Configuration Note

AudioCodes Professional Services – Interoperability Lab

Connecting Zoom Phone Local Survivability with AudioCodes SBC

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Abbreviations and Terminology

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

Related Documentation

Table 1: Table Caption

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
SIP Message Manipulation Reference Guide
AudioCodes Configuration Notes

Document Revision Record

LTRT	Description
29370	Initial document release.
29374	TLS Private Key size of 1024 was removed. Optional PSTN breakout configuration was added.
29379	Updates related to new Zoom trusted public certificates.

Documentation Feedback

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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 the Zoom Phone Local Survivability (hereafter, referred to as *ZPLS*) Module, Generic SIP Trunk and the Zoom Phone Cloud 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 to better understand the various configuration options. For more information on AudioCodes' SBC Wizard including the download option, visit AudioCodes Web site at https://www.audiocodes.com/partners/sbc-interoperability-list.

1.1 About the Zoom Phone Local Survivability

Zoom Phone is a cloud-based service that is dependent on IP connectivity to Zoom's datacenters. Customers that are using the Zoom Phone solution at corporate locations are encouraged to deploy redundant and reliable Internet connectivity with sufficient bandwidth at each corporate office as a base requirement.

For some business locations, maintaining telephony service in the event of an outage is critical. Zoom can offer a survivability solution of basic telephony services to provide an additional layer of protection to ensure business continuity. An outage can be the result of an Internet service failure at a business location, or a failure in multiple Zoom datacenters that prevent client devices from reaching Zoom Phone components.

The Zoom Phone Local Survivability (ZPLS) module leverages the platform and Operating System (OS) provided by the Zoom Module and is distributed as a Linux-based appliance that is spun up on an on-premises VMware ESXi host. The ZPLS module does not affect the phone service during normal operations. Phone clients and devices in survivable Phone Sites register to the corresponding ZPLS module and are able to maintain a subset of Phone features when connectivity to Zoom Phone is lost. When connectivity to the Zoom Phone cloud returns, clients and devices re-register back to the cloud. During the outage, neither the administrator nor the end user is required to take any action to enable survivability. The failover and fallback process is seamless and automatic.

For more details about ZPLS please refer to Zoom Help Center at: https://support.zoom.us/hc/en-us/articles/8427359971853-Zoom-Phone-Local-Survivability.

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

2 Environment Information

This section describes interoperability environment.

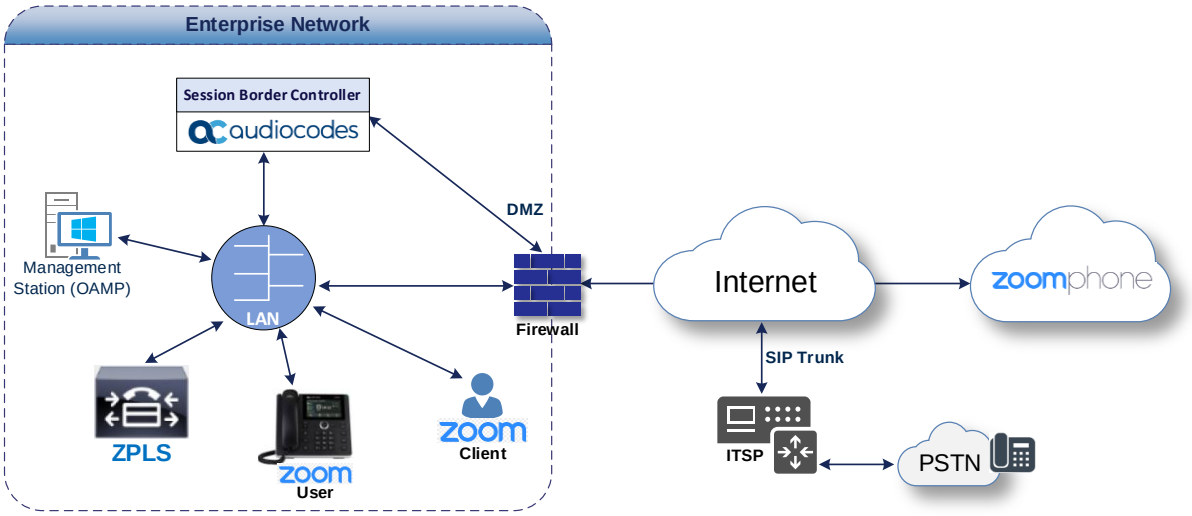
2.1 Interoperability Topology

The interoperability between AudioCodes SBC and Zoom Phone Local Survivability with the Generic SIP Trunk and Zoom Phone system done using the following topology setup:

- Enterprise deployed with Zoom Phone Local Survivability Module (ZPLS) and the administrator's management station, located on the LAN.
- Enterprise connected to the Zoom Phone System located on the WAN for enhanced communication within the Enterprise.
- Enterprise wishes to offer its employees enterprise-voice capabilities and to connect the Enterprise to the PSTN network in case of Zoom Phone System connectivity outage using AudioCodes's SIP Trunking service.
- AudioCodes SBC is implemented to interconnect between the Zoom Phone Local Survivability Module, SIP Trunk and the Zoom Phone system.
  - **Session:** Real-time voice session using the IP-based Session Initiation Protocol (SIP).
  - **Border:** IP-to-IP network border. The Zoom Phone Local Survivability Module is located in the Enterprise LAN. The AudioCodes's SIP Trunk and the Zoom Phone system are located in the public network.

