



A MITEL
PRODUCT
GUIDE

Zoom Phone with MiVoice Business and MiVoice Border Gateway

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

Document Version 1.3

November 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.
1.2	27-May-2025	Updated Prerequisites and Licensing section.
1.3	24-November-2025	Revised the entire document

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 4.

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.

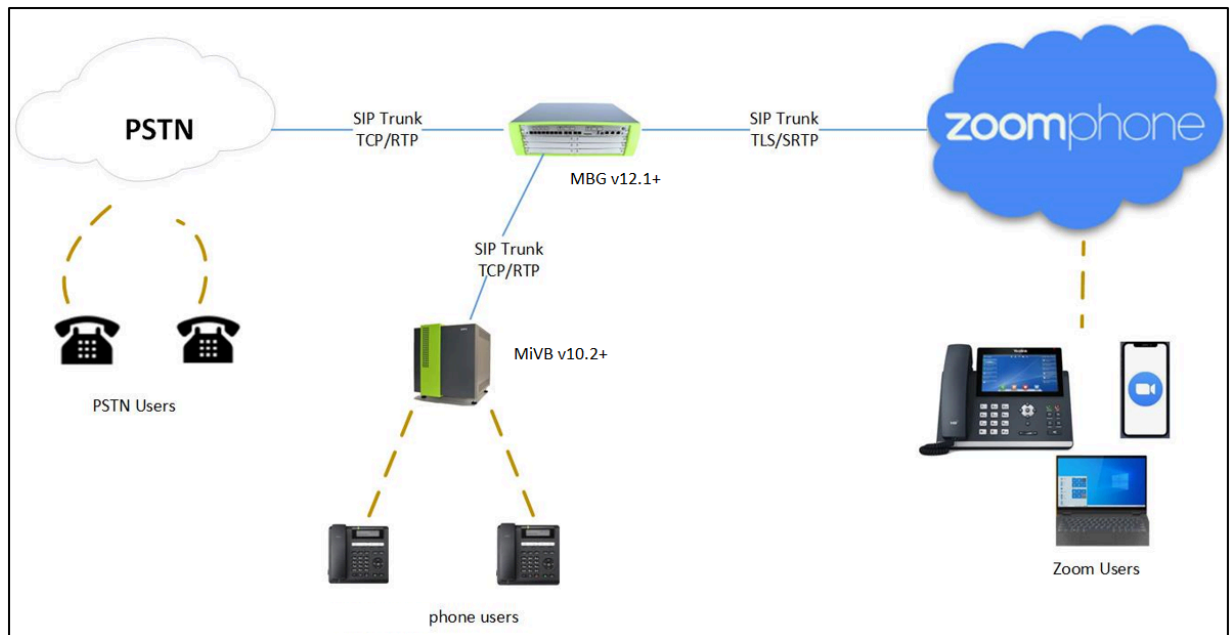


Figure 1: Network Topology Block Diagram

2.1 Prerequisites

Supported product versions

Product	SW Version (minimum)
Zoom Workplace app	6.4.1 or greater
MiVoice Border Gateway	12.2.0.72
MiVoice Business with active SWA (Mitel Software Assurance)	10.3.0.16

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 Phone Users](#)
- [Adding Your SBC](#)
- [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.

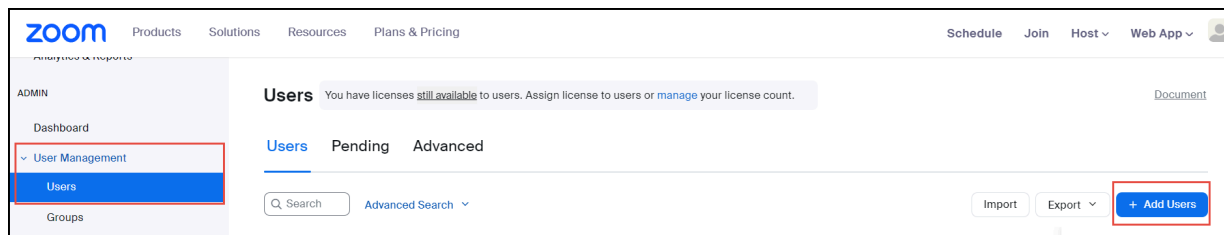
To set up users for the Zoom and MiVoice Business integration, you must first add users to your Zoom account and assign licenses to them.

3.1 Adding 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
Zoom Meetings (0 available)

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

Add Cancel

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](#).

3.1.1 Assigning a Calling Plan to a phone user

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

Prerequisite

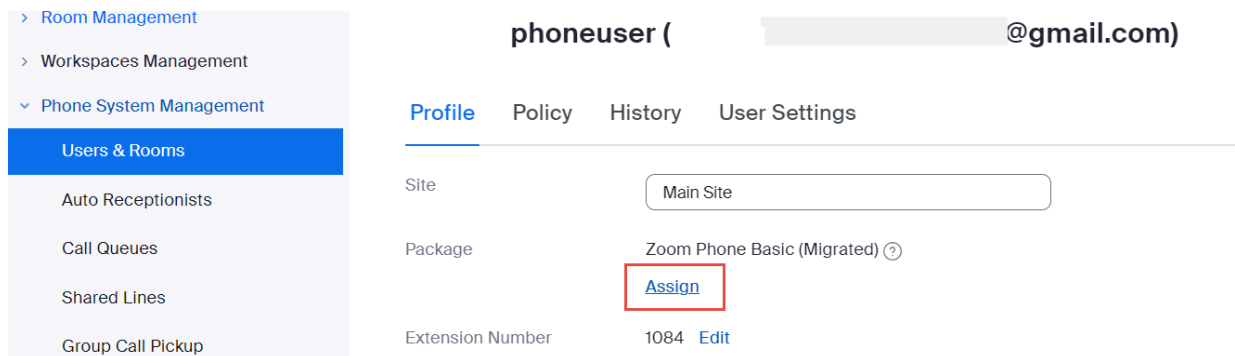
1. You are an administrator with the privilege to edit account settings.
2. You have assigned licenses to the phone users. For more information, refer to [How to assign licenses](#).

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.

The screenshot shows the Zoom Web Portal configuration page for a user named 'phoneuser'. The left sidebar contains the following menu items: Analytics & Reports, ADMIN, Dashboard, > User Management, > Device Management, > Room Management, > Workspaces Management, > Phone System Management (expanded), Users & Rooms (selected), Auto Receptionists, and Call Queues. The main content area has tabs for Profile, Policy, History, and User Settings. The Profile tab is active, showing fields for Site (Main Site), Package (Select Package), Extension Number, and Emergency Address (Default: 3701 W PLANO PKWY, STE 300 STE 300, PLANO, Texas 75075, United States). The Package dropdown menu is open, showing two options: 'US/CA Unlimited Calling Plan (9 Available)' and 'Zoom Phone Power Pack (19 Available)'. The 'US/CA Unlimited Calling Plan' option is highlighted with a red box.

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.

