

Ribbon ATA SBC Edge 1K_2K R9.0 Interop with Zoom : Interoperability Guide



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Interoperable Vendors



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Document Overview

This document outlines the configuration best practices for the Ribbon solution covering the Edge series ATA (Analog Telephone Adaptor)/IAD (Integrated Access Device) when deployed with Zoom.

The Ribbon Edge is a low-density analog gateway or ATA/IAD device that gives small businesses, SOHOs and branch offices with analog voice infrastructures an easy, cost-effective way to capitalize on Voice over Internet Protocol (VoIP) services. The Ribbon Edge ATA/IAD offers a survivable branch office solution, providing support for integrating analog endpoints and the Public Switched Telephone Network (PSTN) and support for all Session Initiation Protocol (SIP) calls. The Ribbon Edge supports any-to-any connectivity between analog and SIP devices, enabling branch offices to rapidly migrate analog phones onto SIP-based networks and communicate seamlessly.

The interoperability compliance testing focuses on verifying inbound and outbound call flows between Ribbon Edge ATA & Zoom Cloud.

This guide contains the following configuration sections:

- [Section A: Ribbon SBC Edge Configuration](#)
 - Captures general Edge ATA/IAD configurations for provisioning with Zoom.
- [Section B: Zoom Configuration](#)
 - Captures the Zoom configuration.

Non-Goals

It is not the goal of this guide to provide detailed configurations that meets the requirements of every customer. Use this guide as a starting point and build the SBC configurations as ATA in consultation with network design and deployment engineers.

Audience

This is a technical document intended for telecommunications engineers with the purpose of configuring the Ribbon SBC.

To perform this interop, you need:

- to use graphical user interface (GUI) or command line interface (CLI) of the Ribbon product.
- to understand the basic concepts of TCP/UDP/TLS and IP/Routing.
- to have SIP/RTP/SRTP to complete the configuration and for troubleshooting.

**Note**

This configuration guide is offered as a convenience to Ribbon customers. The specifications and information regarding the product in this guide are subject to change without notice. All statements, information, and recommendations in this guide are believed to be accurate but are presented without warranty of any kind, express or implied, and are provided "AS IS". Users must take full responsibility for the application of the specifications and information in this guide.

Prerequisites

The following aspects are required before proceeding with the interop:

- Ribbon Edge series
- Public IP Addresses
- Zoom Go account - a special type of Zoom account that has option to provision the ATA/IAD device.
- TLS Certificates for Ribbon Edge series
 - Refer to [TLS Configuration between Ribbon Edge and Zoom](#)

Product and Device Details

The following equipment and software were used for the sample configuration provided:

Table 1: Requirements

	Equipment	Software Version
Ribbon Communications	Ribbon Edge SBC 1K	V9.0.4b595
Zoom	Zoom Desktop app	5.7.7 (1105)
	Zoom Mobile app	5.8.1 (2403)
Third-party Phones	Beetel Analog Phone	NA

**Note**

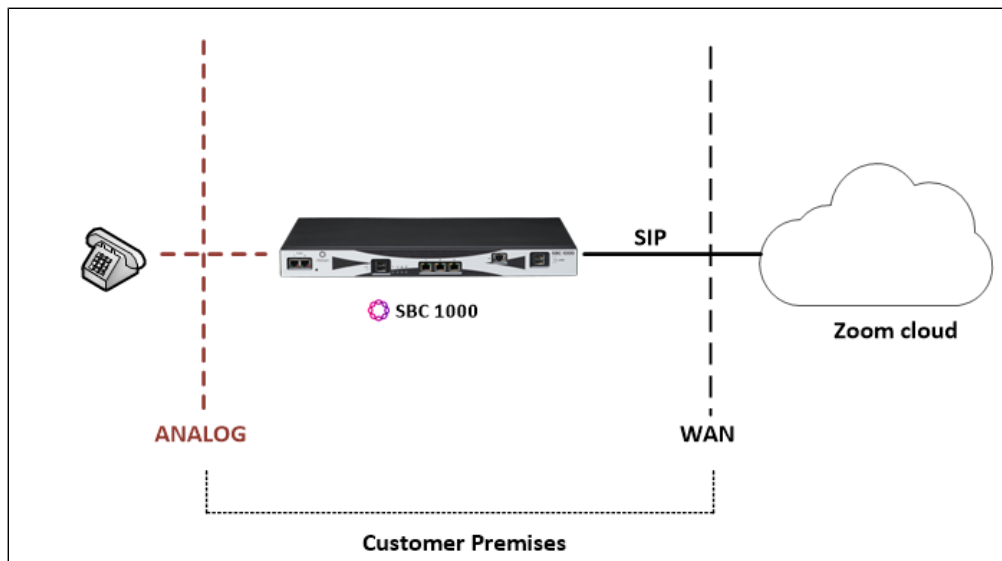
- The Ribbon Edge portfolio includes the SBC 1000 and SBC 2000. As a result, this configuration guide is valid for both devices.
- Zoom Desktop app version is 5.7.7 (1105) or later.
- Zoom Mobile app version is 5.8.1 (2403) or later.

Network Topology Diagram

This section covers the Ribbon Edge deployment topology and the Interoperability Test Lab Topology.

Ribbon Edge Deployment Topology

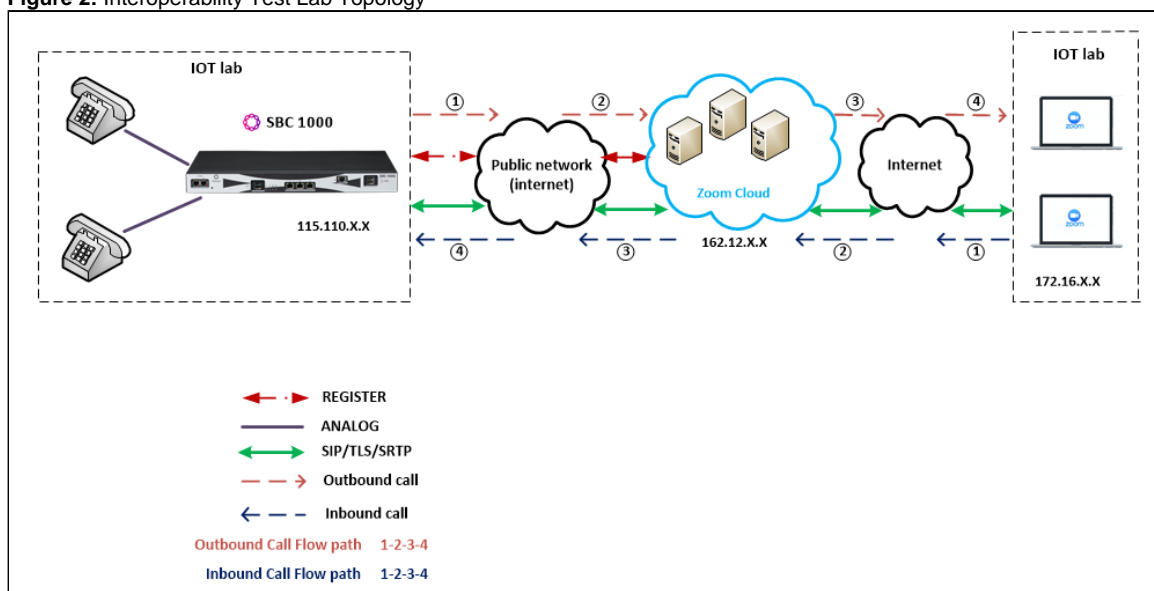
Figure 1: Ribbon Edge Deployment Topology



Interoperability Test Lab Topology

The following lab topology diagram shows connectivity between Zoom and Ribbon Edge ATA/IAD.