The figure below illustrates this interoperability topology:

Figure 1: Layout of an Interoperability Test Environment



2.1.1 Environment Setup

The interoperability topology includes the following environment setup:

Table 2: Environment Setup

Area	Setup
Network	■ Zoom Phone Local Survivability Module is located on the LAN. ■ Both, Zoom Phone system and Generic SIP Trunk environments are located on the WAN.
Signaling Transcoding	■ Both Zoom Phone systems (ZPLS and Cloud) operates with SIP-over-TLS transport type. ■ Generic SIP Trunk can operate with SIP-over-UDP or SIP-over-TCP or SIP-over-TLS transport type (depends on the particular provider).
Codecs Transcoding	■ Both Zoom Phone systems support OPUS, G.711A-law, G.711U-law and G.729 coders. ■ Generic SIP Trunk supports G.711A-law, G.711U-law, and G.729 coders (or other coders, depending on the requirement).
Media Transcoding	■ Both Zoom Phone systems operate with SRTP media type. ■ Generic SIP Trunk operates with RTP media type.

3 Configuring Zoom Phone Local Survivability Module

The current document describe configuration of the AudioCodes SBC interconnected to the Zoom Phone Local Survivability (ZPLS) Module, installed on any VMWare platform. For configuration of the Zoom Phone Local Survivability Module, refer to the appropriated Section in the Mediant 800C SBC with Zoom Phone Local Survivability Deployment Guide or to the Zoom Help Center at https://support.zoom.us/hc/en-us/articles/360001297663-Getting-started-with-Zoom-Phone-admin-.

Before you begin configuration:

- Ensure your Zoom account has Zoom Node Monthly/Annual and Zoom Phone Hybrid subscriptions and SIP groups are enabled on it. Multiple Sites is enabled for Zoom Phone. Contact your Zoom account team for assistance.
- Make sure you have Zoom Portal owner or admin credentials with appropriated privileges (e.g., to manage Zoom Node).

4 Configuring AudioCodes SBC

This section shows how to configure AudioCodes SBC for interworking between the Zoom Phone Local Survivability, the Zoom Phone system and the Generic SIP Trunk. These configuration procedures are based on the interoperability topology described in Section 2.1, and includes the following main areas:

- **SBC LAN interface:** Zoom Phone Local Survivability Module and Management Station.
- **SBC WAN interface:** Generic SIP Trunking and the Zoom Phone System environment.

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

4.1 Validating AudioCodes SBC License and Version

Zoom has successfully conducted validation tests with AudioCodes' Mediant SBC Ver. 7.40A.250. The previous certified firmware version is 7.20A.258.

- For interconnection between the Zoom Phone Local Survivability Module, the Zoom Phone system and Generic SIP Trunk based on the configuration described in this section, AudioCodes SBC must be installed with a License Key that includes the following software features:
  - **Number of SBC sessions** [Based on requirements]
  - **DSP Channels** [If media transcoding is needed]
  - **Transcoding sessions** [If media transcoding is needed]
  - **Coders** [Based on requirements]

  For more information about the License Key, contact your AudioCodes sales representative.
- If your SBC is deployed in a virtual environment and transcoding is required, your virtual machine must have a minimum of two vCPUs. For more information, please refer to the appropriate *Installation Manual,* which can be found on AudioCodes website.
- The scope of this 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 website.

4.2 Prerequisites

Before you begin configuration, make sure you have obtained the following for each SBC you wish to pair with Zoom Phone System:

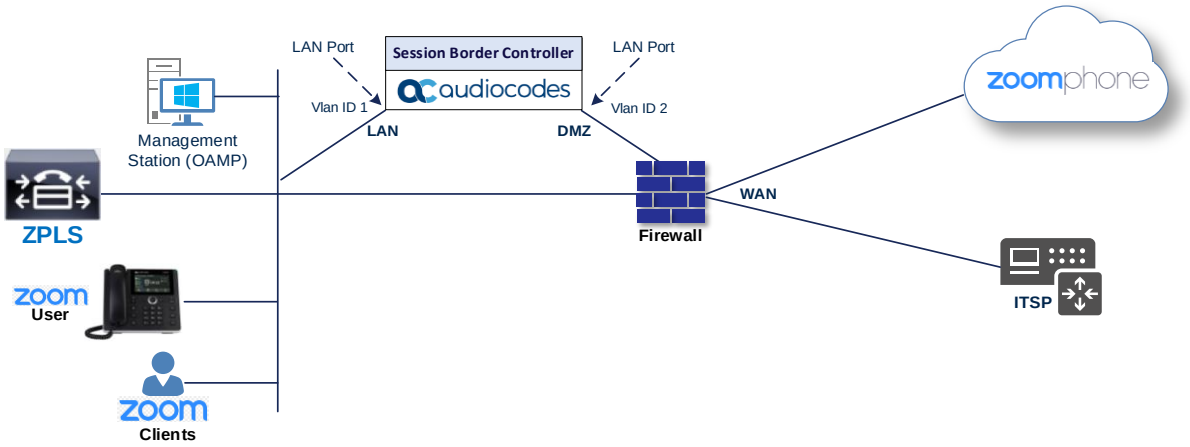
- Public IP address
- Public certificate that is issued by one of the Zoom supported CAs

4.3 Configuring IP Network Interfaces

This section 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:
  - ZPLS Module and Management Servers located on the LAN
  - Zoom Phone system and Generic SIP Trunk, located on the WAN
- SBC connects to the WAN through a DMZ network
- Physical connection: The type of physical connection 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 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 2: Network Interfaces in Interoperability Test Topology



4.3.1 Configuring LAN and WAN VLANs

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

- LAN (assigned the name "LAN_IF")
- WAN (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**).

