# 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 |

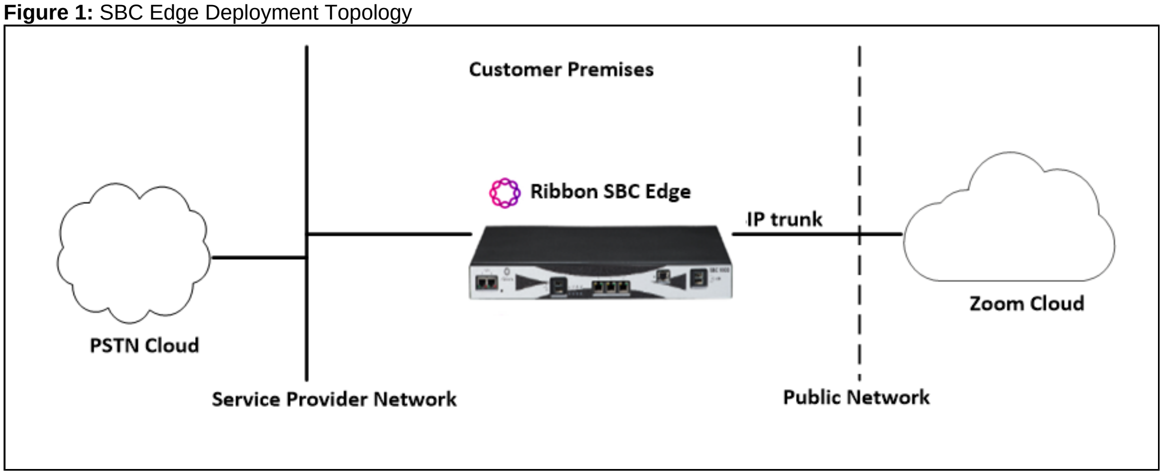
**Note**
Ribbon SBC Edge portfolio includes SBC 1000, SBC 2000 (both are appliance based) and SBC SWeLite (virtualized platform). Software Version is applicable to Ribbon SBC Edge portfolio (1000, 2000, SWeLite) and hence this configuration guide is valid for all these devices.

# Network Topology Diagram

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

## SBC Edge Deployment Topology

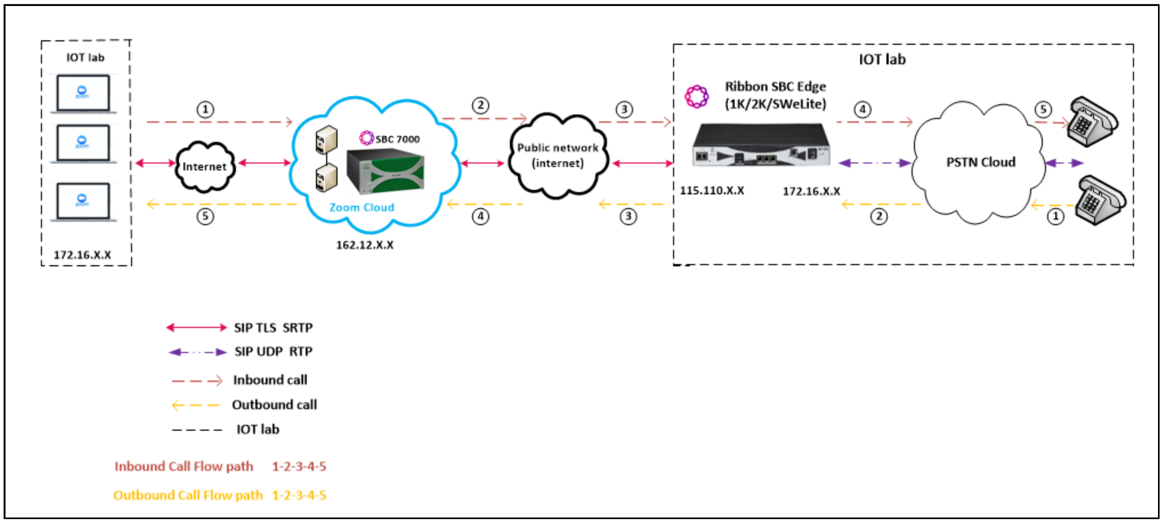
**Figure 1:** SBC Edge Deployment Topology




## 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

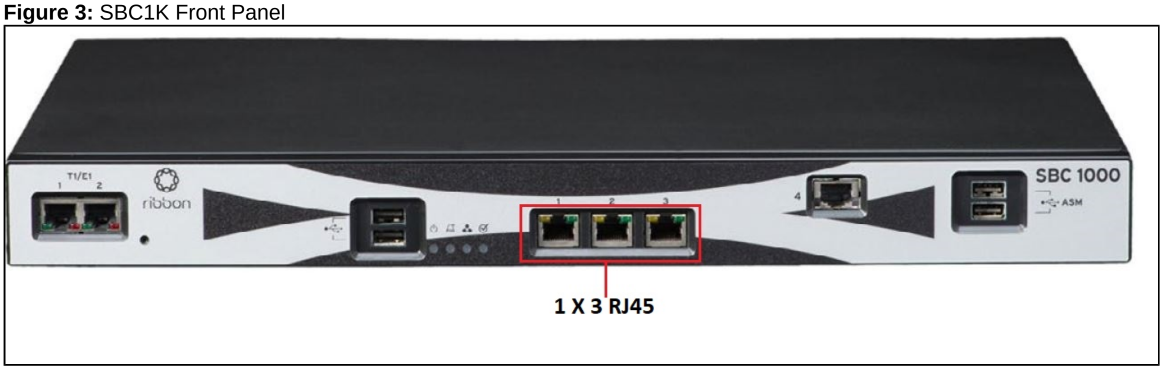
The following SBC Edge configurations are included in this section:

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

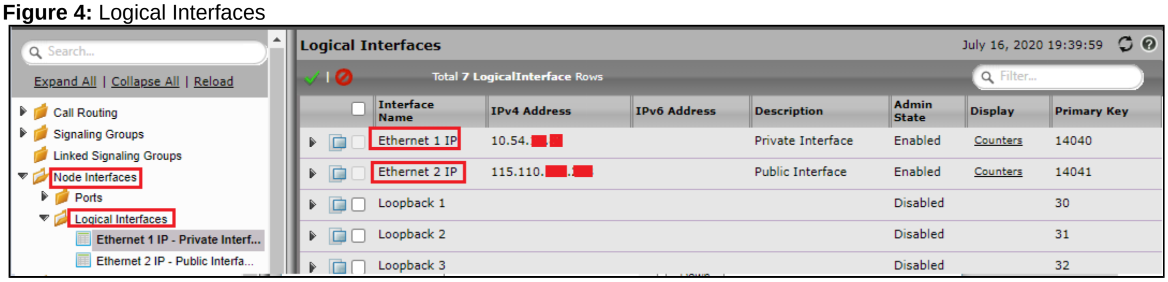
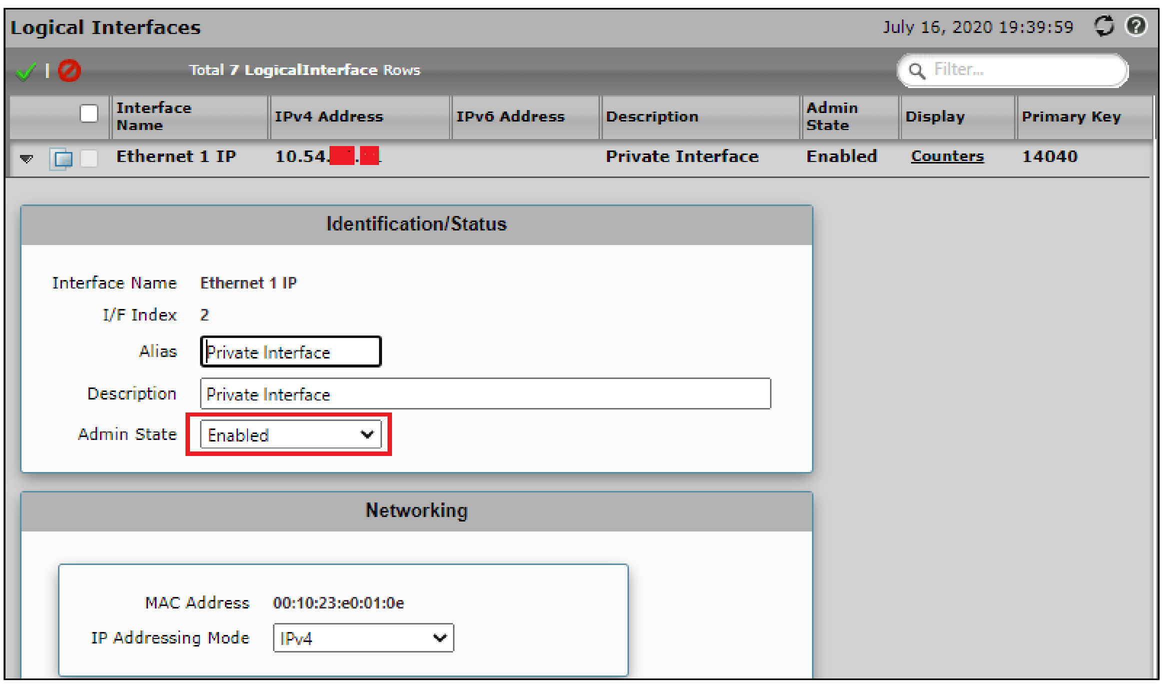



**Figure 5:** Ethernet 1



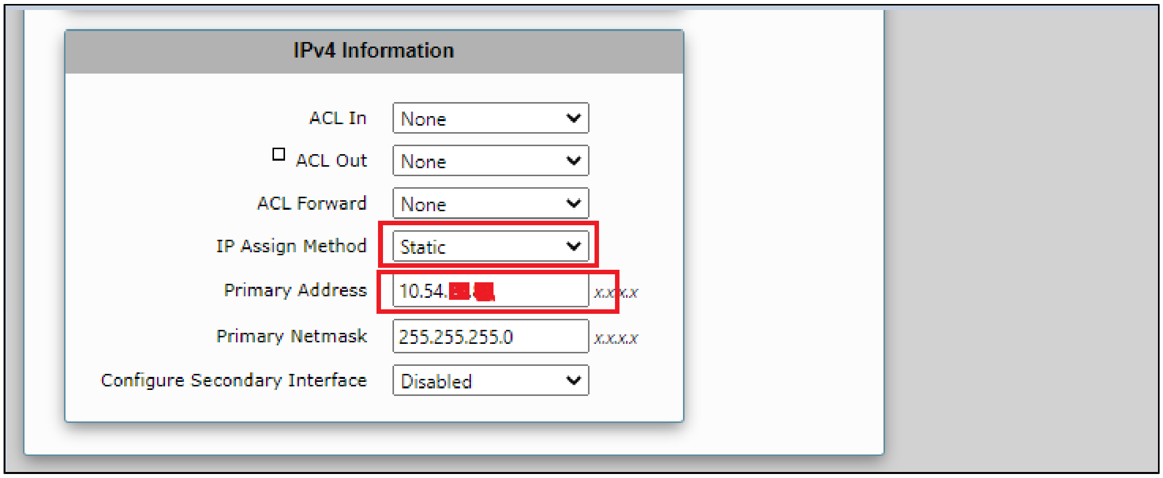
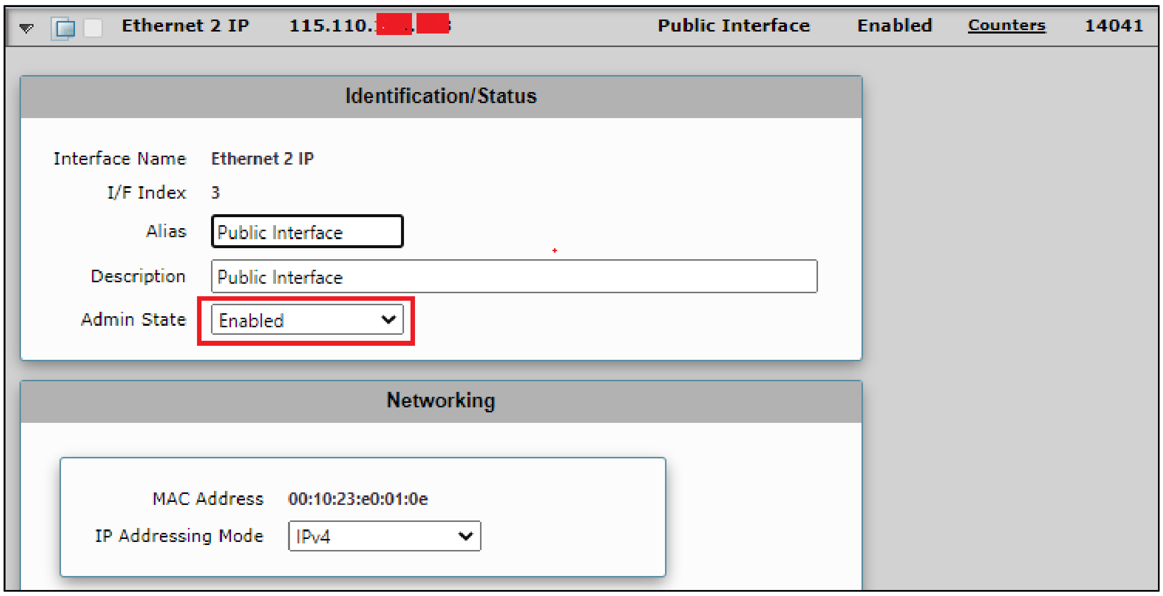
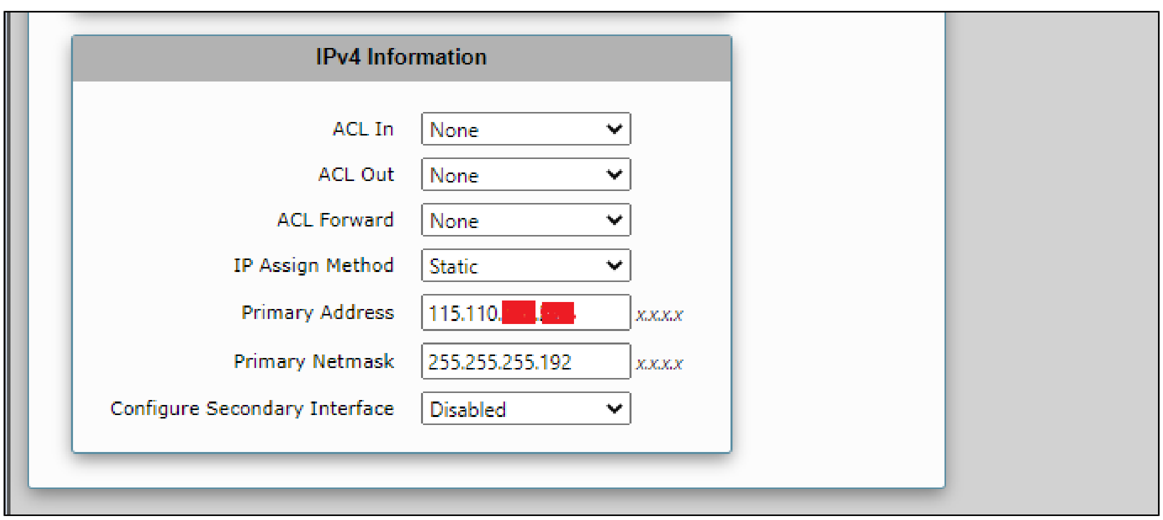



