Configuration Note

AudioCodes Mediant™ Family of Media Gateways & Session Border Controllers

Connecting Zoom Phone Carrier Peering with AudioCodes SBC

Table of Contents

List of Figures

List of Tables

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Notice

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Abbreviations and Terminology

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

Related Documentation

Document Name
Mediant 500 Gateway & E-SBC User's Manual
Mediant 500L Gateway & E-SBC User's Manual
Mediant 800 Gateway & E-SBC User's Manual
Mediant 1000B Gateway & E-SBC User's Manual
Mediant 2600 E-SBC User's Manual
Mediant 4000 SBC User's Manual
Mediant 9000 SBC User's Manual
Mediant Software SBC User's Manual
Gateway and SBC CLI Reference Guide
SIP Message Manipulation Reference Guide
AudioCodes Configuration Notes

Document Revision Record

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29340	Initial document release.
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Documentation Feedback

AudioCodes continually strives to produce high quality documentation. If you have any comments (suggestions or errors) regarding this document, please fill out the Documentation Feedback form on our website at https://online.audiocodes.com/documentation-feedback.

1 Introduction

This document describes how to connect Zoom Phone System to multiple customers using AudioCodes' SBC in Hosting mode and refers to the AudioCodes SBC configuration only.

This document is intended for IT or telephony professionals.

Note: To zoom in on screenshots of Web interface configuration examples, press Ctrl and +.

1.1 About the Zoom Phone System

Zoom Phone is a fully featured cloud PBX designed with security, reliability, scalability and centralized management in mind. Zoom Phone was built from the ground up to seamlessly integrate with the Zoom Collaboration platform to deliver a feature-rich UCaaS user experience. Zoom Phone offers various deployment options providing organizations with the flexibility to migrate and deploy the platform in a manner that best suits their requirements. Zoom Phone leverages global carrier relationships to deliver PSTN connectivity in many regions of the world offering phone number portability to Zoom in most regions thereby simplifying the telephony environment with one partner for your PBX and PSTN connectivity needs. While native Zoom Phone meets the requirements of most organizations, it’s understood that some organizations have environments that may need additional functionality for global support or migration strategies. For organizations with such diverse requirements of their telephony environments, Zoom’s Premise Peering solution is offered.

Zoom Phone Premise Peering provides organizations with flexibility and seamless options to migrate their voice workloads to the cloud. This is accomplished by providing two connection types; Premise Peering PSTN (formally referred to as Bring Your Own Carrier - BYOC) and/or Premise Peering PBX (formally referred to as Bring Your Own PBX - BYOP). Zoom Phone Premise Peering PSTN enables organizations to leverage their existing telephony carrier PSTN environment for Zoom Phone connectivity. Using this functionality organizations can connect Zoom Phone with virtually any telephony carrier.

1.2 About AudioCodes SBC Product Series

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

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 the 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. The SBC can be offered as a Virtualized SBC, supporting the following platforms: Hyper-V, AWS, AZURE, AWP, KVM and VMWare.

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2 Configuring Zoom Phone System

For configuring the Zoom Phone System, refer to Zoom Help Center at https://support.zoom.us/hc/en-us/articles/360001297663-Getting-started-with-Zoom-Phone-admin-.

Notes: Before you begin configuration:

- Contact your Zoom Representative to enable SIP groups and set up SIP trunks that are directed toward your SBC for your Zoom Phone account.
- Make sure you have Zoom Portal admin credentials. Be aware that each customer needs to have a Zoom Phone admin account and all Zoom Phone related configuration will be done by the customer and not by the carrier.

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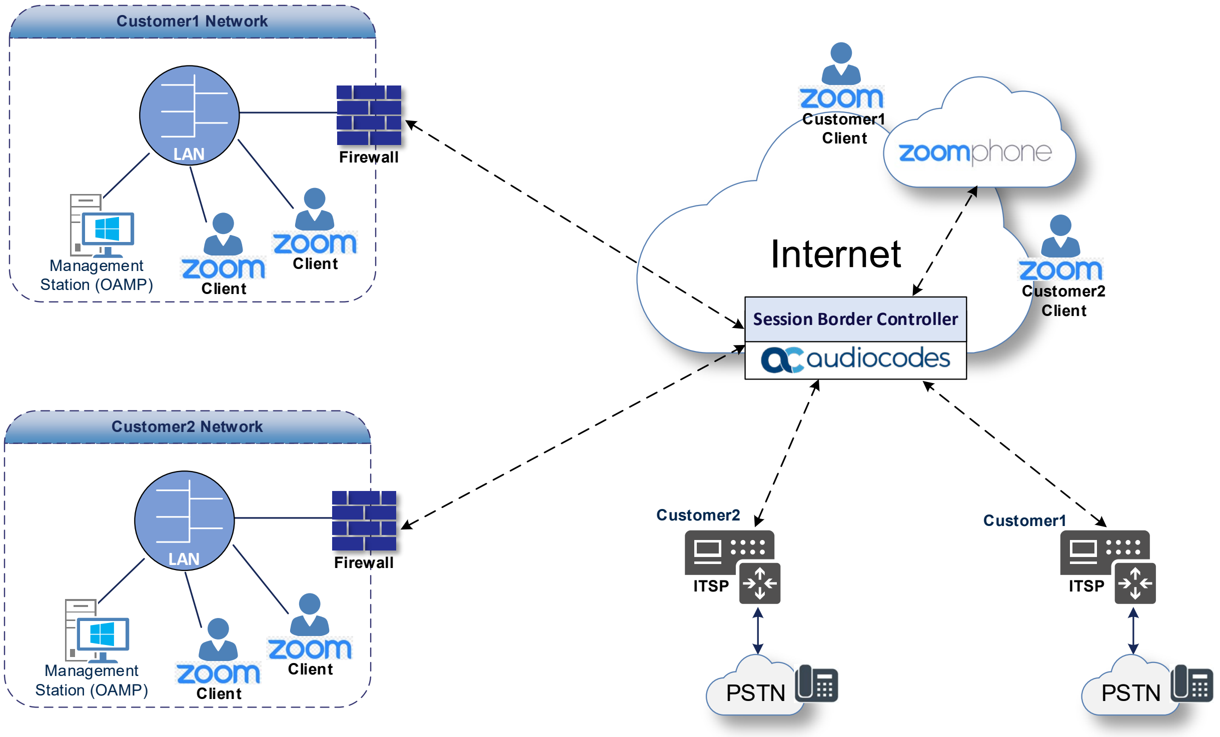
3 Configuring AudioCodes' SBC

This section shows how to configure AudioCodes' SBC for internetworking with Zoom Phone System. The figure below shows an example of the connection topology for the Zoom Phone System Premise Peering Mode. Multiple connection entities are shown in the figure:

- Zoom Phone Systems
- Service Provider Customers SIP Trunks

Figure 3-1: Connection Topology - Network Interfaces



Note: This document shows how to pair between the AudioCodes' hosting SBC and the Zoom Phone System with a Customers SIP Trunks. For detailed configuration of other entities in the deployment such as the SIP Trunk Provider and the local IP-PBX, see AudioCodes' SIP Trunk Configuration Notes (in the Interoperability suite of documents).

3.1 Prerequisites

Before you begin configuration, make sure you have obtained the following for each Hosting SBC you wish to pair:

- Public IP address
- Public certificate that is issued by one of the Zoom supported CAs

3.2 Validate AudioCodes SBC License

Zoom has successfully conducted validation tests with AudioCodes' Mediant SBC Ver. 7.20A.258. For implementing the configuration described in this document, the AudioCodes SBC must be installed with a License Key that includes the following features:

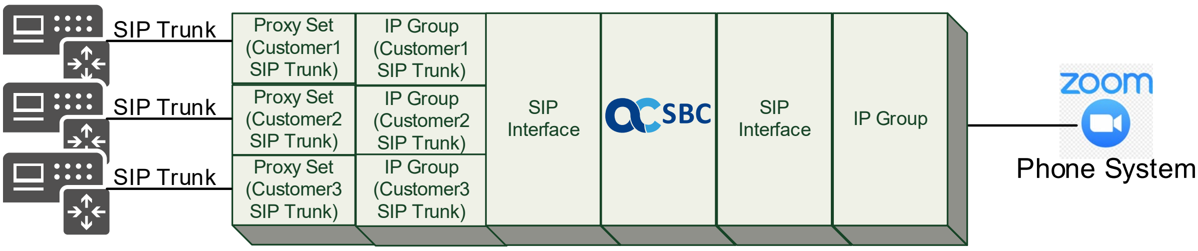
- Number of SBC sessions (based on requirements)
- Transcoding sessions (only if media transcoding is needed)
- Coders (based on requirements)

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

3.3 SBC Configuration Concept

The figure below illustrates the underlying concept of the configuration of AudioCodes' SBC device.

Figure 3-2: SBC Configuration Concept



The routing from the SIP Trunks to Zoom Phone System and vice versa is dependent on the Class 4 switch routing method. The routing decision can be based on:

- Customer DID Range
- Trunk Context (TGRP)
- IP Interface
- SIP Interface (UDP/TCP Port)
- Host name

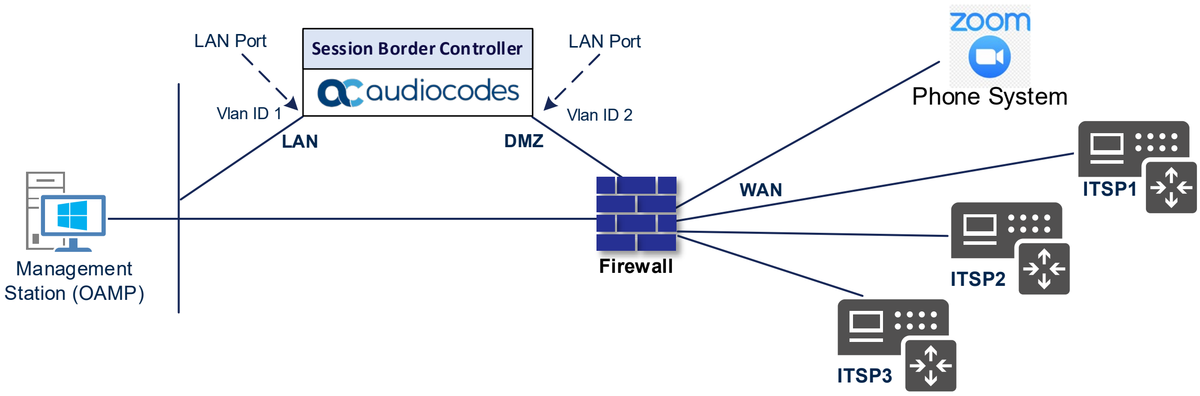
The configuration shown in this document is based on a Customer DID Range using a Dial Plan. For more information, see the AudioCodes' Documentation suite.