   There will be one existing row for VLAN ID 1 and underlying interface GROUP_1.
2. Add another VLAN ID 2 for the WAN side.

4.3.2 Configuring 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 the IP network 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: Configuration Example of the Network Interface 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

4.4 Configuring 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.

The procedure involves the following main steps:

- Configure the NTP Server Address
- 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 certificates on the SBC

4.4.1 Configuring the NTP Server Address

This section describes how to configure the NTP server's IP address. It is recommended to implement an NTP server (local NTP server or another global NTP 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**).
3. Click **Apply**.

4.4.2 Creating a TLS Context for Zoom Phone System

The section below describes how to request a certificate for the SBC WAN interface and configure it. The certificate is used by the SBC to authenticate the connection with the Zoom Phone System.

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 4: New TLS Context

Index	Name	TLS Version
1	Zoom (arbitrary descriptive name)	TLSv1.2 and TLSv1.3
All other parameters can be left unchanged with their default values.		

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

4.4.3 Generating a CSR and Obtaining 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. Fill in the rest of the request fields according to your security provider's instructions.
    d. Click the **Create CSR** button; a textual certificate signing request is displayed in the area below the button.
4. Copy the CSR from the line "----BEGIN CERTIFICATE REQUEST" 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.

4.4.4 Deploying the SBC Signed and Trusted by Zoom Root Certificates

After obtaining the SBC signed certificate from the CA, download trusted by Zoom Public Root Certificates and install the following:

- SBC certificate signed by the public CA authority that was authorized by Zoom (refer to Appendix B on page 34)
- Trusted by Zoom Public Root certificates

Currently, Zoom Data Centers (DC) uses DigiCert public CA certificates. Zoom is currently in the process of transitioning root certificate to **DigiCert Global Root G2** and **DigiCert TLS RSA4096 Root G5** certificate, which begins after December 1st, 2023. Therefore, to establish a TLS connection with Zoom Phone infrastructure, download and install as trusted root following public CA certificates:

- https://cacerts.digicert.com/DigiCertGlobalRootG2.crt.pem
- https://cacerts.digicert.com/DigiCertTLSRSA4096RootG5.crt.pem
- https://cacerts.digicert.com/DigiCertGlobalG2TLSRSASHA2562020CA1-1.crt.pem
- https://cacerts.digicert.com/DigiCertG5TLSRSA4096SHA3842021CA1-1.crt.pem

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.
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.
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.
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 trusted by Zoom public CA certificates (obtained from the link at the beginning of this section) to load.
5. Click **OK**; the certificate is loaded to the device and listed in the Trusted Certificates store.

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

4.5 Configuring Media Realms

This section describes how to configure Media Realms. Media Realms allows the dividing of the UDP port ranges for use on different interfaces. In the example below, the following Media Realms are configured:

- One for the IP interface towards the ZPLS Module, with the UDP port starting at 20000 and the number of media session legs is 100 (you need to calculate number of media session legs based on your usage).
- One for the IP interface towards the Zoom Phone System, with the UDP port starting at 10000 and the number of media session legs 100 (you need to calculate number of media session legs based on your usage).
- One for the IP interface towards Generic SIP Trunk, with the UDP port range starting at 6000 and the number of media session legs 100.

To configure Media Realms:

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

Table 5: Configuration Example Media Realms in Media Realm Table

Index	Name	IPv4 Interface Name	Port Range Start	Number of Media Session Legs
0	MR-ZPLS (arbitrary name)	LAN_IF	20000	100 (media sessions assigned with port range)
1	MR-Zoom (arbitrary name)	WAN_IF	10000	100 (media sessions assigned with port range)
2	MR-SIPTrunk (arbitrary name)	WAN_IF	6000	100 (media sessions assigned with port range)
All other parameters can be left unchanged at their default values.				

4.6 Configuring SIP Signaling Interfaces

This section shows how to configure a 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 Generic SIP Trunk 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 SIP Interfaces:

1. Open the SIP Interfaces table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **SIP Interfaces**).
2. Configure SIP Interfaces. You can use the default SIP Interface (Index 0), but 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 6: Configured SIP Interfaces in SIP Interface Table

Index	Name	Network Interface	Application Type	UDP Port	TCP Port	TLS Port	Media Realm
0	SI_ZPLS (arbitrary name)	LAN_IF	SBC	0	0	5061 (according to requirement)	MR-ZPLS
1	SI_Zoom (arbitrary name)	WAN_IF	SBC	0	0	5061 (according to requirement)	MR-Zoom
2	SI_SIPTrunk (arbitrary name)	WAN_IF	SBC	5060 (according to requirement)	0	0	MR-SIP Trunk
All other parameters can be left unchanged at their default values.							

For enhanced security, AudioCodes recommends implementing a Mutual TLS connection with the Zoom Phone System. For required configuration, see section 4.19.1 on page 32.

