
Ribbon SBC Edge Configuration with Zoom BYOC

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Document Overview

This document outlines the configuration best practices for the Ribbon SBC Edge (SBC 1K, 2K, SWeLite) when deployed with Zoom Bring Your Own Carrier (BYOC). This means that for all subscribers catering to Zoom customers, the PSTN calls terminating through the local SBC Edge are directly connected to the Service Provider of their choice.

A Session Border Controller (SBC) is a network element deployed to protect SIP based Voice over Internet Protocol (VoIP) networks. Early deployments of SBCs were focused on the borders between two service provider networks in a peering environment. This role has now expanded to include significant deployments between a service provider's access network and a backbone network to provide service to residential and/or enterprise customers. The interoperability compliance testing focuses on verifying inbound and outbound call flows between Ribbon SBC 1K/2K /SWeLite and Zoom cloud. Ribbon SBC 1K/2K/SWeLite is deployed on the customer site to resolve any potential numbering format issue between Zoom and the customer's existing carrier dial plan numbering.

This guide contains the following sections:

- [Section A: SBC Edge Configuration](#)
 - Captures general SBC Edge configurations for deploying with Zoom BYOC.
- [Section B: Zoom Web BYOC configuration](#)
 - Captures the Zoom BYOC configuration.
 - Test all basic calls, along with the supplementary features like call hold, call transfer, and conference with configurations from Section A and Section B.
 - Configure Advanced supplementary features on Zoom as mentioned in [Supplementary Services Configuration on Zoom](#). These include:
 - Auto Receptionist
 - Call Flip
 - Shared Line Appearance (SLA) or Call Delegation
 - Shared Line Group (SLG)



Note

SBC 1K, 2K and SWeLite are represented as SBC Edge in the subsequent sections.



References

For additional information on Zoom, refer to <https://zoom.us>

For additional information on the Ribbon SBC, refer to <https://ribboncommunications.com/>

Non-Goals

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

Audience

This is a technical document intended for telecommunications engineers with the purpose of configuring both the Ribbon SBCs and the third-party product. Steps will require navigating the third-party product as well as the Ribbon SBC Command Line Interface (CLI). Understanding of the basic concepts of TCP/UDP, IP/Routing, and SIP/RTP is needed to complete the configuration and any necessary 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.

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 SBC 1000 /2000	V08.01.00-526
	Ribbon SWeLite	V08.01.05-239
Zoom	Zoom app Desktop	5.0.5(26213.0602)
	Zoom app Mobile	5.0.5(26211.0602)
Third-party Equipment	Kapanga Softphone	1.00
	Phonerlite	2.77
	Zoiper	5.3.8



Note

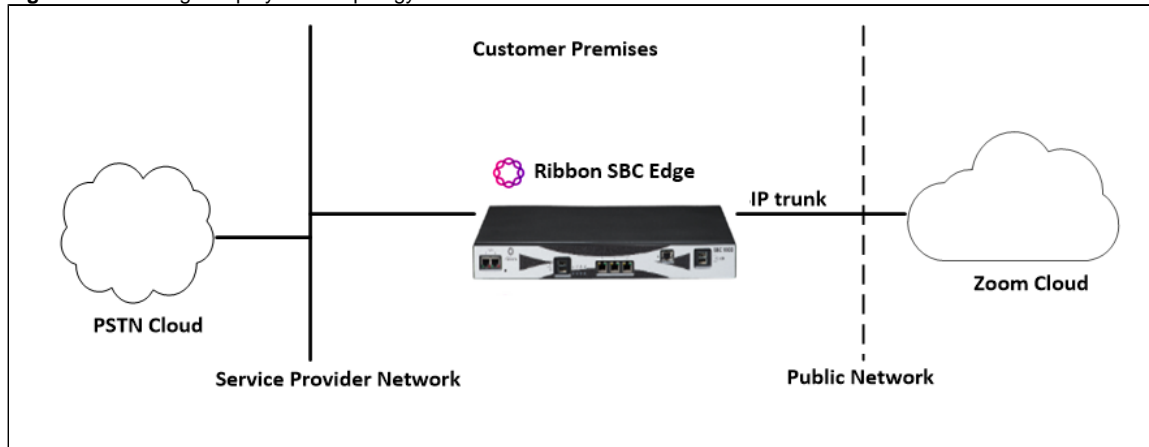
Ribbon SBC Edge portfolio includes SBC 1000, SBC 2000 (both are appliance based) and SBC SWeLite (virtualized platform). Software Version is applicable to Ribbon SBC Edge portfolio (1000, 2000, SWeLite) and hence this configuration guide is valid for all these devices.

Network Topology Diagram

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

SBC Edge Deployment Topology

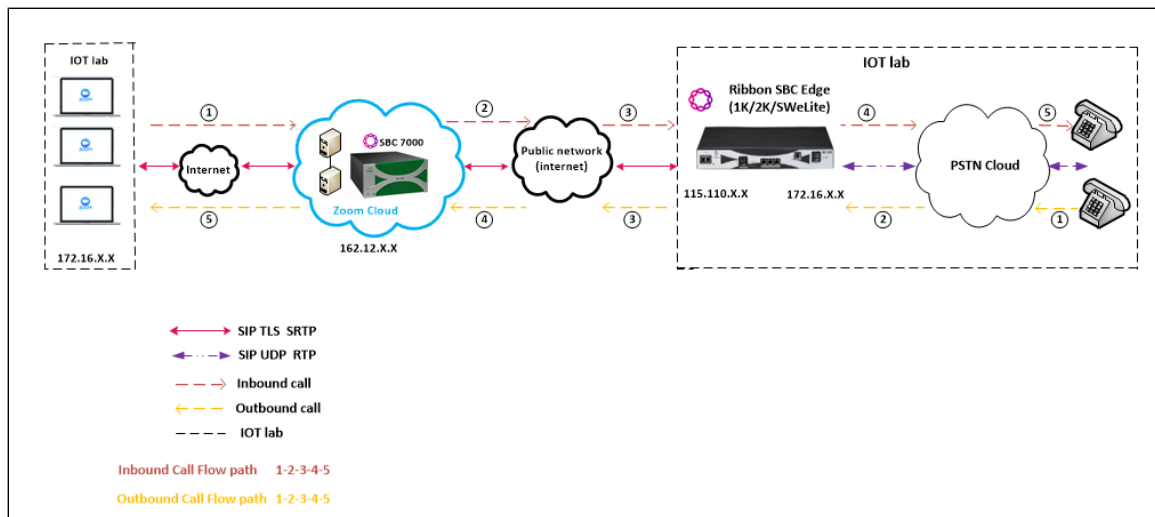
Figure 1: SBC Edge Deployment Topology



Interoperability Test Lab Topology

The following lab topology diagram shows connectivity between Zoom and Ribbon SBC Edge (1K/2K/SWeLite).