3.4 Configure IP Network Interfaces

This section describes how to configure the SBC's IP network interfaces. There are several ways to deploy the SBC:

- SBC interfaces with the following IP entities:
  - Zoom Phone System
  - Customers SIP Trunks
- Physical connection: The type of physical connection depends on the method used to connect to the Enterprise's network. In this example topology, SBC connects to the LAN and DMZ using dedicated Ethernet 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 3-3: Network Interfaces in the Topology with all entities on the WAN



3.4.1 Configure LAN and WAN VLANs

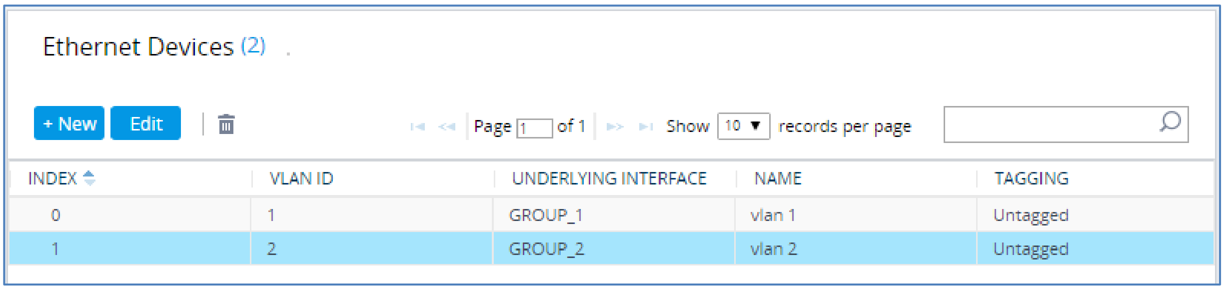
This section describes how to define VLANs for each of the following interfaces:

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

➢ **To configure 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.

Figure 3-4: Configured VLANs in the Ethernet Device Table


Ethernet Devices (2)

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

INDEX	VLAN ID	UNDERLYING INTERFACE	NAME	TAGGING
0	1	GROUP_1	vlan 1	Untagged
1	2	GROUP_2	vlan 2	Untagged

3.4.2 Configure Network Interfaces

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

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

➢ **To configure network parameters for both LAN and WAN interfaces:**

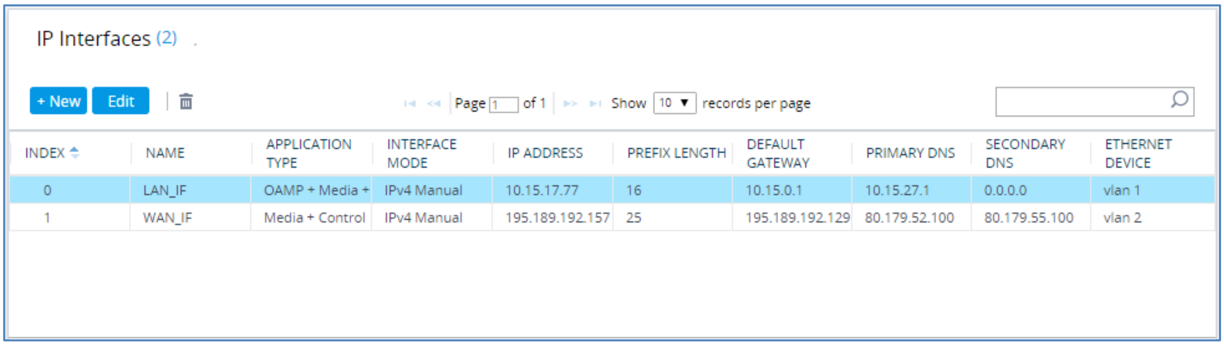
1. Open the IP Interfaces table (**Setup** menu > **IP Network** tab > **Core Entities** folder **> IP Interfaces**).
2. Configure the IP interfaces as follows (your network parameters might be different):

Table 3-1: Configuration Example of the IP Interfaces Table

Index	Application Types	Interface Mode	IP Address	Prefix Length	Gateway	DNS	I/F Name	Ethernet Device
0	OAMP+ Media + Control	IPv4 Manual	10.15.77.77	16	10.15.0.1	10.15.27.1	LAN_IF	vlan 1
1	Media + Control (as this interface points to the internet, enabling OAMP is not recommended)	IPv4 Manual	195.189.192.157 (DMZ IP address of SBC)	25	195.189.192.129 (router's IP address)	According to your Internet provider's instructions	WAN_IF	vlan 2

The configured IP network interfaces are shown below:

Figure 3-5: Configuration Example of the 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

3.5 Configure TLS Context for Zoom

This section describes how to configure the SBC for using a TLS connection with the Zoom Phone System. This configuration is essential for a secure SIP TLS connection.

This certificate module is based on the GoDaddy Certificate Chain. For more certificate structure options, refer to the Zoom Phone System documentation.

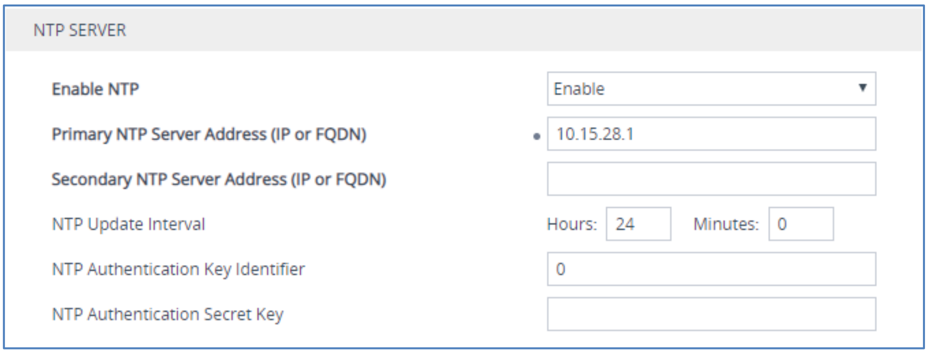
3.5.1 Configure the NTP Server Address

This section describes how to configure the NTP server's IP address. It is recommended to implement an NTP server (Microsoft NTP server or another global server) to ensure that the SBC receives the current date and time. This is necessary for validating certificates of remote parties. It is important, that the NTP server is located on the OAMP IP Interface (LAN_IF in our case) or will be accessible through it.

➢ **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., 10.15.28.1).

Figure 3-6: Configuring NTP Server Address



3. Click **Apply**.

3.5.2 Create a TLS Context for Zoom Phone System

The section below describes how to request a certificate for the SBC WAN interface and configure it, based on the example of the GoDaddy Global Root CA. The certificate is used by the SBC to authenticate the connection with the Zoom Phone System.

The procedure involves the following main steps:

- Create a TLS Context for Zoom Phone System
- Generate a Certificate Signing Request (CSR) and obtain the certificate from a supported Certification Authority
- Deploy the SBC and Root / Intermediate certificates on the SBC

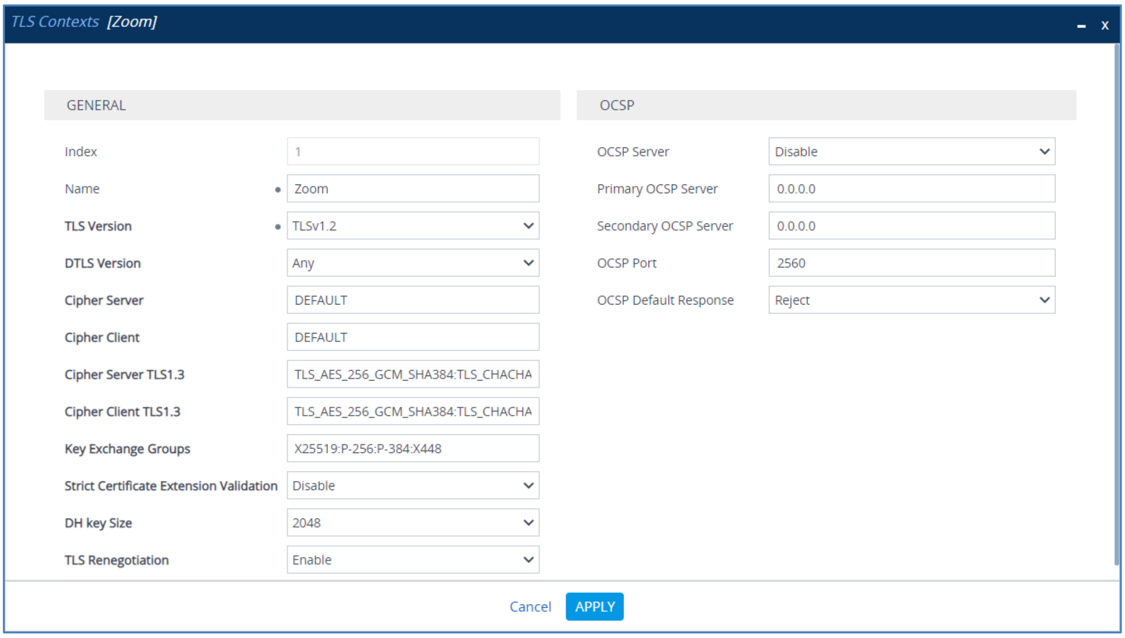
➢ **To create a TLS Context for Zoom Phone System:**

1. Open the TLS Contexts page (**Setup** menu > **IP Network** tab > **Security** folder > **TLS Contexts**).
2. Create a new TLS Context by clicking **+New**, and then configure the parameters using the table below as reference.

Table 3-2: New TLS Context

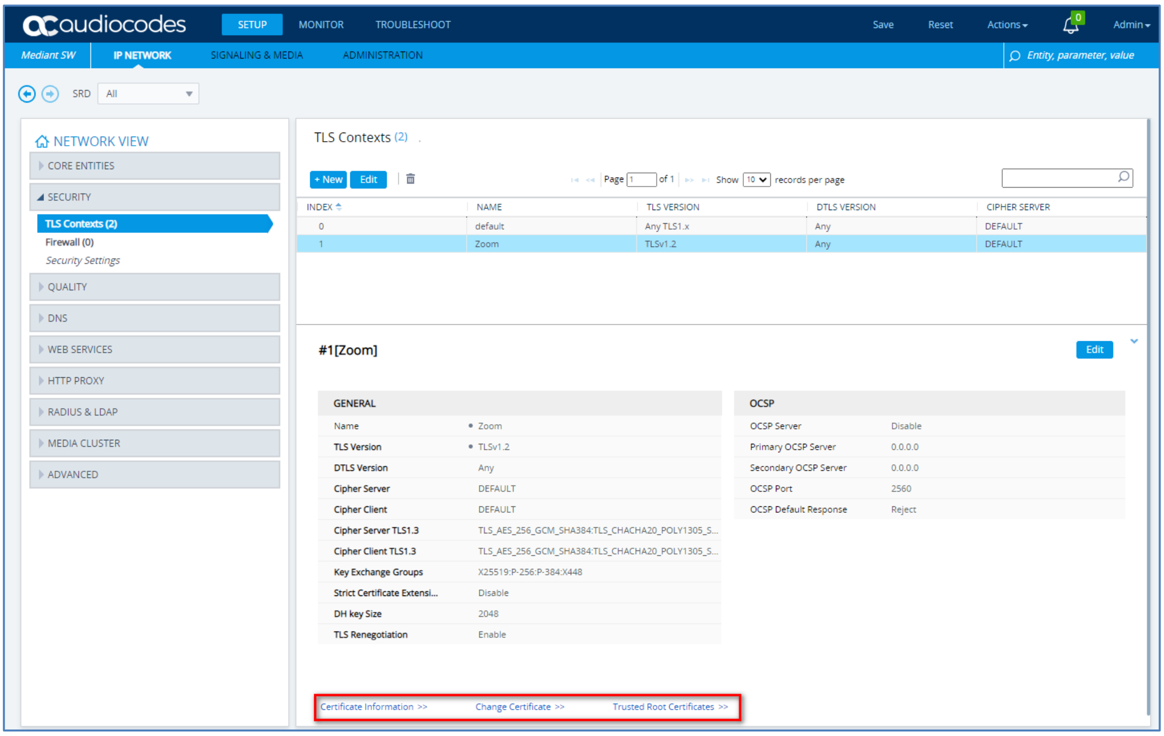
Index	Name	TLS Version
1	Zoom (arbitrary descriptive name)	TLSv1.2
All other parameters can be left unchanged with their default values.		

