A MITEL
PRODUCT
GUIDE

Zoom Phone with MiVoice Border Gateway and MiVoice Business

Solution Guide for Bring Your Own Carrier (BYOC) and Bring Your Own PBX (BYOP)

Document Version 1.1

February 2025

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History of Changes 1

Version	Date	Change
1.0	10-August-2020	Template design
1.1	17-February-2025	Updated the configuration guide for Mitel MiVoice Border Gateway and MiVoice Business configurations

Introduction 2

This chapter contains the following sections:

- Prerequisites
- Related documentation

This document outlines the process of connecting the **MiVoice Border Gateway (MBG)** and **MiVoice Business (MiVB)** to **Zoom Phone** using Bring Your Own Carrier (BYOC)[1] and Bring Your Own PBX (BYOP)[2] configurations.

This document provides instructions on how to set up **MiVoice Border Gateway (MBG)/MiVoice Business (MiVB)** for interoperability between Generic SIP Trunk and Zoom Phone environment. The interoperability compliance testing focuses on verifying inbound and outbound call flows between MiVoice Border Gateway (MBG)/MiVoice Business (MiVB) and Zoom cloud.

This hybrid integration model allows organizations to leverage Zoom's cloud platform while maintaining their existing MiVoice Business infrastructure for telephony features and PSTN connectivity. This solution is particularly valuable for organizations already using Zoom as their primary collaboration platform who want to preserve their investment in MiVoice Business for call management.

How it works:

The integration allows Zoom Phone to connect to the MiVoice Business system through a Generic SIP Trunk. MiVoice Business and MiVoice Border Gateway manage the communication between Zoom Phone and external networks, including the PSTN (Public Switched Telephone Network). MiVoice Business handles SIP message manipulation and call routing, ensuring proper communication between Zoom Phone and external networks (like PSTN). It also sets up signaling paths to Zoom Phone data centers and the SSP (PSTN provider), ensuring smooth call flow *to* and *from* Zoom Phone and the PSTN. Zoom Phone takes care of the cloud-based communication features, while MiVoice Border Gateway links Zoom Phone and your on-premises infrastructure, ensuring smooth integration.

This solution provides secure traffic management, allowing users to retain their MiVoice Business system while benefiting from Zoom's cloud features. Proper configuration of both MiVoice Business and MiVoice Border Gateway within the user environment is essential for successful deployment. Once MiVoice Business is configured, they can use the SBC to route calls, secure communication, and manage traffic between Zoom Phone and PSTN networks.

For detailed Zoom Phone settings and configuration, please refer to the official Zoom support page under the Settings and Configuration for Zoom Phone section and the following Zoom Web Portal Configuration on page 5.

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1 **Bring Your Own Carrier (BYOC):** Connecting your existing telecom provider (carrier) to Zoom Phone.
2 **Bring Your Own PBX (BYOP):** Integrating your existing phone system (PBX) with Zoom Phone.

Important:
Initial releases of MiVoice Border Gateway, for Zoom DO NOT require a Zoom BYOC/BYOP license. However, this license will be required for future releases. During this transition, MiVoice Border Gateway Zoom BYOC/BYOP licenses will NOT BE NEEDED as part of the Zoom subscription.

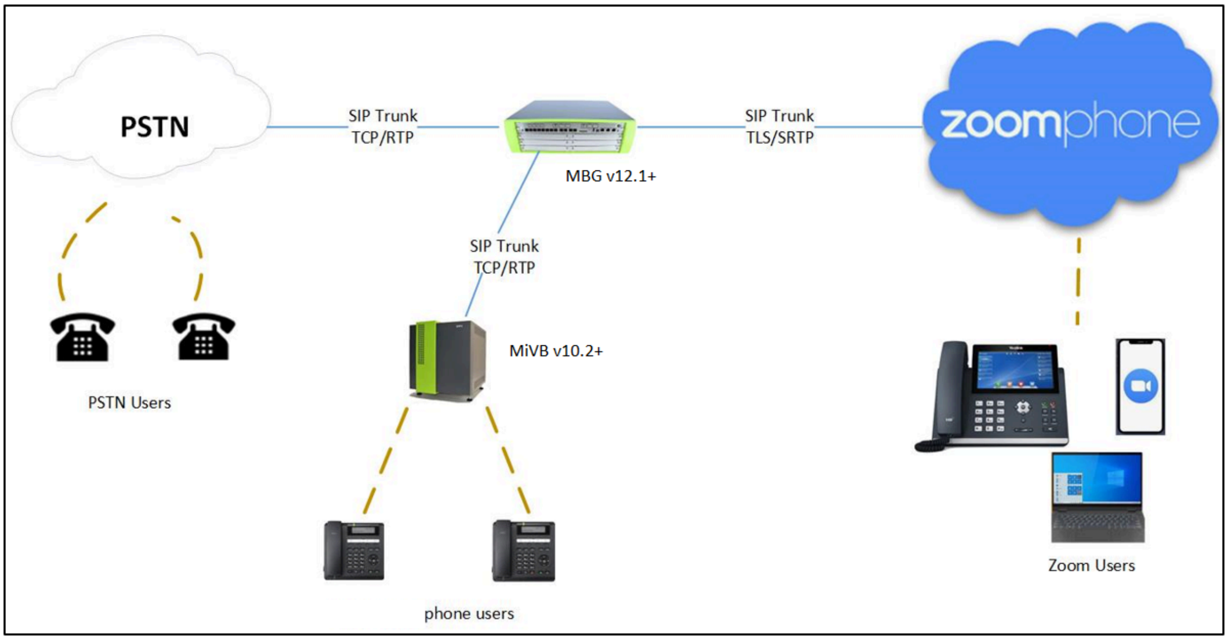


Figure 1: Network Topology Block Diagram

2.1 Prerequisites

Supported product versions

Product	SW Version (minimum)
Zoom Workplace app	6.2.0
MiVoice Border Gateway	12.1.0.110
MiVoice Business	10.2.0.54

2.2 Related documentation

- For additional information on MiVoice Border Gateway (MBG), refer MiVoice Border Gateway documentation.
- For additional information on **MiVoice Business (MiVB)** , refer MiVoice Business documentation.
- For additional information on the Zoom Configurations, refer to the official Zoom Support page.

Zoom Web Portal Configuration 3

This chapter contains the following sections:

- Adding Your SBC
- Adding Phone Users
- Adding BYOC Phone Numbers
- Adding BYOP numbers

This section guides you in preparing the environment for integrating and operating with external Bring Your Own Carrier (BYOC) DID phone numbers. It also explains how to add these numbers and map them to the corresponding endpoint devices, such as IP phones and other SIP devices.

Important:
Initial releases of MiVoice Border Gateway for Zoom DO NOT require a Zoom BYOC/BYOP license. However, this license will be required for future releases. During this transition, MiVoice Border Gateway Zoom BYOC/BYOP licenses will NOT BE NEEDED as part of the Zoom subscription.

To set up users for the Zoom and MiVoice Business integration, you must first add users to your Zoom account and assign licenses to them.

3.1 Adding Your SBC

Follow the instructions below to add your SBC in the Zoom Web Portal.

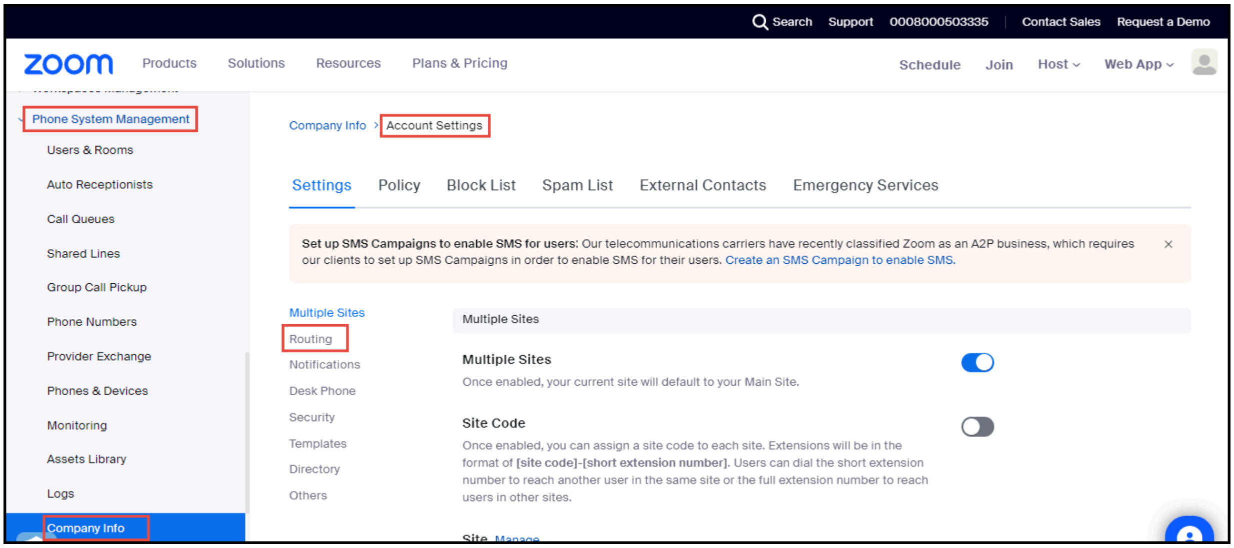
Prerequisites

1. You are an administrator.
2. You have completed the initial Zoom Phone setup.
3. You have configured appropriate firewall rules for connectivity. For more information, refer to Zoom network firewall or proxy server settings.
4. You have a public IP address for SIP trunk connectivity.

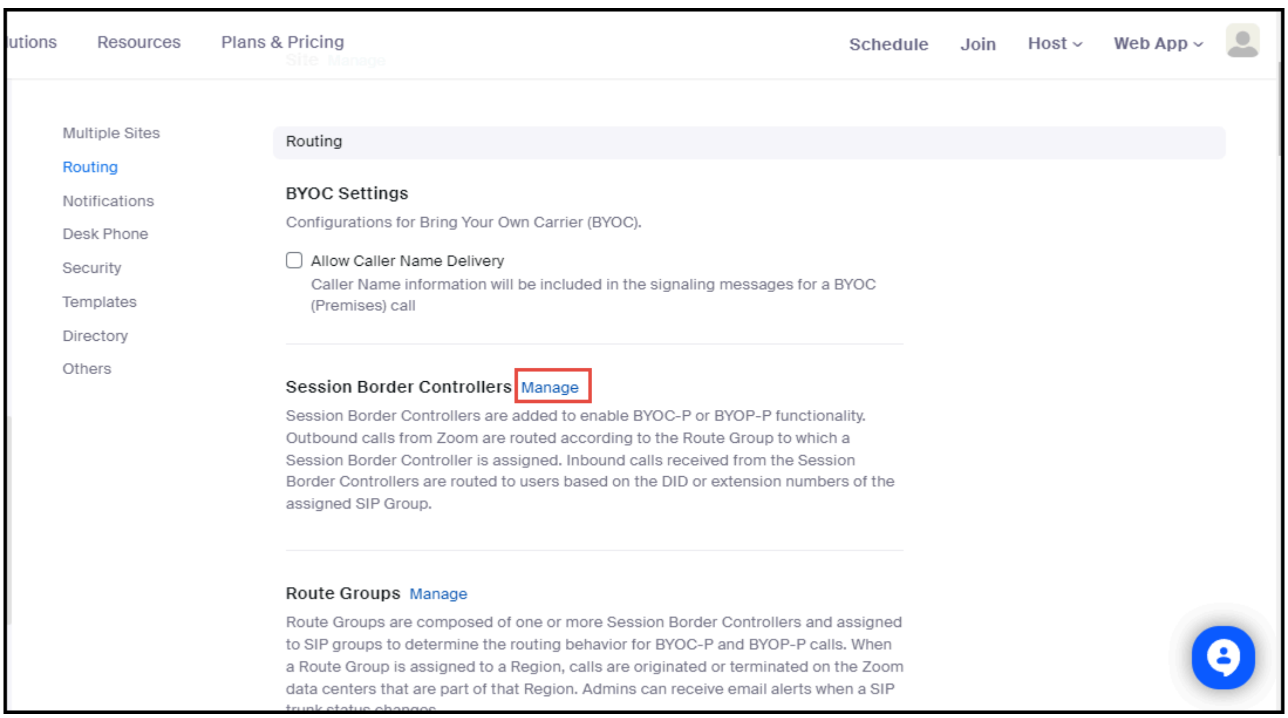
Adding your SBC

1. Log in to the **Zoom Admin Portal**.

2. Navigate to **Phone System Management** > **Company Info** > **Account Settings** > **Routing.**



3. Locate **the Session Border Controllers** section and click **Manage**.

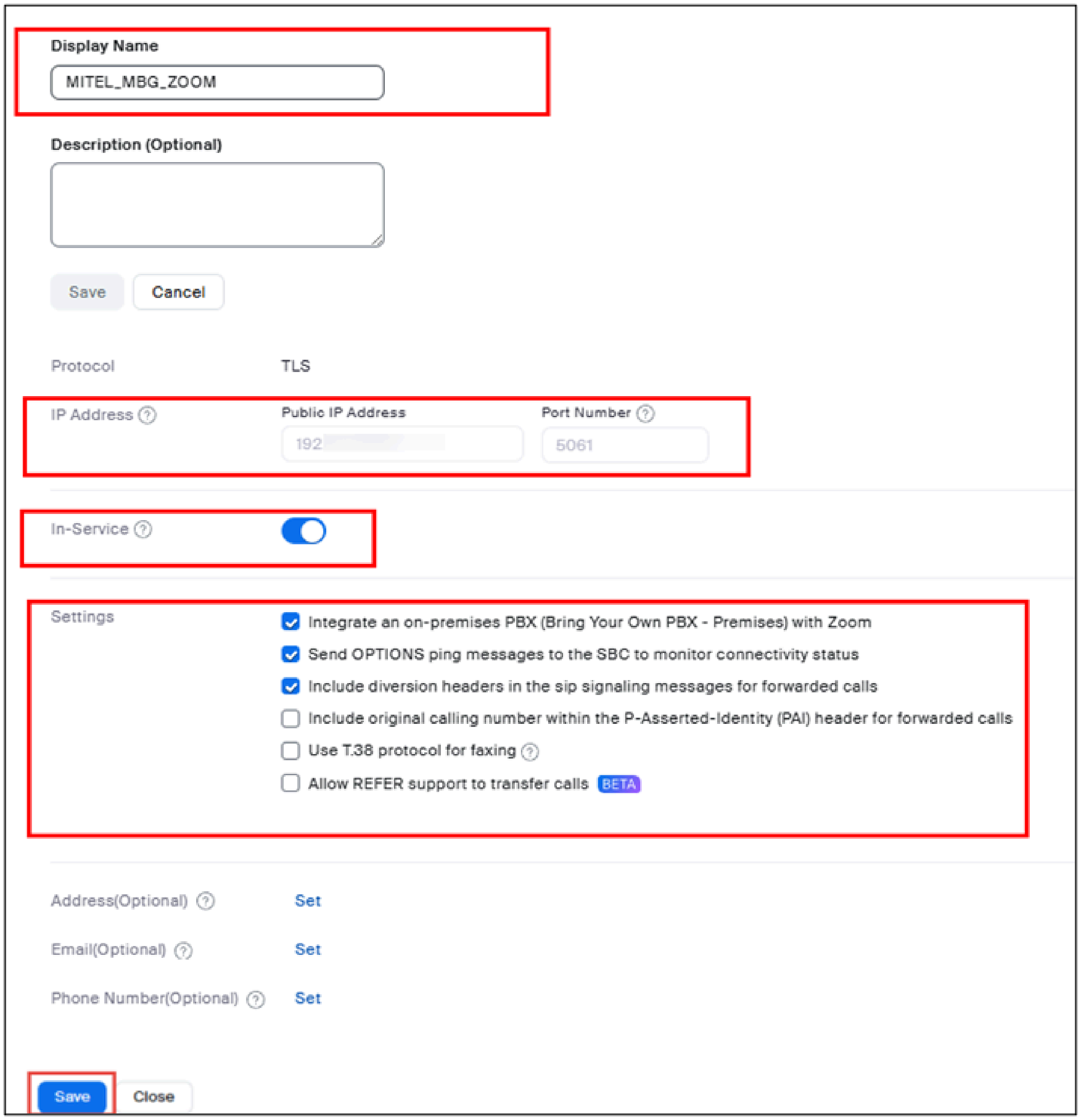


4. Click **Add.**

5. Configure the following:

   a. **Display Name:** Type the display name of your choice. For example, **Mitel_MBG_ZOOM**.
   b. **IP Address:** Enter the IP address of the MBG interface facing towards Zoom and configure the port number (for example, 5061).
   c. **In-Service:** Click the toggle button to enable the **In-Service option**.
   d. Under the **Settings** section, check the following check boxes**:**

      - **Integrate an on-premises PBX (Bring Tour OWN PBX-Premises) with Zoom**
      - **Send OPTIONS ping messages to the SBC to monitor connectivity status**
      - **Include diversion headers in the sip signaling messages for forwarded calls**



6. Click **Save**.

Note:

To ensure Zoom's network allows traffic from your MBG, contact your **Zoom representative** to **whitelist** the SBC's **IP address** and **port** in Zoom's **Access Control Lists (ACLs)**. Once the **whitelisting** is done, you can start sending traffic (i.e., calls or data) between your system and Zoom.

Use **SIP OPTIONS** to check that the connection between your SBC and Zoom is working correctly after the transport is established.

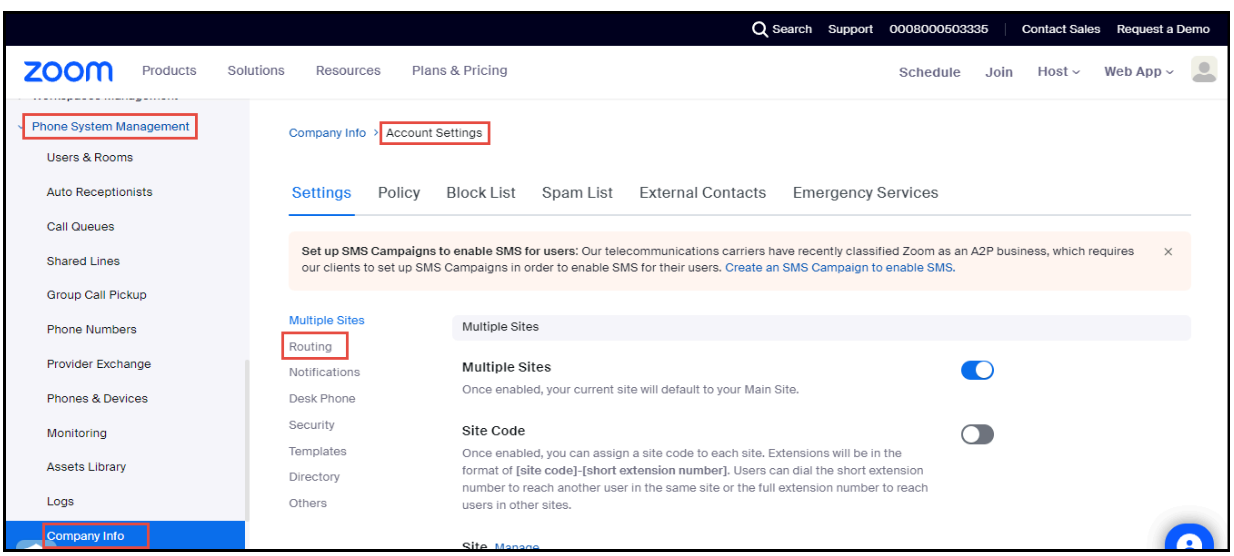
3.1.1 Configuring the Route Group

Route Groups are collections of Session Border Controllers (SBCs), which manage and route voice traffic across a network. A Route Group determines how calls are routed and handled by directing them to specific SIP endpoints. The **Region** setting ensures that calls are routed through the appropriate Zoom data centers based on their geographic location.

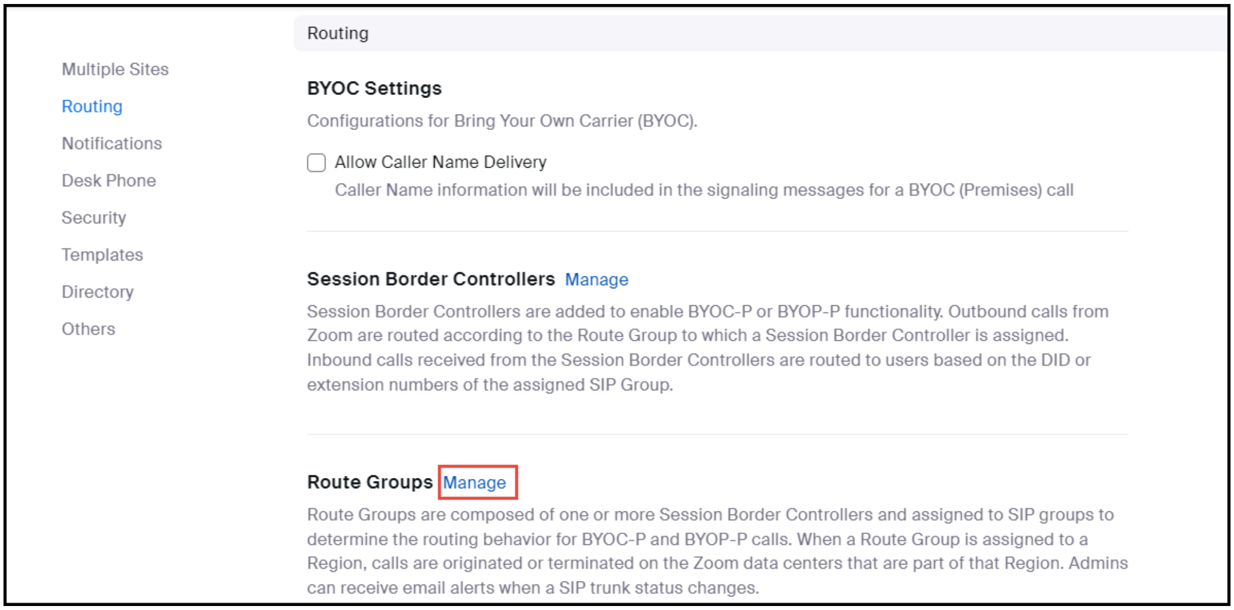
Note:
These configurations (Route Group, SIP Group, and Routing Rule) will take effect once phone numbers are added and assigned to the appropriate users. Until then, the routing logic will be in place, but calls will not be routed as expected.

To add a Route Group:

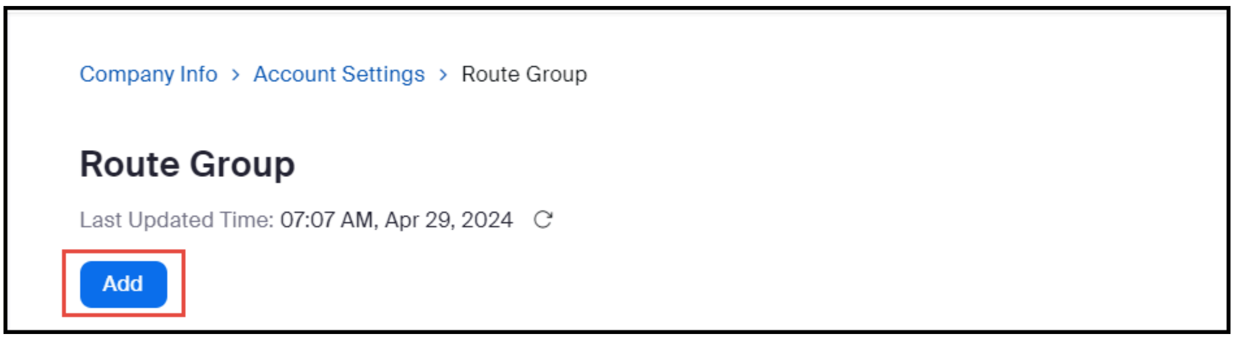
1. Navigate to **Phone System Management** > **Company Info** > **Account Settings** > **Routing**.