4.7 Configuring Proxy Sets and Proxy Address

This section 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 topology, Proxy Sets need to be configured for the following IP entities:

- ZPLS Module
- Zoom Phone Cloud system
- Generic SIP Trunk

The Proxy Sets will later be 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. Configure Proxy Sets as shown in the table below:

Table 7: Configuration Example Proxy Sets in Proxy Sets Table

Index	Name	SBC IPv4 SIP Interface	TLS Context Name	Proxy Keep-Alive	Keep-Alive Failure Responses	Redundancy Mode	Proxy Hot Swap
1	**ZPLS** (arbitrary name)	SI_ZPLS	Zoom[1]	Using Options	-	-	-
2	**Zoom DCs** (arbitrary name)	SI_Zoom	Zoom[2]	Using Options	503	Homing	Enable
3	**SIPTrunk** (arbitrary name)	SI_SIPTrunk	Default	Using Options	According to SIP Trunk requirement	According to SIP Trunk requirement	According to SIP Trunk requirement

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

[1] Configured in Section 4.4.
[2] Configured in Section 4.4.

4.7.1 Configuring a Proxy Address

This section shows how to configure a Proxy Address.

To configure a Proxy Address for ZPLS Module:

1. Open the Proxy Sets table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **Proxy Sets**).
2. Click the Proxy Set **ZPLS**, and then click the Proxy Address link located below the table; the Proxy Address table opens.
3. Click **+New,** and then configure the address of the Proxy Set according to the parameters described in the table below:

Table 8: Configuration Proxy Address for ZPLS Module

Index	Proxy Address	Transport Type	Proxy Priority	Proxy Random Weight
0	xxx.xxx.xxx.xxx:5061 (ZPLS Module IP and port)	TLS	0	0

4. Click **Apply**.

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 DCs**, and then click the Proxy Address link located below the table; the Proxy Address table opens.
2. Click **+New,** and then configure the address of the Proxy Set according to the parameters described in the table below:

Table 9: Configuration Proxy Address for Zoom Phone System

Index	Proxy Address	Transport Type	Proxy Priority	Proxy Random Weight
0	213.19.144.198:5061	TLS	0	0
1	213.19.140.198:5061	TLS	0	0

3. Click **Apply**.

The current example is based on configuration Zoom Europe Data Center's IP address. In your implementation, the IP address may be different according to your region. Refer to Appendix A on page 33 for a list of FQDNs / IP addresses of other Zoom Regional Data Centers.

To configure a Proxy Address for SIP Trunk:

1. Open the Proxy Sets table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **Proxy Sets**) and then click the Proxy Set **SIPTrunk**, and then click the **Proxy Address** link located below the table; the Proxy Address table opens.
2. Click **+New,** and then configure the address of the Proxy Set according to the parameters described in the table below:

Table 10: Configuration Proxy Address for SIP Trunk

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

3. Click **Apply**.

4.8 Configuring Coders

This section describes how to configure coders (termed *Coder Group*). As the Zoom Phone systems supports the OPUS and G.729 coders while the network connection to Generic SIP Trunk may restrict operation with other dedicated coders list, you need to add a Coder Group with the supported coders for each leg, the Zoom Phone systems and the Generic SIP Trunk.

> The Coder Group ID for this entity is assigned to its corresponding IP Profile in the next step.

To configure coders for Zoom Phone systems:

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 follows:

Coder Name	Packetization Time	Rate	Payload Type	Silence Suppression
Opus	20	N/A	102	N/A
G.729	20	64	9	Disabled

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

> Repeat the same procedure for each Generic SIP Trunk if it's required.

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

To set a preferred coder for the Generic SIP Trunk:

1. Open the Allowed Audio Coders Groups table (**Setup** menu > **Signaling & Media** tab > **Coders & Profiles** folder > **Allowed Audio Coders Groups**).
2. Click **New,** and then configure a name for the Allowed Audio Coders Group for Generic SIP Trunk (e.g., *SIPTrunk Allowed Coders*).
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	G.729
1	G.711 U-law
2	G.711 A-law

To set a preferred coder for the Zoom Phone systems:

1. Open the Allowed Audio Coders Groups table (**Setup** menu > **Signaling & Media** tab > **Coders & Profiles** folder > **Allowed Audio Coders Groups**).
2. Click **New,** and then configure a name for the Allowed Audio Coders Group for Zoom Phone system (e.g., *Zoom Allowed Coders*).
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.711 U-law
2	G.711 A-law
3	G.729

6. Open the Media Settings page (**Setup** menu > **Signaling & Media** tab > **Media** folder > **Media Settings**).
7. From the '**Extended Coders Behavior**' drop-down list, select **Include Extensions**.
8. Click **Apply**.

4.9 Configuring IP Profiles

This section 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).

To configure IP Profile for the Zoom Phone system:

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

Table 11: Configuration Example: Zoom IP Profile

Parameter	Value
General	
Index	**1**
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**.

To configure an IP Profile for the Generic SIP Trunk:

1. Open the IP Profiles table (**Setup** menu > **Signaling & Media** tab > **Coders & Profiles** folder > **IP Profiles**).
2. Click **+New** add the IP Profile for the Generic SIP Trunk. Configure the parameters using the table below as reference.

