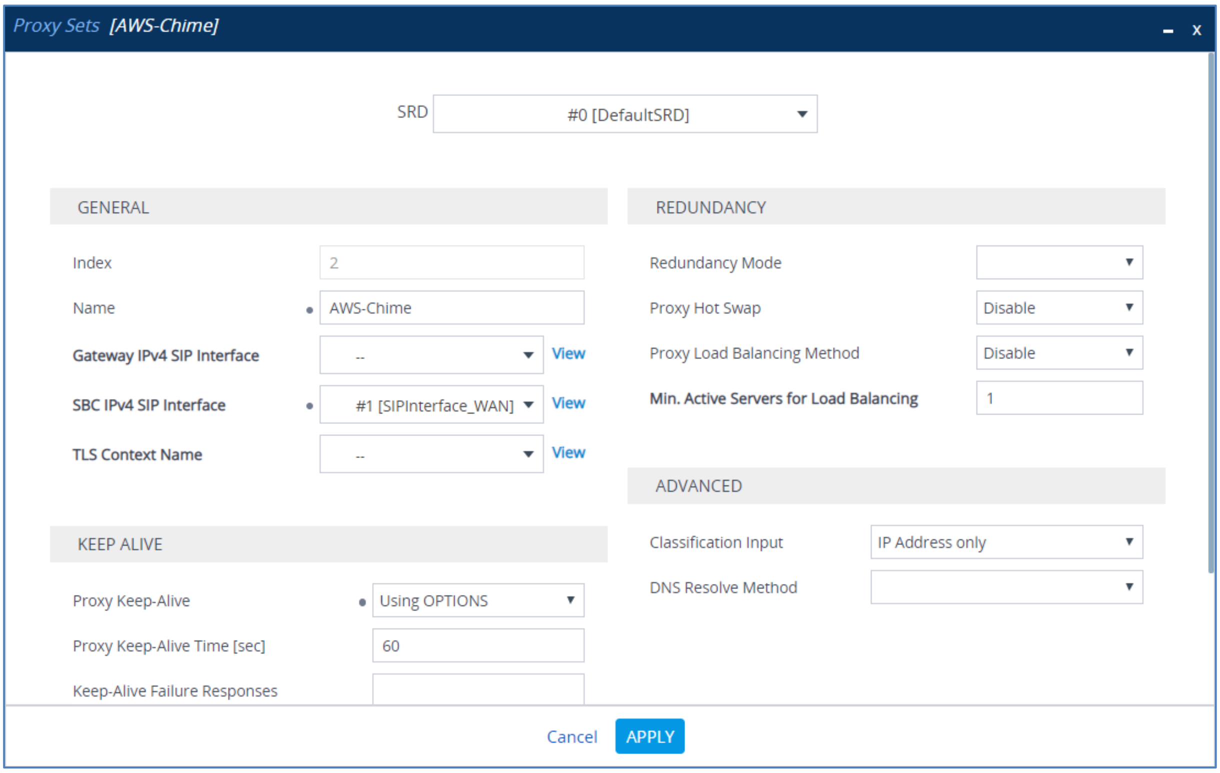
Parameter	Value
Index	**0**
Proxy Address	**10.15.28.101:5060** (IP-PBX IP address / FQDN and destination port)
Transport Type	**TCP**

d. Click **Apply**.

3. Configure a Proxy Set for the AWS Chime Voice Connector:

Parameter	Value
Index	**2**
Name	**AWS-Chime**
SBC IPv4 SIP Interface	**SIPInterface_WAN**
Proxy Keep-Alive	**Using Options**

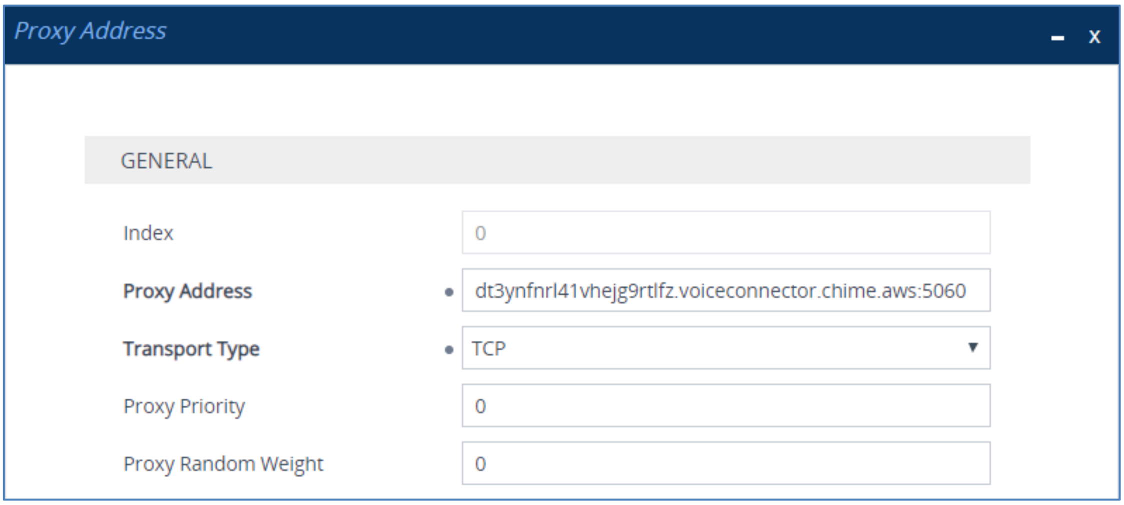
Figure 5-10: Configuring Proxy Set for AWS Chime Voice Connector



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 5-11: Configuring Proxy Address for AWS Chime Voice Connector



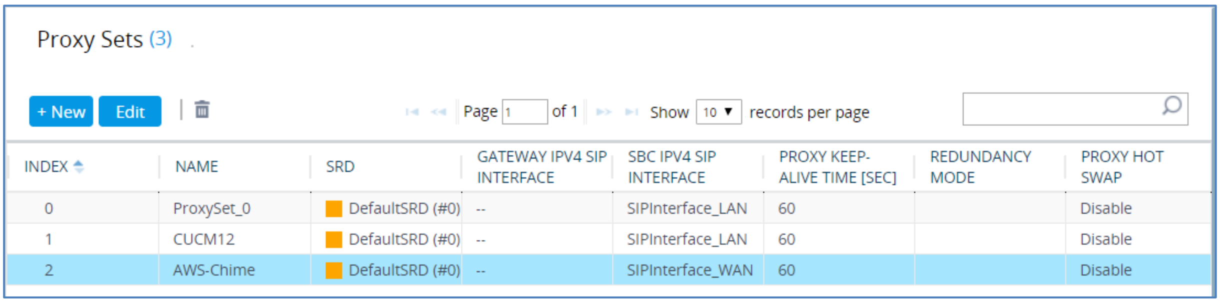
c. Configure the address of the Proxy Set according to the parameters described in the table below.

Parameter	Value
Index	**0**
Proxy Address	**dt3ynfnrl41vhejg9rtlfz.voiceconnector.chime.aws:5060** (FQDN and destination port of your enterprise voice connector ID)
Transport Type	**TCP** or **TLS** (according to connection requirement)

d. Click **Apply**.

The configured Proxy Sets are shown in the figure below:

Figure 5-12: Configured Proxy Sets in Proxy Sets Table



5.5 Step 5: Configure Coders

This step describes how to configure coders (termed *Coder Group*). As Cisco CUCM may support different coders while the AWS Chime Voice Connector supports only G.711 U-law coder, you need to add a Coder Group with the G.711 U-law coder for the AWS Chime Voice Connector.

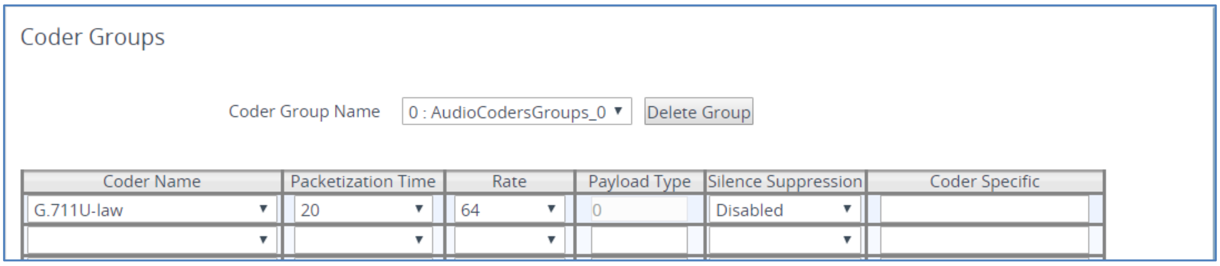
Note that the Coder Group ID for this entity will be assigned to its corresponding IP Profile in the next step.