2. Locate the **Route Groups** section and click **Manage.**



3. Click **Add.**

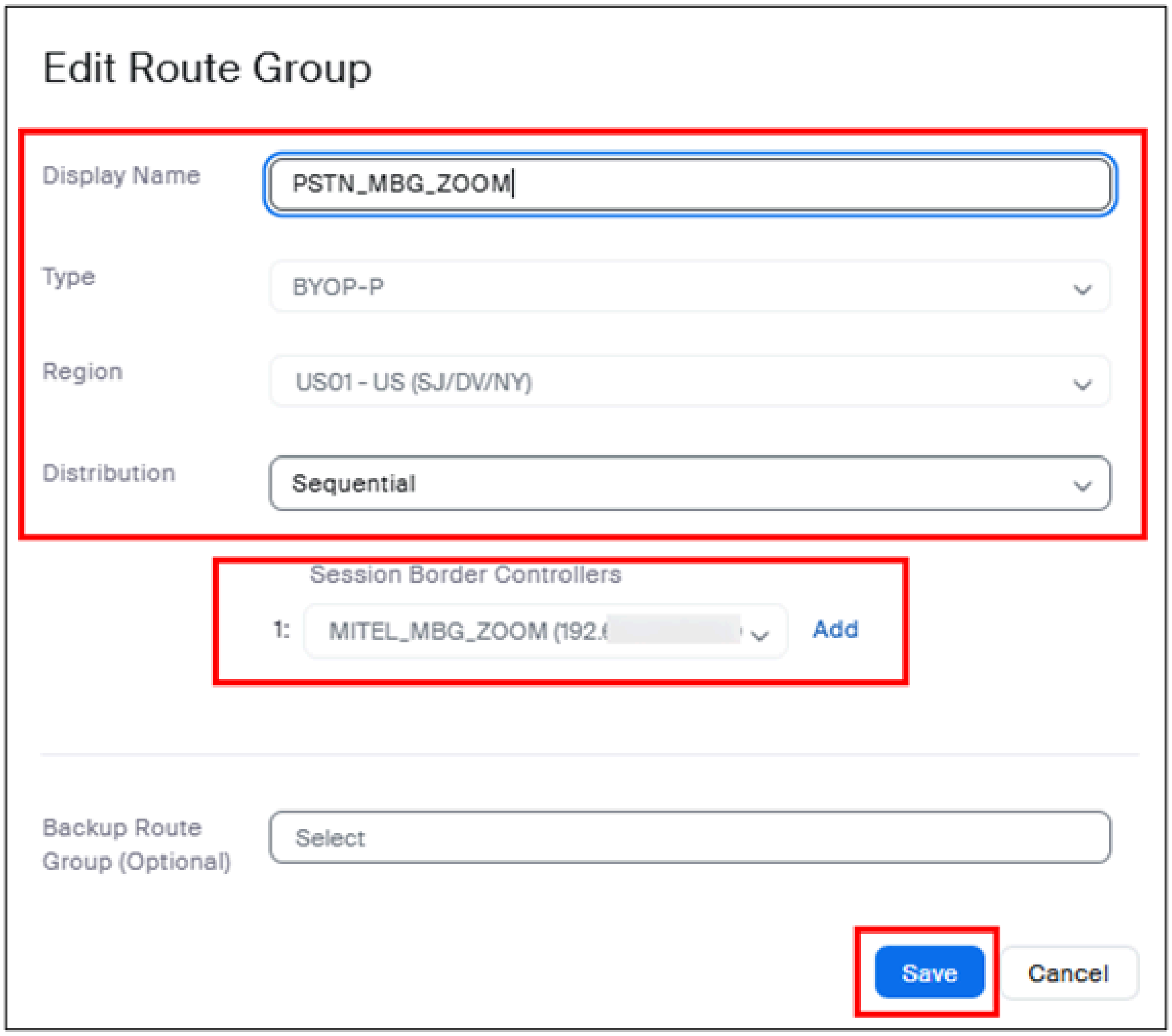


4. Configure the following:

   a. **Display Name:** Type the display name of your choice. For example, **PSTN_MBG_ZOOM**.
   b. From the **Type** drop-down menu**,** select **BYOC-P**.
   c. From the **Region** drop-down menu, select the region code for your location. The format will be similar to: **US01-US(SJ/DV/NY)**

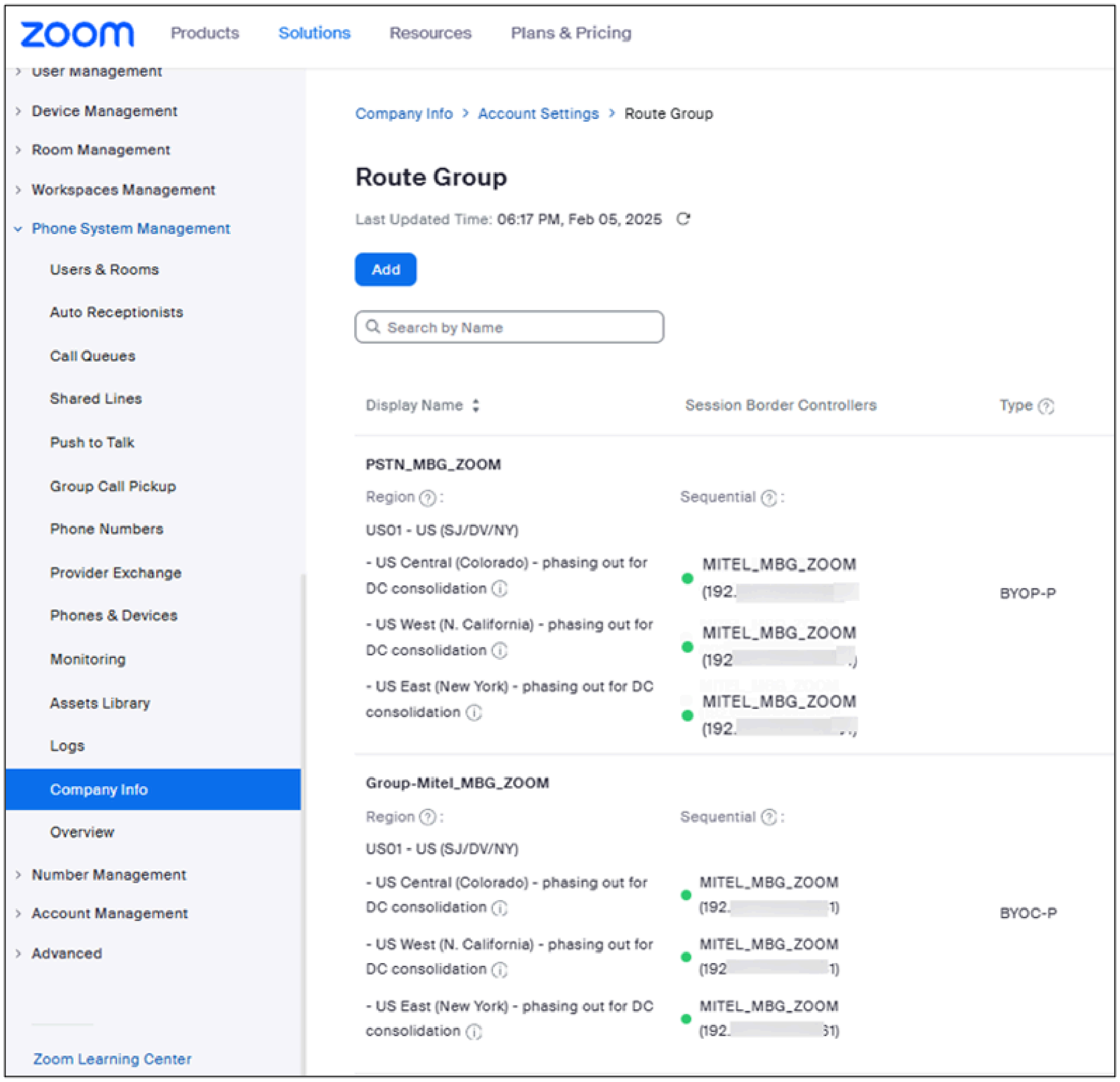
Note:
The format given above is an example.Choose the zone (SJ/DV/NY etc.) that is geographically closest to your SBC installation location.

d. From the **Distribution** drop-down menu, select **Sequential** and then from the **Session Border Controllers** drop-down menu, select the MITEL_MBG_ZOOM that was created in Adding Your SBC section.

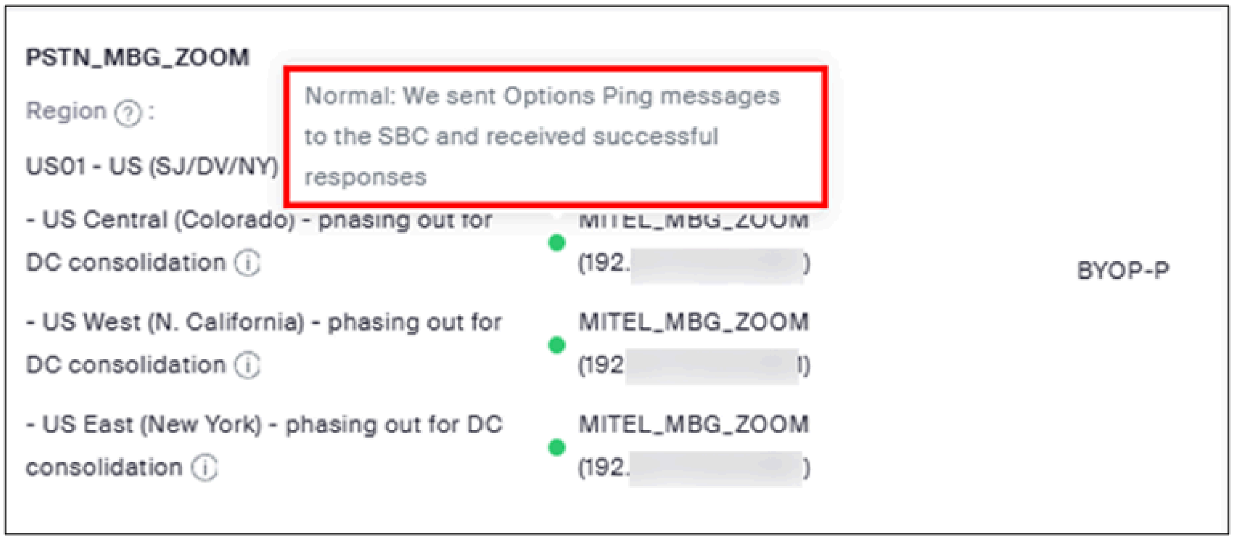


5. Click **Save**.

A green light indicates that the trunk status is active, as shown below:



6. Optional: Hover over the green LED icon to view the trunk status, as shown below:

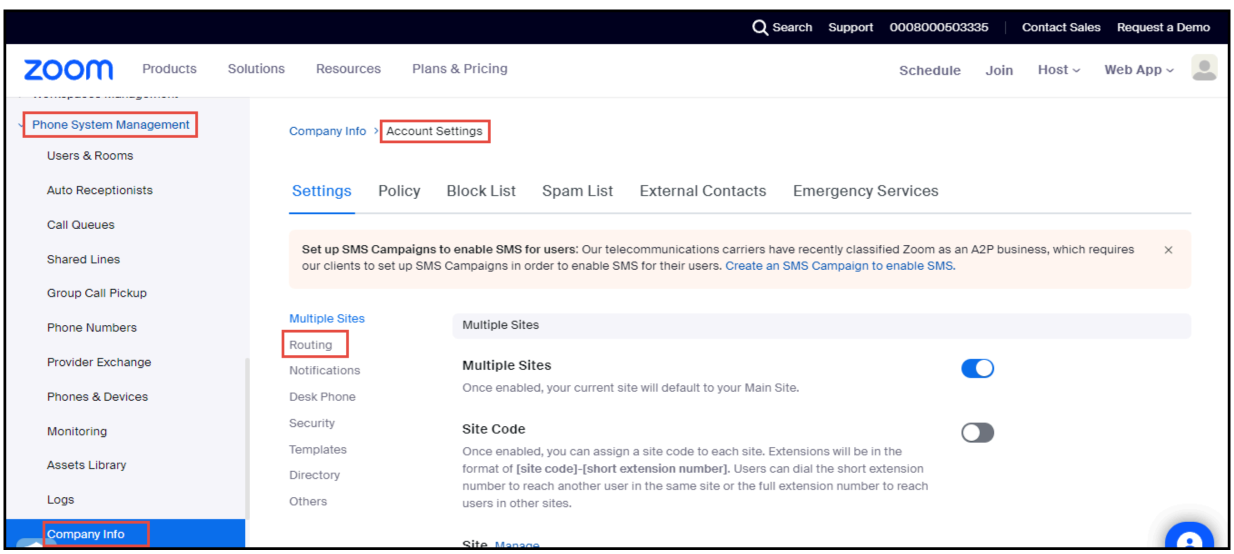


3.1.2 Configuring the SIP Group

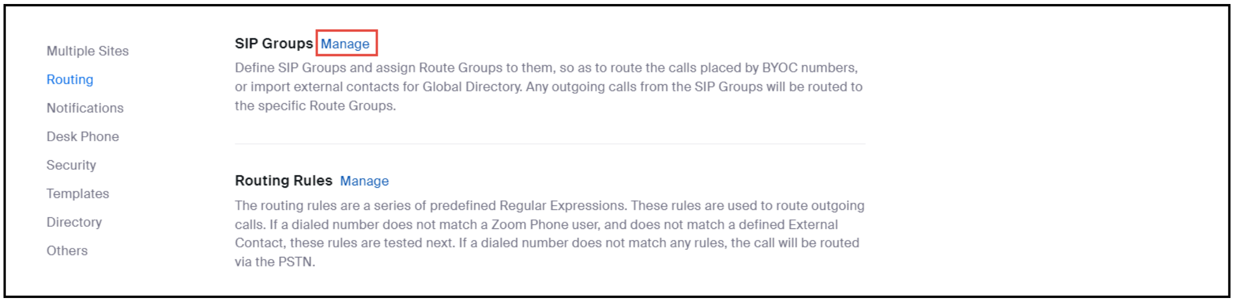
Follow the instructions below to configure SIP groups and assign Route Groups to them, in order to route calls placed by BYOC numbers. This step is mandatory for uploading the BYOC numbers.

To add a SIP Group:

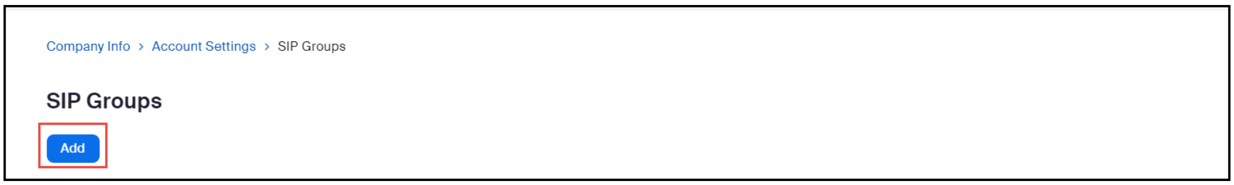
1. Navigate to **Phone System Management** > **Company Info** > **Account Settings** > **Routing**.



2. Locate the **SIP Groups** section and click **Manage.**



3. Click **Add.**



4. Configure the following:

   a. **Display Name:** Type the display name of your choice. For example, **SIP_Group_Mitel_MBG_Zoom.**
   b. From the **Route** drop-down menu**,** select the **Group-Mitel_MBG_ZOOM (BYOC)** group, created in Configuring the Route Group section.

5. Click **Save.**

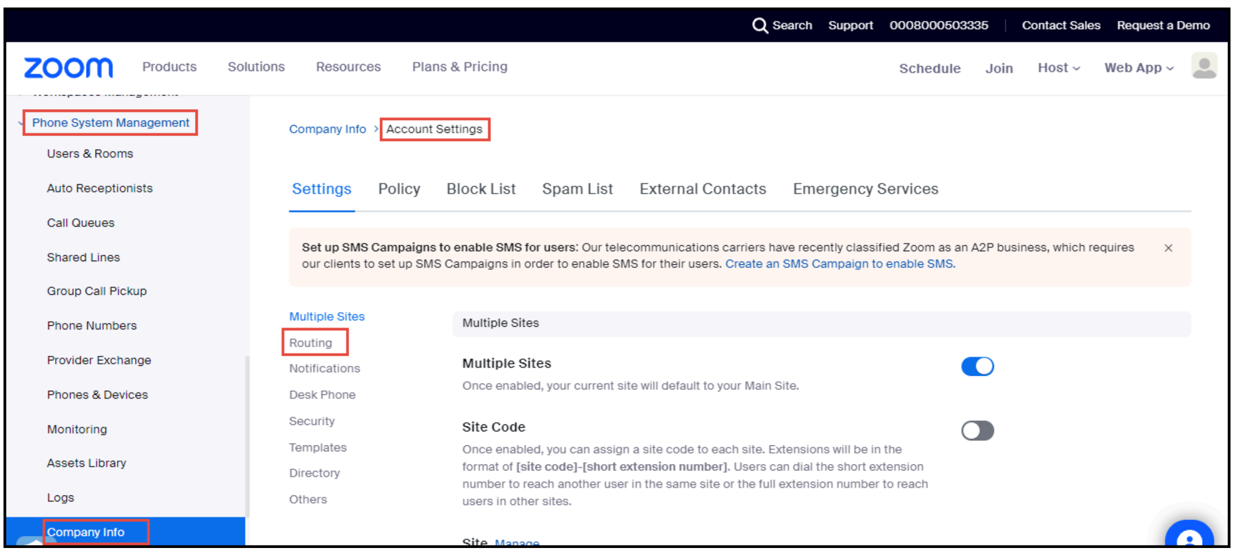
3.1.3 Configuring the Routing Rule

When configuring a **BYOC (Bring Your Own Carrier)** setup, you might create a routing rule to specify that calls from certain users or departments go through your MBG or network route.To add a Routing Rule for outbound calls:

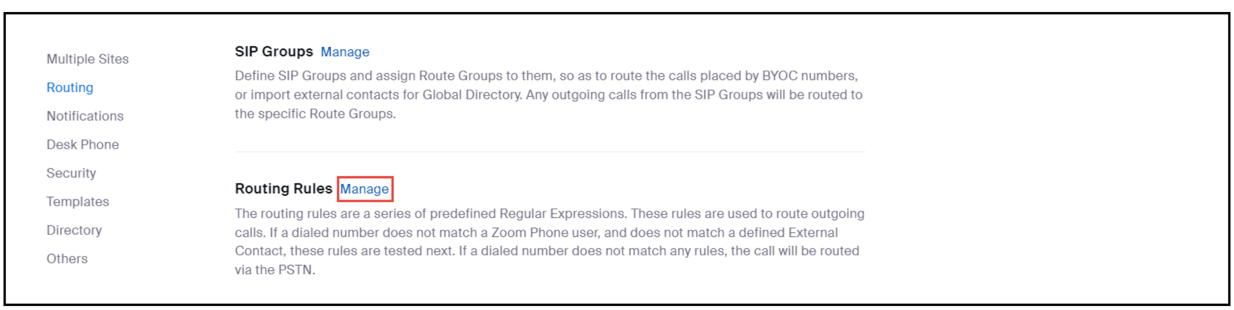
> **Note**:
> Ensure that your Session Border Controller is properly configured and connected before setting up routing rules. Additionally, phone users must be provisioned and assigned to the correct phone numbers for routing rules to function correctly.

To add the Routing Rule:

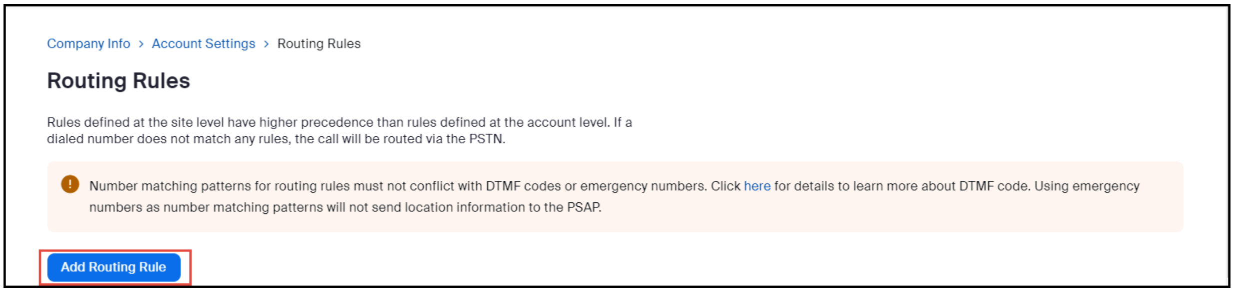
1. Navigate to **Phone System Management** > **Company Info** > **Account Settings** > **Routing**.



2. Locate the **Routing Rule** section and click **Manage.**

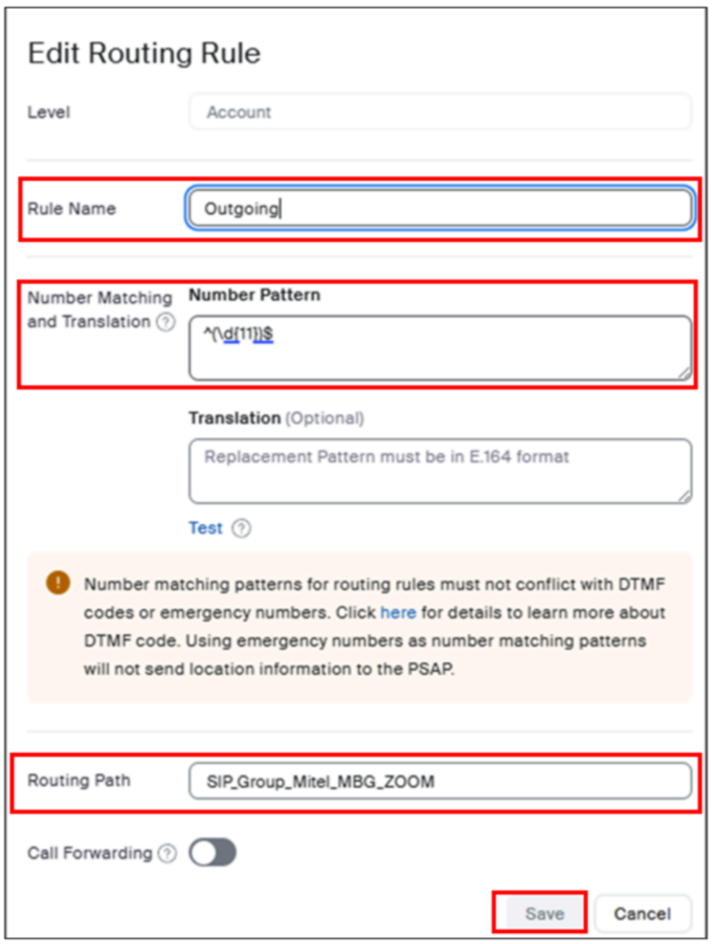


3. Click **Add Routing Rule** to add your rule for outbound calls.



4. Configure the following:

   a. **Rule Name:** Type the rule name of your choice. For example, **Outgoing**.
   b. **Number Matching and Translation:** Enter the **`^(\d{11})$`** Number Pattern (as given below)
   c. **Routing path:** Select the **SIP_Group_Mitel_MBG_ZOOM** routing path, created in Configuring SIP Group section.



5. Click **Save**.

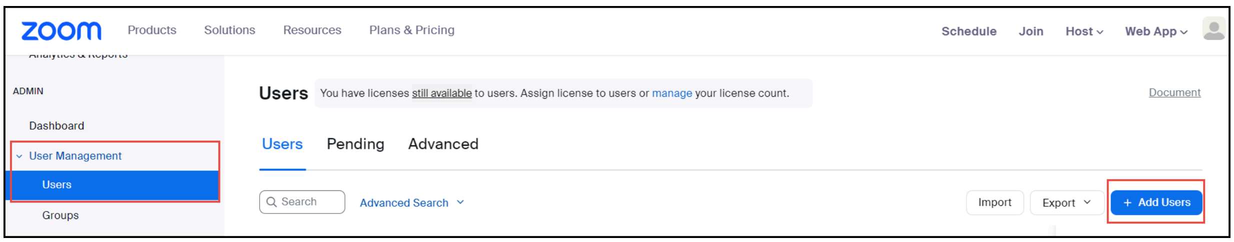
3.2 Adding Phone Users

Follow the instructions below to add Zoom Phone Users. For more details, please refer to the official Zoom support page on How to add a new user.