Figure 2: Interoperability Test Lab Topology



Section A: SBC Edge Configuration

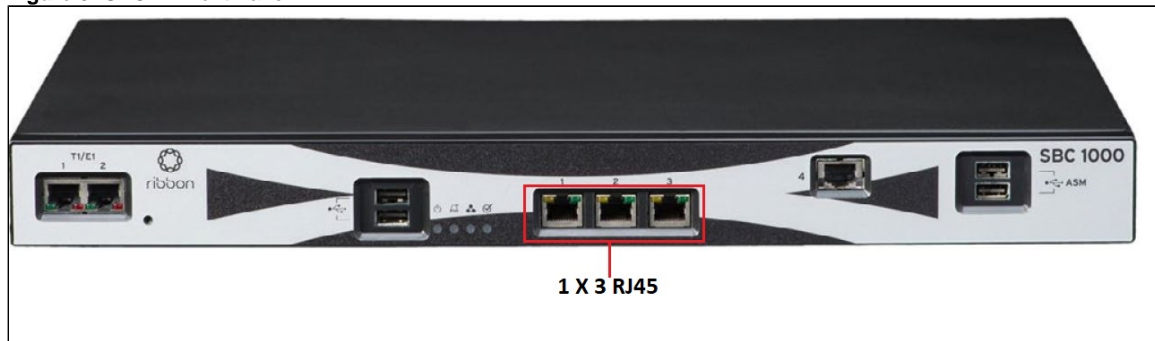
The following SBC Edge configurations are included in this section:

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. [PSTN Leg Configuration](#)
9. [Zoom Leg Configuration](#)

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

1. Connectivity

Figure 3: SBC1K Front Panel



**Note**

SBC1K 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

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

Identification/Status

Interface Name: Ethernet 1 IP
I/F Index: 2
Alias: Private Interface
Description: Private Interface
Admin State: Enabled

Networking

MAC Address: 00:10:23:e0:01:0e
IP Addressing Mode: IPv4

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

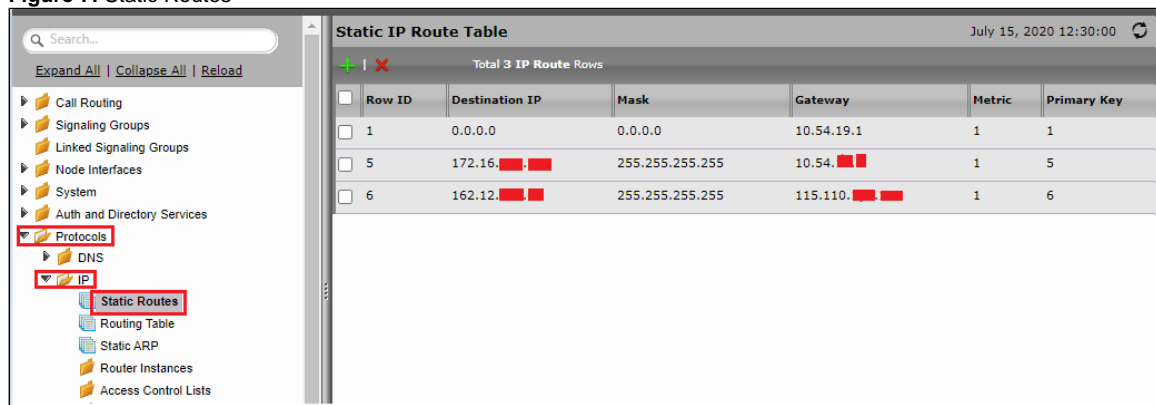
- For smaller networks with just one or two routes, configuring static routing is preferable. This is often more efficient since a link is not being wasted by exchanging dynamic routing information.
- For networks that have a LAN side Gateway on Voice VLAN or Multi-Switch Edge Devices (MSEs) with Voice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

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

Figure 7: Static Routes



Row ID	Destination IP	Mask	Gateway	Metric	Primary Key
1	0.0.0.0	0.0.0.0	10.54.19.1	1	1
5	172.16.1.0	255.255.255.255	10.54.19.1	1	5
6	162.12.1.0	255.255.255.255	115.110.1.1	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, GoDaddy is used as a Trusted CA.
- Enable Zoom BYOC 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
bi5jb20xHTAbBgkqhkiG9w0BCQEWDnVzZXl0b21lcnMuaW50ZXJvcGRvbWFP
UzELMAkGA1UECBMCTG9wZS50b21lcnMuaW50ZXJvcGRvbWFPJlVDCCAIw
DQYJKoZIhvcNAQEBBQADggEPADCCAQoCggEBAP1m1uHXRgbKsGLGeOPwKFNOLuwi
FOgv0AugqrefvK5+Ru938w5OyrRsZZ5KN58vS/BI7tkqvZeqFZTEToUq23qvMADO
2OxJkZQzghZ5dk39On1THemRYa7tdBtmyyD1F8XRFPEUaANOfTrLzyMPvFnJuls
sTNmjA76/i3Qg+80kY0X2266uoTzs2puNEOIKpqZ6yxWngEyp50BDgZUKx53U6Yy
OyJNlLpXTUYeDMwDtsICM0j3YdV6KbcA/Z6ZMLHvis3B34q8c4gm0wEjwVLbknd4
t/gub6+ZQPGXVphgg3W6E8GUFVyzC6b36oHhCS6NJVT6qkNMKnKxRhkFLBUCAwEA
AaBIMEYGCSqSIB3DQEDjE5MDcwCQYDVR0TBAlwADALBgNVHQ8EBAMCBAAwHQYD
VR0IBBYwFAYIKwYBBQUHAWEGCCsGAQUFBwMCMAOGCSqSIB3DQEBcwUAA4IBAQD0
f0b+nhanA06rQxrjoGffcpPdjlCFt3SQQIAcxb7eR49BpSjzVINfO38IPmJgvYD8
w/h2JTLExyzbkPKTIVdKaHb920ZgrGta5JYFaOYx9mHBrZhCIMZc6qhv+58H9T
1K1r3wUelyR5e2PwKPP03LyFNvP4PbNc3XA0zh53mhZEqs9EEcRP+J3rqxVoaFUa
-----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							
Total 3 Certificate Rows							
<input type="checkbox"/>	Common Name	Issuer	Start Validity	Expiration	Key Length	Display	Primary Key
<input type="checkbox"/>	Go Daddy Secure Cert...	Go Daddy Root Certif...	May 3, 2011	May 3, 2031	2048		2
<input type="checkbox"/>	Go Daddy Root Certif...	Go Daddy Root Certif...	Aug 31, 2009	Dec 31, 2037	2048		3

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

TLS Profile

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

- TLS Protocol as "TLS 1.2 Only".
- Mutual Authentication "Enabled".
- Validate Server FQDN as "Disabled".
- Certificate as "SBC Edge Certificate".