Figure 2: Interoperability Test Lab Topology



Section A: Ribbon SBC Edge Configuration

The following SBC Edge configurations are included in this section:

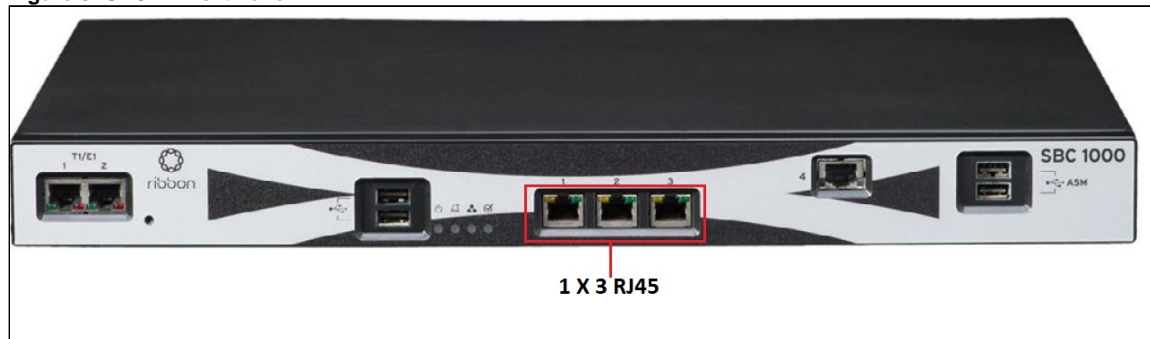
1. [Connectivity](#)
2. [Network](#)
3. [Static Routes](#)
4. [TLS Configuration between Ribbon SBC Edge and Zoom](#)
5. [Media Profile](#)
6. [SRTP Profile](#)
7. [SIP Profile](#)
8. [Analog Leg Configuration](#)
9. [Zoom Leg Configuration](#)

- The SBC Edge can connect to the network as mentioned in [Connectivity](#) and [Network](#).
- Zoom prefers transport as TLS. Establishing a TLS connection between the SBC Edge and Zoom is covered under [TLS Configuration between Ribbon SBC Edge and Zoom](#).
- Generic configurations related to the SBC Edge are covered under [Media Profile](#), [SRTP Profile](#) and [SIP Profile](#).

- The SBC Edge specific configuration related to PSTN is covered under [Analog Leg Configuration](#).
- The SBC Edge specific configuration related to Zoom is covered under [Zoom Leg Configuration](#).

1. Connectivity

Figure 3: SBC1K Front Panel



Note

The SBC 1000/2000 is connected to the network as follows:

Ethernet 1: RJ45 "1" is connected towards the PSTN leg.

Ethernet 2: RJ45 "2" is connected towards the Zoom leg.

2. Network

Configure Ethernet 1 and Ethernet 2 of SBC 1000/2000 with the IP as follows:

Navigate to **Node Interfaces > Logical Interfaces**.

Figure 4: Logical Interfaces

Logical Interfaces							
Total 7 LogicalInterface Rows							
Interface Name	IPv4 Address	IPv6 Address	Description	Admin State	Display	Primary Key	
Ethernet 1 IP	10.54.1.1		Private Interface	Enabled	Counters	14040	
Ethernet 2 IP	115.110.1.1		Public Interface	Enabled	Counters	14041	
Loopback 1				Disabled		30	
Loopback 2				Disabled		31	
Loopback 3				Disabled		32	

Figure 5: Ethernet 1

Logical Interfaces July 16, 2020 19:39:59

Total 7 LogicalInterface Rows Filter...

Interface Name	IPv4 Address	IPv6 Address	Description	Admin State	Display	Primary Key
Ethernet 1 IP	10.54. . .		Private Interface	Enabled	Counters	14040

Identification/Status

Interface Name Ethernet 1 IP

I/F Index 2

Alias

Description

Admin State Enabled

Networking

MAC Address 00:10:23:e0:01:0e

IP Addressing Mode IPv4

IPv4 Information

ACL In None

☐ ACL Out None

ACL Forward None

IP Assign Method Static

Primary Address 10.54. . . X.X.X.X

Primary Netmask 255.255.255.0 X.X.X.X

Configure Secondary Interface Disabled

Figure 6: Ethernet 2

Interface Name	IPv4 Address	IPv6 Address	Description	Admin State	Display	Primary Key
Ethernet 2 IP	115.110. . .		Public Interface	Enabled	Counters	14041

Identification/Status

Interface Name Ethernet 2 IP

I/F Index 3

Alias

Description

Admin State Enabled

Networking

MAC Address 00:10:23:e0:01:0e

IP Addressing Mode IPv4

IPv4 Information

ACL In None

ACL Out None

ACL Forward None

IP Assign Method Static

Primary Address 115.110. XXXX x.x.x.x

Primary Netmask 255.255.255.192 x.x.x.x

Configure Secondary Interface Disabled

3. Static Routes

Static routes are used to create communication to remote networks. In a production environment, static routes are mainly configured for routing from a specific network to a network that can only be accessed through one point or one interface (single path access or default route).



Tip

- For smaller networks with just one or two routes, configuring static routing is preferable. This is often more efficient since a link is not being wasted by exchanging dynamic routing information.
- For networks that have a LAN side Gateway on Voice VLAN or Multi-Switch Edge Devices (MSEs) with Voice VLAN towards the SBC Edge static routing configurations are not required.

Add Static routes need to be added towards Eth1 interface 172.16.X.X(PSTN) and Eth2 interface 162.12.X.0(Zoom), as Zoom uses multiple IPs in this subnet.

Default static route is towards the Eth1 that is in a private network.

- Navigate to **Settings > Protocol > IP > Static Routes** to configure the routes.

Figure 7: Static Routes

[Expand All](#) | [Collapse All](#) | [Reload](#)

- ▶ Call Routing
- ▶ Signaling Groups
- ▶ Linked Signaling Groups
- ▶ Node Interfaces
- ▶ System
- ▶ Auth and Directory Services
- ▶ **Protocols**
 - ▶ DNS
 - ▶ **IP**
 - Static Routes**
 - Routing Table
 - Static ARP
 - Router Instances
 - Access Control Lists

Static IP Route Table

July 15, 2020 12:30:00 ↻

⊕ ⊖ ✕
Total 3 IP Route Rows

	Row ID	Destination IP	Mask	Gateway	Metric	Primary Key
<input type="checkbox"/>	1	0.0.0.0	0.0.0.0	10.54.19.1	1	1
<input type="checkbox"/>	5	172.16. XXXX	255.255.255.255	10.54. XXXX	1	5
<input type="checkbox"/>	6	162.12. XXXX	255.255.255.255	115.110. XXXX	1	6

4. TLS Configuration Between Ribbon SBC Edge and Zoom

Prerequisites:

- For TLS to work on the public side of network, a trusted CA (Certificate Authority) is needed. In this scenario, use GoDaddy as a Trusted CA.
- Enable Zoom trunk with TLS/SRTP.

Request a certificate for the SBC and configure it based on the example using GoDaddy as follows:

1. Generate a Certificate Signing Request (CSR) and obtain the certificate from a Certification Authority.
2. Import the Public CA Root/Intermediate Certificate and SBC certificate on the SBC.

Step 1: Generate a Certificate Signing Request and obtain the certificate from a Certification Authority (CA).

1. Navigate to **Settings > Security > SBC Certificates**.
2. Click **Generate SBC Edge CSR**.
3. Enter data in the required fields.
4. Click **OK**. After the Certificate Signing request finishes generating, copy the result to the clipboard.