3.2 Adding Your SBC

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

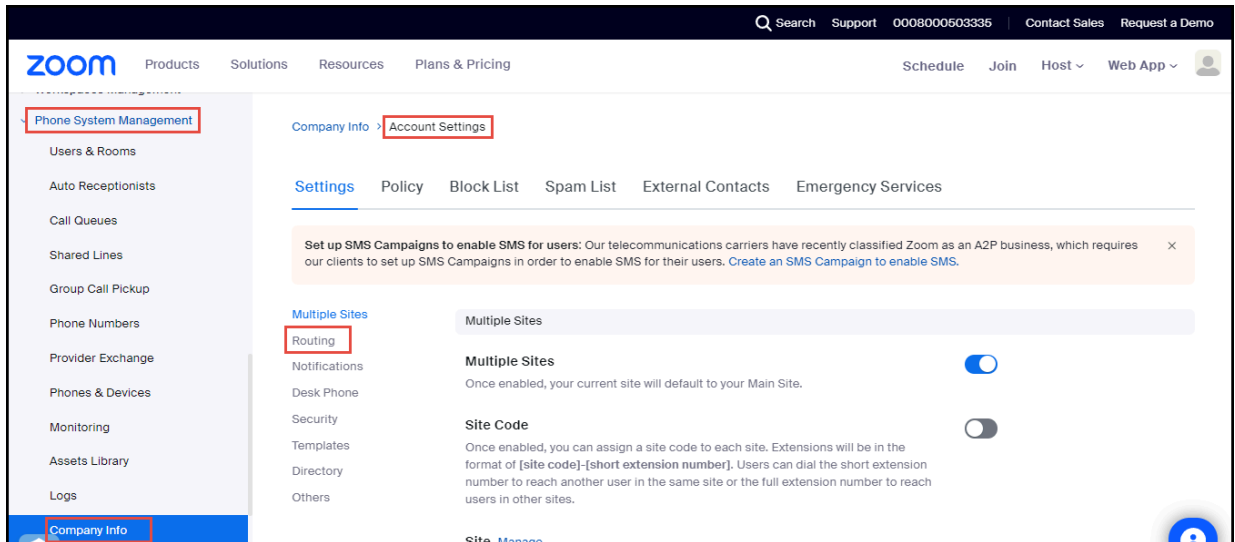
Prerequisites

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.

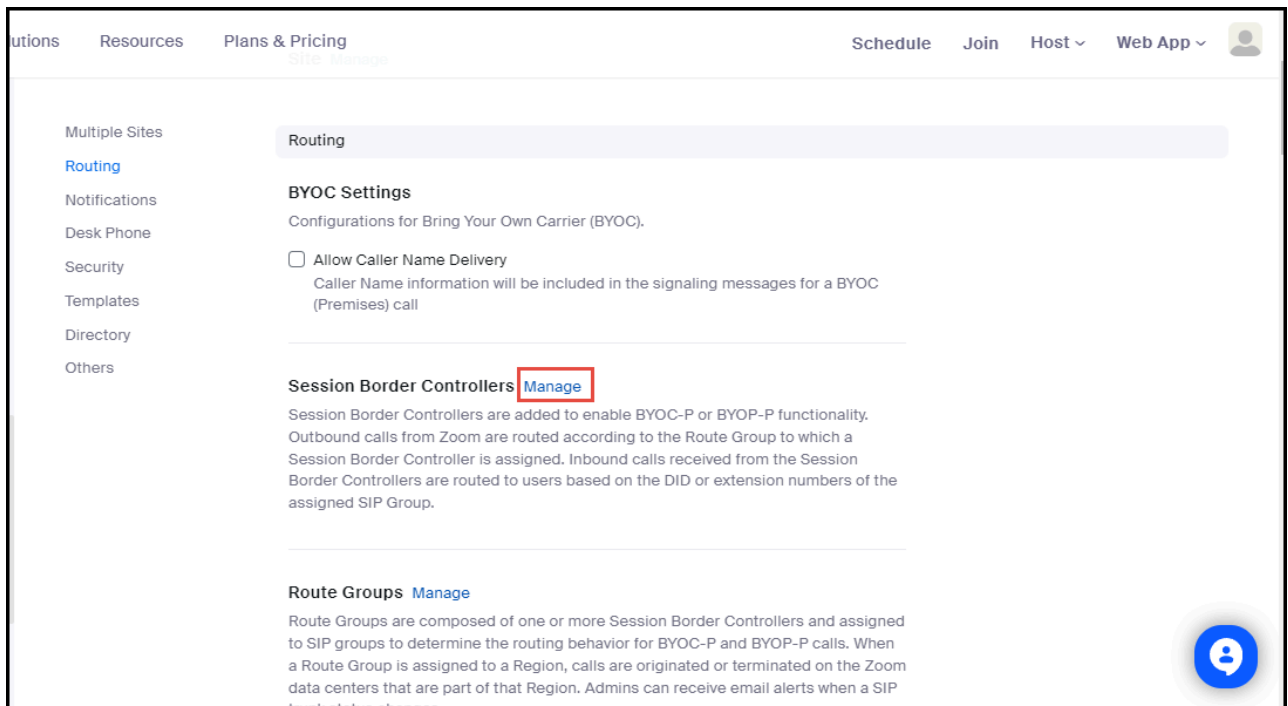
Adding your SBC

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

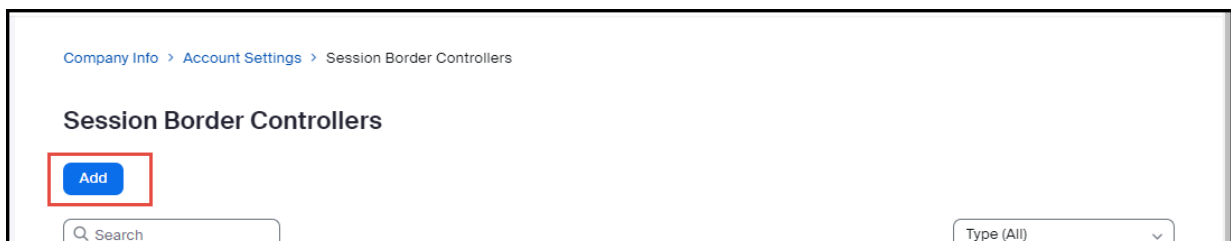
2. Navigate to **Phone System Management > Company Info > Account Settings > Routing**.



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 Set-side Streaming Address 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 the Zoom Web Portal configuration form. Red boxes highlight the following sections:

- Display Name:** A text input field containing "MITEL_MBG_ZOOM".
- Description (Optional):** An empty text input field.
- 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" (with a question mark), "Public IP Address" (containing "192"), and "Port Number" (containing "5061").
- In-Service:** A toggle switch that is turned on (blue).
- Settings:** A section containing several 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 (with a question mark)
 - ☐ Allow REFER support to transfer calls (with a "BETA" label)
- Address(Optional):** A section with three rows, each containing a label (Address, Email, Phone Number) and a "Set" link.
- Save/Close:** Two buttons at the bottom left of the form.

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.

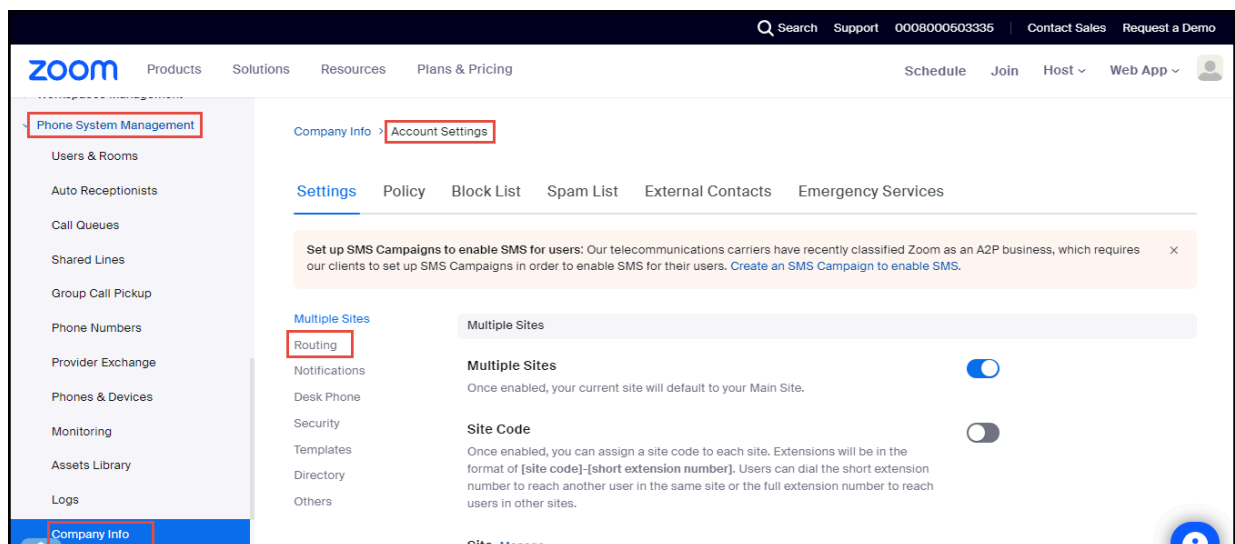
3.2.1 Configuring the Route Group

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**.