Figure 11: Default TLS Profile

TLS Profile July 17, 2020 20:32:54

Total 1 TLS Profile Row

Description	Primary Key
Default TLS Profile	1

Description: Default TLS Profile

TLS Parameters

Common Attributes

TLS Protocol: TLS 1.2 Only
 Mutual Authentication: Enabled
 Handshake Inactivity Timeout: 30 secs [1..30]

Client Attributes

Client Cipher List:

- TLS_ECDHE_RSA_WITH_AES_256_CBC_SHA384
- TLS_ECDHE_RSA_WITH_AES_128_CBC_SHA256
- TLS_ECDHE_RSA_WITH_3DES_EDE_CBC_SHA
- TLS_RSA_WITH_AES_256_CBC_SHA256
- TLS_RSA_WITH_AES_128_CBC_SHA256
- TLS_RSA_WITH_AES256_CBC_SHA
- TLS_RSA_WITH_AES128_CBC_SHA

Validate Server FQDN: Disabled
 Certificate: SBC Edge Certificate

Server Attribute

Validate Client FQDN: Disabled
 Certificate: SBC Edge Certificate

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 12: Media Profile

Media Profiles July 15, 2020 17:57:08

Create Media Profile

Total 2 Media Profile Rows

Codec	Description	Primary Key
G.711 A-Law	Default G711A	1
G.711 μ-Law	Default G711u	2

Figure 13: G711-A law

Media Profiles
July 15, 2020 17:57:08

Create Media Profile
Total 2 Media Profile Rows

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

Voice Codec Configuration

Description

Codec

Payload Size
 ms

Apply

Figure 14: G711 Mu law

Media Profiles
July 15, 2020 17:57:08

Create Media Profile
Total 2 Media Profile Rows

	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

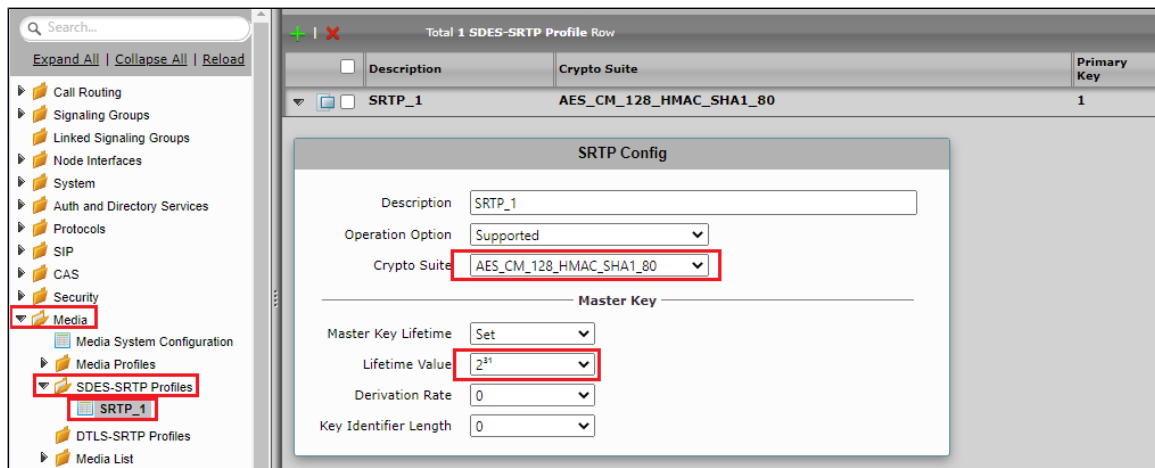
Apply

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_80".
- Set the LifeTime Value as shown in the diagram.

