
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

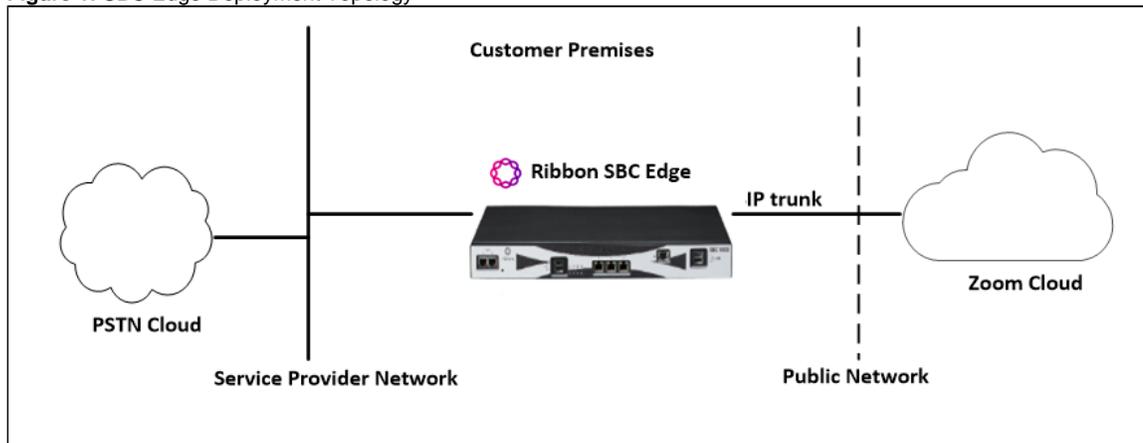
Note
 Ribbon SBC Edge portfolio includes SBC 1000, SBC 2000 (both are appliance based) and SBC SWeLite (virtualized platform). Software Version is applicable to Ribbon SBC Edge portfolio (1000, 2000, SWeLite) and hence this configuration guide is valid for all these devices.

Network Topology Diagram

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

SBC Edge Deployment Topology

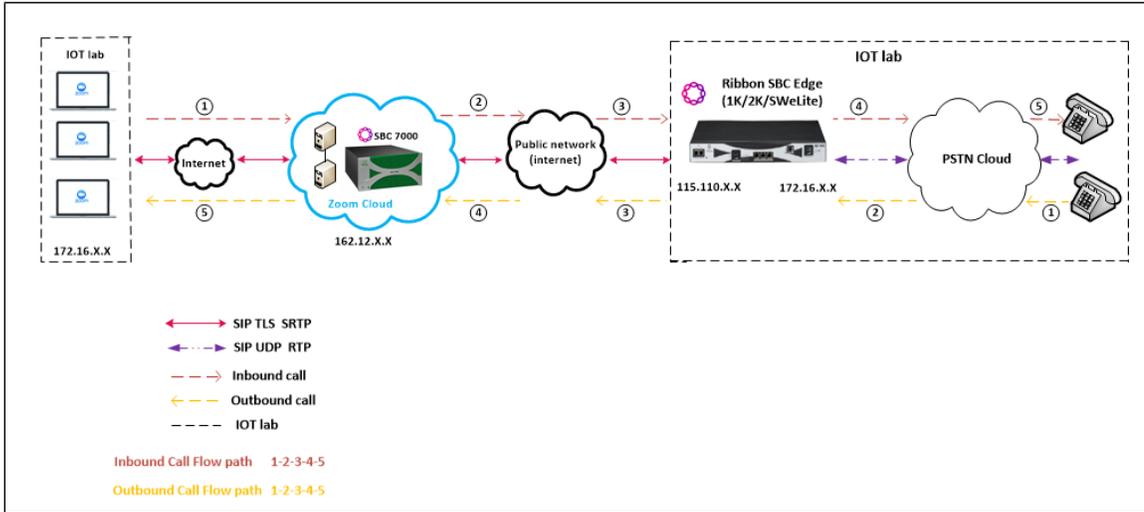
Figure 1: SBC Edge Deployment Topology



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

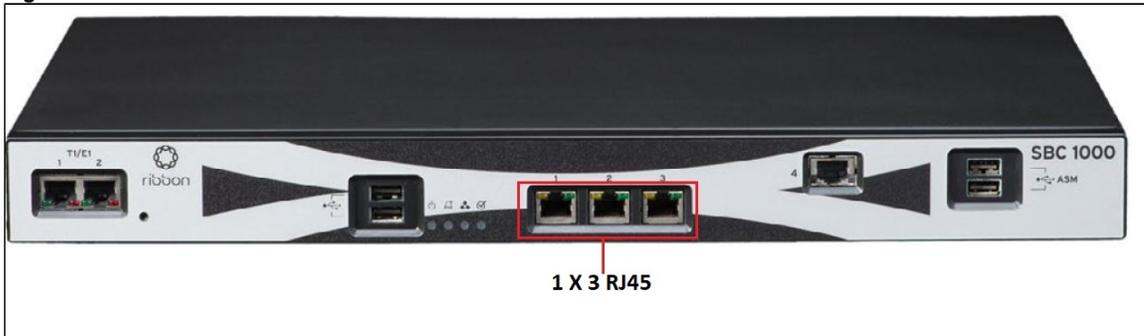
The following SBC Edge configurations are included in this section:

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. . .		Private Interface	Enabled	Counters	14040
Ethernet 2 IP	115.110. . .		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

IPv4 Information

ACL In	None
<input type="checkbox"/> ACL Out	None
ACL Forward	None
IP Assign Method	Static
Primary Address	10.54.
Primary Netmask	255.255.255.0
Configure Secondary Interface	Disabled

Figure 6: Ethernet 2

Ethernet 2 IP 115.110.
Public Interface Enabled [Counters](#) 14041

Identification/Status

Interface Name	Ethernet 2 IP
I/F Index	3
Alias	Public Interface
Description	Public Interface
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.
Primary Netmask	255.255.255.192
Configure Secondary Interface	Disabled

Tip
To configure Ethernet 1 and Ethernet 2 of SBC SWeLite, Navigate to **Networking Interfaces > Logical Interfaces**.

