



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

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

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.

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.

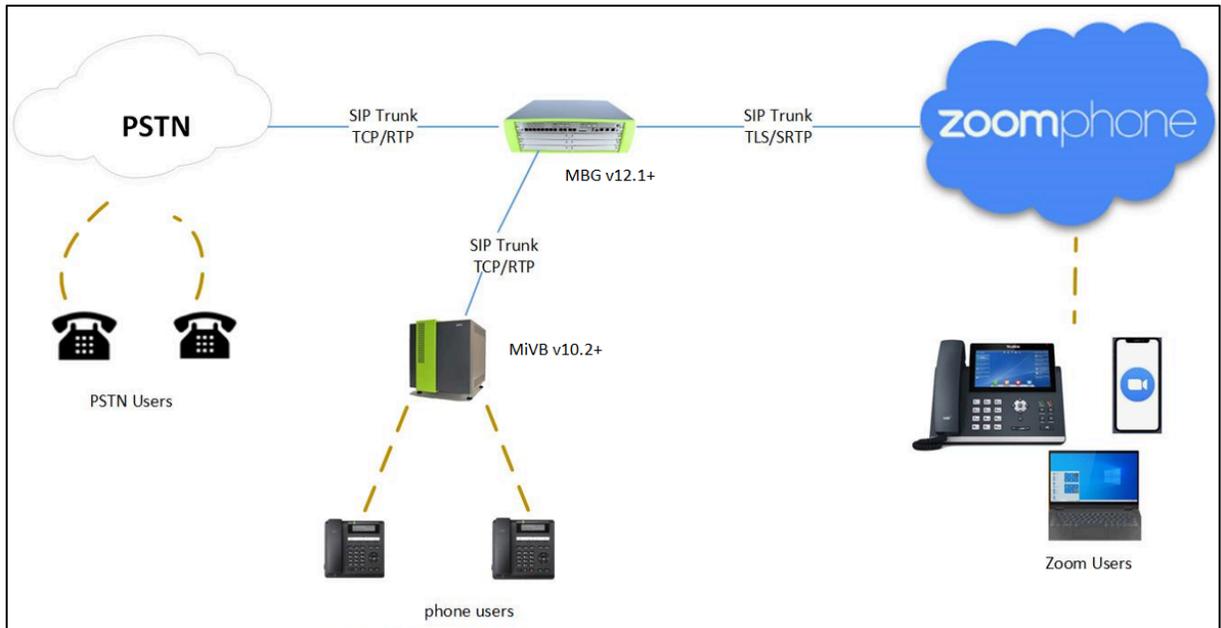


Figure 1: Network Topology Block Diagram

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.

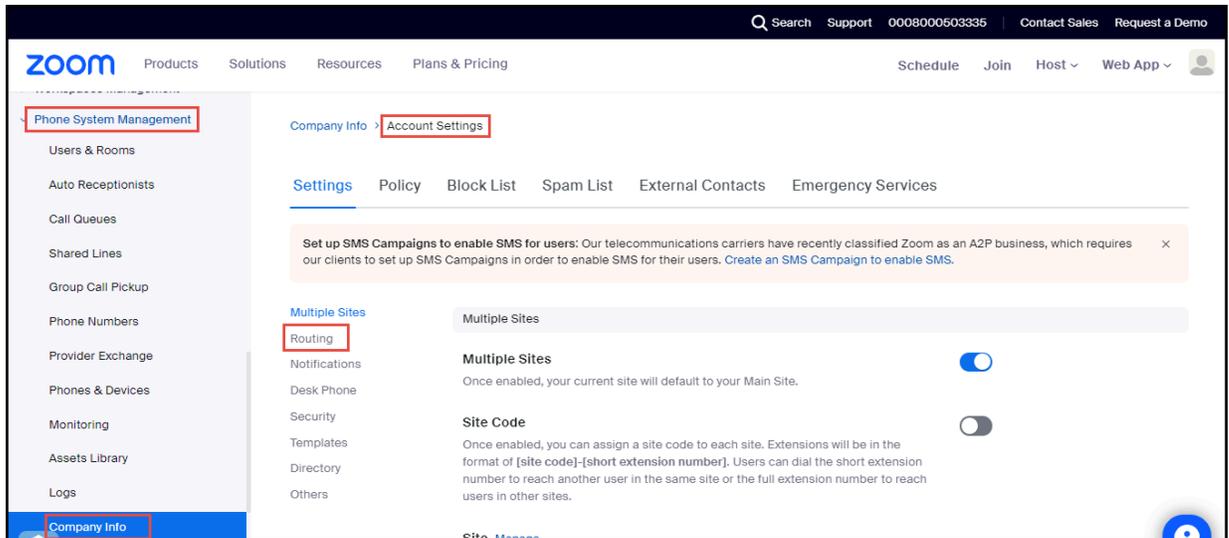
Prerequisites

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.

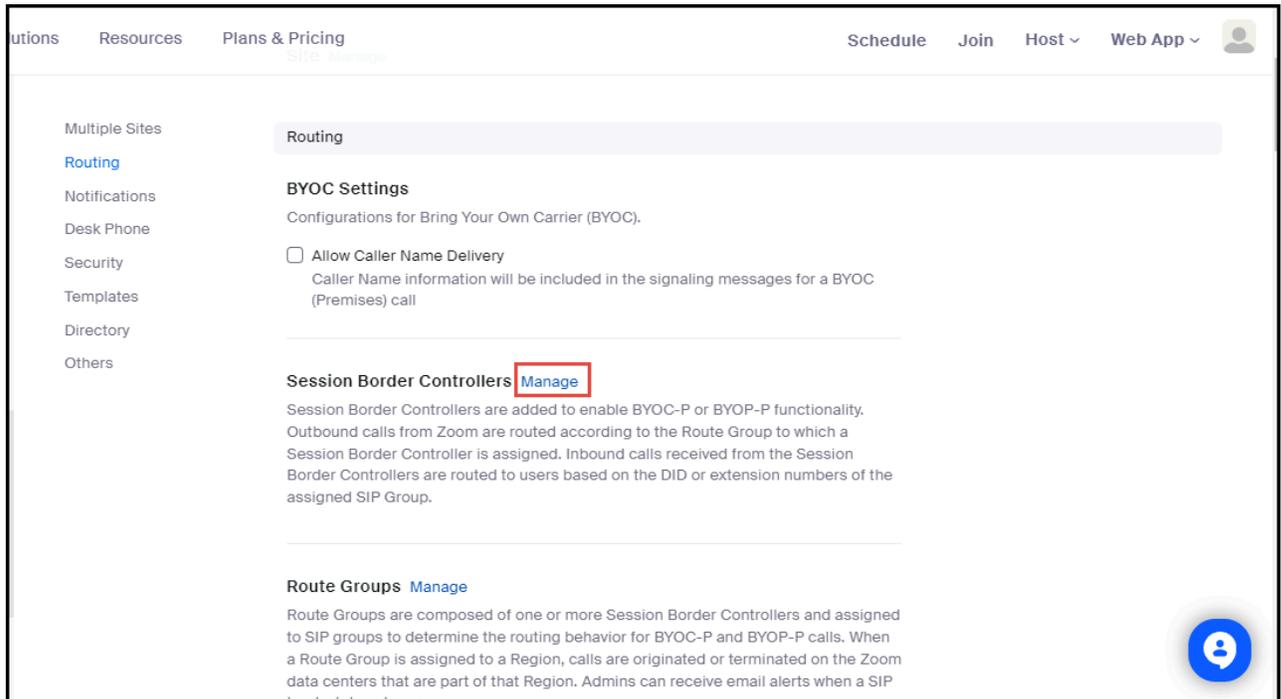
Adding your SBC

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 Your 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**

The screenshot shows a configuration form for Zoom. The following fields and sections are highlighted with red boxes:

- Display Name:** A text input field containing "MITEL_MBG_ZOOM".
- Description (Optional):** An empty text area.
- Save/Cancel:** Two buttons below the description field.
- Protocol:** A dropdown menu set to "TLS".
- IP Address:** A section containing three input fields: "IP Address" (empty), "Public IP Address" (containing "192"), and "Port Number" (containing "5061").
- In-Service:** A toggle switch that is turned on.
- Settings:** A section with five checkboxes:
 - Integrate an on-premises PBX (Bring Your 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
 - Include original calling number within the P-Asserted-Identity (PAI) header for forwarded calls
 - Use T.38 protocol for faxing
 - Allow REFER support to transfer calls **BETA**
- Address(Optional):** A field with a "Set" link.
- Email(Optional):** A field with a "Set" link.
- Phone Number(Optional):** A field with a "Set" link.
- Save/Close:** Two buttons at the bottom left.

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

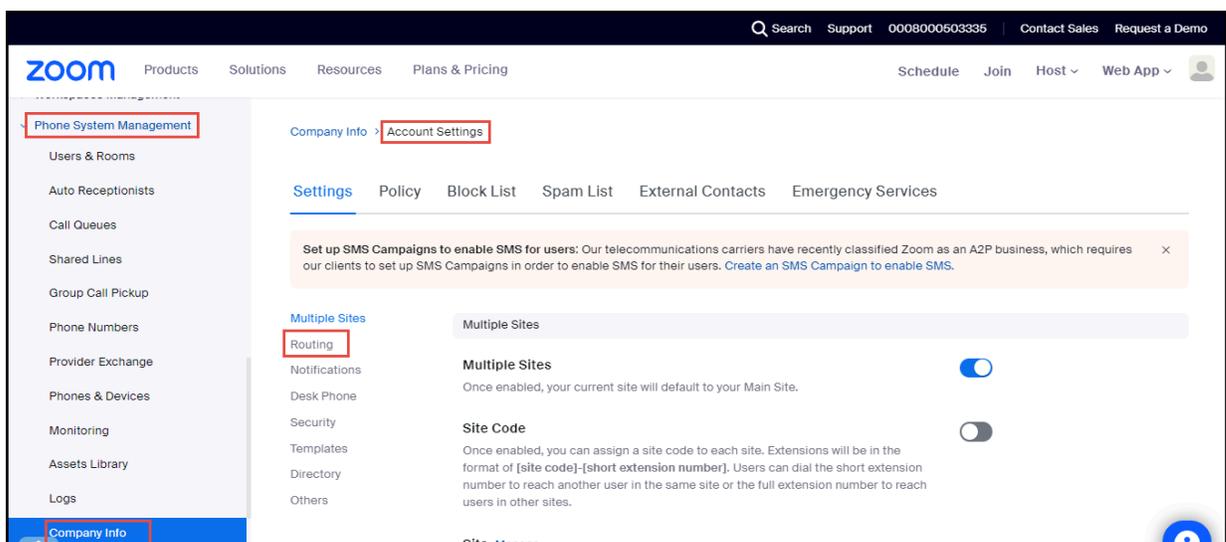
Route Groups are collections of Session Border Controllers (SBCs), which manage and route voice traffic across a network. A Route Group determines how calls are routed and handled by directing them to specific SIP endpoints. The **Region** setting ensures that calls are routed through the appropriate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