Figure 15: SDES-SRTP Profile

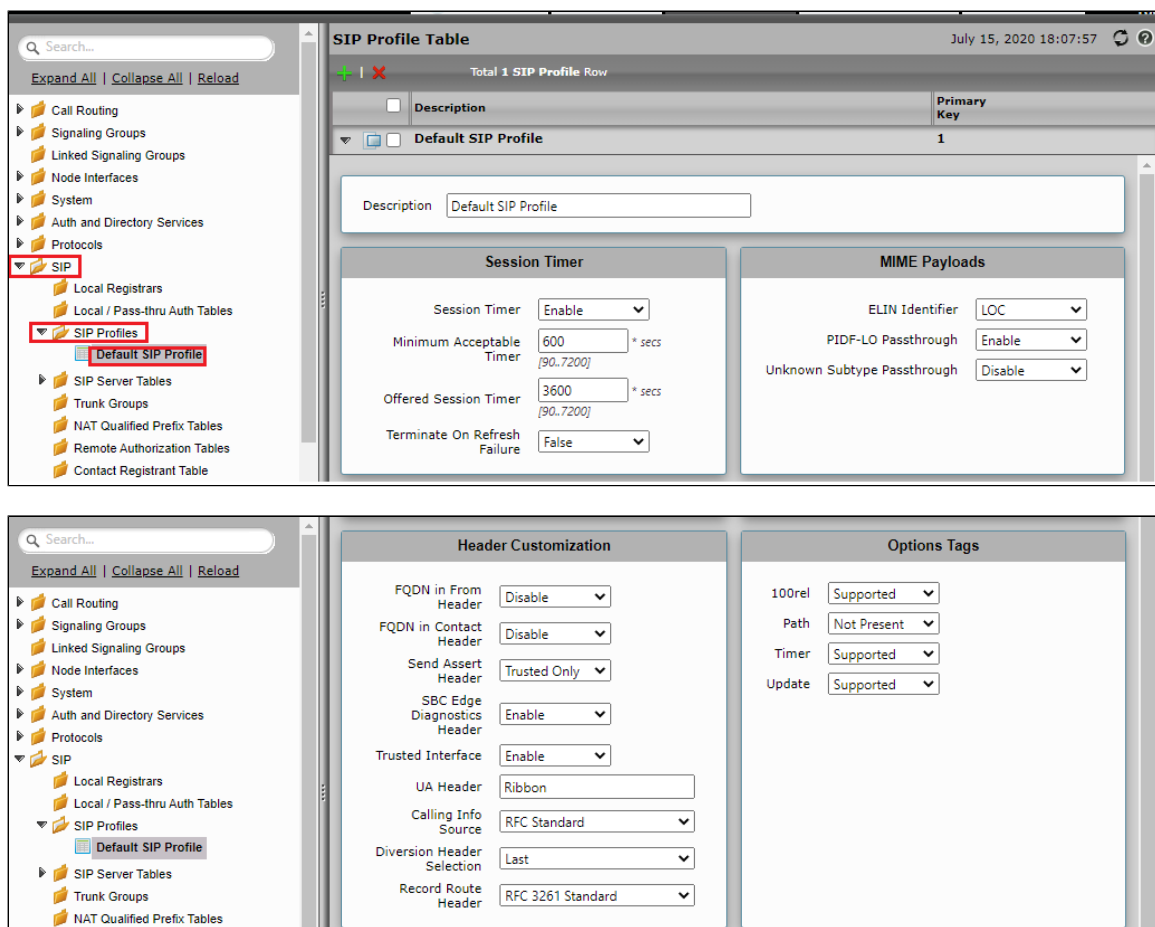


7. SIP Profile

SIP profile is used to modify the different sip parameters like Session timers, SIP Header Customization, SDP Customization. *Default SIP profile* has been used in the current test setup.

- Navigate to **SIP > SIP Profiles > Default SIP Profile**.

Figure 16: SIP profile



Timers

Transport Timeout Timer: 5000 ms [5000..32000]

Maximum Retransmissions: RFC Standard

Redundancy Retry Timer: 180000 ms [5000..180000]

RFC Timers

Timer T1: 500 ms [100..10000]

Timer T2: 4000 ms [1000..80000] (>= T1)

Timer T4: 5000 ms [1000..100000]

Timer D: 32000 ms [5000..640000]

Timer B: 32000 ms

Timer F: 32000 ms

Timer H: 32000 ms (64*TimerT1)

Timer J: 4000 ms [4000..640000]

SDP Customization

Send Number of Audio Channels: True

Connection Info in Media Section: True

Origin Field Username: SBC (default: SBC)

Session Name: VoipCall (default: VoipCall)

Digit Transmission Preference: RFC 2833/Voice

SDP Handling Preference: Legacy Audio/Fax

8. PSTN Leg Configuration

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

1. [Media List](#)
2. [SIP Server Tables](#)
3. [Signaling Group](#)
4. [Transformation](#)
5. [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 as defined in Media Profile.

- "Add/Edit" to add the different Media profile created earlier.
- Set RTCP mode to "RTCP".
- Set Silence Suppression to "disabled".

Figure 17: Media List

Media List View July 15, 2020 18:26:18

Total 2 Media List Rows

Description	Primary Key
Default Media List	1

Description: Default Media List

Media Profiles List: Default G711u, Default G711A

Up, Down, Add/Edit, Remove buttons

SDES-SRTP Profile: None (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: Disabled

Gain Control		Digit Relay	
Receive Gain	0 [-14..+6] dB	Digit (DTMF) Relay Type	RFC 2833
Transmit Gain	0 [-14..+6] dB	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. SIP Server Tables

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 PSTN IP (for example, 172.16.X.X in our case).
- Keep the default transport protocol, which is "UDP".

Figure 18: SIP

The screenshot displays the configuration interface for SIP Server Tables. On the left, a navigation pane shows a tree structure where 'SIP' is expanded, and 'SIP Server Tables' and 'IP_PBX' are highlighted with red boxes. The main panel, titled 'IP_PBX', shows a table with one row. The table has columns for Host / Domain, Server Lookup, Port, Protocol, Display Counters, Priority, and Primary Key. The row contains the values: 172.16.X.X, IP/FQDN, 5060, UDP, Counters, 1, and 1.

Host / Domain	Server Lookup	Port	Protocol	Display Counters	Priority	Primary Key
172.16.X.X	IP/FQDN	5060	UDP	Counters	1	1

SIP Channels and Routing

Action Set Table: None +

Call Routing Table: IP_PBX_RT +

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

SIP Profile: Default SIP Profile +

SIP Mode: Basic Call

Agent Type: Back-to-Back User Agent

Interop Mode: Standard

SIP Server Table: IP_PBX +

Load Balancing: Round Robin

Channel Hunting: Most Idle

Notify Lync CAC Profile: Disable

Media Information

Supported Audio/Fax Modes: DSP Proxy Direct Add/Edit Remove

Supported Video/Application Modes: Disabled

Media List ID: Default Media List +

Play Ringback: Auto on 180

Tone Table: Default Tone Table +

Play Congestion Tone: Disable

Early 183: Disable

Allow Refresh SDP: Enable

SIP IP Details

Signaling/Media Source IP: Ethernet 1 IP (10.54. .)

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 2 SIP Listen Port Rows

Port	Protocol	TLS Profile ID
5060	UDP	N/A
5060	TCP	N/A

Federated IP/FQDN

Total 1 SIP Federated IP Row

IP/FQDN	Netmask/Prefix
172.16. .	255.255.255.255

4. Transformation

Example:

A customer has an existing carrier that only accepts the U.S.A. domestic "10-digit" dial plan numbering format. For example: (XXX) YYY-ZZZZ. Where XXX=area code, YYY-ZZZZ=7-digit phone number. Zoom is using the E.164 numbering format: +(country code)(phone number). This creates a phone number format incompatibility issue between Zoom and the customer carrier. Zoom expects to receive calls in E.164 numbering format, while the customer carrier expects the USA 10-digit domestic numbering format. SBC Edge is introduced to solve the numbering interop issue between the two entities. SBC Edge inserts a "+1" for all U.S. phone numbers destined for Zoom, and removes "+1" for all U.S. phone numbers destined for customer carrier(s).



Note

Ribbon SBC Edge can be programmed for different country E.164 code mapping in addition to the U.S. dial plan.

"Add_plusOne" transformation rule is required for outgoing call towards Zoom.

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

Figure 20: Transformation

Search...