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.X.X	255.255.255.255	10.54.X.X	1	5
6	162.12.X.X	255.255.255.255	115.110.X.X	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

```

-----BEGIN CERTIFICATE REQUEST-----
MIIDCzCCAfmCAQAwfjEmMCQGA1UEAxQdKi5jdXN0b21lcnMuaW50ZXJvcGRvbWVp
bi5jb20xHTAbBgkqhkiG9w0BCQEWLnVzZXlxcXhJiYm4uY29tMQswCQYDVQQGEwJV
UzELMAkGA1UECBMCTG9uZjEAMBgNVBAoTBTBvbnVzMQswCQYDVQQLLwJVDCCASlW
DQYJKoZIhvcNAQEBBQADggEPADCCAQoCggEBAP1m1uHXRgBkKsGLGeOPwKFNOLuwi
FOgv0AugqrefvK5+Ru938w5OyrRsZZ5KN58vS/BI7tkqvZeqFZTEToUq23qvMADO
2OxJkZQzghZ5dk39On1THemRYa7tdBtmyyD1F8XRFPEUaANOFtrLzyMPvFnJuls
sTNmjA76/i3Qg+80kY0X2266uoTzs2puNEOIKppqZ6yxWngEyp50BDgZUKx53U6Yy
OyJNILpXTUYeDMwDtsICM0j3YdV6KbcA/Z6ZMLHvis3B34q8c4gm0wEjwVLbknd4
t/gub6+ZQPXVphgg3W6E8GUFVyzC6b36oHhCS6NJVT6qkNMKnKxRhkflBUCAwEA
AaBIMEYGCSqGSib3DQEJDJE5MDcwCQYDVR0TBAlwADALBgNVHQ8EBAMCBaAwHQYD
VR0IBBYwFAYIKwYBBQUHAWEGCCsGAQUFBwMCMAOGCSqGSib3DQEBCwUAA4IBAQD0
f0b+nhanA06rQxrjoGffcpPdjlCFt3SQQAcb7eR49BpSjzVINfO38IPmJgvYD8
w/h2JTFLExyzbkPKTIVdKaHb920ZgrGta5JYFaOyXf9mHBrZhCIMZc6qhv+58H9T
1K1r3wUelyR5e2PwKPP03LyFNvP4PbNc3XA0zh53mhZEqs9EEcRP+J3rxvVoaFUa

```

Copy CSR

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

The screenshot shows the 'SBC Primary Certificate' validation interface. At the top, there are 'Import' and 'Export' buttons and a timestamp 'July 16, 2020 15:54:32'. The interface is divided into three main sections: 'Subject', 'Issuer', and 'Certificate'.

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

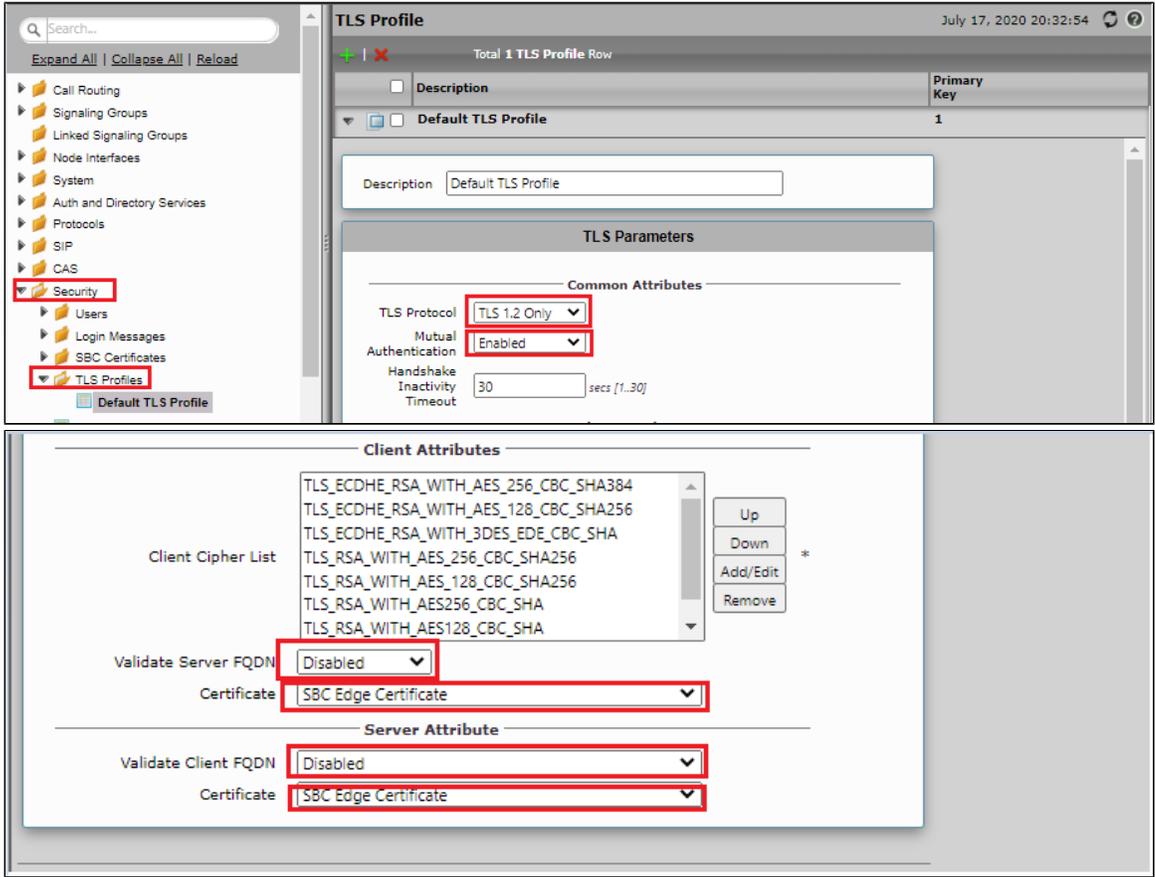
TLS Profile is required for the TLS handshake between SBC Edge and Zoom. This profile defines cipher suites supported by SBC Edge.