Table 12: Configuration Example: Generic SIP Trunk IP Profile

Parameter	Value
General	
Index	**2**
Name	**SIPTrunk**
Media Security	
SBC Media Security Mode	**Not Secured**
SBC Media	
Extension Coders Group	**AudioCodersGroups_2**
Allowed Audio Coders	**SIPTrunk Allowed Coders**
Allowed Coders Mode	**Restriction and Preference** (reorder coders according to Allowed Coders including extension coders)
SBC Signaling	
P-Asserted-Identity Header Mode	**Add** (required for anonymous calls)

3. Click **Apply**.

4.10 Configuring SIP Response Codes for Alternative Routing Reasons

This section describes how to configure the SBC's handling of SIP error responses received from Zoom Phone Cloud system 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. This feature works together with the Proxy Hot Swap feature, which is configured in the Proxy Sets table. Alternative routing based on SIP responses is configured using two tables with 'parent-child' relationships:

- Alternative Reasons Set table ('parent'): Defines the name of the Alternative Reasons Set.
- Alternative Reasons Rules table ('child'): Defines SIP response codes per Alternative Reasons Set.

To apply your configured alternative routing reason rules, you need to assign the Alternative Reasons Set for which you configured the rules, to the Zoom Phone system IP Group in the IP Groups table, using the 'SBC Alternative Routing Reasons Set' parameter.

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 Set**).
2. Click **New** and configure a name for the Alternative Routing Reasons Set (e.g., *Alt. Route Reasons*).
3. Click **Apply**.
4. Select the index row of the Alternative Reasons Set that you added, and then click the Alternative Reasons Rules link located at the bottom of the page; the Alternative Reasons Rules table opens.
5. Click **New** and select **503 Service Unavailable** from the 'Release Cause Code' drop-down list.
6. Click **Apply**.

Additional SIP responses can be added to the table, based on requirements.

4.11 Configuring 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 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 topology, IP Groups must be configured for the following IP entities:

- ZPLS Module
- Zoom Phone system
- Generic SIP Trunk

To configure IP Groups:

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

Parameter	Value
Index	**1**
Name	**ZPLS** (arbitrary descriptive name)
Type	**Server**
Proxy Set	**ZPLS**
IP Profile	**Zoom**
Media Realm	**MR- ZPLS**
SIP Group Name	(According to requirement)
Proxy Keep-Alive using IP Group settings	**Enable**
All other parameters can be left unchanged with their default values.	

3. Configure an IP Group for the Zoom Phone system:

Parameter	Value
Index	**2**
Name	**Zoom DCs** (arbitrary descriptive name)
Type	**Server**
Proxy Set	**Zoom DCs**
IP Profile	**Zoom**
Media Realm	**MR-Zoom**
SIP Group Name	(According to requirement)
SBC Alternative Routing Reason Set	**Alt. Route Reasons** (created in section 4.10 on page 19)
Proxy Keep-Alive using IP Group settings	**Enable**
All other parameters can be left unchanged with their default values.	

4. Configure an IP Group for the SIP Trunk:

Parameter	Value
Index	**3**
Name	**SIPTrunk** (arbitrary descriptive name)
Type	**Server**
Proxy Set	**SIPTrunk**
IP Profile	**SIPTrunk**
Media Realm	**MR-SIPTrunk**
SIP Group Name	(According to ITSP requirement)
All other parameters can be left unchanged with their default values.	

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

4.12 Configuring 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 configure media security:

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.
3. Click **Apply**.

4.13 Configuring 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.

For the interoperability topology, the following IP-to-IP routing rules need to be configured:

- Terminate SIP OPTIONS messages on the SBC that are received from any entity
- Calls from Zoom Phone Cloud system to Generic SIP Trunk
- Calls from Generic SIP Trunk to Zoom Phone Cloud system
- If the Zoom Phone Cloud system is not available, route calls from the Generic SIP Trunk to the ZPLS Module
- Calls from ZPLS Module to Generic SIP Trunk

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:

Table 13: Configuration IP-to-IP Routing Rules

Index	Name	Alternative Route Options	Source IP Group	Request Type	Dest Type	Dest IP Group	Internal Action
0	Terminate OPTIONS		Any	OPTIONS	Internal		Reply(Response='200')
1	Zoom to ITSP (arbitrary name)		Zoom DCs		IP Group	SIPTrunk	
2	ITSP to Zoom (arbitrary name)		SIPTrunk		IP Group	Zoom DCs	
3	ITSP to ZPLS (arbitrary name)	Alternative Route Ignore Inputs	SIPTrunk		IP Group	ZPLS	
4	ZPLS to ITSP (arbitrary name)		ZPLS		IP Group	SIPTrunk	

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

4.14 Configuring Number Manipulation Rules

This section describes how to configure IP-to-IP manipulation rules. These rules manipulate the SIP Request-URI user part (source or destination number). The manipulation rules use the configured IP Groups (as configured in Section 4.11 on page 20) to denote the source and destination of the call.

> Adapt the manipulation table according to your environment dial plan.

For example, for this interoperability topology, a manipulation is configured to add the "+" (plus sign) to the destination number (if it not exists) for calls from the Generic SIP Trunk IP Group to the Zoom DCs IP Group for any destination username pattern.

To configure a number manipulation rule:

1. Open the Outbound Manipulations table (**Setup** menu > **Signaling & Media** tab > **SBC** folder > **Manipulation** > **Outbound Manipulations**).
2. Configure the rules according to your setup.