Prerequisites

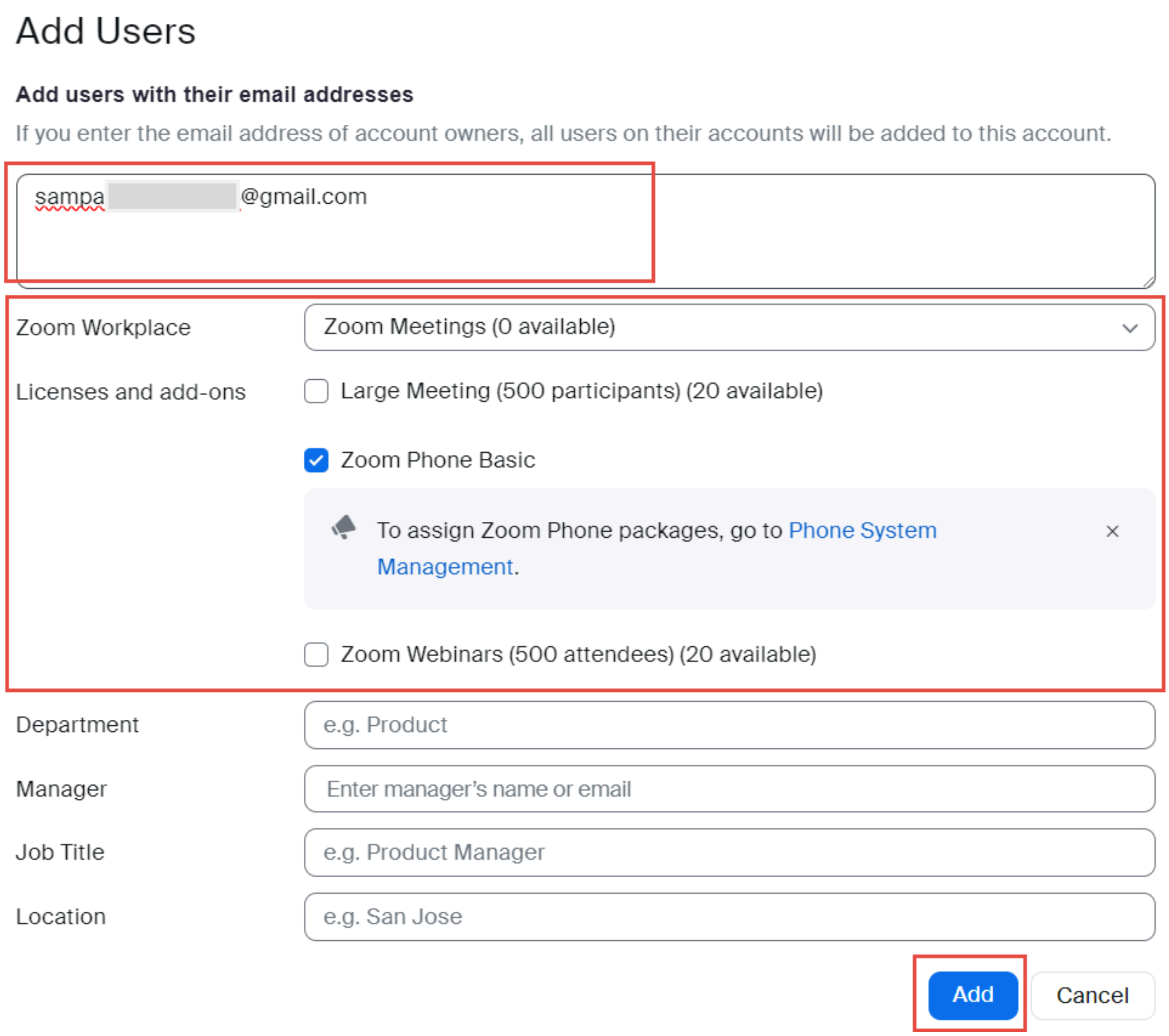
1. You have a Pro, Business, or Enterprise Zoom Phone account.
2. You are an administrator with the privilege to edit account settings.
3. You have completed the initial Zoom Phone setup. For more information, refer to Getting started with Zoom Phone (admin).

1. Log in to the **Zoom web portal**.
2. Navigate to **User Management** > **Users** > **Add Users.**



3. Configure the following in the **Add Users** pop-up:

   a. Enter the user's email address**.** To add multiple users with the same settings, enter multiple email addresses separated by commas: **,** .
   b. From the **Zoom Workplace** drop-down menu, select the available Zoom Workplace licenses to assign, such as **Zoom Meetings**.
   c. In the **Licenses and add-ons** section, check the **Zoom Phone Basic** checkbox.
   d. Click **Add**.



The new user(s) will appear on the **Pending** tab of the User Management section.

Next steps

You can now assign licenses to users. After purchasing your Zoom One licenses, during the setup of Zoom Phone for your account, you can choose either to assign Zoom Phone packages automatically or manually to your Zoom One users. Before assigning a license to a phone user, ensure that automatic phone assignment for Zoom One licenses is disabled for your account. For more information, refer to the official Zoom support page.

With automatic assignment disabled, you can proceed to assign licenses to the phone user(s). For more information, refer to How to assign Zoom licenses.

3.2.1 Assigning a Calling Plan to a phone user

You can assign a calling plan to phone users to enable outbound calling.

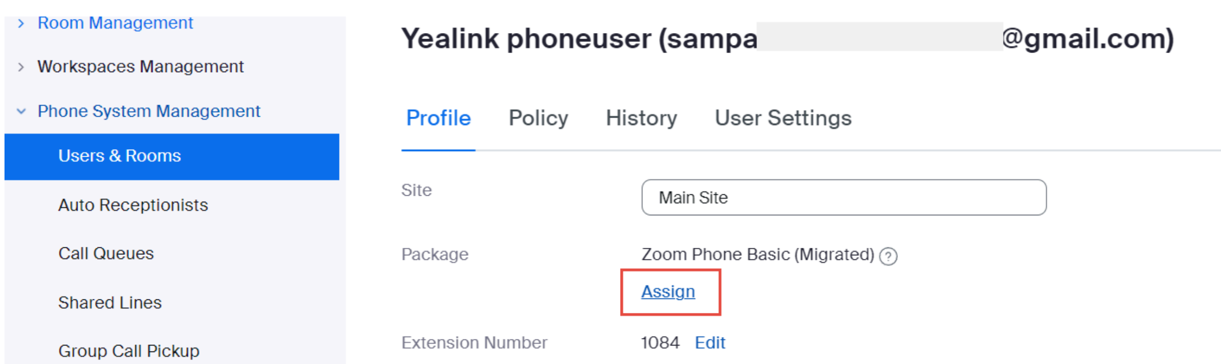
Prerequisite

1. You are an administrator with the privilege to edit account settings.
2. You have assigned licenses to the phone users. For more information, refer to How to assign licenses.

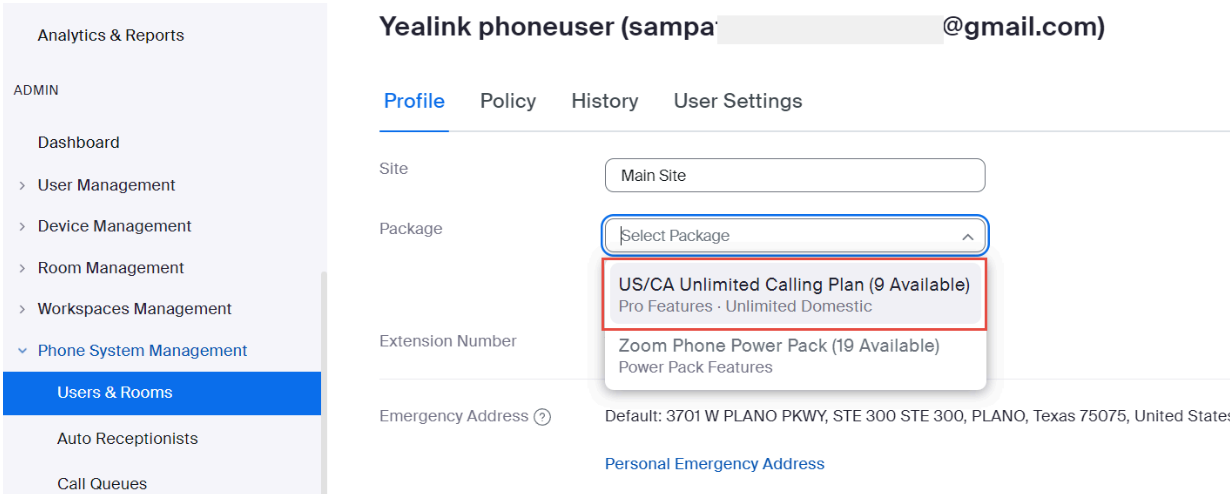
1. Navigate to **Phone System Management** > **Users & Rooms**.
2. Select the user for whom you want to add a calling plan and click **Assign**.

> **Note**:
> When a Zoom Phone license is assigned to a user, an extension number is automatically assigned to them.

3. Under the **Profile** tab, locate the **Package** section and click **Assign**.



4. From the **Package** drop-down menu, select **US/CA Unlimited Calling Plan**, as shown below.



5. Click **Confirm**.

Note:
If you do not add a calling plan package for the user, you will not be able to make outgoing calls through the SIP Trunk for BYOC/BYOP.

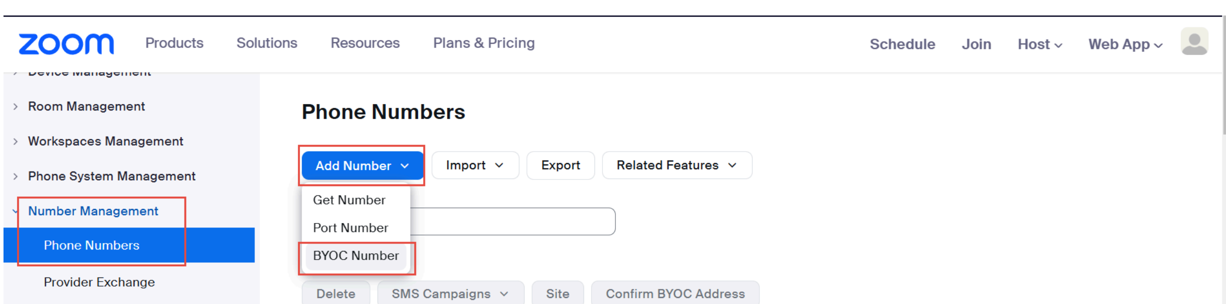
3.3 Adding BYOC Phone Numbers

You can add the BYOC phone numbers as shown below.

Prerequisite

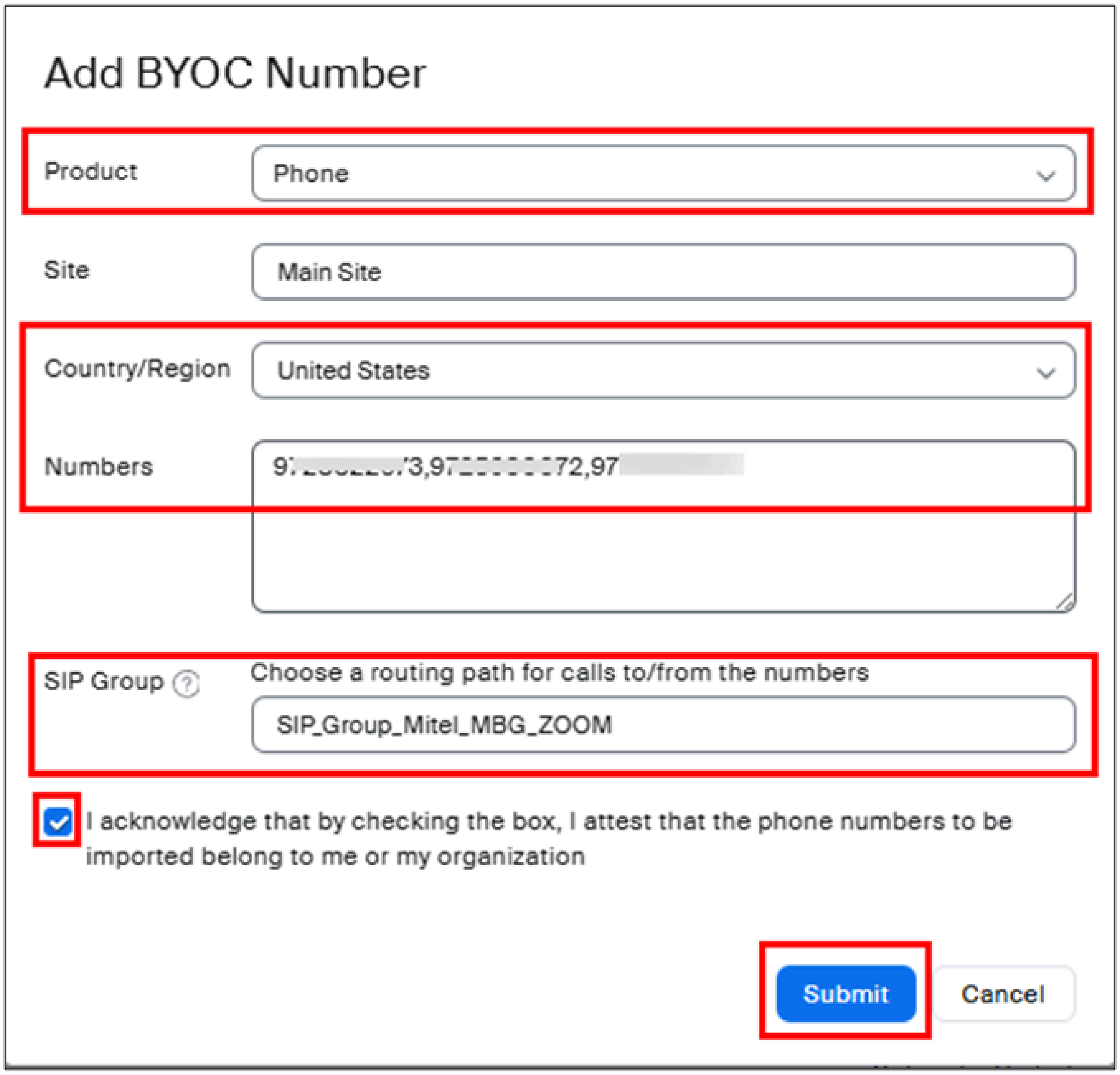
1. You are an administrator with the privilege to edit account settings.

1. Log in the **Zoom web portal**.
2. Navigate to **Number Management** > **Phone numbers.**
3. From the **Add Number** drop-down menu, select **BYOC Number.**



4. In the **Add BYOC Numbers** window:

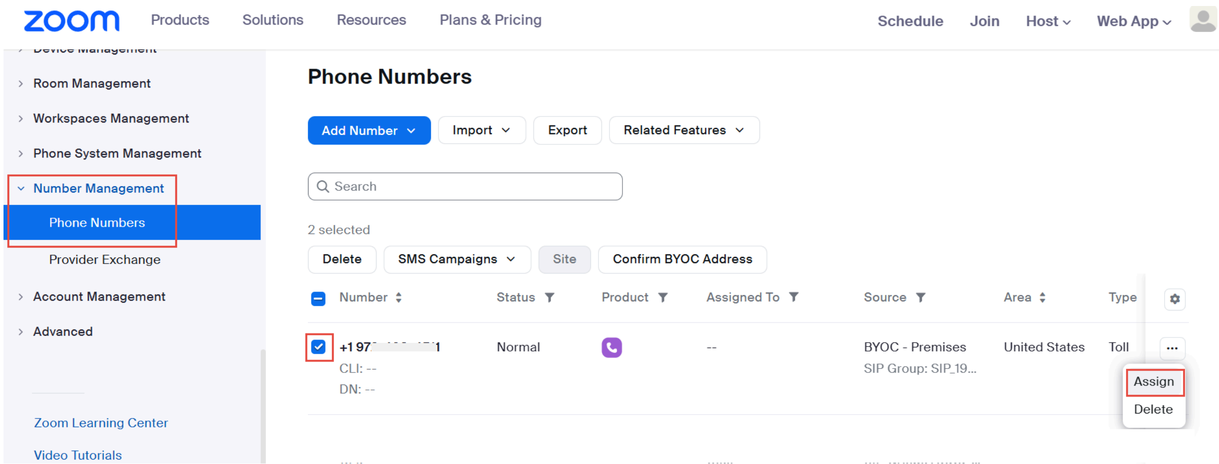
   a. From the **Product** drop-down menu, select **Phone**.
   b. From the **Country/Region** drop-down menu, select the country to which the phone numbers belong. For example, United States.
   c. In the **Numbers** field, enter the phone numbers separated by **' , '**, as shown in the image below.
   d. From the **SIP System** drop-down menu, select **Zoom Phone**.
   e. From the **SIP Group** drop-down menu**,** select the SIP_Group_Mitel_MBG_ZOOM which was created in Configuring the SIP Group section.
   f. Check the acknowledgment box to consent.
   g. Click **Submit**.



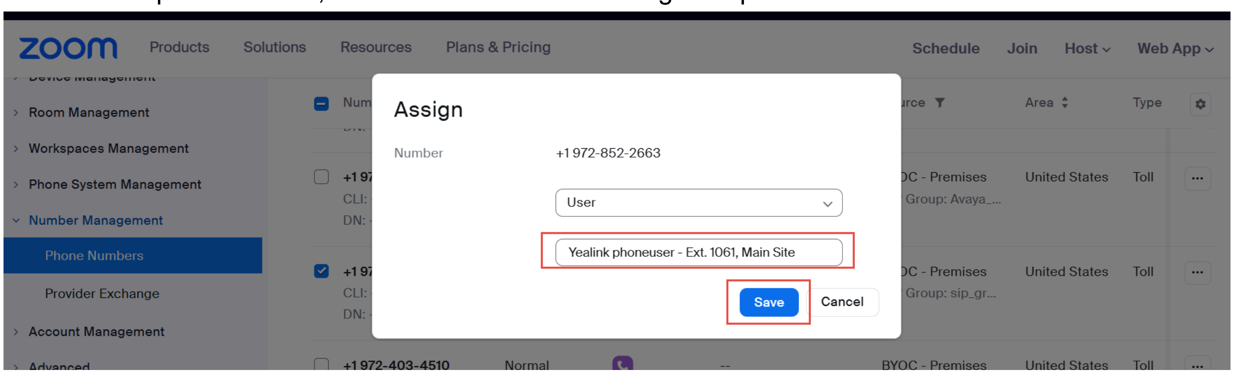
Assigning BYOC numbers

To assign Bring Your Own Carrier (BYOC) numbers to the Zoom phone users:

1. Navigate to **Number Management** > **Phone Numbers.**
2. Select the **phone number** that needs to be assigned to the Zoom phone user.
3. Click **Assign**.



4. From the drop-down menu, select an extensions to assign the phone number to and click **Save**.



The phone number will be assigned to the selected user.

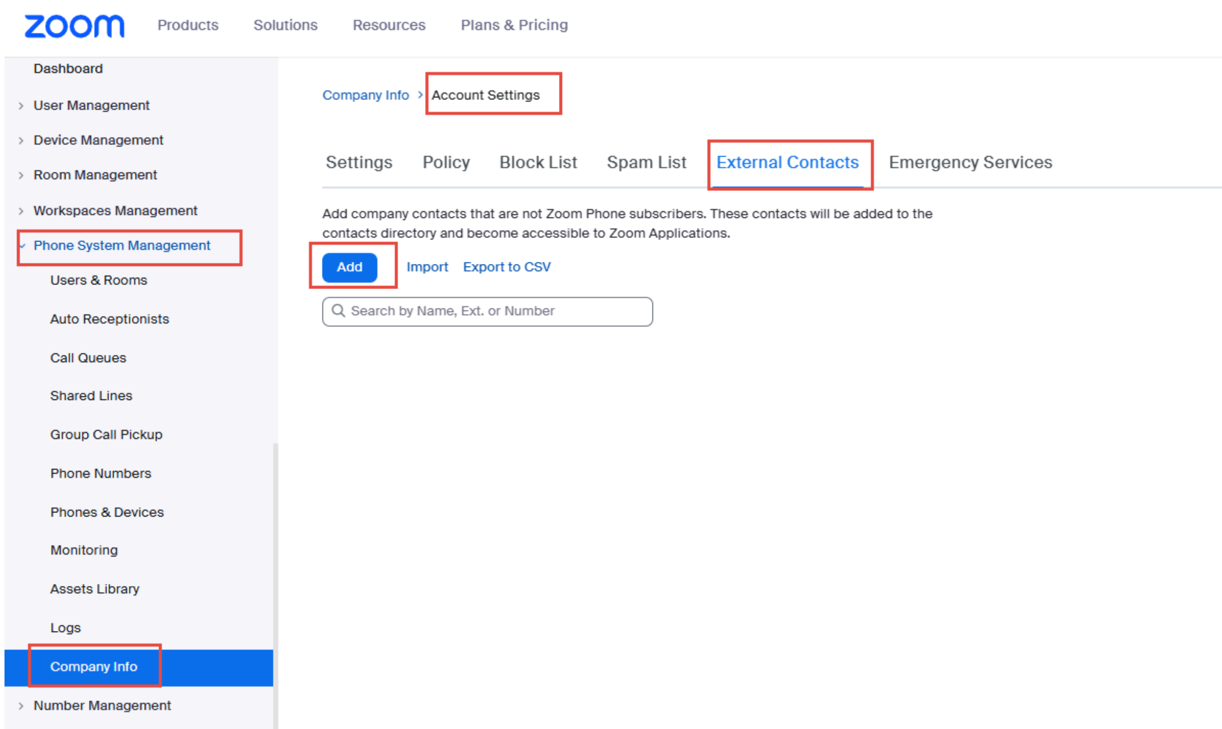
3.4 Adding BYOP numbers

Administrators can add MBG Voice users as External contacts which will be added to the contacts directory and be accessible to Zoom applications.