Multiple Sites
Routing
Notifications
Desk Phone
Security
Templates
Directory
Others

Routing

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:

- 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)**

i 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

Type

BYOC-P

Region

US01 - US (SJ/DV/NY)

Distribution

Sequential

Session Border Controllers

1: MITEL_MBG_ZOOM (192.XXXX.X.)

Add

Backup Route Group (Optional)

Select

Save

Cancel

5. Click **Save**.

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

zoom

ProductsSolutionsResourcesPlans & Pricing

> user management

> Device Management

> Room Management

> Workspaces Management

> Phone System Management

Users & Rooms

Auto Receptionists

Call Queues

Shared Lines

Push to Talk

Group Call Pickup

Phone Numbers

Provider Exchange

Phones & Devices

Monitoring

Assets Library

Logs

Company Info

Overview

> Number Management

> Account Management

> Advanced

Zoom Learning Center

Company Info > Account Settings > Route Group

Route Group

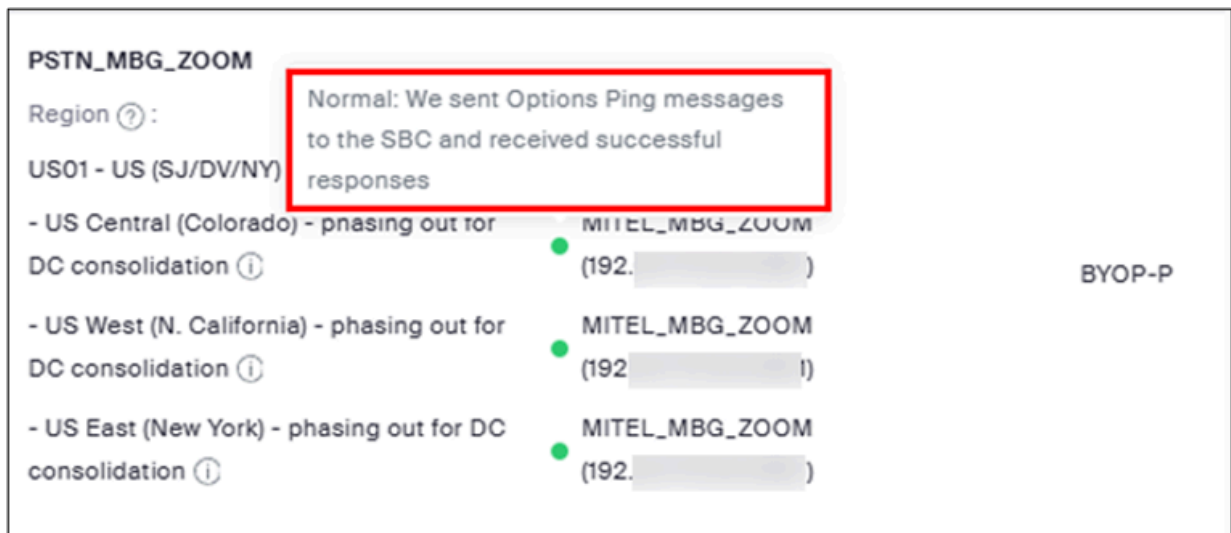
Last Updated Time: 06:17 PM, Feb 05, 2025

Add

Search by Name

Display Name	Session Border Controllers	Type
PSTN_MBG_ZOOM		
Region : US01 - US (SJ/DV/NY)	Sequential : MITEL_MBG_ZOOM (192.)	BYOP-P
- US Central (Colorado) - phasing out for DC consolidation	MITEL_MBG_ZOOM (192.)	
- 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 : MITEL_MBG_ZOOM (192. 1)	BYOC-P
- US Central (Colorado) - phasing out for DC consolidation	MITEL_MBG_ZOOM (192. 1)	
- 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:

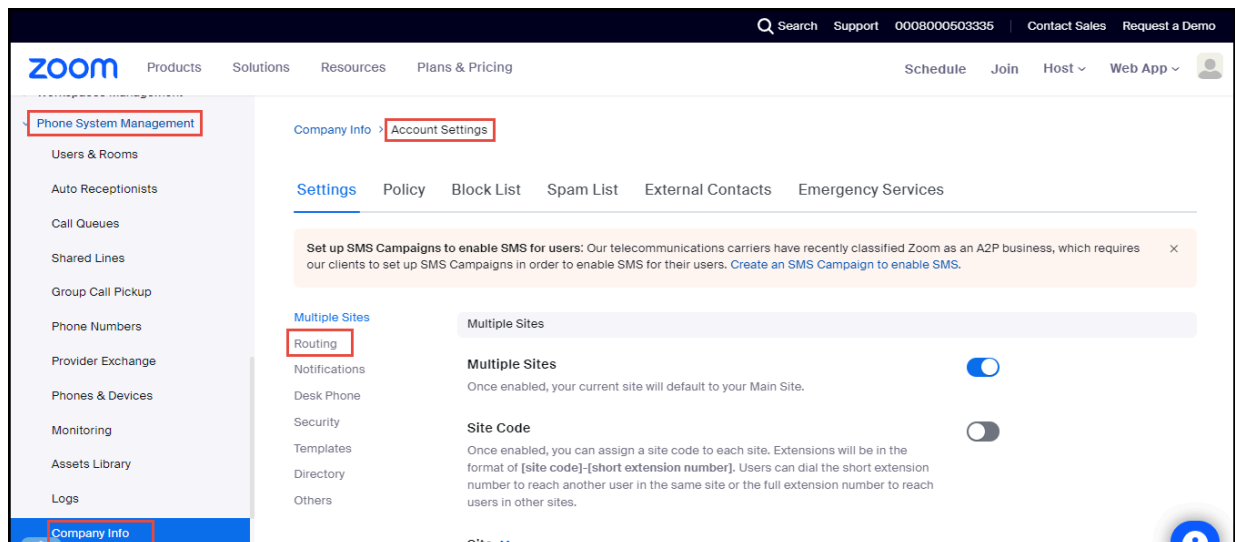


3.2.2 Configuring the SIP Group

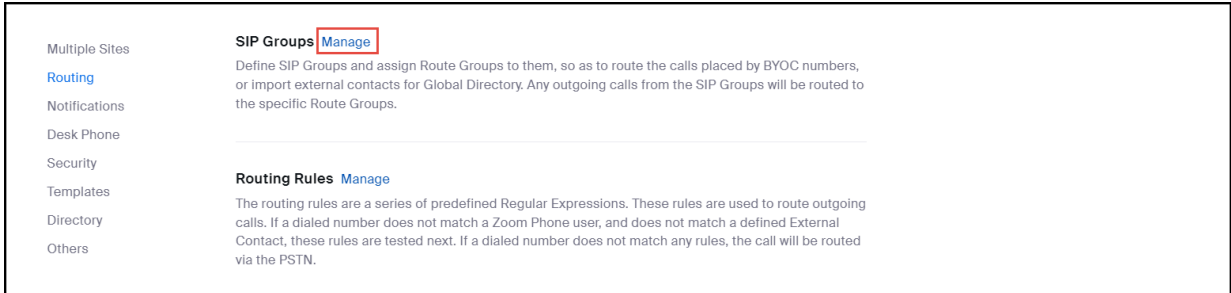
Follow the instructions below to configure SIP groups and assign Route Groups to them, in order to route calls placed by BYOC numbers. This step is mandatory for uploading the BYOC numbers.