**Figure 6:** Ethernet 2




**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**

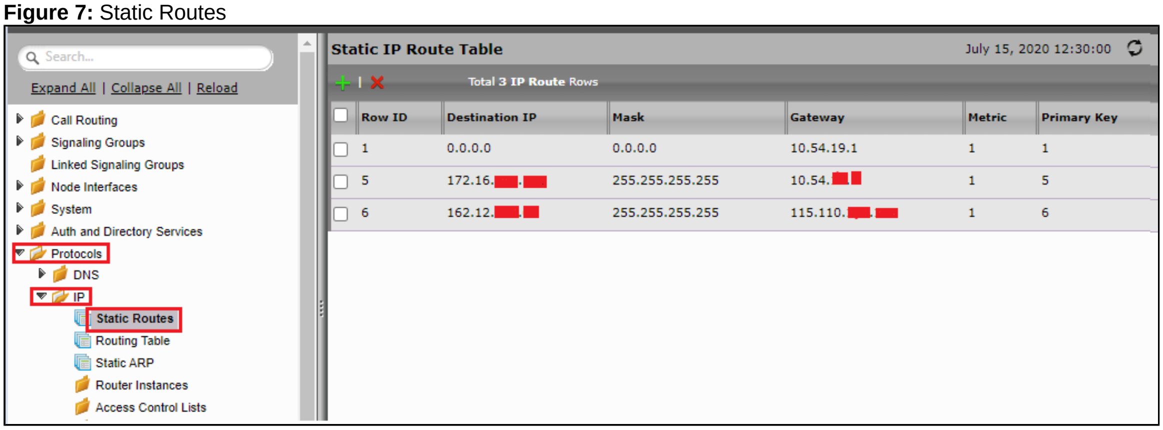
- For smaller networks with just one or two routes, configuring static routing is preferable. This is often more efficient since a link is not being wasted by exchanging dynamic routing information.
- For networks that have a LAN side Gateway on Voice VLAN or Multi-Switch Edge Devices (MSEs) with Voice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




# 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.

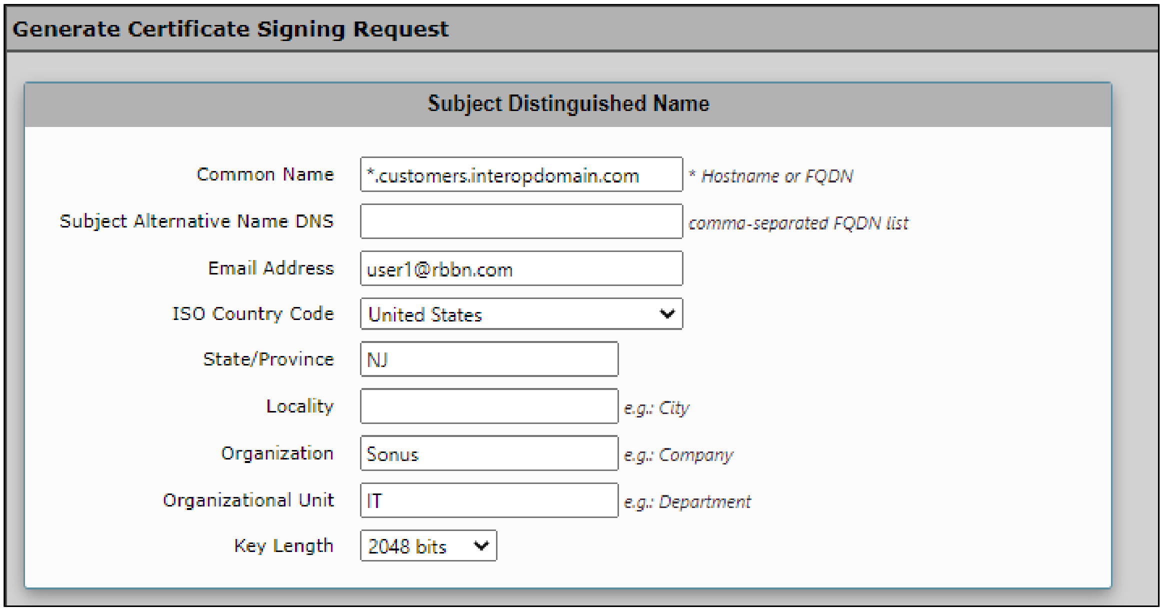
Request a certificate for the SBC and configure it based on the example using GoDaddy as follows:

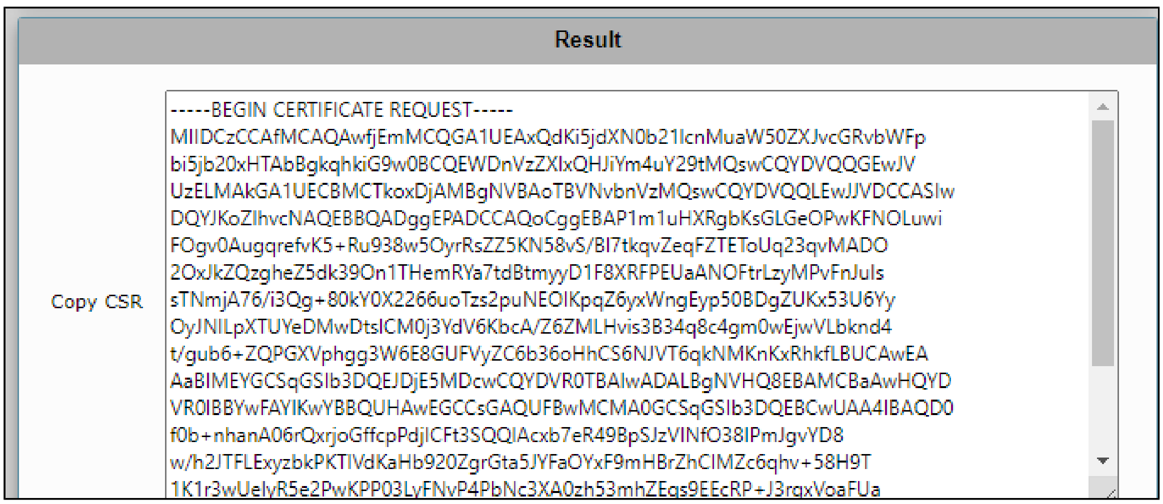
1. Generate a Certificate Signing Request (CSR) and obtain the certificate from a Certification Authority.
2. Import the Public CA Root/Intermediate Certificate and SBC certificate on the SBC.

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




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:

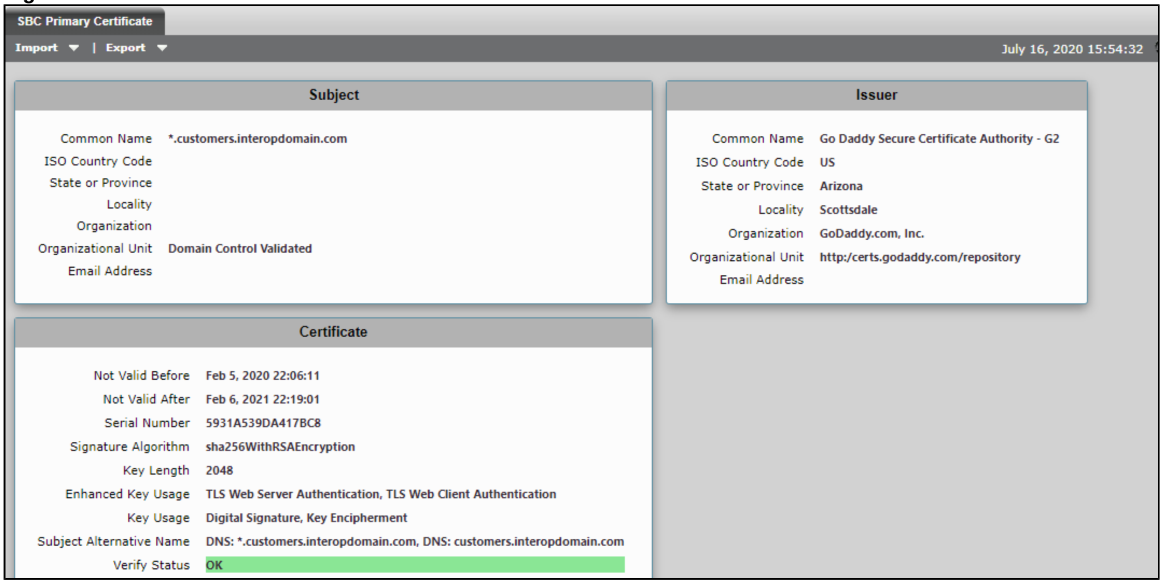
1. Obtain Trusted Root and Intermediary signing certificates from your certification authority.
2. To install Trusted Root/Intermediate Certificates, go to **Settings > Security > SBC Certificates > Trusted Root Certificates.**
3. Click **Import** and select the trusted root certificates.
4. To install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

**Figure 9:** Trusted CA certificate table

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

7. Validate the certificate is installed correctly.

**Figure 10:** Validate certificate




**TLS Profile**

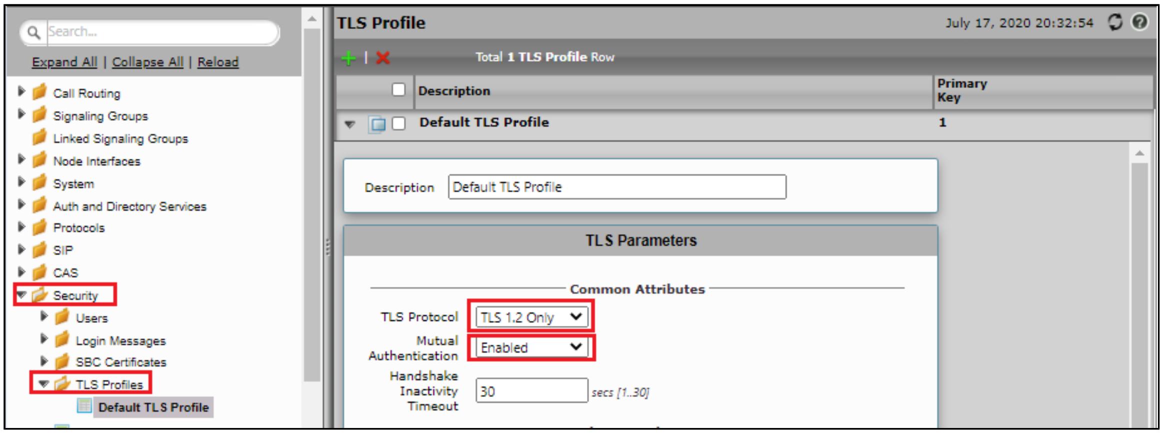
TLS Profile is required for the TLS handshake between SBC Edge and Zoom. This profile defines cipher suites supported by SBC Edge.