To add Bring Your Own PBX (BYOP) numbers:

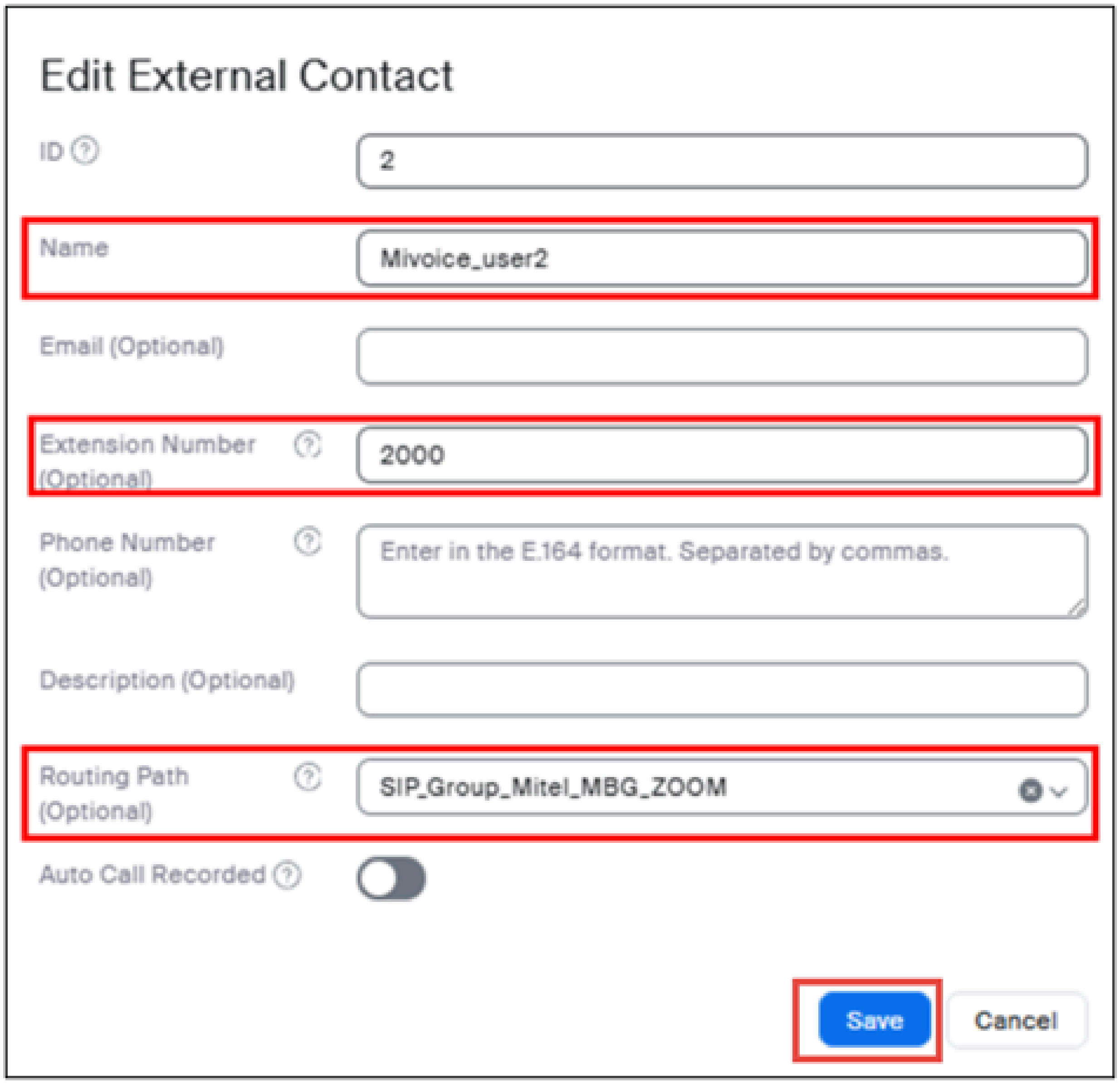
1. Navigate to **Phone System Management** > **Company Info** > **Account Settings** > **External Contacts.**

2. Click **Add.**


zoom
Products
Solutions
Resources
Plans & Pricing
Dashboard
User Management
Device Management
Room Management
Workspaces Management
Phone System Management
Users & Rooms
Auto Receptionists
Call Queues
Shared Lines
Group Call Pickup
Phone Numbers
Phones & Devices
Monitoring
Assets Library
Logs
Company Info
Number Management
Company Info > Account Settings
Settings
Policy
Block List
Spam List
External Contacts
Emergency Services
Add company contacts that are not Zoom Phone subscribers. These contacts will be added to the contacts directory and become accessible to Zoom Applications.
Add
Import
Export to CSV
Search by Name, Ext. or Number

3. In the **Edit External contact** pop-up, configure the following:

- **Name:** Type the name of the MiVB Voice user. For example, **Mivoice_user2**
- In the **Extension Number** field, enter the extension number of the MiVB Voice user.
- From the **Routing path** drop-down menu**,** select the **SIP Group** "SIP_Group_Mitel_MBG_ZOOM" created in **Configuring the SIP Group on page 12** section.



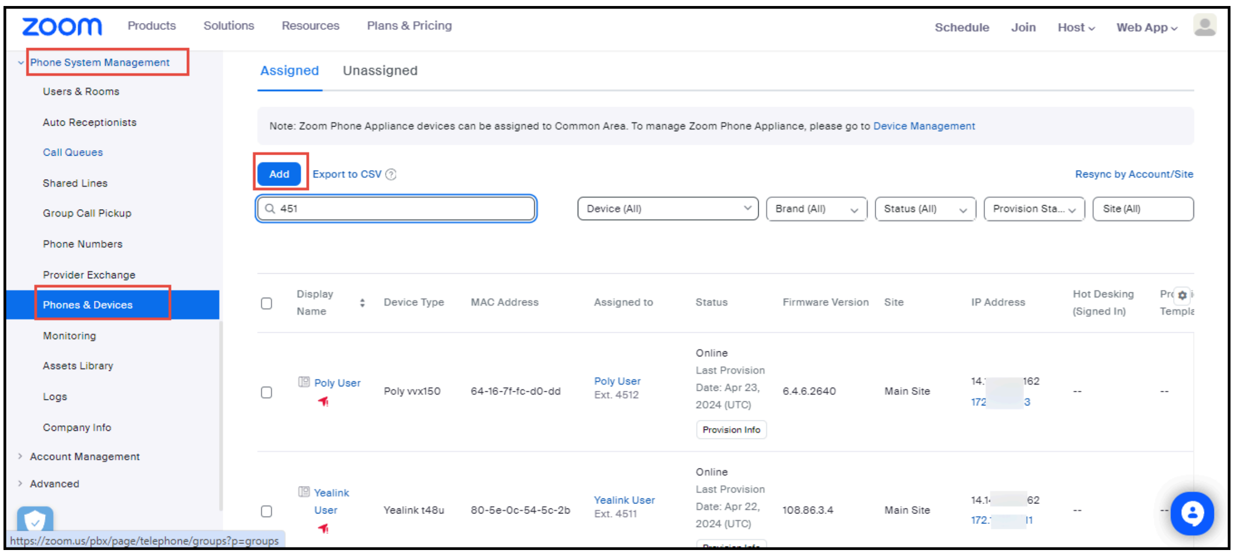
4. Click **Save**.

Provisioning Desk Phones for Zoom Phone users

4

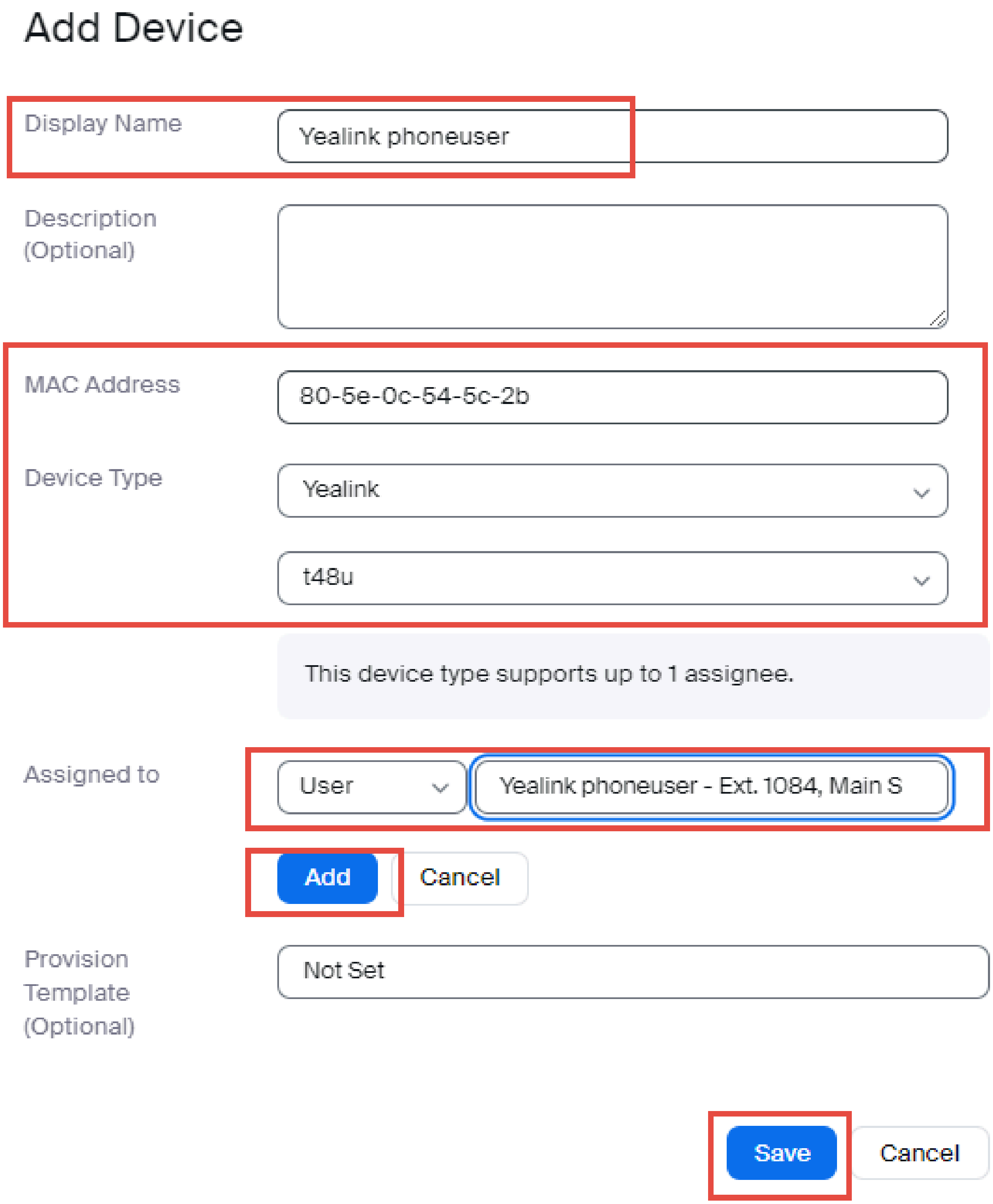
Follow the instructions below to provision Desk phones for Zoom Phone users. Zoom-certified vendor phone models are used for this test and will be available after provisioning.

1. Navigate to **Phone System Management** > **Phones & Devices.**
2. Click **Add.**



3. In the **Add Device** pop-up, configure the following:

   a. **Display Name:** Type the display name for the phone. For example, **Yealink phoneuser**.
   b. **MAC Address:** Enter the **MAC address** of the phone.
   c. **Device Type:** Select the device type. For example, **Yealink t48u**.
   d. From the **Assigned to** drop-down menu, select the user to whom you want to assign the phone number and click **Add**.
   e. Click **Save**.


Add Device
Display Name
Yealink phoneuser
Description (Optional)
MAC Address
80-5e-0c-54-5c-2b
Device Type
Yealink
t48u
This device type supports up to 1 assignee.
Assigned to
User
Yealink phoneuser - Ext. 1084, Main S
Add
Cancel
Provision Template (Optional)
Not Set
Save
Cancel

MBG/MiVB Configuration: BYOP/BYOC 5

This chapter contains the following sections:

- MBG Configuration for BYOP/BYOC
- MiVB Configuration for BYOP/BYOC

The **MiVoice Border Gateway (MBG)** is a highly scalable SBC solution supporting broad SIP interoperability, advanced media handling and robust security. The MBG enables enterprises to deliver voice services, such as SIP trunking and unified communications. The **MiVoice Business IP-PBX (MiVB)** provides the call routing and number modification facilities. The SIP signaling for incoming and outgoing calls to Zoom clients will always pass through the MiVoice Business.

5.1 MBG Configuration for BYOP/BYOC

This chapter provides a comprehensive guide to configuring the MiVoice Border Gateway (MBG) for seamless inter-working with Zoom Direct Routing. To add support for Zoom Bring Your Own Provider (BYOP) or Bring Your Own Carrier (BYOC), SIP Trunks need to be configured on the MBG and MiVoice Business (MiVB) system.

The MBG should be set up in accordance with the customer's desired network configuration. It may be deployed in a **Demilitarized Zone (DMZ)** or operate in **Server/Gateway mode**, depending on the network requirements and security policies of the organization.

In section Adding Your SBC on page 5, the MBG IP address was configured into Zoom system.The number of SIP Trunks required depends on the regional setup within the Zoom account. Based on the regions defined, it may be necessary to establish **two or more SIP Trunks** connecting to the IP addresses provided by Zoom.

Steps for MBG Configuration to Support Zoom

To successfully integrate MBG with Zoom Direct Routing, the following two key configuration steps must be performed:

1. Add an Integrated Communication Platform (ICP) for MiVoice Business PBX.
2. Create two or more SIP trunks, each corresponding to a different Zoom region.

5.1.1 Licensing & Network Configuration

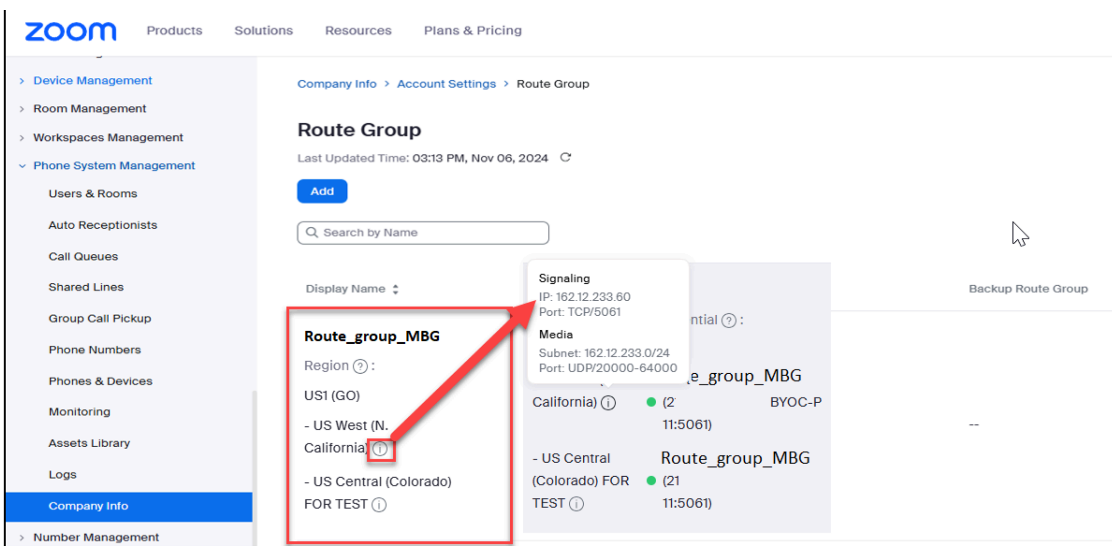
During the installation of the Mitel Border Gateway (MBG), you will be required to enter network configuration details for both the **WAN (Wide Area Network) and LAN (Local Area Network)**. These steps are standard for all MBG installations. For detailed instructions on setting up network configuration refer to the MBG Installation and Configuration Guide.

> **Important**:
> Initial releases of MiVoice Border Gateway, for Zoom DO NOT require a Zoom BYOC/BYOP license. However, this license will be required for future releases. During this transition, MiVoice Border Gateway Zoom BYOC/BYOP licenses will NOT BE NEEDED as part of the Zoom subscription.

5.1.2 Finding Zoom IP Addresses for SIP Trunking

The Configuring the Route group section displays the regions available for connection under the **Route_group_MBG** field. Click the symbol, where you can find the Zoom IP address for each region as shown in the image.

> **Note**:
> It would be better if the Fully Qualified Domain Names (FQDNs) were known to ensure accurate configuration and troubleshooting.



5.1.3 Configuring Certificates

Zoom Phone System allows only TLS connections for SIP traffic from SBCs with a certificate signed by one of Zoom trusted Certification Authorities.

Important:

The list of trusted root authorities for Zoom services is maintained by Zoom and may change over time. Including static information from internal documents is not recommended due to potential changes without notice. Always rely on official Zoom documentation or support channels. For the most accurate and up-to-date information, users must contact Zoom Support directly.

To contact Zoom Support, visit the Zoom Support Contact Page or reach out to your Zoom account representative.

For the MBG TLS interconnection to the Zoom Phone System, three files in '**pem**' format are required from the Certification Authority:

- A certificate authority or certification authority (CA) certificate (for example, "ca_chain.pem"). The CA certificate contains a public key and the owner's identity, ensuring an entity can be trusted.
- Server certificate for MBG (for example, "certificate.pem").
- MBG server certificate private key used for the CSR to CA (for example, "privatekey.pem").

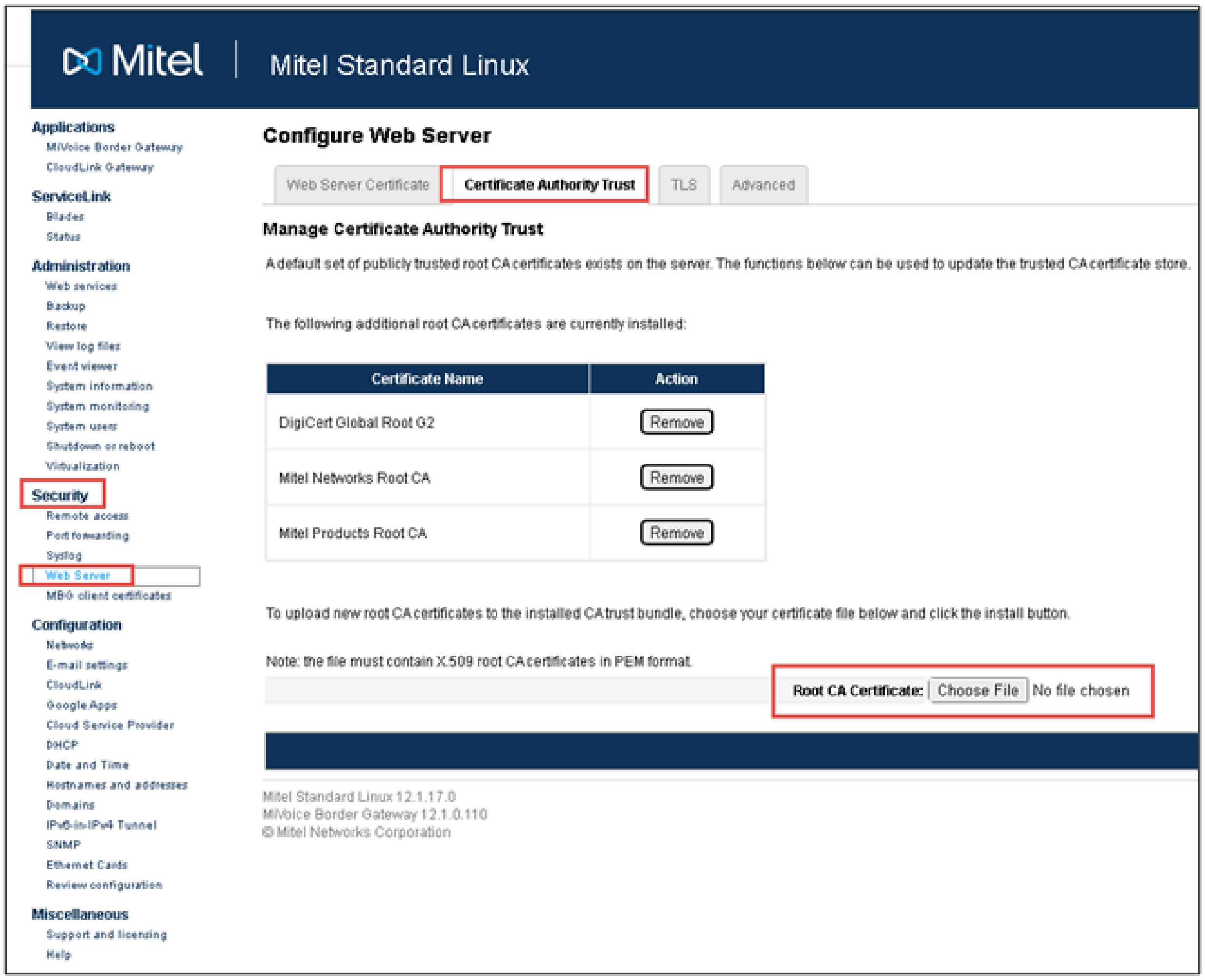
Note:
Zoom trusted root CA certificates can be downloaded directly from the Zoom portal, Updating root Certificate for Zoom Services.

The files mentioned above must be uploaded to MBG for the TLS connection with the Zoom Phone System interface.

To Upload the Root CA and Wildcard Certificates :

1. Navigate to **Security> Web server > Certificate AuthorityTrust > Root CA Certificate**.

2. Click **Choose file**.



3. Select the certificates to be uploaded.

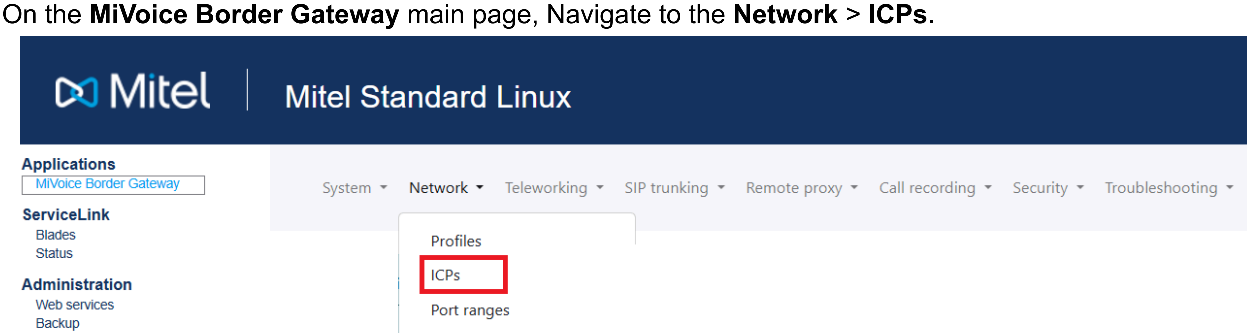
4. Once the Certificates are uploaded, click **Install Root CA Certificates**.

5.1.4 Adding a Network ICP

Before adding SIP Trunks to the MBG, a Network ICP/PBX must be created to serve as the connection point for the trunk. The MBG has been tested with multiple PBX systems operating behind it, ensuring compatibility and reliable communication.

To add Network ICP:

1. On the **MiVoice Border Gateway** main page, Navigate to the **Network** > **ICPs**.



2. Click the **'+' icon** to add an ICP.

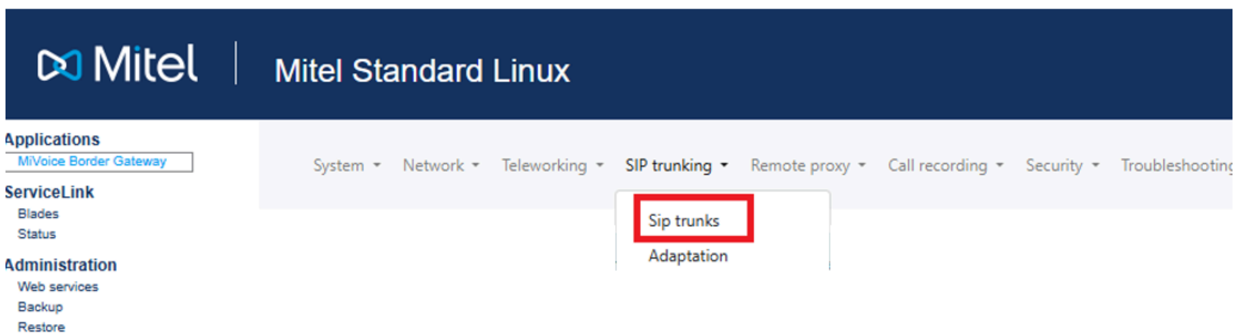
3. In the **Manage ICP** window, Configure the following:

   a. **Name**: Enter the name that you choose. (example: ZOOMMIVB).
   b. **Type**: From the Type drop-down menu, select the **MiVoice Business**.
   c. **SIP Capabilities**: From the SIP capabilities drop-down menu, select the **UDP, TCP, TLS** option.
   d. **Hostname or IP Address**: Enter the static IP address or FQDN.
   e. Click **Save**.

5.1.5 Adding SIP Trunks

For each Zoom region, create a separate SIP Trunk by navigating to the SIP trunks configuration page.

1. On the **MiVoice Border Gateway** main page, Navigate to the **SIP trunking** > **Sip trunks**.



2. Click the **'+' icon** to add a new SIP Trunk.
3. In the **Manage SIP trunk** window, Configure the following:

   a. Profile field

      i. Select the **Enabled** check box to enable SIP Trunking.
      ii. **Name**: Enter the name. (For example: ZOOMTLS1, for trunk-1).

   b. Connection field

      i. **Transport protocol**: From the Transport protocol drop-down, select the **TLS**.
      ii. **Remote trunk endpoint address**: Enter the Zoom provided IP address. (Refer Finding Zoom IP Address for SIP Trunking section to find the Zoom IP address for SIP Trunking).
      iii. **Remote trunk endpoint port**: Enter the Remote trunk endpoint port as **5061**.
      iv. **Outgoing TLS trust profile:** From the drop-down menu, select **MTLS using installed Web certificate**.

   c.

> **Note**:
> If your global settings for RTP security are already configured as desired, you can use them. In this example, we demonstrate a setup where RTP is **encrypted on the trunk-side** and allows **either encrypted or not on the ICP side**.