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:

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

The screenshot shows the 'Edit SIP Group' form. The form has the following fields and controls:

- Display Name:** A text input field containing 'SIP_Group_Mitel_MBG_ZOOM'. The entire field is highlighted with a red box.
- Send SIP Group Name in SIP header:** A checkbox that is currently unchecked, followed by a help icon (?).
- Route Group:** A dropdown menu showing 'Group-Mitel_MBG_ZOOM (BYOC)'. The entire field is highlighted with a red box.
- Description (Optional):** A text input field containing the placeholder text 'Enter'.
- Buttons:** At the bottom right, there are two buttons: 'Save' and 'Cancel'. The 'Save' button is highlighted with a red box.

5. Click **Save**.

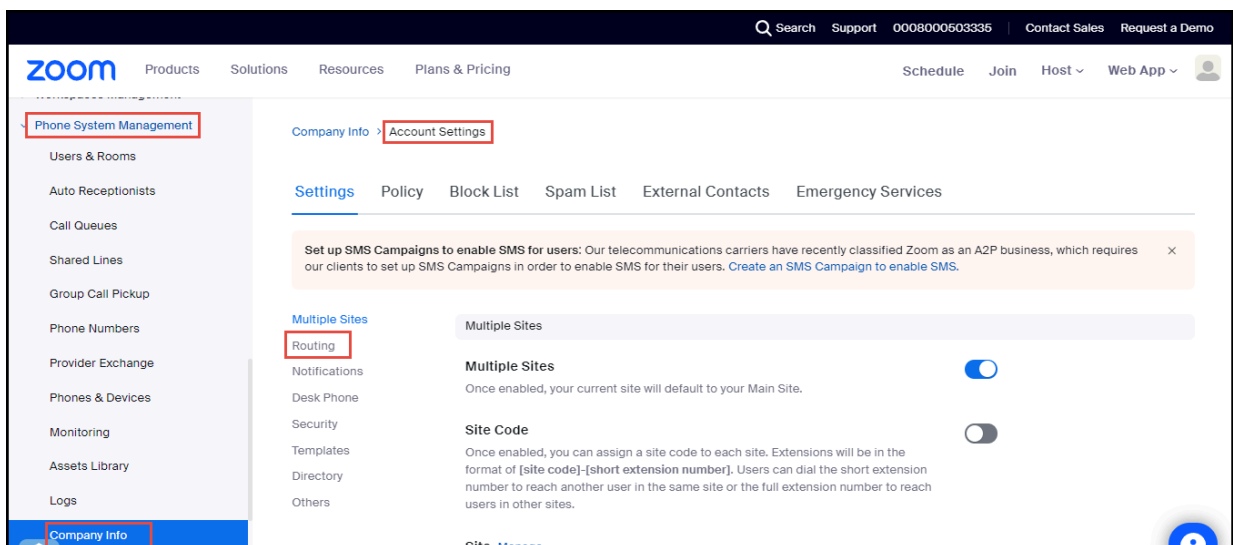
3.2.3 Configuring the Routing Rule

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:

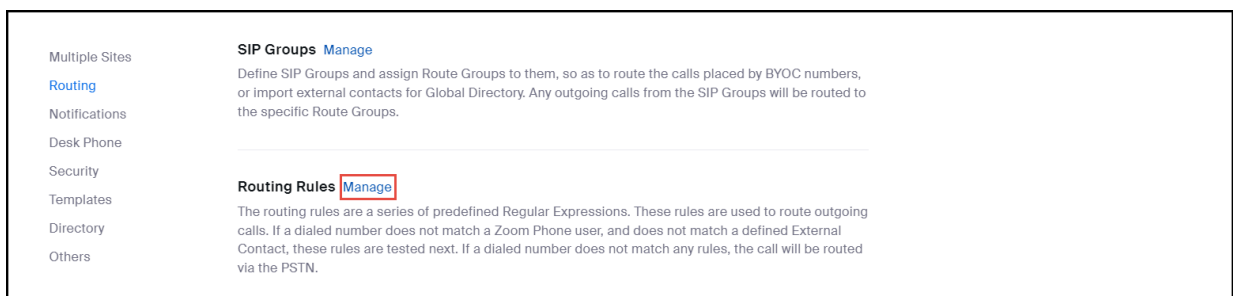
Note: Ensure that your Session Border Controller is properly configured and connected before setting up routing rules. Additionally, phone users must be provisioned and assigned to the correct phone numbers for routing rules to function correctly.

To add the Routing Rule:

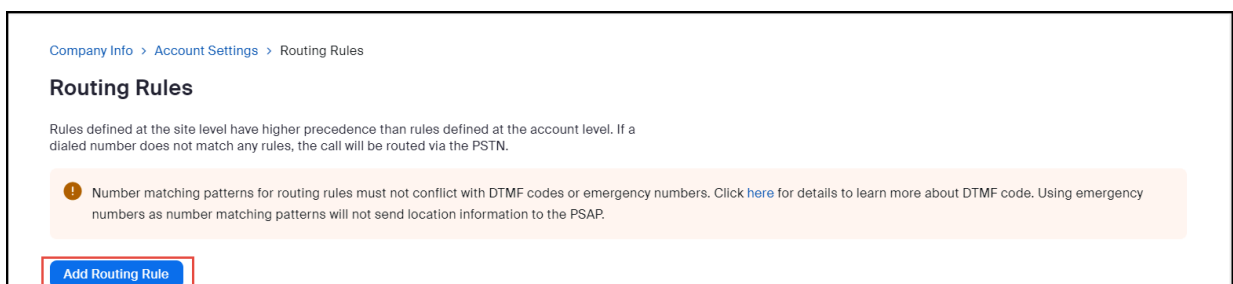
1. Navigate to **Phone System Management > Company Info > Account Settings > Routing**.



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.

Edit Routing Rule

Level: Account

Rule Name: Outgoing

Number Matching and Translation: Number Pattern: ^\\\\d{11})\$

Translation (Optional): Replacement Pattern must be in E.164 format

Test

Warning: Number matching patterns for routing rules must not conflict with DTMF codes or emergency numbers. Click [here](#) for details to learn more about DTMF code. Using emergency numbers as number matching patterns will not send location information to the PSAP.

Routing Path: SIP_Group_Mitel_MBG_ZOOM

Call Forwarding: ☐

Save Cancel

5. Click **Save**.

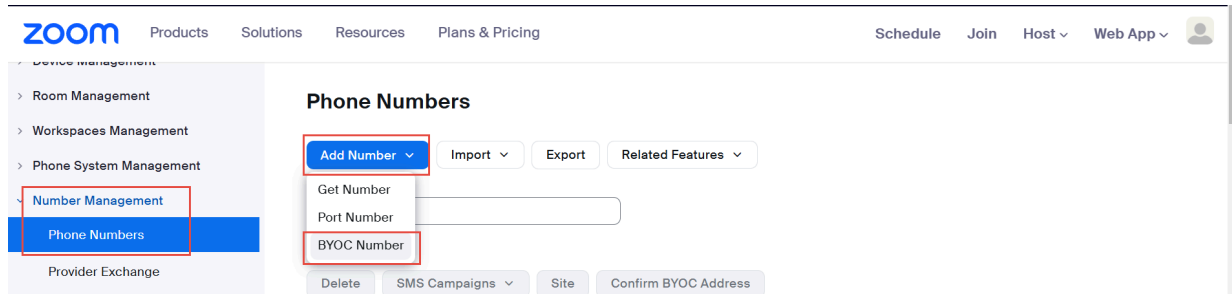
3.3 Adding BYOC Phone Numbers

You can add the BYOC phone numbers provided by your service provider 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**.