Figure 3-7: Configuration of TLS Context for Zoom Phone System



3. Click **Apply**; you should see the new TLS Context and option to manage the certificates at the bottom of 'TLS Context' table.

Figure 3-8: TLS Context for Zoom Phone System and Interface to Manage the Certificates



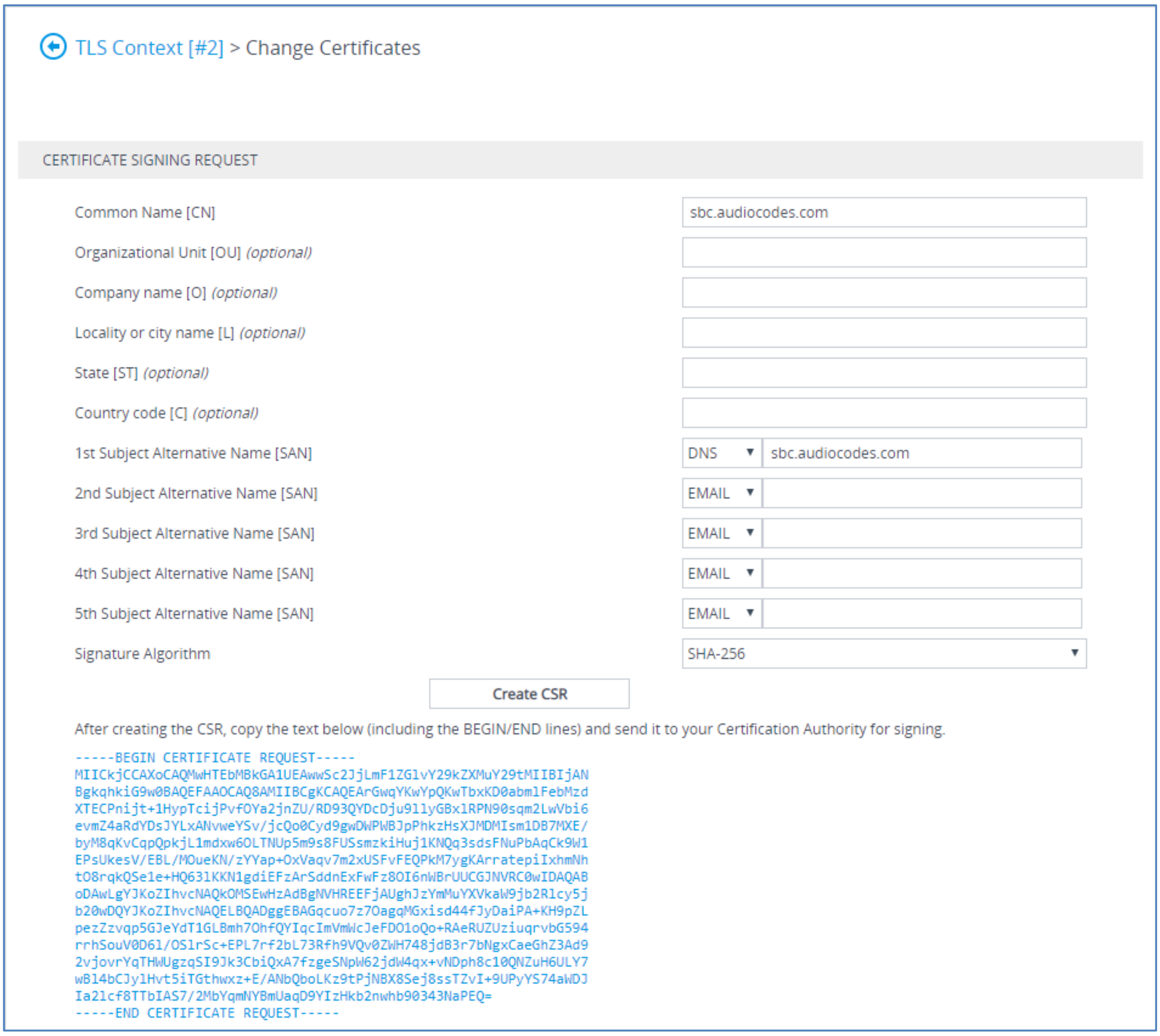
3.5.3 Generate a CSR and Obtain the Certificate from a Supported CA

This section shows how to generate a Certificate Signing Request (CSR) and obtain the certificate from a supported Certification Authority.

➢ **To generate a Certificate Signing Request (CSR) and obtain the certificate from a supported Certification Authority:**

1. Open the TLS Contexts page (**Setup** menu > **IP Network** tab > **Security** folder > **TLS Contexts**).
2. In the TLS Contexts page, select the Zoom TLS Context index row, and then click the Change Certificate link located below the table; the Context Certificates page appears.
3. Under the Certificate Signing Request group, do the following:
   a. In the 'Common Name [CN]' field, enter the SBC FQDN name (for example, sbc.audiocodes.com).
   b. In the '1st Subject Alternative Name [SAN]' field, change the type to 'DNS' and enter the SBC FQDN name (based on example above, sbc.audiocodes.com).
   c. Change the 'Private Key Size' based on the requirements of your Certification Authority. Many CAs do not support private key of size 1024.
   d. To change the key size on TLS Context, go to: **Generate New Private Key and Self-Signed Certificate**, change the 'Private Key Size' and then click **Generate Private-Key**. To use 2048 as a Private Key Size value, you can click **Generate Private-Key** without changing the default key size value.
   e. Fill in the rest of the request fields according to your security provider's instructions.
   f. Click the **Create CSR** button; a textual certificate signing request is displayed in the area below the button:

Figure 3-9: Example of Certificate Signing Request Page



4. Copy the CSR from the line "----BEGIN CERTIFICATE" to "END CERTIFICATE REQUEST----" to a text file (such as Notepad), and then save it to a folder on your computer with the file name, for example certreq.txt.
5. Send certreq.txt file to the Certified Authority Administrator for signing.

3.5.4 Deploy the SBC and Root / Intermediate Certificates on the SBC

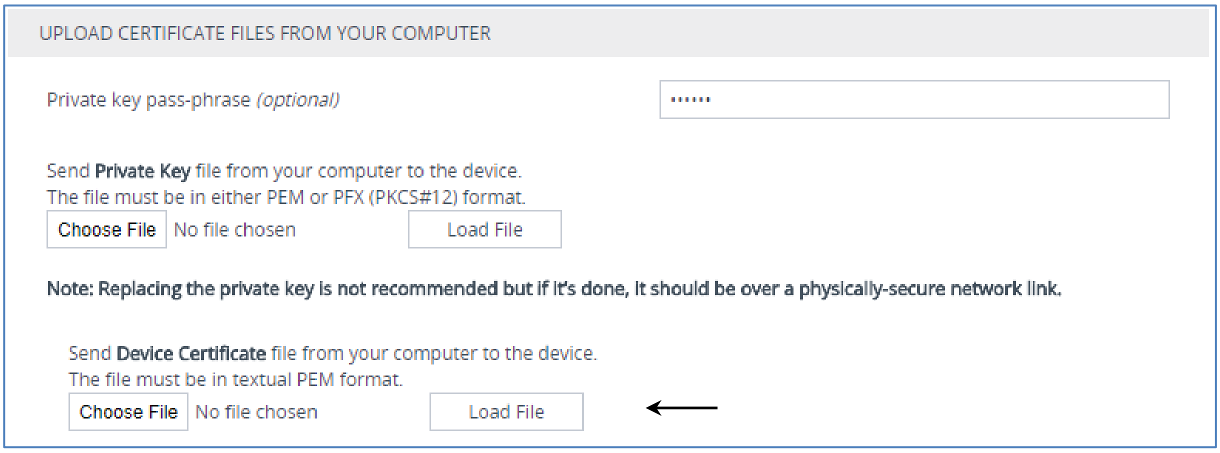
After obtaining the SBC signed and Trusted Root/Intermediate Certificate from the CA, install the following:

- SBC certificate
- Root / Intermediate certificates:

➢ **To install the SBC certificate:**

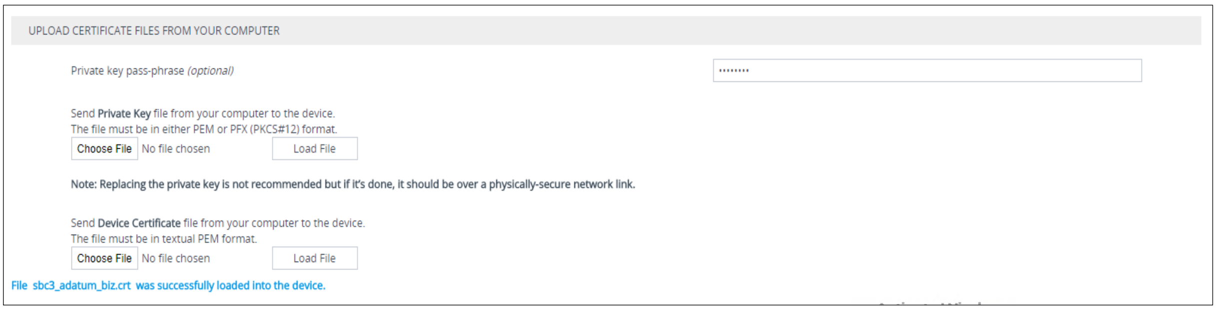
1. In the SBC's Web interface, return to the TLS Contexts page and do the following:
   a. In the TLS Contexts page, select the required TLS Context index row, and then click the **Change Certificate** link located below the table; the Context Certificates page appears.
   b. Scroll down to the Upload certificates files from your computer group, click the **Choose File** button corresponding to the 'Send Device Certificate...' field, navigate to the certificate file obtained from the CA, and then click **Load File** to upload the certificate to the SBC.