Figure 8: Generate Certificate Signing Request

Generate Certificate Signing Request

Subject Distinguished Name

Common Name * Hostname or FQDN

Subject Alternative Name DNS comma-separated FQDN list

Email Address

ISO Country Code ▼

State/Province

Locality e.g.: City

Organization e.g.: Company

Organizational Unit e.g.: Department

Key Length ▼

Result

Copy CSR

```
-----BEGIN CERTIFICATE REQUEST-----
MIIDCzCCAfMCAQAwfjEmMCQGA1UEAxQdKi5jdXN0b21lcnMuaW50ZXJvcGRvbWFP
bi5jb20xHTAbBgkqhkiG9w0BCEQWVzZXIuYm4uY29tMQswCQYDVQQLJwJVDCCASl
UzELMAkGA1UECBMCToxDjAMBgNVBAoTBVNVbnVzMQswCQYDVQQLJwJVDCCASl
DQYJKoZIhvcNAQEBBQADggEPADCCAQoCggEBAP1m1uHXRgbKsGLGeOPwKFNOLuwi
FOgv0AugqrefvK5+Ru938w5OyrRsZZ5KN58vS/BI7tkqvZeqFZTEToUq23qvMADO
2OxJkZQzgheZ5dk39On1THemRYa7tdBtmyyD1F8XRFPEUaANOFtrLzyMPvFnJuls
sTNmjA76/i3Qg+80kY0X2266uoTzs2puNEOlKpqZ6yxWngEyp508DgZUKx53U6Yy
OyJNlLpXTUYeDMwDtslCM0j3YdV6KbcA/Z6ZMLHvis3B34q8c4gm0wEjwVLbknd4
t/gub6+ZQPGXVphgg3W6E8GUFVYzC6b36oHhCS6NJVT6qkNMKnKxRhkfLBUCAwEA
AaBIMEYGCsGSIb3DQEJDDjE5MDcwCQYDVROTBAlwADALBgNVHQ8EBAMCBaAwHQYD
VR0IBBYwFAYIKwYBBQUHAWEGCCsGAQUFBwMCMA0GCsGSIb3DQEBCwUAA4IBAQQD0
f0b+nhanA06rQxrjoGffcpPdjlCFt3SQQIAcxb7eR49BpSJzVINfO38IPmJgvYD8
w/h2JTFLExyzbkPKTIVdKaHb920ZgrGta5JYFaOYxF9mHBrZhCIMZc6qhv+58H9T
1K1r3wUelyR5e2PwKPP03LyFNvP4PbNc3XA0zh53mhZEas9EEcRP+J3raxVoaFUa
-----END CERTIFICATE REQUEST-----
```

5. Use the generated CSR text from the clipboard to obtain the certificate.

Step 2: Deploy the Root/Intermediate and SBC Certificates on the SBC.

After receiving the certificates from the certification authority, install the SBC Certificate and Root/Intermediate Certificates as follows:

1. Obtain Trusted Root and Intermediary signing certificates from your certification authority.
2. To install Trusted Root/Intermediate Certificates, go to **Settings > Security > SBC Certificates > Trusted Root Certificates**.
3. Click **Import** and select the trusted root certificates.
4. To install the SBC certificate, open **Settings > Security > SBC Certificates > SBC Edge Certificate**.
5. Validate the certificate is installed correctly.

Figure 9: Trusted CA certificate table

Trusted CA Certificate Table							July 16, 2020 15:48:59		
		Total 3 Certificate Rows							
	Common Name	Issuer	Start Validity	Expiration	Key Length	Display	Primary Key		
	Go Daddy Secure Cert...	Go Daddy Root Certif...	May 3, 2011	May 3, 2031	2048		2		
	Go Daddy Root Certif...	Go Daddy Root Certif...	Aug 31, 2009	Dec 31, 2037	2048		3		

6. Click **Import** and select **X.509 Signed Certificate**.

7. Validate the certificate is installed correctly.

Figure 10: Validate certificate

SBC Primary Certificate

Import

Export

July 16, 2020 15:54:32

Subject

Common Name *.customers.interopdomain.com
ISO Country Code
State or Province
Locality
Organization
Organizational Unit Domain Control Validated
Email Address

Issuer

Common Name Go Daddy Secure Certificate Authority - G2
ISO Country Code US
State or Province Arizona
Locality Scottsdale
Organization GoDaddy.com, Inc.
Organizational Unit http://certs.godaddy.com/repository
Email Address

Certificate

Not Valid Before Feb 5, 2020 22:06:11
Not Valid After Feb 6, 2021 22:19:01
Serial Number 5931A539DA417BC8
Signature Algorithm sha256WithRSAEncryption
Key Length 2048
Enhanced Key Usage TLS Web Server Authentication, TLS Web Client Authentication
Key Usage Digital Signature, Key Encipherment
Subject Alternative Name DNS: *.customers.interopdomain.com, DNS: customers.interopdomain.com
Verify Status OK

The following certificates need to be downloaded from Zoom Portal and uploaded into an SBC Edge Under Trusted CA Certificate.

Figure 11: Zoom Certificates

Trusted CA Certificate Table

October 29, 2021 16:37:43

Total 8 Certificate Rows

Filter...

<input type="checkbox"/>	Common Name	Issuer	Start Validity	Expiration	Key Length	Display	Primary Key
<div><div>▶</div><div><div></div><div></div></div></div>	DigiCert Global Root...	DigiCert Global Root...	Nov 10, 2006	Nov 10, 2031	2048		2
<div><div>▶</div><div><div></div><div></div></div></div>	DigiCert Global Root...	DigiCert Global Root...	Aug 1, 2013	Jan 15, 2038	2048		3
<div><div>▶</div><div><div></div><div></div></div></div>	DigiCert Global Root...	DigiCert Global Root...	Aug 1, 2013	Jan 15, 2038	384		4



As mentioned in Zoom Portal "Please download [DigiCert Global Root CA](#), [DigiCert Global Root G2](#), [DigiCert Global Root G3](#) and import to your IP phone if they are not in the trust list of the device."

TLS Profile

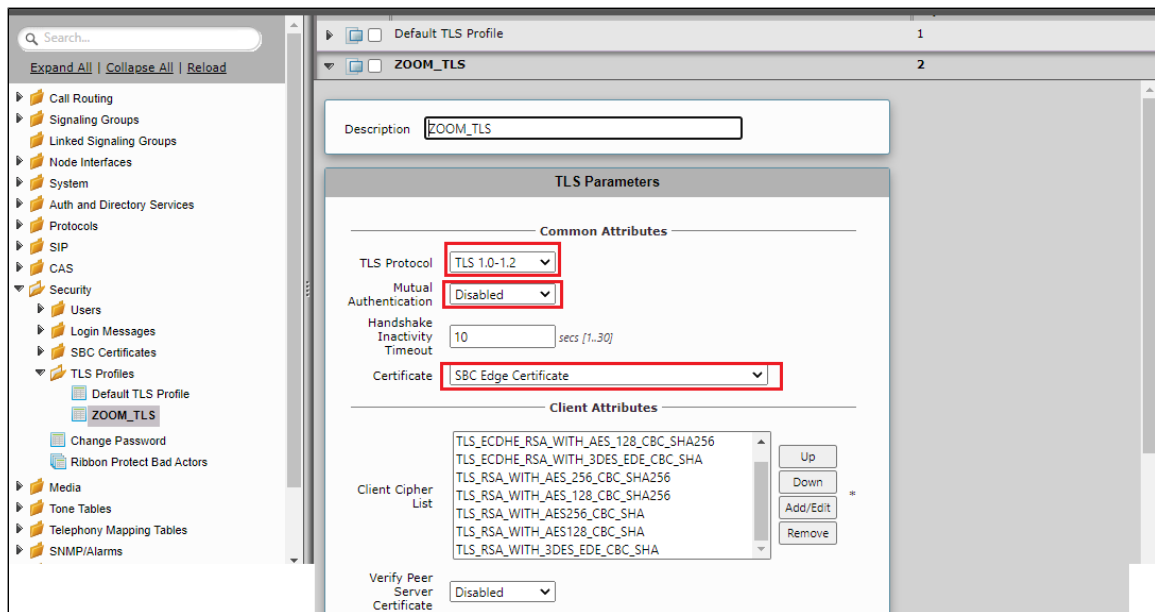
TLS Profile is required for the TLS handshake between the SBC Edge and Zoom. This profile defines cipher suites supported by the SBC Edge.

Default TLS Profile need to be attached to SIP Server Table on Zoom leg.