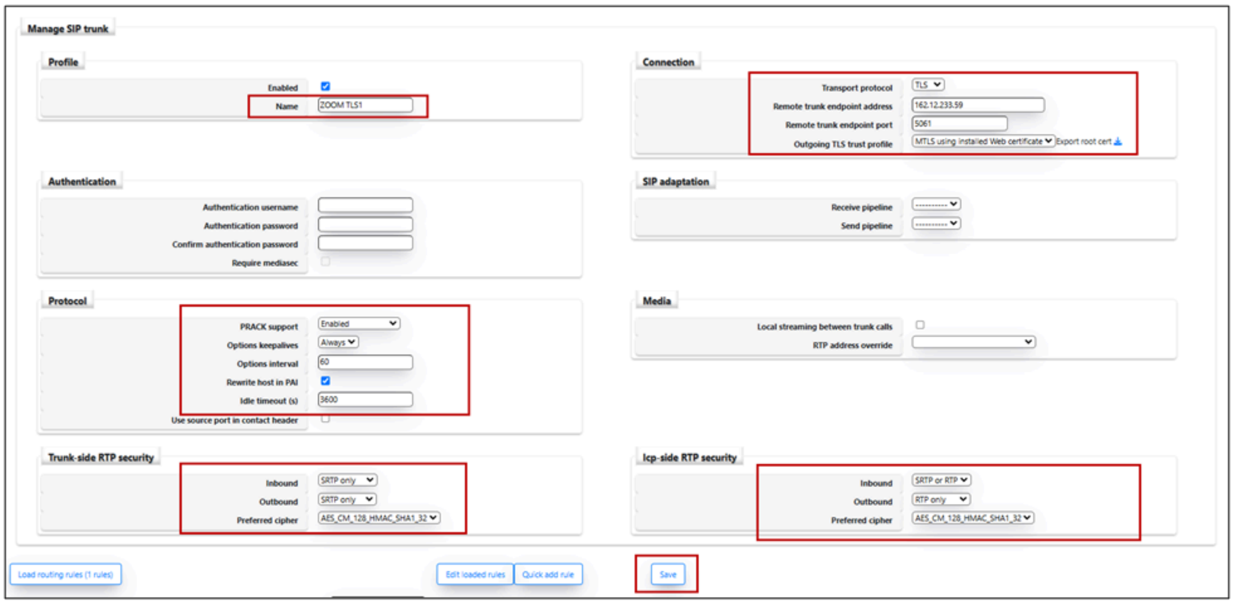
d. Trunk-side RTP security field

i. **Inbound**: From the drop-down menu, select **SRTP only**.
ii. **Outbound**: From the drop-down menu, select **SRTP only**.
iii. **Preferred cipher:** From the drop-down menu, select **AES_CM_128_HMAC_SHA1_32**.

e. ICP-side RTP security

i. **Inbound**: From the drop-down menu, you can select **SRTP or RTP** .
ii. **Outbound**: From the drop-down menu, select **RTP**.
iii. **Preferred cipher:** From the drop-down menu, select **AES_CM_128_HMAC_SHA1_32**.

4. Click **Save**.

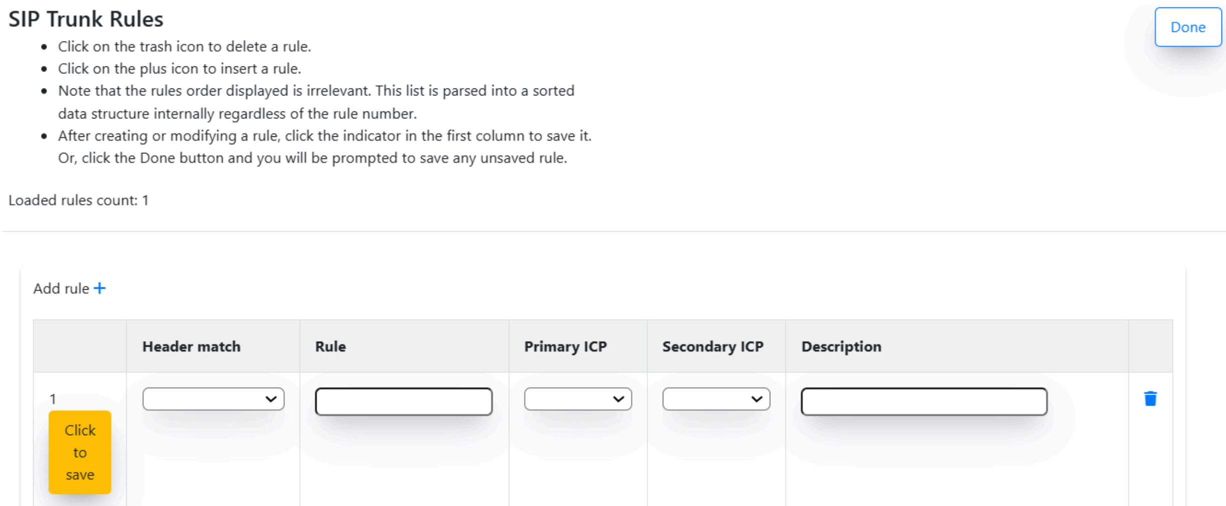


To add a Rule:

1. Click **Quick add rule button** at the bottom of the form.

> **Note**:
> A warning message will appear, and this option will only be enabled after saving the new trunk definition.

2. The **SIP Trunk Rules** form will open. Click **'+' icon** to add a rule.



3. Configure the following:

   a. **Header match**: From the drop-down menu, select **Request URI**.
   b. **Rule** : Enter *
   c. **Primary ICP**: select the MiVB Name you configured in section Adding a Network ICP from the drop-down menu. (Example: **ZOOMMIVB**).
   d. Click **Click to Save** option followed by **Done**.

> **Note**:
> To view the added SIP trunk, navigate to **SIP Trunking > SIP Trunks** from the top menu.

Now, repeat these steps to add any additional SIP trunks required for different Zoom regions. Here 3 Zoom trunks are considered. (Example: **ZOOMTr2** and **ZOOMTr3**).

5.2 MiVB Configuration for BYOP/BYOC

This chapter describes the MiVoice Business configuration for connecting to MiVoice Border Gateway. The purpose of this connectivity is for MiVoice Business to provide the necessary SIP message manipulation and call routing facilities to MBG so that the latter can interconnect to Zoom Phone SBC and calls between Zoom clients and PSTN subscribers are feasible.

In the MiVoice Business, you must configure with SIP Trunks that use the MBG as an Outbound Proxy Server to reach the Zoom regions. The MiVB is configured using the ESM configuration interface. The System Administration Tool provides the forms used for configuration, as described below.

5.2.1 Licensing

The Mitel MiVoice Business system to function correctly, it must have both valid licensing and a proper network configuration. Before connecting the Mitel MiVoice Business system to **COX SIP Trunking**, you need to confirm that the system has a sufficient number of **SIP trunk licenses**. SIP trunk licenses allow the MiVoice Business system to establish and manage SIP-based voice connections. Without enough licenses, the system may not be able to handle the required number of simultaneous SIP calls. This can be configured in the License and Option Selection form.

> **Note**:
> The Mitel MiVoice Business (MiVB) system to function correctly, it must have both valid licensing and a proper network configuration. During the installation process, the system will prompt the user to enter network addresses (such as IP addresses, subnet masks, and gateway settings) and to apply a **valid license** to activate the required features. Since these steps are standard across all MiVB installations, they are not covered in this document. For detailed instructions, refer MiVoice Business Technician's Handbook.

To Configure the SIP trunk Licenses:

1. Navigate to **Licenses** > **License and Option Selection** form.

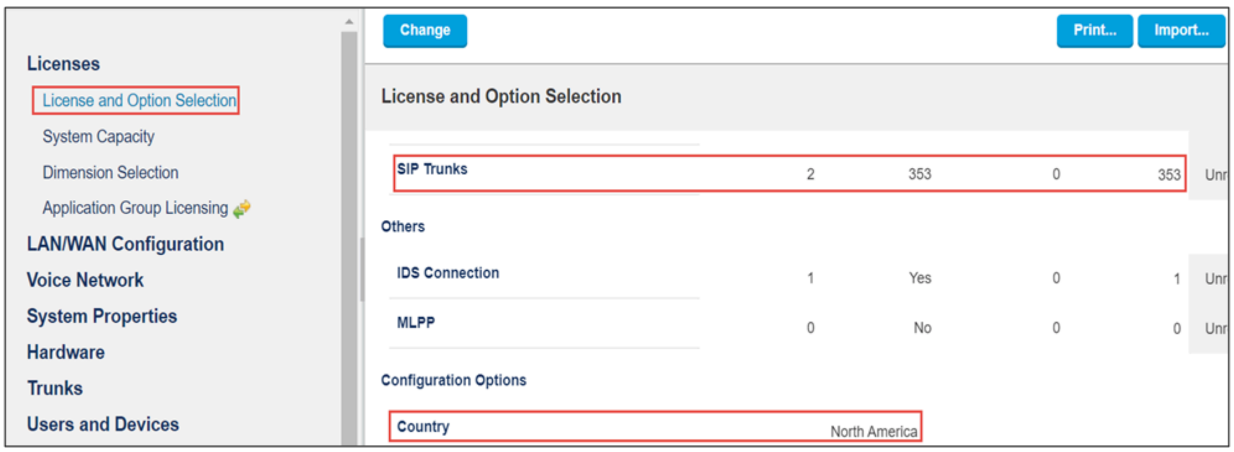


Figure 2: Licensing

2. Enter the total number of licenses in the **SIP Trunks** field.
3. In **Country** field, Select your country from the Country drop-down menu. (Example: **North America**).
4. Click **Save**.

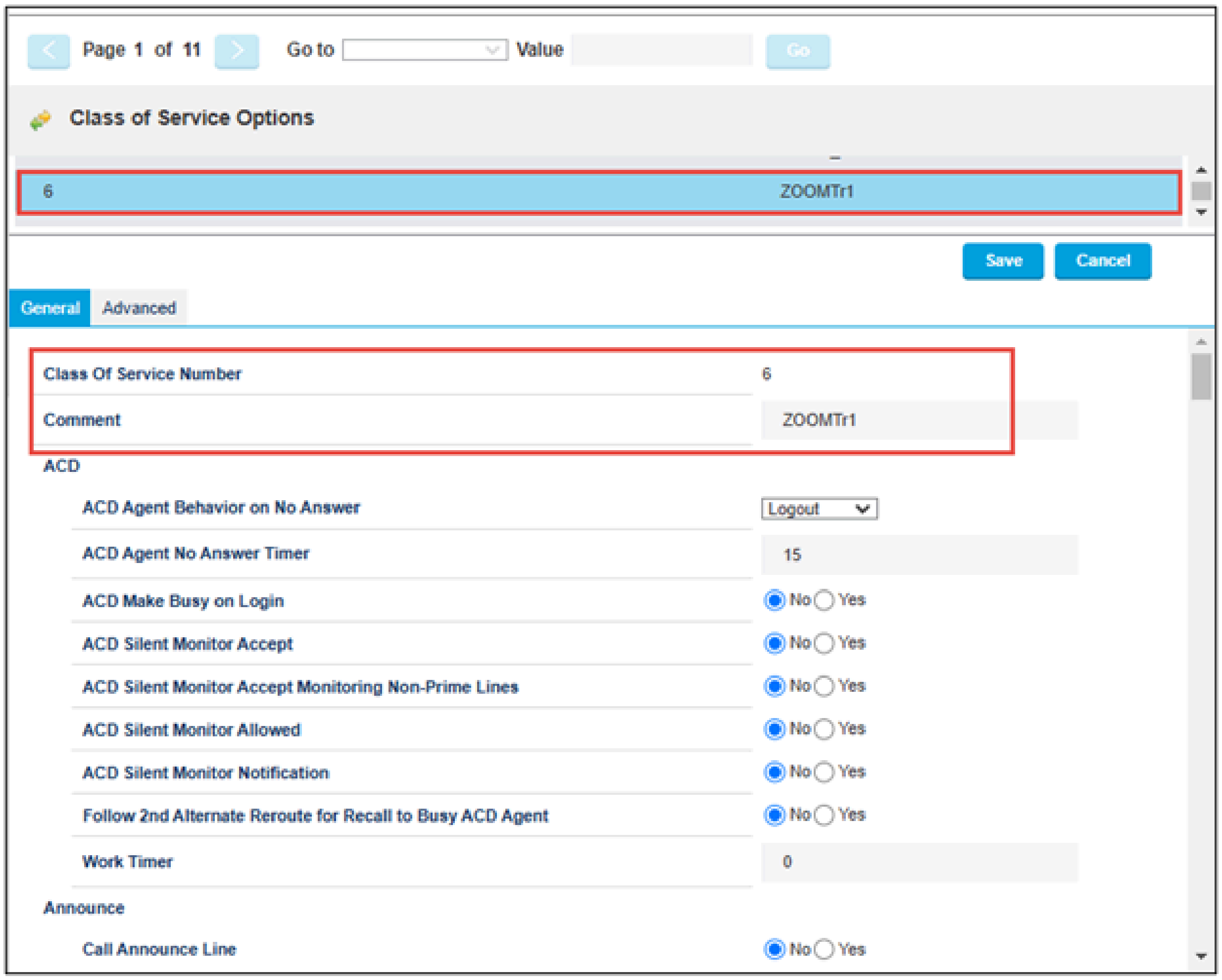
> **Note**:
> The entered number (example: 353) is the maximum number of SIP trunk sessions that can be configured in the MiVoice Business system to be used with all service providers, applications, and SIP trunking devices.

5.2.2 Class of Service

The Class of Service Options form is used to create or edit a Class of Service and specify its options. Classes of Service, identified by Class of Service numbers, are referenced in the Trunk Service Assignment form for SIP trunks.

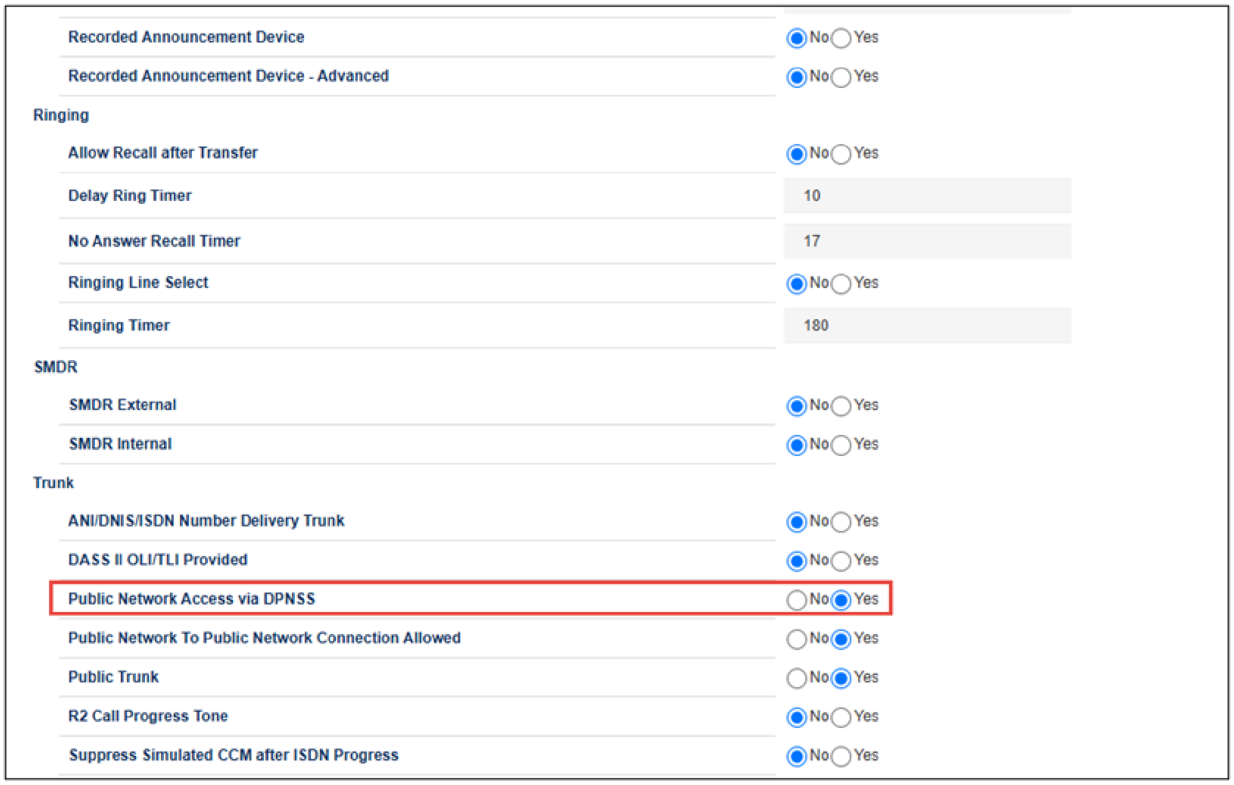
To Configure Class of Service option Zoom Trunk:

1. Navigate to **System Properties** > **System Feature Settings** > **Class of Service Options.**
2. Click **Change**.
3. **Class o Service Number** : Enter **6** for ZOOMTr1.
4. **Comment** : Enter the Zoom trunk name. (Example: **ZOOMTr1**).



5. Search for **Public Network Access via DPNSS** and Select **Yes**.

6. Leave all other fields as default.



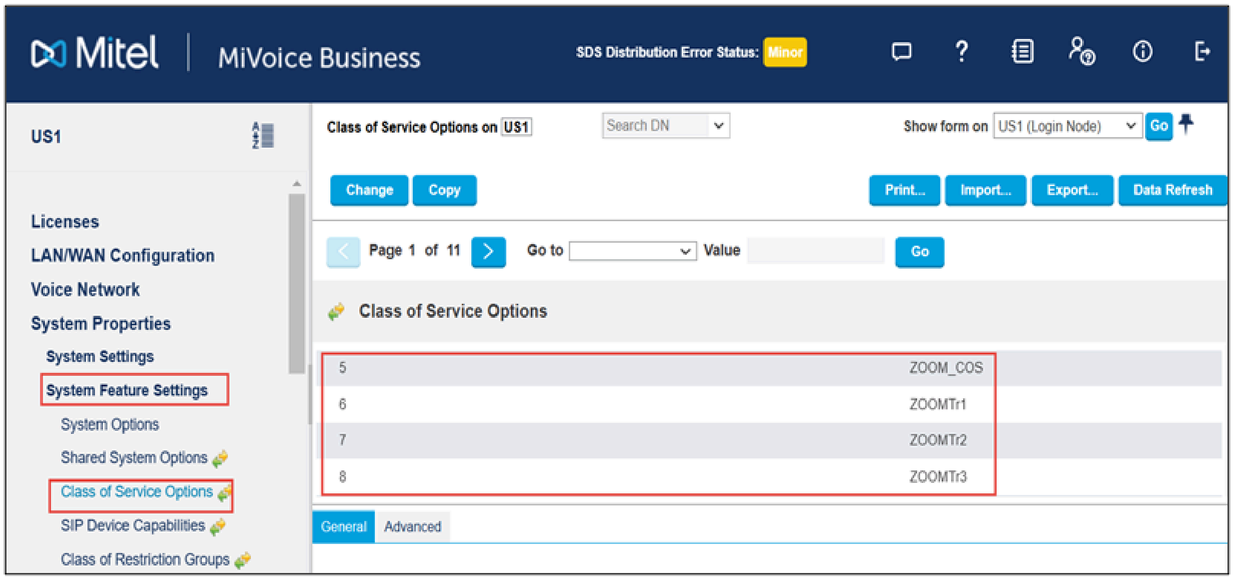
7. Click **Save**.

Similarly Add Class of Service Options for other 2 Zoom Trunks **ZOOMTr2** and **ZOOMTr3** respectively.

To Configure Class of Service option for PSTN Trunk:

1. Navigate to **System Properties** > **System Feature Settings** > **Class of Service Options.**
2. Click **Change**.
3. **Class o Service Number** : Enter **5** for ZOOM_COS.
4. **Comment** : Enter the PSTN trunk name. (Example: **ZOOM_COS**).
5. Search for **Public Network Access via DPNSS** and Select **Yes**.
6. Leave all other fields as default.
7. Click **Save**.

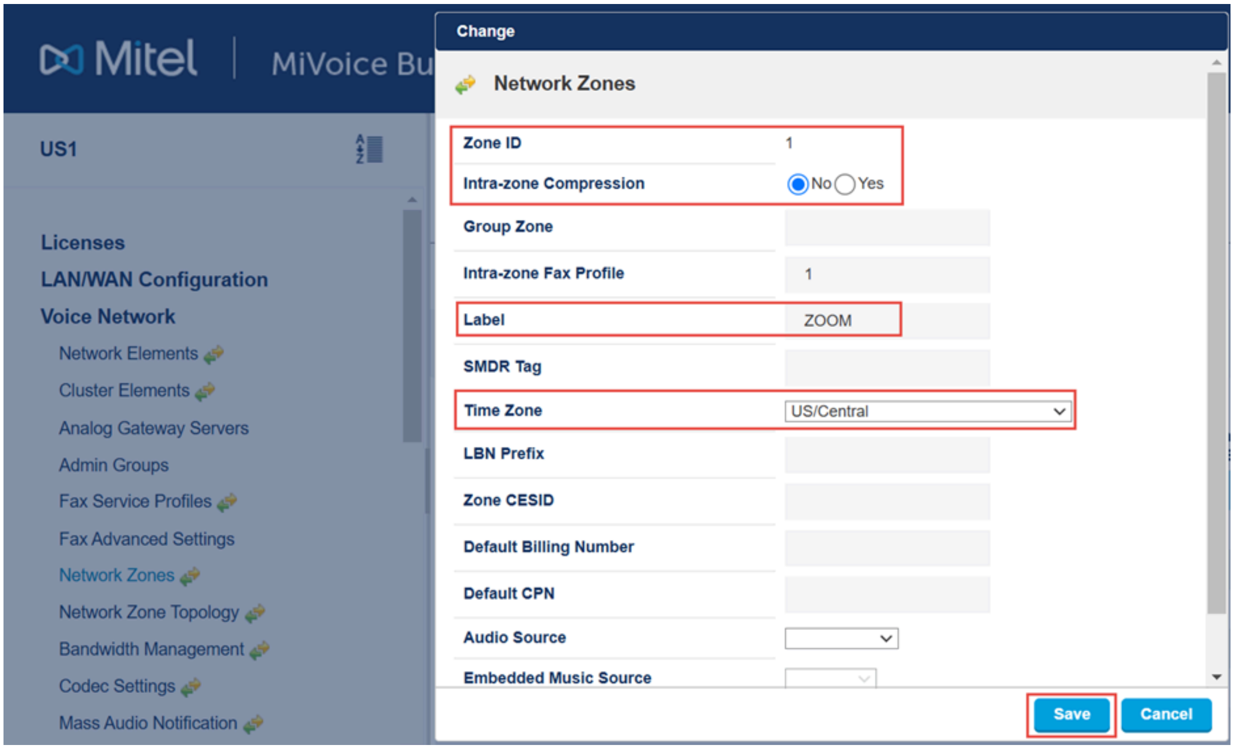
The **Class of Services** added for the **Zoom Trunks** and **PSTN Trunks** are listed in the **Class of Service Options** form as shown in the below figure.



5.2.3 Network Zone Assignment

To configure the Network Zone Assignment:

1. Navigate to **Network > Network Zones** form.
2. Select Desired Zone ID (**1** is used for this setup).
3. Click **Change**.
4. **Intra-Zone Compression**: Select **No**.
5. **Label**: Enter the label as **ZOOM**.
6. **Time Zone**: Select the proper time zone from the drop-down menu.
7. Leave all other fields as default.
8. Click **Save**.

