A MITEL PRODUCT GUIDE

MiVoice MX-ONE

Zoom with MX-ONE and MBG - Administrator Guide (BYOC and BYOP)

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Powering connections

Notices

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Contents

Introduction 1

This chapter contains the following sections:

- Prerequisites
- Related Documentation

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

This document provides instructions on setting up MiVoice Border Gateway (MBG)/MX-ONE for interoperability between the Generic SIP Trunk and Zoom Phone environments. The interoperability compliance testing focuses on verifying inbound and outbound call flows between MiVoice Border Gateway (MBG)/MX-ONE and Zoom Cloud.

This hybrid integration model allows organization to leverage Zoom's cloud platform while maintaining their existing MX-ONE 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 MX-ONE for call management.

How it works:

The integration allows Zoom Phone to connect to the MX-ONE system through a Generic SIP Trunk. MX-ONE and MiVoice Border Gateway manage the communication between Zoom Phone and external networks, including the PSTN (Public Switched Telephone Network). MX-ONE 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 MX-ONE system while benefiting from Zoom's cloud features. Proper configuration of both MX-ONE and MiVoice Border Gateway within the user environment is essential for successful deployment. Once MX-ONE 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.

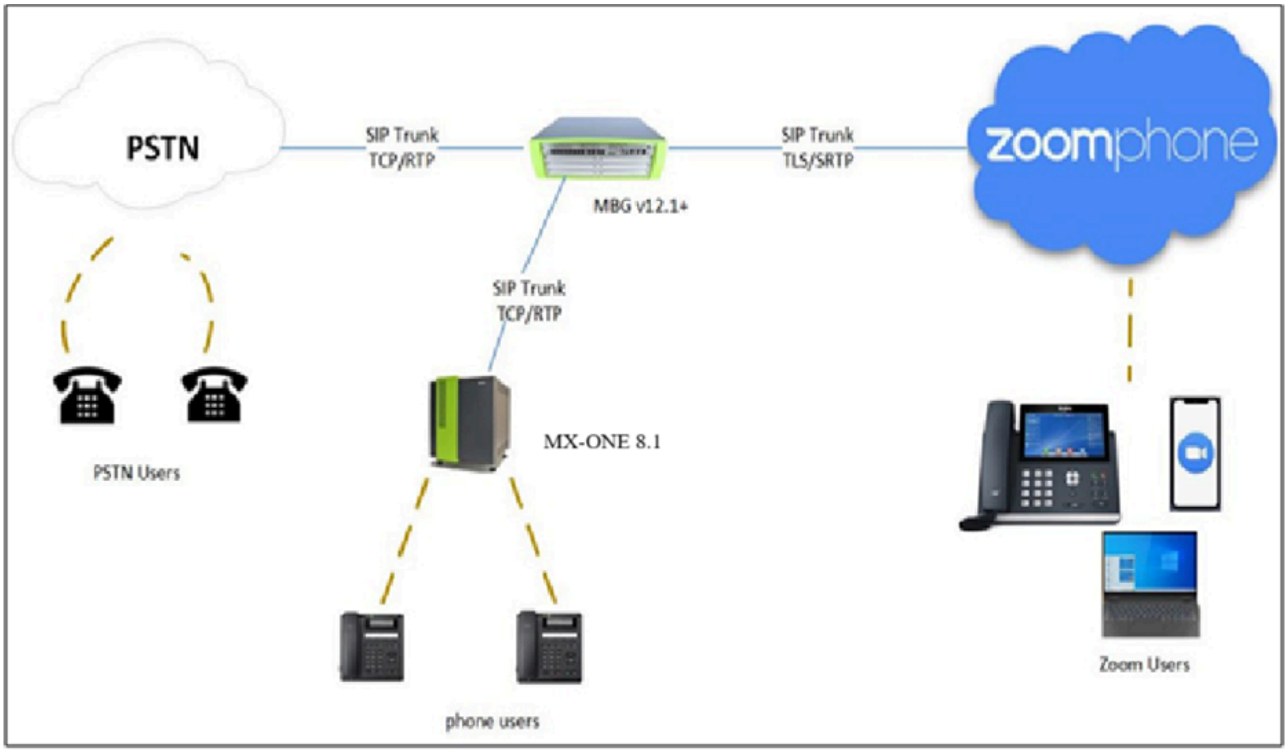


Figure 1: Network Topology Block Diagram

Note: 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.