Navigate to **Security > TLS Profiles**. Use the Default TLS Profile with the following modifications:

- TLS Protocol as "TLS 1.0-1.2".
- Mutual Authentication "Disabled".
- Certificate as "SBC Edge Certificate".

Figure 12: Default TLS Profile



5. Media Profile

To create a Media Profile:

- Navigate to **Settings > Media > Media Profiles**.
- From the drop-down select **Create Media Profile > Voice Codec Profile**.

Figure 13: Media Profile

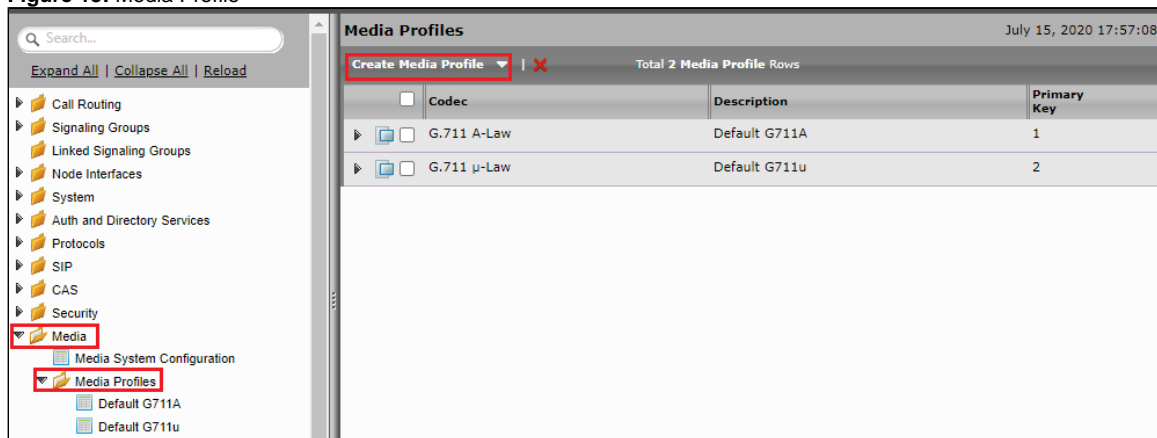


Figure 14: G711-A law

Media Profiles July 15, 2020 17:57:08

Create Media Profile | X Total 2 Media Profile Rows

<input type="checkbox"/>	Codec	Description	Primary Key
<input type="checkbox"/>	G.711 A-Law	Default G711A	1

Voice Codec Configuration

Description

Codec

Payload Size ms

Apply

Figure 15: G711 Mu law

Media Profiles July 15, 2020 17:57:08

Create Media Profile | X Total 2 Media Profile Rows

<input type="checkbox"/>	Codec	Description	Primary Key
<input type="checkbox"/>	G.711 A-Law	Default G711A	1
<input type="checkbox"/>	G.711 μ -Law	Default G711u	2

Voice Codec Configuration

Description

Codec

Payload Size ms

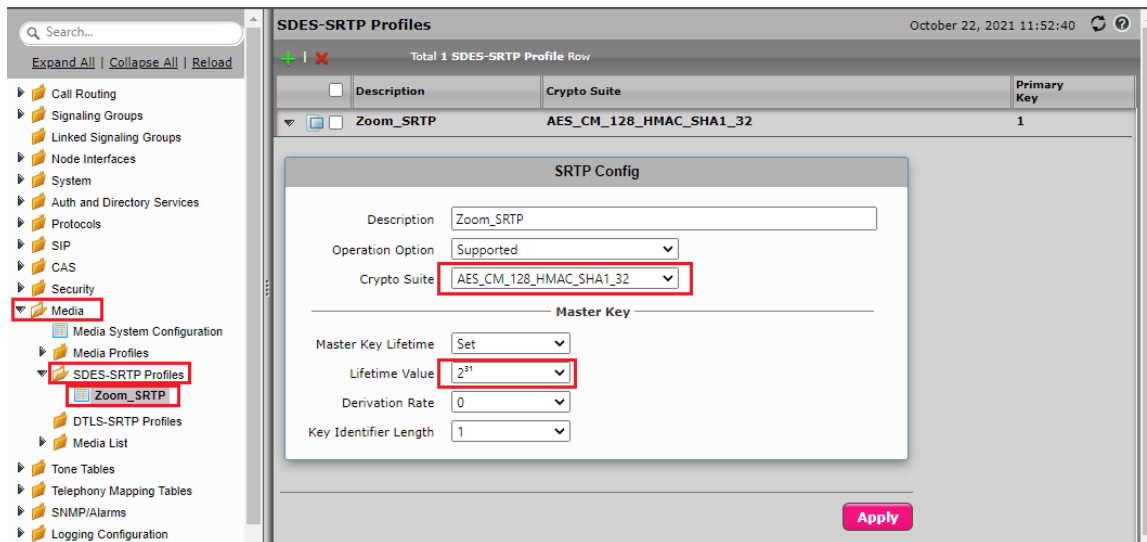
Apply

6. SRTP Profile

To create a SRTP Profile:

- Navigate to **Settings > Media > SDES-SRTP Profiles**.
- Select the Crypto Suite as "AES_CM_128_HMAC_SHA1_32".
- Set the LifeTime Value as shown in the diagram.

Figure 16: SDES-SRTP Profile

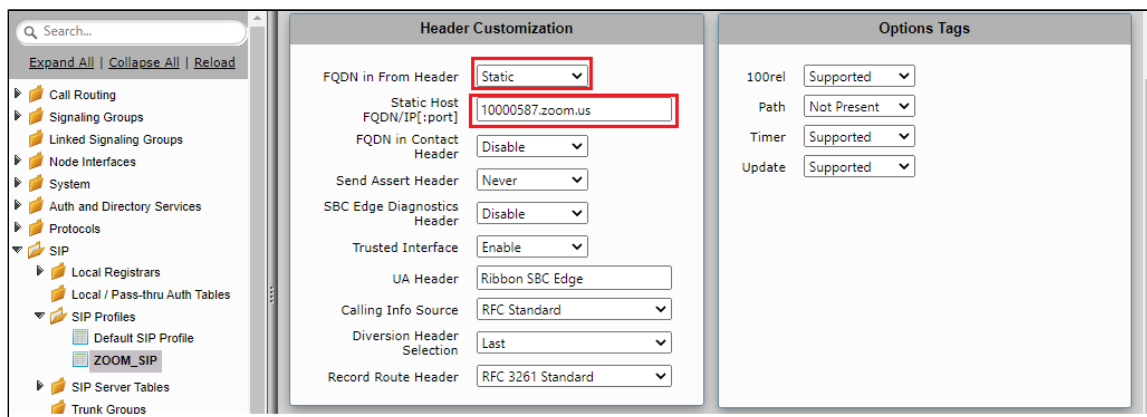
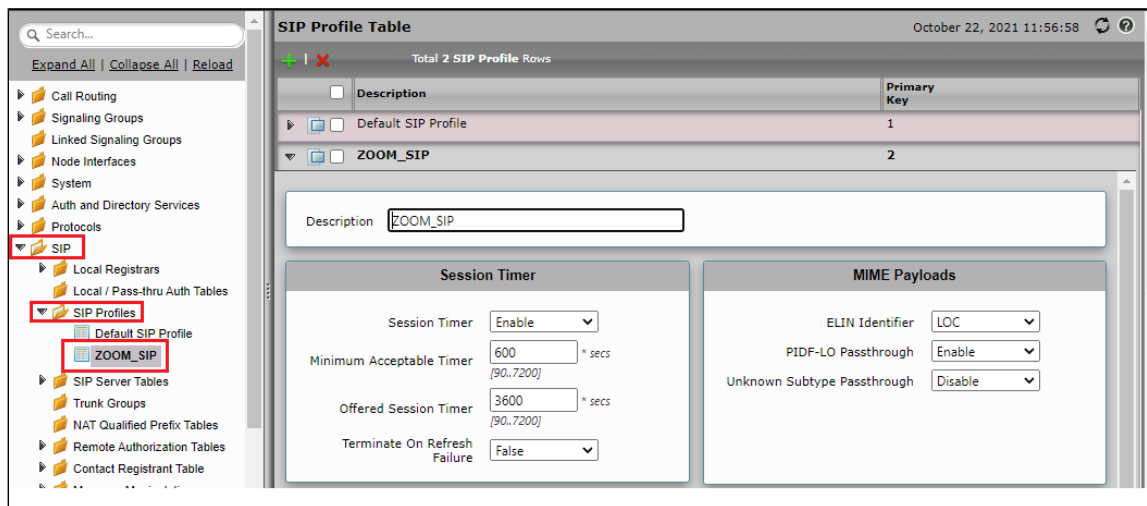


7. SIP Profile

A SIP profile is used to modify the different SIP parameters like Session timers, SIP Header Customization, and SDP Customization. Create a new SIP Profile "ZOOM_SIP".