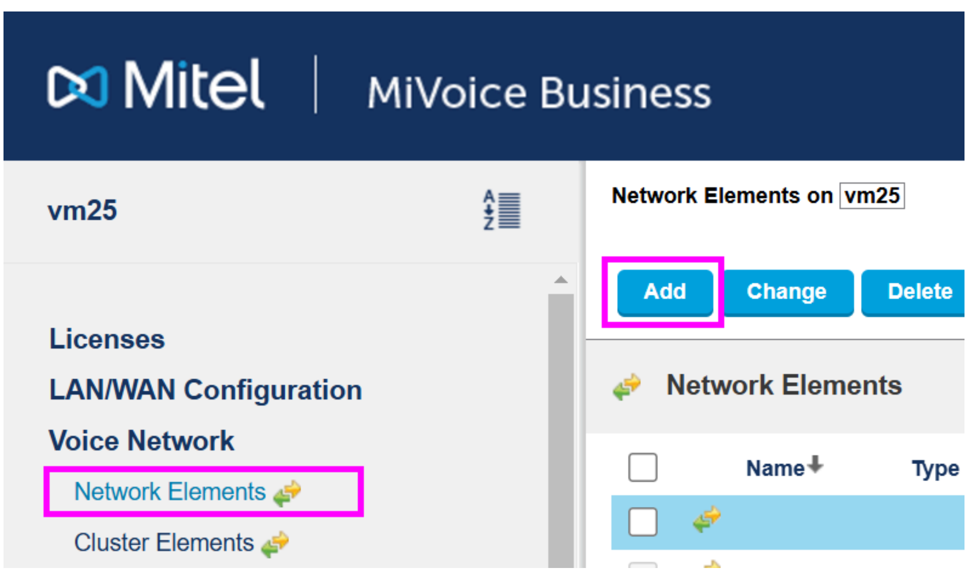


5.2.4 Adding Network Elements

You will need to create a Network Element for each Zoom region you are trying to access and the MBG which will act as your Outbound Proxy. This is done from the Network Element form.

To add network element:

1. Navigate to **Voice Network** > **Network Elements** form.
2. Click **Add**.



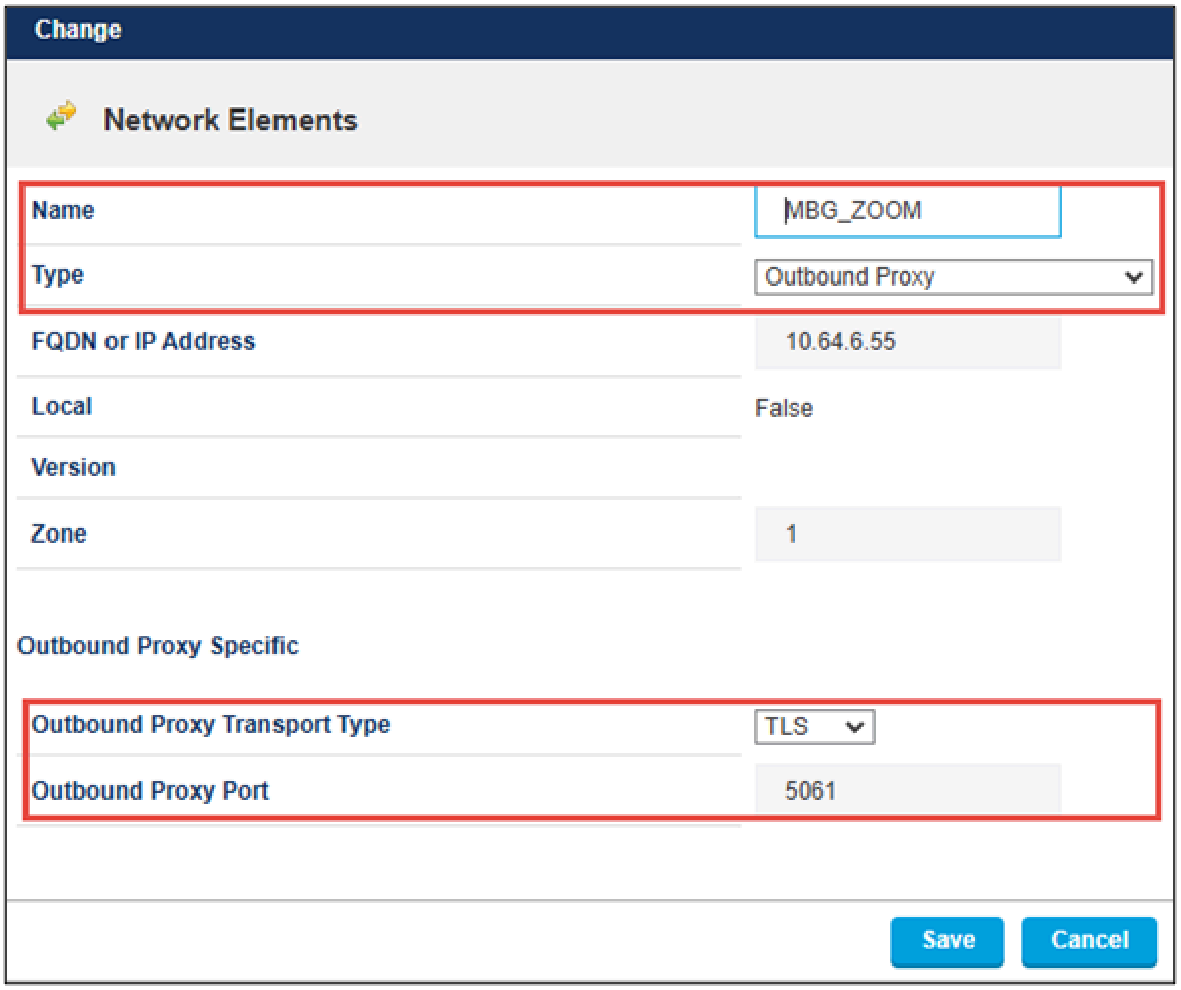
To add MBG Outbound Proxy:

1. Configure the following:

   a. **Name**: Enter the name. (Example: MBG_ZOOM).
   b. **Type**: From the drop-down menu, Select **Outbound Proxy**.
   c. **FQDN or IP Address**: Enter the MiVoice Border Gateway LAN IP address. For example, **10.64.6.55** is used).

   > **Note**:
   > MBG LAN IP address can be seen on the MBG dashboard in the Server-Manager.

   d. **Outbound Proxy Transport Type**: Select **TLS** from the drop-down menu.
   e. **Outbound Proxy Port**: Enter **5061**.
2. Click **Save**.



5.2.4.1 Adding PSTN Outbound Proxy

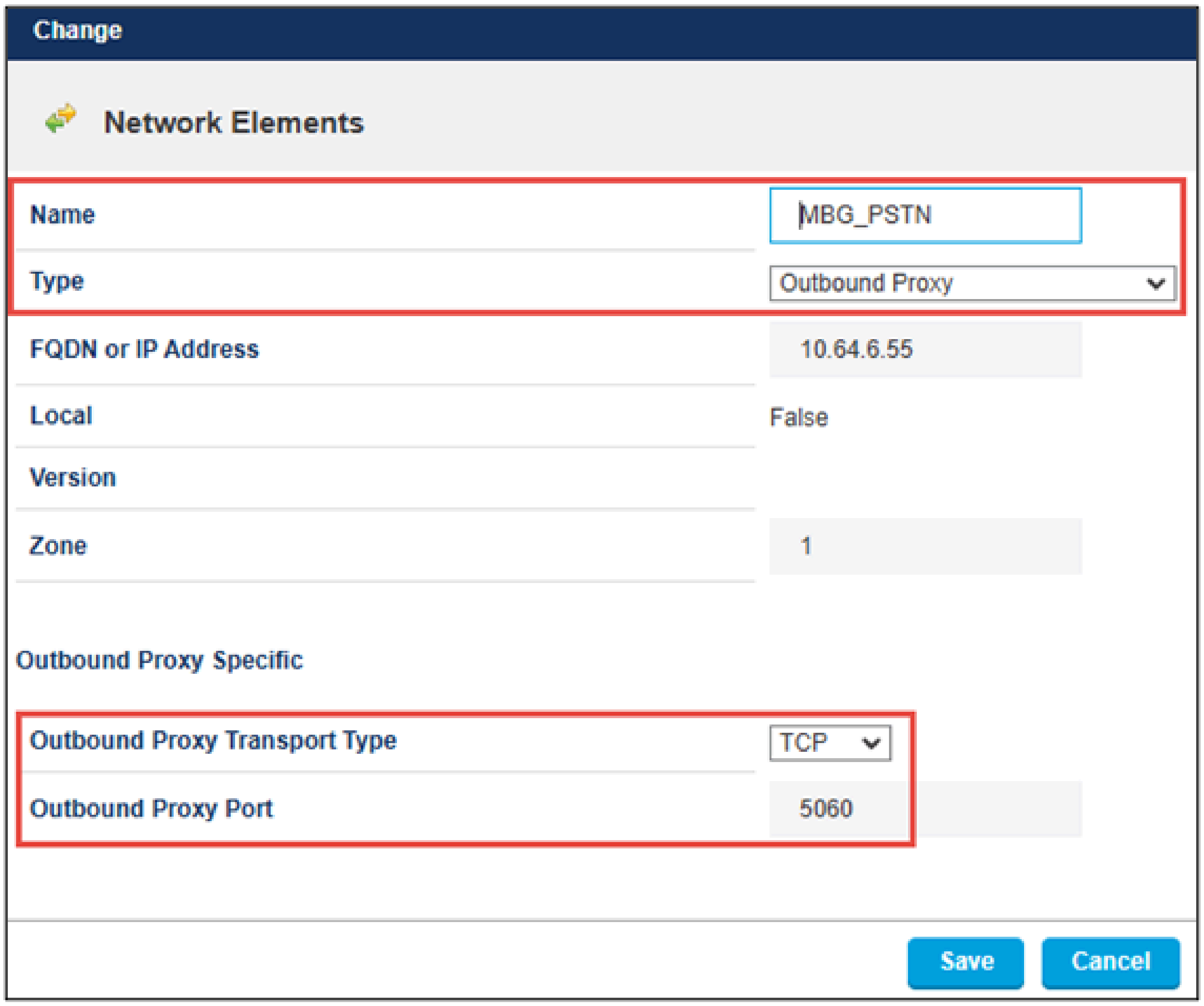
To connect with PSTN:

1. Configure the following:

   a. **Name**: Enter the name. (Example: MBG_PSTN).
   b. **Type**: From the drop-down menu, Select **Outbound Proxy**.
   c. **FQDN or IP Address**: Enter the MiVoice Border Gateway LAN IP address. For example, **10.64.6.55** is used).

   > **Note**:
   > The MBG LAN IP address can be seen on the MBG dashboard in the Server-Manager.

   d. **Outbound Proxy Transport Type**: Select **TCP** from the drop-down menu.
   e. **Outbound Proxy Port**: Enter **5060**.
2. Click **Save**.



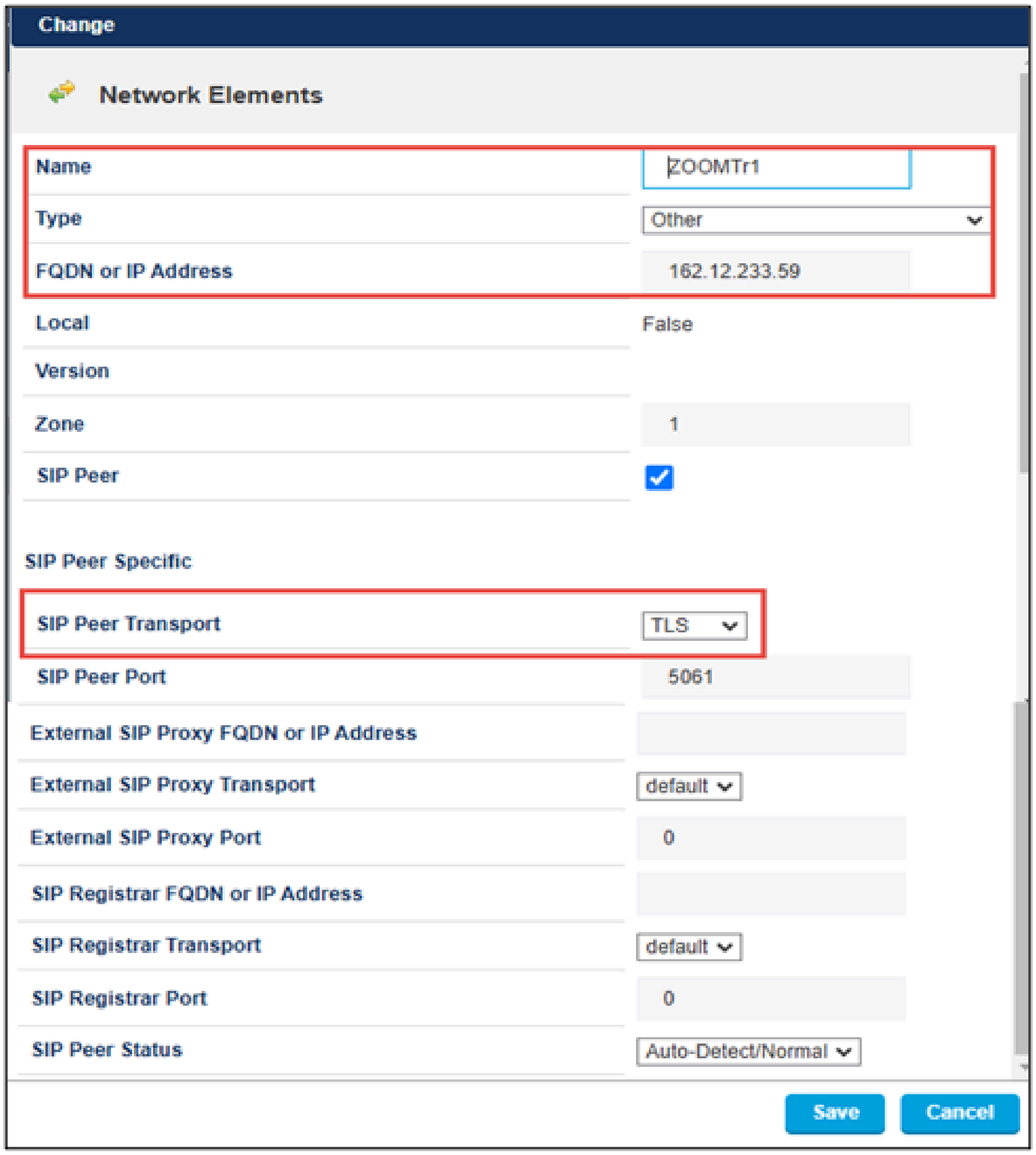
5.2.4.2 Adding Zoom Region Network Elements

After creating the MBG and PSTN Network elements, the next step is to add the Zoom Region Network elements. Three Zoom regions are considered here, and each Zoom region must be configured separately.

To configure the Zoom Region Network Elements, follow these steps:

1. Navigate to **Voice Network** > **Network Elements** form.
2. Click **Add**.
3. Configure the following:

    a. **Name**: Enter the name. (**ZOOMTr1**, this is an example for ZOOM Trunk 1).
    b. **Type**: From the drop-down menu, Select **Others.**
    c. **FQDN or IP Address**: Enter the Zoom IP address. (Refer Finding Zoom IP Address for SIP Trunking section to find the Zoom IP address for each zoom region. For example, **162.12.233.59** is used).
    d. **Zone**: Enter the Zone number. (Example: 1 is used here).
    e. **SIP Peer**: Select the **SIP Peer** check box.
    f. **SIP Peer Transport**: Select **TLS** from the drop-down menu.
    g. Click **Save**.



Follow the same steps to add Network element for other two Zoom regions (**ZOOMTr2** and **ZoomTr3**).

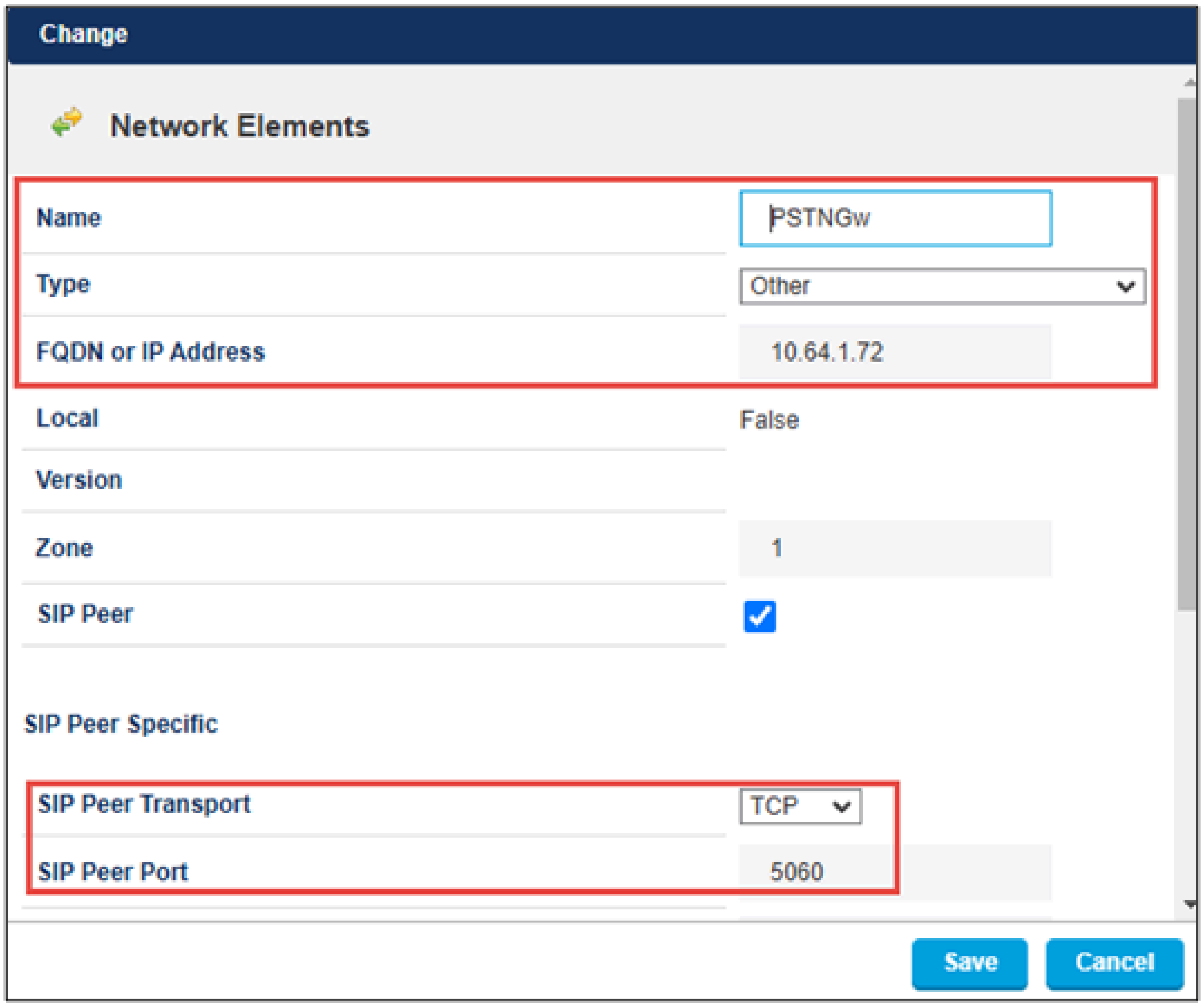
5.2.4.3 Adding PSTN Region Network Element

To configure the PSTN Region Network Elements, follow these steps:

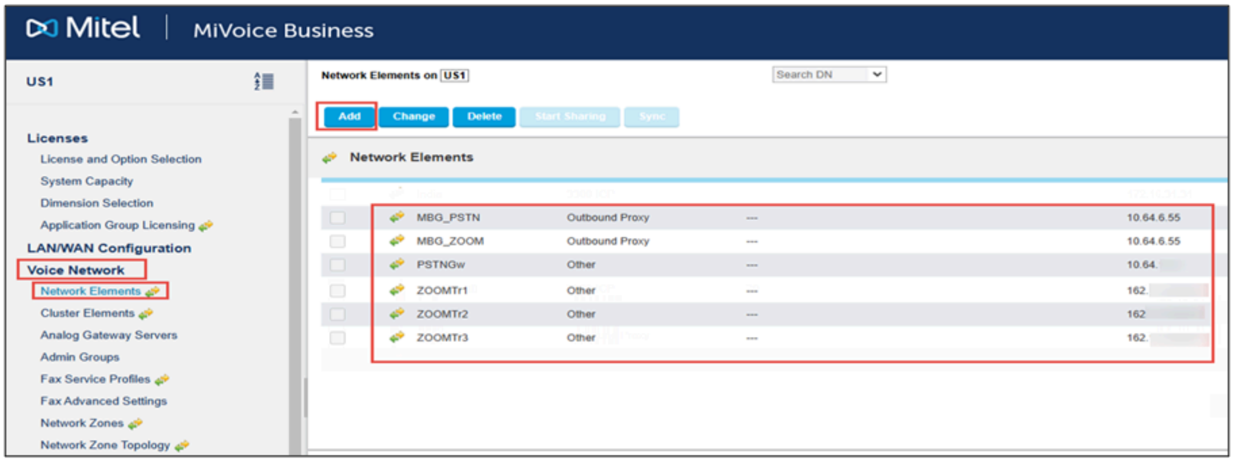
1. Navigate to **Voice Network** > **Network Elements** form.
2. Click **Add**.

3. Configure the following:

   a. **Name**: Enter the name. (Example: **PSTNGw**).
   b. **Type**: From the drop-down menu, Select **Others**.
   c. **FQDN or IP Address**: Enter the PSTN Gateway IP address.
   d. **Zone**: Enter the Zone number. (Example: 1 is used here).
   e. **SIP Peer**: Select the **SIP Peer** check box.
   f. **SIP Peer Transport**: Select **TCP** from the drop-down menu.
   g. Click **Save**.



You can see the list of network elements added under the Network Elements form.



5.2.5 Creating a SIP Peer Profile for each Zoom SIP Trunk

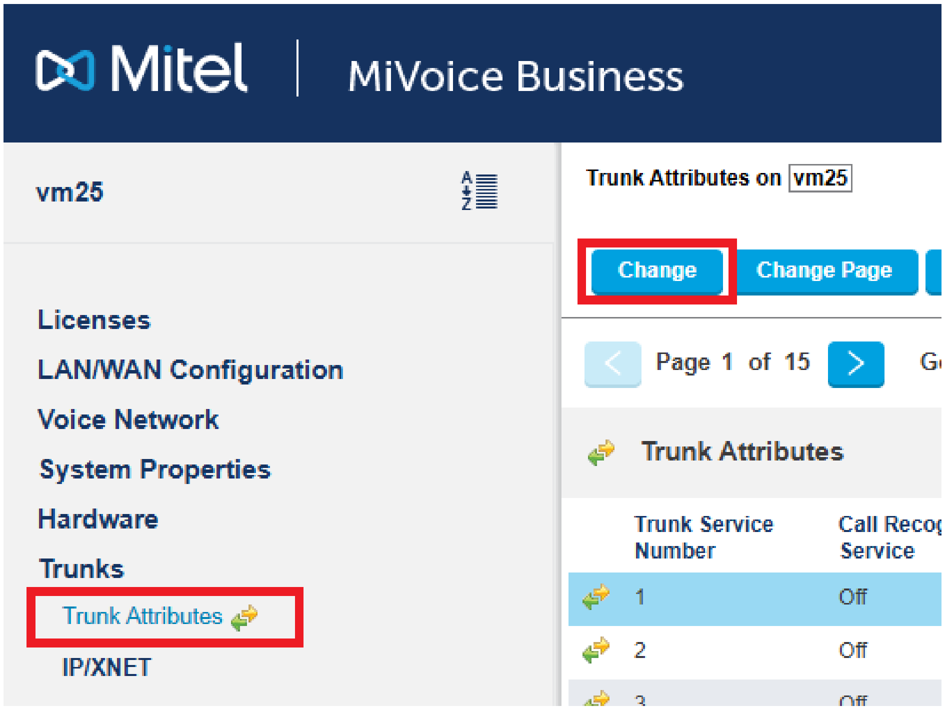
The next step is to create a Trunk Attribute that can be used for each SIP Peer.

5.2.5.1 Configuring the Trunk Attributes

The Trunk Attributes form can define the incoming call handling and class of service for the trunk.