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



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

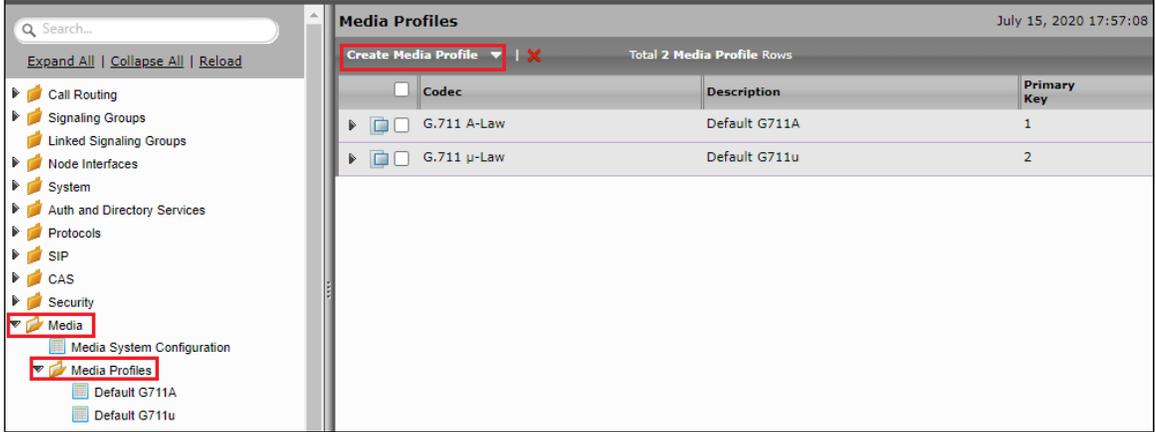


Figure 13: G711-A law

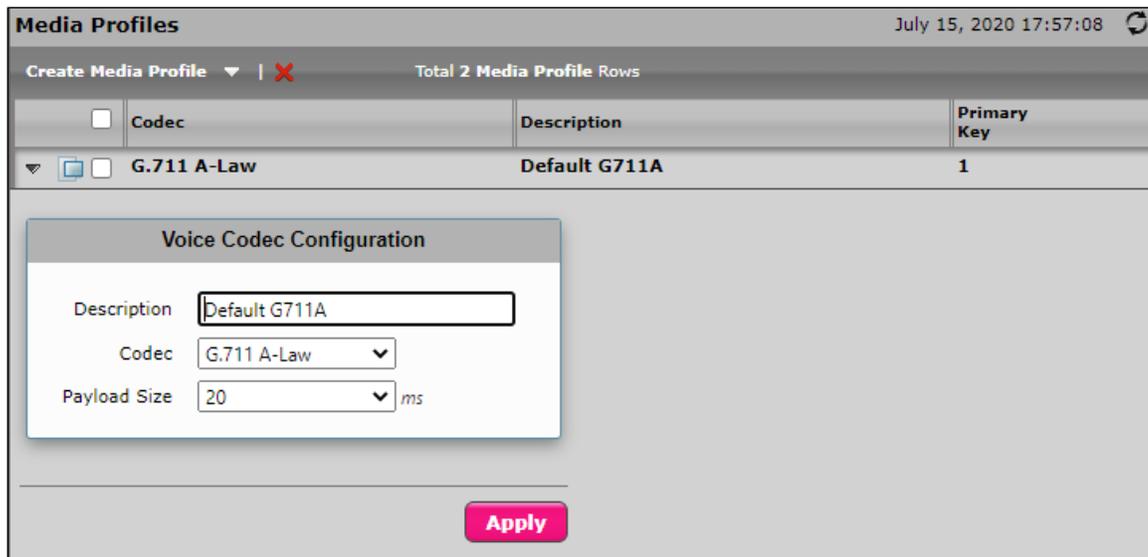
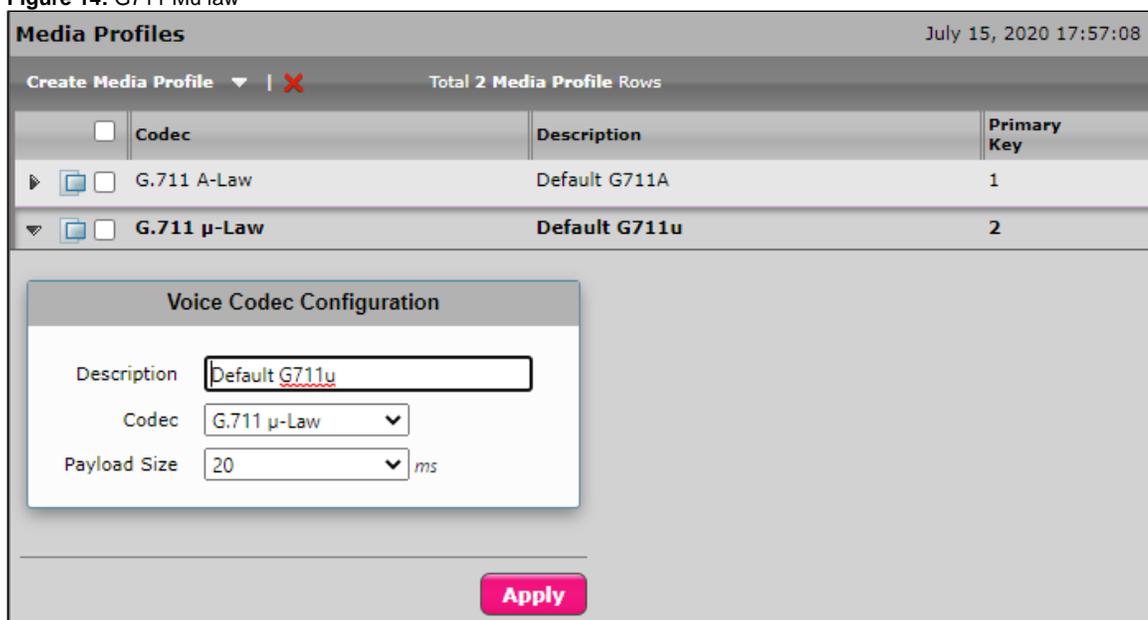