1.1 Prerequisites

Supported product versions

Product	SW Version (minimum)
Zoom Workplace app	6.2.0
MiVoice Border Gateway	12.1.0.110
MiVoice MX-ONE	8.1

1.2 Related Documentation

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

Zoom Web Portal Configuration 2

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 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.

Note: 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 MX-ONE integration, you must first add users to your Zoom account and assign licenses to them.

2.1 Adding Your SBC

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

Prerequisites

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

Adding your SBC

1. Log in to the **Zoom Admin Portal**.
2. Navigate to **Phone System Management** > **Company Info** > **Account Settings** > **Routing.**
3. Locate **the Session Border Controllers** section and click **Manage**.
4. Click **Add**.

5. Configure the following:

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

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

Note:

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

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

2.1.1 Configuring the Route Group

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

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

To add a Route Group:

1. Navigate to **Phone System Management** > **Company Info** > **Account Settings** > **Routing**.
2. Locate the **Route Groups** section and click **Manage**.
3. Click **Add**.

4. Configure the following:

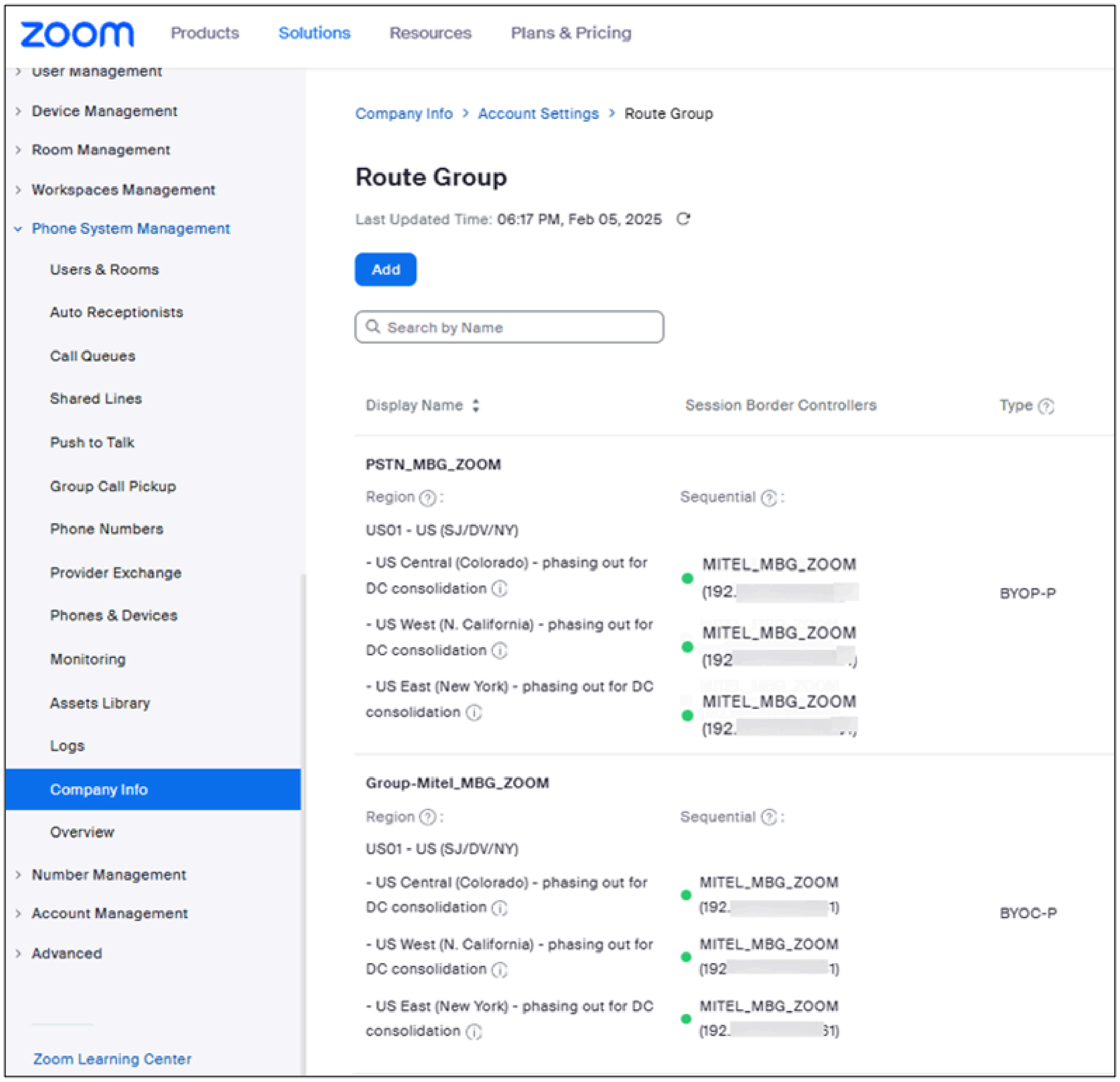
   a. **Display Name:** Type the display name of your choice. For example, **PSTN_MBG_ZOOM**.
   b. From the **Type** drop-down menu, select **BYOC-P**.
   c. From the **Region** drop-down menu, select the region code for your location. The format will be similar to: **US01-US(SJ/DV/NY)**