1. Navigate to **Trunks** > **Trunk Attributes** form.
2. Click **Change.**

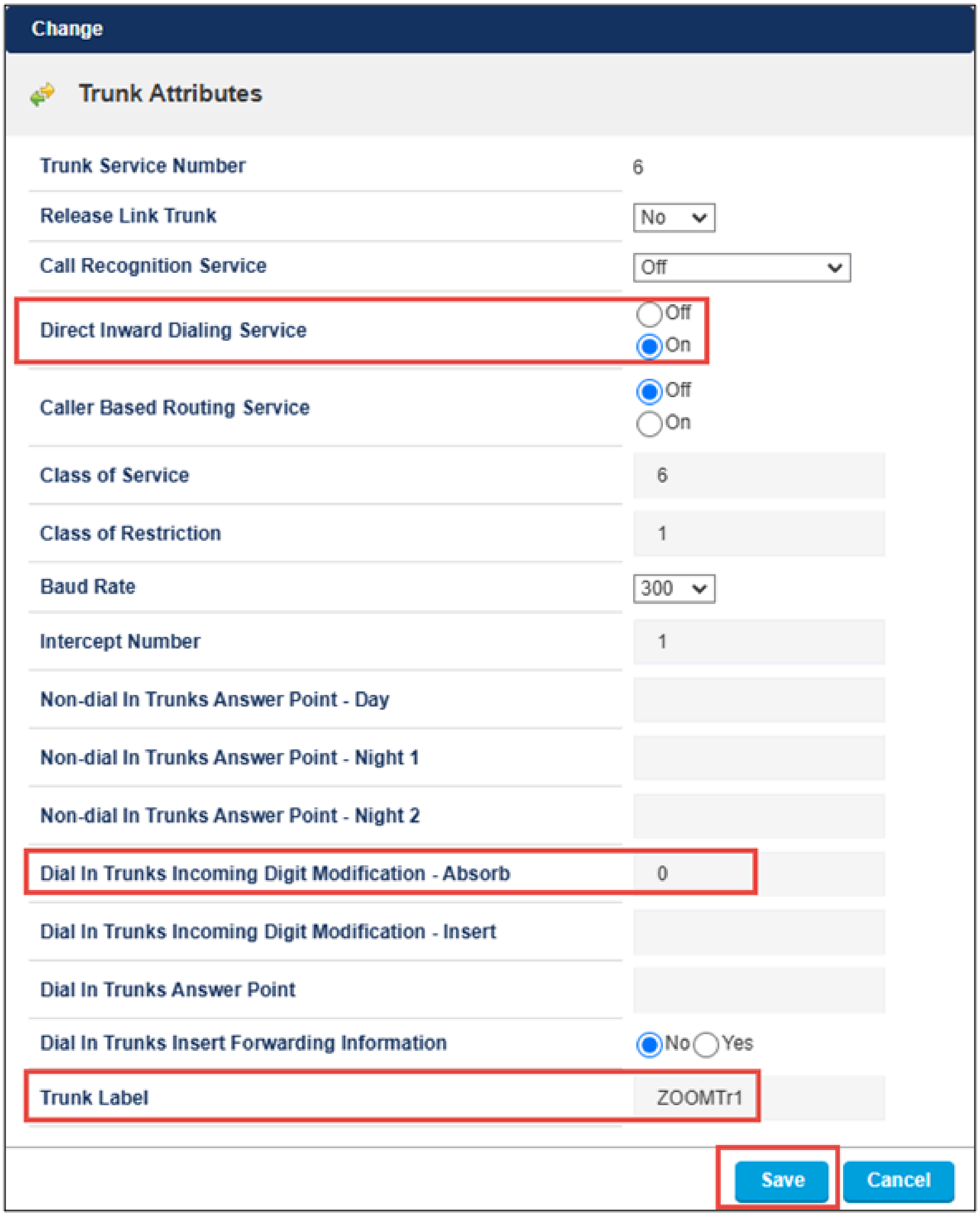
> **Note**:
> Create Trunk Attributes for all Zoom Trunks and PSTN Trunk. As an example, Here 3 Zoom Trunks are considered.



To Configure Trunk Attribute for Zoom Trunks, Do the following:

1. **Direct Inward Dialing Service**: Select **On** for this setup.
2. **Class of Service**: Enter **6**, that was created in the Class of service section.
3. **Dial In Trunks Incoming Digit Modification - Absorb** : Enter 0. (The means on incoming calls we will strip 0 digits).
4. **Trunk Label**: Enter **ZoomTr1** (This example is for **Zoom Trunk 1)**.
5. Leave the other field values as default.
6. Click **Save**.

Note:
Similarly, add Trunk Attributes for all Zoom Trunks (ZOOMTr2 and ZOOMTr3).



To Configure Trunk Attribute for PSTN Trunks, Do the following:

1. **Direct Inward Dialing Service**: Select **On** for this setup.
2. **Class of Service**: Enter **5**, that was created in the Class of service section.
3. **Dial In Trunks Incoming Digit Modification - Absorb** : Enter 0. (The means on incoming calls we will strip 0 digits).

4. **Trunk Label**: Enter Trunk Label. (Example: **SIPTrunk is used).**
5. Leave the other field values as default.
6. Click **Save**.

5.2.5.2 Configuring the SIP Peer Profiles

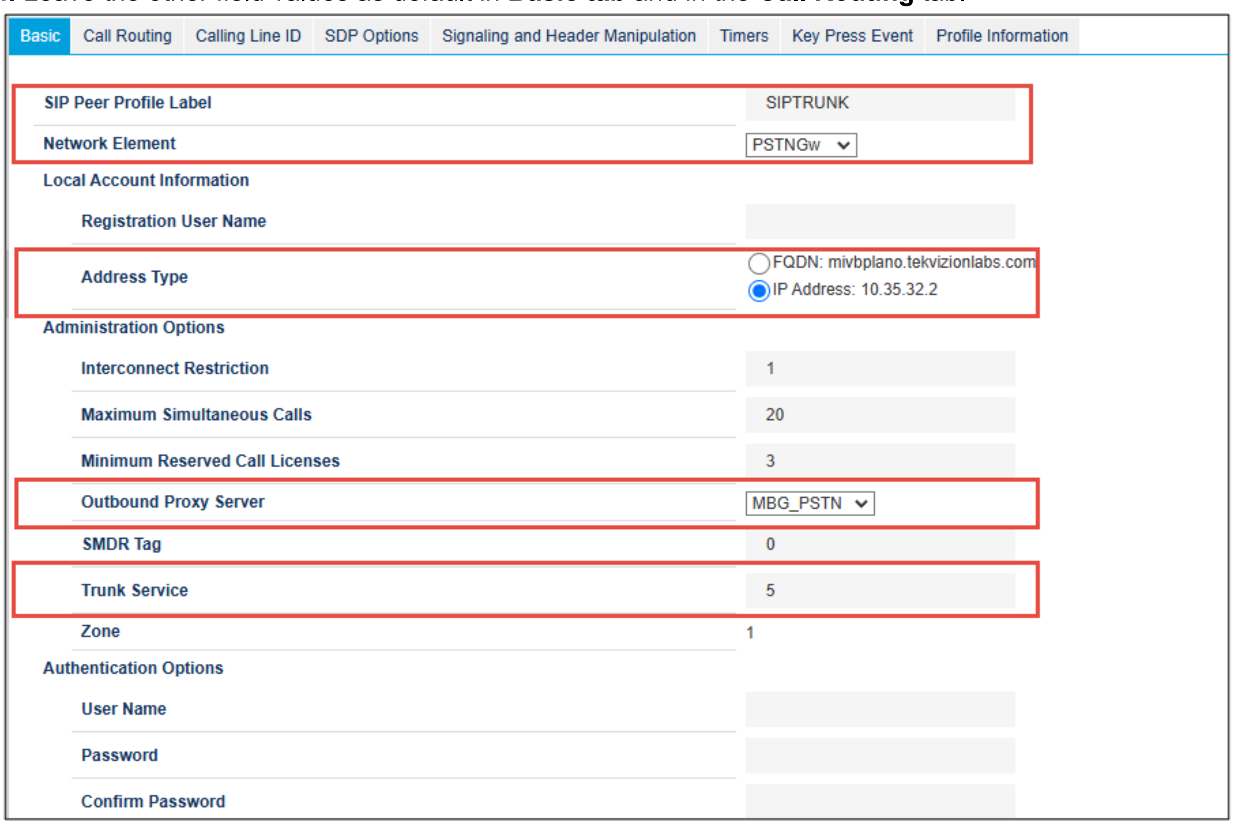
After creating Trunk Attributes, the next step is to add the SIP Peer under the **SIP Peer Profile** form. Similar to the Trunk Attributes creation, SIP Peer Profiles need to be created separately for each Zoom Trunk and for the PSTN Trunk by following the below mentioned steps.

1. Navigate to **Trunks** > **SIP** > **SIP Peer Profile** form.
2. Click **Add**.

To Configure SIP Peer Profiles for PSTN Trunks:

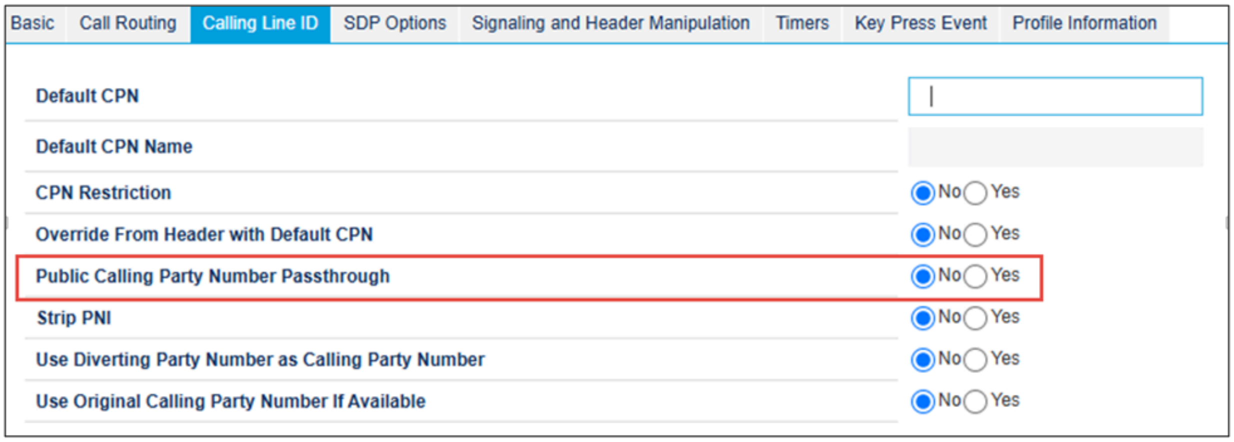
1. Navigate to **Basic** tab.

   a. **SIP Peer Profile Label**: Enter the label for SIP Peer Profile. (Example: SIP Trunk is used here).
   b. **Network Element**: Select the newly created network element for PSTN Region, **PSTNGw** from the section Adding PSTN Region Network elements from the drop-down menu.
   c. **Address Type:** Enter the Mitel MiVoice Business LAN IP address.
   d. **Outbound Proxy Server**: Select **MBG_PSTN** from the drop-down menu.
   e. Trunk Service: Newly created Trunk Service number **5** is used here.
   f. Leave the other field values as default in **Basic tab** and in the **Call Routing** tab.



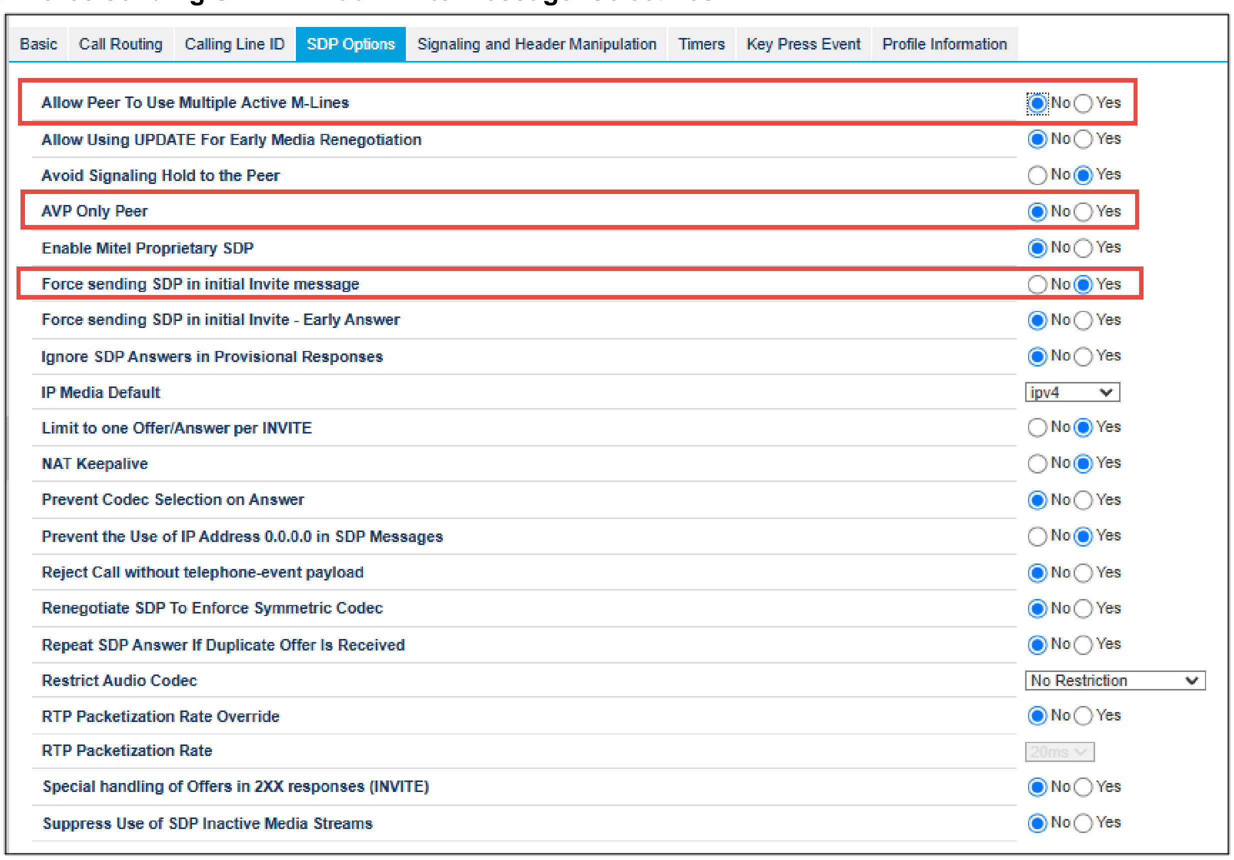
2. In the **Calling Line ID tab**,

a. Public Calling Party Number Passthrough : Select **No** for this setup.
b. Leave the other field values as default.



3. In the **SDP Options** tab,

a. Public Calling Party Number Passthrough: Select **No**.
b. Allow Peer To User Multiple Active M-Lines: Select **No**.
c. AVP Only Peer: Select **No**.
d. Force sending SDP in initial Invite message: Select **Yes**.

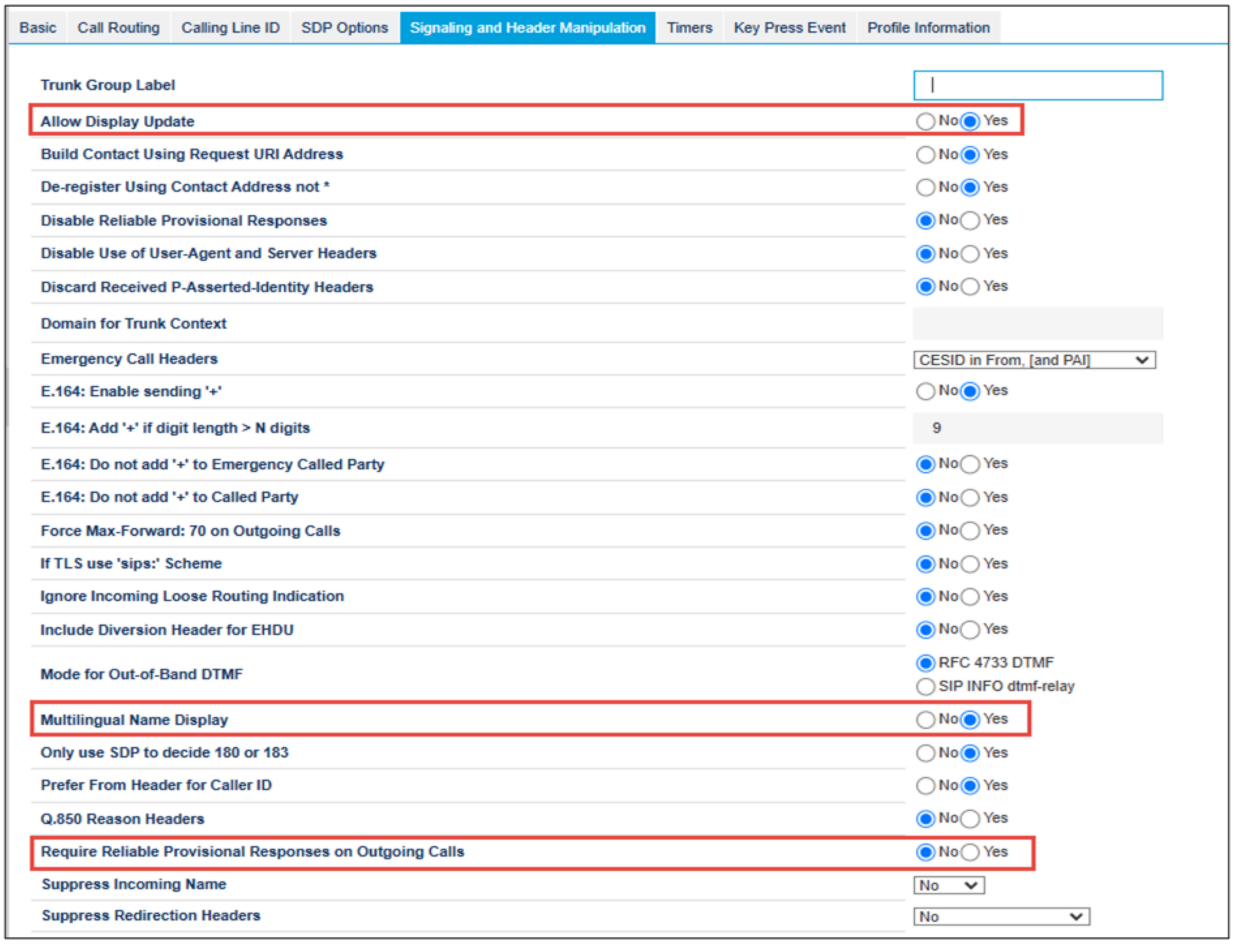


4. In the **Signaling and Header Manipulation** tab,

a. Allow Display Update: Select **Yes.**

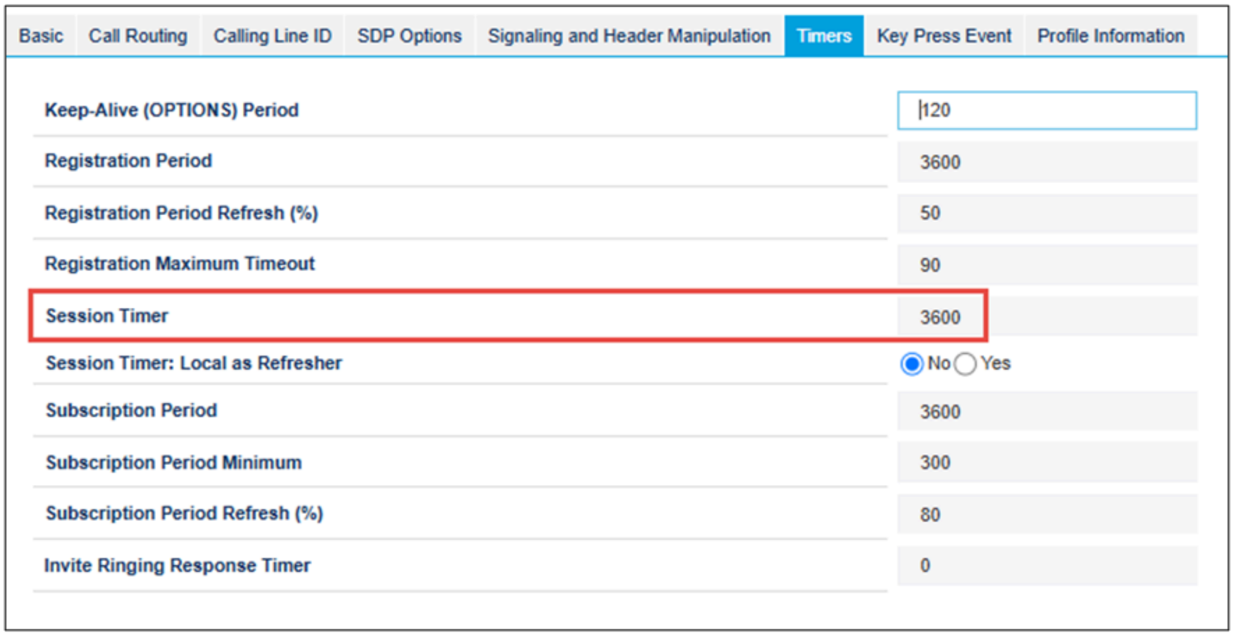
b. Multilingual Name Display: Select **Yes**.

c. Require Reliable Provisional Responses on Outgoing Calls: Select **No.**



5. In the **Timers** tab,

a. Set **Session Timer** as **3600.**



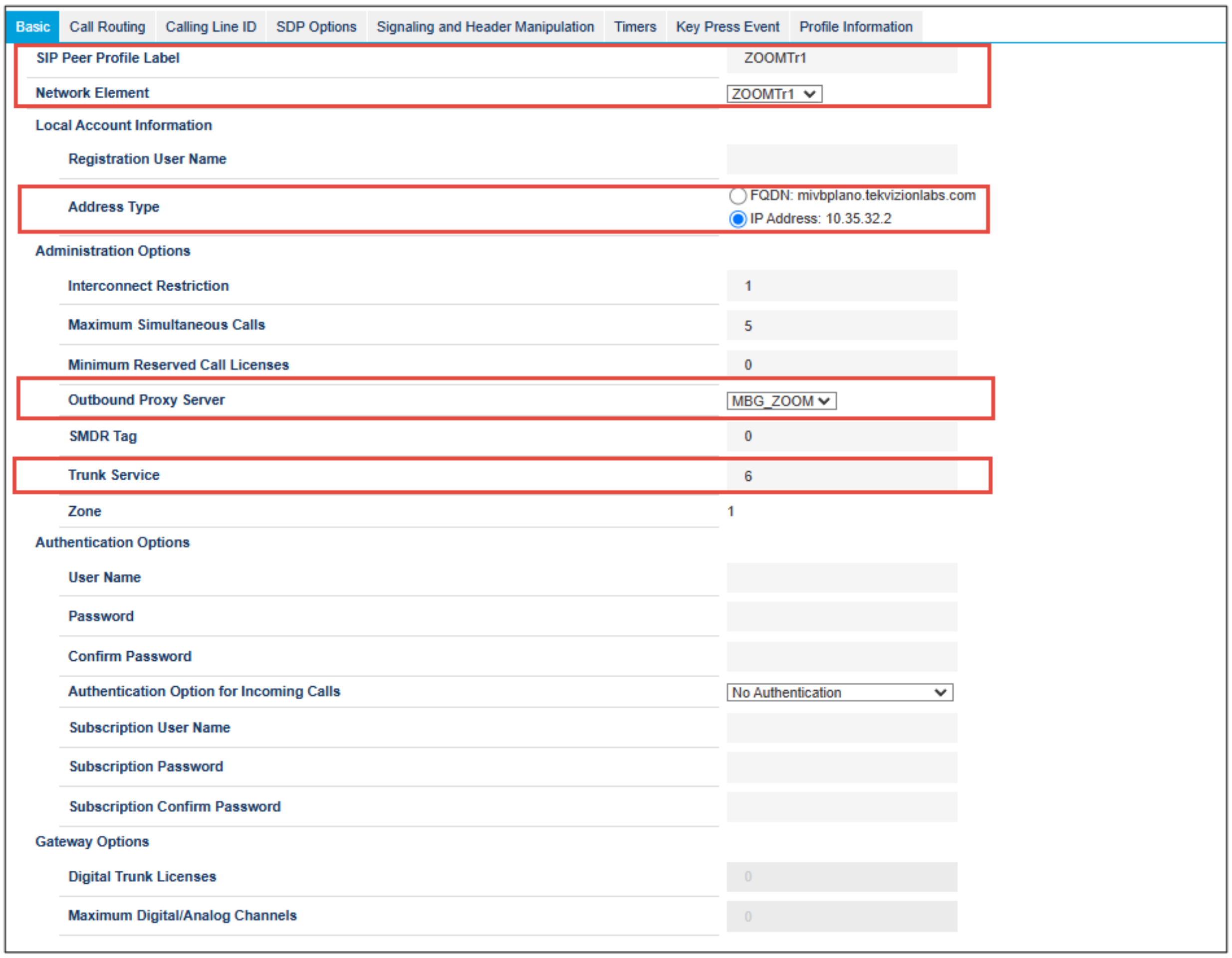
6. Leave the other field values as default.