Figure 14: G711 Mu law

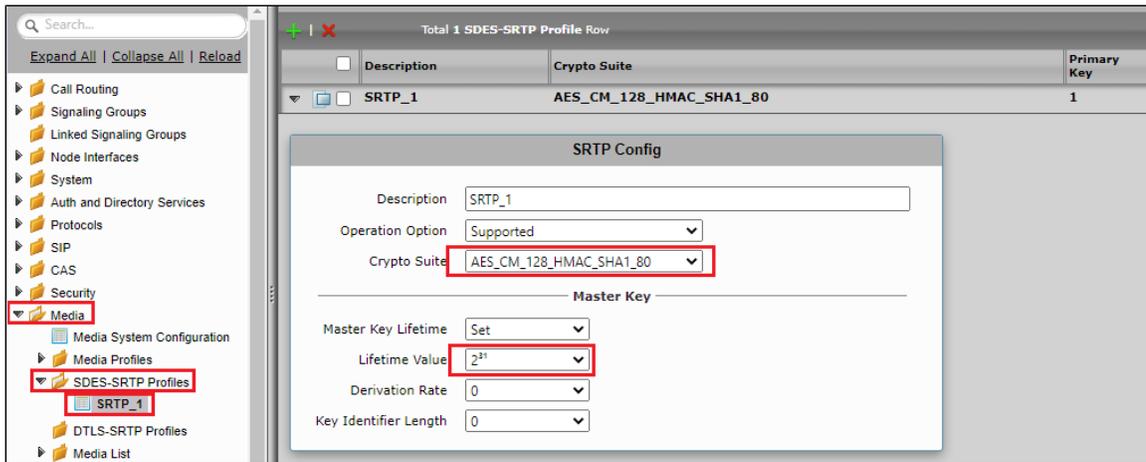


6. SRTP Profile

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

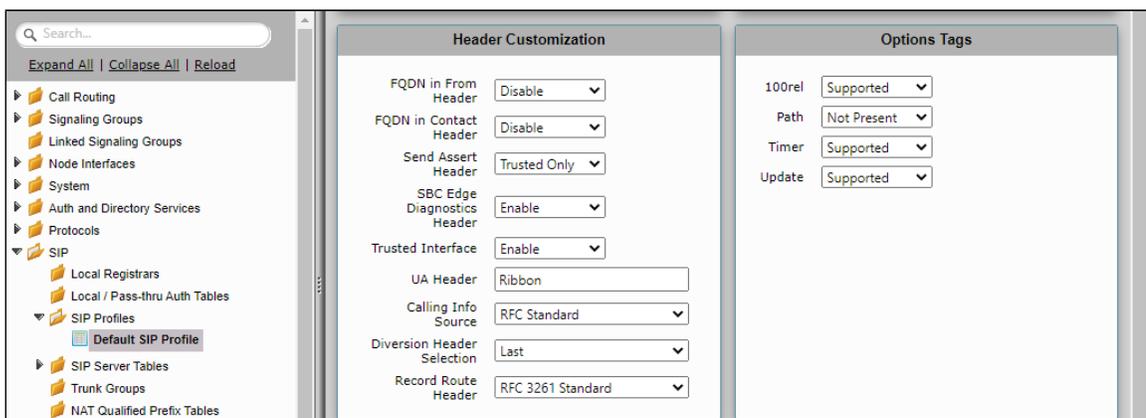
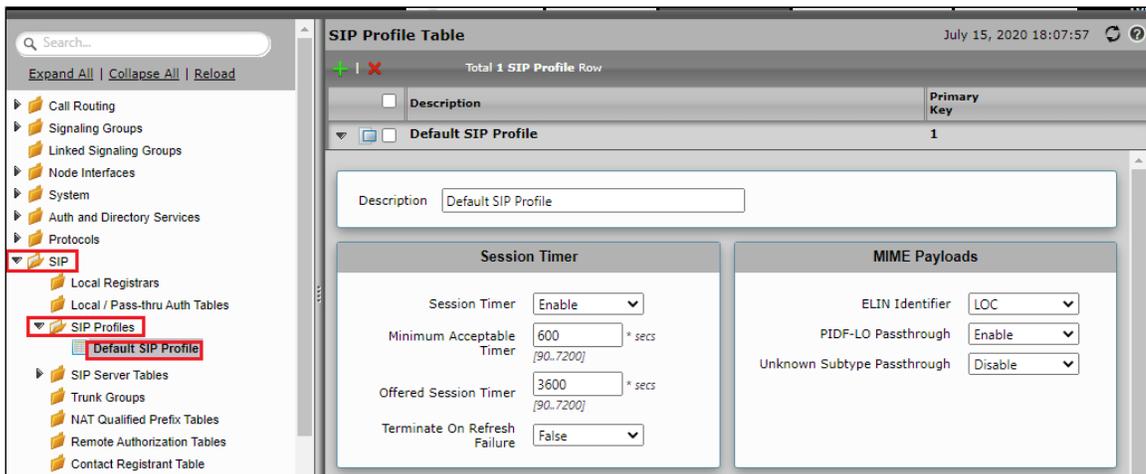


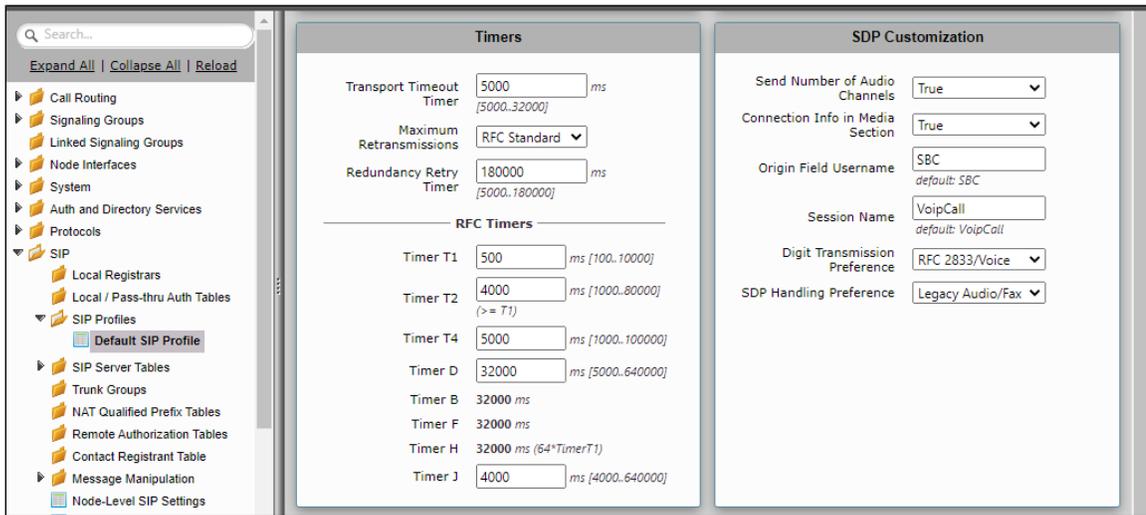
7. SIP Profile

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





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:

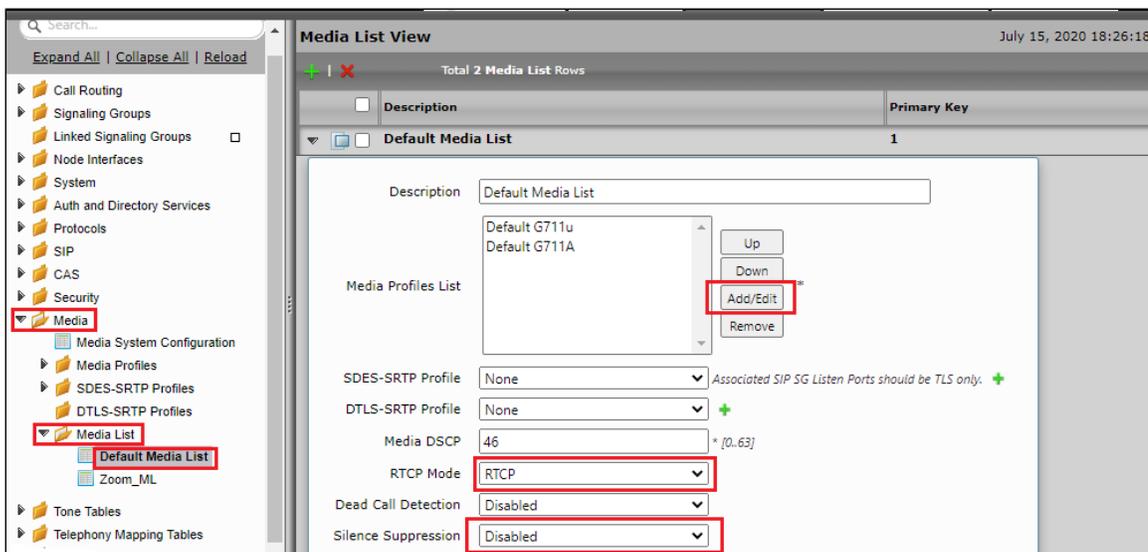
1. [Media List](#)
2. [SIP Server Tables](#)
3. [Signaling Group](#)
4. [Transformation](#)
5. [Call Routing Table](#)

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



Gain Control		Digit Relay	
Receive Gain	<input type="text" value="0"/> [-14..+6] dB	Digit (DTMF) Relay Type	<input type="text" value="RFC 2833"/>
Transmit Gain	<input type="text" value="0"/> [-14..+6] dB	Digit Relay Payload Type	<input type="text" value="101"/> [96..127]
Passthrough/Tone Detection			
Modem Passthrough	<input type="text" value="Enabled"/>		
Fax Passthrough	<input type="text" value="Enabled"/>		
CNG Tone Detection	<input type="text" value="Disabled"/>		
Fax Tone Detection	<input type="text" value="Enabled"/>		
DTMF Signal to Noise	<input type="text" value="0"/> [-3..+6] dB		
DTMF Minimum Level	<input type="text" value="-38"/> [-48..-14] dBm0		

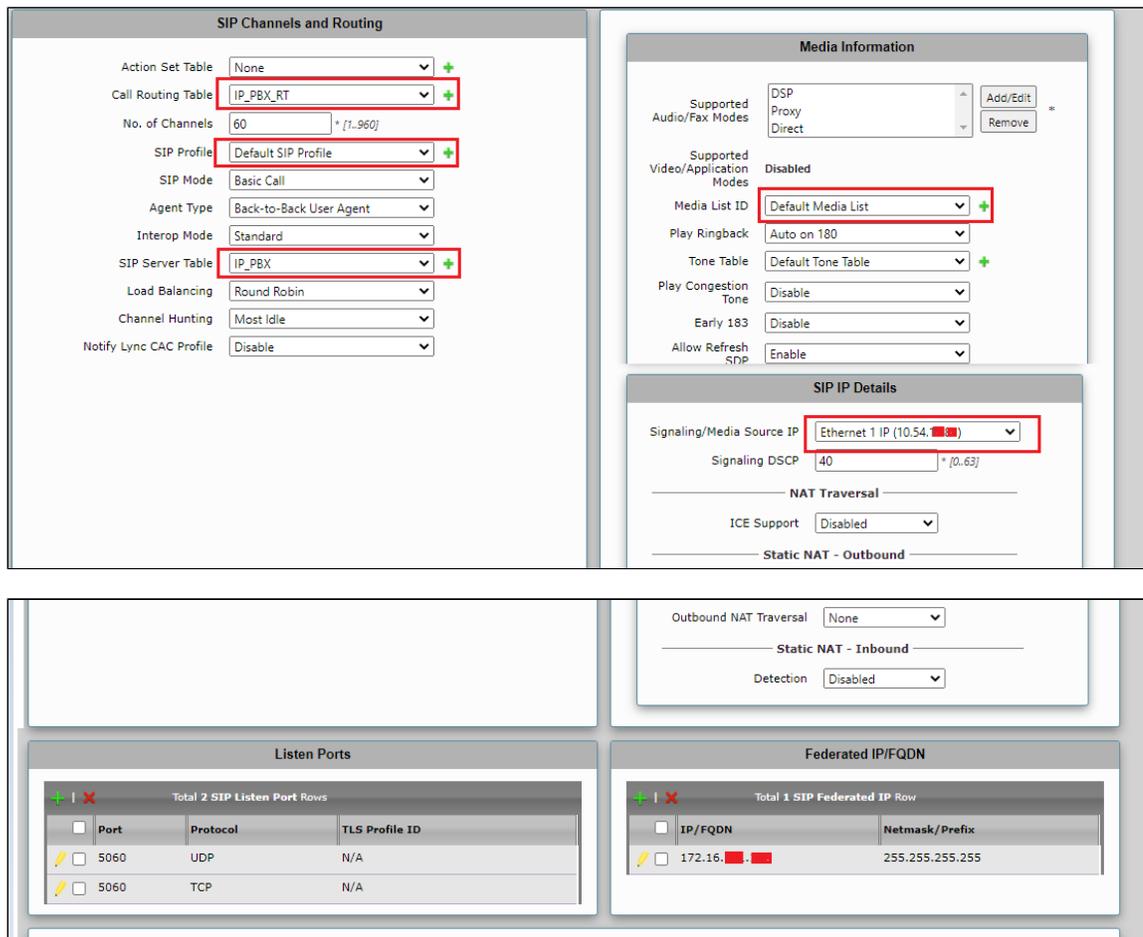
2. SIP Server Tables

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

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



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).