Note:

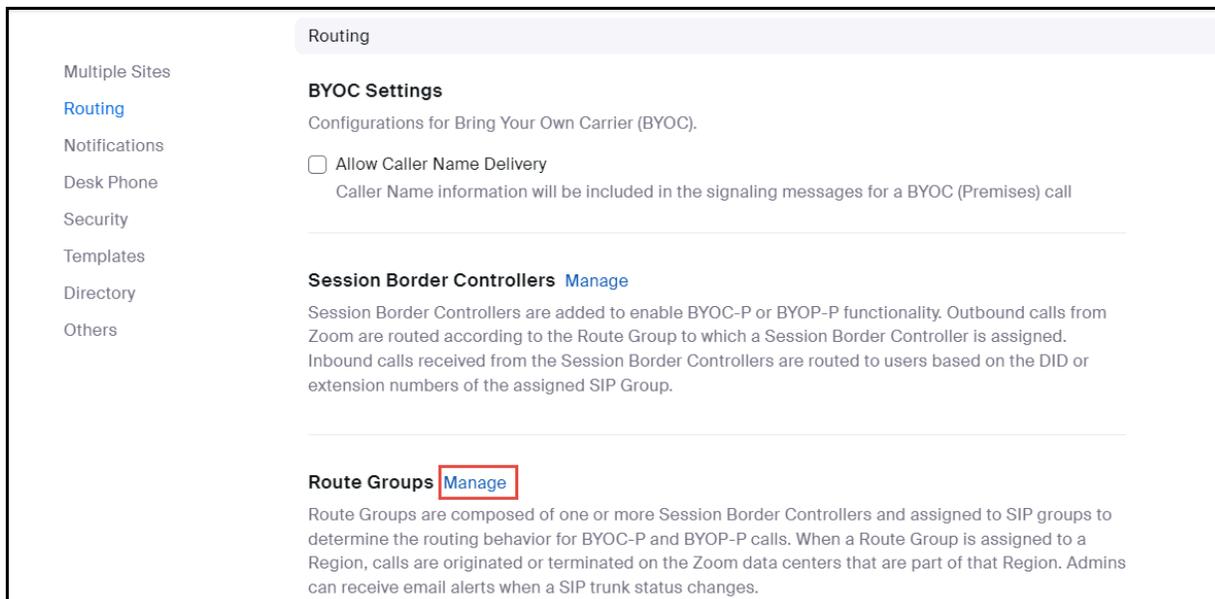
These configurations (Route Group, SIP Group, and Routing Rule) will take effect once phone numbers are added and assigned to the appropriate users. Until then, the routing logic will be in place, but calls will not be routed as expected.

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**.



Routing

Multiple Sites
Routing
Notifications
Desk Phone
Security
Templates
Directory
Others

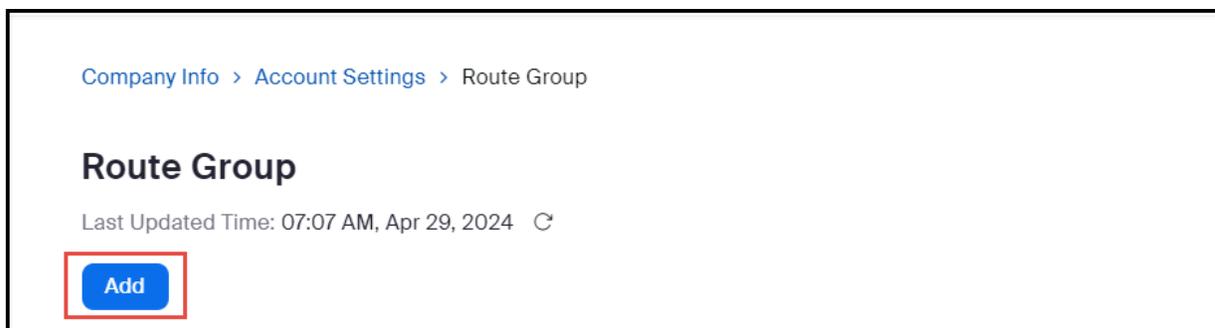
BYOC Settings
Configurations for Bring Your Own Carrier (BYOC).

Allow Caller Name Delivery
Caller Name information will be included in the signaling messages for a BYOC (Premises) call

Session Border Controllers [Manage](#)
Session Border Controllers are added to enable BYOC-P or BYOP-P functionality. Outbound calls from Zoom are routed according to the Route Group to which a Session Border Controller is assigned. Inbound calls received from the Session Border Controllers are routed to users based on the DID or extension numbers of the assigned SIP Group.

Route Groups [Manage](#)
Route Groups are composed of one or more Session Border Controllers and assigned to SIP groups to determine the routing behavior for BYOC-P and BYOP-P calls. When a Route Group is assigned to a Region, calls are originated or terminated on the Zoom data centers that are part of that Region. Admins can receive email alerts when a SIP trunk status changes.

3. Click **Add**.



Company Info > Account Settings > Route Group

Route Group

Last Updated Time: 07:07 AM, Apr 29, 2024 [🔄](#)

[Add](#)

4. Configure the following:

- Display Name:** Type the display name of your choice. For example, **PSTN_MBG_ZOOM**.
- From the **Type** drop-down menu, select **BYOC-P**.
- 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.

Edit Route Group

Display Name	<input type="text" value="PSTN_MBG_ZOOM"/>
Type	<input type="text" value="BYOP-P"/>
Region	<input type="text" value="US01 - US (SJ/DV/NY)"/>
Distribution	<input type="text" value="Sequential"/>

Session Border Controllers

1: [Add](#)

Backup Route Group (Optional)

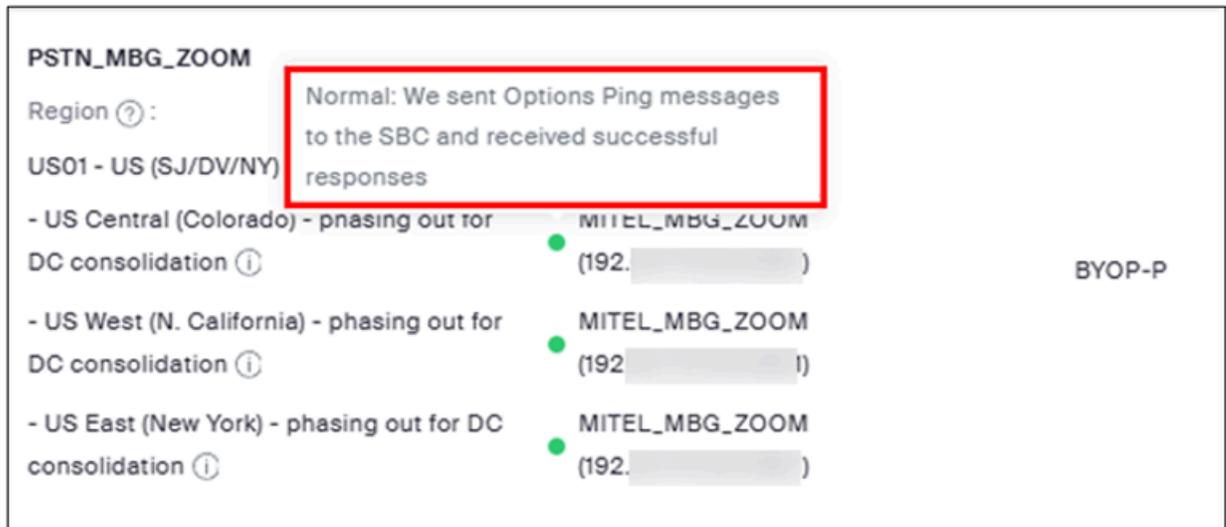
5. Click **Save**.

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

The screenshot shows the Zoom web portal interface for configuring a Route Group. The left sidebar contains navigation options, with 'Company Info' selected. The main content area shows the 'Route Group' configuration page, including a search bar and a table of route groups.

Display Name	Session Border Controllers	Type
PSTN_MBG_ZOOM		
Region: US01 - US (SJ/DV/NY)	Sequential:	
- US Central (Colorado) - phasing out for DC consolidation	● MITEL_MBG_ZOOM (192.)	BYOP-P
- US West (N. California) - phasing out for DC consolidation	● MITEL_MBG_ZOOM (192.)	
- US East (New York) - phasing out for DC consolidation	● MITEL_MBG_ZOOM (192.)	
Group-Mitel_MBG_ZOOM		
Region: US01 - US (SJ/DV/NY)	Sequential:	
- US Central (Colorado) - phasing out for DC consolidation	● MITEL_MBG_ZOOM (192.1)	BYOC-P
- US West (N. California) - phasing out for DC consolidation	● MITEL_MBG_ZOOM (192.1)	
- US East (New York) - phasing out for DC consolidation	● MITEL_MBG_ZOOM (192.31)	