➢ **To configure coders:**

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

Parameter	Value
Coder Group Name	**AudioCodersGroups_0**
Coder Name	**G.711 U-law**

Figure 5-13: Configuring Coder Group for AWS Chime Voice Connector



5.6 Step 6: 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:

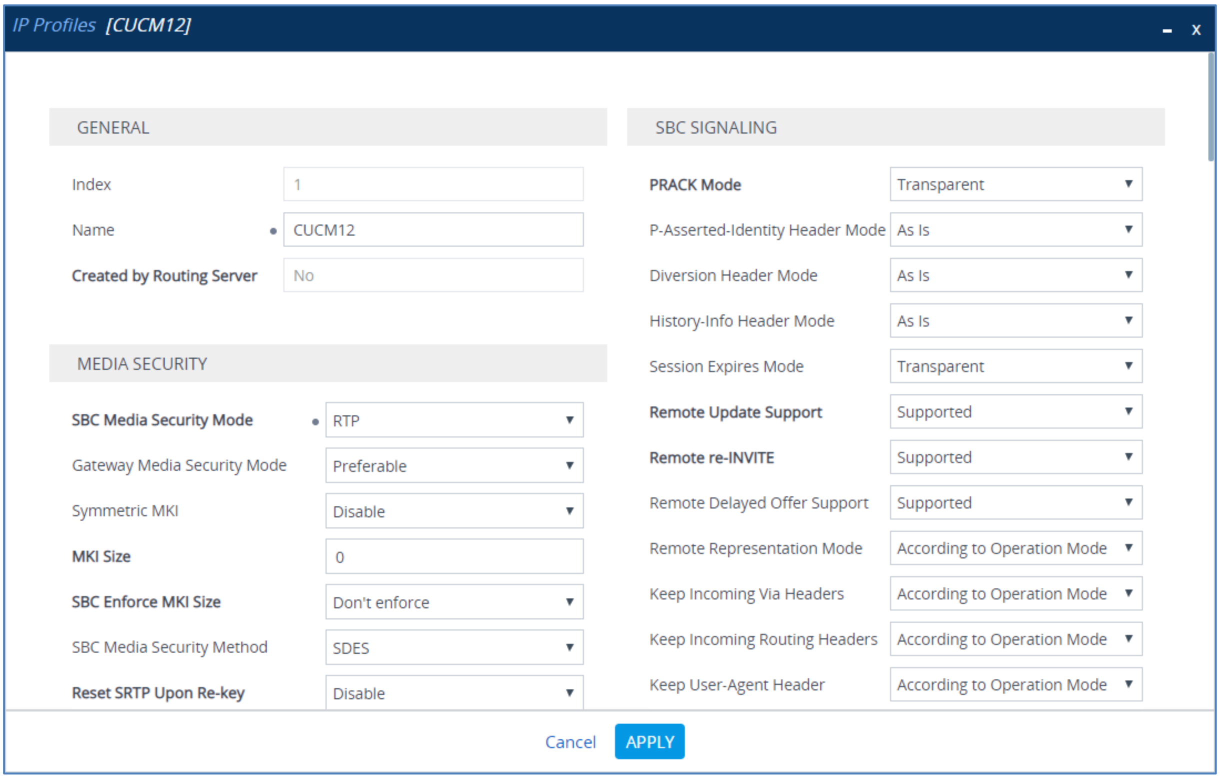
- Cisco CUCM – to operate in non-secure mode using RTP and SIP over TCP
- AWS Chime Voice Connector – to operate in non-secure mode using RTP and SIP over UDP

➢ **To configure IP Profile for CISCO CUCM:**

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	**CUCM12**
Media Security	
SBC Media Security Mode	**RTP**

Figure 5-14: Configuring IP Profile for Cisco CUCM



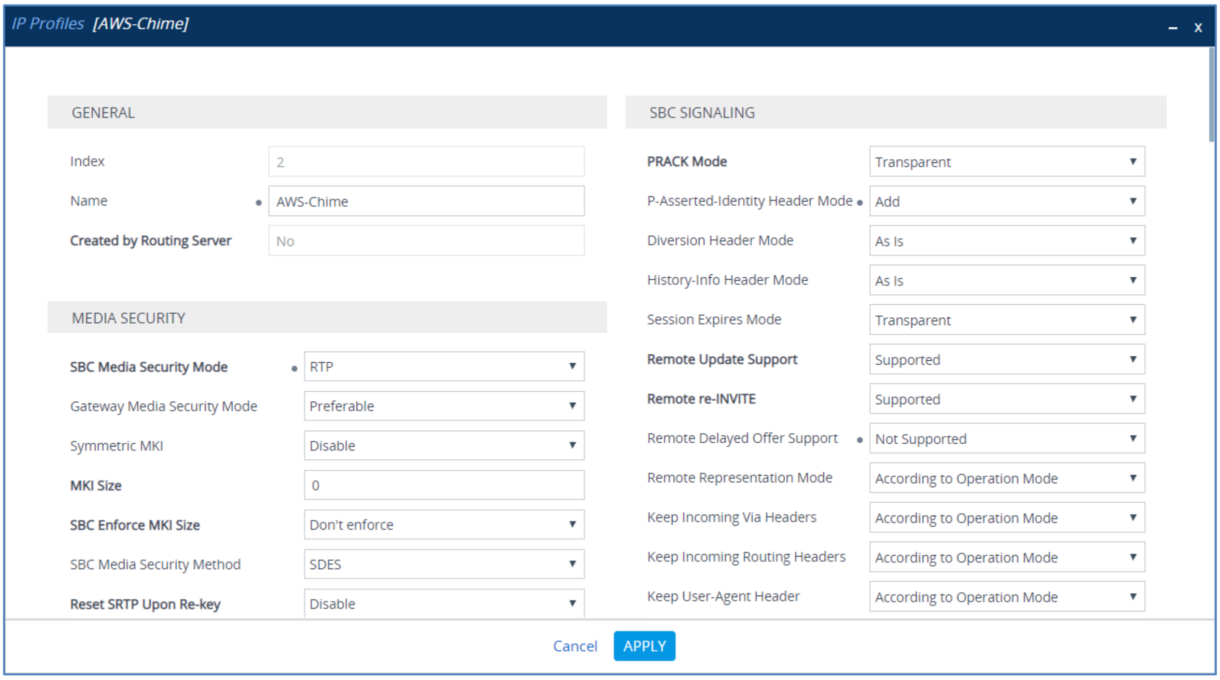
3. Click **Apply**.

➢ To configure an IP Profile for the AWS Chime Voice Connector:

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

Parameter	Value
General	
Index	**2**
Name	**AWS-Chime**
Media Security	
SBC Media Security Mode	**RTP** or **SRTP** (according to connection requirement)
SBC Media	
Extension Coders Group	**AudioCodersGroups_0**
RFC 2833 Mode	**Extended** (in case CUCM is configured without support for RFC 2833)
SBC Signaling	
P-Asserted-Identity Header Mode	**Add** (required for anonymous calls)
Remote Delayed Offer Support	**Not Supported**

Figure 5-15: Configuring IP Profile for AWS Chime Voice Connector



2. Click **Apply**.

5.7 Step 7: Configure IP Groups

This step describes how to configure IP Groups. The IP Group represents an IP entity on the network with which the 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:

- Cisco CUCM located on LAN
- AWS Chime Voice Connector 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 Cisco CUCM:

Parameter	Value
Index	**1**
Name	**CUCM12**
Type	**Server**
Proxy Set	**CUCM12**
IP Profile	**CUCM12**
Media Realm	**MRLan**
SIP Group Name	(FQDN of your enterprise AWS Chime Voice Connector ID)

3. Configure an IP Group for the AWS Chime Voice Connector:

Parameter	Value
Index	**2**
Name	**AWS-Chime**
Topology Location	**Up**
Type	**Server**
Proxy Set	**AWS-Chime**
IP Profile	**AWS-Chime**
Media Realm	**MRWan**
SIP Group Name	(FQDN of your enterprise AWS Chime voice connector ID)

The configured IP Groups are shown in the figure below:

Figure 5-16: 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	OUTBOUN MESSAGE MANIPULA SET
0	Default_IPG	DefaultSF	Server	Not Configur	ProxySet_0	--	--		Disable	-1	-1
1	CUCM12	DefaultSF	Server	Not Configur	CUCM12	CUCM12	MRLan	dt3ynfnrl41v	Enable	-1	2
2	AWS-Chime	DefaultSF	Server	Not Configur	AWS-Chime	AWS-Chime	MRWan	dt3ynfnrl41v	Enable	-1	-1

5.8 Step 8: SIP TLS Connection Configuration (optional)

This section describes how to configure the SBC for using a TLS connection with the AWS Chime Voice Connector. This is essential for a secure SIP TLS connection and highly recommended by Amazon.

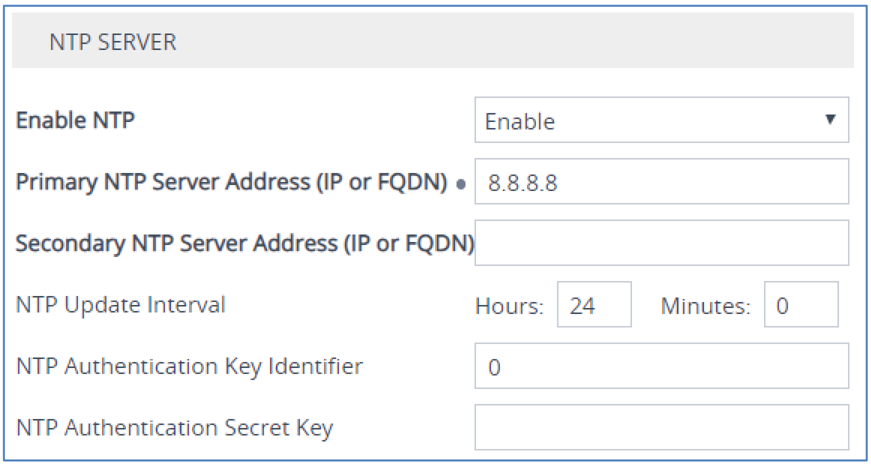
5.8.1 Step 8a: Configure the NTP Server Address

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 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., **8.8.8.8**).