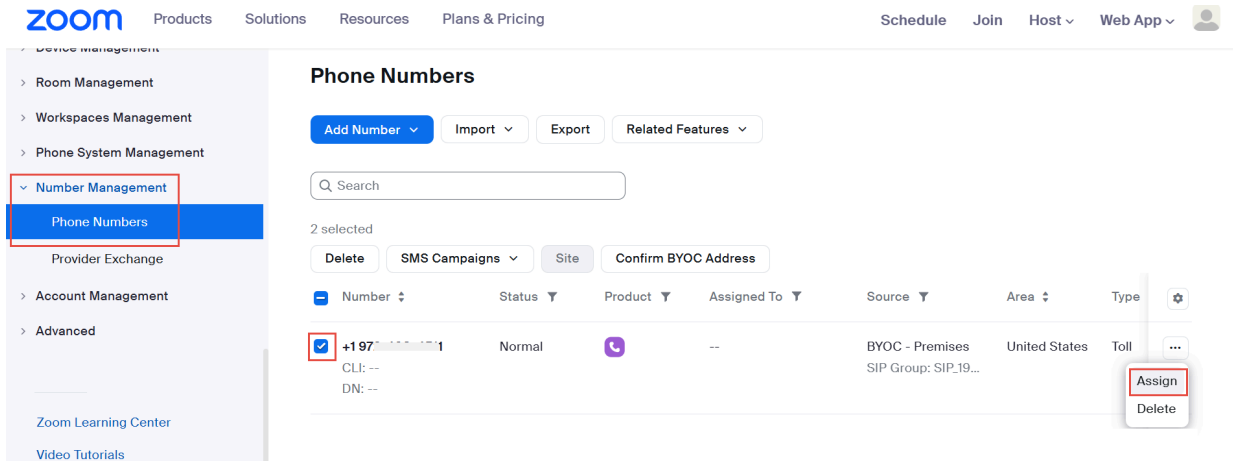
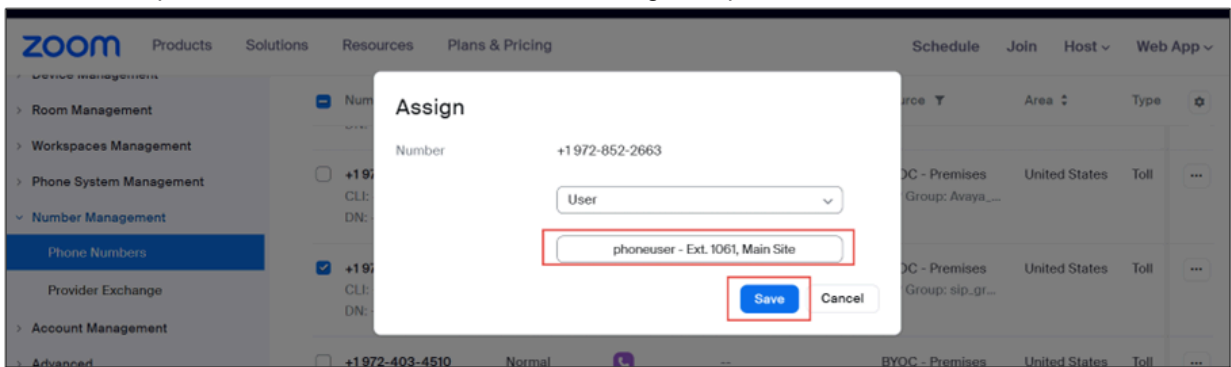
Add BYOC Number

Product	Phone
Site	Main Site
Country/Region	United States
Numbers	9720022073,9720022072,9720022071
SIP Group ?	Choose a routing path for calls to/from the numbers SIP_Group_Mitel_MBG_ZOOM
<input checked="" type="checkbox"/> I acknowledge that by checking the box, I attest that the phone numbers to be imported belong to me or my organization	
<input type="button" value="Submit"/> <input type="button" value="Cancel"/>	

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 extension to assign the phone number to and click **Save**.

The phone number will be assigned to the selected user.

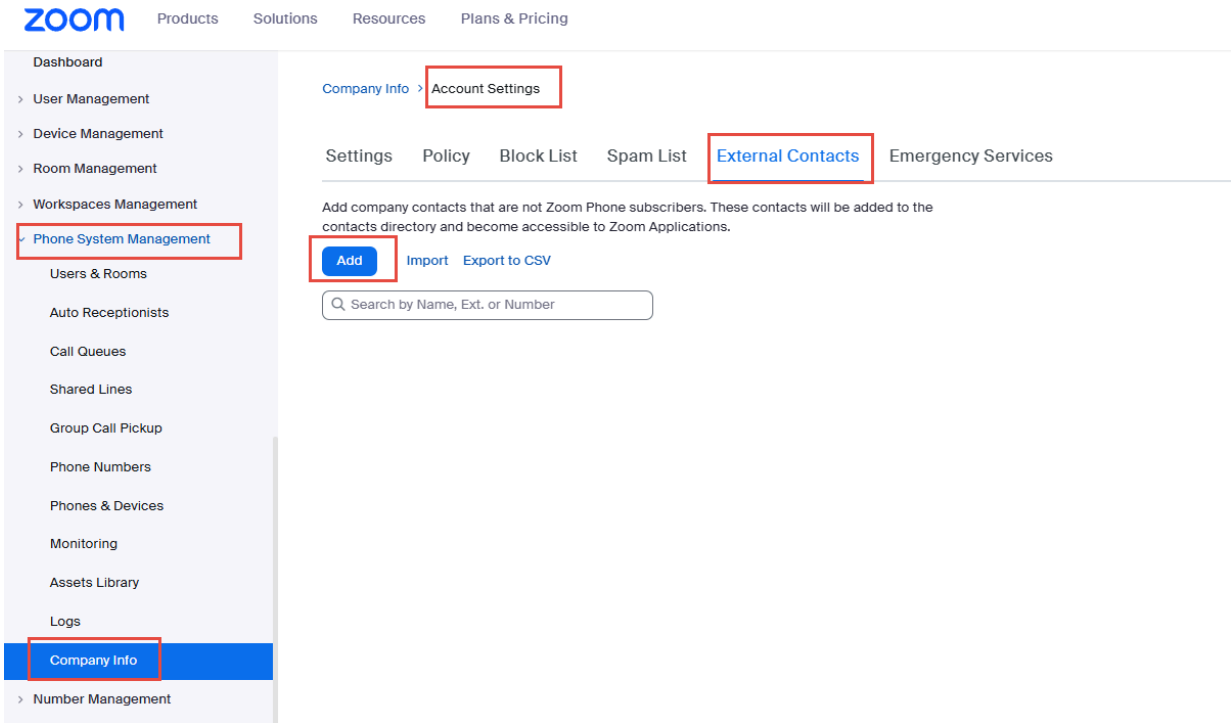
3.4 Adding BYOP numbers

Administrators can add MiVB 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 14](#) section.

The screenshot shows the 'Edit External Contact' form. The fields are as follows:

- ID** (with a help icon): 2
- Name**: Mivoice_user2 (highlighted with a red box)
- Email (Optional)**: (empty)
- Extension Number (Optional)** (with a help icon): 2000 (highlighted with a red box)
- 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 box)
- Auto Call Recorded** (with a help icon): (toggle switch is off)
- Buttons**: Save (highlighted with a red box) and Cancel

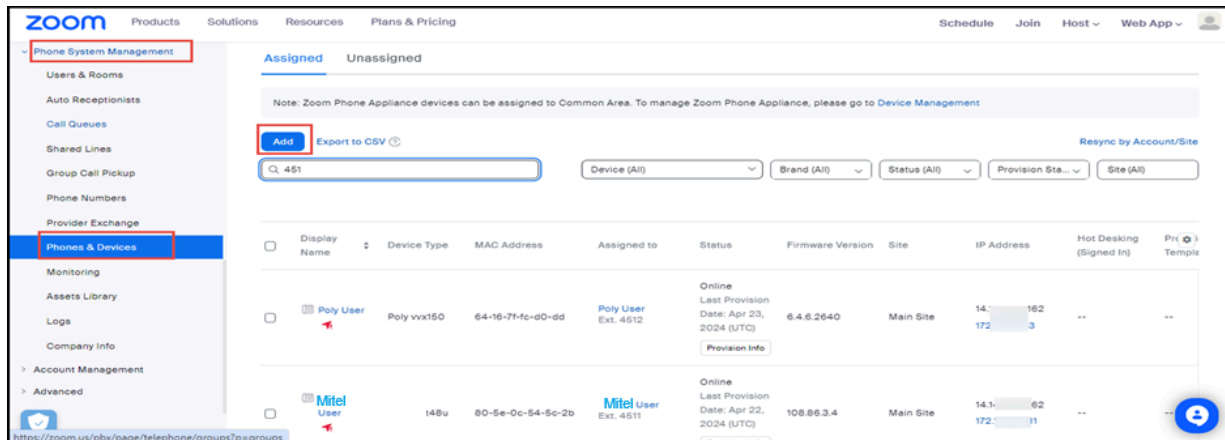
4. Click **Save**.