Expand All | Collapse All | Reload

Call Routing

Transformation

Add_plusOne

Passthrough Untouched

Remove_plusOne

Add_plusOne

July 16, 2020 18:47:10

Total 1 Transformation Entry Row

Admin State	Input Field Type	Input Field Value	Output Field Type	Output Field Value	Match Type	Description	Primary Key
On	Called Address/Number	(.*)	Called Address/Number	+1\1	Optional (Match One)	Add_plus_one	1

Description

Admin State

Match Type

Input Field

Type

Value

Output Field

Type

Value

5. 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 of which 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 "Enabled".
 - Media list as "Zoom_ML".



Tip

Attach the Media List and Destination Signaling Groups which 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
 - IP_PBX_RT
 - Zoom_RT

IP_PBX_RT
July 15, 2020 22:28:52

Display Counters
Total 1 Call Route Entry Row

Admin State	Priority	Transformation Table	Destination Type	First Signaling Group	Description	Fork Call	Primary Key
<input checked="" type="checkbox"/>	1	Add_plusOne	Normal	(SIP) Zoom_SG	IP_PBX	No	1

Admin State	Priority	Transformation Table	Destination Type	First Signaling Group	Description	Fork Call	Primary Key
<input type="checkbox"/>	1	Add_plusOne	Normal	(SIP) Zoom_SG	IP_PBX	No	1

Route Details

Description:

Admin State:

Route Priority:

Call Priority:

Number/Name Transformation Table: +

Time of Day Restriction: +

Destination Information

Destination Type:

Message Translation Table: +

Cause Code Reroutes: +

Cancel Others upon Forwarding:

Fork Call:

Destination Signaling Groups

Up

Down

Add/Edit *

Remove

Enable Maximum Call Duration:

Maximum Call Duration: [1..10080] min.

Media

Audio/Fax Stream Mode:

Video/Application Stream Mode:

Media Transcoding:

Media List: +

Quality of Service

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

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

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

Enable Min MOS Threshold:

Enable Max. R/T Delay:

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

Enable Max. Jitter:

Max. Jitter: ms [1..3000]

9. Zoom Leg Configuration

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

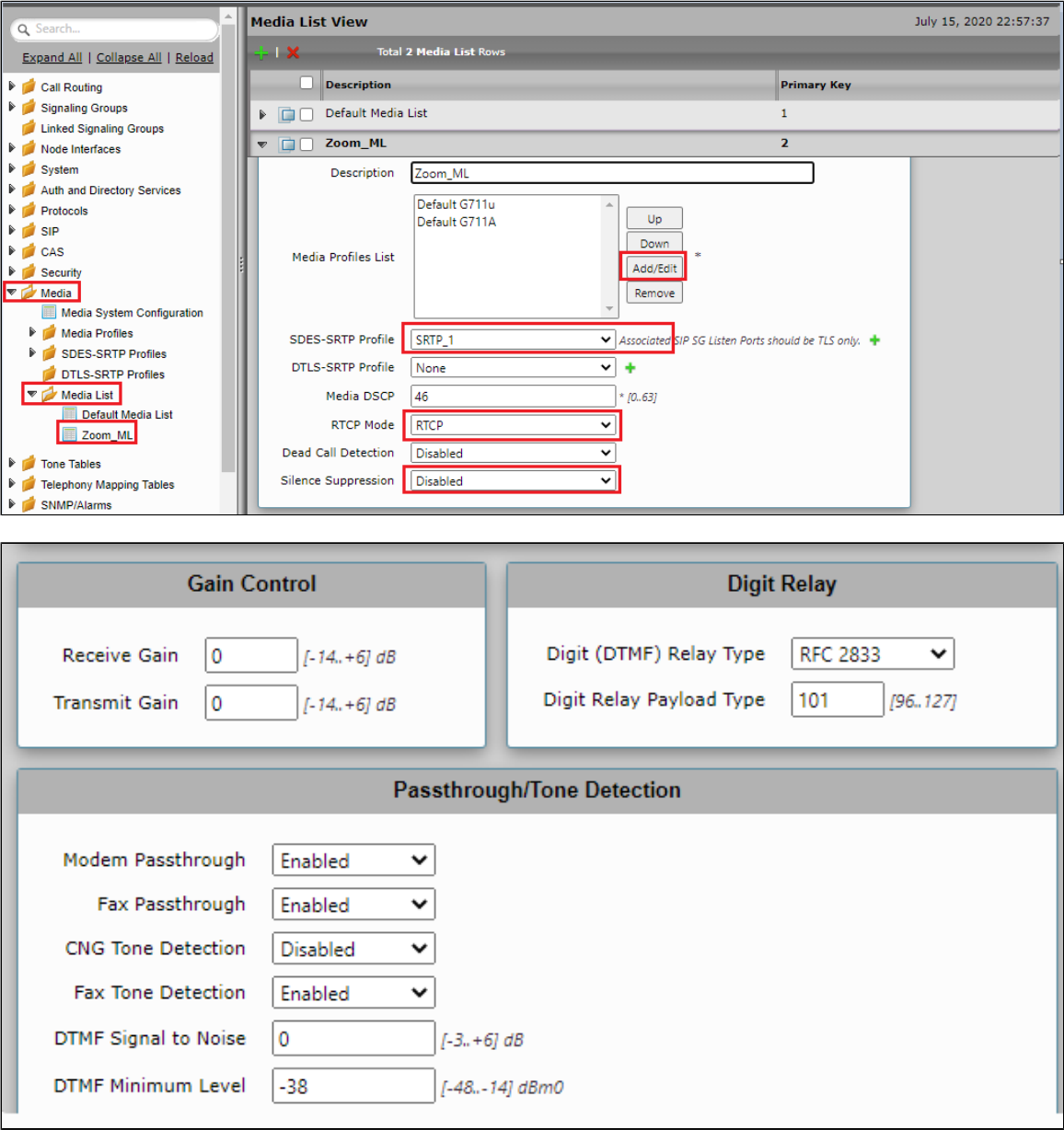
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3. [Signaling Group](#).
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1. Media List

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 "SRTP_1" as created earlier.
- Set RTCP mode to "RTCP".
- Set Silence Suppression to "disabled".