Figure 5-17: Configuring NTP Server Address



3. Click **Apply**.

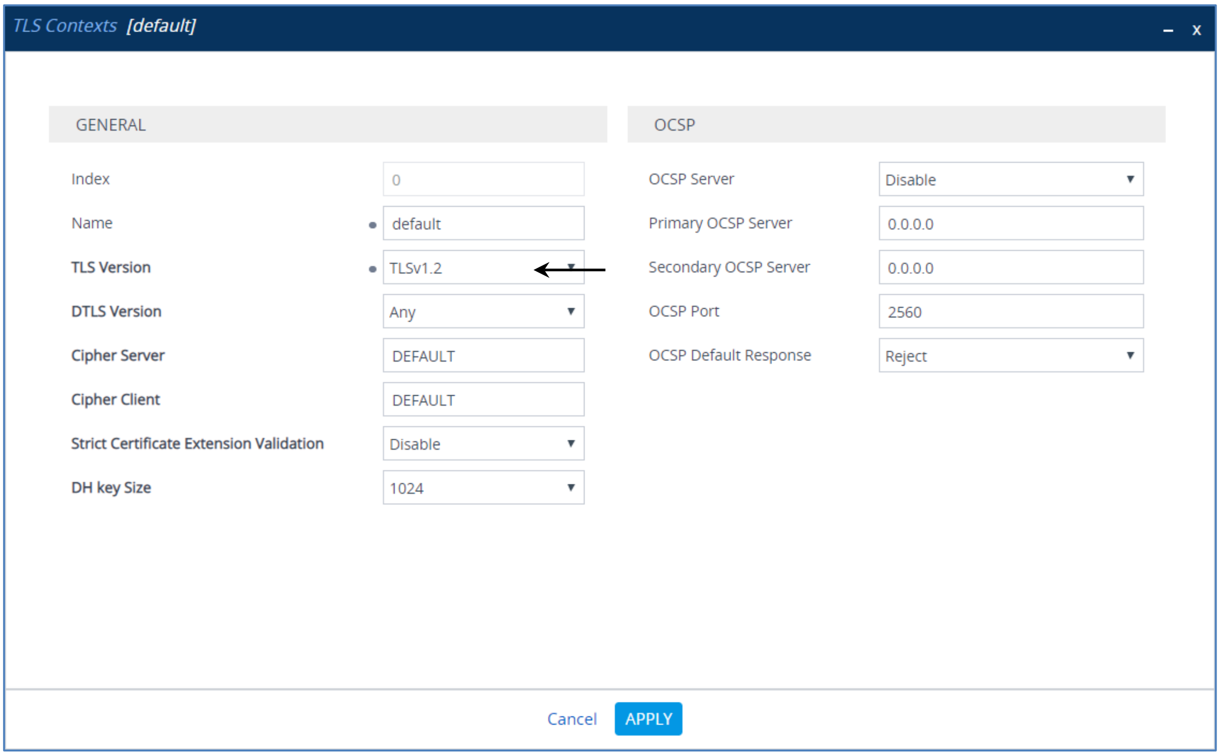
5.8.2 Step 8b: Configure the TLS version

This step describes how to configure the SBC to use TLS version 1.2 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 '**TLSv1.2**'

Figure 5-18: Configuring TLS version



4. Click **Apply**.

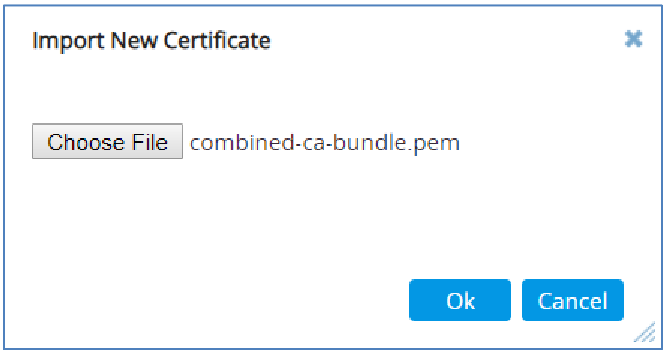
5.8.3 Step 8c: Deploy Amazon Trusted Root Certificate

This step describes how to import the Amazon Chime root certificate. Currently the Amazon Chime Voice Connector service uses a wildcard certificate (*.voiceconnector.chime.aws). To trust this certificate, your SBC *must* import this certificate to its Trusted Certificates storage. Download the certificate from https://s3.amazonaws.com/voice-connector-certs/combined-ca-bundle.pem. Follow the steps below to import the certificate to the Trusted Root storage.

➢ **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 **Trusted Root Certificates** link, located at the bottom of the TLS Contexts page; the Trusted Certificates page appears.
3. Click the **Import** button, and then select the certificate file to load:

Figure 5-19: Importing Root Certificate into Trusted Certificates Store



4. Click **OK**; the certificate is loaded to the device and listed in the Trusted Certificates store.

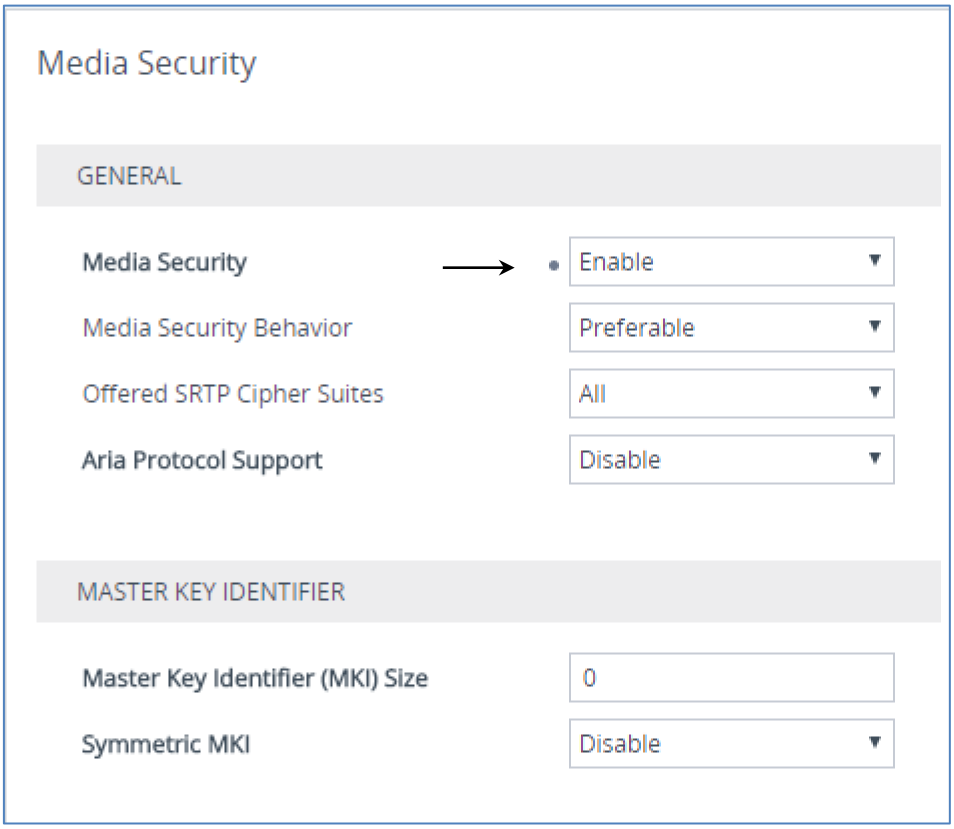
5.9 Step 9: Configure SRTP (optional)

This step describes how to configure media security. If the AWS Chime Voice Connector requires SRTP, configure the SBC to operate in the same manner. Note that SRTP is enabled for the AWS Chime Voice Connector when you configure an IP Profile (see Section 5.5 on page 34).

➢ **To configure media security:**

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

Figure 5-20: Configuring SRTP



2. From the 'Media Security' drop-down list, select **Enable** to enable SRTP.
3. Click **Apply**.

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

This step describes how to configure IP-to-IP call routing rules. These rules define the routes for forwarding SIP messages (e.g., INVITE) received from one IP entity to another. The 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 5.7 on page 33,) 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 Cisco CUCM (LAN) and AWS Chime Voice Connector (DMZ):

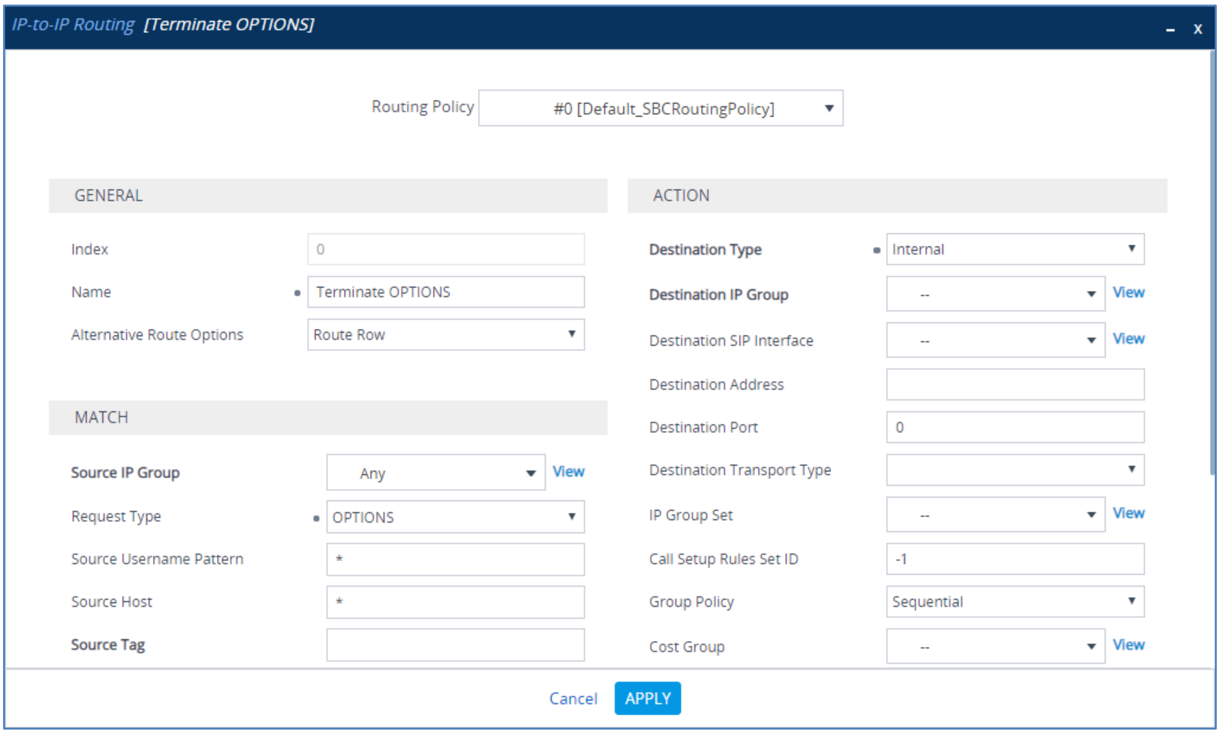
- Terminate SIP OPTIONS messages on the SBC that are received from the both LAN and DMZ
- Calls from Cisco CUCM to AWS Chime Voice Connector
- Calls from AWS Chime Voice Connector to Cisco CUCM

➢ **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	**Internal**
Internal Action	**Reply (Response='200')**