- Navigate to **SIP > SIP Profiles > ZOOM_SIP**.

Figure 17: SIP profile



Search...

Expand All | Collapse All | Reload

- Call Routing
- Signaling Groups
 - Linked Signaling Groups
- Node Interfaces
- System
- Auth and Directory Services
- Protocols
 - SIP
 - Local Registrars
 - Local / Pass-thru Auth Tables
 - SIP Profiles
 - Default SIP Profile
 - ZOOM_SIP**
 - SIP Server Tables
 - Trunk Groups
 - NAT Qualified Prefix Tables
 - Remote Authorization Tables
 - Contact Registrant Table
 - Message Manipulation
 - Node-Level SIP Settings

Timers

Transport Timeout Timer ms
[5000..32000]

Maximum Retransmissions

Redundancy Retry Timer ms
[5000..180000]

RFC Timers

Timer T1 ms [100..10000]

Timer T2 ms [1000..80000]
(>= T1)

Timer T4 ms [1000..100000]

Timer D ms [5000..640000]

Timer B ms

Timer F ms

Timer H ms (64*TimerT1)

Timer J ms [4000..640000]

SDP Customization

Send Number of Audio Channels

Connection Info in Media Section

Origin Field Username default: SBC

Session Name default: VoipCall

Digit Transmission Preference

SDP Handling Preference

8. Analog Leg Configuration

Create profiles with a specific set of characteristics corresponding to Analog (PSTN). This includes configuration of the following entities on the Analog leg:

1. [CAS Profile](#)
2. [Signaling Groups](#)
3. [Transformation](#)
4. [Call Routing Table](#)

1. CAS Profile

- Navigate to **CAS > CAS Signaling Profiles > Create CAS Profile**.

Figure 18: CAS Profile

Create CAS Loop Start FXS Side Profile

Description

Zoom

Loop Start FXS Properties

Loop Start Type

Forward Disconnect

Forward Disconnect Duration

700

* ms [100..3000]

Disconnect Tone Generation

Disabled

Flashhook Signal Detection

Enabled

Maximum Flashhook Duration

700

* ms [50..1000]

Minimum Flashhook Duration

200

* ms [50..1000]

Inter-Digit Timeout

4000

* ms [250..30000]

Ringing Cadence

Cadence On

2000

* ms [50..9000]

Cadence Off

4000

* ms [50..9000]

Double Cadence

No

2. Signaling Groups

Signaling groups allow telephony channels to be grouped together for the purposes of routing and shared configuration. The signaling groups are the entity where calls are routed, as well as the location where Call Routes are selected. They are also the location where the Tone Tables and Action Sets are selected.

- Navigate to **Settings > Signaling Groups > Create Signaling Group**.
- From the drop-down select "CAS Signaling Group".
- Set CAS Signaling Profile as "(FXS)Zoom".
- Set Tone Table as "Default Tone Table".
- Assign the Channel Phone Number "809" for Port 1:1.
- Assign the Channel Phone Number "810" for Port 1:2.

Tip

Set Call Routing table as "Analog_Zoom" that is created in the [Call Routing Table](#) section.

Figure 19: Signalling Groups

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Description

Line Type

Analogue

Admin State

Enabled

Service Status

Up

Channels and Routing

Direction

Bidirectional

Channel Hunting

Own Number

Tone Table

Default Tone Table

Action Set Table

None

Call Routing Table

Analog_Zoom

No Channel Available Override

34: No Circuit/Channel Available

CAS Protocol

CAS Signaling Profile

(FXS) Zoom

Supplementary Services Profile

None

Caller ID Type

Disabled

Play Ringback

Always

Call Forwarding Feature

Disable

Assigned Channels

Total 2 CAS Channel Rows

	Port Name	Channel Phone Number	Hotline Enabled	Hotline Number	Call Forwarding Activated	Call Forwarding Number
	1:1	809	No		No	
	1:2	810	No		No	

Apply

3. Transformation

Transformation table is used to map the 3 digit analog line to 20 digit Zoom Username. In the current setup, "analog_sip" transformation rule applied for outgoing call towards Zoom.

Navigate to **Settings > Call Routing > Transformation**.

Figure 20: Transformation

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Search...

Expand All | Collapse All | Reload

Call Routing

Transformation

Analog_to_SIP

Passthrough Untouched

SIP_Analog

October 22, 2021 17:05:29

Total 3 Transformation Entry Rows

	Admin State	Input Field Type	Input Field Value	Output Field Type	Output Field Value	Match Type	Description	Primary Key
	Enabled	Calling Address/Number	809	Calling Address/Number	77746483298489165551	Optional (Match One)	analog_sip	1
	Enabled	Calling Address/Number	810	Calling Address/Number	20407385877802799170	Optional (Match One)	2nd line	2

Description: analog_sip

Admin State: Enabled

Match Type: Optional (Match One)

Input Field

Type: Calling Address/Number

Value: 809

Output Field

Type: Calling Address/Number

Value: 77746483298489165551

Description: 2nd line

Admin State: Enabled

Match Type: Optional (Match One)

Input Field

Type: Calling Address/Number

Value: 810

Output Field

Type: Calling Address/Number

Value: 20407385877802799170

4. Call Routing Table

Call Routing allows calls to be carried between signaling groups, thus allowing calls to be carried between ports, and between protocols (like ISDN to SIP). Routes are defined by Call Routing Tables, which allow for flexible configuration where calls are carried and how they are translated. These tables are one of the central connection points of the system, linking Transformation Tables, Message Translations, Cause Code Reroute Tables, Media Lists and the three types of Signaling Groups (ISDN, SIP and CAS).

- Navigate to **Settings > Call Routing > Call Routing Table**.
- Set Number/Name Transformation Table as "Add_plusOne" as created in an earlier step.
- Destination Signaling Groups as "(SIP)Zoom_SG" in the Media. Select the following options:
 - Audio/Fax Stream Mode as "DSP".
 - Media Transcoding as "Disabled".
 - Media list as "Default Media List".



Tip

Attach the Media List and Destination Signaling Groups that are created in [Zoom Leg Configuration](#).

Figure 21: Call Routing Table

Search...

Expand All | Collapse All | Reload

Call Routing

Transformation

Time of Day Table

Call Routing Table

Default Route Table

Analog_Zoom

Zoom_Analog

Analog_Zoom

October 22, 2021 17:08:49

Display Counters

Total 1 Call Route Entry Row

Admin State	Priority	Transformation Table	Destination Type	First Signaling Group	Description	Fork Call	Primary Key
1	1	Analog_to_SIP	Normal	(SIP) Zoom	Analog_SIP	No	1

Admin State	Priority	Transformation Table	Destination Type	First Signaling Group	Description	Fork Call	Primary Key
1	1	Analog_to_SIP	Normal	(SIP) Zoom	Analog_SIP	No	1

Route Details

Description: Analog_SIP

Admin State: Enabled

Route Priority: 1

Call Priority: Normal

Number/Name Transformation Table: Analog_to_SIP

Time of Day Restriction: None

Destination Information

Destination Type: Normal

Message Translation Table: None

Cause Code Reroutes: None

Cancel Others upon Forwarding: Disabled

Fork Call: No

Destination Signaling Groups: (SIP) Zoom

Enable Maximum Call Duration: Disabled

Media

Audio/Fax Stream Mode: DSP

Video/Application Stream Mode: Disabled

Media Transcoding: Disabled

Media List: Default Media List

Quality of Service

Quality Metrics Number of Calls: 10 [1..100]

Quality Metrics Time Before Retry: 10 [1-60] min.