Figure 3-10: Uploading the Certificate Obtained from the Certification Authority



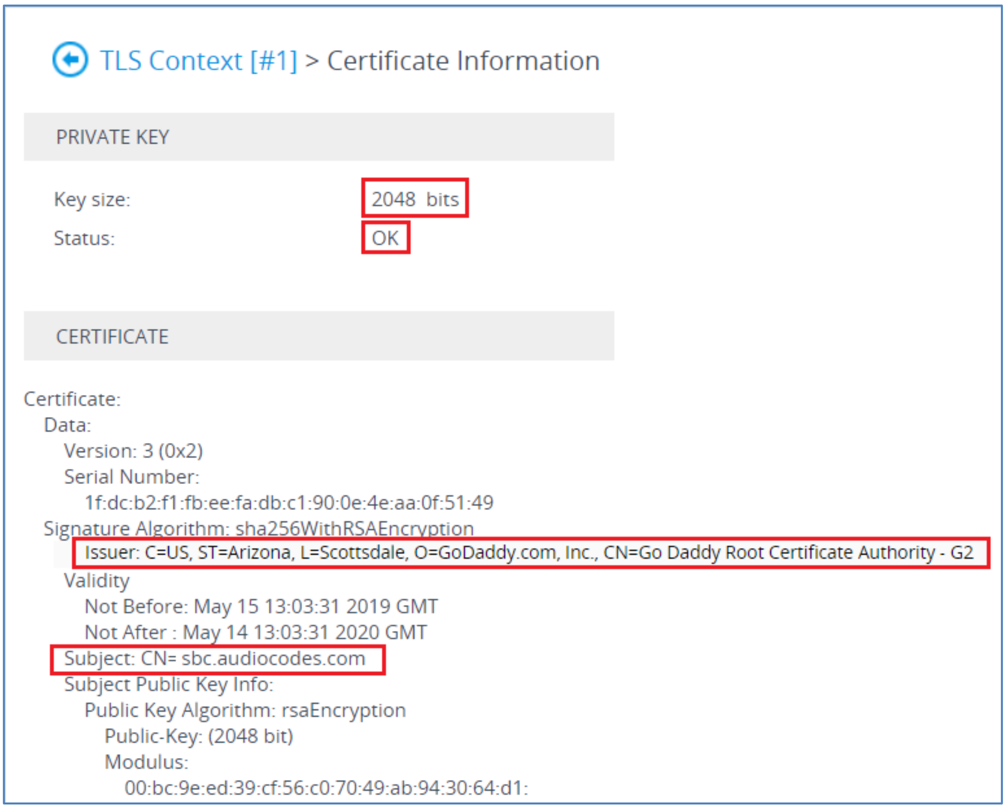
2. Validate that the certificate was uploaded correctly. A message indicating that the certificate was uploaded successfully is displayed in blue in the lower part of the page:

Figure 3-11: Message Indicating Successful Upload of the Certificate



3. In the SBC's Web interface, return to the TLS Contexts page, select the required TLS Context index row, and then click the **Certificate Information** link, located at the bottom of the TLS. Then validate the Key size, certificate status and Subject Name:

Figure 3-12: Certificate Information Example



4. In the SBC's Web interface, return to the TLS Contexts page.
   a. In the TLS Contexts page, select the required TLS Context index row, and then click the Trusted Root **Certificates** link, located at the bottom of the TLS Contexts page; the Trusted Certificates page appears.
   b. Click the **Import** button, and then select all Root/Intermediate Certificates obtained from your Certification Authority to load.
5. Click **OK**; the certificate is loaded to the device and listed in the Trusted Certificates store:

Figure 3-13: Configured Trusted Certificates Page

Note: The above method creates a signed certificate for an explicit device, on which a Certificate Sign Request was generated (and signed with private key).

3.6 Configure Media Realms

Media Realms allow dividing the UDP port ranges for use on different interfaces. In the example below, two Media Realms are configured:

- One for the IP interface towards the Zoom Phone System, with the UDP port starting at 10000 and the number of media session legs 1000 (you need to calculate number of media session legs based on your usage).
- One for the IP interface towards Customers SIP Trunks, with the UDP port range starting at 6000 and the number of media session legs 1000.

➢ To configure Media Realms:

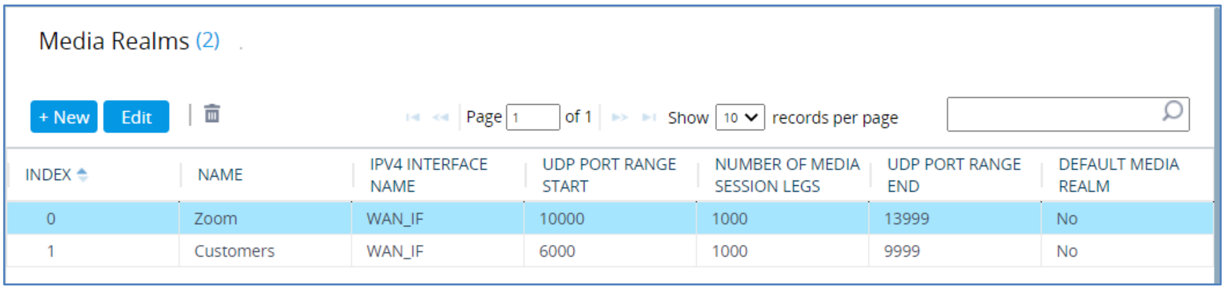
1. Open the Media Realms table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **Media Realms**).
2. Configure Media Realms as follows (you can use the default Media Realm - Index 0 - however modify it):

Table 3-3: Configuration Example Media Realms in Media Realm Table

Index	Name	Topology Location	IPv4 Interface Name	Port Range Start	Number of Media Session Legs
0	Zoom (arbitrary name)	Up	WAN_IF	10000	1000 (media sessions assigned with port range)
1	Customers (arbitrary name)		WAN_IF	6000	1000 (media sessions assigned with port range)

The configured Media Realms are shown in the figure below:

Figure 3-14: Configuration Example Media Realms in Media Realm Table



3.7 Configure SIP Signaling Interfaces

This section shows how to configure SIP Signaling Interfaces. A SIP Interface defines a listening port and type (UDP, TCP, or TLS) for SIP signaling traffic on a specific logical IP network interface (configured in the Interface Table above) and Media Realm.

Note that the configuration of a SIP interface for the SIP Trunks shows an example and your configuration might be different. For specific configuration of interfaces pointing to SIP trunks and/or a third-party PSTN environment connected to the SBC, see the trunk / environment vendor documentation.

AudioCodes also offers a comprehensive suite of documents covering the interconnection between different trunks and equipment.

➢ **To configure a SIP interfaces:**

1. Open the SIP Interface table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **SIP Interfaces**).
2. Configure SIP Interfaces. You can use the default SIP Interface (Index 0), however modify it as shown in the table below. The table below shows an example of the configuration. You can change some parameters according to your requirements.

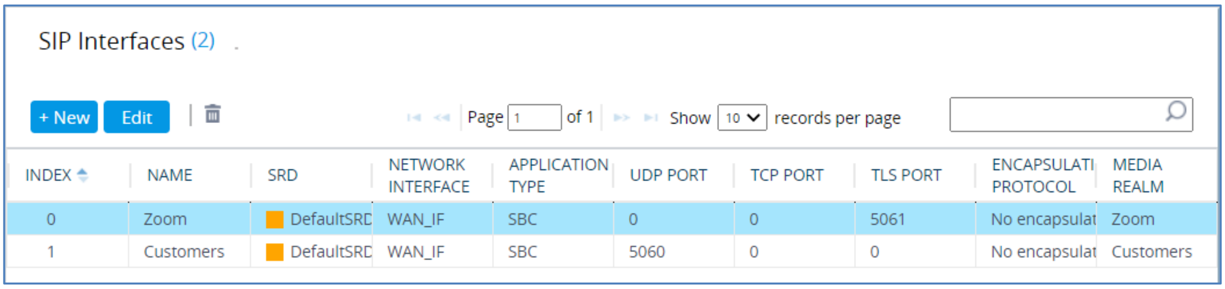
Table 3-4: Configuration Example of SIP Signaling Interfaces

Index	Name	Network Interface	Application Type	UDP Port	TCP Port	TLS Port	Classification Failure Response Type	Media Realm
0	Zoom (arbitrary name)	WAN_IF	SBC	0	0	5061	0 (Recommended to prevent DoS attacks)	Zoom
1	Customers (arbitrary name)	WAN_IF	SBC	5060 (according to Service Provider requirement)	0	0	0 (Recommended to prevent DoS attacks)	Customers

Note: For enhanced security, AudioCodes recommends implementing a Mutual TLS connection with the Zoom Phone System. For required configuration, see Section 3.18.1 on page 44.

The configured SIP Interfaces are shown in the figure below:

Figure 3-15: Configuration Example of SIP Signaling Interfaces



3.8 Configure Proxy Sets and Proxy Address

3.8.1 Configure Proxy Sets

The Proxy Set and Proxy Address defines TLS parameters, IP interfaces, FQDN and the remote entity's port. Proxy Sets can also be used to configure load balancing between multiple servers. The example below covers configuration of a Proxy Sets for Zoom Phone System and Customers SIP Trunks. Note that the configuration of a Proxy Set for the SIP Trunks shows an example and your configuration might be different. For specific configuration of interfaces directed to SIP Trunks and/or the third-party PSTN environment connected to the SBC, see the trunk/environment vendor's documentation. AudioCodes also offers a comprehensive suite of documents covering the interconnection between different trunks and the equipment.
The Proxy Sets will later be applied to the VoIP network by assigning them to IP Groups.

➢ **To configure a Proxy Sets:**

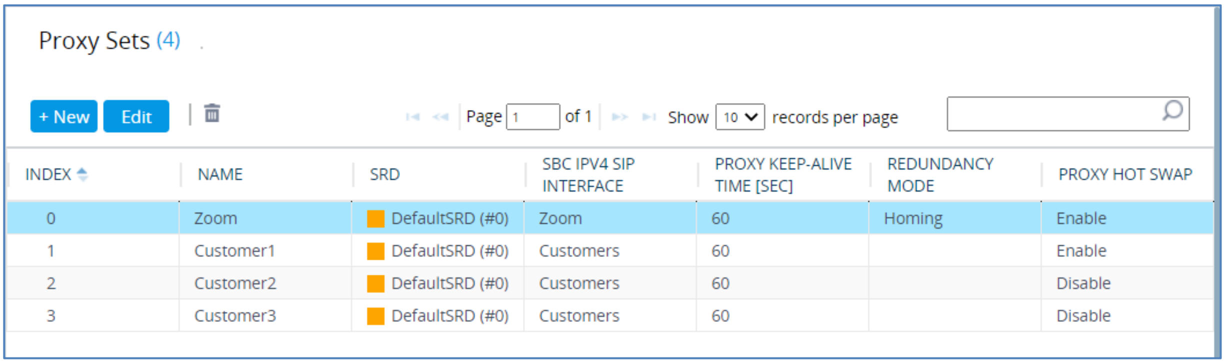
1. Open the Proxy Sets table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **Proxy Sets**).
2. Configure Proxy Sets as shown in the table below:

Table 3-5: Configuration Example Proxy Sets in Proxy Sets Table

Index	Name	SBC IPv4 SIP Interface	TLS Context Name	Proxy Keep-Alive	Redundancy Mode	Proxy Hot Swap
0	Zoom (arbitrary name)	Zoom	Zoom	Using Options	Homing	Enable
1	Customer1 (arbitrary name)	Customers	Default	Using Options	-	-
2	Customer2 (arbitrary name)	Customers	Default	Using Options	-	-
3	Customer3 (arbitrary name)	Customers	Default	Using Options	-	-

The configured Proxy Sets are shown in the figure below:

Figure 3-16: Configuration Example Proxy Sets in Proxy Sets Table



INDEX	NAME	SRD	SBC IPV4 SIP INTERFACE	PROXY KEEP-ALIVE TIME [SEC]	REDUNDANCY MODE	PROXY HOT SWAP
0	Zoom	DefaultSRD (#0)	Zoom	60	Homing	Enable
1	Customer1	DefaultSRD (#0)	Customers	60		Enable
2	Customer2	DefaultSRD (#0)	Customers	60		Disable
3	Customer3	DefaultSRD (#0)	Customers	60		Disable

Note: On Hybrid SBCs (with Onboard PSTN interfaces) it's recommended to leave Proxy Set 0 unconfigured for possible future use for PSTN Fallback.