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

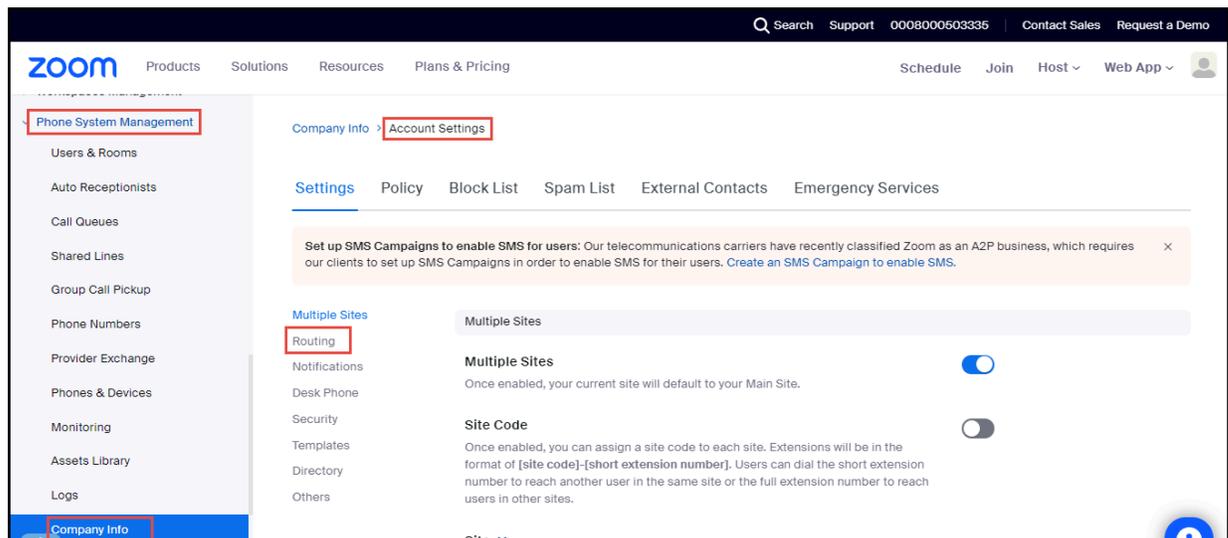


3.1.2 Configuring the SIP Group

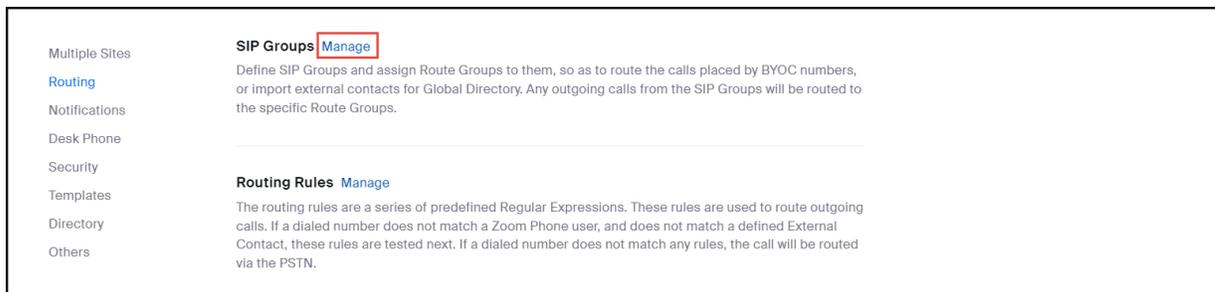
Follow the instructions below to configure SIP groups and assign Route Groups to them, in order to route calls placed by BYOC numbers. This step is mandatory for uploading the BYOC numbers.

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

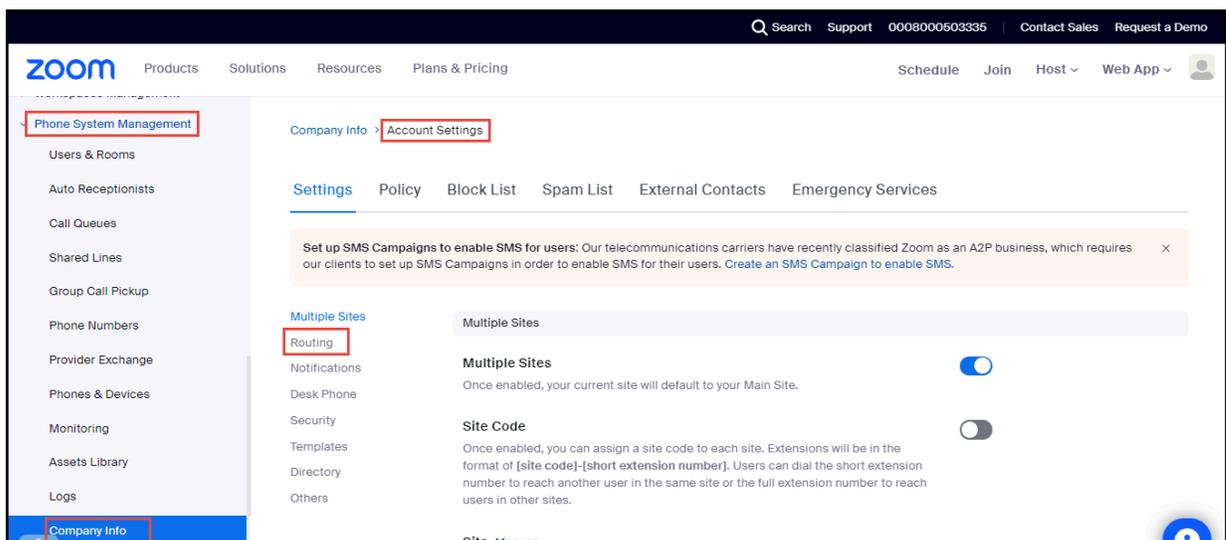
When configuring a **BYOC (Bring Your Own Carrier)** setup, you might create a routing rule to specify that calls from certain users or departments go through your MBG or network route. To add a Routing Rule for outbound calls:

Note:

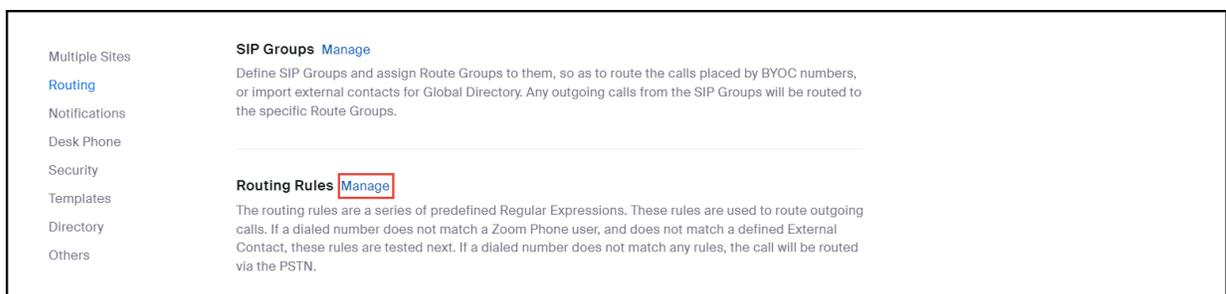
Ensure that your Session Border Controller is properly configured and connected before setting up routing rules. Additionally, phone users must be provisioned and assigned to the correct phone numbers for routing rules to function correctly.

To add the Routing Rule:

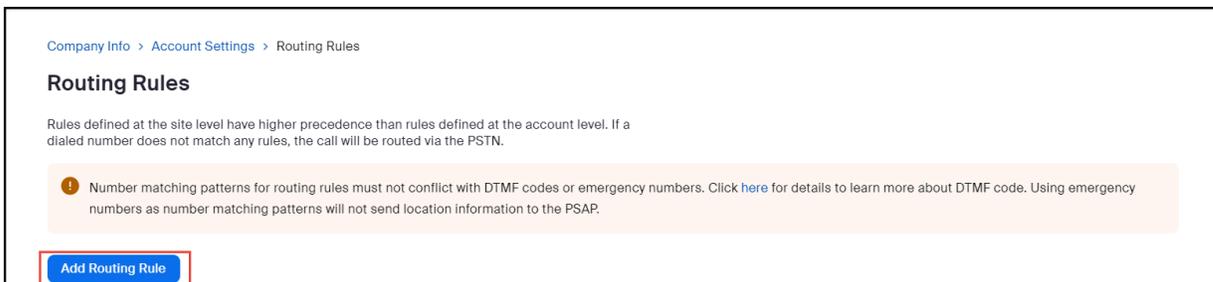
1. Navigate to **Phone System Management > Company Info > Account Settings > Routing**.



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

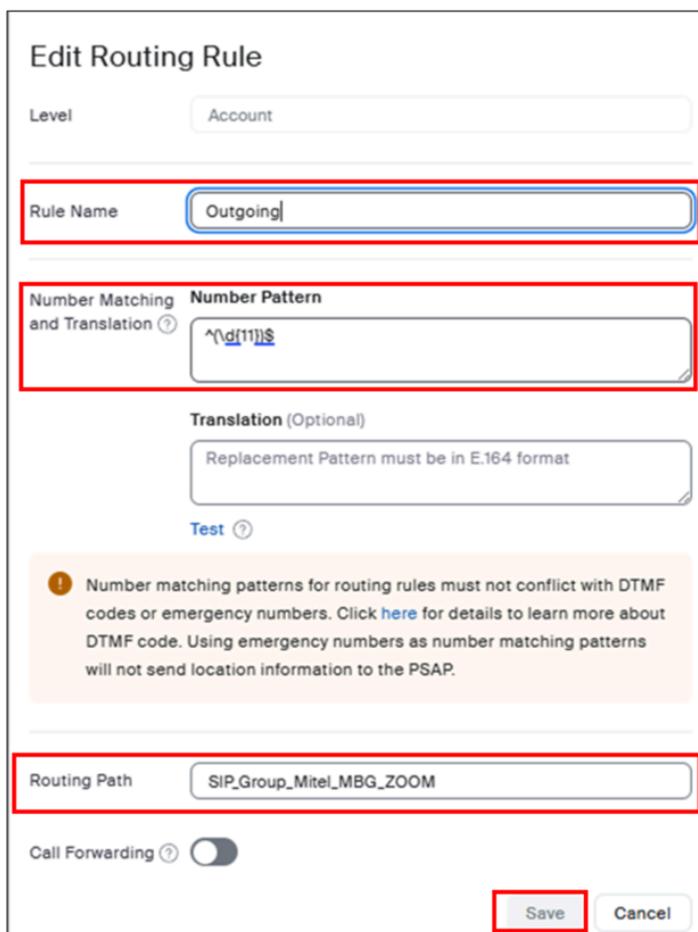


3. Click **Add Routing Rule** to add your rule for outbound calls.



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**.

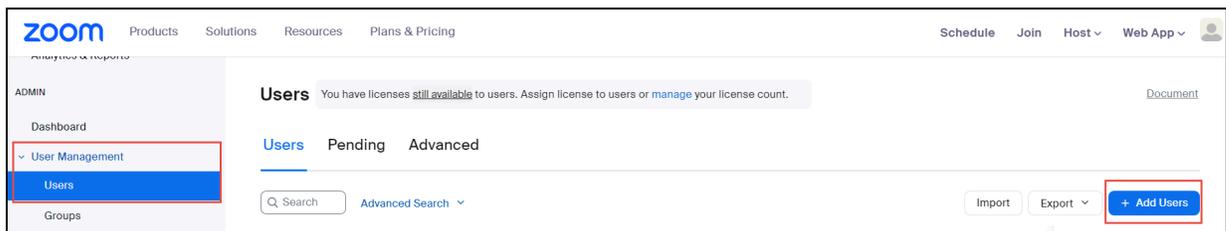
3.2 Adding Phone Users

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**.

Add Users

Add users with their email addresses

If you enter the email address of account owners, all users on their accounts will be added to this account.

Zoom Workplace

Licenses and add-ons

Large Meeting (500 participants) (20 available)

Zoom Phone Basic

 To assign Zoom Phone packages, go to [Phone System Management](#).