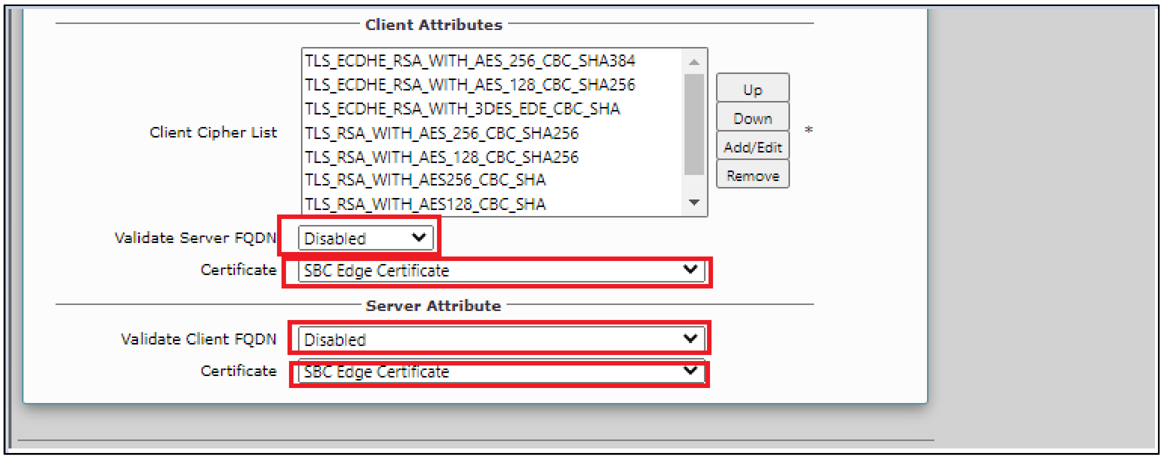
Default TLS Profile need to be attached to SIP Server Table on Zoom leg.

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

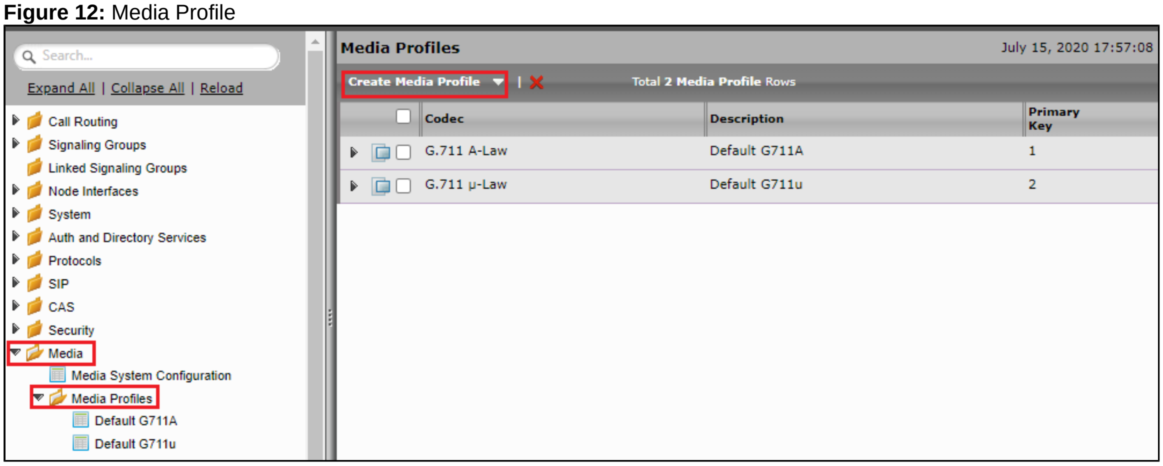



**Figure 13:** G711-A law

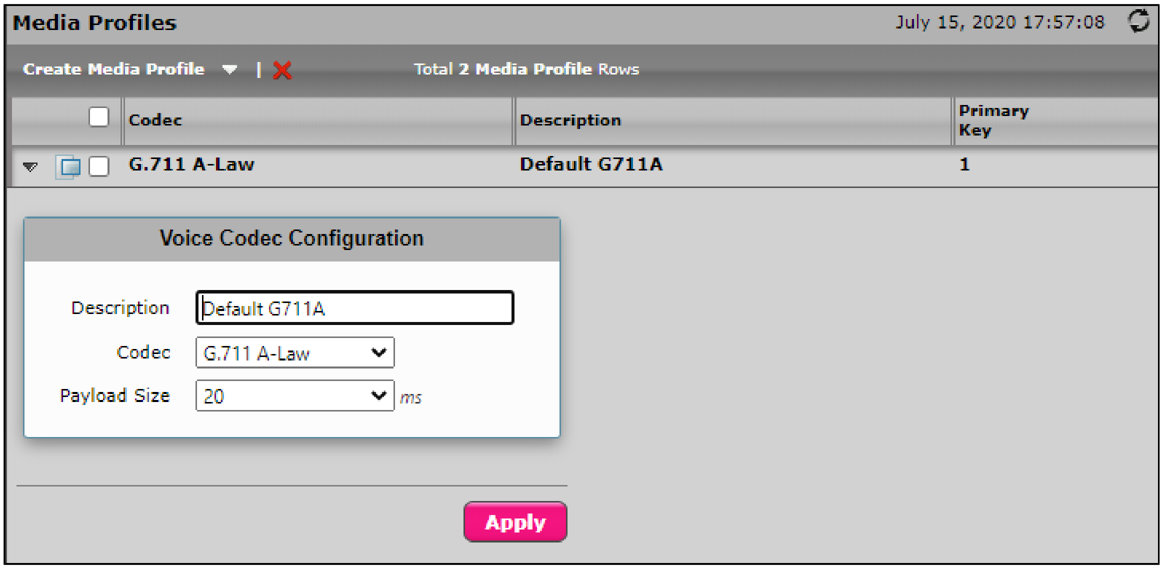
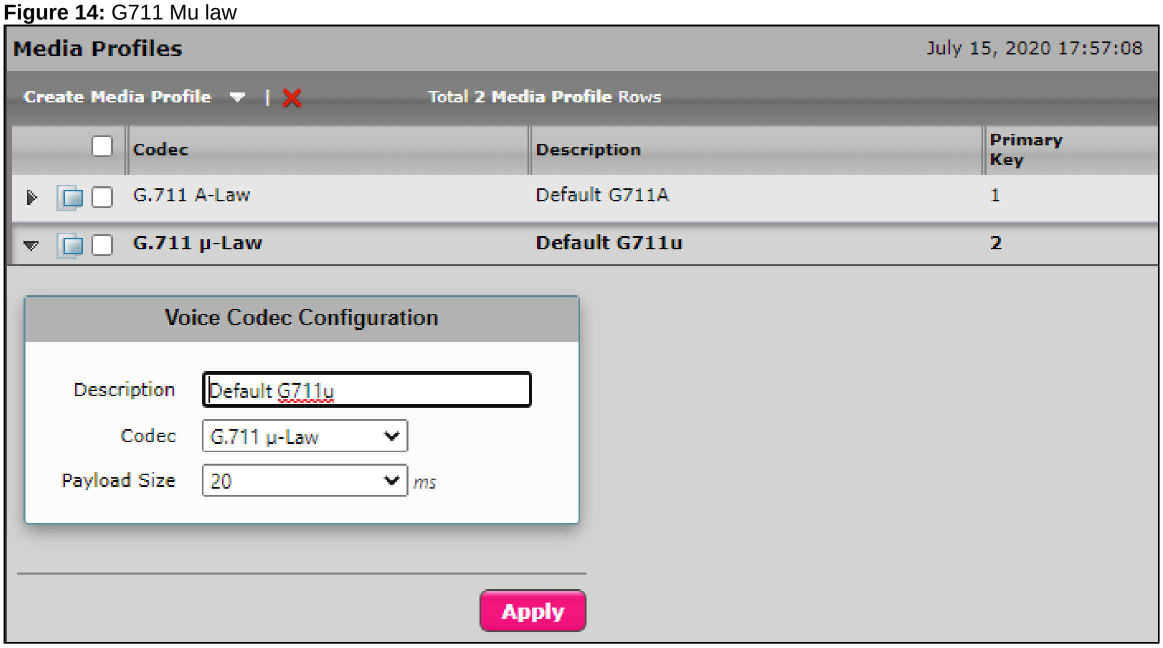