Min. ASR Threshold: 0 [% [0..100]]

Enable Min MOS Threshold: Disabled

Enable Max. R/T Delay: Enabled

Max. R/T Delay: 65535 ms [1..65535]

Enable Max. Jitter: Enabled

Max. Jitter: 3000 ms [1..3000]

9. Zoom Leg Configuration

Create profiles with a specific set of characteristics corresponding to Zoom. This includes configuring the following entities on the Zoom leg:

1. [Media List](#).
2. [Contact Registrant Table](#).
3. [Remote Authorization Tables](#).
4. [SIP Server Tables](#).
5. [Signaling Group](#).
6. [Transformation](#).
7. [Call Routing Table](#).

1. Media List

Media List allows you to specify a set of codecs used for the call. They contain a list of codecs, defined in [Media Profile](#).

- "Add/Edit" to add the different Media profile as created earlier.

- As the Zoom leg would be SRTP, attach the SDES-SRTP Profile as "Zoom_SRTP" as created earlier.
- Set RTCP mode to "RTCP".
- Set Silence Suppression to "enabled".

Figure 22: Media List

Media List View October 22, 2021 19:41:34

Total 2 Media List Rows

Description	Primary Key
Default Media List	1

Description: Default Media List

Media Profiles List:

- Default G711A
- Default G711u

Buttons: Up, Down, **Add/Edit**, Remove

SDES-SRTP Profile: Zoom_SRTP associated SIP SG Listen Ports should be TLS only: +

DTLS-SRTP Profile: None +

Media DSCP: 46 * [0..63]

RTCP Mode: RTCP

Dead Call Detection: Disabled

Silence Suppression: Enabled

Gain Control

Receive Gain: 0 [-14..+6] dB

Transmit Gain: 0 [-14..+6] dB

Digit Relay

Digit (DTMF) Relay Type: RFC 2833

Digit Relay Payload Type: 101 [96..127]

Passthrough/Tone Detection

Modem Passthrough: Enabled

Fax Passthrough: Enabled

CNG Tone Detection: Disabled

Fax Tone Detection: Enabled

DTMF Signal to Noise: 0 [-3..+6] dB

DTMF Minimum Level: -38 [-48..-14] dBm0

2. Contact Registrant Table

The Contact registrant table contains the address of record URI required for REGISTER with Zoom.

- Navigate to **Settings > SIP > Contact Registrant Table**.
- Type of Address of Record "Remote".
- Address of Record URI.
- Add the Contact URI Username.

Figure 23: Contact Registrant Table - Port 1 Config

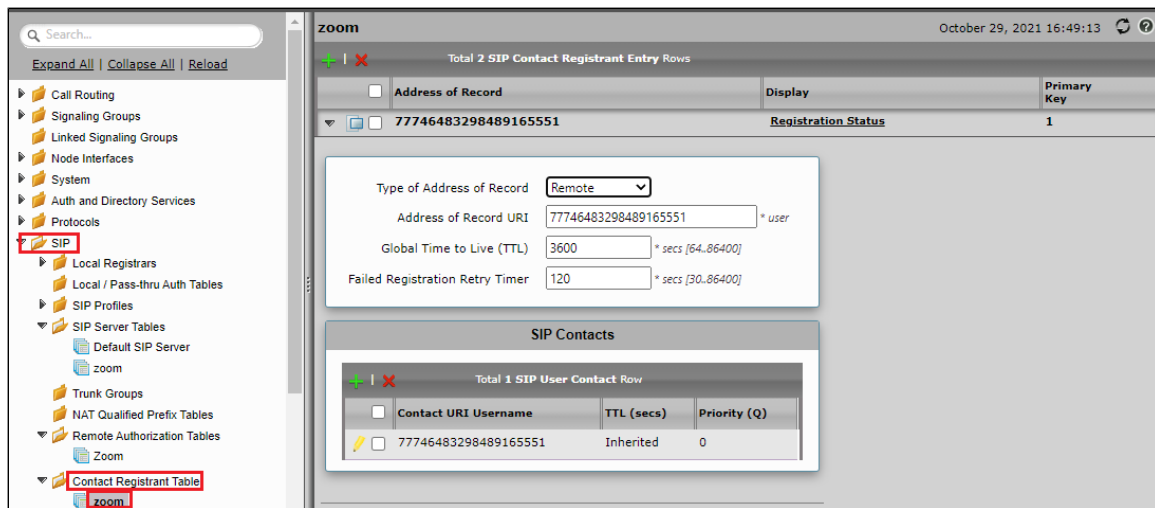
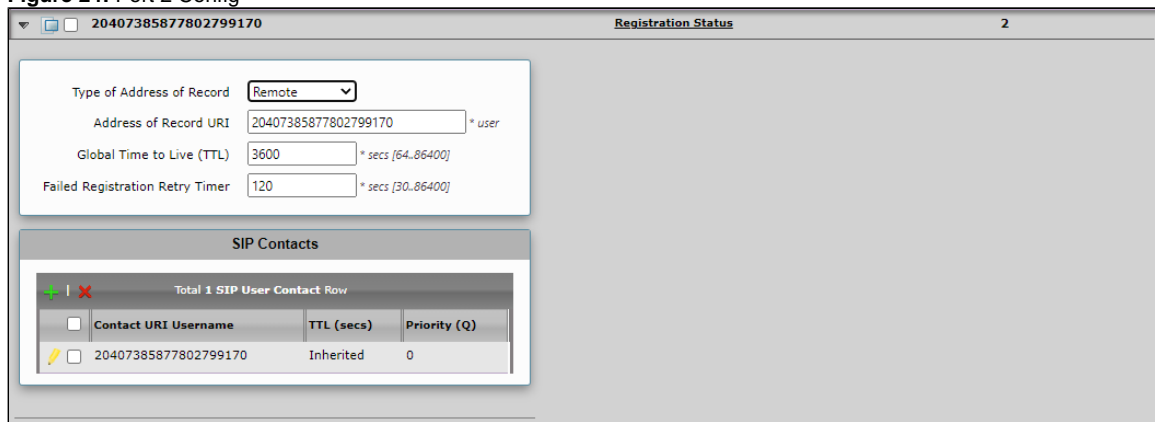


Figure 24: Port 2 Config



3. Remote Authorization Tables

A Remote authorization table contains a realm, authentication ID and password, used when the REGISTER and INVITE is challenged by Zoom.

- Navigate to **Settings > SIP > Remote Authorization Tables**.
- Realm as "10000587.zoom.us".
- Authentication ID .
- From URI User Match "Regex".