Zoom Webinars (500 attendees) (20 available)

Department

Manager

Job Title

Location

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:

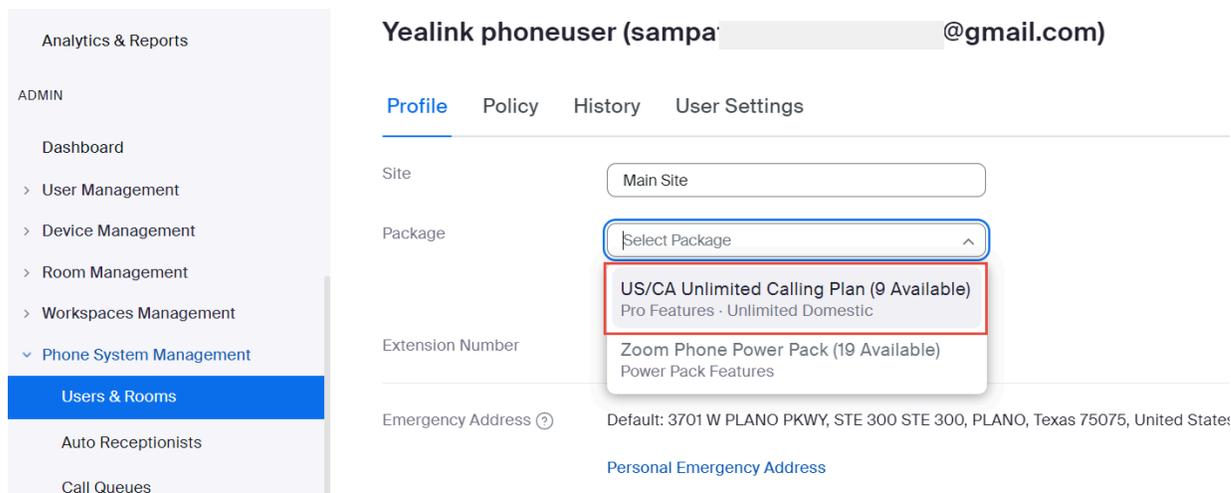
When a Zoom Phone license is assigned to a user, an extension number is automatically assigned to them.

The screenshot shows the Zoom Web Portal navigation menu on the left. The 'Phone System Management' option is expanded, and 'Users & Rooms' is highlighted in blue. A red box highlights the 'Users & Rooms' link in the sidebar and the corresponding user entry in the main content area. The user entry is 'Yealink phoneuser' with extension number 1084, and it is marked as 'Online' and 'Active'.

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

The screenshot shows the Zoom Web Portal user profile page for 'Yealink phoneuser (sampa@gmail.com)'. The 'Profile' tab is selected, and the 'Package' section is highlighted. The 'Assign' button is highlighted with a red box. The 'Package' section shows 'Zoom Phone Basic (Migrated)' with an 'Assign' button. The 'Extension Number' is 1084, and there is an 'Edit' link next to it.

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.

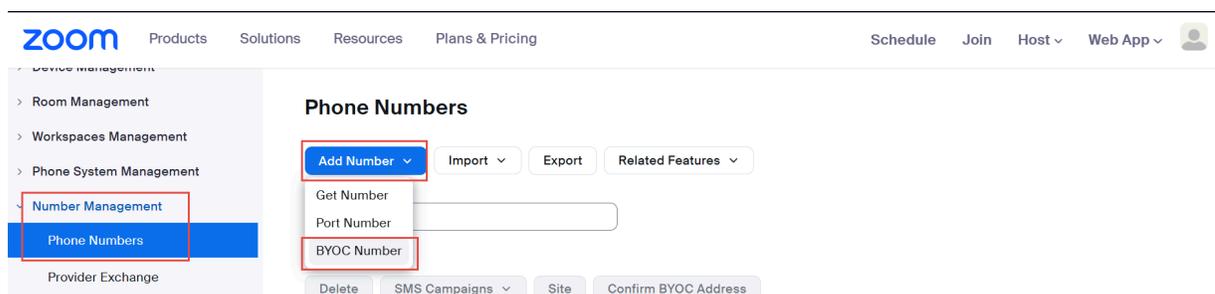
3.3 Adding BYOC Phone Numbers

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**.

Add BYOC Number

Product

Site

Country/Region

Numbers

SIP Group ?
Choose a routing path for calls to/from the numbers

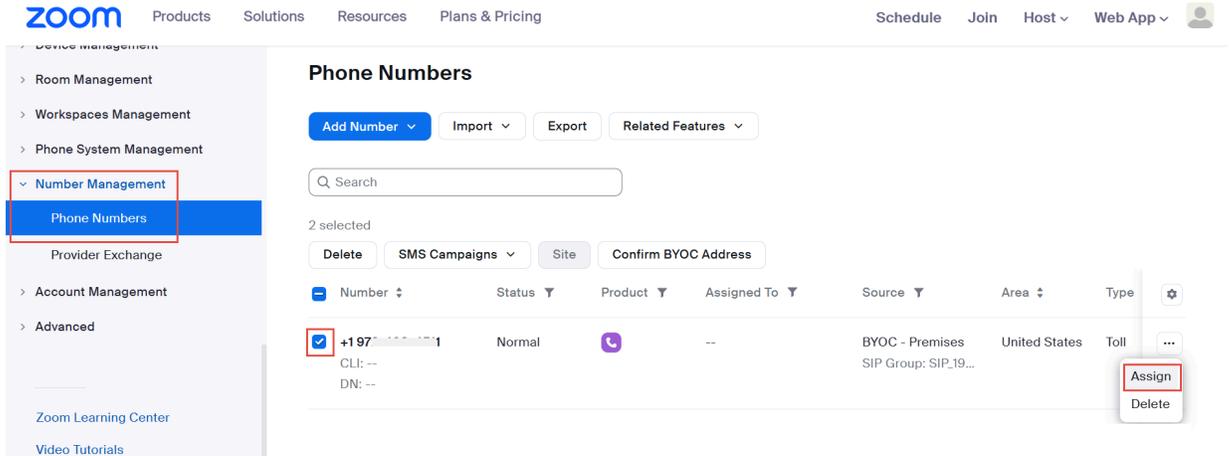
I acknowledge that by checking the box, I attest that the phone numbers to be imported belong to me or my organization

Submit

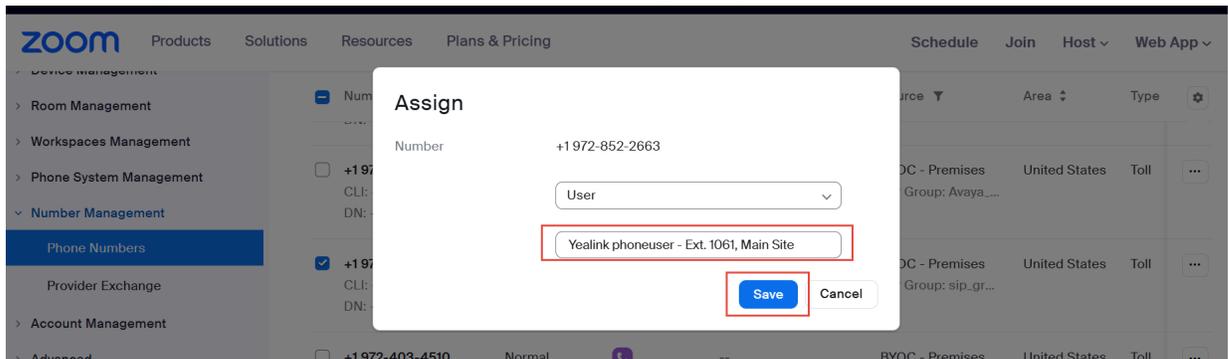
Assigning BYOC numbers

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.

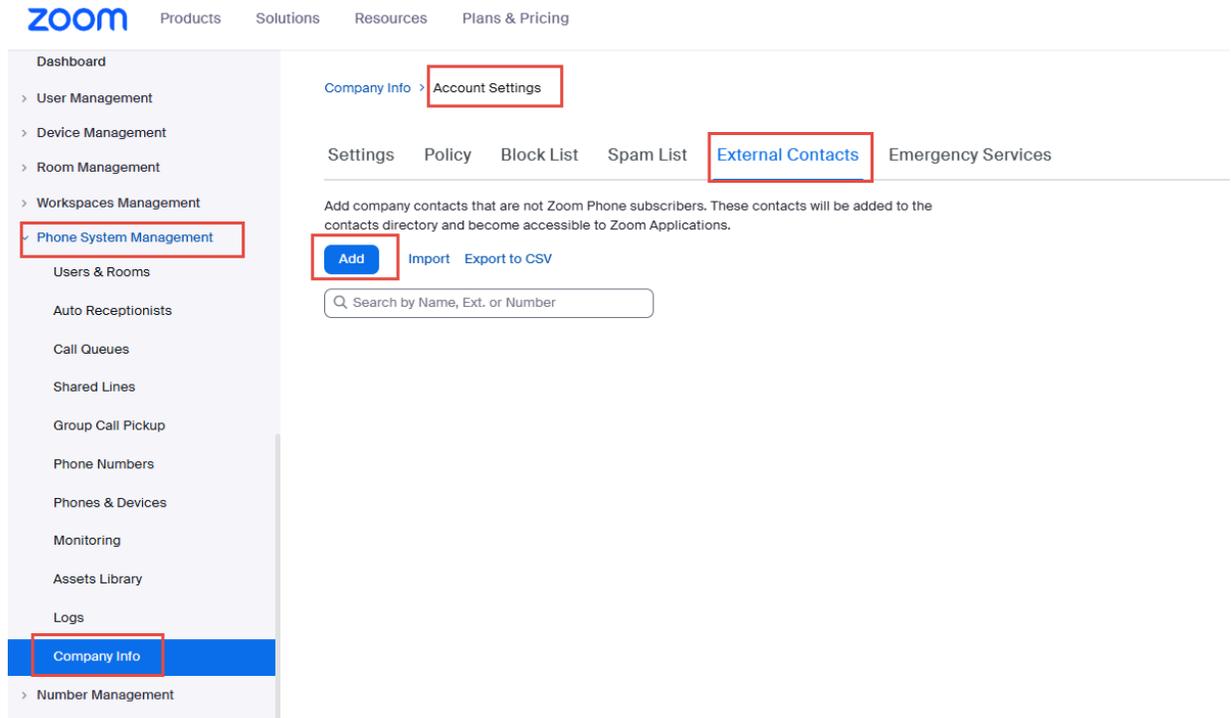
3.4 Adding BYOP numbers

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.



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.