Figure 22: Media List

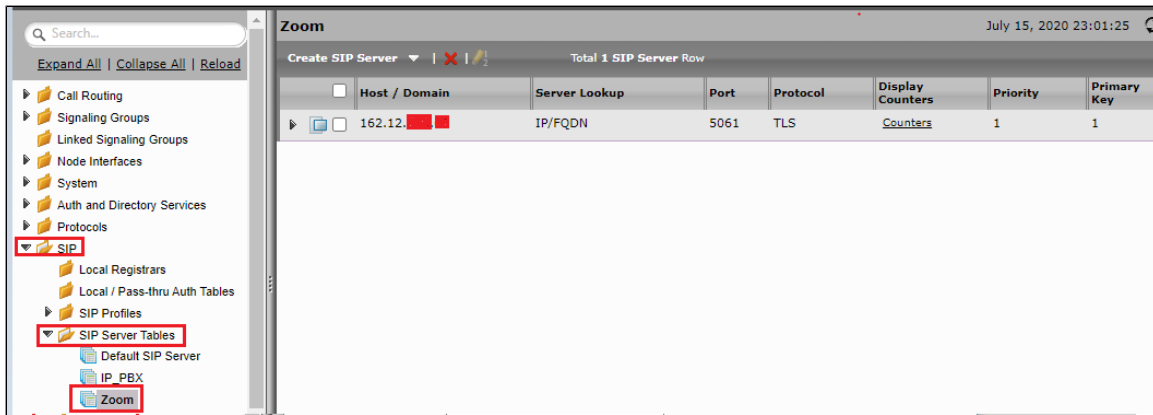


2. SIP Server Tables

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 IP (for example, 162.12.X.X in our case).
- Configure Transport protocol as "TLS".
- Set TLS Profile as "Default TLS Profile" as created in the section [TLS Profile](#).

Figure 23: SIP



Host / Domain	Server Lookup	Port	Protocol	Display Counters	Priority	Primary Key
162.12.X.X	IP/FQDN	5061	TLS	Counters	1	1

Server Host

Server Lookup: IP/FQDN

Priority: 1

Host FQDN/IP: 162.12.X.X

Port: 5061

Protocol: TLS

TLS Profile: Default TLS Profile

Transport

Monitor: SIP Options

Keep Alive Frequency: 30 * secs [30..300]

Recover Frequency: 5 * secs [5..300]

Local Username: SBC1K * Local Username of SBC Edge

Peer Username: SBC1K * Peer Username of sip server

Remote Authorization and Contacts

Remote Authorization Table: None

Contact Registrant Table: None

Session URI Validation: Liberal

Connection Reuse

Reuse: True

Sockets: 4

Reuse Timeout: Forever

3. Signaling Groups

Signaling groups allow telephony channels to be grouped together for the purposes of routing and shared configuration. They are the entity to which calls are routed, as well as the location from which Call Routes are selected. They are also the location from which 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 "Default SIP Profile".
- 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 Call Routing table as "Zoom_RT" as created in the [Call Routing Table](#) section.

Figure 24: Signalling Groups

Search...

Expand All | Collapse All | Reload

Call Routing

Signaling Groups

(SIP) IP_PBX_SG

(SIP) Zoom_SG

Linked Signaling Groups

Node Interfaces

System

Auth and Directory Services

Protocols

SIP

Signaling Group Table

July 16, 2020 10:41:13

Create Signaling Group

Total 2 Signaling Group Rows

Type	Description	Admin State	Service Status	Display	Primary Key
SIP	IP_PBX_SG	Up	Up	Counters Channels Sessions	1
SIP	Zoom_SG	Up	Up	Counters Channels Sessions	2

Description: Zoom_SG

Admin State: Enabled

Service Status: Up

SIP Channels and Routing

Action Set Table: None

Call Routing Table: Zoom_RT

No. of Channels: 60

SIP Profile: Default SIP Profile

SIP Mode: Basic Call

Agent Type: Back-to-Back User Agent

Interop Mode: Standard

SIP Server Table: Zoom

Load Balancing: Round Robin

Channel Hunting: Most Idle

Notify Lync CAC Profile: Disable

Media Information

Supported Audio/Fax Modes: DSP, Proxy, Direct

Supported Video/Application Modes: Disabled

Media List ID: Zoom_ML

Play Ringback: Auto on 180

Tone Table: Default Tone Table

Play Congestion Tone: Disable

Early 183: Disable

Allow Refresh SDP: Enable

SIP IP Details

Signaling/Media Source IP: Ethernet 2 IP (115.110. . .)

Signaling DSCP: 40

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
5061	TLS	Default TLS Profile

Federated IP/FQDN

Total 1 SIP Federated IP Row

IP/FQDN	Netmask/Prefix
162.12. . .	255.255.255.0

4. Transformation

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

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

Figure 25: Transformation

Search...

Expand All | Collapse All | Reload

Call Routing

Transformation

Add_plusOne

Passthrough Untouched

Remove_plusOne

Remove_plusOne

July 16, 2020 18:54:04

Total 1 Transformation Entry Row

Admin State	Input Field Type	Input Field Value	Output Field Type	Output Field Value	Match Type	Description	Primary Key
Up	Called Address/Number	\+1(.*)	Called Address/Number	\1	Optional (Match One)	Remove_plusOne	1

Description

Admin State

Match Type

Input Field

Type

Value

Output Field

Type

Value

5. 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 of which 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 which were created earlier in the [PSTN Leg Configuration](#) section.

Figure 26: Call Routing Table

Search...

Expand All | Collapse All | Reload

Call Routing

Transformation

Time of Day Table

Call Routing Table

Default Route Table

IP_PBX_RT

Zoom_RT

Zoom_RT

July 16, 2020 11:36:56

✓

✗

+

−

↺

↻

⚙

Display Counters

Total 1 Call Route Entry Row

	Admin State	Priority	Transformation Table	Destination Type	First Signaling Group	Description	Fork Call	Primary Key
	<input checked="" type="checkbox"/>	1	Remove_plusOne	Normal	(SIP) IP_PBX_SG	mstsZoom	No	1

Admin State	Priority	Transformation Table	Destination Type	First Signaling Group	Description	Fork Call	Primary Key
<input checked="" type="checkbox"/>	1	Remove_plusOne	Normal	(SIP) IP_PBX_SG	mstsZoom	No	1

Route Details

Description:

Admin State:

Route Priority:

Call Priority:

Number/Name Transformation Table:

Time of Day Restriction:

Destination Information

Destination Type:

Message Translation Table:

Cause Code Reroutes:

Cancel Others upon Forwarding:

Fork Call:

Destination Signaling Groups

Up

Down

Add/Edit

Remove

Enable Maximum Call Duration:

Maximum Call Duration: [1..10080] min.