**Figure 14:** G711 Mu law

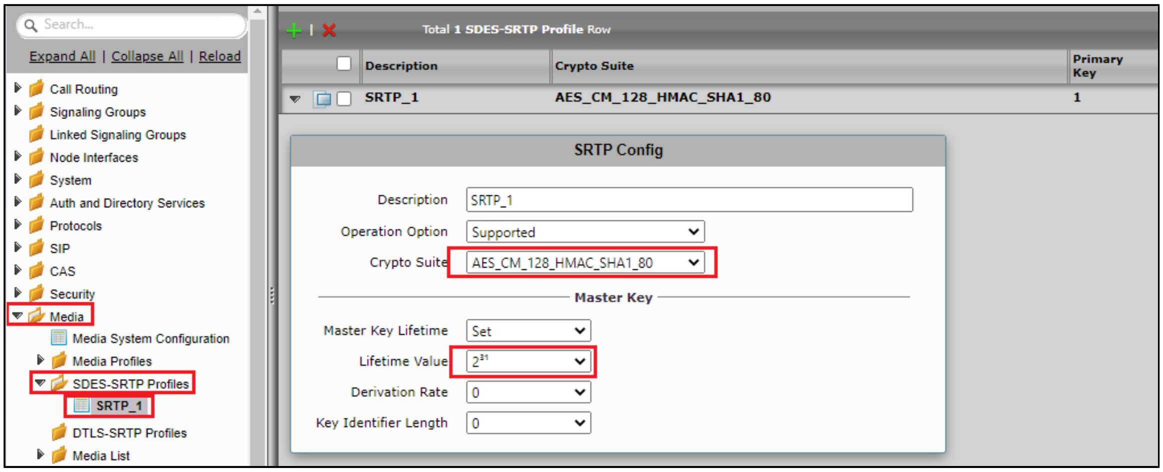



# 6. SRTP Profile

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

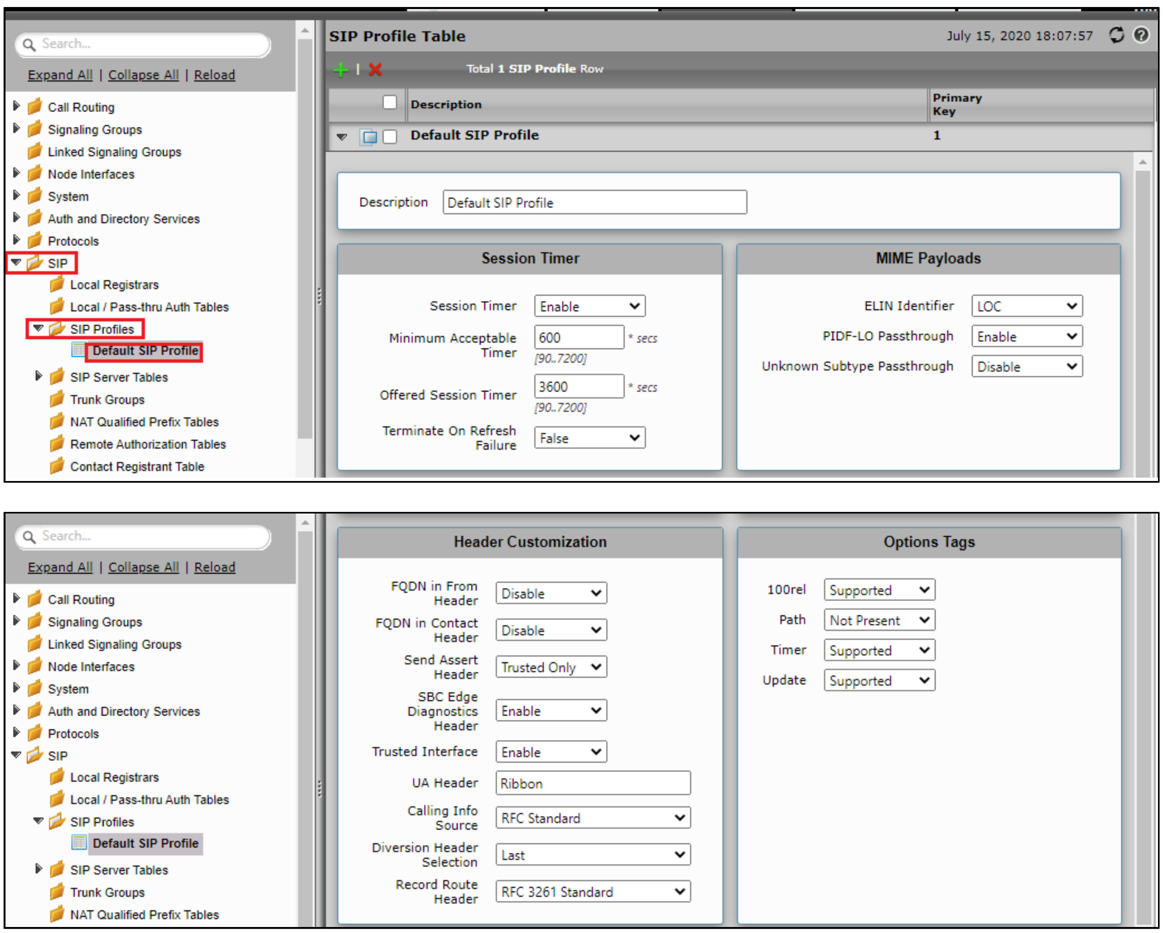


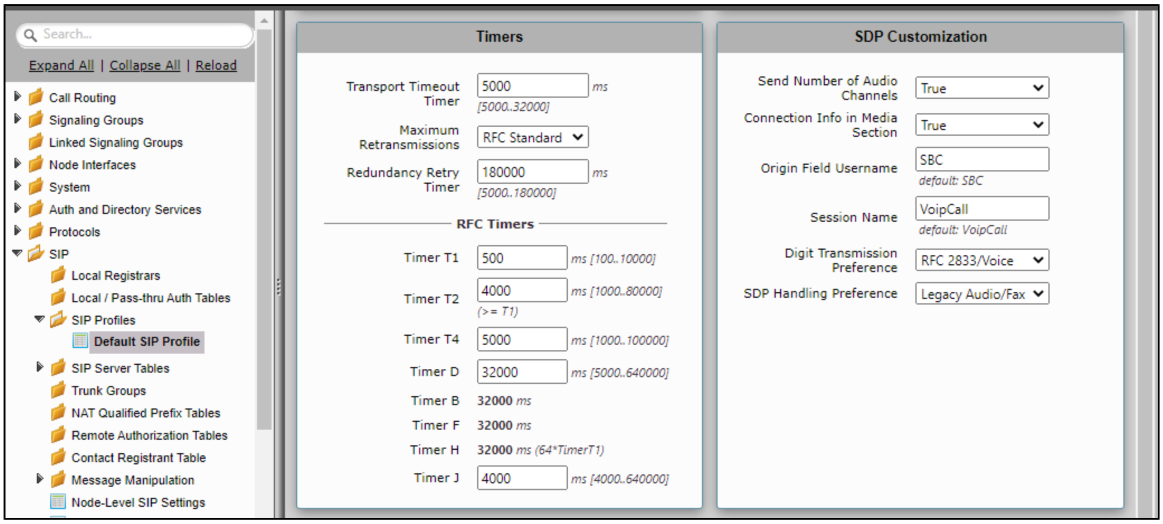

## 7. SIP Profile

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:

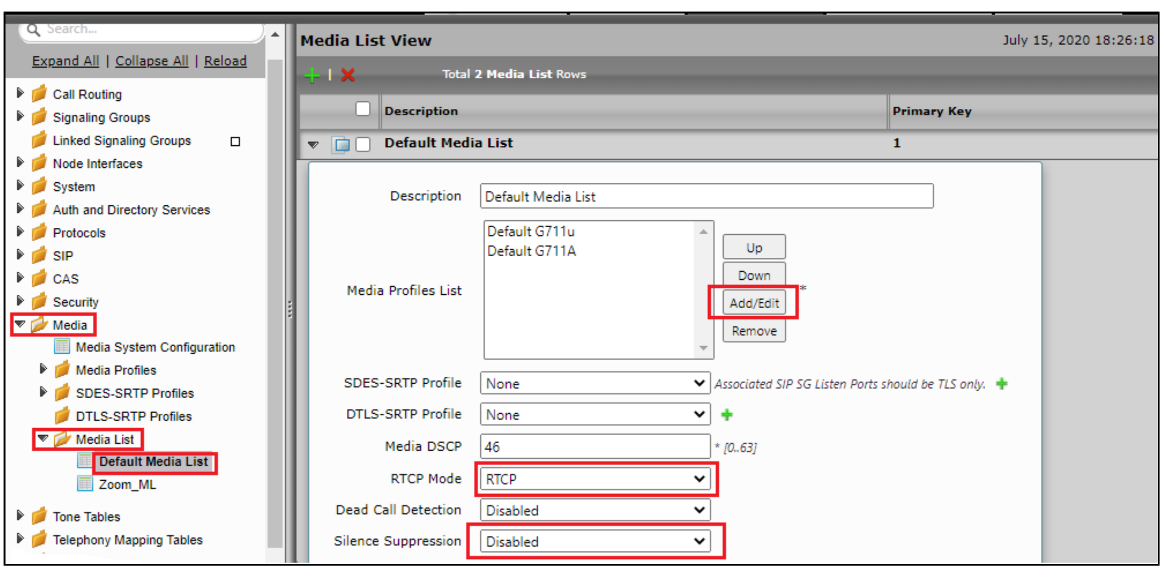
1. Media List
2. SIP Server Tables
3. Signaling Group
4. Transformation
5. Call Routing Table