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

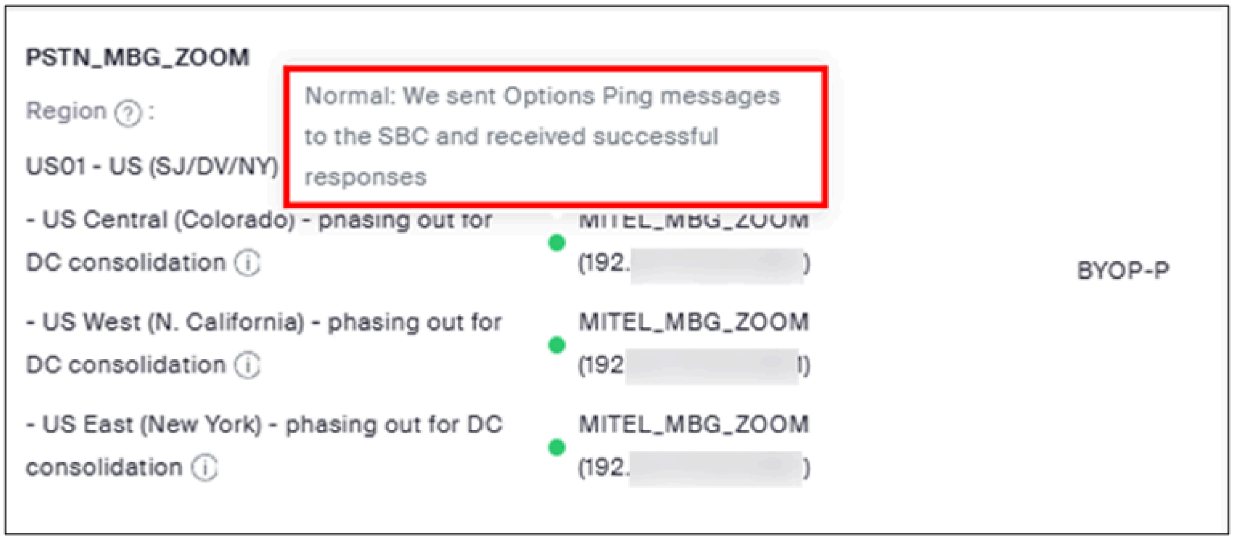
   d. From the **Distribution** drop-down menu, select **Sequential** and then from the **Session Border Controllers** drop-down menu, select the *MITEL_MBG_ZOOM* that was created in Adding Your SBC section.

5. Click **Save**.

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



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



2.1.2 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**.

2.1.3 Configuring the Routing Rule

When configuring a BYOC 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**.

2.2 Adding Phone Users

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

Prerequisites

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

1. Log in to the **Zoom web portal**.
2. Navigate to **User Management** > **Users** > **Add Users**.
3. Configure the following in the **Add Users** pop-up:

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

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

Next steps

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

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

2.2.1 Assigning a Calling Plan to a phone user

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

Prerequisite

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

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

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

3. Under the **Profile** tab, locate the **Package** section and click **Assign**.
4. From the **Package** drop-down menu, select **US/CA Unlimited Calling Plan**, as shown below.
5. Click **Confirm**.

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

2.3 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**.

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**.

2.4 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 MX-ONE user. For example, **mxone_user2**
   - In the **Extension Number** field, enter the extension number of the MX-ONE 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 8** section.
4. Click **Save**.