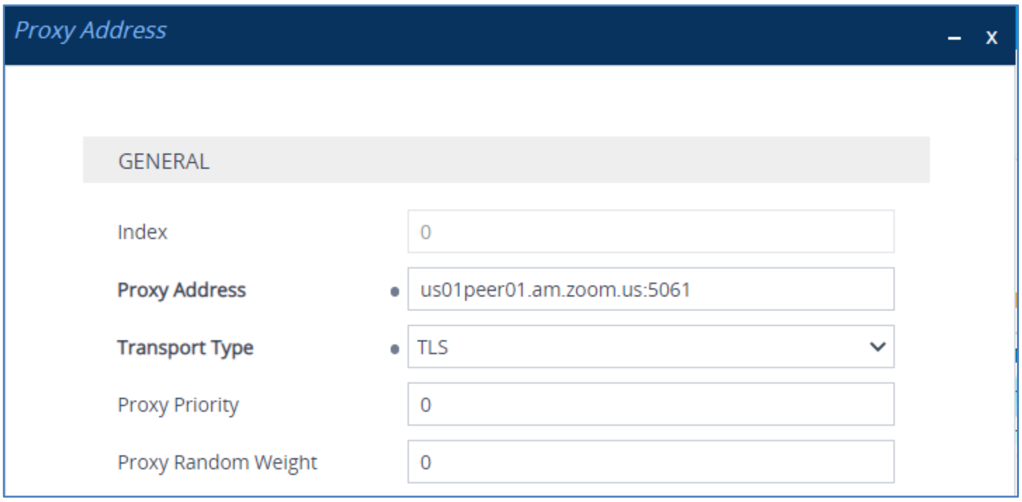
3.8.2 Configure a Proxy Address

This section shows how to configure a Proxy Address.

➢ **To configure a Proxy Address for Zoom:**

1. Open the Proxy Sets table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **Proxy Sets**) click the **Proxy Set Zoom**, and then click the Proxy Address link located below the table; the Proxy Address table opens.
2. Click **+New**; the following dialog box appears:

Figure 3-17: Configuring Proxy Address for Zoom Phone System Interface



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

Table 3-6: Configuration Proxy Address for Zoom Phone System

Index	Proxy Address	Transport Type	Proxy Priority	Proxy Random Weight
0	us01peer01.am.zoom.us:5061	TLS	0	0
1	us01peer01.fr.zoom.us:5061	TLS	0	0

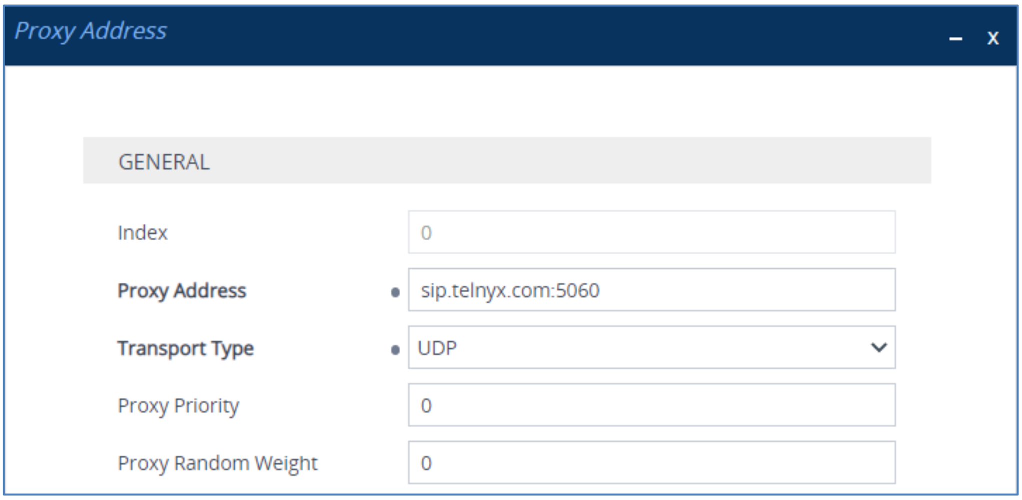
4. Click **Apply**.

Note: The current example is based on configuration Zoom Europe Data Center's IP address (FQDN). In your implementation, the IP address may be different according to your region. Refer to Appendix A on page 49 for a list of FQDNs / IP addresses of other Zoom Regional Data Centers.

➢ To configure a Proxy Address for Customers SIP Trunks:

1. Open the Proxy Sets table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **Proxy Sets**) click the Proxy Set Customer1, and then click the Proxy Address link located below the table; the Proxy Address table opens.
2. Click **+New**; the following dialog box appears:

Figure 3-18: Configuring Proxy Address for Customer 1 SIP Trunk



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

Table 3-7: Configuration Proxy Address for Customer 1 SIP Trunk

Index	Proxy Address	Transport Type	Proxy Priority	Proxy Random Weight
0	sip.telnyx.com:5060 (SIP Trunk IP / FQDN and port)	UDP	0	0

4. Click **Apply**.

3.9 Configure the Dial Plan Table (Customer DIDs)

For deployments requiring hundreds of routing rules (which may exceed the maximum number of rules that can be configured in the IP-to-IP Routing table), you can employ tags to represent the many different calling (source URI user name) and called (destination URI user name) prefix numbers in your routing rules. Tags are typically implemented when you have users of many different called and/or calling numbers that need to be routed to the same destination (e.g., IP Group or IP address). In such a scenario, instead of configuring many routing rules to match all the required prefix numbers, you need only to configure a single routing rule using the tag to represent all the possible prefix numbers.

The Dial Plan (e.g., Customers) will be configured with a customer tag per prefix.

➢ **To configure Dial Plans:**

1. Open the Dial Plan table (**Setup** menu > **Signaling & Media** tab **> SIP Definitions** folder > **Dial Plan**).
2. Click **New** and then configure a Dial Plan name (e.g., Customers) according to the parameters described in the table below.
3. Click **Apply**.
4. In the Dial Plan table, select the row for which you want to configure dial plan rules and then click the Dial Plan Rule link located below the table; the Dial Plan Rule table appears.
5. Click **New**; the following dialog box appears:

Figure 3-19: Dial Plan Rule Table - Add Dialog Box

6. Configure a dial plan rule according to the parameters described in the table below.

Table 3-8: Dial Plan Carrier Customers

Index	Name	Prefix	Tag
0	Customer1	+19098[0000-9999]	Customer1_Name (arbitrary name)
1	Customer 2	+17093[0000-9999]	Customer2_Name (arbitrary name)
2	Customer 3	+18097[0000-9999]	Customer3_Name (arbitrary name)

7. Click **Apply** and then save your settings to flash memory.

3.10 Configure Call Setup Rules

This section describes how to configure Call Setup Rules based on customer DID range (Dial Plan). Call Setup rules define various sequences that are run upon receipt of an incoming call (dialog) at call setup, before the device routes the call to its destination.

Configured Call Setup Rules need be assigned to a specific IP Group.

➢ **To configure a Call Setup Rules based on customer DID range (Dial Plan):**

1. Open the Call Setup Rules table (**Setup** menu > **Signaling & Media** tab > **SIP Definitions** folder > **Call Setup Rules**).
2. Click **New** and configure Call Setup rules according to the parameters described in the table below.

Table 3-9: Call Setup Rules Table

Index	Rules Set ID	Name	Query Type	Query Target	Search Key	Condition	Action Subject	Action Type	Action Value
0	0	Customer DstTags	Dial Plan	Customers	Param.Call.Src.User	DialPlan.Found exists	DstTags	Modify	DialPlan.Result
1	0	X-TO-CARRIER Header				Header.X-TO-CARRIER exists	Var.Session.X-TO-CARRIER	Modify	Header.X-TO-CARRIER
2	1	Zoom DstTags					DstTags	Modify	'Zoom'
3	1	X-TO-CARRIER to Zoom	Dial Plan	Customers	Param.Call.Dst.User	DialPlan.Found exists	Var.Session.X-TO-CARRIER	Modify	DialPlan.Result

3. Click **Apply** and then save your settings to flash memory.

Rule Index	Description
0	For messages, received from Zoom, the Dial Plan is queried according to user part of the From header. Tag value from the matched row will be assigned to DstTags, which will be used for routing.
1	For messages received from Zoom, the value of the X-TO-CARRIER header is assigned to the 'X-TO-CARRIER' session variable, which will be added to the outgoing messages towards the customers' SIP Trunks.
2	For messages received from customers' SIP Trunks, the value 'Zoom' is assigned to DstTags, which will be used for routing towards the Zoom Phone System.
3	For messages, received from customers' SIP Trunks, the Dial Plan is queried according to user part of the Request-URI header. The Tag value from the matched row will be assigned to the 'X-TO-CARRIER' session variable, which will be added as the X-TO-CARRIER header to the outgoing messages towards Zoom.

3.11 Configure Message Manipulation Rules

This section 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).

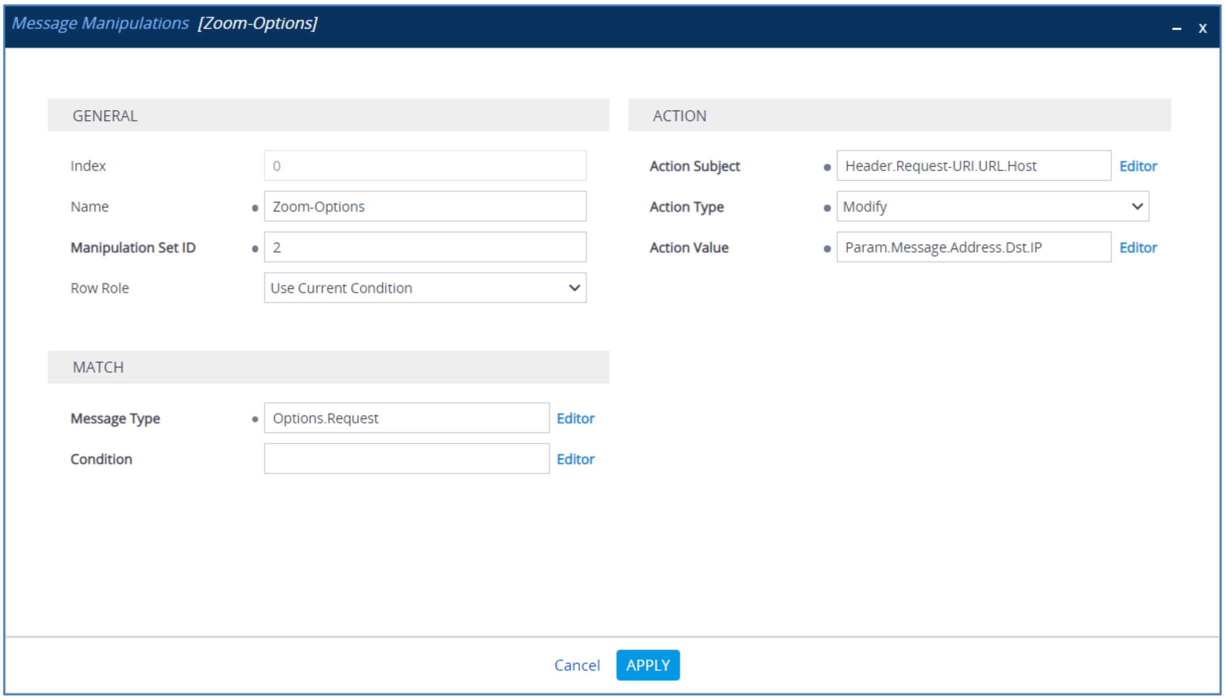
Once you have configured the SIP message manipulation rules, you need to assign them to the relevant IP Group (in the IP Group table) and determine whether they must be applied to inbound or outbound messages.