Figure 5-21: Configuring IP-to-IP Routing Rule for Terminating SIP OPTIONS

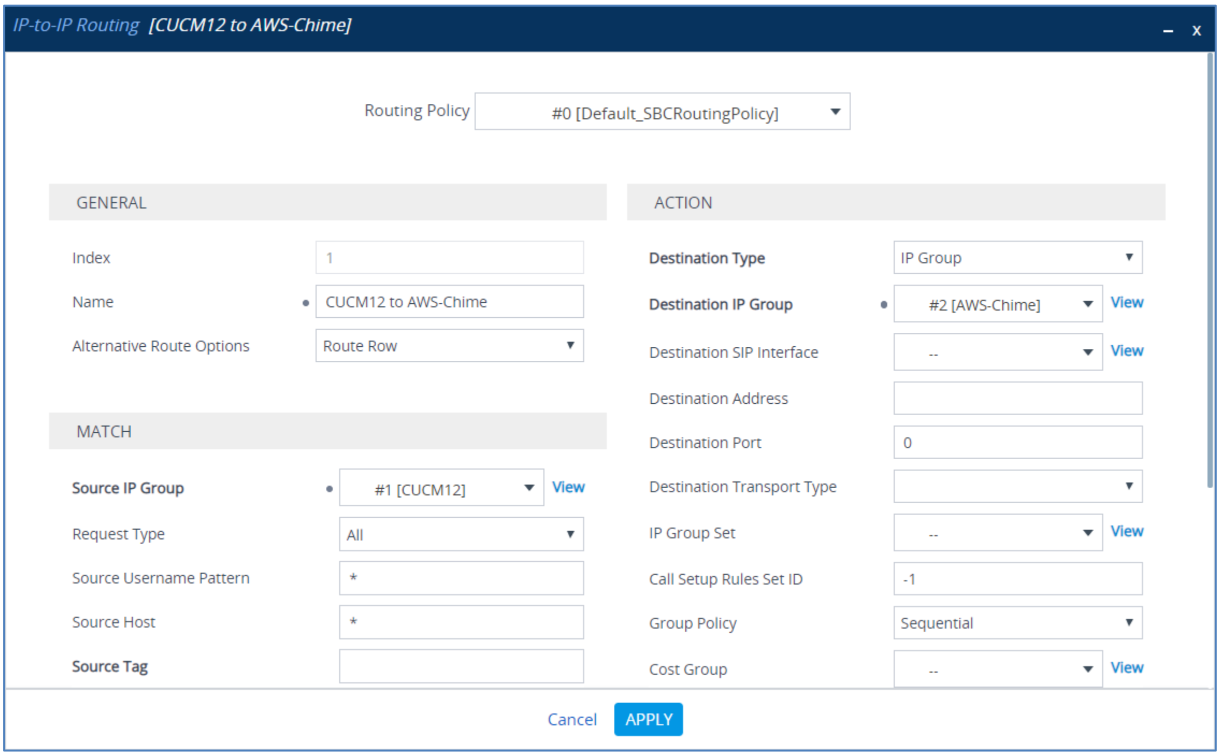


b. Click **Apply**.

4. Configure a rule to route calls from Cisco CUCM to AWS Chime Voice Connector:
   a. Click **New**, and then configure the parameters as follows:

Parameter	Value
Index	**2**
Route Name	**CUCM12 to AWS-Chime** (arbitrary descriptive name)
Source IP Group	**CUCM12**
Destination Type	**IP Group**
Destination IP Group	**AWS-Chime**

Figure 5-22: Configuring IP-to-IP Routing Rule for Cisco CUCM to AWS-Chime

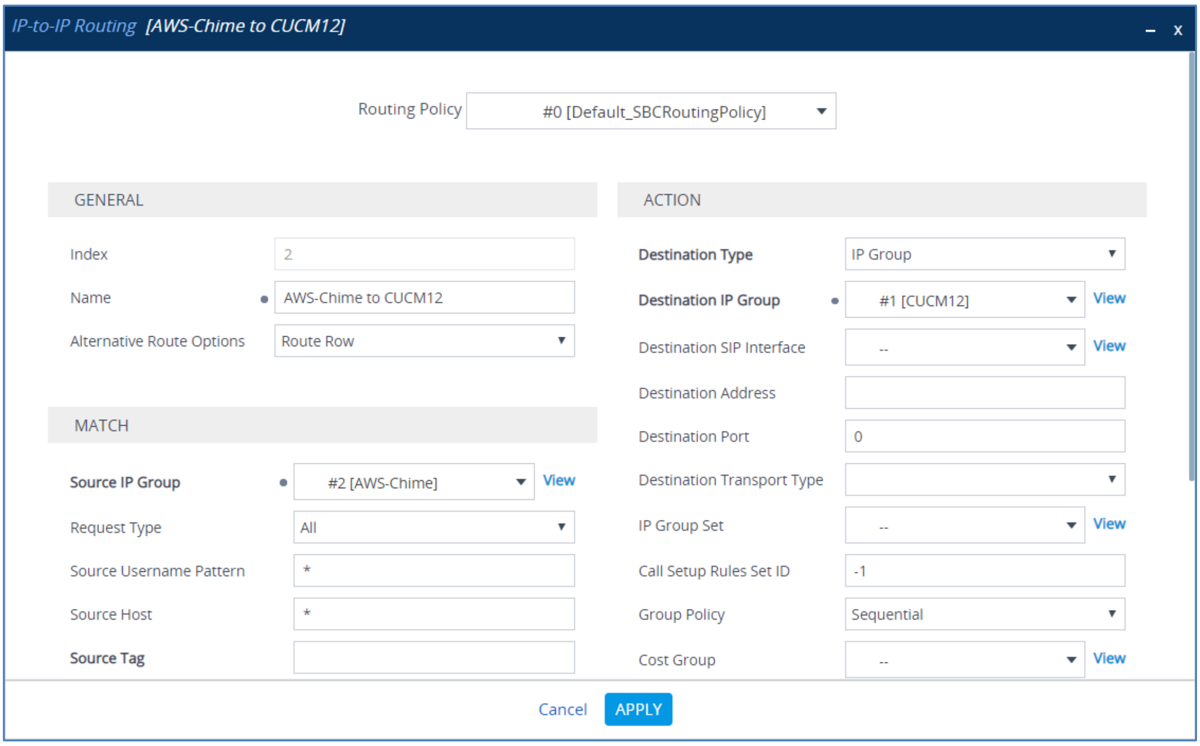


   b. Click **Apply**.

5. Configure rule to route calls from AWS Chime Voice Connector to Cisco CUCM:
    a. Click **New**, and then configure the parameters as follows:

Parameter	Value
Index	**4**
Route Name	**AWS-Chime to CUCM12** (arbitrary descriptive name)
Source IP Group	**AWS-Chime**
Destination Type	**IP Group**
Destination IP Group	**CUCM12**

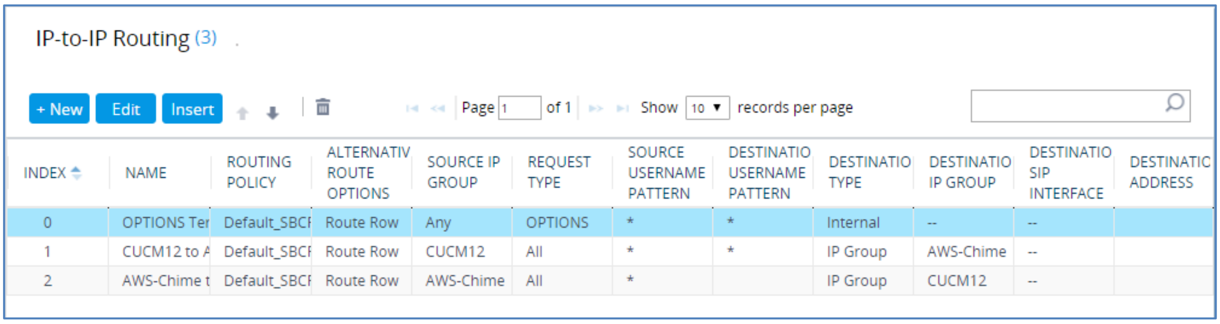
Figure 5-23: Configuring IP-to-IP Routing Rule for AWS-Chime to Cisco CUCM Server



    b. Click **Apply.**

The configured routing rules are shown in the figure below:

Figure 5-24: Configured IP-to-IP Routing Rules in IP-to-IP Routing Table



INDEX	NAME	ROUTING POLICY	ALTERNATIV ROUTE OPTIONS	SOURCE IP GROUP	REQUEST TYPE	SOURCE USERNAME PATTERN	DESTINATIO USERNAME PATTERN	DESTINATIO TYPE	DESTINATIO IP GROUP	DESTINATIO SIP INTERFACE	DESTINATIC ADDRESS
0	OPTIONS Ter	Default_SBCI	Route Row	Any	OPTIONS	*	*	Internal	--	--	
1	CUCM12 to A	Default_SBCI	Route Row	CUCM12	All	*	*	IP Group	AWS-Chime	--	
2	AWS-Chime t	Default_SBCI	Route Row	AWS-Chime	All	*		IP Group	CUCM12	--	

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

5.11 Step 11: 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 5.7 on page 33) to denote the source and destination of the call.