Media

Audio/Fax Stream Mode:

Video/Application Stream Mode:

Media Transcoding:

Media List:

Quality of Service

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

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

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

Enable Min MOS Threshold:

Enable Max. R/T Delay:

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

Enable Max. Jitter:

Max. Jitter: ms [1..3000]

Section B: Zoom Web BYOC Configuration

Prerequisites:

- Zoom Go BYOC account: A special type of Zoom account that has outbound/inbound SIP trunk that peers between the Zoom Phone Cloud and the customer's PSTN carrier connection.
- Customer's existing carrier/carrier equipment: Any carrier offering PSTN services. Carrier equipment can be router/gateway or another SBC that supports SIP trunk connectivity. Carrier has provided several DID's to use as external BYOC numbers.
- Trunk Registration: BYOC is a "static" trunk between 2 static IP endpoints, therefore no trunk registration is done here.



Note

Ensure a Zoom BYOC SIP trunk is built between Zoom SBC and Ribbon SBC Edge deployed on a customer site.

Once the Zoom Go account is available, Login to Zoom Web BYOC portal at <https://go.zoom.us/>.

The following Zoom BYOC configurations are included in this section:

- [Add External Number](#)
- [Create Zoom Users](#)
- [Supplementary services configuration on Zoom](#)

Add External Number

Navigate to **Phone Systems Management > Phone Numbers > External**.

Select **Add** to add external phone numbers provided by your carrier into the Zoom portal. These numbers are the DID numbers provided by your carrier.

Figure 27: Add External Number

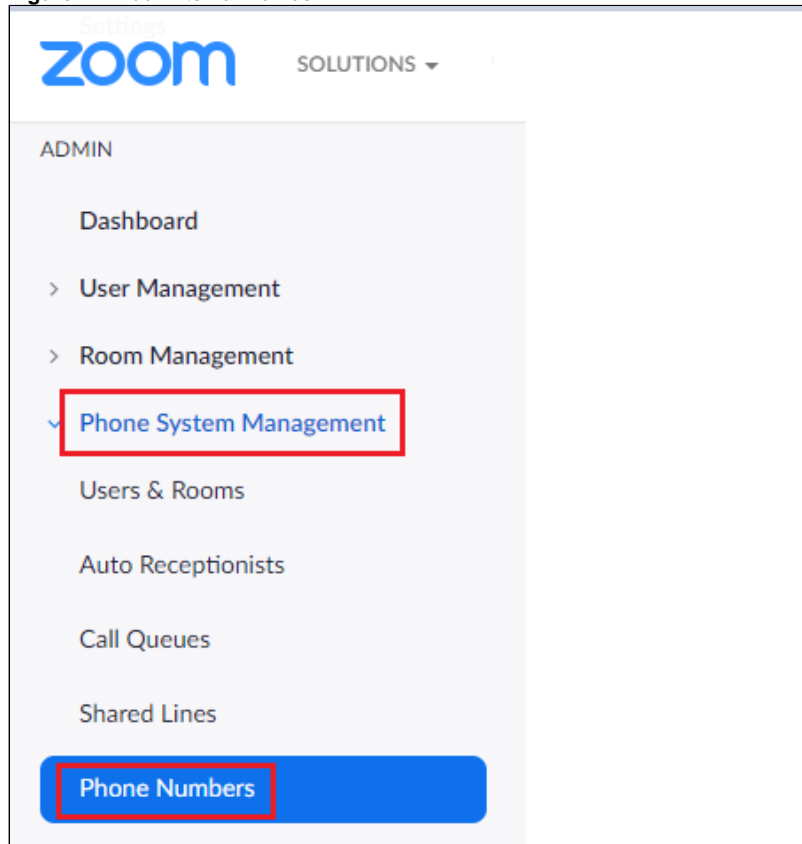
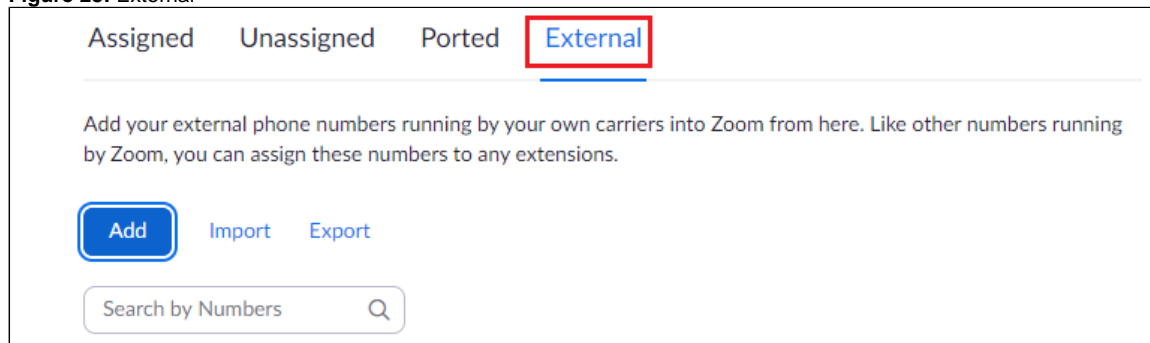


Figure 28: External



1. Select **BYOC** as the carrier.
2. Enter the existing customer phone numbers (from carrier) separated by commas.
3. Click **Submit**.

Figure 29: Add External Number

Add External Numbers

Carrier BYOC ▼

Numbers

+15125671233

Example: +19991234567, +19991234568

Cancel
Submit

Check the external numbers have been created successfully as shown below.

Figure 30: External Number created successfully

Assigned Unassigned Ported External

Add your external phone numbers running by your own carriers into Zoom from here. Like other numbers running by Zoom, you can assign these numbers to any extensions.

Add Import Export

Search by Numbers

Number Type (All)

Number	Number Type	Carrier	Country	Submission Date
(512) 567-1233	Toll Number	BYOC	United States	May 8, 2020, 12:05 AM

Create Zoom Users

Zoom Users are created in order to login to Zoom clients on desktop or mobile. The steps for creating a user are as follows:

1. Navigate to **User Management > Users**. Click **Add** to create new Zoom users.
2. Navigate to **Phone System Management > Users & Rooms**. Check that the User status is **"Active"**.
3. Navigate to **Assign Calling Plan > Assign BYOC Calling Plan**. Click **"Confirm and Assign Numbers"**.