The screenshot shows the 'Edit External Contact' form with the following fields and values:

- ID** (with a help icon): 2
- Name**: Mivoice_user2 (highlighted with a red border)
- Email (Optional)**: (empty)
- Extension Number (Optional)** (with a help icon): 2000 (highlighted with a red border)
- Phone Number (Optional)** (with a help icon): Enter in the E.164 format. Separated by commas.
- Description (Optional)**: (empty)
- Routing Path (Optional)** (with a help icon): SIP_Group_Mitel_MBG_ZOOM (highlighted with a red border)
- Auto Call Recorded** (with a help icon): (toggle switch is turned off)

At the bottom right, there are two buttons: **Save** (highlighted with a red border) and **Cancel**.

4. Click **Save**.

Add Device

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

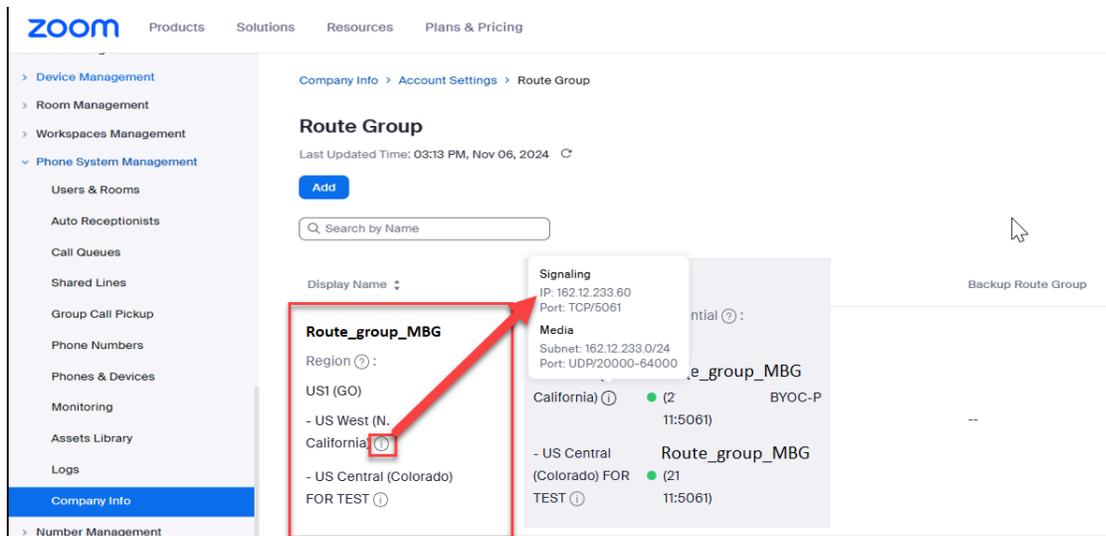
Initial releases of MiVoice Border Gateway, for Zoom DO NOT require a Zoom BYOC/BYOP license. However, this license will be required for future releases. During this transition, MiVoice Border Gateway Zoom BYOC/BYOP licenses will NOT BE NEEDED as part of the Zoom subscription.

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").

i 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**.

The screenshot shows the 'Configure Web Server' page in the Mitel Standard Linux management interface. The left sidebar contains a navigation menu with categories: Applications, ServiceLink, Administration, Security, Web Server, Configuration, and Miscellaneous. The 'Web Server' menu item is highlighted with a red box. The main content area is titled 'Configure Web Server' and has three tabs: 'Web Server Certificate', 'Certificate Authority Trust' (which is selected and highlighted with a red box), and 'Advanced'. Below the tabs, the section is titled 'Manage Certificate Authority Trust'. A paragraph explains that a default set of publicly trusted root CA certificates exists on the server and that the functions below can be used to update the trusted CA certificate store. Below this, a table lists currently installed additional root CA certificates:

Certificate Name	Action
DigiCert Global Root G2	<input type="button" value="Remove"/>
Mitel Networks Root CA	<input type="button" value="Remove"/>
Mitel Products Root CA	<input type="button" value="Remove"/>

Below the table, a note states: 'To upload new root CA certificates to the installed CA trust bundle, choose your certificate file below and click the install button.' Another note specifies: 'Note: the file must contain X.509 root CA certificates in PEM format.' At the bottom of the main content area, there is a 'Root CA Certificate:' label followed by a 'Choose File' button and the text 'No file chosen'. The 'Choose File' button is highlighted with a red box. The footer of the page displays system information: 'Mitel Standard Linux 12.1.17.0', 'MiVoice Border Gateway 12.1.0.110', and '© Mitel Networks Corporation'.

3. Select the certificates to be uploaded.

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

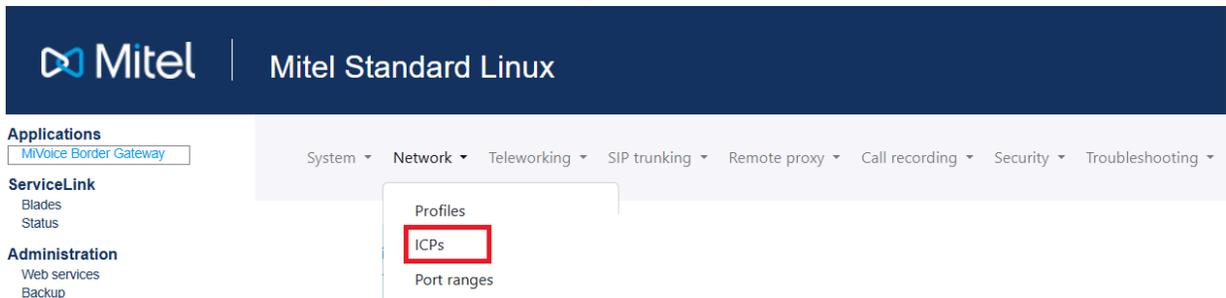


5.1.4 Adding a Network ICP

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:

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



- 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**.

Page updated: Tue Feb 04 2020 01:44:03 GMT-0800 (Pacific Standard Time)
The following is a form for modifying an icp entry. You may edit this information as you wish, and click on the "Save" button below when you are done.

Manage ICP

Name: ZOOMMIVB

Type: MiVoice Business

SIP capabilities: UDP, TCP, TLS Export root cert

Hostname or IP address: 10.35.32.2

MiNet installer password: [password field]

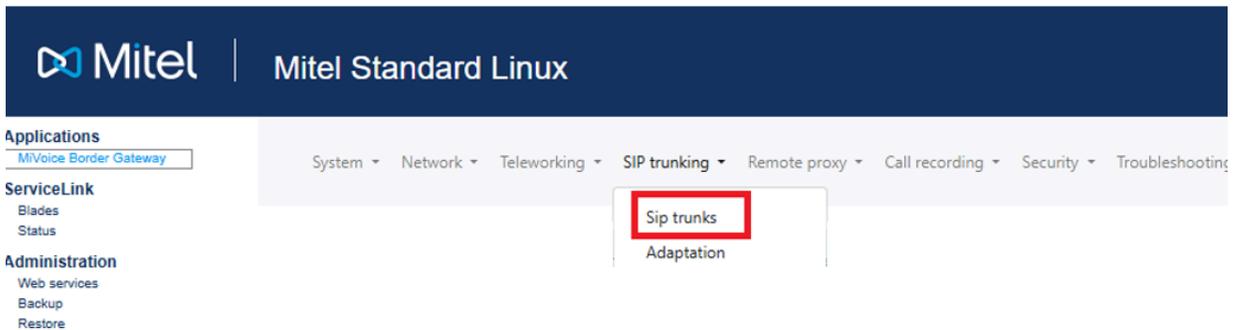
Indirect call recording capable:

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.

SIP Trunk Rules

- Click on the trash icon to delete a rule.
- Click on the plus icon to insert a rule.
- Note that the rules order displayed is irrelevant. This list is parsed into a sorted data structure internally regardless of the rule number.
- After creating or modifying a rule, click the indicator in the first column to save it. Or, click the Done button and you will be prompted to save any unsaved rule.

Done

Loaded rules count: 1

Add rule +

	Header match	Rule	Primary ICP	Secondary ICP	Description	
1	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	
	<input type="button" value="Click to save"/>					

3. Configure the following:

- Header match:** From the drop-down menu, select **Request URI**.
- Rule :** Enter *
- Primary ICP:** select the MiVB Name you configured in section [Adding a Network ICP](#) from the drop-down menu. (Example: **ZOOMMIVB**).
- 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.

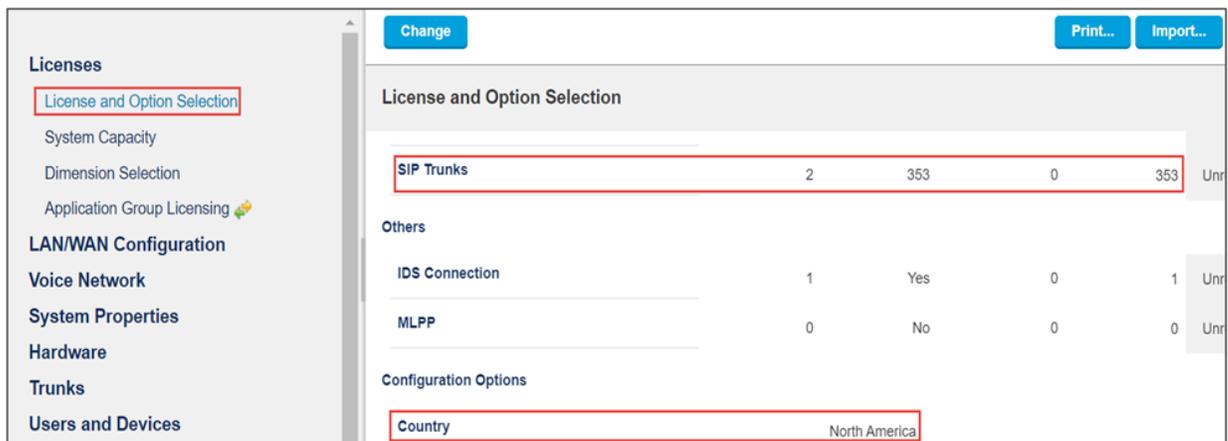


Figure 2: Licensing

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 of Service Number** : Enter **6** for ZOOMTr1.
4. **Comment** : Enter the Zoom trunk name. (Example: **ZOOMTr1**).

The screenshot displays the 'Class of Service Options' configuration page. At the top, there is a navigation bar with 'Page 1 of 11', a 'Go to' field, and a 'Go' button. Below this is a table with one entry: Class of Service Number 6 and Comment ZOOMTr1. The table is highlighted with a red border. Below the table are two tabs: 'General' and 'Advanced'. The 'General' tab is active. Below the tabs are several configuration options for ACD and Announce. The 'Class of Service Number' and 'Comment' fields are highlighted with a red border. The 'ACD' section includes options like 'ACD Agent Behavior on No Answer' (Logout), 'ACD Agent No Answer Timer' (15), 'ACD Make Busy on Login' (No), 'ACD Silent Monitor Accept' (No), 'ACD Silent Monitor Accept Monitoring Non-Prime Lines' (No), 'ACD Silent Monitor Allowed' (No), 'ACD Silent Monitor Notification' (No), and 'Follow 2nd Alternate Reroute for Recall to Busy ACD Agent' (No). The 'Work Timer' is set to 0. The 'Announce' section includes 'Call Announce Line' (No).