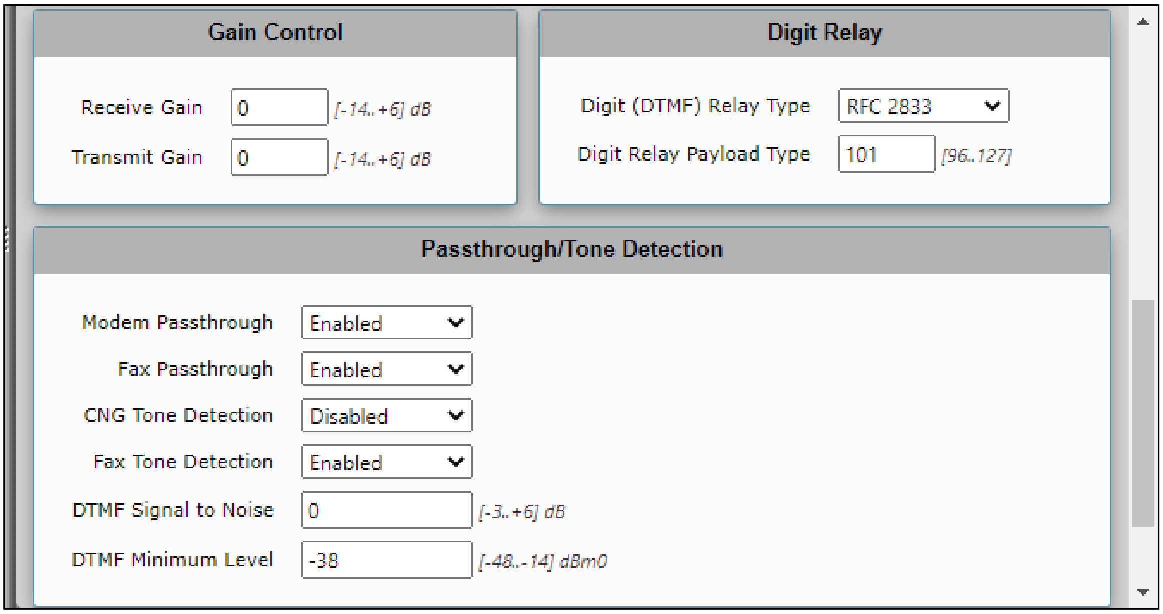
### 1. Media List

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



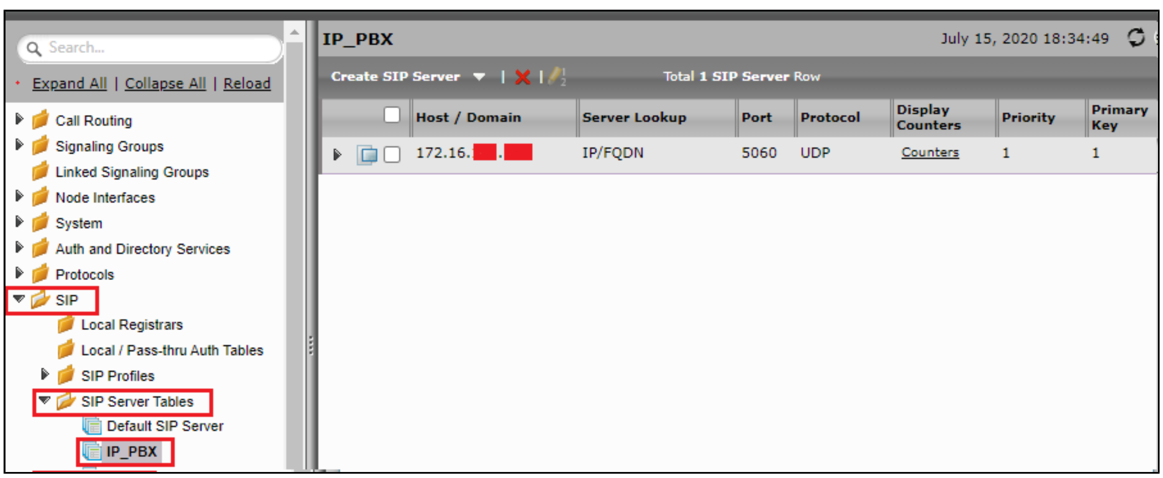


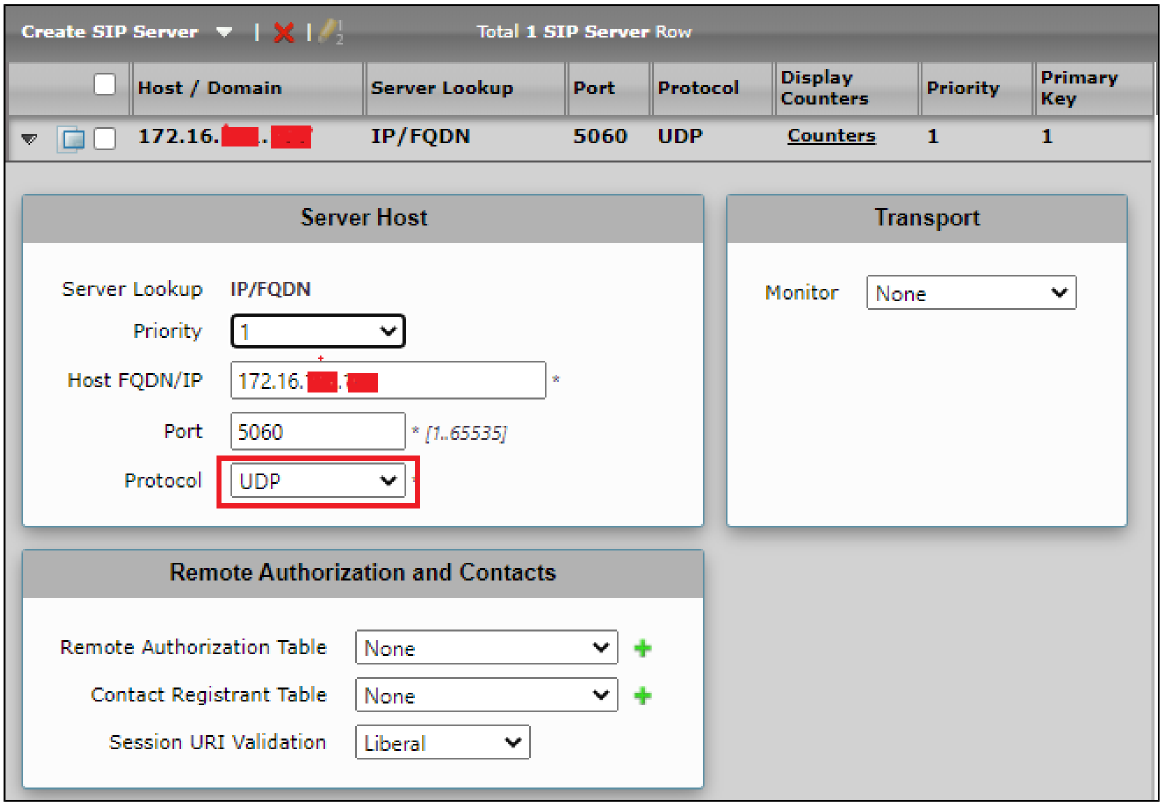

## 2. SIP Server Tables

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





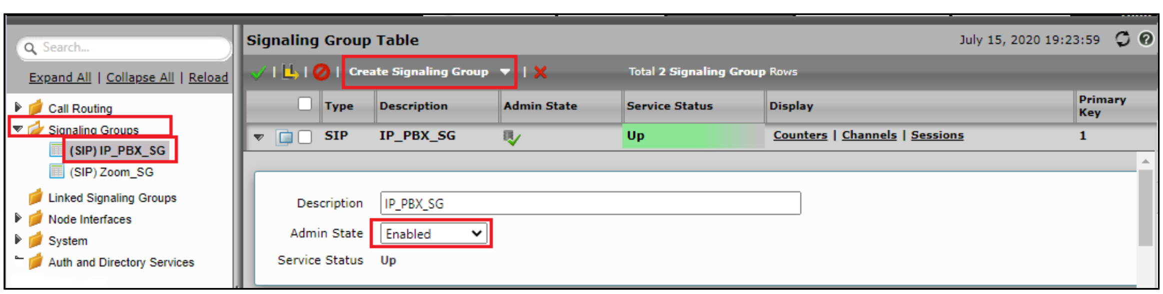

## 3. Signaling Groups

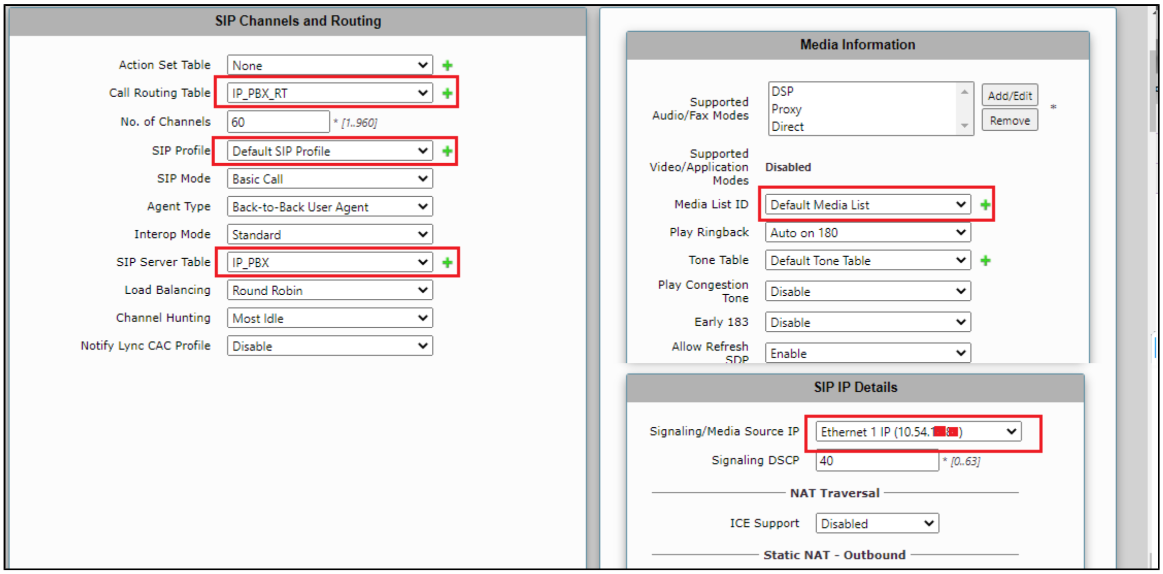
Signaling groups allow telephony channels to be grouped together for the purposes of routing and shared configuration. They are the entity to which calls are routed, as well as the location from which Call Routes are selected. They are also the location from which Tone Tables and Action Sets are selected.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

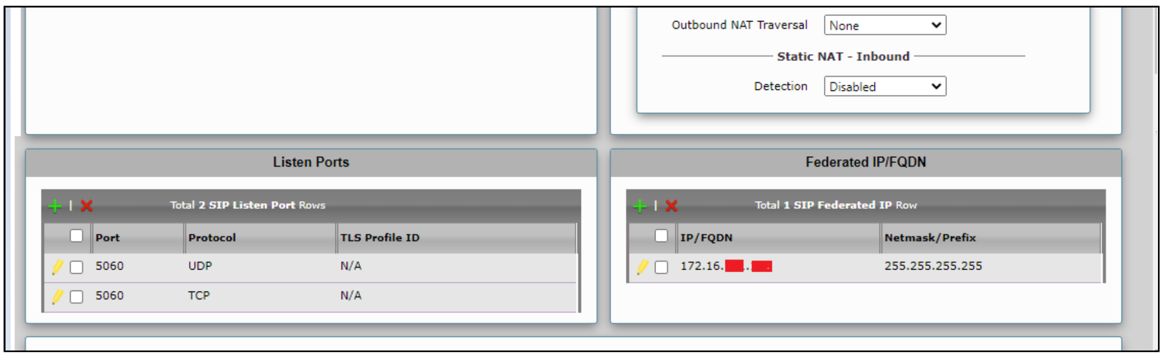
- Navigate to **Settings > Signaling Groups > Create Signaling Group.**
- From the drop-down select "SIP Signaling Group".
- Set SIP Profile as "Default SIP Profile".
- Set SIP Server Table as "IP_PBX".
- Set Media List ID as "Default Media List".
- Configure Signaling/Media Source IP as "Ethernet 1 IP(10.54.X.X)".
- Configure Federated IP as PSTN IP (172.16.X.X).

> **Tip**
> Set Call Routing table as "IP_PBX_RT" which is created in the Call Routing Table section.

**Figure 19:** Signalling Groups






## 4. Transformation

Example:

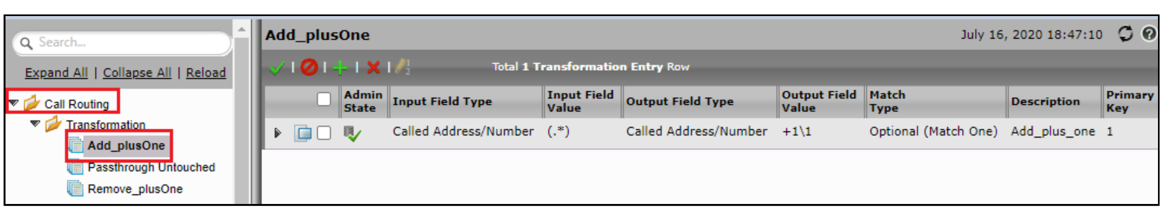
A customer has an existing carrier that only accepts the U.S.A. domestic "10-digit" dial plan numbering format. For example: (XXX) YYY-ZZZZ. Where XXX=area code, YYY-ZZZZ=7-digit phone number. Zoom is using the E.164 numbering format: +(country code)(phone number). This creates a phone number format 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).

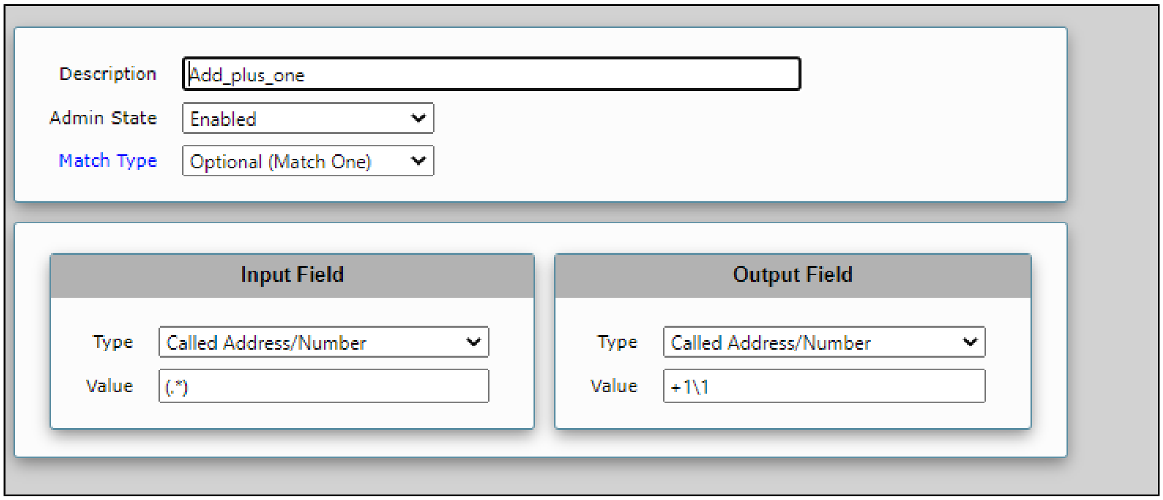
> **Note**
> Ribbon SBC Edge can be programmed for different country E.164 code mapping in addition to the U.S. dial plan.

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

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

**Figure 20:** Transformation





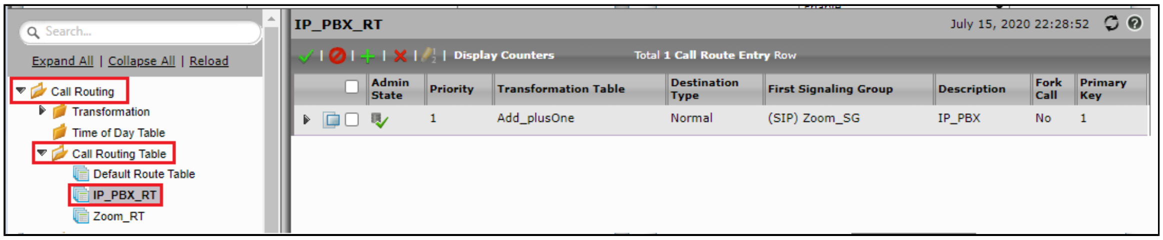

## 5. Call Routing Table

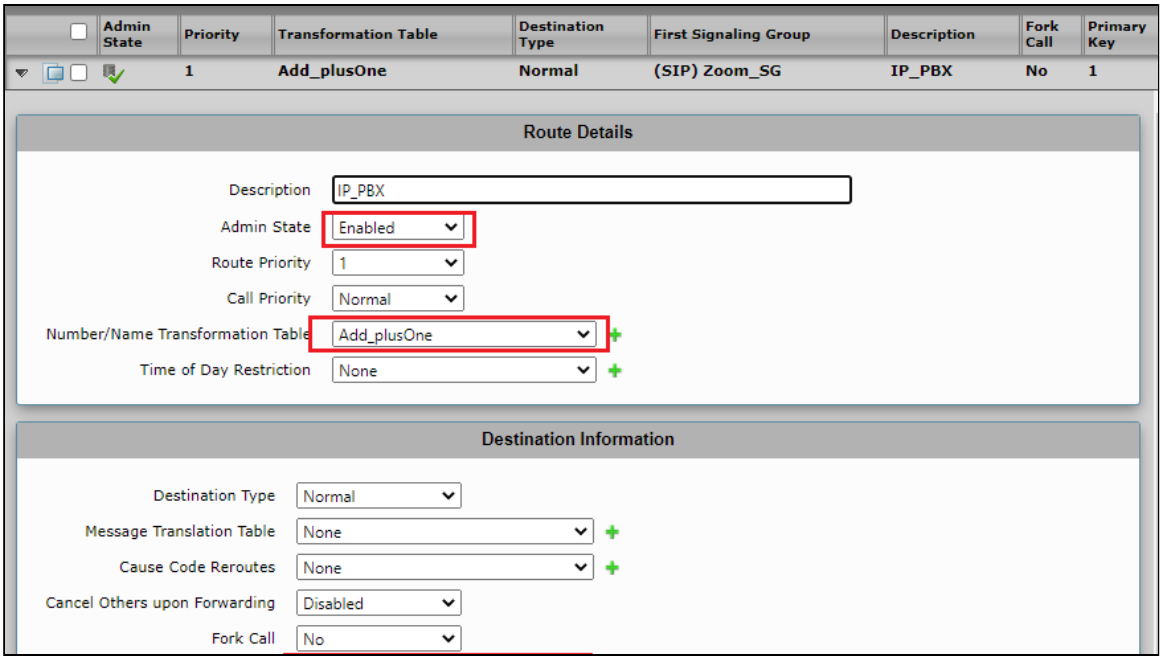
Call Routing allows calls to be carried between signaling groups, thus allowing calls to be carried between ports, and between protocols (like ISDN to SIP). Routes are defined by Call Routing Tables, which allow for flexible configuration of which calls are carried and how they are translated. These 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).