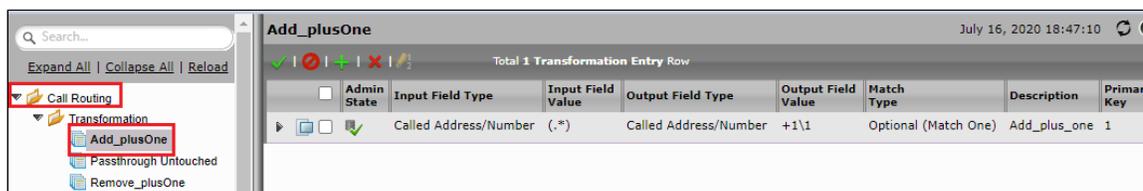
Note

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



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

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:

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:

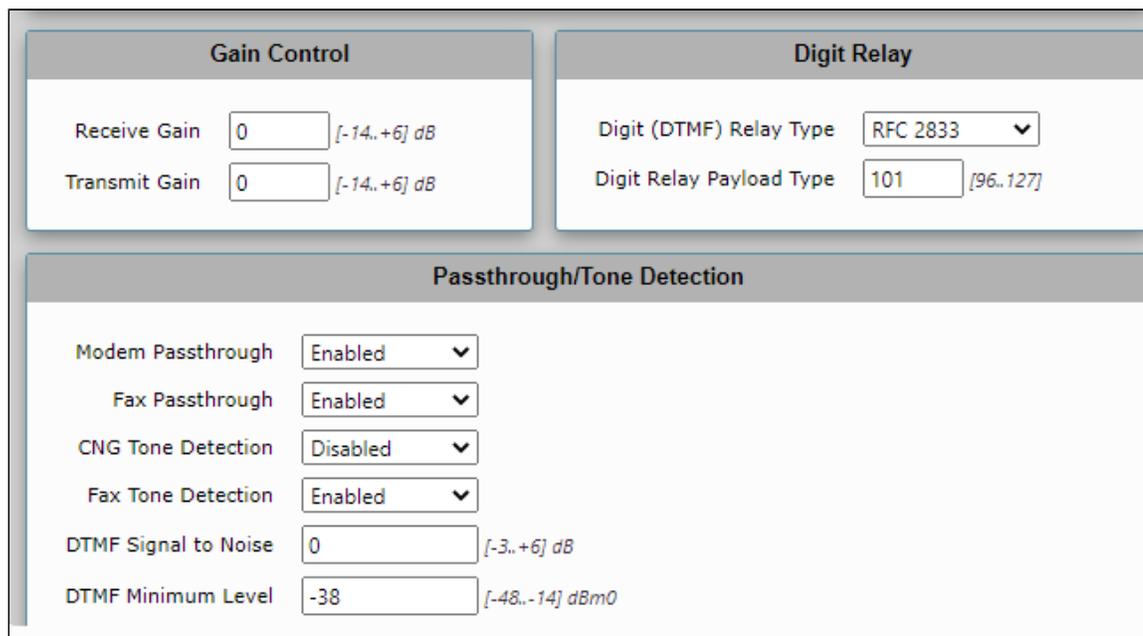
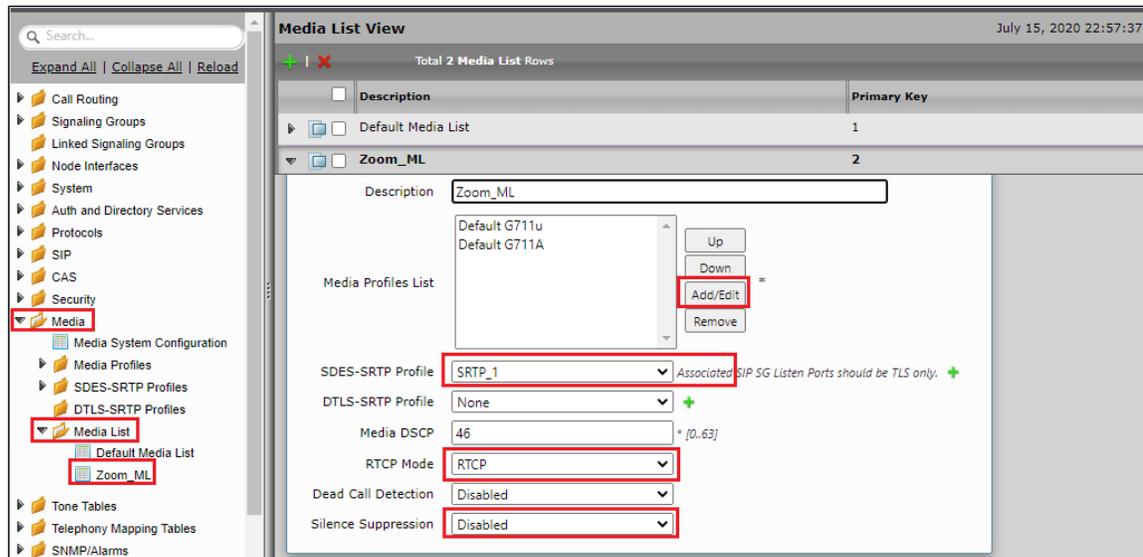
1. [Media List](#).
2. [SIP Server Tables](#).
3. [Signaling Group](#).
4. [Transformation](#).
5. [Call Routing Table](#).

1. Media List

Media List allows you to specify a set of codecs used for the call. They contain a list of codecs, 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