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, **Mitel phoneuser**.
 - b. **MAC Address:** Enter the **MAC address** of the phone.
 - c. **Device Type:** Select the device type. For example, **Mitel**.
 - 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

Mitel phoneuser

Description
(Optional)

MAC Address

80-5e-0c-54-5c-2b

Device Type

Mitel

t48u

This device type supports up to 1 assignee.

Assigned to

User

Mitel phoneuser - Ext. 1084, Main S

Add

Cancel

Provision
Template
(Optional)

Not Set

Save

Cancel

To enable Zoom Workplace integration with MiVoice Business, the following licenses are required for every user:

1. **Zoom Hybrid License subscription:** Required to support Zoom Phone with MiVoice Business and MiVoice Border Gateway.

Note: Zoom Supported License per user includes Zoom Workplace Business/Business+, Zoom Workplace Essential/sEnterprise/Enterprise+/Enterprise Premier, Legacy Meeting Licenses ENH/EAH.

2. **MiVoice Business Multi-Device User License:** Required to support both desktop and mobile Zoom Clients.
3. **MBG Teleworker License:** A separate license is required for each Zoom client - one for the desktop client and one for the mobile device.
4. **Embedded Voicemail Mailbox (Optional):** If voicemail is required, a MiVoice Business embedded mailbox license must be used.

If the above licenses are not readily available, they can be added using the part numbers specified below:

Description	Part No	Quantity per user
Mitel Zoom Hybrid License	See CPQ under Subscription Offers / Mitel Zoom Hybrid License Subscription	1
MiVoice Business Multi-Device User License	54005328	1
MBG Teleworker License	54004572	2
Embedded Voicemail Mailbox (Optional)	54000297	1

MBG/MiVB Configuration: BYOP/BYOC 6

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.

6.1 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 7, 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.

6.1.1 Network Configuration

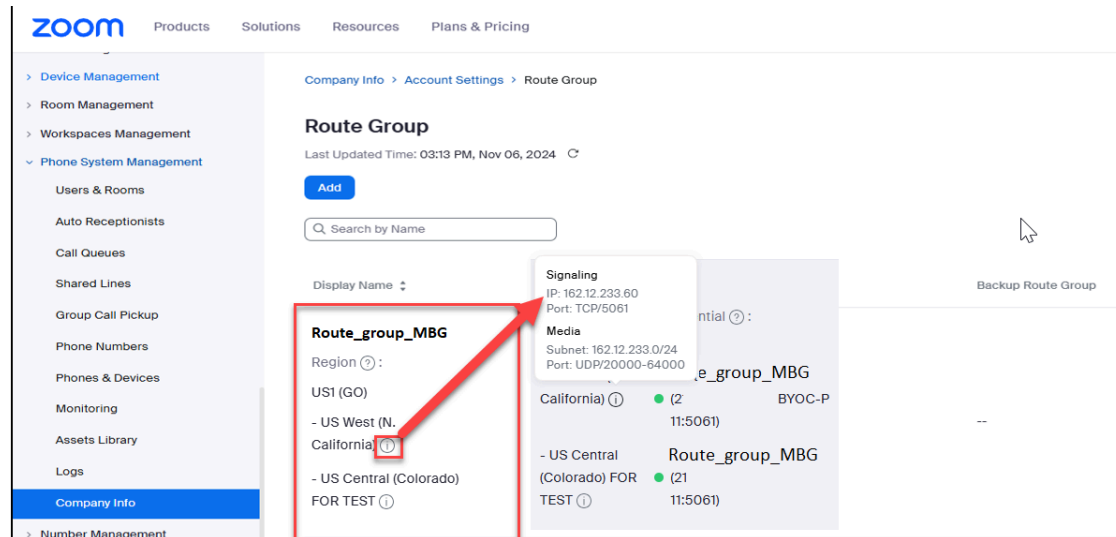
During the installation of the Mitel Border Gateway (MBG), you will be required to enter network configuration details for both the **WAN (Wide Area Network)** and **LAN (Local Area Network)**. These steps are standard for all MBG installations. For detailed instructions on setting up network configuration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

6.1.2 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.



6.1.3 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 Certificates :

1. Navigate to **Security > Web server > Certificate Authority Trust > Root CA Certificate**.
2. Click **Choose file**.

Mitel | Mitel Standard Linux

Configure Web Server

Web Server Certificate | **Certificate Authority Trust** | TLS | Advanced

Manage Certificate Authority Trust

A default set of publicly trusted root CA certificates exists on the server. The functions below can be used to update the trusted CA certificate store.

The following additional root CA certificates are currently installed:

Certificate Name	Action
DigiCert Global Root G2	<button>Remove</button>
Mitel Networks Root CA	<button>Remove</button>
Mitel Products Root CA	<button>Remove</button>

To upload new root CA certificates to the installed CA trust bundle, choose your certificate file below and click the install button.

Note: the file must contain X.509 root CA certificates in PEM format.

Root CA Certificate: No file chosen

Mitel Standard Linux 12.1.17.0
MiVoice Border Gateway 12.1.0.110
© Mitel Networks Corporation

3. Select the certificates to be uploaded.

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

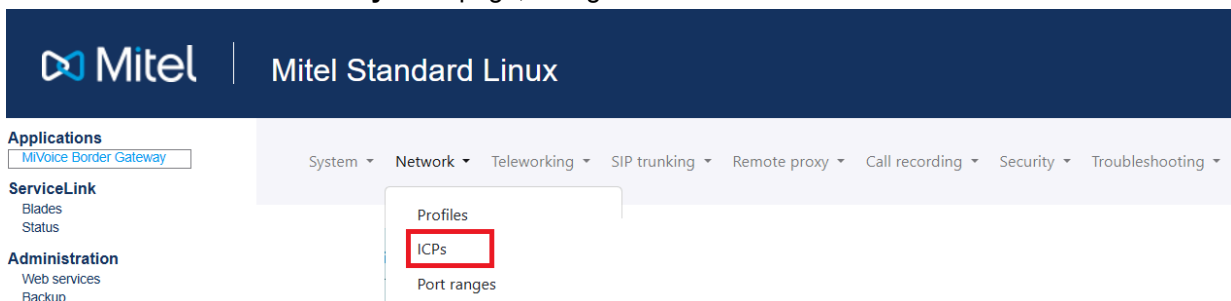


6.1.4 Adding a Network ICP

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: ZOOMMVB).
- 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 opened: Tue Feb 04 2020 11:40:23 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: ZOOMMVB

Type: MiVoice Business

SIP capabilities: UDP, TCP, TLS | Export root cert

Hostname or IP address: 10.35.32.2

MiNet installer password:

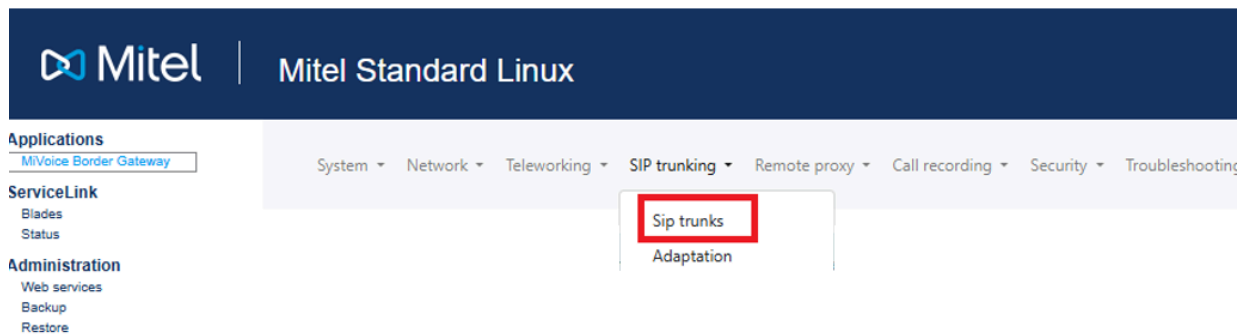
Indirect call recording capable: ☐

Save

6.1.5 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

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**).