The table below shows an example of configured IP-to-IP outbound manipulation rules for calls between the Zoom DCs IP Group and Generic SIP Trunk IP Group:

Rule Index	Description
0	Calls from SIP Trunk IP Group to Zoom DCs IP Group with the prefix destination number "+", do nothing.
1	Calls from SIP Trunk IP Group to Zoom DCs IP Group with any destination number between 1 to 9, add "+" to the prefix of the destination number.

4.15 Configuring 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 DCs:

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 DCs IP Group. This rule applies to OPTIONS messages sent to the Zoom DCs IP Group. This replaces the host part of the SIP Request-URI Header with the destination (Zoom Phone DC 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**

3. Configure another manipulation rule (Manipulation Set 1) for Zoom DCs IP Group. This rule applies to messages received from the Zoom DCs 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**

4. Assign Manipulation Set IDs 1 and 2 to the Zoom DCs IP Group:
   a. Open the IP Groups table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **IP Groups**).
   b. Select the row of the Zoom DCs IP Group, and then click **Edit**.
   c. Set the 'Inbound Message Manipulation Set' field to **1**.
   d. Set the 'Outbound Message Manipulation Set' field to **2**.
   e. Click **Apply**.

In your implementation, connectivity to the SIP Trunk 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.

4.16 Configuring Registration Accounts (Optional)

This section describes how to configure SIP registration accounts. This is required so that the SBC can register with the Generic SIP Trunk on behalf of the Zoom Phone systems. The Generic SIP Trunk requires registration and authentication to provide service.

In the interoperability topology, the Served IP Group is the ZPLS Module and Zoom DCs IP Groups and the Serving IP Group is Generic SIP Trunk IP Group.

> Configure Registration Account only if this is required by SIP Trunk.

To configure a registration account:

1. Open the Accounts table (**Setup** menu > **Signaling & Media** tab > **SIP Definitions** folder > **Accounts**).
2. Click **New** and configure the account for the ZPLS Module IP Group according to the provided information, for example:

Parameter	Value
Served IP Group	**ZPLS**
Application Type	**SBC**
Serving IP Group	**SIPTrunk**
Host Name	As provided by the SIP Trunk provider
Register	**Regular**
Contact User	Trunk main line as provided by the SIP Trunk provider
Username	As provided by the SIP Trunk provider
Password	As provided by the SIP Trunk provider

3. Click **Apply**.
4. Click **New** and configure the account for Zoom DCs IP Group according to the provided information, for example:

Parameter	Value
Served IP Group	**Zoom DCs**
Application Type	**SBC**
Serving IP Group	**SIPTrunk**
Host Name	As provided by the SIP Trunk provider
Register	**Regular**
Contact User	Trunk main line as provided by the SIP Trunk provider
Username	As provided by the SIP Trunk provider
Password	As provided by the SIP Trunk provider

5. Click **Apply**.

4.17 Configuring Firewall Settings (Optional)

As an additional security measure, 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 14: 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	64.211.144.247	32	0	65535	TCP	Enable	WAN_IF	Allow
11	<SIP Trunk IP address>	32	0	65535	TCP	Enable	WAN_IF	Allow
49	0.0.0.0	0	0	65535	Any	Enable	WAN_IF	Block

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 11.

4.18 Configuring PSTN Breakout (Optional)

This section describes configuring AudioCodes Mediant 800C SBC for connecting to a PSTN network. This solution can be used for PSTN breakout when connectivity between SBC and SIP Trunk is also dropped, but there is a PSTN connection to the Telephony services.

> As configuration settings of Gateway functionality (especially PSTN interface) may vary widely between customers, this document describes an example configuration. However, if you need assistance in your Gateway configuration and you have a valid support agreement with AudioCodes, please contact AudioCodes Professional Services (who also perform PoC testing, if required).

4.18.1 Configuring TDM Bus Clock Settings

This section describes the configuration of the TDM and clock timing parameters. In a traditional TDM service network such as PSTN, both ends of the TDM connection must be synchronized. If synchronization is not achieved, voice frames are either dropped (to prevent a buffer overflow condition) or inserted (to prevent an underflow condition). In both cases, connection quality and reliability are affected.

AudioCodes Gateway can be configured to recover clock from the PSTN line or to act as clock source to PSTN line (internal clock).

To configure synchronization based on clock from PSTN line:

1. Open the TDM Bus Settings page (**Setup** menu > **Signaling & Media** tab > **Gateway** folder > **TDM Bus Settings**).
2. From the 'TDM Bus Clock Source' drop-down list, select **Network** to recover the clock from the line interface.
3. In the 'TDM Bus Local Reference' field, enter the trunk from which the clock is derived.

> The E1/T1 trunk should recover the clock from the remote side (see below description of the 'Clock Master' parameter).

4. Enable automatic switchover to the next available "slave" trunk if the device detects that the local-reference trunk is no longer capable of supplying the clock to the system:
   a. From the 'TDM Bus PSTN Auto FallBack Clock' drop-down list, select **Enable**.
   b. From the 'TDM Bus PSTN Auto Clock Reverting' drop-down list, select **Enable** to enable the device to switch back to a previous trunk that returns to service if it has higher switchover priority.
5. Configure the PSTN trunk to recover/derive clock from/to the remote side of the PSTN trunk (i.e., clock slave or clock master): In the Trunk Settings page, configure the 'Clock Master' parameter to one of the following:
   - Recovered - to recover clock (i.e., slave)
   - Generated - to transmit clock (i.e., master)

4.18.2 Configuring Trunk Settings

This section describes the configuration of the PSTN Trunk parameters. This includes selecting the PSTN protocol and configuring related parameters.