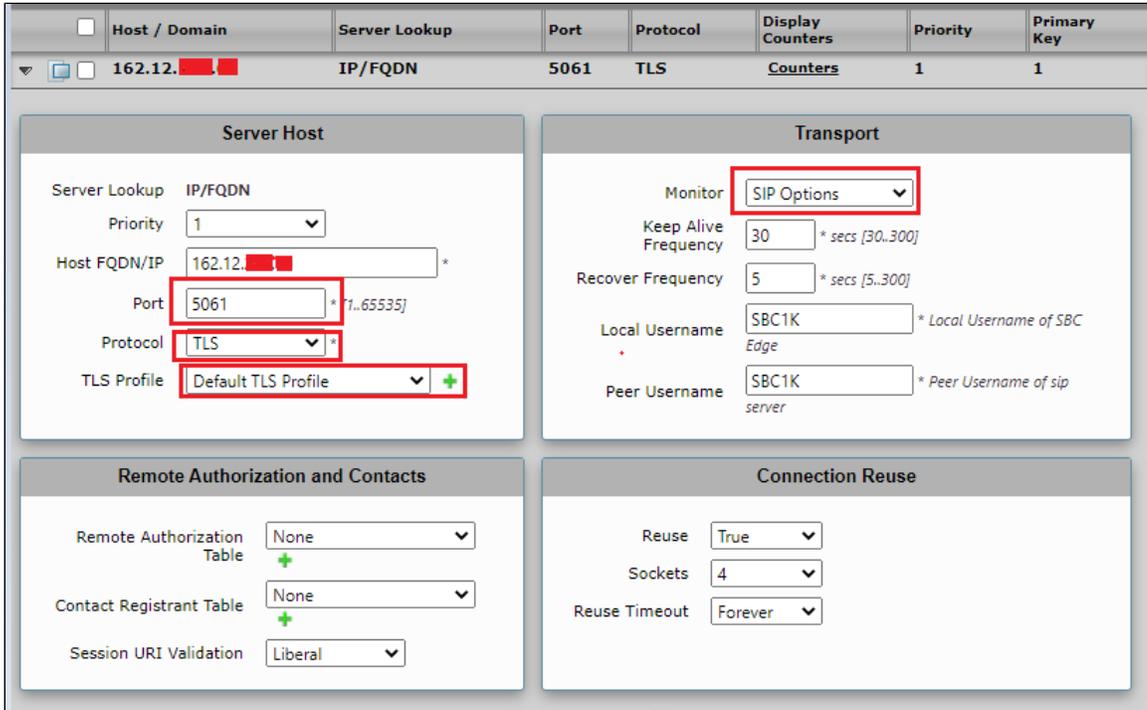
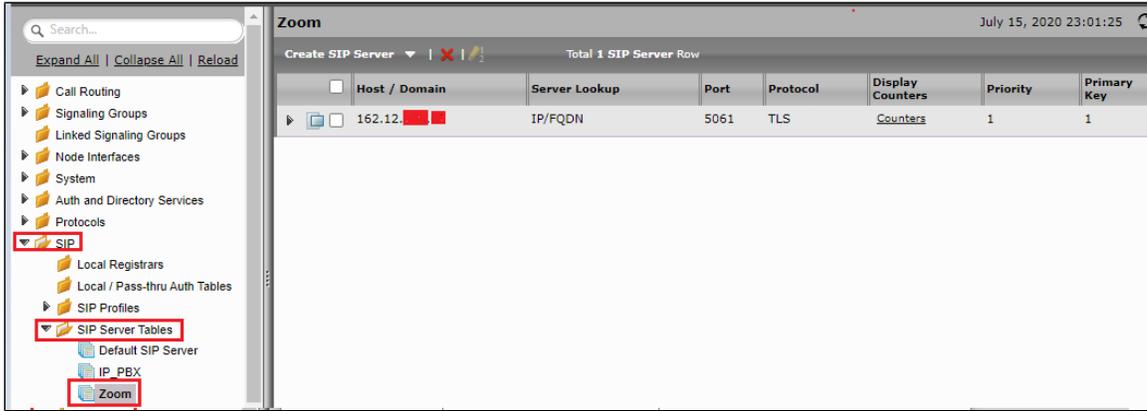


2. SIP Server Tables

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



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

Signaling Group Table July 16, 2020 10:41:13

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

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 * [1..960]

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 * [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
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

Remove_plusOne July 16, 2020 18:54:04

Expand All | Collapse All | Reload

Call Routing

- Transformation
 - Add_plusOne
 - Passthrough Untouched
 - Remove_plusOne

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

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 type="checkbox"/>	1	Remove_plusOne	Normal	(SIP) IP_PBX_SG	mstsZoom	No	1

Route Details

Description: mstsZoom

Admin State: Enabled

Route Priority: 1

Call Priority: Normal

Number/Name Transformation Table: Remove_plusOne

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) IP_PBX_SG

Buttons: Up, Down, Add/Edit, Remove

Enable Maximum Call Duration: Enabled

Maximum Call Duration: 1440 [1..10080] min.

Media

Audio/Fax Stream Mode: DSP

Video/Application Stream Mode: Disabled

Media Transcoding: Enabled

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]

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:

1. [Add External Number](#)
2. [Create Zoom Users](#)
3. [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

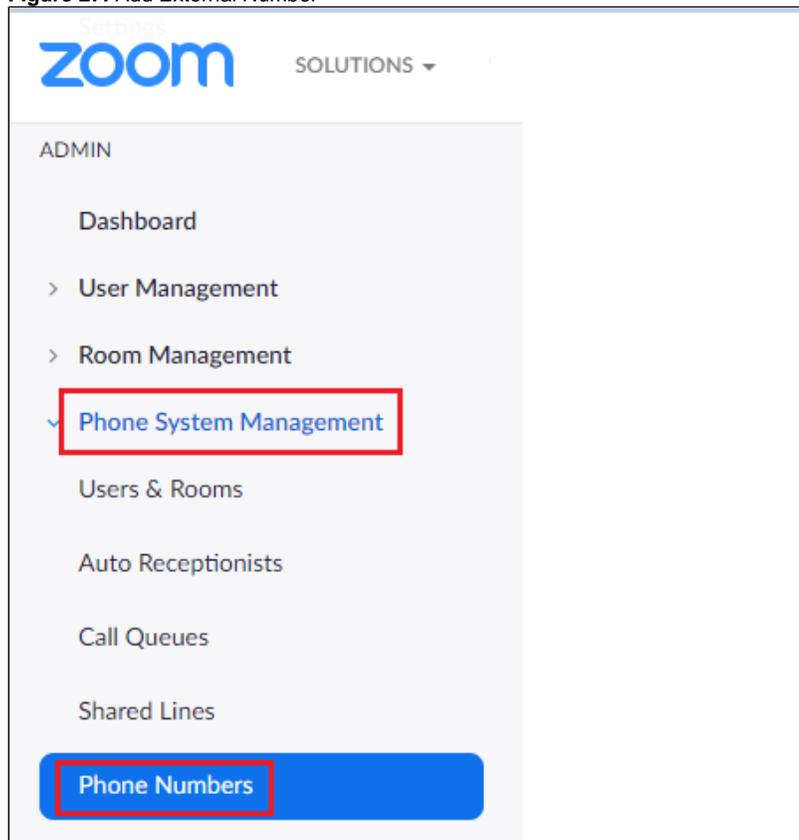
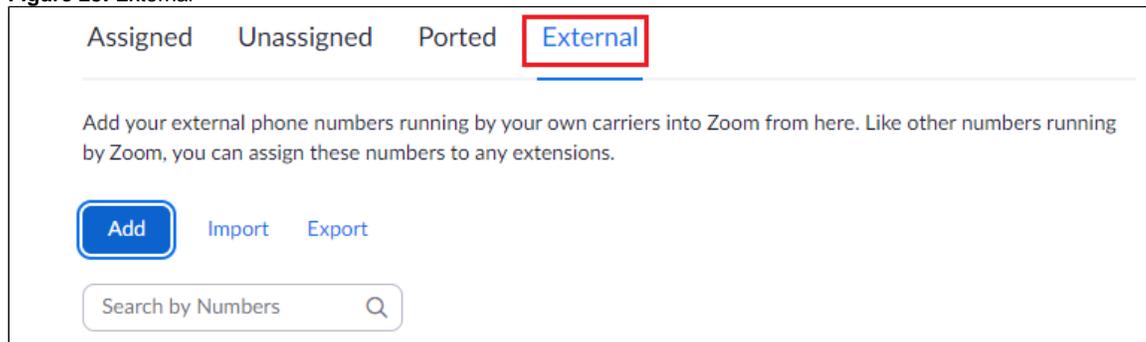