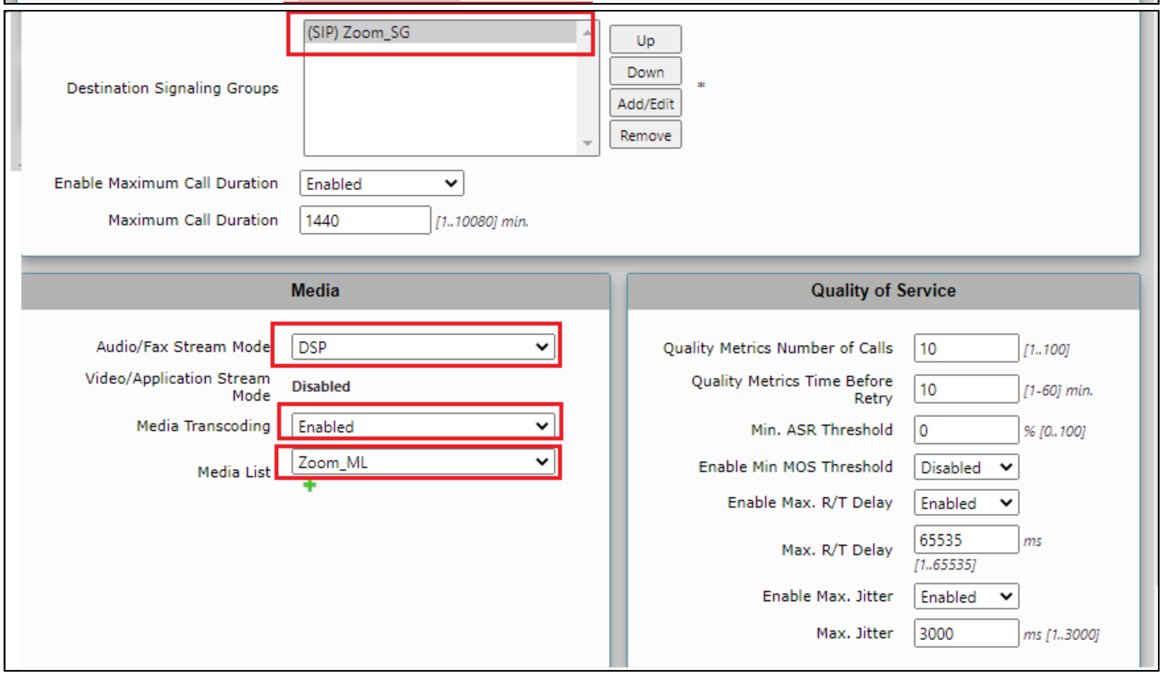
- Navigate to **Settings > Call Routing > Call Routing Table.**
- Set Number/Name Transformation Table as "Add_plusOne" as created in an earlier step.
- Destination Signaling Groups as "(SIP)Zoom_SG" in the Media. Select the following options:
    - Audio/Fax Stream Mode as "DSP".
    - Media Transcoding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







## 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:

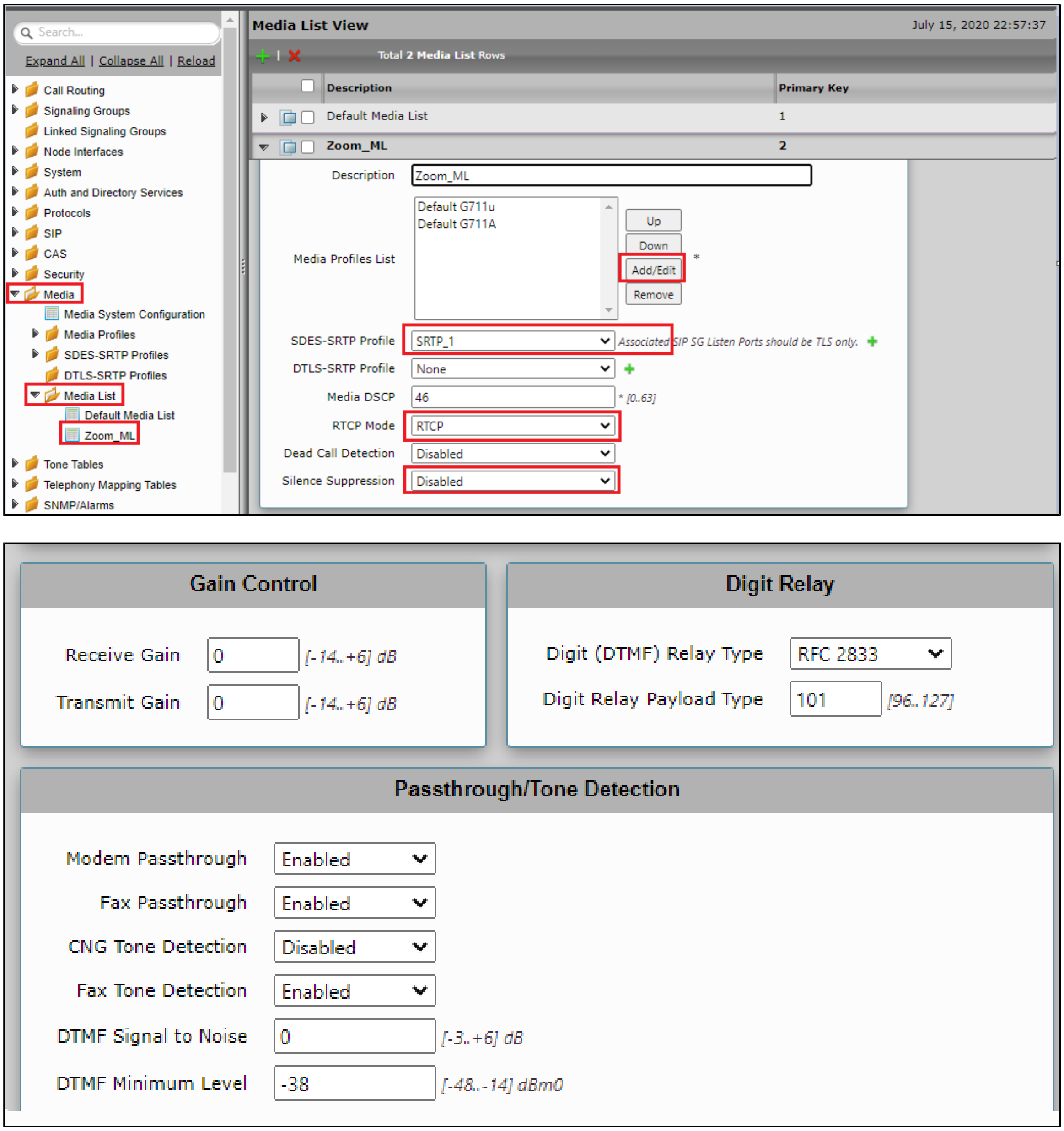
1. Media List.
2. SIP Server Tables.
3. Signaling Group.
4. Transformation.
5. Call Routing Table.