Figure 25: Remote Authorization Table - Port 1 Config

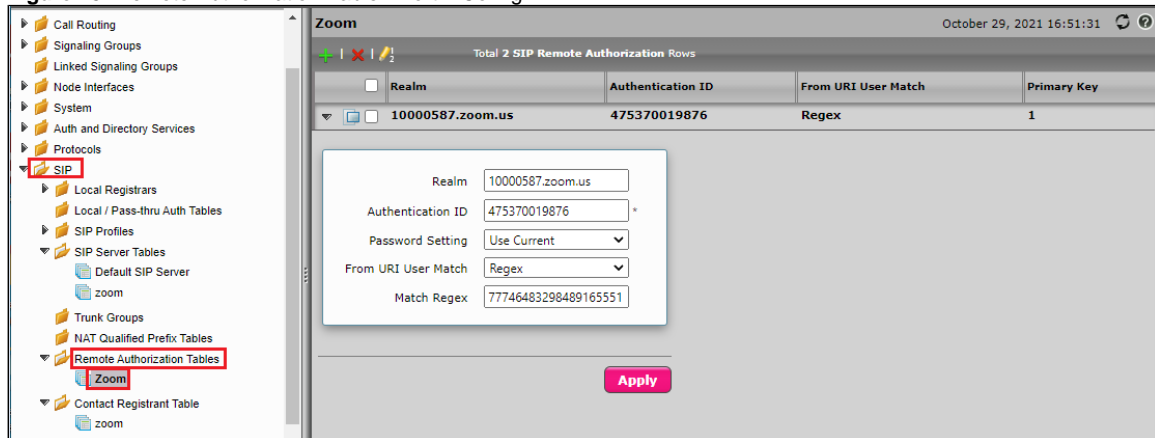


Figure 26: Port 2 Config

10000587.zoom.us 247646854027 Regex 2

Realm 10000587.zoom.us

Authentication ID 247646854027 *

Password Setting Use Current

From URI User Match Regex

Match Regex 20407385877802799170

Apply

4. SIP Server Tables

The SIP Server Tables contain information about the SIP devices connected to the SBC Edge. The entries in the tables provide information about the IP Addresses, ports and transport protocols used to communicate with each server.

- Navigate to **Settings > SIP > SIP Server Tables > Create SIP Server**.
- From the drop-down select "IP/FQDN".
- Configure the SIP server table with Zoom Fqdn (for example, 10000587.zoom.us in our case).
- Port as "5091".
- Configure Transport protocol as "TLS".
- Set TLS Profile as "ZOOM_TLS" as created in the section [TLS Profile](#).

Figure 27: SIP

Search...

Expand All | Collapse All | Reload

Call Routing

Transformation

Time of Day Table

Call Routing Table

Call Actions

Signaling Groups

Linked Signaling Groups

Node Interfaces

System

Auth and Directory Services

Protocols

SIP

Local Registrars

Local / Pass-thru Auth Tables

SIP Profiles

SIP Server Tables

Default SIP Server

zoom

zoom October 22, 2021 19:48:05

Create SIP Server Total 1 SIP Server Row

Host / Domain	Server Lookup	Port	Protocol	Display Counters	Priority	Primary Key
10000587.zoom.us	IP/FQDN	5091	TLS	Counters	1	1

Host / Domain	Server Lookup	Port	Protocol	Display Counters	Priority	Primary Key
10000587.zoom.us	IP/FQDN	5091	TLS	Counters	1	1

Server Host

Server Lookup: IP/FQDN

Priority: 1

Host FQDN/IP: 10000587.zoom.us

Host IP Version: IPv4

Port: 5091

Protocol: TLS

TLS Profile: ZOOM_TLS

Transport

Monitor: None

Remote Authorization and Contacts

Remote Authorization Table: Zoom

Contact Registrant Table: zoom

Clear Remote Registration on Startup: True

Contact URI Randomizer: False

Stagger Registration: False

Retry Non-State Nonce: True

Authorization on Refresh: True

Session URI Validation: Liberal

Connection Reuse

Reuse: True

Sockets: 4

Reuse Timeout: Forever

5. Signaling Groups

Signaling groups allow telephony channels to be grouped together for the purposes of routing and shared configuration. The signaling groups are the entity where calls are routed, as well as the location where Call Routes are selected. They are also the location where Tone Tables and Action Sets are selected. In the case of SIP, they specify protocol settings and link to server, media and mapping tables.

- Navigate to **Settings > Signaling Groups > Create Signaling Group**.
- From the drop-down select "SIP Signaling Group".
- Set SIP Profile as "ZOOM_SIP".
- Set SIP Server Table as "zoom".
- Set Media List ID as "Zoom_ML".
- Set Signaling/Media Source IP as "Ethernet 2 IP(115.110.X.X)".
- Configure Federated IP as Zoom IP (162.12.X.X).



Tip

Set the Call Routing table as "Zoom_RT" as created in the [Call Routing Table](#) section.

Figure 28: Signalling Groups

Search...

Expand All | Collapse All | Reload

- Call Routing
- Signaling Groups**
 - (SIP) Zoom**
 - (CAS) analog
- Linked Signaling Groups
- Node Interfaces
- System
- Auth and Directory Services
- Protocols

Signaling Group Table

October 22, 2021 19:55:19

Total 2 Signaling Group Rows

Type	Description	Admin State	Service Status	Display	Primary Key
SIP	Zoom	Enabled	Up	Counters Channels Sessions	1

Description: Zoom

Admin State: Enabled

Service Status: Up

SIP Channels and Routing

Action Set Table: None +

Call Routing Table: Zoom_Analog +

No. of Channels: 60 [1..960]

SIP Profile: ZOOM_SIP +

SIP Mode: Basic Call

Agent Type: Back-to-Back User Agent

Interop Mode: Standard

SIP Server Table: zoom +

Load Balancing: Priority: Register Active Only

Channel Hunting: Round Robin

Notify Lync CAC Profile: Disable

Media Information

Supported Audio/Fax Modes: DSP + Remove

Supported Video/Application Modes: Disabled

Media List ID: Default Media List +

Play Ringback: Auto on 180/183

Tone Table: Default Tone Table +

Play Congestion Tone: Disable

Early 183: Disable

Allow Refresh: Enable

Challenge Request: Disable

Outbound Proxy IP/FQDN:

Outbound Proxy Port: [1..65535]

No Channel Available Override: 34: No Circuit/Channel Available

Call Setup Response Timer: 255 [180..750] secs

Call Proceeding Timer: 180 [24..750] secs

QoE Reporting: Disabled

Use Register as Keep Alive: Enable

Forked Call Answered Too Soon: Disable

Mapping Tables

SIP To Q.850 Override Table: Default (RFC4497) +

Q.850 To SIP Override Table: Default (RFC4497) +

Pass-thru Peer SIP Response Code: Enable

SIP Failover: +

Teams Local Media Optimization: Disable

Signaling/Media Source IP: Ethernet 2 IP (115.110.170.209)

Signaling DSCP: 40 [0..63]

NAT Traversal

ICE Support: Disabled

Static NAT - Outbound

Outbound NAT Traversal: None

Static NAT - Inbound

Detection: Disabled

Listen Ports

Total 1 SIP Listen Port Row

Port	Protocol	TLS Profile ID
5091	TLS	ZOOM_TLS

Federated IP/FQDN

Total 1 SIP Federated IP Row

IP/FQDN	Netmask/Prefix
192.204.1.1	255.255.255.0

Message Manipulation: Disabled

6. Transformation

The "Remove_plusOne" transformation rule is required for the call towards PSTN.

Navigate to **Settings > Call Routing > Transformation**.

Figure 29: Transformation

Search...

Expand All | Collapse All | Reload

Call Routing

Transformation

Analog_to_SIP

Passthrough Untouched

SIP_Analog

SIP_Analog

October 22, 2021 19:59:57

Total 2 Transformation Entry Rows

Admin State	Input Field Type	Input Field Value	Output Field Type	Output Field Value	Match Type	Description	Primary Key
<input checked="" type="checkbox"/>	Called Address/Number	77746483298489165551	Called Address/Number	809	Optional (Match One)	SIP_analog	1
<input checked="" type="checkbox"/>	Called Address/Number	20407385877802799170	Called Address/Number	810	Optional (Match One)	To_2ndline	2

The image displays two screenshots of a web-based configuration interface for a Call Routing Table. Each screenshot shows a configuration form for a specific routing rule.