Provisioning Desk Phones for Zoom Phone Users 3

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

1. Navigate to **Phone System Management** > **Phones & Devices**.
2. Click **Add**.
3. In the **Add Device** pop-up, configure the following:

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

Configuration for BYOP/BYOC 4

This chapter contains the following sections:

- MBG Configuration for BYOP/BYOC
- MX-ONE Configuration for BYOP/BYOC

The 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 MX-ONE IP-PBX 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 MX-ONE.

4.1 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 MX-ONE 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 4, 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 MX-ONE PBX.
2. Create two or more SIP trunks, each corresponding to a different Zoom region.

4.1.1 Licensing & Network Configuration

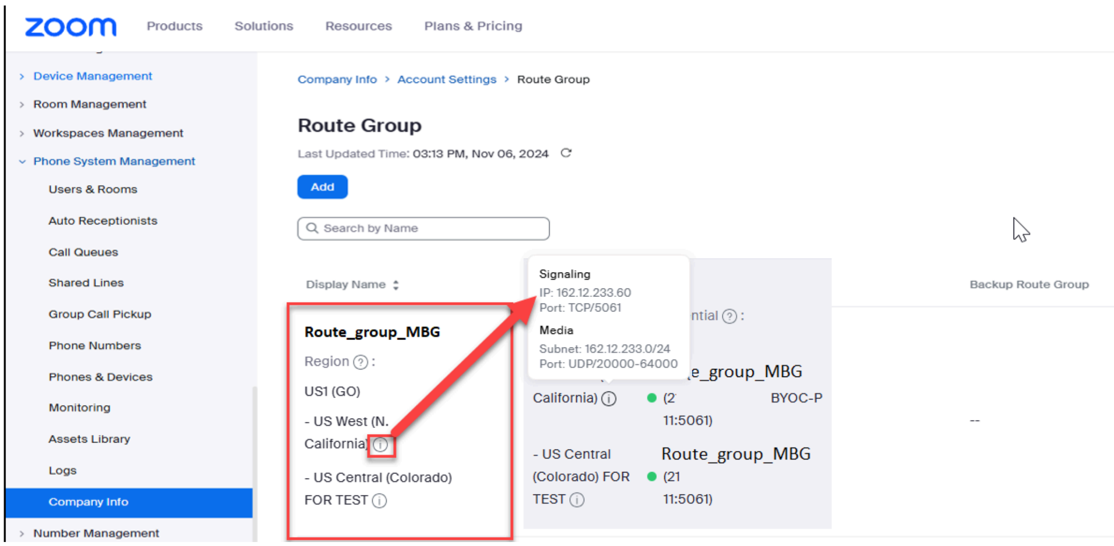
During the installation of the Mitel Border Gateway (MBG), you will be required to enter network configuration details for both the **WAN (Wide Area Network) and LAN (Local Area Network)**. These steps are standard for all MBG installations. For detailed instructions on setting up network configuration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

> **Note**: 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.

4.1.2 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.



4.1.3 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.