To configure Trunk settings:

1. Open the Trunk Settings page (**Setup** menu > **Signaling & Media** tab > **Gateway** folder > **Trunks & Groups** > **Trunks**).
2. Select the trunk that you want to configure by clicking the required trunk number icon.
3. To configure a new trunk:
   - Configure the trunk parameters as required.
   - Click the **Apply Trunk Settings** button.

> The trunk parameters should be configured according to the remote side.

4. The most commonly used parameters, which you need to configure are the protocol type (e.g., E1 Euro ISDN), the clock master of the trunk (Recovered/Generated), and the ISDN Termination Side (User/Network side).

4.18.3 Configuring Trunk Groups

This section describes the Trunk Groups configuration. A Trunk Group is a logical group of physical trunks and channels. A Trunk Group can include multiple trunks and a range of channels. To enable and activate the channels, you need to configure the Trunk Group and assign it telephone numbers. Channels that are not configured in this table are disabled. Once you have configured your Trunk Group, you can use it for call routing.

To configure a Trunk Group:

1. Open the Trunk Settings page (**Setup** menu > **Signaling & Media** tab > **Gateway** folder > **Trunks & Groups** > **Trunk Groups**).
2. Configure Trunk Group as shown in the table below:

Table 15: Example of the Trunk Group Configuration

Group Index	Module	From Trunk	To Trunk	Channels	Phone Number	Trunk Group ID	Tel Profile Name
1	Module 1 PRI	1	1	1-31	-	1	None

> This is just example; your configuration can be different.

3. Click **Apply**

4.18.4 Configuring Trunk Group Settings

This section describes the Trunk Group Settings table, which lets you configure various settings per Trunk Group, configured in the previous section. The main configuration includes channel select method, which defines how the device allocates incoming IP-to-Tel calls to the channels of a Trunk Group.

To configure Trunk Group Settings:

1. Open the Trunk Settings page (**Setup** menu > **Signaling & Media** tab > **Gateway** folder > **Trunks & Groups** > **Trunk Group Settings**).
2. Configure Trunk Group Settings as shown in the table below:

Table 16: Example of the Trunk Group Settings

Index	Name	Trunk Group ID	Channel Select Mode
0	PSTN Breakout	1	Channel Cyclic Ascending
All other parameters can be left unchanged with their default values.			

This is just example; your configuration can be different.

3. Click **Apply**.

4.18.5 Configuring IP-to-Tel Routing Rule

This section describes the Gateway IP to PSTN Routing settings. For call routing from the ZPLS to the PSTN trunk, you need to configure an IP-to-Tel routing rule. In other words, you need to route calls from the IP Group of the ZPLS to the Trunk Group that you configured for the PSTN trunk (i.e., ID 1) in the Section 4.18.3.

To configure an IP-to-Tel routing rule:

1. Open the IP-to-Tel Routing page (**Setup** menu > **Signaling & Media** tab > **Gateway** folder > **Routing** > **IP-to-Tel Routing**).
2. Configure IP-to-Tel Routing Rule as shown in the table below:

Table 17: Example of the IP-to-Tel Routing Rule Configuration

Index	Name	Destination Phone Prefix	Destination Type	Trunk Group ID
0	PSTN Breakout (arbitrary name)	*	Trunk Group	1
All other parameters can be left unchanged with their default values.				

The asterisk (*) value of the 'Destination Phone Prefix' parameter denotes all dialed calls.

3. Click **Apply**.

4.18.6 Configuring Tel-to-IP Routing Rule

This section describes the Gateway PSTN to IP Routing settings. To receive calls from the PSTN network, you need to add rules to route calls received from the E1 trunk (e.g., Trunk Group ID 1) to the ZPLS Module.

To configure a Tel-to-IP routing rule:

1. Open the Tel-to-IP Routing page (**Setup** menu > **Signaling & Media** tab > **Gateway** folder > **Routing** > **Tel-to-IP Routing**).
2. Configure Tel-to-IP Routing Rules as shown in the table below:

Table 18: Example of the Tel-to-IP Routing Rules Configuration

Index	Name	Source Trunk Group ID	Destination IP Group
0	PSTN to ZPLS (arbitrary name)	1	ZPLS
All other parameters can be left unchanged with their default values.			

3. Click **Apply**.

4.18.7 Adapting SBC Routing Table with local PSTN Breakout

This section describes how to change SBC routing rules to route calls from the ZPLS to the local PSTN. Following IP-to-IP routing rule needs to be added for this purpose.

To configure IP-to-IP routing rule:

1. Open the IP-to-IP Routing table (**Setup** menu > **Signaling & Media** tab > **SBC** folder > **Routing** > **IP-to-IP Routing**).
2. Add following routing rule at the end of the table:

Table 19: Configuration IP-to-IP Routing Rules with local PSTN Breakout

Index	Name	Alternative Route Options	Source IP Group	Request Type	Dest Type
5	ZPLS to PSTN (arbitrary name)	Alternative Route Ignore Inputs	ZPLS	INVITE	Gateway

4.19 Miscellaneous Configuration

This section describes miscellaneous SBC configuration.