Figure 28: External



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

zoom 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

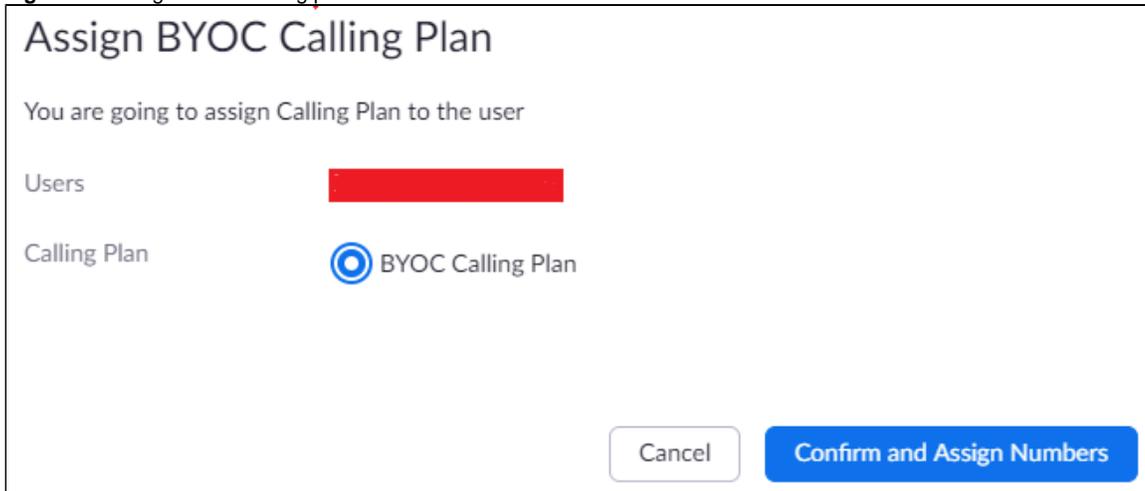
Search by Name, Ext. or Number

Plan (All) Status (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	--	--	--	Active	Assign Calling Plan <input type="button" value="v"/>

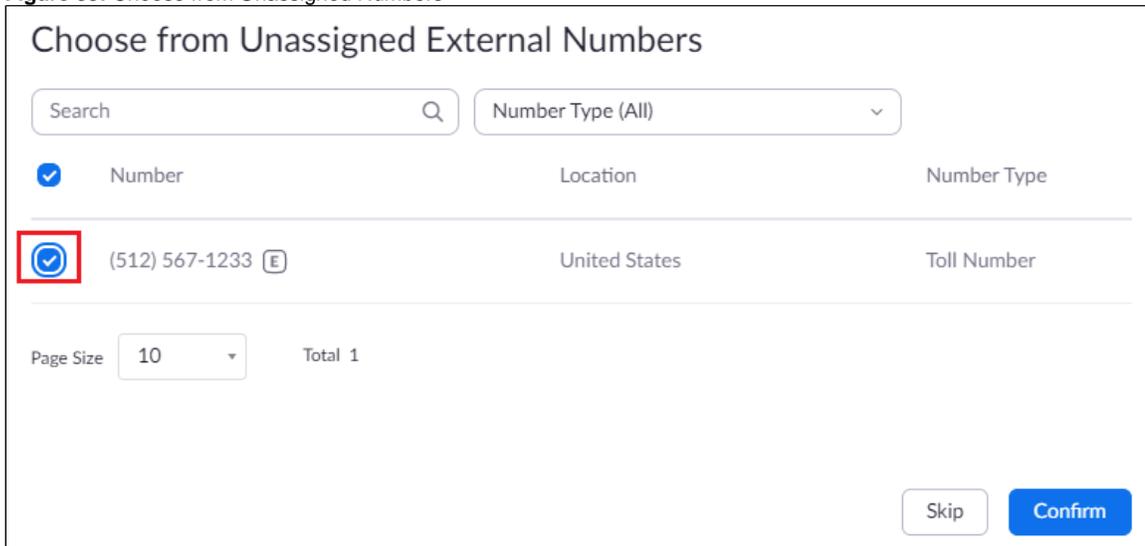
Figure 32: Assign BYOC calling plan



The dialog box is titled "Assign BYOC Calling Plan". It contains the text "You are going to assign Calling Plan to the user". Below this, there are two fields: "Users" with a red rectangular placeholder, and "Calling Plan" with a radio button selected next to "BYOC Calling Plan". At the bottom right, there are two buttons: "Cancel" and "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 features a search bar and a "Number Type (All)" dropdown menu. Below these is a table with columns for "Number", "Location", and "Number Type". The first row in the table is highlighted with a red box around the checkmark in the "Number" column. The table contains one row: (512) 567-1233 (E), United States, Toll Number. At the bottom left, there is a "Page Size" dropdown set to "10" and "Total 1". At the bottom right, there are "Skip" and "Confirm" buttons.

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

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

<input type="button" value="Add"/> <input type="button" value="Import"/> <input type="button" value="Export"/>						
<input type="text" value="Search by Name, Ext. or Number"/>					<input type="button" value="Plan (All)"/>	
<input type="button" value="Assign Numbers"/> <input type="button" value="Assign Calling Plan"/> <input type="button" value="Apply Settings"/> <input type="button" value="Remove"/>						
<input type="checkbox"/>	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/>