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

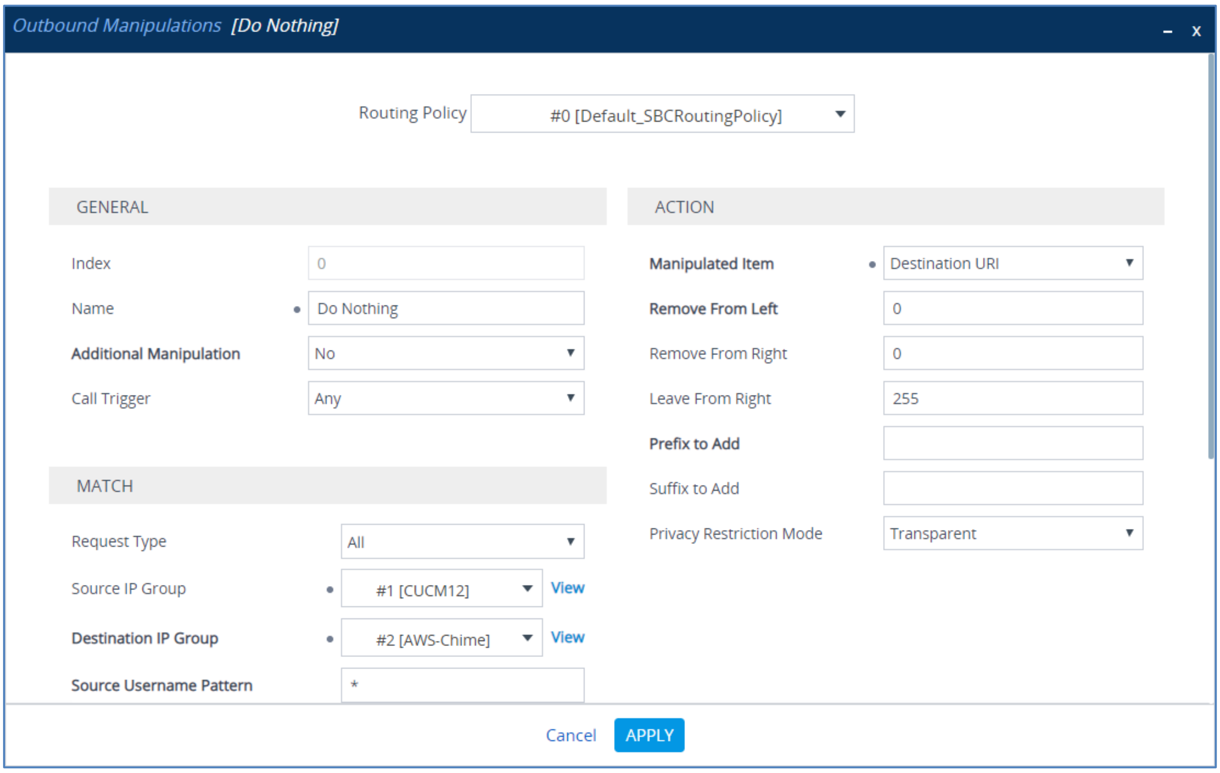
For example, the AWS Chime requires that the dialed number be displayed in E.164 format. However the Cisco CUCM doesn't support the "+" (plus sign) in the phone number. So, for the interoperability, a manipulation is configured to add the "+" (plus sign) to the destination number (if it does not exist) for calls from the CUCM Server IP Group to the AWS Chime Voice Connector IP Group for any destination username pattern. In the opposite direction, strip the "+" (plus sign) from the phone number for calls from the AWS Chime Voice Connector IP Group to the CUCM Server IP Group.

➢ **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	**Do Nothing**
Source IP Group	**CUCM12**
Destination IP Group	**AWS-Chime**
Destination Username Pattern	**+** (plus sign)
Manipulated Item	**Destination URI**

Figure 5-25: Configuring IP-to-IP Outbound Manipulation Rule

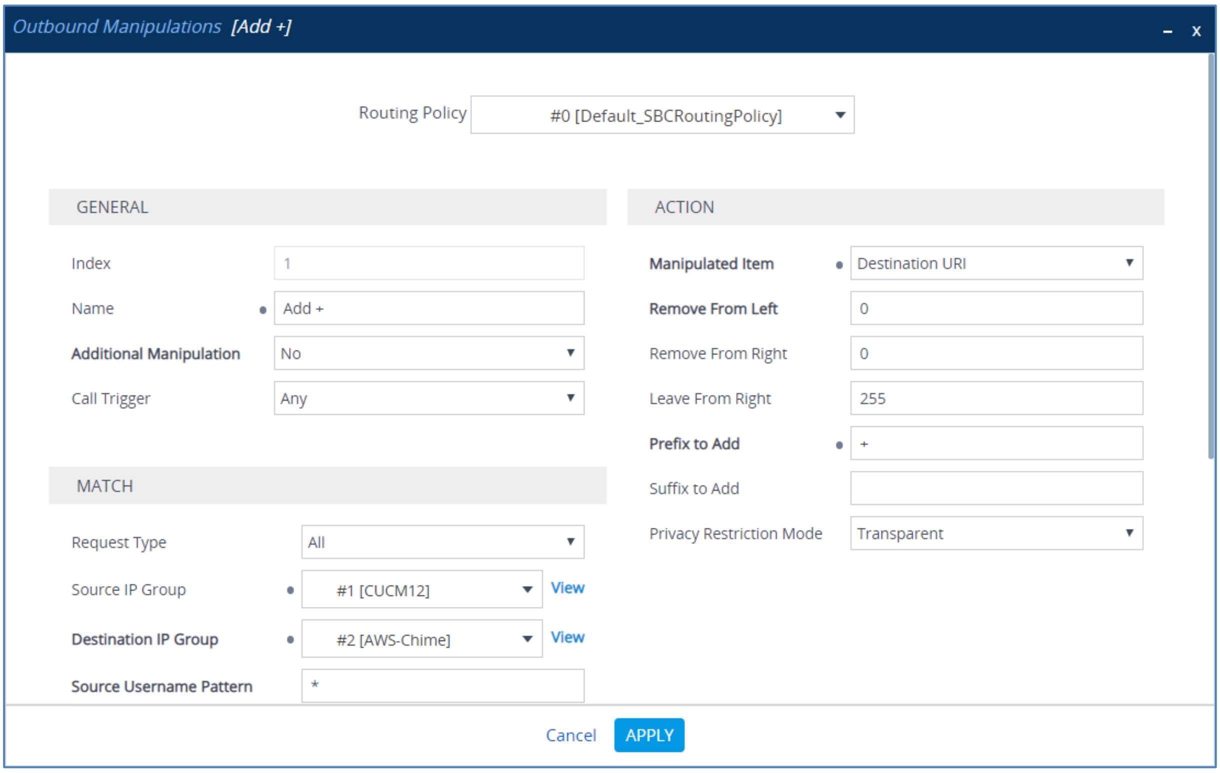


3. Click **Apply**.

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

Parameter	Value
Index	**1**
Name	**Add +**
Source IP Group	**CUCM12**
Destination IP Group	**AWS-Chime**
Destination Username Pattern	* (asterisk sign)
Manipulated Item	**Destination URI**
Prefix to Add	**+** (plus sign)

Figure 5-26: Configuring IP-to-IP Outbound Manipulation Rule

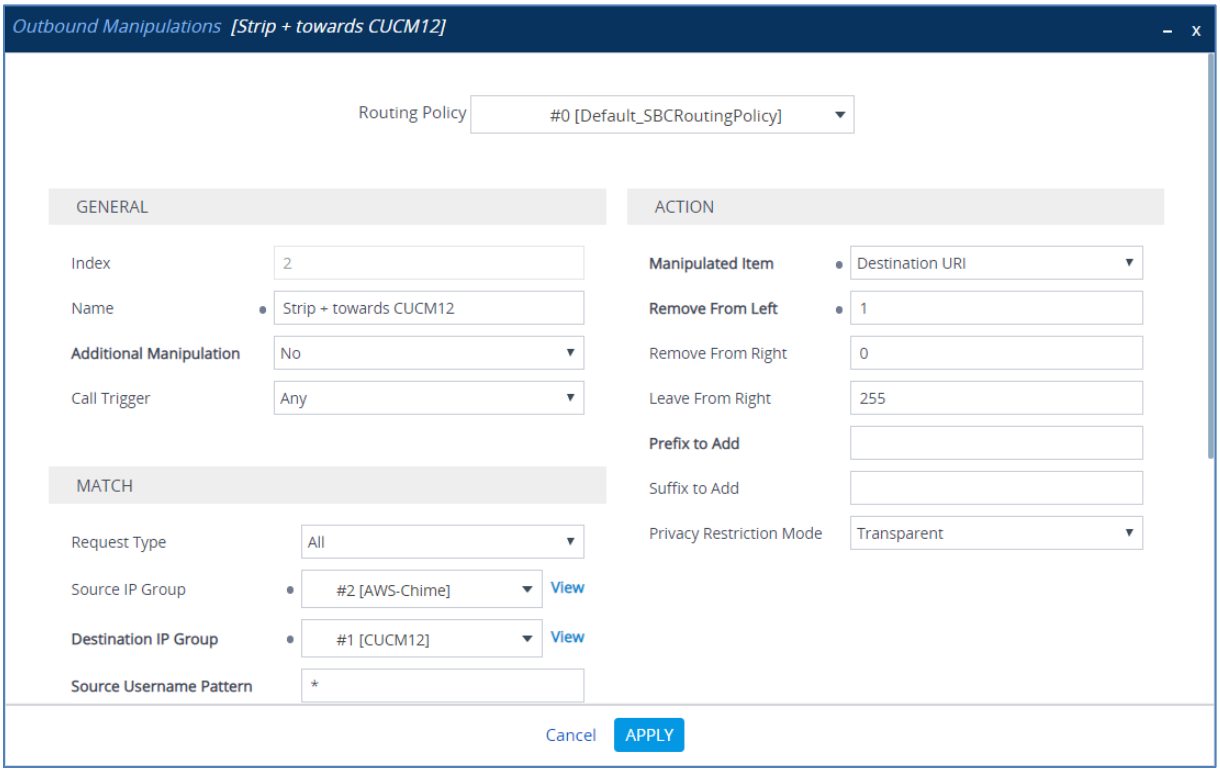


5. Click **Apply**.

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

Parameter	Value
Index	**2**
Name	**Strip + towards CUCM12**
Source IP Group	**AWS-Chime**
Destination IP Group	**CUCM12**
Destination Username Pattern	**+** (plus sign)
Manipulated Item	**Destination URI**
Remove From Left	**1**