➢ To configure SIP message manipulation rule for Zoom:

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

Parameter	Value
Index	0
Name	Zoom-OPTIONS (arbitrary name)
Manipulation Set ID	2
Message Type	Options.Request
Action Subject	Header.Request-URI.URL.Host
Action Type	Modify
Action Value	Param.Message.Address.Dst.IP

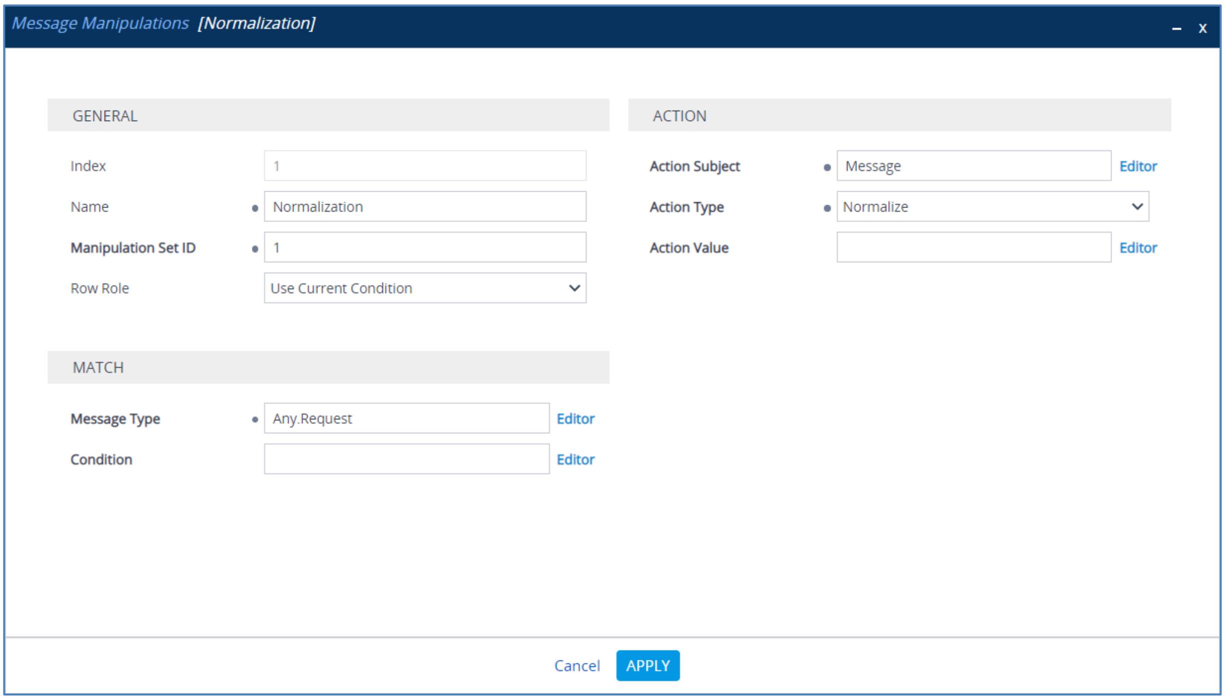
Figure 3-20: Configuring SIP Message Manipulation Rule 0 (for Zoom IP Group)



3. Configure another manipulation rule (Manipulation Set 1) for Zoom IP Group. This rule applies to messages received from the Zoom IP Group. This rule performs normalization of the messages received from Zoom Phone System.

Parameter	Value
Index	1
Name	Normalization
Manipulation Set ID	1
Message Type	Any.Request
Action Subject	Message
Action Type	Normalize

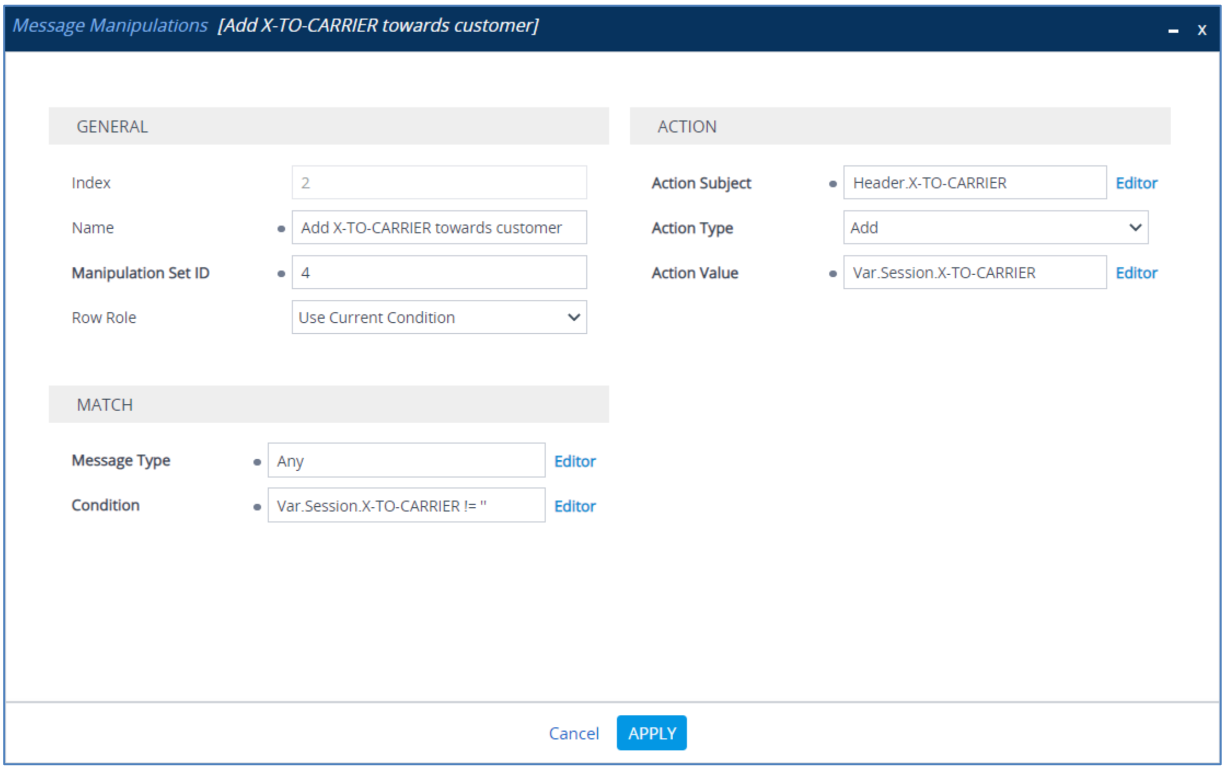
Figure 3-21: Configuring SIP Message Manipulation Rule 1 (for Zoom IP Group)



4. Configure another manipulation rule (Manipulation Set 4) for Customers SIP Trunks IP Groups (if required). This rule applies to messages sent to the Customers SIP Trunks IP Groups. This rule adds X-TO-CARRIER SIP Header with the value from the messages received from the Zoom Phone System.

Parameter	Value
Index	2
Name	Add X-TO-CARRIER towards customer
Manipulation Set ID	4
Message Type	Any
Condition	Var.Session.X-TO-CARRIER != ''
Action Subject	Header.X-TO-CARRIER
Action Type	Add
Action Value	Var.Session.X-TO-CARRIER

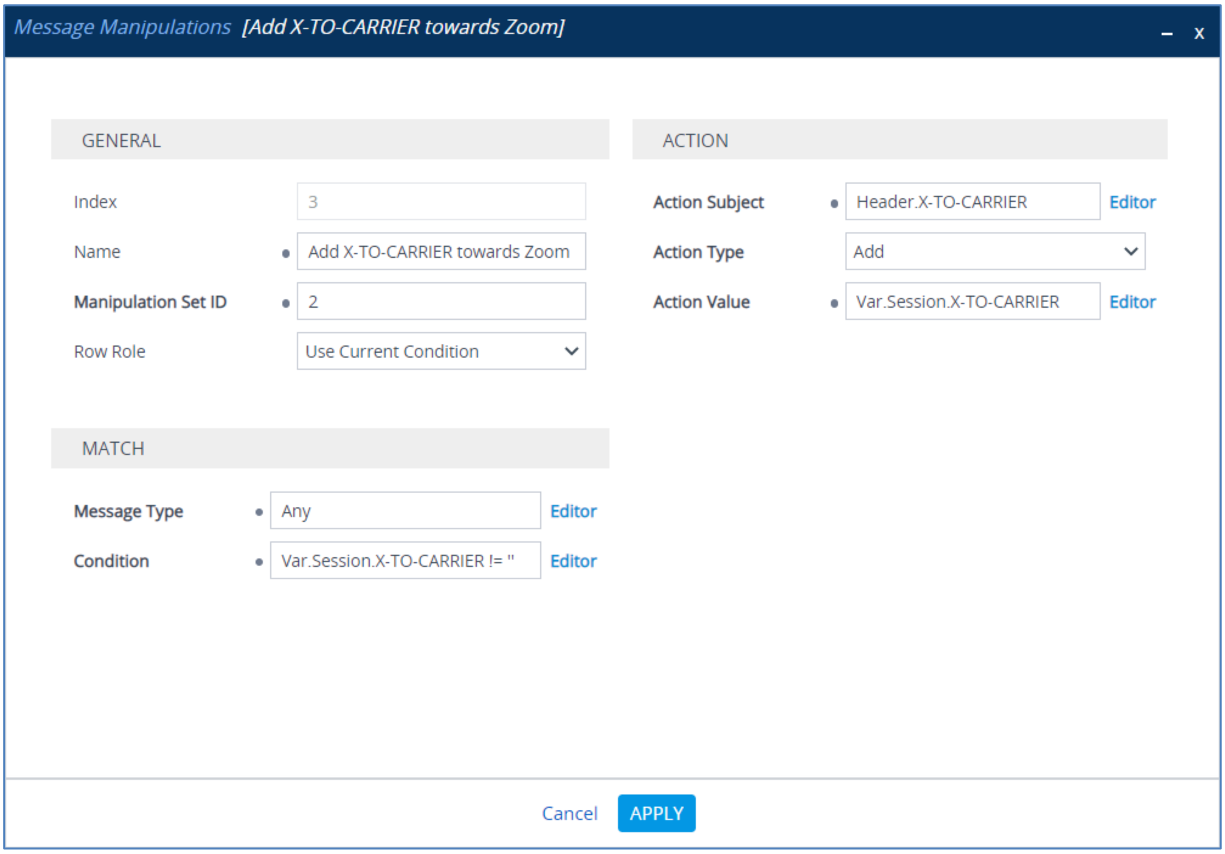
Figure 3-22: Configuring SIP Message Manipulation Rule 2 (for Customers SIP Trunks)



5. Configure another manipulation rule (Manipulation Set 2) for the Zoom IP Group. This rule applies to messages sent to the Zoom IP Group. This rule adds the X-TO-CARRIER SIP Header with the value extracted from the Dial Plan.

Parameter	Value
Index	2
Name	Add X-TO-CARRIER towards Zoom
Manipulation Set ID	2
Message Type	Any
Condition	Var.Session.X-TO-CARRIER != ''
Action Subject	Header.X-TO-CARRIER
Action Type	Add
Action Value	Var.Session.X-TO-CARRIER

Figure 3-23: Configuring SIP Message Manipulation Rule 3 (for Zoom IP Group)



Note: In your implementation connectivity to customers SIP Trunks may require additional message manipulation rules. Refer to the appropriate SIP Trunk Implementation Guide or contact an AudioCodes representative to order Professional Services from AudioCodes, and our Professional Services team will help you with your configuration.