Top Screenshot:

- Header:** Called Address/Number 77746483298489165551 | Called Address/Number 809 | Optional (Match One) | SIP_analog 1
- Description:** SIP_analog
- Admin State:** Enabled
- Match Type:** Optional (Match One)
- Input Field:**
 - Type: Called Address/Number
 - Value: 77746483298489165551
- Output Field:**
 - Type: Called Address/Number
 - Value: 809

Bottom Screenshot:

- Header:** Called Address/Number 20407385877802799170 | Called Address/Number 810 | Optional (Match One) | To_2ndline 2
- Description:** To_2ndline
- Admin State:** Enabled
- Match Type:** Optional (Match One)
- Input Field:**
 - Type: Called Address/Number
 - Value: 20407385877802799170
- Output Field:**
 - Type: Called Address/Number
 - Value: 810

7. Call Routing Table

Call Routing allows calls to be carried between signaling groups, thus allowing calls to be carried between ports, and between protocols (such as ISDN to SIP). Routes are defined by Call Routing Tables, which allow for flexible configuration where calls are carried, and how they are translated. These tables are one of the central connection points of the system, linking Transformation Tables, Message Translations, Cause Code Reroute Tables, Media Lists and the three types of Signaling Groups (ISDN, SIP and CAS).

- Navigate to **Settings > Call Routing > Call Routing Table**.
- Set Number/Name Transformation Table as "Remove_plusOne" as created in an earlier step.
- Destination Signaling Groups as "(SIP)IP_PBX_SG" In the Media, select the following options:
 - Audio/Fax Stream Mode as "DSP".
 - Media Transcoding as "Enabled".
 - Media list as "Default Media List".



Tip

Attach the Media List and Destination Signaling Groups that are created in the [Analog Leg Configuration](#) section.

Figure 30: Call Routing Table

Zoom_Analog

October 22, 2021 20:02:02

Expand All | Collapse All | Reload

Search...

Call Routing

- Transformation
- Time of Day Table
- Call Routing Table
 - Default Route Table
 - Analog_Zoom
 - Zoom_Analog

Admin State	Priority	Transformation Table	Destination Type	First Signaling Group	Description	Fork Call	Primary Key
<input checked="" type="checkbox"/>	1	SIP_Analog	Normal	(CAS) analog	Zoom_analog	No	1

Route Details

Description: Zoom_analog

Admin State: Enabled

Route Priority: 1

Call Priority: Normal

Number/Name Transformation Table: SIP_Analog

Time of Day Restriction: None

Destination Information

Destination Type: Normal

Message Translation Table: None

Cause Code Reroutes: None

Cancel Others upon Forwarding: Disabled

Fork Call: No

Destination Signaling Groups: (CAS) analog

Buttons: Up, Down, Add/Edit, Remove

Enable Maximum Call Duration: Disabled

Media

Audio/Fax Stream Mode: DSP

Video/Application Stream Mode: Disabled

Media Transcoding: Disabled

Media List: Default Media List

Quality of Service

Quality Metrics Number of Calls: 10 [1..100]

Quality Metrics Time Before Retry: 10 min [1-60]

Min. ASR Threshold: 0 % [0..100]

Enable Min MOS Threshold: Disabled

Enable Max. R/T Delay: Enabled

Max. R/T Delay: 65535 ms [1..65535]

Enable Max. Jitter: Enabled

Max. Jitter: 3000 ms [1..3000]

Route Details

Description

Zoom_analog

Admin State

Enabled

Route Priority

1

Call Priority

Normal

Number/Name Transformation Table

SIP_Analog

+

Time of Day Restriction

None

+

Destination Information	
Destination Type	Normal
Message Translation Table	None
Cause Code Reroutes	None
Cancel Others upon Forwarding	Disabled
Fork Call	No
Destination Signaling Groups	<div> (CAS) analog </div> <div> Up Down Add/Edit </div>

Remove

Enable Maximum Call Duration

Disabled

Media	
Audio/Fax Stream Mode	<input type="text" value="DSP"/>
Video/Application Stream Mode	Disabled
Media Transcoding	<input type="text" value="Disabled"/>
Media List	<input type="text" value="Default Media List"/>

Quality of Service	
Quality Metrics Number of Calls	<input type="text" value="10"/> [1..100]
Quality Metrics Time Before Retry	<input type="text" value="10"/> [1-60] min.
Min. ASR Threshold	<input type="text" value="0"/> % [0..100]
Enable Min MOS Threshold	<input type="text" value="Disabled"/>
Enable Max. R/T Delay	<input type="text" value="Enabled"/>
Max. R/T Delay	<input type="text" value="65535"/> ms [1..65535]
Enable Max. Jitter	<input type="text" value="Enabled"/>
Max. Jitter	<input type="text" value="3000"/> ms [1..3000]

Section B: Zoom Configuration

The SBC 1000 can be provisioned as an ATA/IAD with Zoom, contact Zoom for Provisioning details.

The details provided by Zoom to configure Analog(FXS) Port 1 and Port 2 for the SBC 1000 are as follows:

```
Ribbon SBC 1K Generic Device Account details
MAC: 00:10:23:e1:04:2f

SIP Account SBC-1K-FXS1
Extension: 809

1. SIP Domain: 10000587.zoom.us
2. Outbound Proxy: gossip0h.sc.zoom.us:5091
3. User Name: 77746483298489165551
4. Authorization ID: 475370019876
5. Password: xxxxx
=====

SIP Account 1SBC-1K-FXS2
Extension: 810

1. SIP Domain: 10000587.zoom.us
2. Outbound Proxy: gossip0h.sc.zoom.us:5091
3. User Name: 20407385877802799170
4. Authorization ID: 247646854027
5. Password: xxxxx
```

Configuring Supplementary Services on Zoom

Zoom supports multiple supplementary services. To configure different supplementary services in Zoom, refer to the following links:

- Auto Receptionist: https://support.zoom.us/hc/en-us/articles/360001297663-Getting-started-with-Zoom-Phone-admin-#h_a625f531-94c6-4291-909e-3d68ad685b68
- Call Flip: <https://support.zoom.us/hc/en-us/articles/360034613311-Using-Call-Flip>
- Shared Line Appearance (SLA) or Call Delegation: <https://support.zoom.us/hc/en-us/articles/360032881731>
- Shared Line Group (SLG): <https://support.zoom.us/hc/en-us/articles/360038850792/>

Supplementary Services and Features Coverage

The following checklist depicts the set of services/features covered through the configuration defined in this Interop Guide.

Sr. No.	Supplementary Features/Services	Coverage
1	Basic Registration over TLS	✓
2	Basic Call Setup	✓
3	Basic Call Termination	✓
4	Auto Receptionist (Auto Attendant)	✓
5	Call Hold/Resume	✓
6	Call Transfer - Blind (Cold transfer)	✓
7	Call Transfer - Consult (Warm transfer)	✓
8	Conference	✓
9	Call Waiting	✓
10	Call Queue	✓
11	Shared Line Group (SLG)	✓
12	Shared Line Appearance (SLA) or Call Delegation	✓
13	Call Recording	✓
14	Call Flip	✓

15	Call Park	✓
----	-----------	---

Legend

✓	Supported
✗	Not Supported
N/A	Not Applicable

Caveats

The following issues are observed during the testing:

- Ringback tone not heard on analog phone, when call is dialed from Analog Phone behind SBC 1K/2K to Zoom Client.
- The SBC Edge sends 503 Service Unavailable for the Out-of-Dialog NOTIFY received from Zoom Server.

The issues above must fixed in the upcoming SBC 1000 and 2000 software release.

Support

For any support related queries about this guide, contact your local Ribbon representative, or use the details below:

- Sales and Support: 1-833-742-2661
- Other Queries: 1-877-412-8867
- Website: <https://ribboncommunications.com/about-us>

References

For detailed information about Ribbon products & solutions, refer to:

<https://ribboncommunications.com/products>

For information about Zoom products & solutions, refer to:

<https://zoom.us/>

Conclusion

This Interoperability Guide describes a successful configuration covering Zoom interop with Ribbon Edge ATA/IAD.

All features and capabilities tested are detailed within this document - any limitations, notes or observations are also recorded in order to provide the reader with an accurate understanding of what has been covered, and what has not.

Configuration guidance is provided to enable the reader to replicate the same base setup - there maybe additional configuration changes required to suit the exact deployment environment.

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