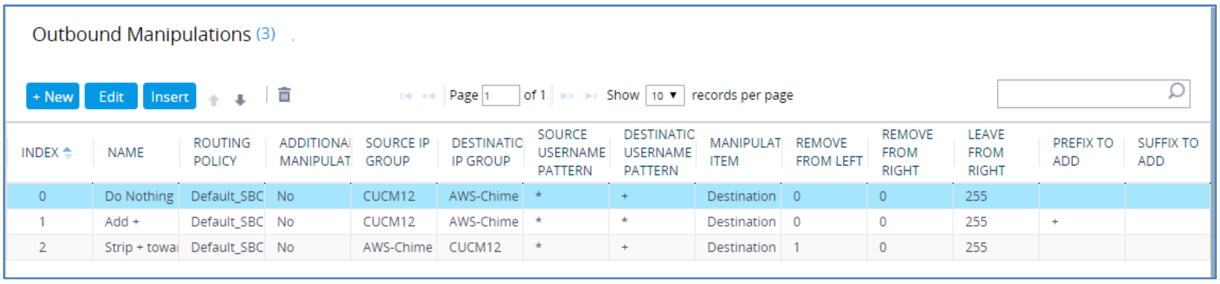
Figure 5-27: Configuring IP-to-IP Outbound Manipulation Rule



7. Click **Apply**.

The figure below shows an example of configured IP-to-IP outbound manipulation rules for calls between CUCM Server IP Group and AWS Chime Voice Connector IP Group:

Figure 5-28: Example of Configured IP-to-IP Outbound Manipulation Rules


Outbound Manipulations (3)

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

INDEX	NAME	ROUTING POLICY	ADDITIONAL MANIPULAT	SOURCE IP GROUP	DESTINATIC IP GROUP	SOURCE USERNAME PATTERN	DESTINATIC USERNAME PATTERN	MANIPULAT ITEM	REMOVE FROM LEFT	REMOVE FROM RIGHT	LEAVE FROM RIGHT	PREFIX TO ADD	SUFFIX TO ADD
0	Do Nothing	Default_SBC	No	CUCM12	AWS-Chime	*	+	Destination	0	0	255		
1	Add +	Default_SBC	No	CUCM12	AWS-Chime	*	*	Destination	0	0	255	+	
2	Strip + towa	Default_SBC	No	AWS-Chime	CUCM12	*	+	Destination	1	0	255		

Rule Index	Description
0	Calls from CUCM12 IP Group to AWS-Chime IP Group with the prefix destination number "+", do nothing.
1	Calls from CUCM12 IP Group to AWS-Chime IP Group with any destination number (*), add "+" to the prefix of the destination number.
2	Calls from AWS-Chime IP Group to CUCM12 IP Group with the prefix destination number "+", remove one character from the left (remove “+”).

5.12 Step 12: Configure Message Manipulation Rules

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

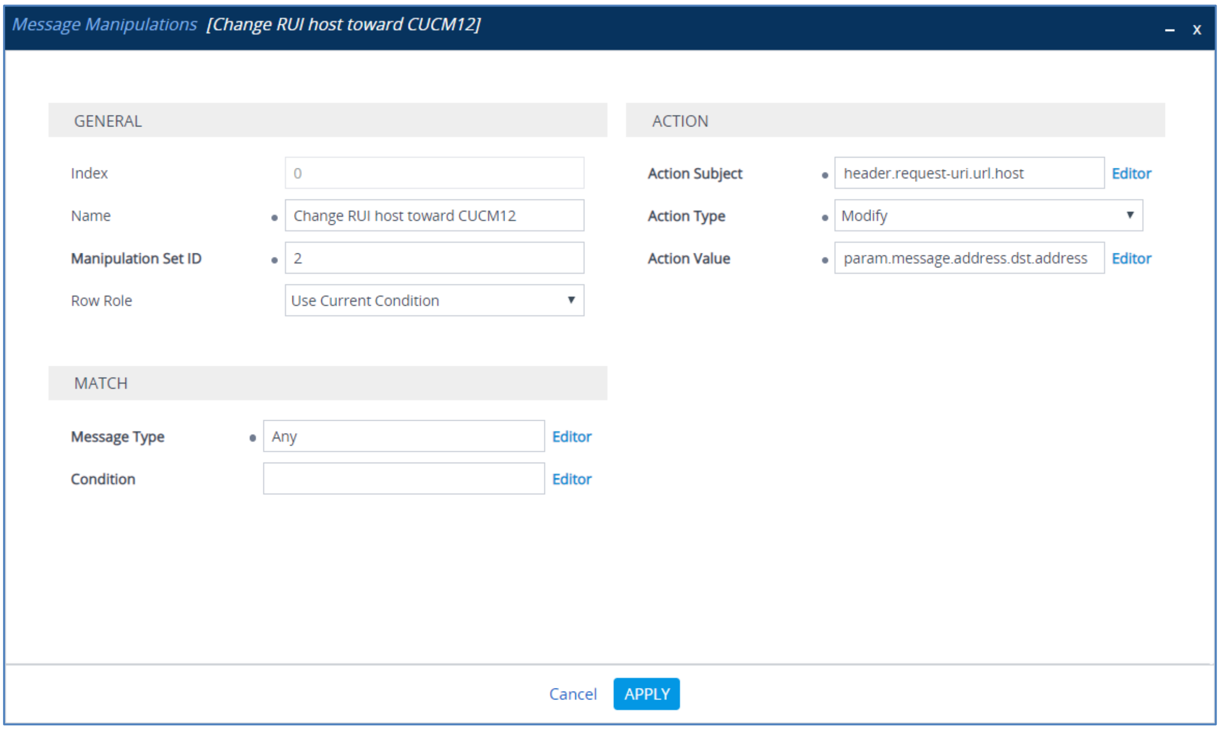
Once you have configured the SIP message manipulation rules, you need to assign them to the relevant IP Group (in the IP Group table) and determine whether they must be applied to inbound or outbound messages.

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

1. Open the Message Manipulations page (**Setup** menu > **Signaling & Media** tab > **Message Manipulation** folder > **Message Manipulations**).
2. Configure a manipulation rule (Manipulation Set 2) for the Cisco CUCM server. This rule applies to messages sent to the CUCM Server IP Group. This replaces the host part of the SIP Request-URI Header with the CUCM Server IP address.

Parameter	Value
Index	**0**
Name	**Change R-URI host toward CUCM12**
Manipulation Set ID	**2**
Message Type	**Any.Request**
Action Subject	**Header.Request-URI.URL.Host**
Action Type	**Modify**
Action Value	**Param.Message.Address.Dst.Address**

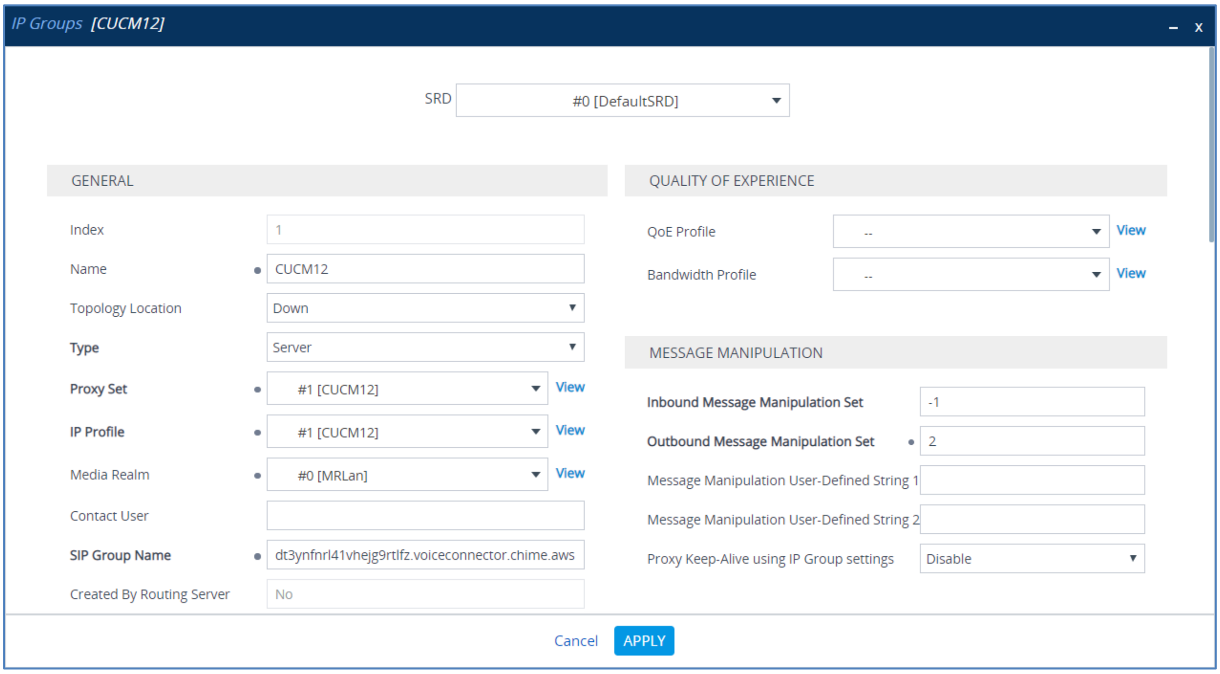
Figure 5-29: Configuring SIP Message Manipulation Rule 0 (for CUCM12)



Note: Due to fact that Cisco CUCM can be configured in different ways (e.g. to use SIP REFER Message or Re-INVITE for Call Transfer scenarios), different Message Manipulation Rules may needrequired to be configured.

3. Assign Manipulation Set ID 2 to the CUCM Server IP Group:
   a. Open the IP Groups table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **IP Groups**).
   b. Select the row of the CUCM Server IP Group, and then click **Edit**.
   c. Set the 'Outbound Message Manipulation Set' field to **2**.

Figure 5-30: Assigning Manipulation Set to the CUCM Server IP Group



   d. Click **Apply**.

5.13 Step 13: Configure Account for Authentication

This step describes how to configure the SIP account for authentication purposes. Amazon Chime Voice Connector requires IP-based whitelisting for outbound calling. Consequently, the SBC needs to be configured with the appropriate credentials using the Accounts Table.