3.12 Configure a Coder Group

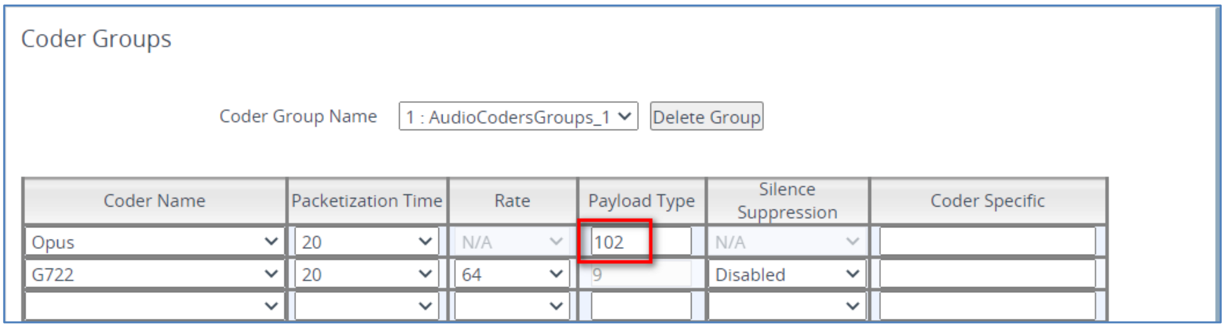
This section describes how to configure coders (termed Coder Groups). As Zoom Phone System supports the OPUS and G.722 coders, while the network connection to the Customers SIP Trunks may restrict operation with a dedicated coders list, you need to add a Coder Group with the supported coders for each leg, the Zoom Phone System and the SIP Trunks.

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

➢ To configure a Coder Group for Zoom:

1. Open the Coder Groups table (**Setup** menu > **Signaling & Media** tab > **Coders & Profiles** folder > **Coder Groups**).
2. From the 'Coder Group Name' dropdown, select **1:Does Not Exist** and add the required codecs as shown in the figure below.

Figure 3-24: Configuring Coder Group for Zoom Phone System



3. Click **Apply** and confirm the configuration change in the prompt that pops up.

Note: Repeat the same procedure for each Customers SIP Trunk if it's required.

The procedure below describes how to configure Allowed Coders Groups to ensure that voice sent to the Zoom Phone System, uses the dedicated coders list whenever possible. Note that the Allowed Coders Group IDs will be assigned to the IP Profiles belonging to the Zoom Phone System, in the next step.

➢ To set a preferred coder for the Zoom Phone System:

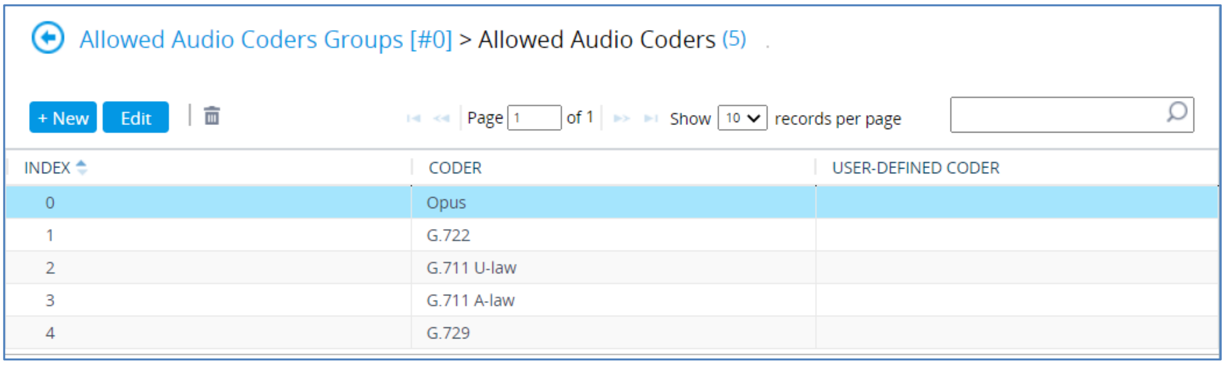
1. Open the Allowed Audio Coders Groups table (**Setup** menu > **Signaling & Media** tab > **Coders & Profiles** folder > **Allowed Audio Coders Groups**).
2. Click **New** and configure a name for the Allowed Audio Coders Group for the Zoom Phone System.

Figure 3-25: Configuring Allowed Coders Group for Zoom Phone System

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:

Index	Coder
0	Opus
1	G.722
2	G.711 U-law
3	G.711 A-law
4	G.729

Figure 3-26: Configuring Allowed Coders for Zoom Phone System



Note: Repeat the same procedure for each Customers SIP Trunk if it's required.

3.13 Configure an IP Profiles

This section describes how to configure IP Profiles. An IP Profile is a set of parameters with user-defined settings related to signaling (e.g., SIP message terminations such as REFER) and media (e.g., coder type). An IP Profile need be assigned to specific IP Group.

➢ **To configure an IP Profile:**

1. Open the Proxy Sets table (**Setup** menu > **Signaling & Media** tab > **Coders & Profiles** folder > **IP Profiles**).
2. Click **+New** to add the IP Profile for Zoom Phone System interface. Configure the parameters using the table below as reference.

Table 3-10: Configuration Example: Zoom IP Profile

Parameter	Value
General	
Name	Zoom (arbitrary descriptive name)
Media Security	
SBC Media Security Mode	Secured
SBC Media	
Extension Coders Group	AudioCodersGroups_1
Allowed Audio Coders	Zoom Allowed Coders
Allowed Coders Mode	Restriction and Preference (reorder coders according to Allowed Coders including extension coders)
RFC 2833 Mode	Extend
SBC Signaling	
Session Expires Mode	Supported
All other parameters can be left unchanged with their default values.	

3. Click **Apply**, and then save your settings to flash memory.

4. Click **+New** to add the IP Profile for the Customer SIP Trunk. Configure the parameters using the table below as reference.

Table 3-11: Configuration Example: Customer 1 SIP Trunk IP Profile

Parameter	Value
General	
Name	Customer 1 (arbitrary descriptive name)
Media Security	
SBC Media Security Mode	Not Secured
SBC Signaling	
P-Asserted-Identity Header Mode	Add (required for anonymous calls)
SBC Forward and Transfer	
Remote REFER Mode	Handle Locally
Remote Replaces Mode	Handle Locally
Remote 3xx Mode	Handle Locally
All other parameters can be left unchanged with their default values.	

5. Click **Apply** and then save your settings to flash memory.

Note: Repeat the same procedure for each Customers SIP Trunk according to SIP Trunk requirements.

3.14 Configure IP Groups

This section 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 SIP Trunk) 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.

➢ **To configure an IP Groups:**

1. Open the IP Groups table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **IP Groups**).
2. Configure IP Group for the Zoom Phone System:

Parameter	Value
Name	Zoom
Topology Location	Up
Type	Server
Proxy Set	Zoom
IP Profile	Zoom
Media Realm	Zoom
Call Setup Rules Set ID	0
Tags	Zoom
Inbound Message Manipulation Set ID	1
Outbound Message Manipulation Set ID	2
Proxy Keep-Alive using IP Group settings	Enable
All other parameters can be left unchanged with their default values.	

3. Configure IP Groups for the Customer's SIP Trunks (for each customer create dedicated IP Group):

Parameter	Value
Name	Customer1 (arbitrary descriptive name)
Type	Server
Proxy Set	Customer1
IP Profile	Customer1
Media Realm	Customers
Call Setup Rules Set ID	1
Tags	<tag per each customer> (as configured in the Dial Plan, refer to Section 3.9 on page 29)
Outbound Message Manipulation Set ID	4 (if required and as configured in Section 3.11 on page 31)
All other parameters can be left unchanged with their default values.	

The configured IP Groups are shown in the figure below:

Figure 3-27: Configuration Example IP Groups in the IP Group Table

IP Groups (4)

+ 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 MANIPULA SET	OUTBOUND MESSAGE MANIPULAT SET
0	Zoom	DefaultS	Server	Not Configu	Zoom	Zoom	Zoom		Enable	1	2
1	Customer1	DefaultS	Server	Not Configu	Customer1	Customer1	Customers		Enable	1	4
2	Customer2	DefaultS	Server	Not Configu	Customer2	Customer2	Customers		Enable	-1	-1
3	Customer3	DefaultS	Server	Not Configu	Customer3	Customer3	Customers		Enable	-1	-1

Note: On Hybrid SBCs (with onboard PSTN interfaces), it's recommended to leave IP Group 0 unconfigured for possible future use for PSTN Fallback.

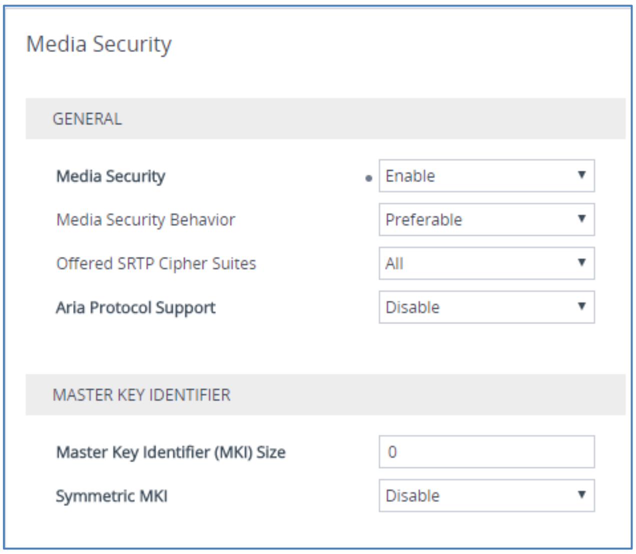
3.15 Configure SRTP

This section describes how to configure media security. The Zoom Phone System Interface needs to use of SRTP only, so you need to configure the SBC to operate in the same manner. By default, SRTP is disabled.

➢ **To enable SRTP:**

1. Open the Media Security page (**Setup** menu > **Signaling & Media** tab > Media folder > Media Security).
2. From the 'Media Security' drop-down list, select Enable to enable SRTP.

Figure 3-28: Configuring Media Security Parameter



3. Click **Apply**.

3.16 Configure IP-to-IP Call Routing Rules

This section 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 example shown below only covers IP-to-IP routing, though you can route the calls from SIP Trunks to Zoom and vice versa. See AudioCodes' SBC documentation for more information on how to route in other scenarios.