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

6. Leave all other fields as default.

Recorded Announcement Device	<input checked="" type="radio"/> No <input type="radio"/> Yes
Recorded Announcement Device - Advanced	<input checked="" type="radio"/> No <input type="radio"/> Yes
Ringing	
Allow Recall after Transfer	<input checked="" type="radio"/> No <input type="radio"/> Yes
Delay Ring Timer	10
No Answer Recall Timer	17
Ringing Line Select	<input checked="" type="radio"/> No <input type="radio"/> Yes
Ringing Timer	180
SMDR	
SMDR External	<input checked="" type="radio"/> No <input type="radio"/> Yes
SMDR Internal	<input checked="" type="radio"/> No <input type="radio"/> Yes
Trunk	
ANI/DNIS/ISDN Number Delivery Trunk	<input checked="" type="radio"/> No <input type="radio"/> Yes
DASS II OLI/TLI Provided	<input checked="" type="radio"/> No <input type="radio"/> Yes
Public Network Access via DPNSS	<input type="radio"/> No <input checked="" type="radio"/> Yes
Public Network To Public Network Connection Allowed	<input type="radio"/> No <input checked="" type="radio"/> Yes
Public Trunk	<input type="radio"/> No <input checked="" type="radio"/> Yes
R2 Call Progress Tone	<input checked="" type="radio"/> No <input type="radio"/> Yes
Suppress Simulated CCM after ISDN Progress	<input checked="" type="radio"/> No <input type="radio"/> Yes

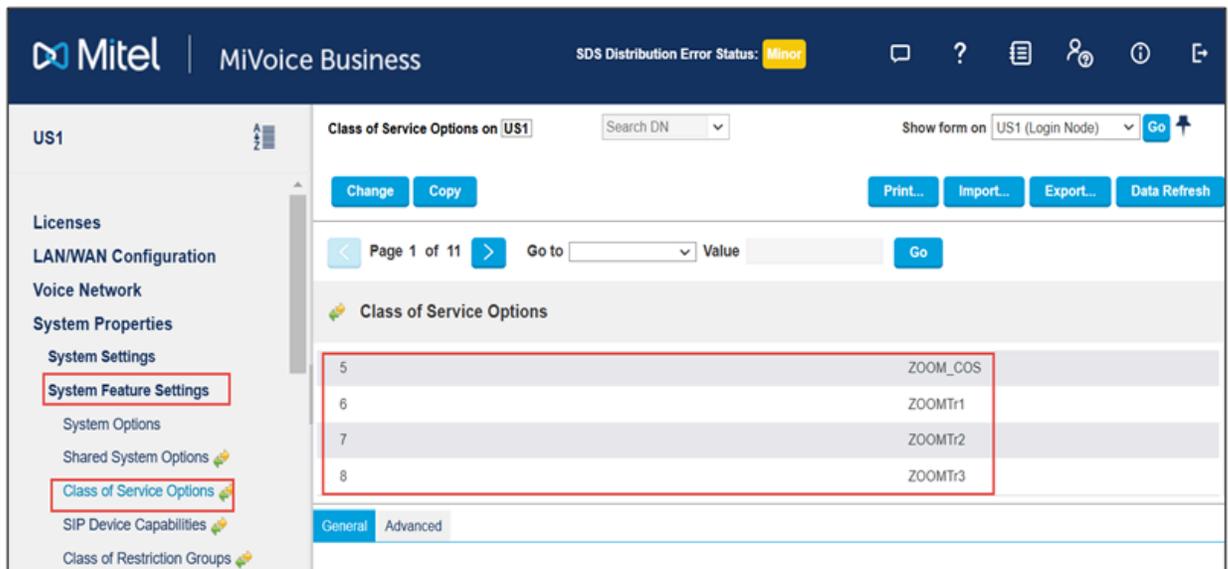
7. Click **Save**.

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**.

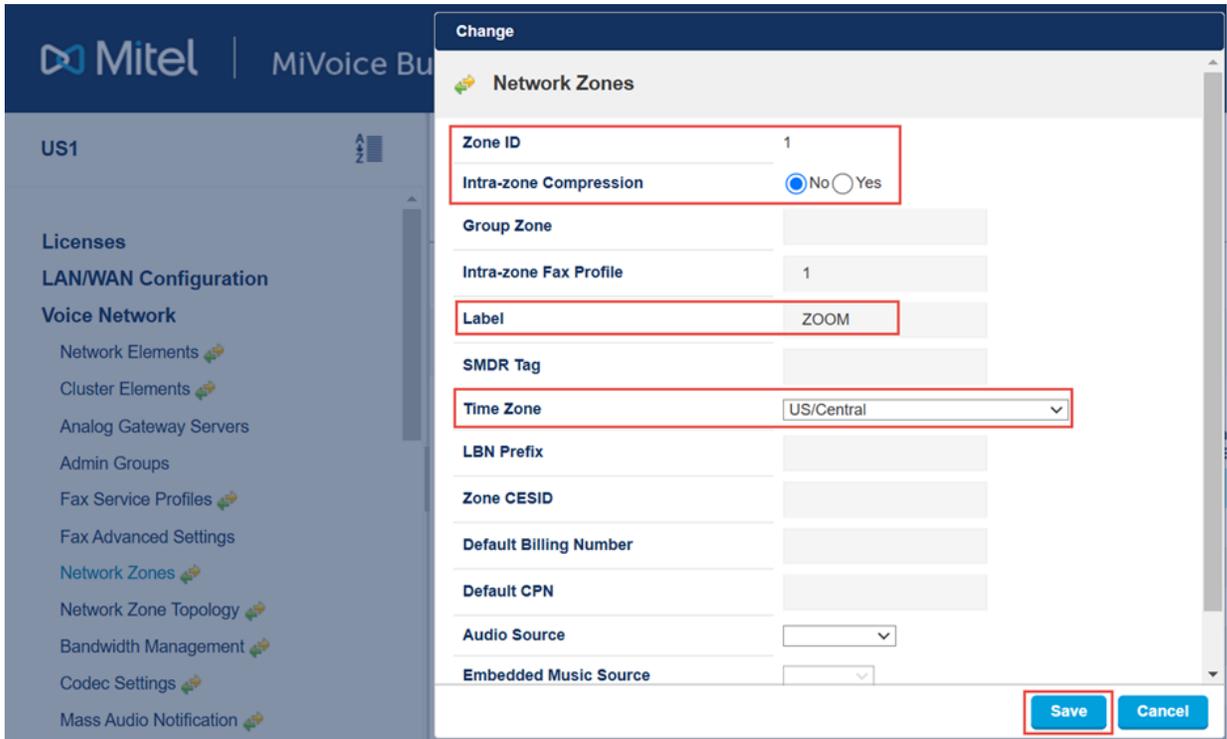
The **Class of Services** added for the **Zoom Trunks** and **PSTN Trunks** are listed in the **Class of Service Options** form as shown in the below figure.



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**.

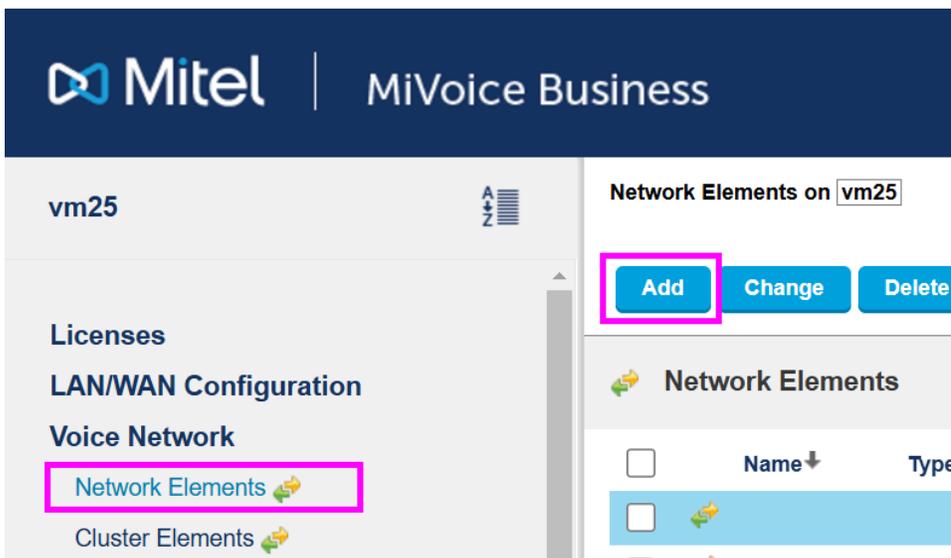


5.2.4 Adding Network Elements

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**.

The screenshot shows a 'Change' dialog box for 'Network Elements'. The fields are as follows:

Name	MBG_ZOOM
Type	Outbound Proxy
FQDN or IP Address	10.64.6.55
Local	False
Version	
Zone	1
Outbound Proxy Specific	
Outbound Proxy Transport Type	TLS
Outbound Proxy Port	5061

Buttons: Save, Cancel

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**.

Change

Network Elements

Name	<input style="width: 90%;" type="text" value="MBG_PSTN"/>
Type	<input style="border-bottom: 1px solid #ccc;" type="text" value="Outbound Proxy"/>
FQDN or IP Address	<input style="width: 90%;" type="text" value="10.64.6.55"/>
Local	False
Version	
Zone	<input style="width: 90%;" type="text" value="1"/>
Outbound Proxy Specific	
Outbound Proxy Transport Type	<input style="width: 90%;" type="text" value="TCP"/>
Outbound Proxy Port	<input style="width: 90%;" type="text" value="5060"/>

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**.