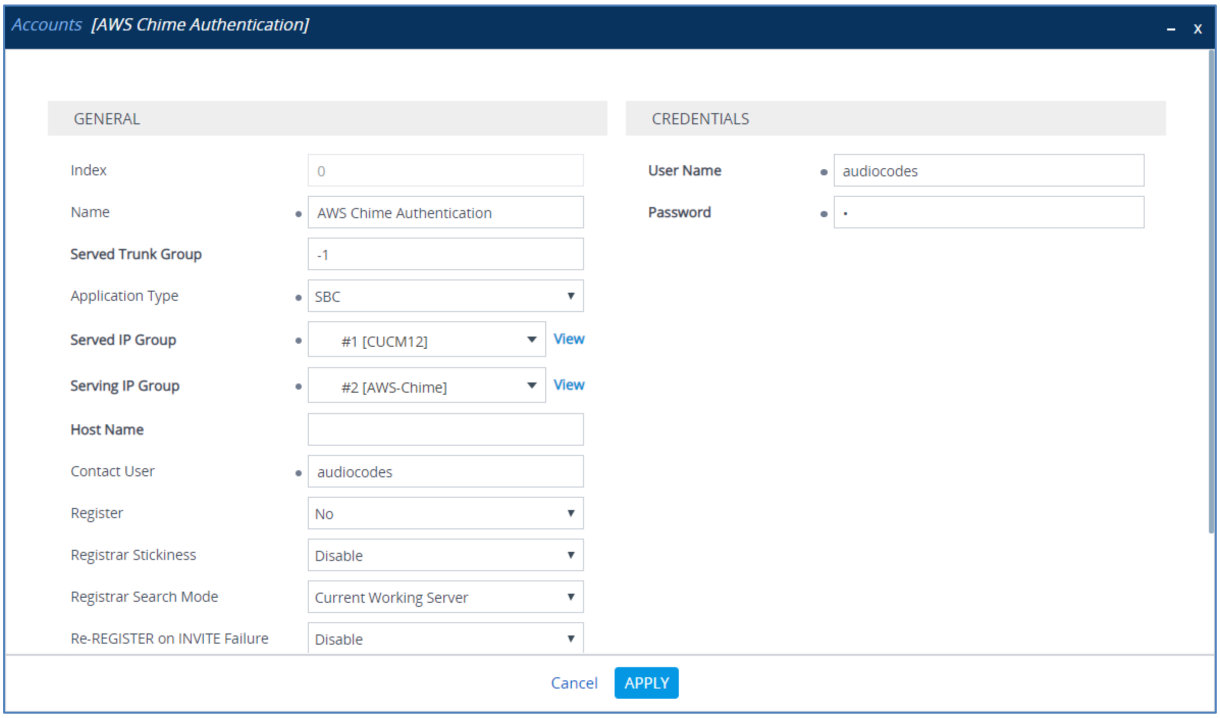
In the interoperability test topology, the Served IP Group is CUCM Server and the Serving IP Group is AWS Chime Voice Connector.

➢ **To configure a SIP account for authentication:**

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
Name	**AWS Chime Authentication** (arbitrary descriptive name)
Application Type	**SBC**
Served IP Group	**CUCM12**
Serving IP Group	**AWS-Chime**
Contact User	**audiocodes** (per Chime Voice Connector configuration)
Username	According to Chime Voice Connector configuration
Password	According to Chime Voice Connector configuration

Figure 5-31: Configuring a SIP Authentication Account



4. Click **Apply**.

5.14 Step 14: Miscellaneous Configuration

This section describes miscellaneous SBC configuration.

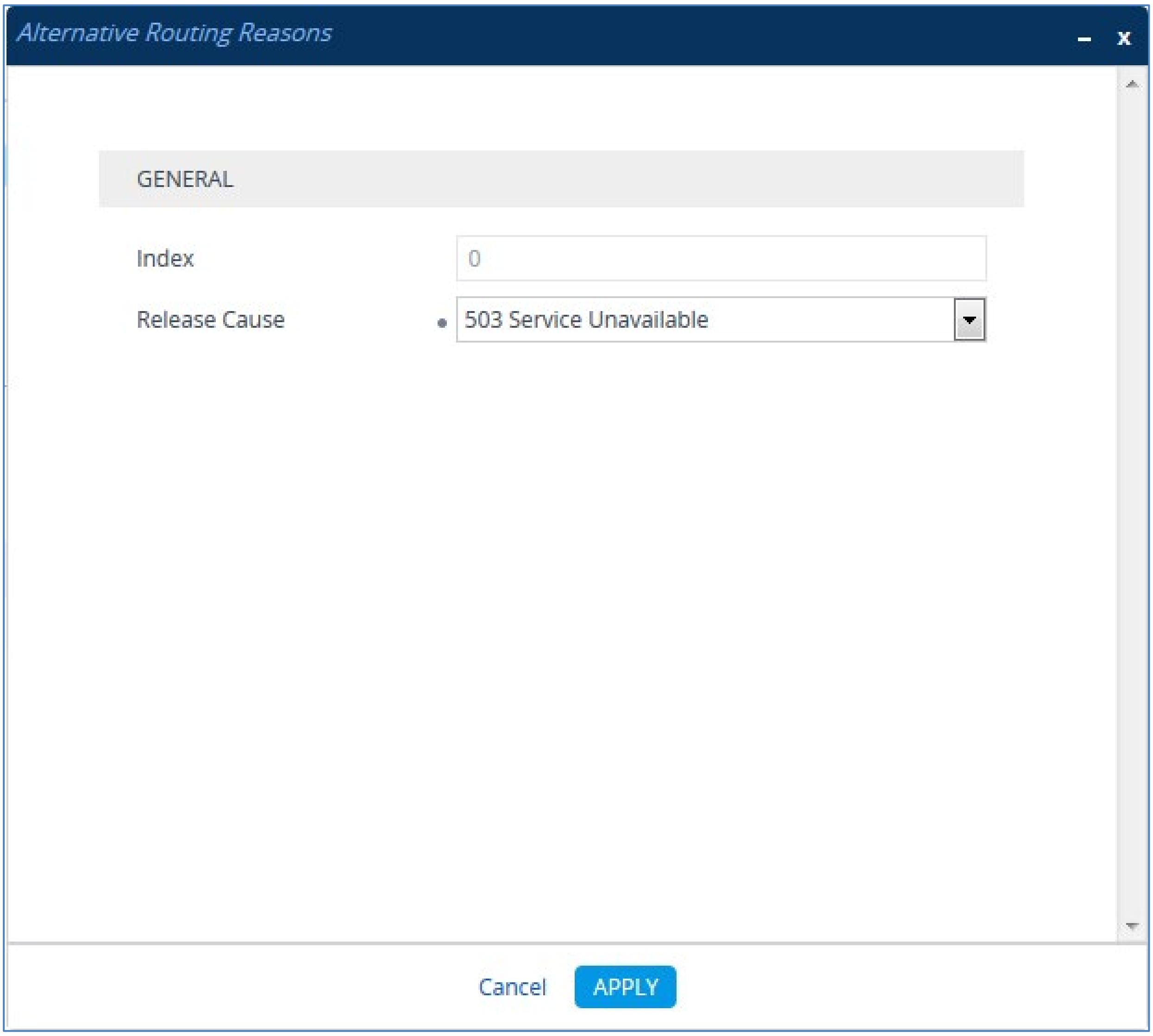
5.14.1 Step 14a: Configure SBC Alternative Routing Reasons

This step describes how to configure the SBC's handling of SIP 503 responses received for outgoing SIP dialog-initiating methods, e.g., INVITE, OPTIONS, and SUBSCRIBE messages. In this case, the 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 5-32: SBC Alternative Routing Reasons Table



4. Click **Apply**.

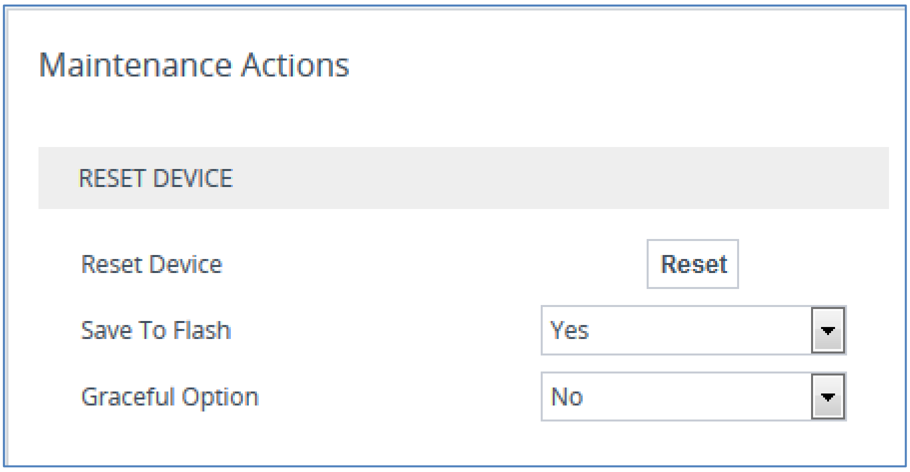
5.15 Step 15: Reset the SBC

After you have completed the configuration of the SBC described in this chapter, save ("burn") the configuration to the 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 5-33: Resetting the 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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A AudioCodes INI File

The *ini* configuration file of the SBC, corresponding to the Web-based configuration as described in Section 4 on page 19, is shown below:

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

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

;Board: M800B
;Board Type: 72
;Serial Number: 5299378
;Slot Number: 1
;Software Version: 7.20A.252.011
;DSP Software Version: 5014AE3_R => 710.16
;Board IP Address: 10.15.77.55
;Board Subnet Mask: 255.255.0.0
;Board Default Gateway: 10.15.0.1
;Ram size: 512M   Flash size: 64M   Core speed: 500Mhz
;Num of DSP Cores: 3
;Num of physical LAN ports: 4
;Profile: NONE
;;;Key features:;Board Type: M800B ;Coders: G723 G729 G728 NETCODER GSM-
FR GSM-EFR AMR EVRC-QCELP G727 ILBC EVRC-B AMR-WB G722 EG711 MS_RTA_NB
MS_RTA_WB SILK_NB SILK_WB SPEEX_NB SPEEX_WB OPUS_NB OPUS_WB ;DSP Voice
features: RTCP-XR ;DATA features: ;Channel Type: DspCh=30 IPMediaDspCh=30
;HA ;E1Trunks=1 ;T1Trunks=1 ;FXSPorts=4 ;FXOPorts=0 ;BRITrunks=4 ;IP
Media: Conf VXML ;QOE features: VoiceQualityMonitoring MediaEnhancement
;Security: IPSEC MediaEncryption StrongEncryption EncryptControlProtocol
;Control Protocols: MGCP SIP SBC=250 TEAMS MSFT FEU=100 TestCall=100
;Default features:;Coders: G711 G726;