### 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

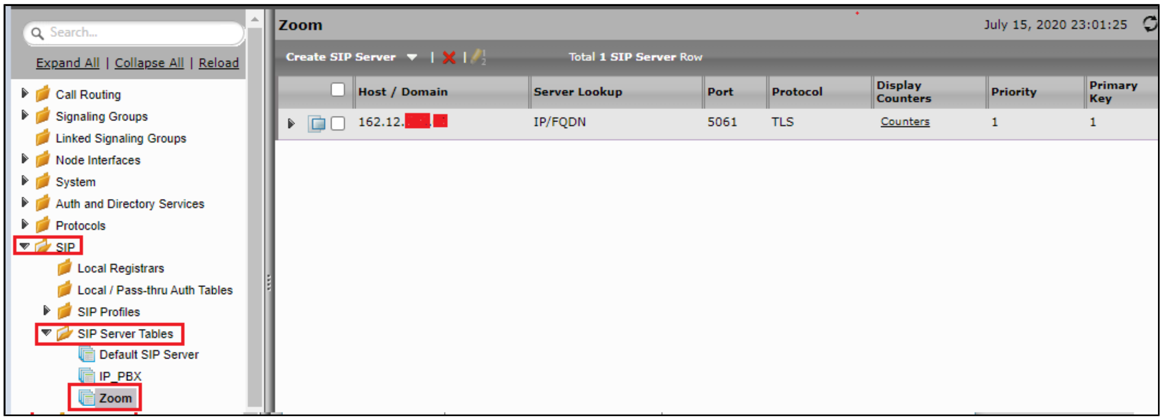


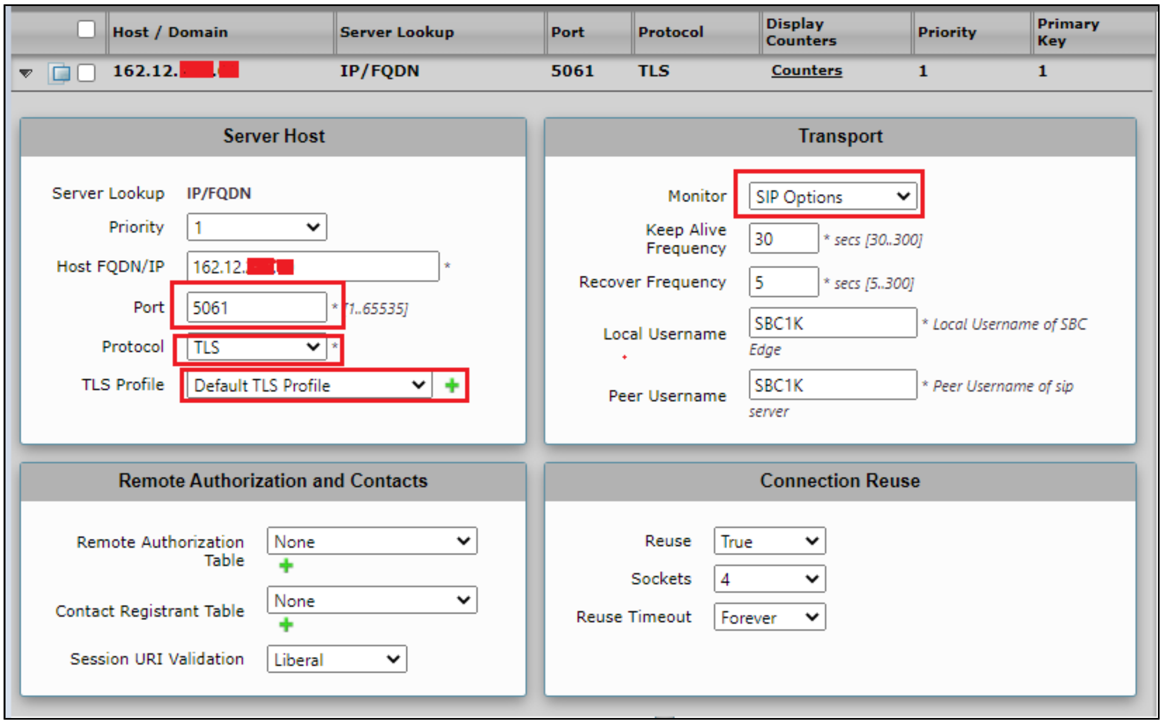

## 2. SIP Server Tables

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



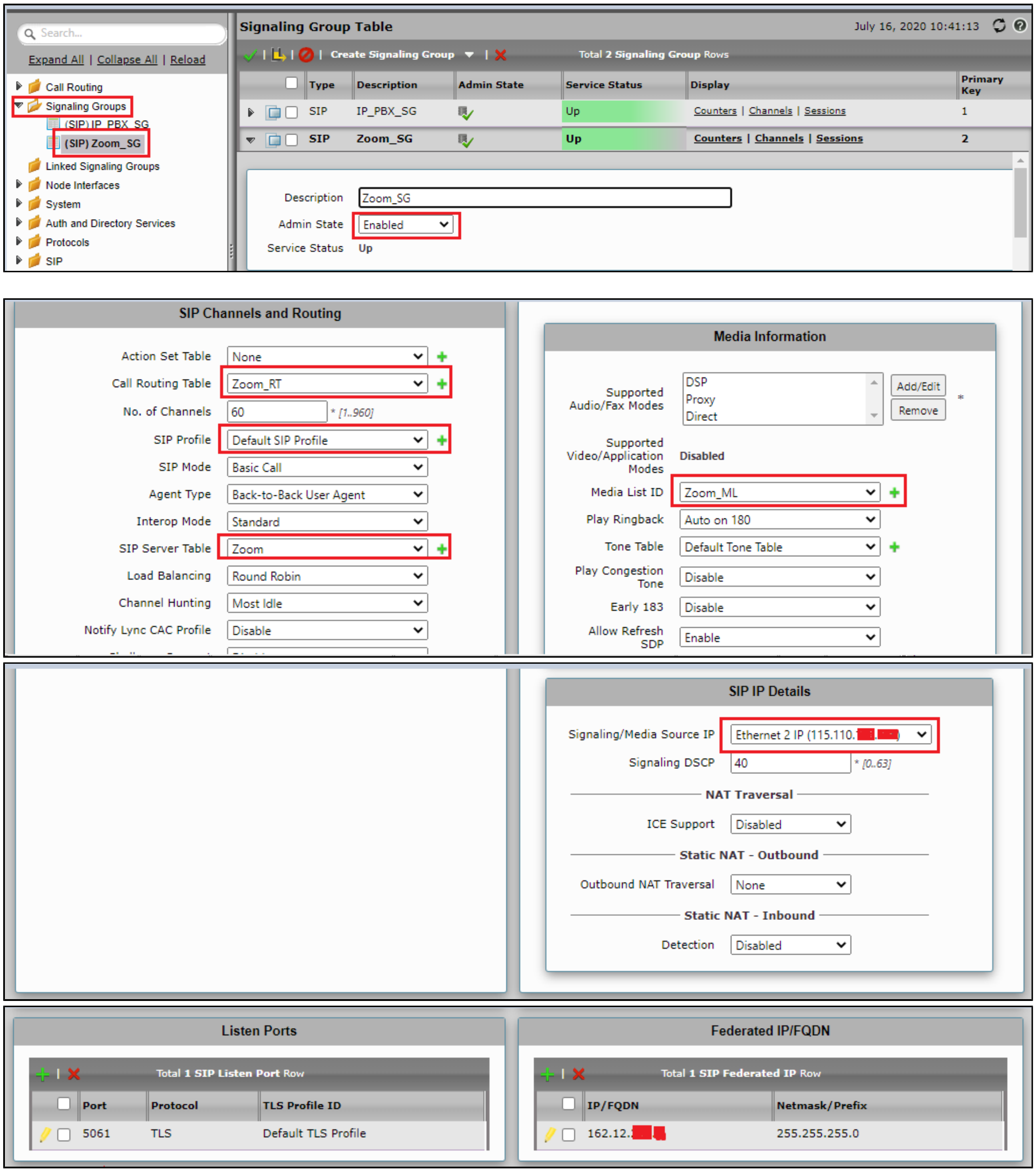

## 3. Signaling Groups

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

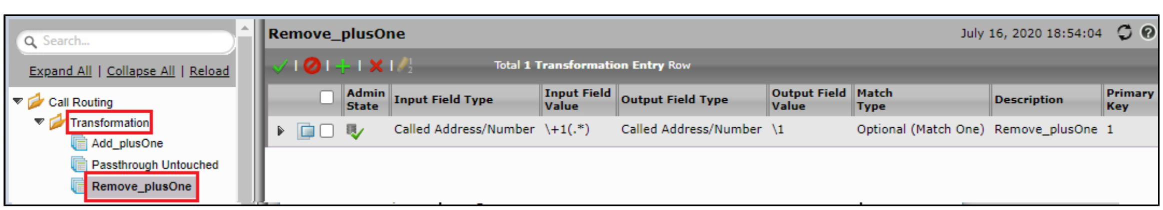


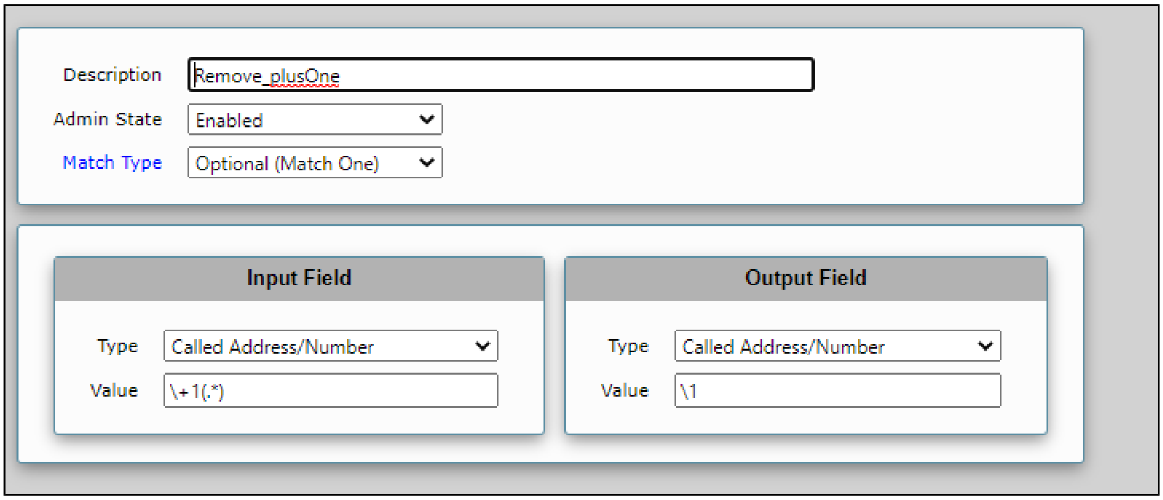

## 4. Transformation

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

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

**Figure 25:** Transformation





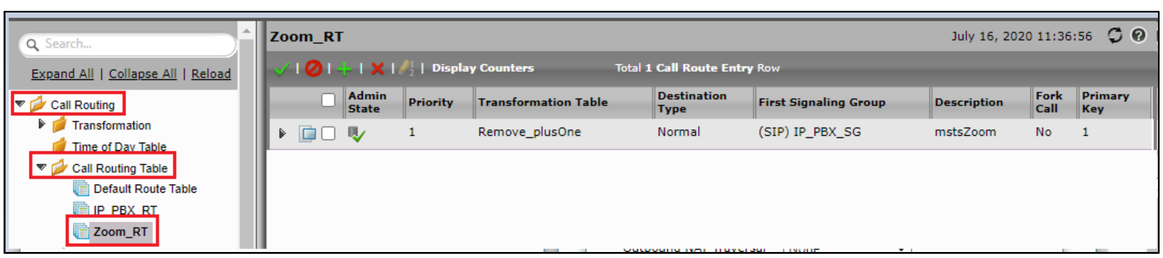

## 5. Call Routing Table

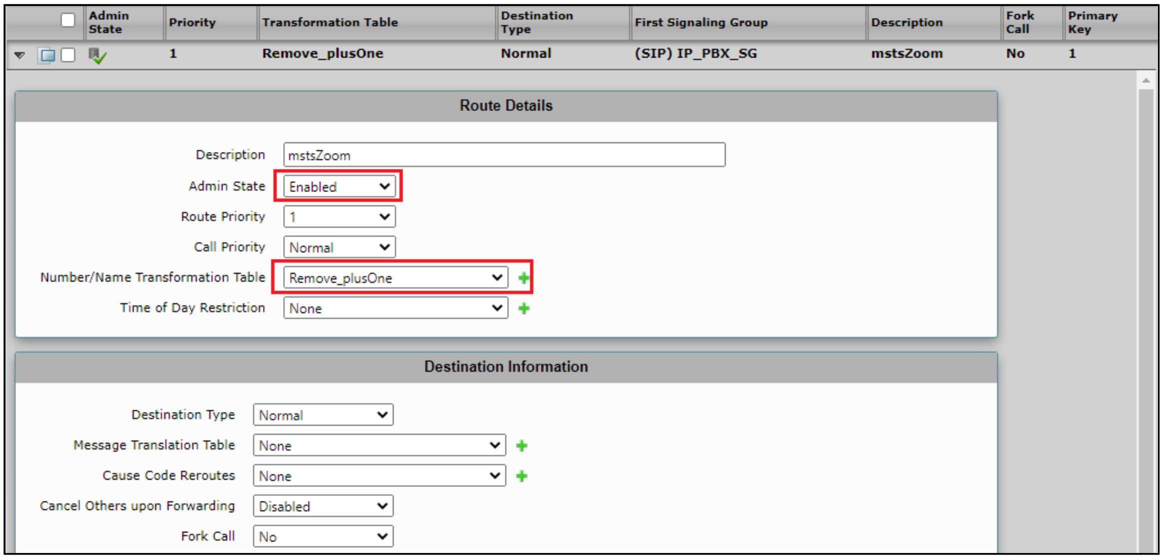
Call Routing allows calls to be carried between signaling groups, thus allowing calls to be carried between ports, and between protocols (like ISDN to SIP). Routes are defined by Call Routing Tables, which allow for flexible configuration of which calls are carried, and how they are translated. These 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).

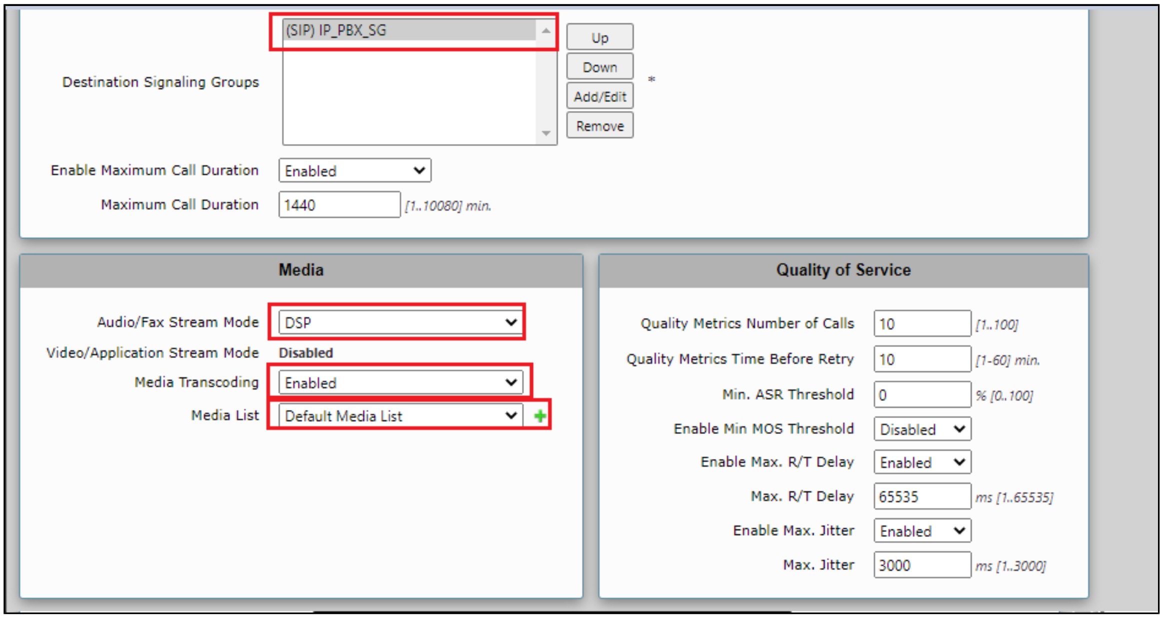
- Navigate to **Settings >Call Routing > Call Routing Table.**
- Set Number/Name Transformation Table as "Remove_plusOne" as created in an earlier step.
- Destination Signaling Groups as "(SIP)IP_PBX_SG" In the Media, select the following options:
  - Audio/Fax Stream Mode as "DSP".
  - Media Transcoding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