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

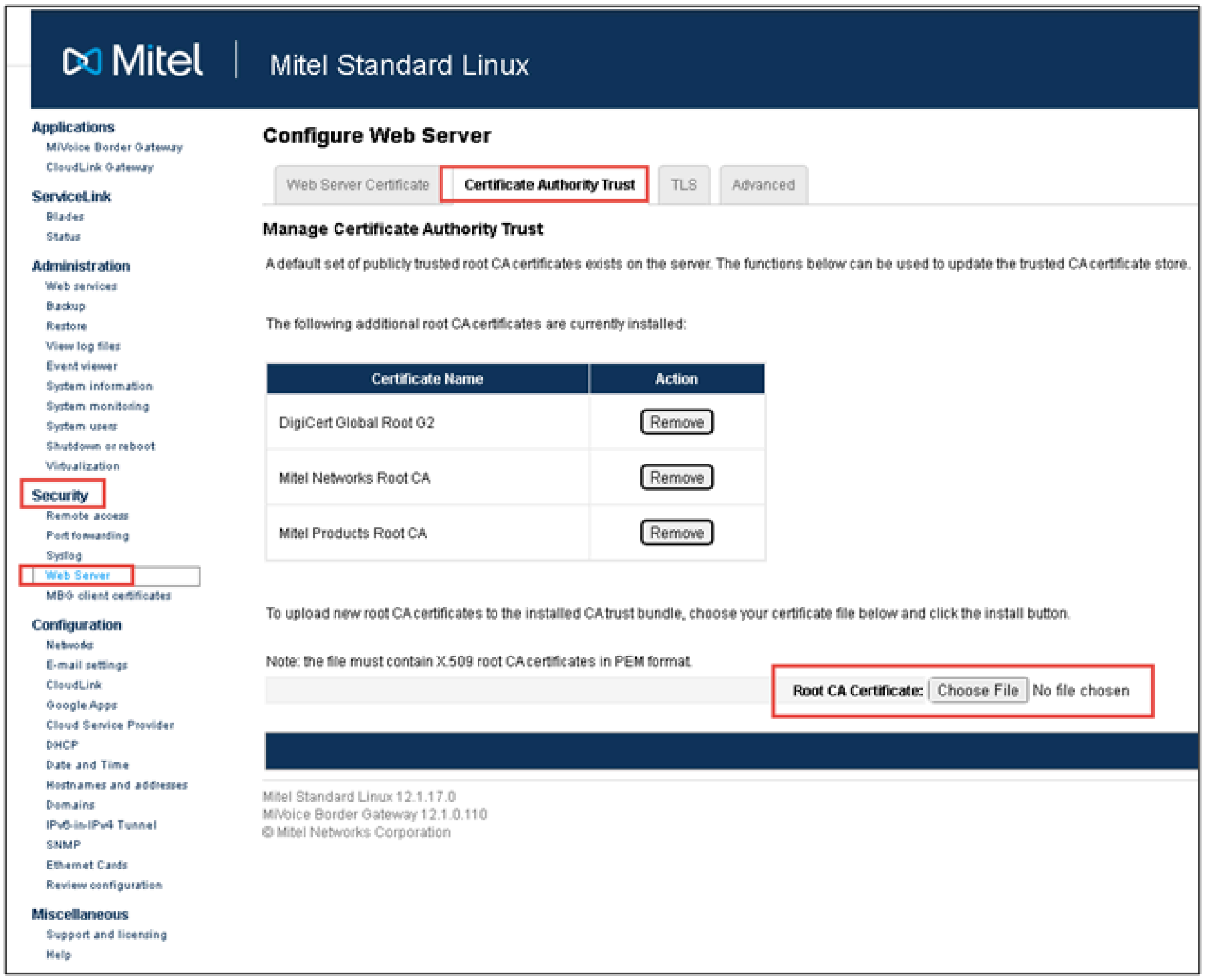
- A certificate authority or certification authority (CA) certificate (for example, "ca_chain.pem"). The CA certificate contains a public key and the owner's identity, ensuring an entity can be trusted.
- Server certificate for MBG (for example, "certificate.pem").
- MBG server certificate private key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

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



3. Select the certificates to be uploaded.

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

4.1.4 Adding a Network ICP

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

To add Network ICP:

1. On the **MiVoice Border Gateway** main page, Navigate to the **Network** > **ICPs**.
2. Click the **'+' icon** to add an ICP.
3. In the **Manage ICP** window, Configure the following:

   a. **Name**: Enter the name that you choose. (example: ZOOMMX-ONE).
   b. **Type**: From the Type drop-down menu, select the **MiVoice MX-ONE**.
   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**.

4.1.5 Adding SIP Trunks

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

To add SIP trunk:

1. On the **MiVoice Border Gateway** main page, Navigate to the **SIP trunking** > **Sip trunks**.
2. Click the **'+' icon** to add a new SIP Trunk.
3. In the **Manage SIP trunk** window, Configure the following:

    a. Profile field

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

    b. Connection field

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

    c.

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

    d. Trunk-side RTP security field

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

    e. ICP-side RTP security

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

4. Click **Save**.

To add a Rule:

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

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

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

3. Configure the following:

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

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

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

4.2 MX-ONE Configuration for BYOP/BYOC

This chapter describes the MiVoice MX-ONE configuration for connecting to the MiVoice Border Gateway. The purpose of this connectivity is for MiVoice MX-ONE 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 MX-ONE, you must configure with SIP Trunks that use the MBG as an Outbound Proxy Server to reach the Zoom regions. The MX-ONE is configured using the PM configuration interface. The System Administration Tool provides the forms used for configuration, as described below.

4.2.1 Licensing

The Mitel MiVoice MX-ONE system to function correctly, it must have both valid licensing and a proper network configuration. Before connecting the Mitel MiVoice MX-ONE system to SIP Trunking, you need to confirm that the system has a sufficient number of SIP trunk licenses. SIP trunk licenses allow the MiVoice MX-ONE 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.