Figure 31: Create Zoom User

[SOLUTIONS](#)
[PLANS & PRICING](#)
[CONTACT SALES](#)

[SCHEDULE A MEETING](#)
[JOIN A MEETING](#)
[HOST A MEETING](#)

Phone

Recordings

Settings

ADMIN

Dashboard

User Management

Room Management

Phone System Management

Users & Rooms

Add

Import

Export

Assign Numbers

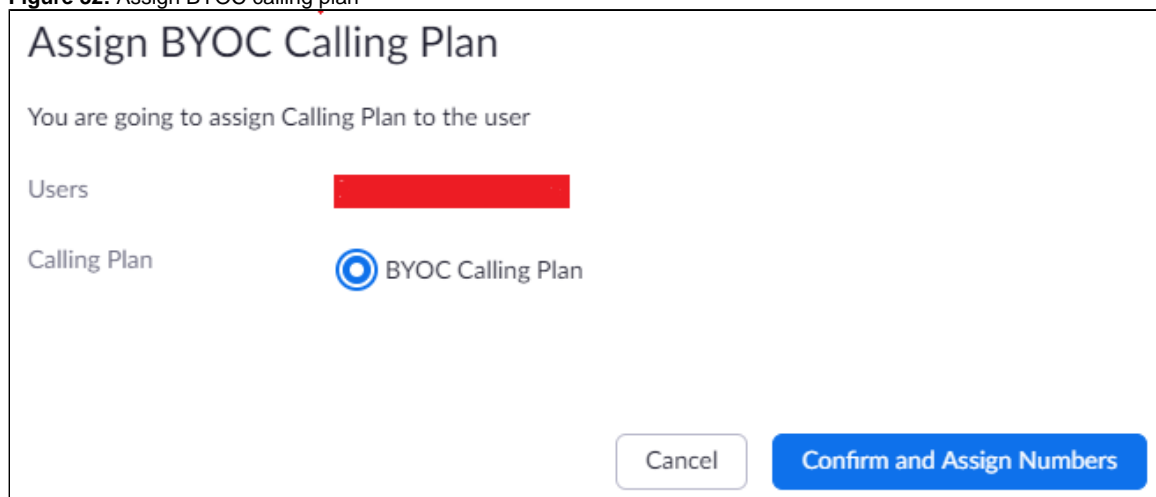
Assign Calling Plan

Apply Settings

Remove

<input type="checkbox"/>	Name	Ext.	Calling Plan(s)	Number(s)	Desk Phone(s)	User Status
<input type="checkbox"/>	[Redacted]	805	--	--	--	Active
<input type="checkbox"/>	[Redacted]					<div>Assign Calling Plan</div>

Figure 32: Assign BYOC calling plan



The dialog box is titled "Assign BYOC Calling Plan". Below the title, it says "You are going to assign Calling Plan to the user". There are two fields: "Users" with a red rectangular placeholder, and "Calling Plan" with a blue circular icon and the text "BYOC Calling Plan". At the bottom right, there are two buttons: "Cancel" and "Confirm and Assign Numbers".

Assign BYOC Calling Plan

You are going to assign Calling Plan to the user

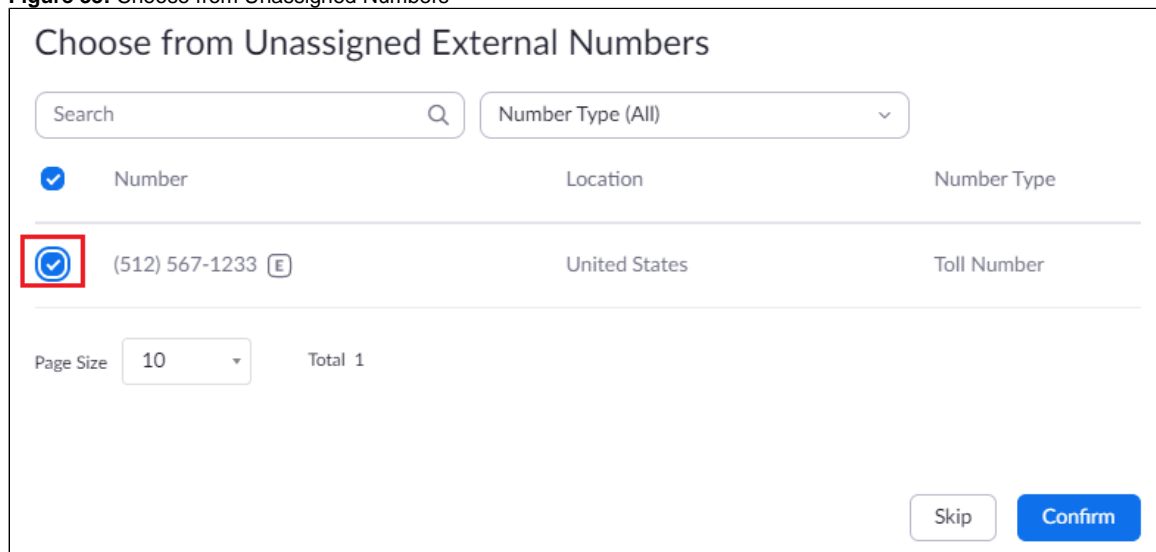
Users [REDACTED]

Calling Plan ● BYOC Calling Plan

Cancel Confirm and Assign Numbers

4. Assign the External Numbers created previously in the Add External Number section.

Figure 33: Choose from Unassigned Numbers



The dialog box is titled "Choose from Unassigned External Numbers". It has a search bar and a dropdown menu for "Number Type (All)". Below is a table with columns: "Number", "Location", and "Number Type". The first row shows a checked checkbox, the number "(512) 567-1233" with an "E" icon, "United States", and "Toll Number". At the bottom, there is a "Page Size" dropdown set to "10", "Total 1", and two buttons: "Skip" and "Confirm".

Choose from Unassigned External Numbers

Search Q Number Type (All) ▼

<input checked="" type="checkbox"/>	Number	Location	Number Type
<input checked="" type="checkbox"/>	(512) 567-1233 E	United States	Toll Number

Page Size 10 ▼ Total 1

Skip Confirm

5. Click **Confirm** to finish. Once the User is assigned with a Calling Plan and Number, it should look like the following example:

Figure 34: Configured User

Add
Import
Export

Search by Name, Ext. or Number
Plan (All)

Assign Numbers
Assign Calling Plan
Apply Settings
Remove

	Name	Ext.	Calling Plan(s)	Number(s)	Desk Phone(s)	User Status
<input type="checkbox"/>	[REDACTED]	805	BYOC	(512) 567-1233 E	--	Active

Supplementary Services Configuration on Zoom

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

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