# 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

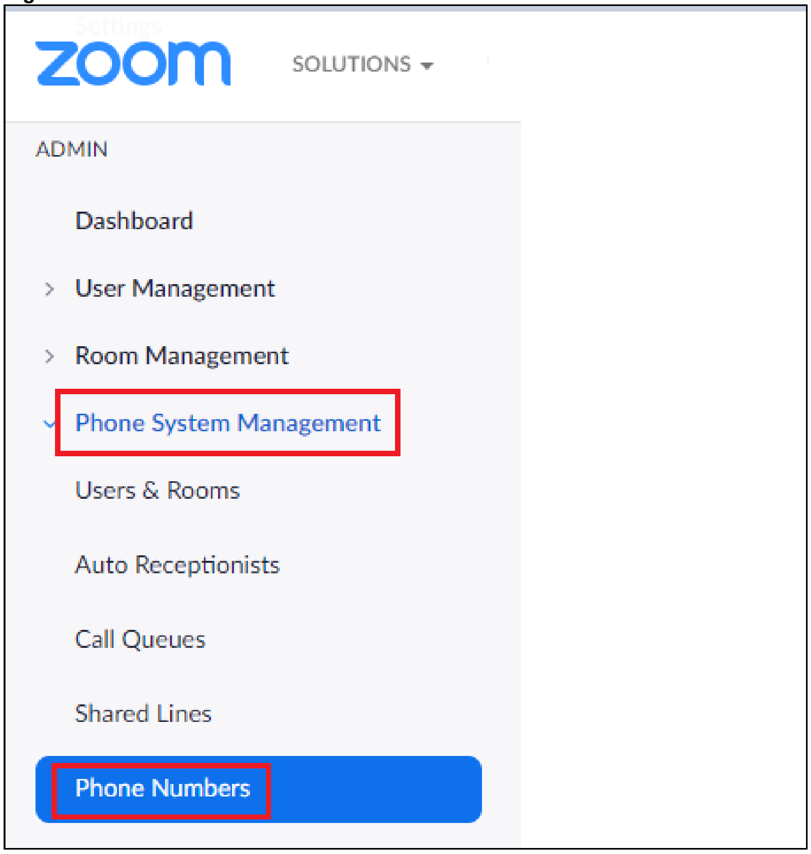
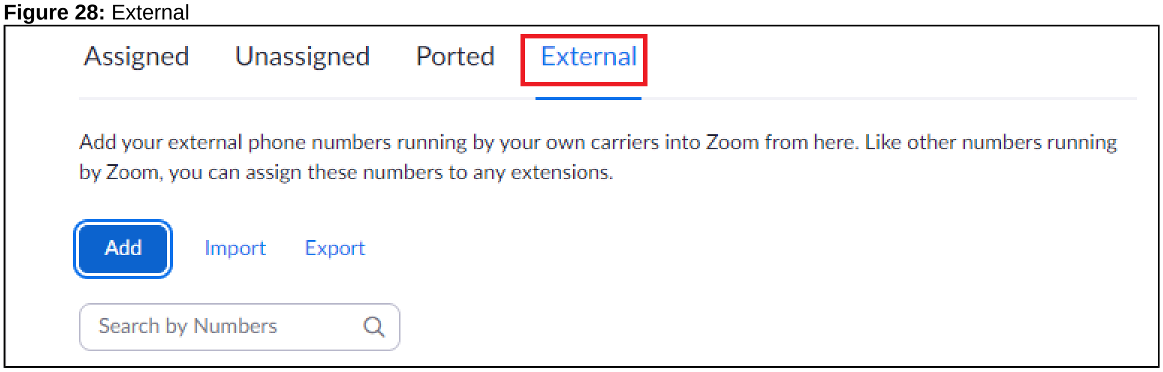
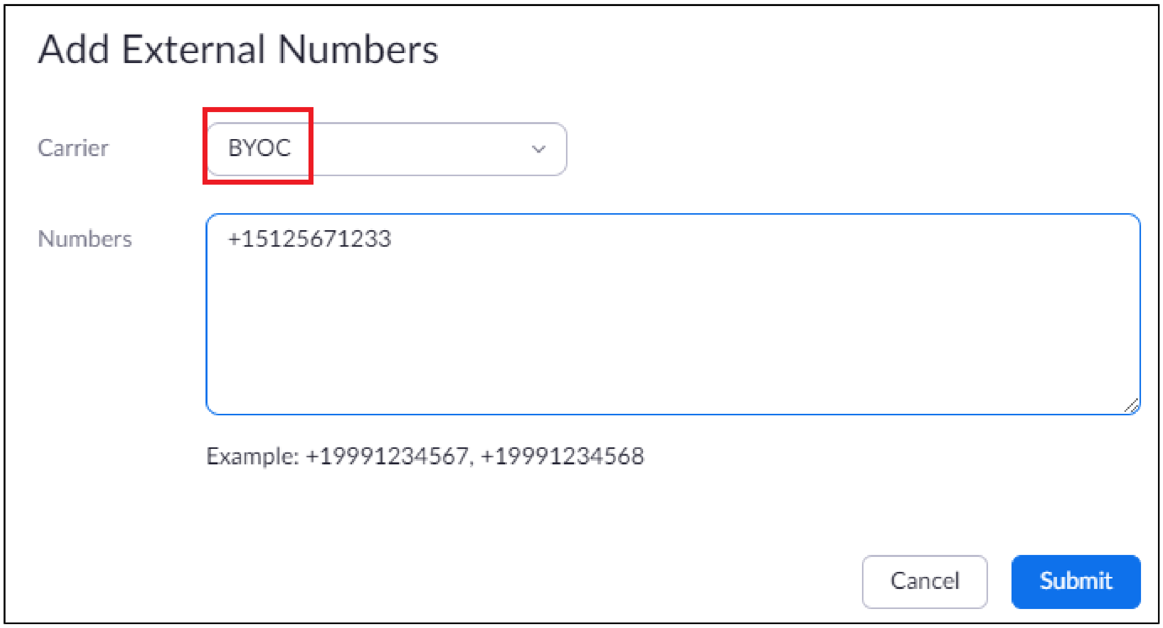



**Figure 28:** External



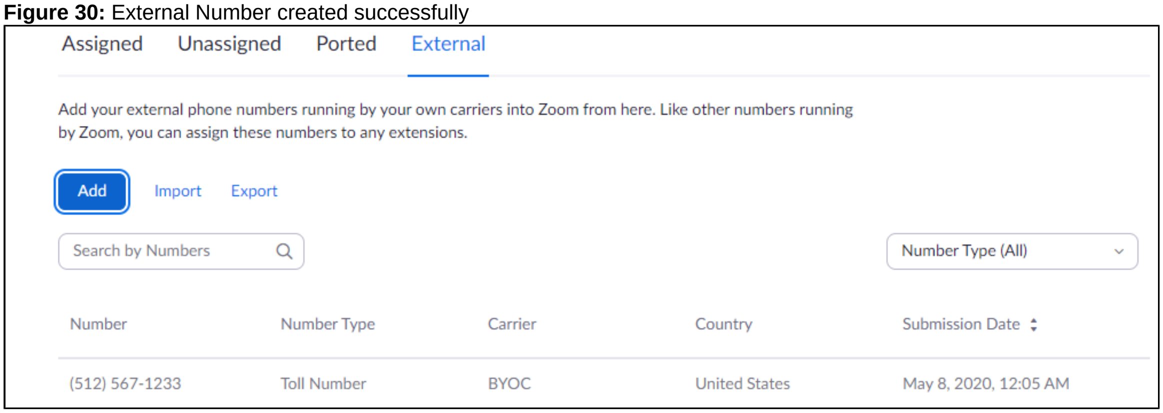

1. Select **BYOC** as the carrier.
2. Enter the existing customer phone numbers (from carrier) separated by commas.
3. Click **Submit**.

**Figure 29:** Add External Number




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

**Figure 30:** External Number created successfully




## 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

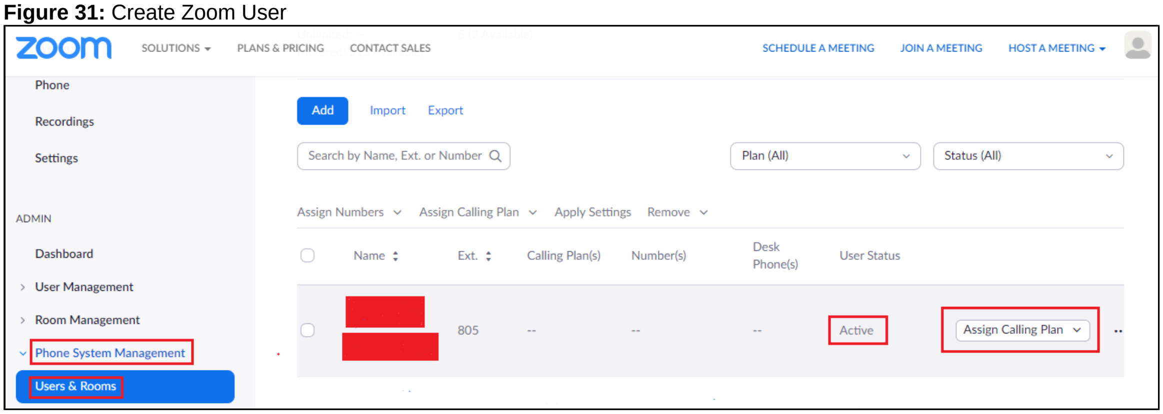
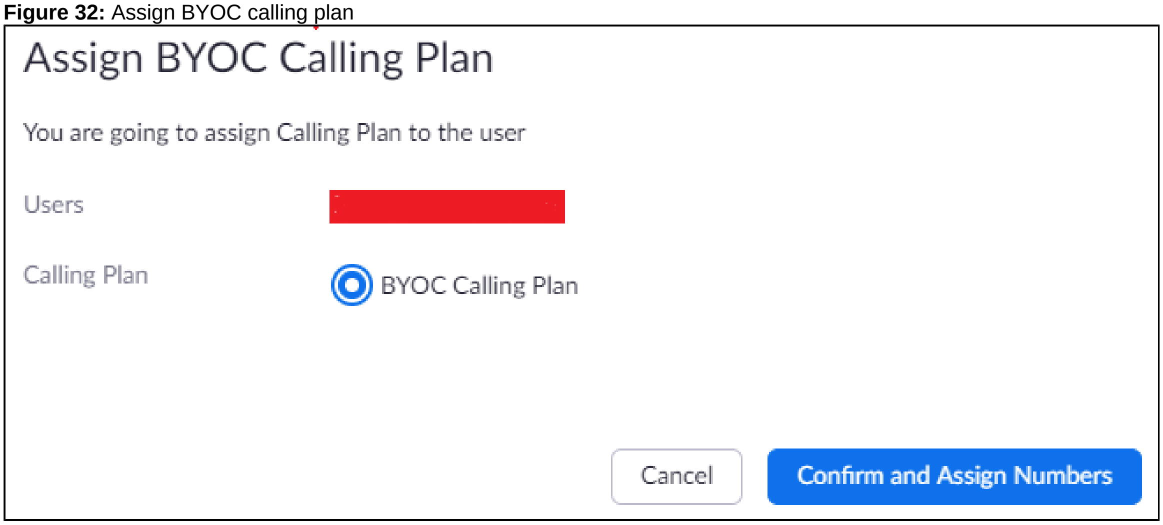
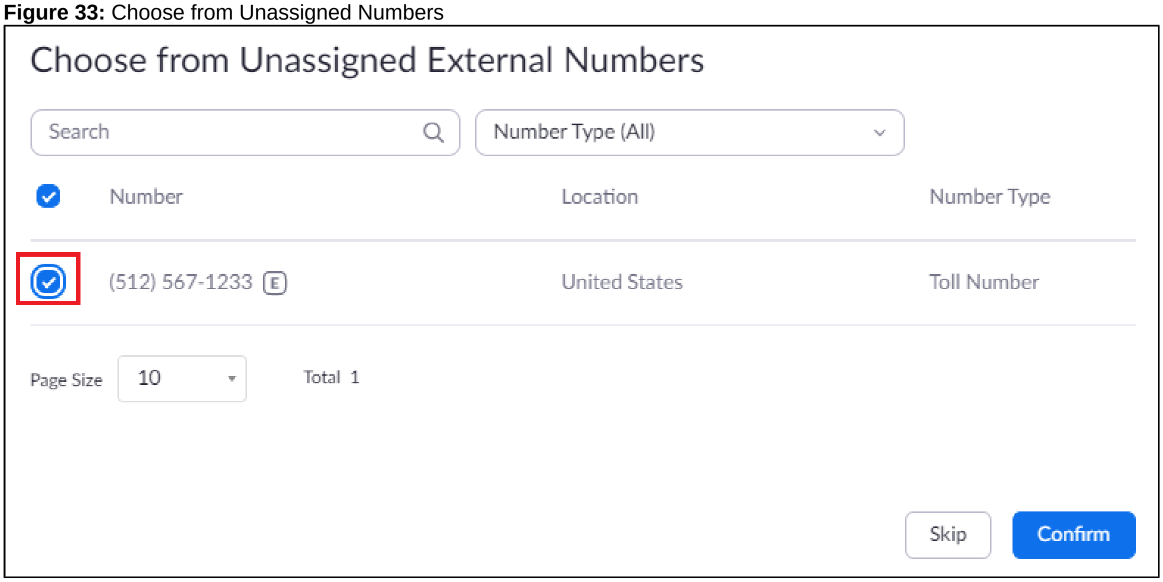


**Figure 32:** Assign BYOC calling plan



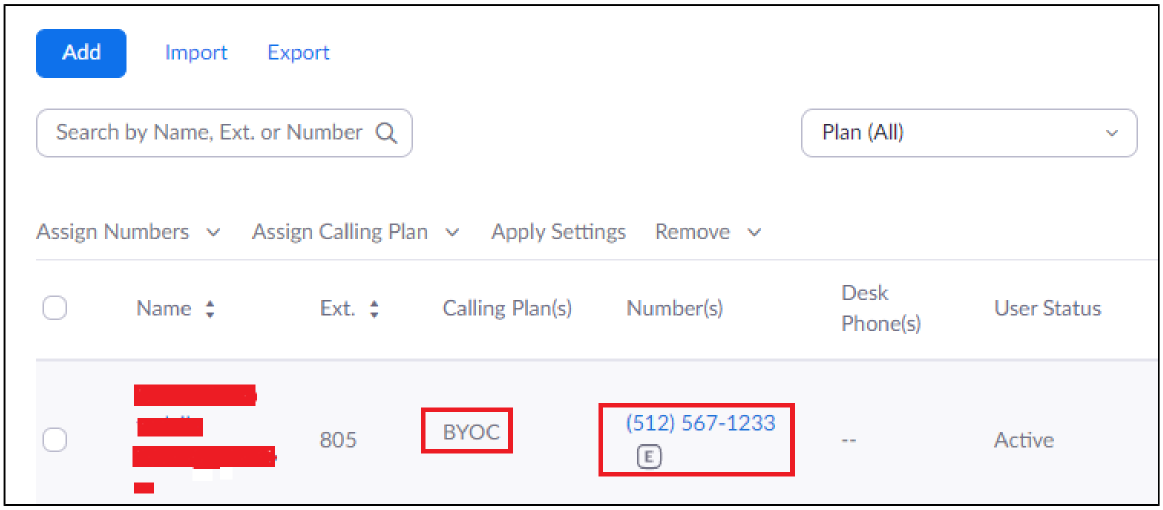

4. Assign the External Numbers created previously in the Add External Number section.

**Figure 33:** Choose from Unassigned Numbers




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




## 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/