6.2 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.

6.2.1 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.

Licenses

- License and Option Selection
- System Capacity
- Dimension Selection
- Application Group Licensing

LAN/WAN Configuration

Voice Network

System Properties

Hardware

Trunks

Users and Devices

License and Option Selection

SIP Trunks

SIP Trunks	2	353	0	353	Unr
------------	---	-----	---	-----	-----

Others

IDS Connection	1	Yes	0	1	Unr
MLPP	0	No	0	0	Unr

Configuration Options

Country: North America

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.

6.2.2 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 shows the 'Class of Service Options' configuration page. At the top, there is a navigation bar with 'Page 1 of 11', a 'Go to' dropdown, and a 'Go' button. Below this is a header section with a green icon and the text 'Class of Service Options'. A table lists the configuration items, with the first item highlighted in blue: Class of Service Number 6, Comment ZOOMTr1. To the right of the table are 'Save' and 'Cancel' buttons. Below the table, there are two tabs: 'General' and 'Advanced'. The 'General' tab is selected. Under the 'General' tab, there are several configuration options: 'Class Of Service Number' (6), 'Comment' (ZOOMTr1), 'ACD' (Logout), 'ACD Agent Behavior on No Answer' (15), '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), 'Follow 2nd Alternate Reroute for Recall to Busy ACD Agent' (No), 'Work Timer' (0), and 'Announce' (No). Each option has a corresponding input field or radio button.

5. Search for **Public Network Access via DPNSS** and Select **Yes**.
6. Leave all other fields at their default settings.
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 of 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**.

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

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.

Mitel | MiVoice Business SDS Distribution Error Status: **Minor**

US1 Class of Service Options on US1 Search DN Show form on US1 (Login Node) Go

Change Copy Print... Import... Export... Data Refresh

Page 1 of 11 Go to Value Go

Class of Service Options	
5	ZOOM_COS
6	ZOOMTr1
7	ZOOMTr2
8	ZOOMTr3

General Advanced

6.2.3 Network Zone Assignment

To configure the Network Zone Assignment:

1. Navigate to **Voice 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**.

The screenshot shows the Mitel MiVoice Business configuration interface. On the left is a sidebar with a menu including 'Licenses', 'LAN/WAN Configuration', 'Voice Network', and various sub-items like 'Network Elements', 'Cluster Elements', 'Analog Gateway Servers', 'Admin Groups', 'Fax Service Profiles', 'Fax Advanced Settings', 'Network Zones', 'Network Zone Topology', 'Bandwidth Management', 'Codec Settings', and 'Mass Audio Notification'. The 'Voice Network' section is expanded, and 'Network Zones' is selected. The main area displays a 'Change' form for 'Network Zones'. The form has a title bar 'Change' and a subtitle 'Network Zones'. It contains the following fields: 'Zone ID' (value: 1), 'Intra-zone Compression' (radio buttons: No selected, Yes), 'Group Zone' (empty), 'Intra-zone Fax Profile' (value: 1), 'Label' (value: ZOOM), 'SMDR Tag' (empty), 'Time Zone' (dropdown menu: US/Central), 'LBN Prefix' (empty), 'Zone CESID' (empty), 'Default Billing Number' (empty), 'Default CPN' (empty), 'Audio Source' (empty), and 'Embedded Music Source' (empty). At the bottom right are 'Save' and 'Cancel' buttons.

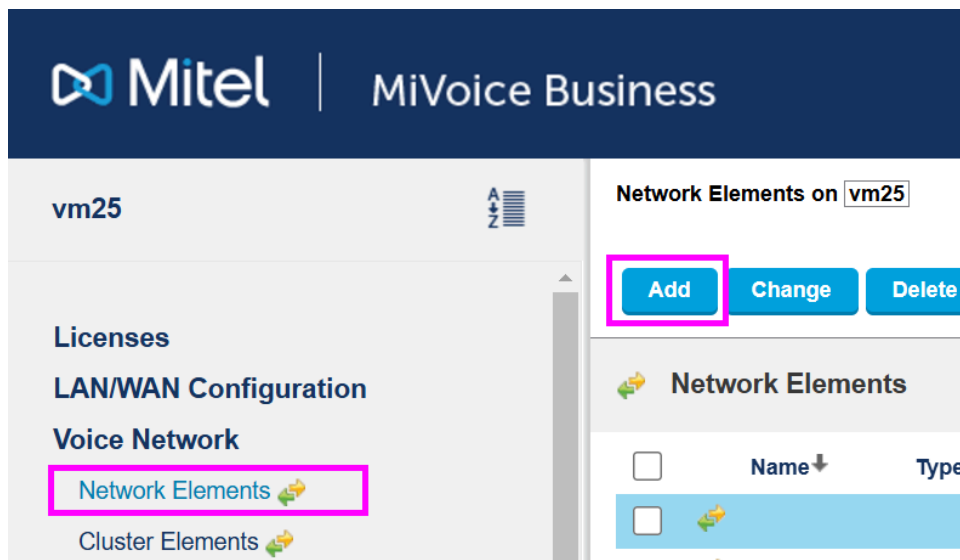
6.2.4 Adding Network Elements

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**.

Change

Network Elements

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

Save **Cancel**

6.2.4.1 Setup as per SIP provider requirements

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	MBG_PSTN
Type	Outbound Proxy ▼
FQDN or IP Address	10.64.6.55
Local	False
Version	
Zone	1
Outbound Proxy Specific	
Outbound Proxy Transport Type	TCP ▼
Outbound Proxy Port	5060

Save **Cancel**

6.2.4.2 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 **Other**.
- 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**).

6.2.4.3 Adding PSTN Region Network Element

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

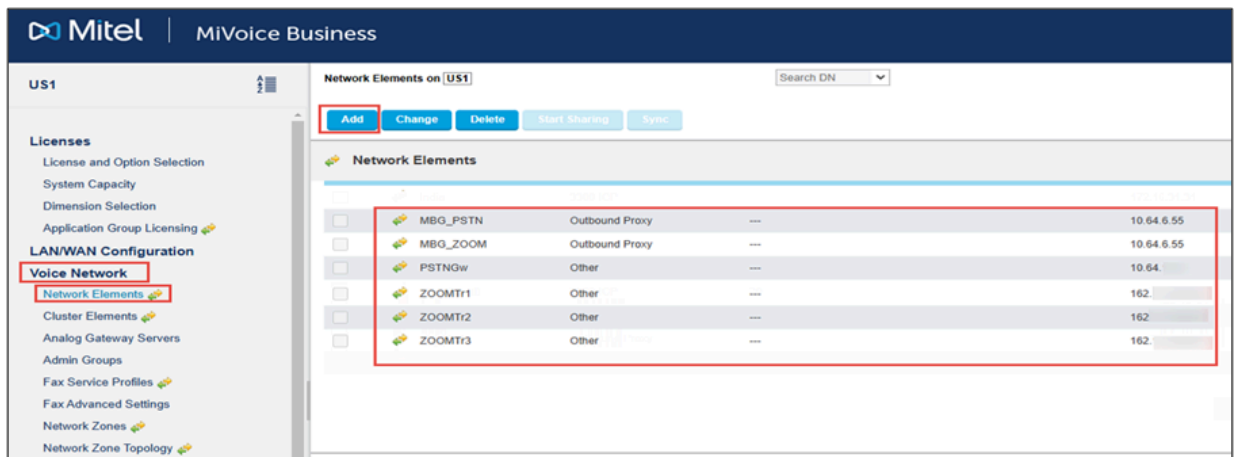
1. Navigate to **Voice Network > Network Elements** form.
2. Click **Add**.
3. Configure the following:
 - a. **Name**: Enter the name. (Example: **PSTNGw**).
 - b. **Type**: From the drop-down menu, Select **Others**.
 - c. **FQDN or IP Address**: Enter the PSTN Gateway IP address.
 - d. **Zone**: Enter 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 web form titled "Change Network Elements". The form contains the following fields and values:

- Name**: PSTNGw
- Type**: Other (selected from a dropdown menu)
- FQDN or IP Address**: 10.64.1.72
- Local**: False
- Version**: (empty field)
- Zone**: 1
- SIP Peer**: ☒
- SIP Peer Specific**:
 - SIP Peer Transport**: TCP (selected from a dropdown menu)
 - SIP Peer Port**: 5060

At the bottom right of the form are two buttons: **Save** and **Cancel**.

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



6.2.5 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.

6.2.5.1 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.

Mitel | MiVoice Business

vm25

Trunk Attributes on vm25

Change Change Page

Page 1 of 15

Trunk Attributes

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**).
- Leave the other field values as default.
- 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

Save

Cancel

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**.

6.2.5.2 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
<div>SIP Peer Profile Label SIPTRUNK</div>							
<div>Network Element PSTNGw</div>							
<div>Local Account Information</div>							
<div>Registration User Name</div>							
<div>Address Type <input type="radio"/> FQDN: mivbplano.tekvizionlabs.com <input checked="" type="radio"/> IP Address: 10.35.32.2</div>							
<div>Administration Options</div>							
<div>Interconnect Restriction 1</div>							
<div>Maximum Simultaneous Calls 20</div>							
<div>Minimum Reserved Call Licenses 3</div>							
<div>Outbound Proxy Server MBG_PSTN</div>							
<div>SMDR Tag 0</div>							
<div>Trunk Service 5</div>							
<div>Zone 1</div>							
<div>Authentication Options</div>							
<div>User Name</div>							
<div>Password</div>							
<div>Confirm Password</div>							

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"/>					
Default CPN Name		<input type="text"/>					
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. **Allow Peer To User Multiple Active M-Lines**: Select **No**.
- b. **AVP Only Peer**: Select **No**.
- c. **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							
Emergency Call Headers				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				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 EHDU				<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				No ▼			
Suppress Redirection Headers				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**.

6.2.5.2.1 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.

The screenshot shows the 'Basic' tab of the SIP Peer Profile configuration. The fields are organized into sections:

- SIP Peer Profile Label:** ZOOMTr1
- Network Element:** ZOOMTr1
- Local Account Information:**
 - Registration User Name: (empty)
- Address Type:**
 - ☐ FQDN: mivbplano.tekvizionlabs.com
 - ☒ 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: (empty)
 - Password: (empty)
 - Confirm Password: (empty)
 - Authentication Option for Incoming Calls: No Authentication
 - Subscription User Name: (empty)
 - Subscription Password: (empty)
 - Subscription Confirm Password: (empty)
- 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. Allow Peer To User Multiple Active M-Lines: Select **No**.
- b. AVP Only Peer: Select **No**.
- c. 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**).

6.2.6 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.

6.2.6.1 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.

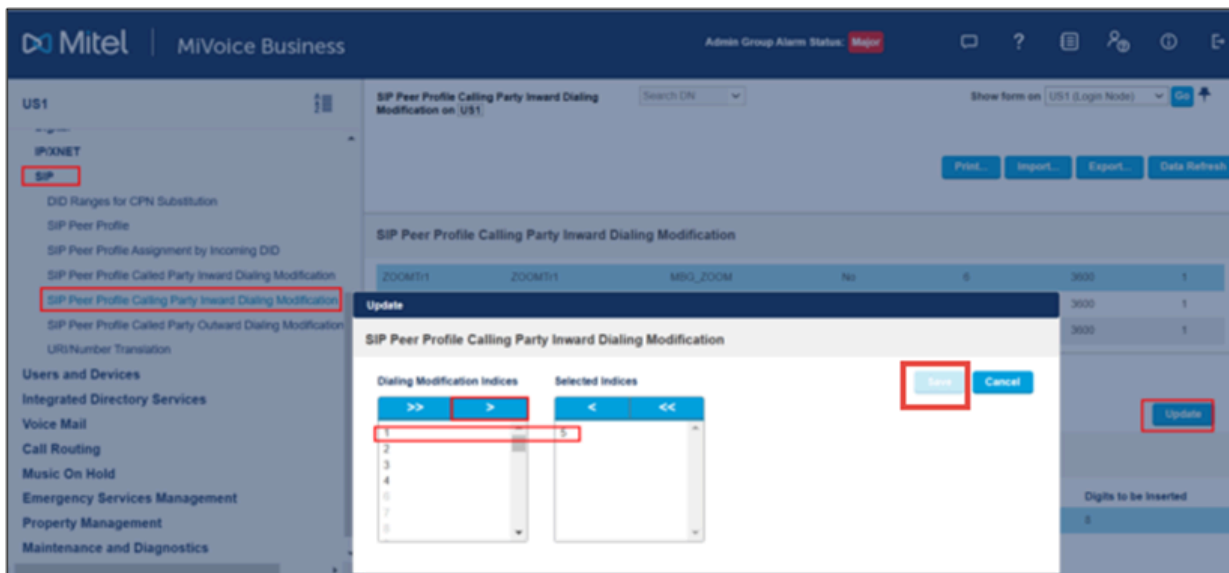
1. Navigate to **System Properties > System Feature Settings > Inward Dialing Modification** form.
2. Select **Change**.
3. Configure the following:
 - a. **Index**: Remember the default Index number. (Default is **5** here).
 - b. Set **Digits to Match**: Enter **x** (any) as a wild card to match any 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**.

6.2.6.2 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 an extension number or a PSTN number from the Zoom clients.



6.2.7 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**.

6.2.7.1 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

SIP Trunk

Trunk Group Number

SIP Peer Profile

SIPTRUNK

PBX Number / Cluster Element ID

COR Group Number

1

Digit Modification Number

2

Digits Before Outpulsing

Route Type

Compression

Off

Save

Cancel

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**).

6.2.7.2 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**.

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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 Route Lists on US1 Search DN

ARS Route Lists Search:

Find a field named: List Number that has a value of:

Change **Clear**

ARS Route Lists

List Number	1st Choice route	2nd Choice route	2nd Choice Warning Tone	3rd Choice route	3rd Choice Warning Tone	4th Choice route
2	11	12	No	13	No	
3			No		No	
4			No		No	

6.2.7.3 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**

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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
8	Unknown	Route	7

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	<input type="button" value="Change to"/>	8	
Number of Digits to Follow	<input type="button" value="Change to"/>	Unknown	-
Termination Type	<input type="button" value="Change to"/>	Route	-
Termination Number	<input type="button" value="Change to"/>	7	

[Preview](#) [Save](#) [Cancel](#)

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**

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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	<input type="button" value="Change to"/>	972598	
Number of Digits to Follow	<input type="button" value="Change to"/>	0	-
Termination Type	<input type="button" value="Change to"/>	List	-
Termination Number	<input type="button" value="Change to"/>	2	

Preview

Save

Cancel