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


[SYSTEM Params]

SyslogServerIP = 10.10.10.10
EnableSyslog = 0
NTPServerUTCOffset = 7200
HALocalMAC = '00908f50dcb2'
TR069ACSPASSWORD = '$1$gQ=='
TR069CONNECTIONREQUESTPASSWORD = '$1$gQ=='
NTPServerIP = '8.8.8.8'
```

```
SBCWizardFilename = 'templates4.zip'

[BSP Params]

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

[Analog Params]


[ControlProtocols Params]

AdminStateLockControl = 0

[PSTN Params]

LineCode = 2
V5ProtocolSide = 0

[Voice Engine Params]

ENABLEMEDIASECURITY = 1
PLThresholdLevelsPerMille_0 = 5
PLThresholdLevelsPerMille_1 = 10
PLThresholdLevelsPerMille_2 = 20
PLThresholdLevelsPerMille_3 = 50
CallProgressTonesFilename = 'usa_tones_13.dat'

[WEB Params]

Languages = 'en-US', '', '', '', '', '', '', '', '', ''

[SIP Params]

GWDEBUGLEVEL = 5
SIPGATEWAYNAME = 'audiocodes@test'
USEGATEWAYNAMEFOROPTIONS = 1
MSLDAPPRIMARYKEY = 'telephoneNumber'
USEPINGPONGKEEPALIVE = 1
ENERGYDETECTORCMD = 587202560
ANSWERDETECTORCMD = 10486144


[ 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, 10.15.77.55, 16, 10.15.0.1, "LAN_IF",
10.15.27.1, , "vlan 1";
InterfaceTable 1 = 5, 10, 195.189.192.150, 24, 195.189.192.129, "WAN_IF",
80.179.52.100, 80.179.55.100, "vlan 2";

[ \InterfaceTable ]


[ 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", 4, 0, "DEFAULT", "DEFAULT", 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 ]


[ 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_SBCRemoveUnKnownCrypto;
IpProfile 1 = "CUCM12", 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,
"", "", "", 0, 2, 0, 0, 0, 0, 0, 8, 300, 400, 0, 0, 0, "", 0, 0, 1, 3, 0,
2, 2, 1, 0, 0, 1, 0, 1, 0, 0, 0, 0, 0, 1, 0, 0, 0, 0, 0, 0, 1, 0, 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, 0;
IpProfile 2 = "AWS-Chime", 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, "",
"AudioCodersGroups_0", 0, 0, "", "", "", 0, 2, 1, 0, 0, 1, 0, 8, 300,
400, 0, 0, 0, "", 0, 0, 1, 3, 0, 2, 2, 0, 0, 0, 1, 0, 1, 0, 0, 0, 0, 0,
1, 0, 0, 0, 0, 0, 0, 1, 0, 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, 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, 50, 6499, 0, "",
"", 0;
CpMediaRealm 1 = "MRWan", "WAN_IF", "", "", "", 7000, 50, 7499, 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_SCTPSecondaryNetworkInterface, SIPInterface_ApplicationType,
SIPInterface_UDPPort, SIPInterface_TCPPort, SIPInterface_TLSPort,
SIPInterface_SCTPPort, SIPInterface_AdditionalUDPPorts,
SIPInterface_AdditionalUDPPortsMode, 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_CallSetupRulesSetId;
SIPInterface 0 = "SIPInterface_LAN", "LAN_IF", "", 2, 5060, 5060, 0, 0,
"", 0, "DefaultSRD", "", "default", -1, 0, 500, -1, 0, "MRLan", 0, -1, -
1, -1, 0, 0, "", "", -1;
SIPInterface 1 = "SIPInterface_WAN", "WAN_IF", "", 2, 0, 5060, 5061, 0,
"", 0, "DefaultSRD", "", "default", -1, 0, 500, -1, 0, "MRWan", 0, -1, -
1, -1, 0, 1, "", "", -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, "",
"", "SIPInterface_LAN", "", "", 1, 1, 10, -1;
ProxySet 1 = "CUCM12", 1, 60, 0, 0, "DefaultSRD", 0, "", -1, -1, "", "",
"SIPInterface_LAN", "", "", 1, 1, 10, -1;
ProxySet 2 = "AWS-Chime", 1, 60, 0, 0, "DefaultSRD", 0, "", -1, -1, "",
"", "SIPInterface_WAN", "", "", 1, 1, 10, -1;

[ \ProxySet ]


[ IPGroup ]

FORMAT IPGroup_Index = IPGroup_Type, IPGroup_Name, IPGroup_ProxySetName,
IPGroup_SIPGroupName, IPGroup_ContactUser, IPGroup_SipReRoutingMode,
IPGroup_AlwaysUseRouteTable, IPGroup_SRDName, IPGroup_MediaRealm,
IPGroup_ClassifyByProxySet, IPGroup_ProfileName,
IPGroup_MaxNumOfRegUsers, IPGroup_InboundManSet, IPGroup_OutboundManSet,
IPGroup_RegistrationMode, IPGroup_AuthenticationMode, IPGroup_MethodList,
IPGroup_SBCServerAuthType, IPGroup_OAuthHTTPService,
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_ProxyKeepAliveUsingIPG;
IPGroup 0 = 0, "Default_IPG", "ProxySet_0", "", "", -1, 0, "DefaultSRD",
"", 0, "", -1, -1, -1, 0, 0, "", -1, "", 0, -1, -1, "", "", "$1$gQ==", 0,
"", "", 0, "", "", 0, 0, "default", 0, 0, -1, 0, 0, 0, "", -1, "", 0, 0,
"", 0;
IPGroup 1 = 0, "CUCM12", "CUCM12",
"dt3ynfnrl41vhejg9rtlfz.voiceconnector.chime.aws", "", -1, 0,
"DefaultSRD", "MRLan", 1, "CUCM12", -1, -1, 2, 0, 0, "", -1, "", 0, -1, -
```

```
1, "", "Admin", "$1$aCkNBwIC", 0, "", "", 0, "", "", 0, 0, "default", 0,
0, -1, 0, 0, 0, "", -1, "", 0, 0, "", 0;
IPGroup 2 = 0, "AWS-Chime", "AWS-Chime",
"dt3ynfnrl41vhejg9rtlfz.voiceconnector.chime.aws", "", -1, 0,
"DefaultSRD", "MRWan", 1, "AWS-Chime", -1, -1, -1, 0, 0, "", -1, "", 0, -
1, -1, "", "Admin", "$1$aCkNBwIC", 0, "", "", 0, "", "", 0, 0, "default",
0, 0, -1, 0, 0, 1, "", -1, "", 0, 0, "", 1;

[ \IPGroup ]


[ ProxyIp ]

FORMAT ProxyIp_Index = ProxyIp_ProxySetId, ProxyIp_ProxyIpIndex,
ProxyIp_IpAddress, ProxyIp_TransportType, ProxyIp_Priority,
ProxyIp_Weight;
ProxyIp 0 = "1", 0, "10.15.28.101:5060", 1, 0, 0;
ProxyIp 1 = "2", 0,
"dt3ynfnrl41vhejg9rtlfz.voiceconnector.chime.aws:5060", 1, 0, 0;

[ \ProxyIp ]


[ Account ]

FORMAT Account_Index = Account_AccountName, 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_ReRegisterOnInviteFailure;
Account 0 = "AWS Chime Authentication", -1, "CUCM12", "AWS-Chime",
"audiocodes", "$1$S3p+fno=", "", "audiocodes", 0, 0, 0, 0, 2, 0, 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, 13, "", "", "", 0, -1, 0,
0, "", "", "", "", "default", "Reply(Response='200')";
IP2IPRouting 1 = "CUCM12 to AWS-Chime", "Default_SBCRoutingPolicy",
"CUCM12", "*", "*", "*", "*", 0, "", "Any", 0, -1, 0, "AWS-Chime", "",
"", 0, -1, 0, 0, "", "", "", "", "default", "";
IP2IPRouting 2 = "AWS-Chime to CUCM12", "Default_SBCRoutingPolicy", "AWS-
Chime", "*", "*", "", "*", 0, "", "Any", 0, -1, 0, "CUCM12", "", "", 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 = "Do Nothing", "Default_SBCRoutingPolicy", 0,
"CUCM12", "AWS-Chime", "*", "*", "+", "*", "*", "", 0, "Any", 0, 1, 0, 0,
255, "", "", 0, "", "";
IPOutboundManipulation 1 = "Add +", "Default_SBCRoutingPolicy", 0,
"CUCM12", "AWS-Chime", "*", "*", "*", "*", "*", "", 0, "Any", 0, 1, 0, 0,
255, "+", "", 0, "", "";
IPOutboundManipulation 2 = "Strip + towards CUCM12",
"Default_SBCRoutingPolicy", 0, "AWS-Chime", "CUCM12", "*", "*", "+", "*",
"*", "", 0, "Any", 0, 1, 1, 0, 255, "", "", 0, "", "";

[ \IPOutboundManipulation ]


[ MessageManipulations ]

FORMAT MessageManipulations_Index =
MessageManipulations_ManipulationName, MessageManipulations_ManSetID,
MessageManipulations_MessageType, MessageManipulations_Condition,
MessageManipulations_ActionSubject, MessageManipulations_ActionType,
MessageManipulations_ActionValue, MessageManipulations_RowRole;
MessageManipulations 0 = "Change RUI host toward CUCM12", 2, "Any", "",
"header.request-uri.url.host", 2, "param.message.address.dst.address", 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 ]


[ AudioCoders ]

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

[ \AudioCoders ]
```

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

AudioCodes Inc.
200 Cottontail Lane
Suite A101E
Somerset NJ 08873
Tel: +1-732-469-0880
Fax: +1-732-469-2298

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

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