The following IP-to-IP Routing rules are defined:

- Terminate SIP OPTIONS messages on the SBC that are received from any entity
- Destination Tag based Routing (from/to Zoom Phone System or Customers SIP Trunks)

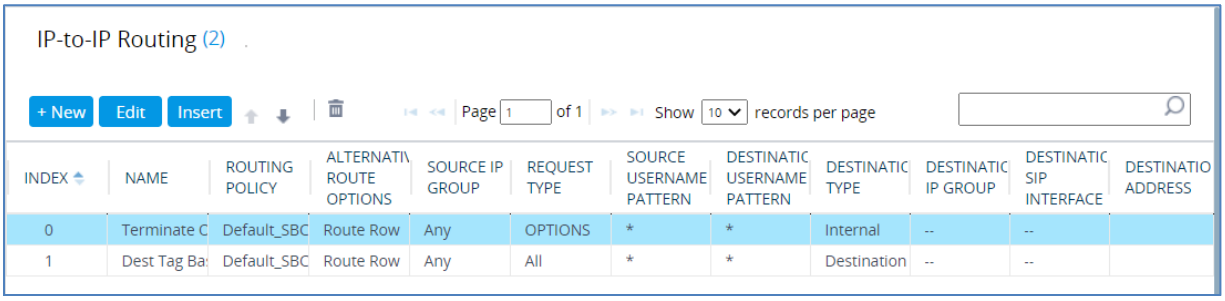
➢ To configure IP-to-IP routing rules:

1. Open the IP-to-IP Routing table (**Setup** menu > **Signaling & Media** tab > **SBC** folder > **Routing** > **IP-to-IP Routing**).
2. Configure routing rules as shown in the table below:

Index	Name	Source IP Group	Request Type	Destination Type	Internal Action
0	Terminate OPTIONS	Any	OPTIONS	Internal	Reply(Response='200')
1	Dest Tag Based Routing (arbitrary name)	Any		Destination Tag	-

The configured routing rules are shown in the figure below:

Figure 3-29: Configured IP-to-IP Routing Rules in IP-to-IP Routing Table



IP-to-IP Routing (2)

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

INDEX	NAME	ROUTING POLICY	ALTERNATIVE ROUTE OPTIONS	SOURCE IP GROUP	REQUEST TYPE	SOURCE USERNAME PATTERN	DESTINATIC USERNAME PATTERN	DESTINATIC TYPE	DESTINATIC IP GROUP	DESTINATIC SIP INTERFACE	DESTINATIO ADDRESS
0	Terminate C	Default_SBC	Route Row	Any	OPTIONS	*	*	Internal	--	--	
1	Dest Tag Ba:	Default_SBC	Route Row	Any	All	*	*	Destination	--	--	

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

3.17 Configure Firewall Settings (Optional)

As extra security, there is an option to configure traffic filtering rules (access list) for incoming traffic on AudioCodes SBC. For each packet received on the configured network interface, the SBC searches the table from top to bottom until the first matching rule is found. The matched rule can permit (allow) or deny (block) the packet. Once a rule in the table is located, subsequent rules further down the table are ignored. If the end of the table is reached without a match, the packet is accepted. Please note that the firewall is stateless. The blocking rules will apply to all incoming packets, including UDP or TCP responses.

➢ **To configure a firewall rule:**

1. Open the Firewall table (**Setup** menu > **IP Network** tab > **Security** folder> **Firewall**).
2. Configure the following Access list rules for WAN IP Interface, based on the list of Zoom Phone System Servers:

Table 3-12: Firewall Table Rules

Index	Source IP	Subnet Prefix	Start Port	End Port	Protocol	Use Specific Interface	Interface ID	Allow Type
0	<Public DNS Server IP> (e.g. 8.8.8.8)	32	0	65535	Any	Enable	WAN_IF	Allow
1	162.12.233.59	32	0	65535	TCP	Enable	WAN_IF	Allow
2	162.12.232.59	32	0	65535	TCP	Enable	WAN_IF	Allow
3	162.12.235.85	32	0	65535	TCP	Enable	WAN_IF	Allow
4	213.19.144.198	32	0	65535	TCP	Enable	WAN_IF	Allow
5	213.244.140.198	32	0	65535	TCP	Enable	WAN_IF	Allow
6	103.122.166.248	32	0	65535	TCP	Enable	WAN_IF	Allow
7	103.122.167.248	32	0	65535	TCP	Enable	WAN_IF	Allow
8	209.9.211.198	32	0	65535	TCP	Enable	WAN_IF	Allow
9	207.226.132.198	32	0	65535	TCP	Enable	WAN_IF	Allow
10	123.123.123.123	32	0	65535	TCP	Enable	WAN_IF	Allow
49	0.0.0.0	0	0	65535	Any	Enable	WAN_IF	Block

Note: Be aware, that if in your configuration, connectivity to SIP Trunk (or other entities) is performed through the same IP Interface as Zoom (WAN_IF in our example), you must add rules to allow traffic from these entities. See an example in the row of index 10.

3.18 Miscellaneous Configuration

This section describes miscellaneous SBC configuration.

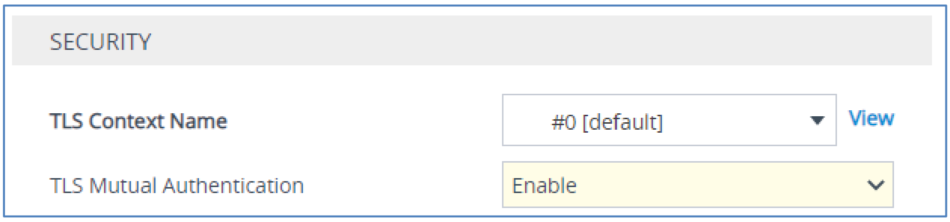
3.18.1 Configuring Mutual TLS Authentication for SIP

This section describes how to configure SBC to work in mutual (two-way) TLS authentication mode.

Note: This section is required only if implementation of MTLS connection with the Zoom Phone System is required and depends on enabling MTLS on the Zoom side.

➢ **To configure mutual TLS authentication for SIP messaging:**

1. Enable two-way authentication on the Zoom SIP Interface: In the SIP Interface table, configure the 'TLS Mutual Authentication' parameter to Enable:

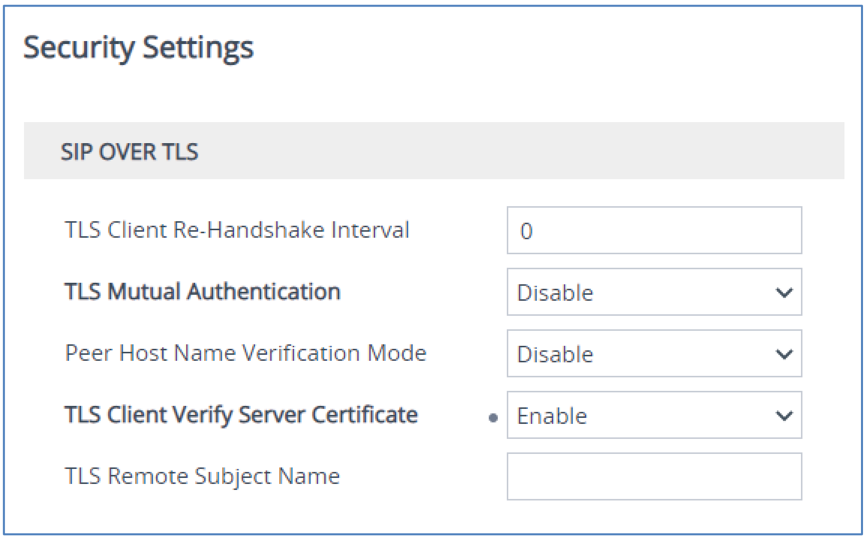


2. Make sure that the TLS certificate is signed by a CA.
3. Make sure that CA certificates are imported into the Trusted Root Certificates table.

In order to further enhance security, it is possible to configure the SBC to verify the server certificates, when it acts as a client for the TLS connection.

➢ **To configure SBC to verify Server certificate:**

1. Open the SBC Security Settings page (**Setup** menu > **IP Network** tab > **Security** folder > **Security Settings**).
2. From the 'TLS Client Verify Server Certificate' drop-down list, select **Enable**:



3. Click **Apply**.

3.18.2 Optimizing CPU Cores Usage for a Specific Service (relevant for Mediant 9000 and Software SBC only)

This section describes how to optimize the SBC's CPU cores usage for a specified profile to achieve maximum capacity for that profile. The supported profiles include:

- SIP profile – improves SIP signaling performance, for example, SIP calls per second (CPS)
- SRTP profile – improves maximum number of SRTP sessions
- Transcoding profile – enables all DSP-required features, for example, transcoding and voice in-band detectors

➢ **To optimize core allocation for a profile:**

1. Open the SBC General Settings page (**Setup** menu > **Signaling & Media** tab > **SBC** folder > **SBC General Settings**).
2. From the 'SBC Performance Profile' drop-down list, select the required profile:

3. Click **Apply**, and then reset the device with a burn-to-flash for your settings to take effect.

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4 Verify the Pairing between the SBC and Zoom Phone System

After you've paired the SBC with Zoom Phone System, validate that the SBC can successfully exchange OPTIONs with Zoom.

➢ **To validate the pairing using SIP OPTIONS:**

1. Open the Proxy Set Status page (**Monitor** > **VOIP Status** > **Proxy Set Status**).
2. Find the Zoom SIP connection and verify that 'Status' is online. If you see a failure, you need to troubleshoot the connection first.

Figure 4-1: Proxy Set Status

Proxy Sets Status

This page refreshes every 60 seconds

PROXY SET ID	NAME	MODE	KEEP ALIVE	ADDRESS	PRIORITY	WEIGHT	SUCCESS COUNT	FAILURE COUNT	STATUS
0	Zoom	Homing	Enabled						ONLINE
				us01peer01.am.zoom.us(213.19.144.198:5061) (*)	-	-	8316	1	ONLINE
				us01peer01.fr.zoom.us(213.244.140.198:5061)	-	-	8314	0	ONLINE
1	Customer1	Parking	Enabled						ONLINE
				sip.telnyx.com(192.76.120.10)(*)	-	-	1	0	ONLINE
2	Customer2	Parking	Enabled						ONLINE
				voipsipbiz.bezeqint.net(212.179.84.133)(*)	-	-	1	0	ONLINE
3	Customer3	Parking	Enabled						ONLINE
				195.189.192.156(*)	-	-	1	0	ONLINE

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A Zoom Data Centers

Connectivity to the Zoom Phone System signaling via Fully Qualified Domain Names (FQDN) depends on the geographical location of the customer SBC(s) and the corresponding Zoom Data Center that the customer would like to send and receive traffic. Zoom Phone System options are currently available in four separate regions across the globe: North America, Europe, APAC and Australia.

Table A-1: Regional instances resolve to the following IP addresses

Region	Traffic Type	Protocol	Ports	A Record	IP Address
North America	Signaling	TCP/TLS	5061	us01peer01.sc.zoom.us	162.12.233.59
	Signaling	TCP/TLS	5061	us01peer01.ny.zoom.us	162.12.232.59
	Signaling	TCP/TLS	5061	us01peer01.dv.zoom.us	162.12.235.85
EMEA	Signaling	TCP/TLS	5061	us01peer01.am.zoom.us	213.19.144.198
	Signaling	TCP/TLS	5061	us01peer01.fr.zoom.us	213.244.140.198
Australia	Signaling	TCP/TLS	5061	us01peer01.sy.zoom.us	103.122.166.248
	Signaling	TCP/TLS	5061	us01peer01.me.zoom.us	103.122.167.248
APAC	Signaling	TCP/TLS	5061	us01peer01.hk.zoom.us	209.9.211.198
	Signaling	TCP/TLS	5061	us01peer01.ty.zoom.us	207.226.132.198

Table A-2: Regional Media Traffic and Ports

Region	Traffic Type	Protocol	Ports	Destination
North America	Media	UDP/SRTP	20000-64000	162.12.232.0/22
EMEA	Media	UDP/SRTP	20000-64000	213.19.144.0/24
	Media	UDP/SRTP	20000-64000	213.244.140.0/24
Australia	Media	UDP/SRTP	20000-64000	103.122.166.0/23
APAC	Media	UDP/SRTP	20000-64000	209.9.211.0/24
	Media	UDP/SRTP	20000-64000	207.226.132.0/24

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