4.19.1 Configuring Mutual TLS Authentication for SIP

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

> 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 with Zoom:

1. Enable two-way authentication on the Zoom SIP Interface:
   a. In the SIP Interface table, assign Zoom TLS context to the Zoom SIP Interface and 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.

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**).
   From the 'TLS Client Verify Server Certificate' drop-down list, select Enable.
2. Click **Apply**.

4.19.2 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, e.g., SIP calls per second (CPS)
- SRTP profile – improves maximum number of SRTP sessions
- Transcoding profile – enables all DSP-required features, e.g., 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 (e.g., *Optimized for transcoding*).
3. Click **Apply**, and then reset the device with a burn-to-flash for your settings to take effect.

> If your SBC is deployed in a virtual environment and transcoding is required, your virtual machine must have a minimum of two vCPUs. For more information, please refer to the appropriate Installation Manual, which can be found on AudioCodes website.

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
South America	Signaling	TCP/TLS	5061	us01peer01.sp.zoom.us	64.211.144.247

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

B Zoom Public Trusted Certificate List

The following table lists the Zoom Public Trusted Certificates.

Table B-1: Zoom Public Trusted Certificate List

Certificate Issuer Organization	Common Name or Certificate Name
Buypass AS-983163327	Buypass Class 2 Root CA
Buypass AS-983163327	Buypass Class 3 Root CA
Baltimore	Baltimore CyberTrust Root
Cybertrust, Inc	Cybertrust Global Root
DigiCert Inc	DigiCert Assured ID Root CA
DigiCert Inc	DigiCert Assured ID Root G2
DigiCert Inc	DigiCert Assured ID Root G3
DigiCert Inc	DigiCert Global Root CA
DigiCert Inc	DigiCert Global Root G2
DigiCert Inc	DigiCert Global Root G3
DigiCert Inc	DigiCert High Assurance EV Root CA
DigiCert Inc	DigiCert Trusted Root G4
GeoTrust Inc.	GeoTrust Global CA
GeoTrust Inc.	GeoTrust Primary Certification Authority
GeoTrust Inc.	GeoTrust Primary Certification Authority - G2
GeoTrust Inc.	GeoTrust Primary Certification Authority - G3
GeoTrust Inc.	GeoTrust Universal CA
GeoTrust Inc.	GeoTrust Universal CA 2
DigiCert Inc	DigiCert Global Root G3
Symantec Corporation	Symantec Class 1 Public Primary Certification Authority - G4
Symantec Corporation	Symantec Class 1 Public Primary Certification Authority - G6
Symantec Corporation	Symantec Class 2 Public Primary Certification Authority - G4
Symantec Corporation	Symantec Class 2 Public Primary Certification Authority - G6
Thawte, Inc.	Thawte Primary Root CA
Thawte, Inc.	Thawte Primary Root CA - G2
Thawte, Inc.	Thawte Primary Root CA - G3
VeriSign, Inc.	VeriSign Class 1 Public Primary Certification Authority - G3
VeriSign, Inc.	VeriSign Class 2 Public Primary Certification Authority - G3
VeriSign, Inc.	VeriSign Class 3 Public Primary Certification Authority - G3
VeriSign, Inc.	VeriSign Class 3 Public Primary Certification Authority - G4
VeriSign, Inc.	VeriSign Class 3 Public Primary Certification Authority - G5
VeriSign, Inc.	VeriSign Universal Root Certification Authority
AffirmTrust	AffirmTrust Commercial

Certificate Issuer Organization	Common Name or Certificate Name
AffirmTrust	AffirmTrust Networking
AffirmTrust	AffirmTrust Premium
AffirmTrust	AffirmTrust Premium ECC
Entrust, Inc.	Entrust Root Certification Authority
Entrust, Inc.	Entrust Root Certification Authority - EC1
Entrust, Inc.	Entrust Root Certification Authority - G2
Entrust, Inc.	Entrust Root Certification Authority - G4
Entrust.net	Entrust.net Certification Authority (2048)
GlobalSign	GlobalSign
GlobalSign	GlobalSign
GlobalSign	GlobalSign
GlobalSign nv-sa	GlobalSign Root CA
The GoDaddy Group, Inc.	Go Daddy Class 2 CA
GoDaddy.com, Inc.	Go Daddy Root Certificate Authority - G2
Starfield Technologies, Inc.	Starfield Class 2 CA
Starfield Technologies, Inc.	Starfield Root Certificate Authority - G2
QuoVadis Limited	QuoVadis Root CA 1 G3
QuoVadis Limited	QuoVadis Root CA 2
QuoVadis Limited	QuoVadis Root CA 2 G3
QuoVadis Limited	QuoVadis Root CA 3
QuoVadis Limited	QuoVadis Root CA 3 G3
QuoVadis Limited	QuoVadis Root Certification Authority
Comodo CA Limited	AAA Certificate Services
AddTrust AB	AddTrust Class 1 CA Root
AddTrust AB	AddTrust External CA Root
COMODO CA Limited	COMODO Certification Authority
COMODO CA Limited	COMODO ECC Certification Authority
COMODO CA Limited	COMODO RSA Certification Authority
The USERTRUST Network	USERTrust ECC Certification Authority
The USERTRUST Network	USERTrust RSA Certification Authority
T-Systems Enterprise Services GmbH	T-TeleSec GlobalRoot Class 2
T-Systems Enterprise Services GmbH	T-TeleSec GlobalRoot Class 3

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