7. Click **Save**.

5.2.5.2.1 Configure the SIP Peer Profile for Zoom Trunks

1. Navigate to **Basic** tab.

   a. **SIP Peer Profile Label**: Enter the label for SIP Peer Profile. (Example: **ZOOMTr1** is used here).
   b. **Network Element**: Select the newly created network element for Zoom Trunk 1, **ZOOMTr1** from the section Adding Zoom Region Network elements from the drop-down menu.
   c. **Address Type:** Enter the Mitel MiVoice Business LAN IP address.
   d. **Outbound Proxy Server**: Select **MBG_Zoom** from the drop-down menu.
   e. **Trunk Service**: Newly created Trunk service number **6** is used here.
   f. Leave the other field values as default in **Basic tab** and in the **Call Routing** tab.



2. In the **Calling Line ID tab**,

   a. Public Calling Party Number Passthrough : Select **No**.
   b. Leave the other field values as default

3. In the **SDP Options** tab,

   a. Public Calling Party Number Passthrough: Select **No**.
   b. Allow Peer To User Multiple Active M-Lines: Select **No**.
   c. AVP Only Peer: Select **No**.
   d. Force sending SDP in initial Invite message: Select **Yes**.
4. In the **Signaling and Header Manipulation** tab,

   a. Allow Display Update: Select **Yes**.
   b. Multilingual Name Display: Select **Yes**.
   c. Require Reliable Provisional Responses on Outgoing Calls: Select **No**.
5. In the **Timers** tab,

   a. Set **Session Timer** as **3600.**
6. Leave the other field values as default.
7. Click **Save**.

Similarly, Configure the other Zoom Trunks (For example: **ZOOMTr2** and **ZOOMTr3**).

5.2.6 Deciding on Outgoing Routing from Zoom Clients over SIP Trunk

The Zoom SIP Trunks will be used on outbound connections to contact both MiVB clients for BYOP and the PSTN for BYOC. One solution would be to add prefix digit(s) for PSTN calls so that they can be routed to the correct PSTN Trunking from the MiVB.

For example, the Zoom client may try to call 972598xxxx or extension xxxx. If the MiVB added a 8 to the trunk call it could be routed to the external PSTN connection where the 8 could be stripped. The extension would have no prefix so the call would simply be routed directly to that extension.

To add prefix digit(s) use the **Inward Dialing Modification** form to setup a rule for adding a prefix.

5.2.6.1 Adding Inward Dialing Modification rules

Using the **Inward Dialing Modification** form you can create a rule to add a prefix. In this example, add a 8 to any digit string greater than 8 digits for routing to a PSTN connection.

1. Navigate to **System Properties** > **System Feature Settings** > **Inward Dialing Modification form**.
2. Select **Change**.

3. Configure the following:

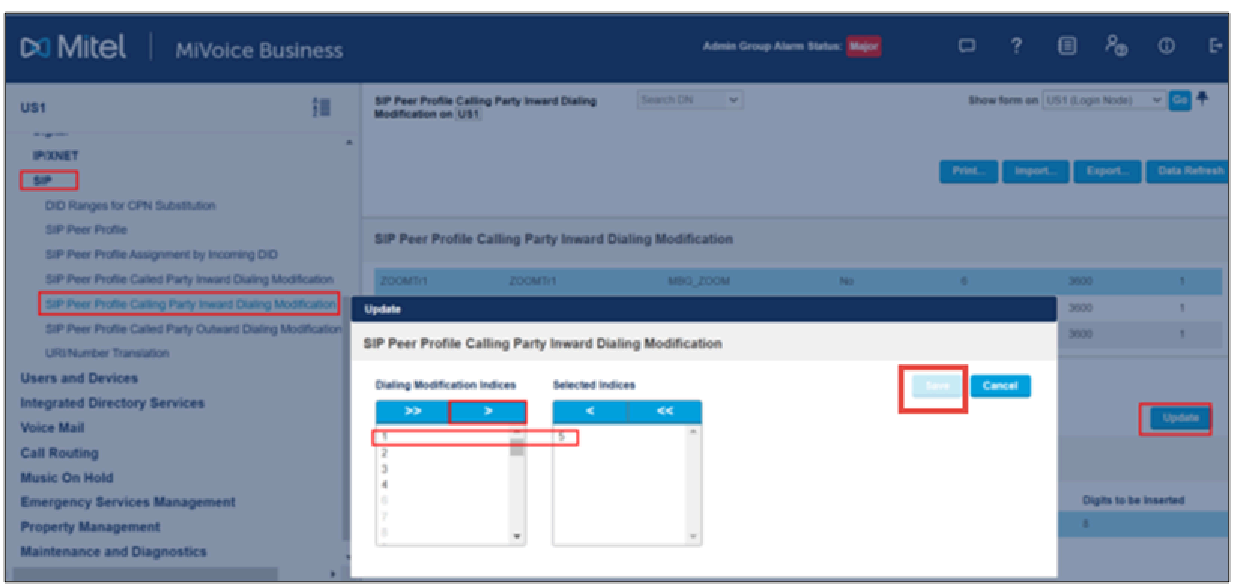
    a. **Index**: Remember the default Index number. (Default is **5** here).
    b. Set **Digits to Match:** Enter **x** (any) as a wild card to match any valid incoming digit (0-9, *, or #).
    c. **Digit Length Operator**: From the drop-down menu, select **greater than or equal** to option.
    d. **Digit Length**: Enter the digit length as **11**.
    e. **Number of Digits to Absorb:** Enter **0**.
    f. **Digits to be Inserted**: Enter **8** (This is our prefix).
    g. Click **Save.**

5.2.6.2 Adding SIP Peer Profile Called Party Inward Dialing Modification

The **Inward Dialing Modification** rule was created in the previous section. Now, it can be applied to the **SIP Peer Profiles** for each Zoom Region by following the below steps.

1. Navigate to **SIP** > **SIP Peer Profile Calling Party Inward Dialing Modification** > **Inward Dialing Modification** form**.**
2. Click **Update**.
3. This opens a selection table where the index (in this case, #5) can be copied by pressing the "**>**" button, transferring it to the Selected Indices table.
4. Click **Save**.

> **Note**:
> Enabling this option allows you to dial either and extension number or a PSTN number from the Zoom clients.



5.2.7 ARS Digit Modification

1. Navigate to **Call Routing** > **Automatic Route Selection (ARS)** > **ARS Digit Modification Plans** form**.**
2. To Configure the PSTN Trunk,

    a. **Digit Modification Number**: 2 is selected for this example (**PSTN trunk**)
    b. Click **Change**.
    c. **Number of Digits to Absorb**: Enter **1**.
    d. Leave all other fields as default.
    e. Click **Save.**
3. To Configure the Zoom Trunk,

    a. **Digit Modification Number**: 10 is selected for this example (**ZOOM trunks**)
    b. Click **Change**.
    c. **Number of Digits to Absorb**: Enter **0**.
    d. Leave all other fields as default.
    e. Click **Save**.

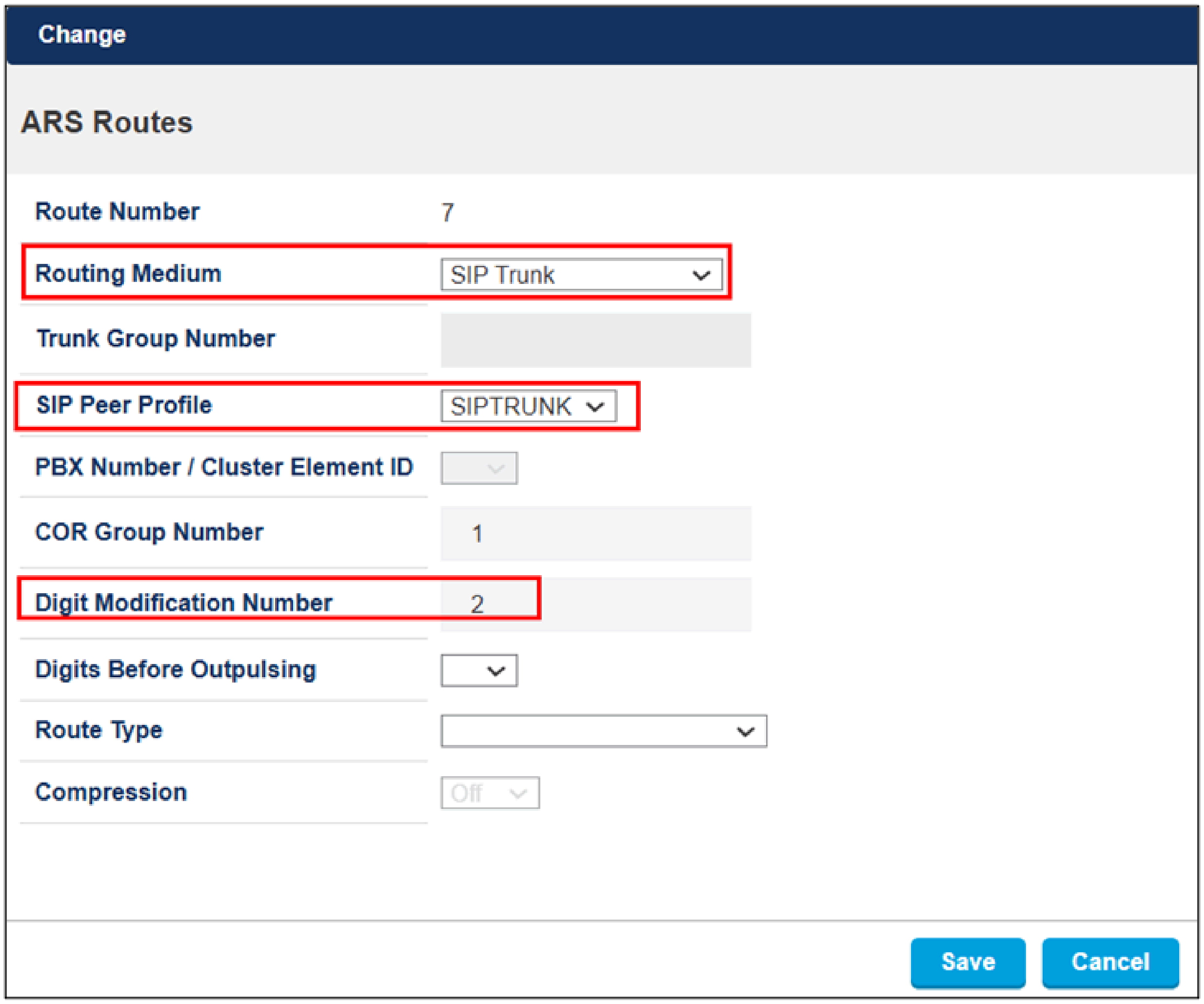
5.2.7.1 Configuring the ARS Routes

To Configure ARS Routes, follow the below mentioned steps:

1. Navigate to **Call Routing** > **Automatic Route Selection** > **ARS Routes** form.
2. Click **Change**.

To Configure Route for PSTN Destination:

1. Select the desired **Route Number**: **7** is used as an example **for PSTN trunk**.
2. Click **Change**
3. **Routing Medium**: Select **SIP Trunk** from the drop-down menu.
4. **SIP Peer Profile**: Select newly created SIP Peer profile **SIPTRUNK** from the drop down menu.
5. **Digit Modification Number**: 2 (**for PSTN Trunk**) which was created in ARS Digit Modification section.
6. Leave all other fields as default
7. Click **Save**



To Configure Route for Zoom Destination:

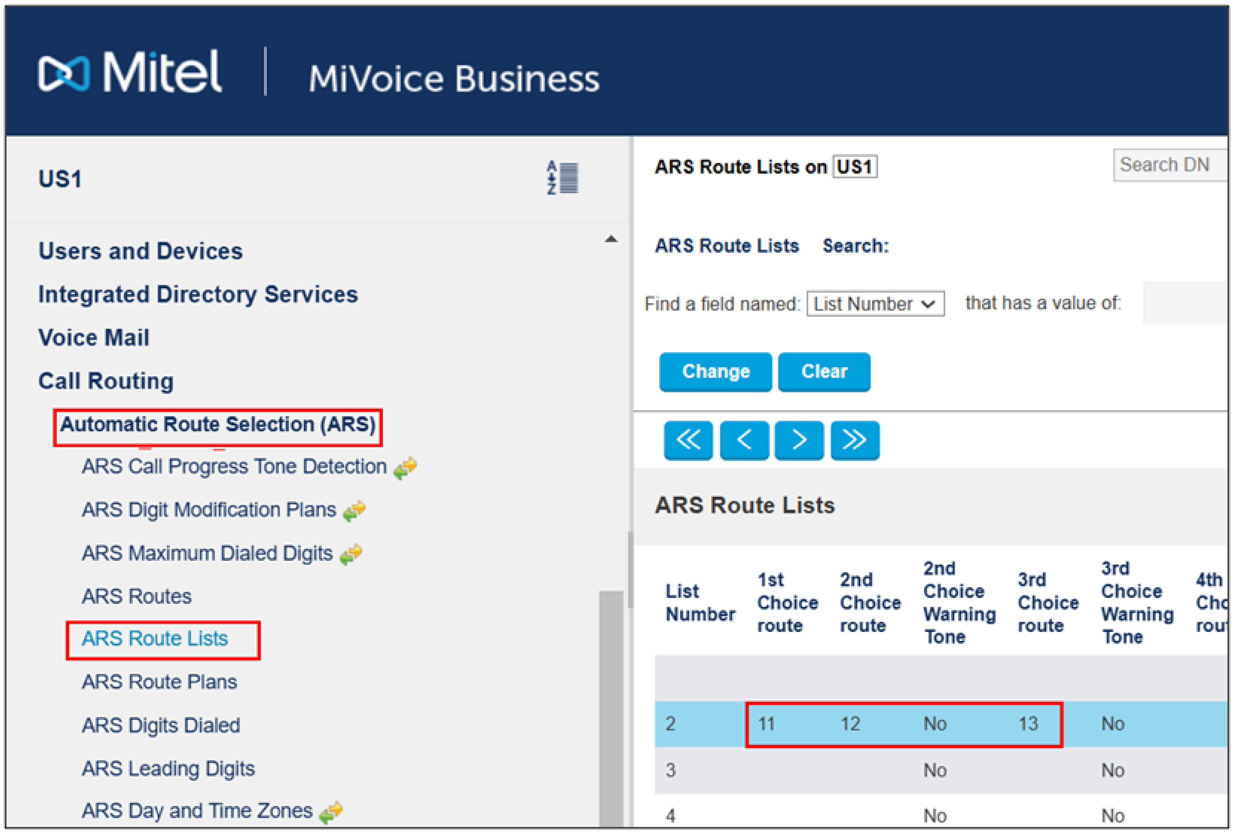
1. Select the desired **Route Number**: **11** is used as an example **for ZOOMTr1**. (Route Number 12 and 13 is used for ZOOMTr2 and ZOOMTr3 respectively)
2. Click **Change**.
3. **Routing Medium**: Select **SIP Trunk** from the drop-down menu.
4. **SIP Peer Profile**: Select **ZOOMTr1** from the drop-down menu for Zoom Trunk 1.
5. **Digit Modification Number**: 10 (**for Zoom Trunk**) which was created in ARS Digit Modification section.
6. Leave all other fields as default
7. Click **Save**

Similarly, Configure the ARS Route for the other Zoom Trunk regions. (Example: **ZOOMTr2** and **ZOOMTr3**).

5.2.7.2 ARS Route List

Add all the three **ARS route** created for Zoom Trunks to a single group.

1. Navigate to **Call Routing** > **Automatic Route Selection** > **ARS Route Lists** form.
2. In the **Change Range Programming - ARS Route Lists** form, add the route numbers of the 1st, 2nd and 3rd choice route.
3. Click **Save**.



5.2.7.3 ARS Digits Dialed

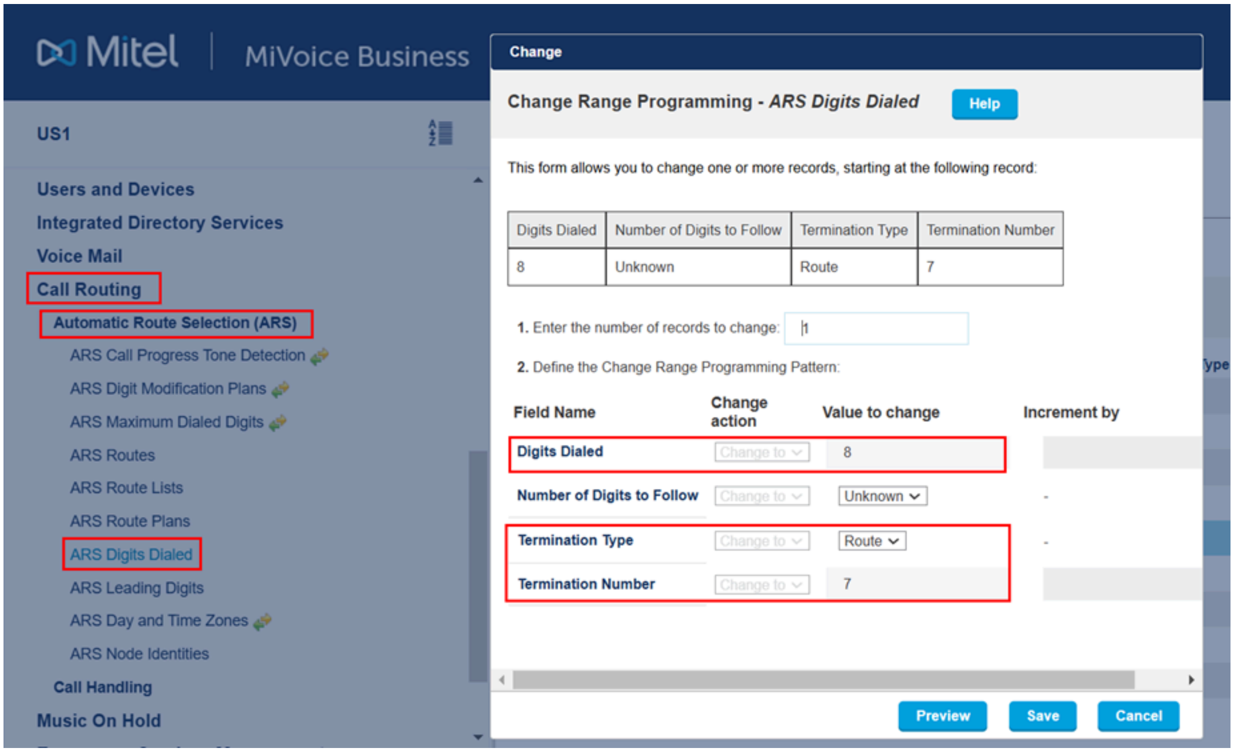
1. Navigate to **Call Routing** > **Automatic Route Selection** > **ARS Digits Dialed** form.

Route calls towards PSTN:

Dialing 8 followed by a number from Zoom Client is used to route call towards PSTN.

To configure the route call towards PSTN:

1. Click **Add**.
2. **Digits Dialed**: Enter **8**.
3. **Number of Digits to Follow**: Select **Unknown,** from the drop-down menu.
4. **Termination Type**: Select **Route**, from the drop-down menu.
5. **Termination Number**: Route Number **7** is used here.
6. Click **Save**

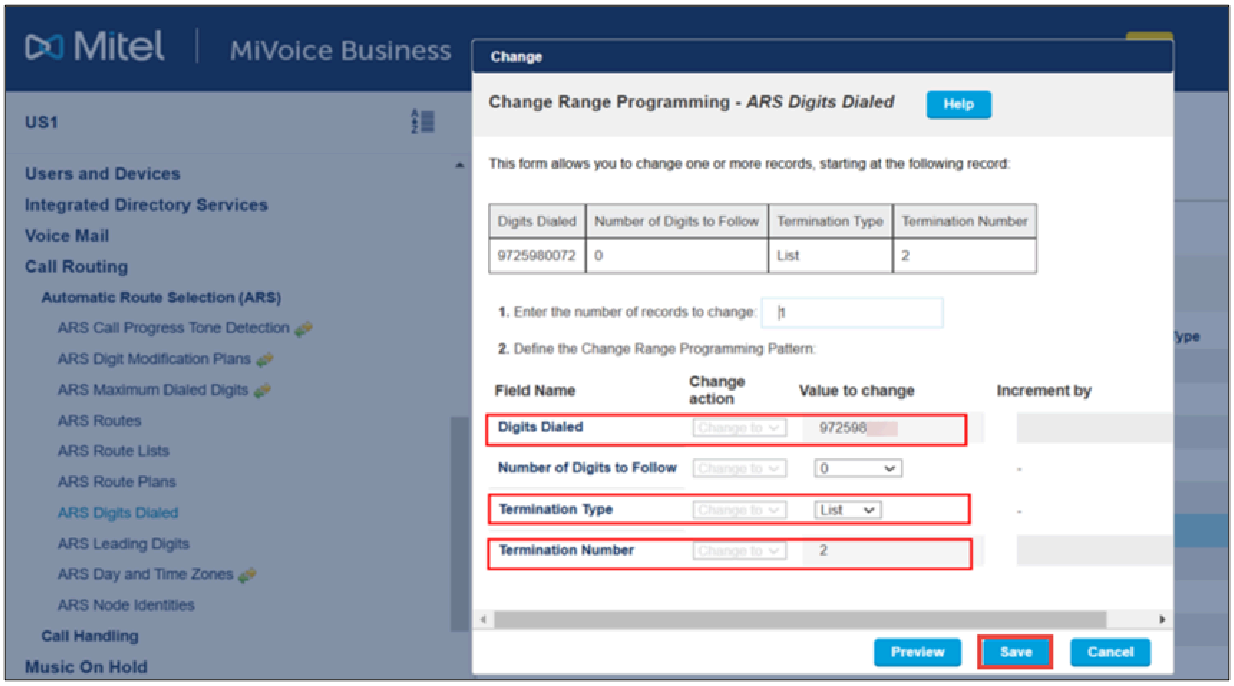


Route calls from PSTN:

Use newly created **ARS Route Lists** to route calls towards Zoom Clients from PSTN.

To configure the route call from PSTN:

1. Click **Add**.
2. **Digits Dialed**: Enter the number **972598XXXX** (DID assigned to Zoom Client).
3. **Number of Digits to Follow**: Select **0,** from the drop-down menu.
4. **Termination Type**: Select **List**, from the drop-down menu.
5. **Termination Number**: List Number **2** is used here.
6. Click **Save**


Mitel
MiVoice Business
Change
Change Range Programming - ARS Digits Dialed
Help
US1
Users and Devices
Integrated Directory Services
Voice Mail
Call Routing
Automatic Route Selection (ARS)
ARS Call Progress Tone Detection
ARS Digit Modification Plans
ARS Maximum Dialed Digits
ARS Routes
ARS Route Lists
ARS Route Plans
ARS Digits Dialed
ARS Leading Digits
ARS Day and Time Zones
ARS Node Identities
Call Handling
Music On Hold
This form allows you to change one or more records, starting at the following record:
Digits Dialed
Number of Digits to Follow
Termination Type
Termination Number
9725980072
0
List
2
1. Enter the number of records to change:
1
2. Define the Change Range Programming Pattern:
Field Name
Change action
Value to change
Increment by
Digits Dialed
972598
Number of Digits to Follow
0
Termination Type
List
Termination Number
2
Preview
Save
Cancel