Change

Network Elements

Name	ZOOMTr1
Type	Other
FQDN or IP Address	162.12.233.59
Local	False
Version	
Zone	1
SIP Peer	<input checked="" type="checkbox"/>
SIP Peer Specific	
SIP Peer Transport	TLS
SIP Peer Port	5061
External SIP Proxy FQDN or IP Address	
External SIP Proxy Transport	default
External SIP Proxy Port	0
SIP Registrar FQDN or IP Address	
SIP Registrar Transport	default
SIP Registrar Port	0
SIP Peer Status	Auto-Detect/Normal

Save **Cancel**

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:

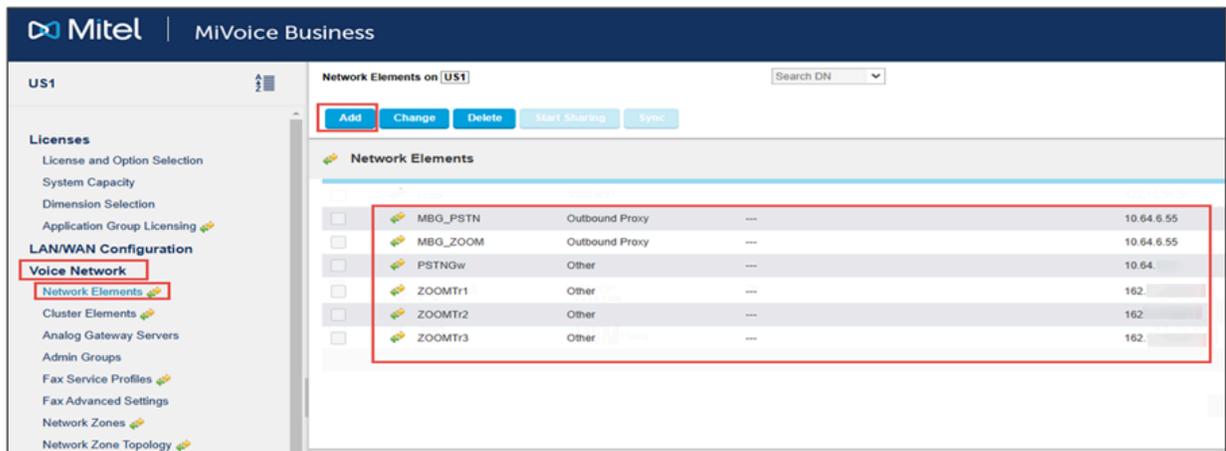
- a. **Name:** Enter the name. (Example: **PSTNGw**).
- b. **Type:** From the drop-down menu, Select **Others**.
- c. **FQDN or IP Address:** Enter the PSTN Gateway IP address.
- d. **Zone:** Enter the Zone number. (Example: 1 is used here).
- e. **SIP Peer:** Select the **SIP Peer** check box.
- f.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

The screenshot shows a 'Change' dialog box for 'Network Elements'. The form contains the following fields and values:

Name	PSTNGw
Type	Other
FQDN or IP Address	10.64.1.72
Local	False
Version	
Zone	1
SIP Peer	<input checked="" type="checkbox"/>
SIP Peer Specific	
SIP Peer Transport	TCP
SIP Peer Port	5060

At the bottom right of the dialog are 'Save' and 'Cancel' buttons.

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.

The screenshot shows the Mitel MiVoice Business configuration interface. The left sidebar contains a navigation menu with the following items: Licenses, LAN/WAN Configuration, Voice Network, System Properties, Hardware, Trunks, and IP/XNET. The 'Trunks' item is selected, and the 'Trunk Attributes' sub-item is highlighted with a red box. The main content area displays 'Trunk Attributes on vm25' with a 'Change' button highlighted in red. Below this is a table with the following data:

Trunk Service Number	Call Recoq Service
1	Off
2	Off
3	Off

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).

Change

✎

Trunk Attributes

Trunk Service Number	6
Release Link Trunk	No ▼
Call Recognition Service	Off ▼
Direct Inward Dialing Service	<input type="radio"/> Off <input checked="" type="radio"/> On
Caller Based Routing Service	<input checked="" type="radio"/> Off <input type="radio"/> On
Class of Service	6
Class of Restriction	1
Baud Rate	300 ▼
Intercept Number	1
Non-dial In Trunks Answer Point - Day	
Non-dial In Trunks Answer Point - Night 1	
Non-dial In Trunks Answer Point - Night 2	
Dial In Trunks Incoming Digit Modification - Absorb	0
Dial In Trunks Incoming Digit Modification - Insert	
Dial In Trunks Answer Point	
Dial In Trunks Insert Forwarding Information	<input checked="" type="radio"/> No <input type="radio"/> Yes
Trunk Label	ZOOMTr1

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.

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	Profile Information
SIP Peer Profile Label		SIPTRUNK					
Network Element		PSTNGw					
Local Account Information							
Registration User Name							
Address Type		<input type="radio"/> FQDN: mivbplano.tekvizionlabs.com <input checked="" type="radio"/> IP Address: 10.35.32.2					
Administration Options							
Interconnect Restriction		1					
Maximum Simultaneous Calls		20					
Minimum Reserved Call Licenses		3					
Outbound Proxy Server		MBG_PSTN					
SMDR Tag		0					
Trunk Service		5					
Zone		1					
Authentication Options							
User Name							
Password							
Confirm Password							

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.

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	Profile Information
Default CPN		<input type="text" value=""/>					
Default CPN Name		<input type="text" value=""/>					
CPN Restriction		<input checked="" type="radio"/> No <input type="radio"/> Yes					
Override From Header with Default CPN		<input checked="" type="radio"/> No <input type="radio"/> Yes					
Public Calling Party Number Passthrough		<input checked="" type="radio"/> No <input type="radio"/> Yes					
Strip PNI		<input checked="" type="radio"/> No <input type="radio"/> Yes					
Use Diverting Party Number as Calling Party Number		<input checked="" type="radio"/> No <input type="radio"/> Yes					
Use Original Calling Party Number If Available		<input checked="" type="radio"/> No <input type="radio"/> Yes					

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**.

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	Profile Information
Allow Peer To Use Multiple Active M-Lines		<input checked="" type="radio"/> No <input type="radio"/> Yes					
Allow Using UPDATE For Early Media Renegotiation		<input checked="" type="radio"/> No <input type="radio"/> Yes					
Avoid Signaling Hold to the Peer		<input type="radio"/> No <input checked="" type="radio"/> Yes					
AVP Only Peer		<input checked="" type="radio"/> No <input type="radio"/> Yes					
Enable Mitel Proprietary SDP		<input checked="" type="radio"/> No <input type="radio"/> Yes					
Force sending SDP in initial Invite message		<input type="radio"/> No <input checked="" type="radio"/> Yes					
Force sending SDP in initial Invite - Early Answer		<input checked="" type="radio"/> No <input type="radio"/> Yes					
Ignore SDP Answers in Provisional Responses		<input checked="" type="radio"/> No <input type="radio"/> Yes					
IP Media Default		ipv4					
Limit to one Offer/Answer per INVITE		<input type="radio"/> No <input checked="" type="radio"/> Yes					
NAT Keepalive		<input type="radio"/> No <input checked="" type="radio"/> Yes					
Prevent Codec Selection on Answer		<input checked="" type="radio"/> No <input type="radio"/> Yes					
Prevent the Use of IP Address 0.0.0.0 in SDP Messages		<input type="radio"/> No <input checked="" type="radio"/> Yes					
Reject Call without telephone-event payload		<input checked="" type="radio"/> No <input type="radio"/> Yes					
Renegotiate SDP To Enforce Symmetric Codec		<input checked="" type="radio"/> No <input type="radio"/> Yes					
Repeat SDP Answer If Duplicate Offer Is Received		<input checked="" type="radio"/> No <input type="radio"/> Yes					
Restrict Audio Codec		No Restriction					
RTP Packetization Rate Override		<input checked="" type="radio"/> No <input type="radio"/> Yes					
RTP Packetization Rate		20ms					
Special handling of Offers in 2XX responses (INVITE)		<input checked="" type="radio"/> No <input type="radio"/> Yes					
Suppress Use of SDP Inactive Media Streams		<input checked="" type="radio"/> No <input type="radio"/> 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**.

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	Profile Information
Trunk Group Label <input type="text" value=""/>							
Allow Display Update <input type="radio"/> No <input checked="" type="radio"/> Yes							
Build Contact Using Request URI Address <input type="radio"/> No <input checked="" type="radio"/> Yes							
De-register Using Contact Address not * <input type="radio"/> No <input checked="" type="radio"/> Yes							
Disable Reliable Provisional Responses <input checked="" type="radio"/> No <input type="radio"/> Yes							
Disable Use of User-Agent and Server Headers <input checked="" type="radio"/> No <input type="radio"/> Yes							
Discard Received P-Asserted-Identity Headers <input checked="" type="radio"/> No <input type="radio"/> Yes							
Domain for Trunk Context <input type="text" value=""/>							
Emergency Call Headers <input type="text" value="CESID in From, [and PAI]"/>							
E.164: Enable sending '+' <input type="radio"/> No <input checked="" type="radio"/> Yes							
E.164: Add '+' if digit length > N digits <input type="text" value="9"/>							
E.164: Do not add '+' to Emergency Called Party <input checked="" type="radio"/> No <input type="radio"/> Yes							
E.164: Do not add '+' to Called Party <input checked="" type="radio"/> No <input type="radio"/> Yes							
Force Max-Forward: 70 on Outgoing Calls <input checked="" type="radio"/> No <input type="radio"/> Yes							
If TLS use 'sips:' Scheme <input checked="" type="radio"/> No <input type="radio"/> Yes							
Ignore Incoming Loose Routing Indication <input checked="" type="radio"/> No <input type="radio"/> Yes							
Include Diversion Header for EHCU <input checked="" type="radio"/> No <input type="radio"/> Yes							
Mode for Out-of-Band DTMF <input checked="" type="radio"/> RFC 4733 DTMF <input type="radio"/> SIP INFO dtmf-relay							
Multilingual Name Display <input type="radio"/> No <input checked="" type="radio"/> Yes							
Only use SDP to decide 180 or 183 <input type="radio"/> No <input checked="" type="radio"/> Yes							
Prefer From Header for Caller ID <input type="radio"/> No <input checked="" type="radio"/> Yes							
Q.850 Reason Headers <input checked="" type="radio"/> No <input type="radio"/> Yes							
Require Reliable Provisional Responses on Outgoing Calls <input checked="" type="radio"/> No <input type="radio"/> Yes							
Suppress Incoming Name <input type="text" value="No"/>							
Suppress Redirection Headers <input type="text" value="No"/>							

5. In the **Timers** tab,

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

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	Profile Information
Keep-Alive (OPTIONS) Period						<input type="text" value="120"/>	
Registration Period						<input type="text" value="3600"/>	
Registration Period Refresh (%)						<input type="text" value="50"/>	
Registration Maximum Timeout						<input type="text" value="90"/>	
Session Timer						<input type="text" value="3600"/>	
Session Timer: Local as Refresher						<input checked="" type="radio"/> No <input type="radio"/> Yes	
Subscription Period						<input type="text" value="3600"/>	
Subscription Period Minimum						<input type="text" value="300"/>	
Subscription Period Refresh (%)						<input type="text" value="80"/>	
Invite Ringing Response Timer						<input type="text" value="0"/>	

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.

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	Profile Information
SIP Peer Profile Label		ZOOMTr1					
Network Element		ZOOMTr1					
Local Account Information							
Registration User Name							
Address Type		<input type="radio"/> FQDN: mivbplano.tekvizionlabs.com <input checked="" type="radio"/> IP Address: 10.35.32.2					
Administration Options							
Interconnect Restriction		1					
Maximum Simultaneous Calls		5					
Minimum Reserved Call Licenses		0					
Outbound Proxy Server		MBG_ZOOM					
SMDR Tag		0					
Trunk Service		6					
Zone		1					
Authentication Options							
User Name							
Password							
Confirm Password							
Authentication Option for Incoming Calls		No Authentication					
Subscription User Name							
Subscription Password							
Subscription Confirm Password							
Gateway Options							
Digital Trunk Licenses		0					
Maximum Digital/Analog Channels		0					

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**.

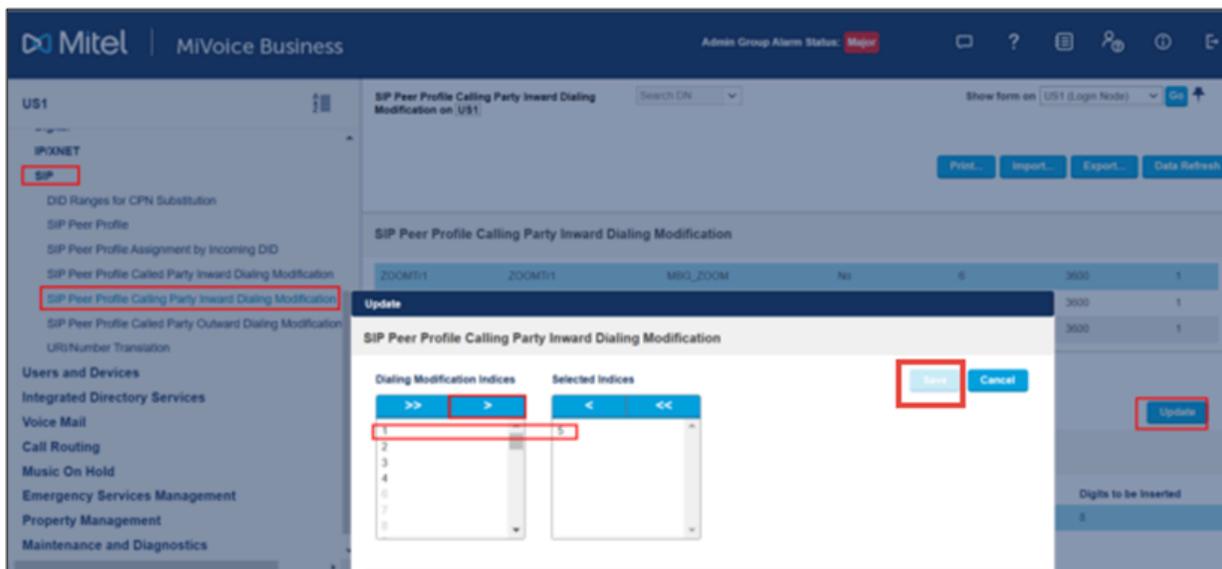
5.2.6.2 Adding SIP Peer Profile Called Party Inward Dialing Modification

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**

Change

ARS Routes

Route Number	7
Routing Medium	<input type="text" value="SIP Trunk"/>
Trunk Group Number	<input type="text" value=""/>
SIP Peer Profile	<input type="text" value="SIPTRUNK"/>
PBX Number / Cluster Element ID	<input type="text" value=""/>
COR Group Number	1
Digit Modification Number	2
Digits Before Outpulsing	<input type="text" value=""/>
Route Type	<input type="text" value=""/>
Compression	<input type="text" value="Off"/>

To Configure Route for Zoom Destination:

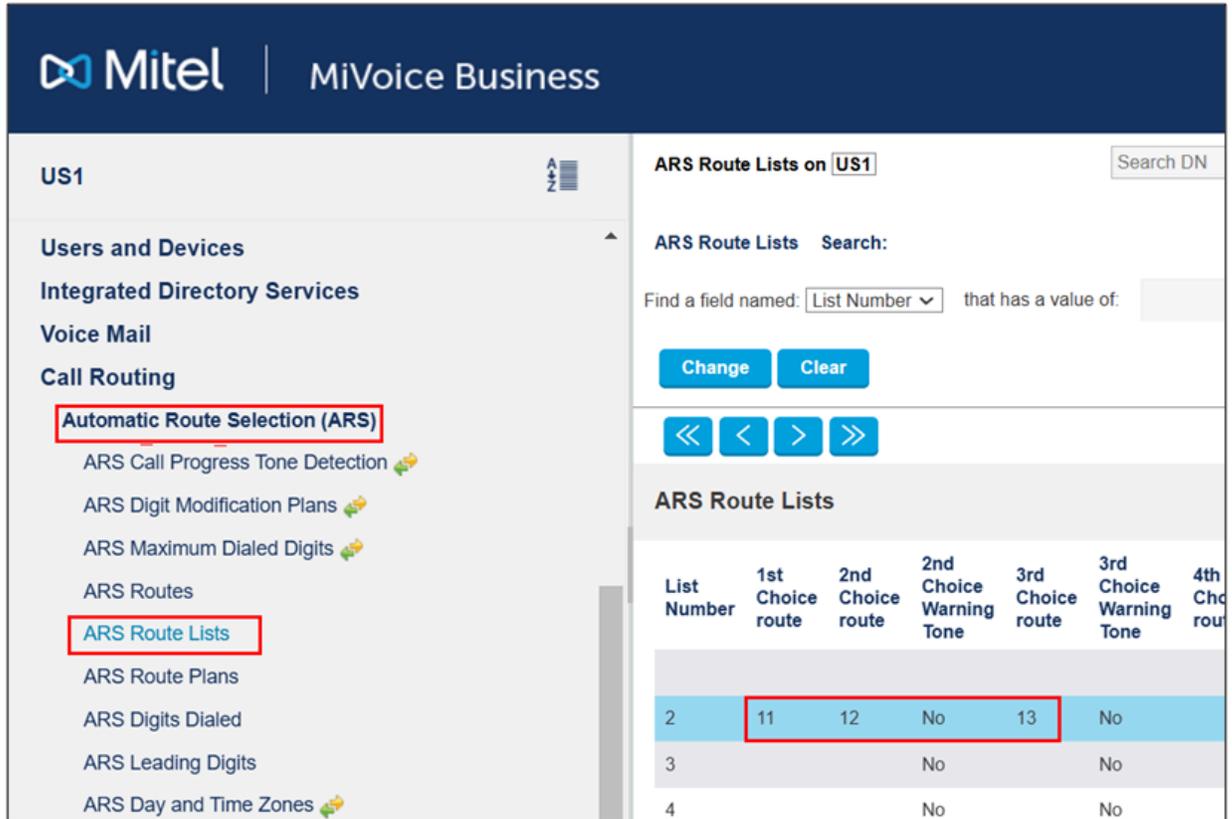
1. Select the desired **Route Number**: 11 is used as an example for **ZOOMTr1**. (Route Number 12 and 13 is used for ZOOMTr2 and ZOOMTr3 respectively)
2. Click **Change**.
3. **Routing Medium**: Select **SIP Trunk** from the drop-down menu.
4. **SIP Peer Profile**: Select **ZOOMTr1** from the drop-down 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

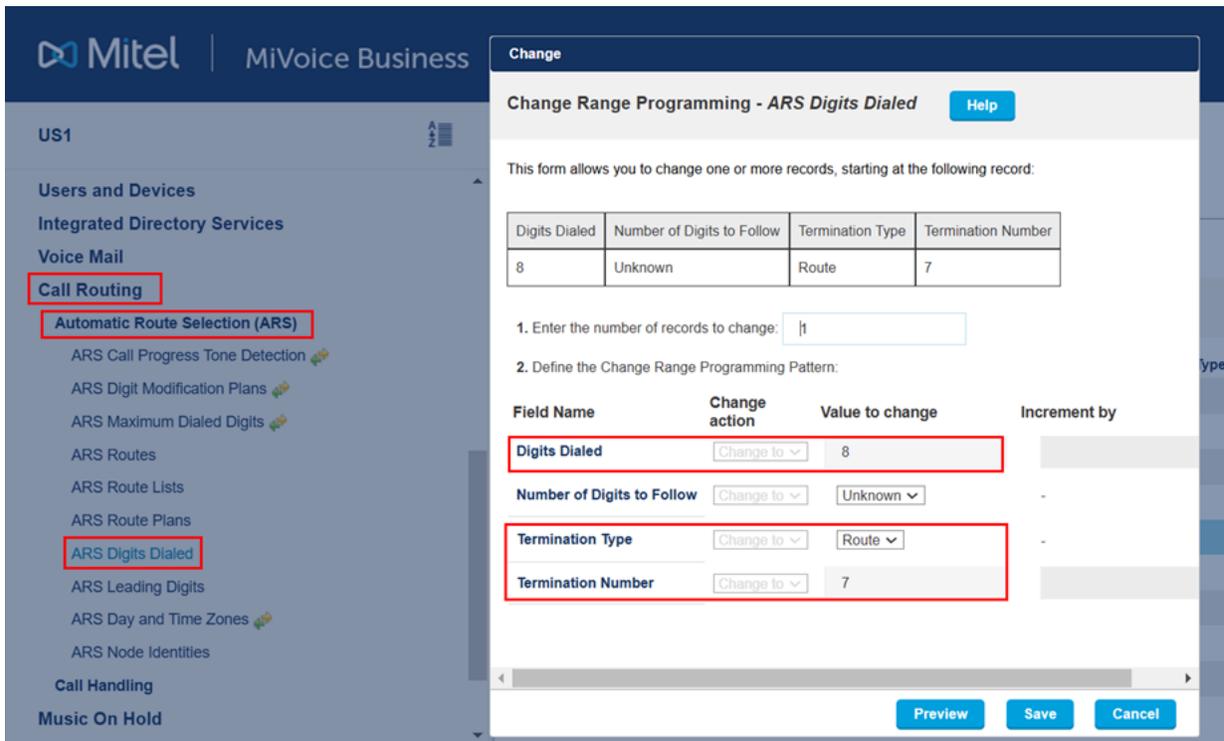
1. Navigate to **Call Routing > Automatic Route Selection > ARS Digits Dialed** form.

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**

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

Change

Change Range Programming - ARS Digits Dialed

Help

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:

2. Define the Change Range Programming Pattern:

Field Name	Change action	Value to change	Increment by
Digits Dialed	Change to	972598	
Number of Digits to Follow	Change to	0	-
Termination Type	Change to	List	-
Termination Number	Change to	2	

Preview

Save

Cancel

