A MITEL
PRODUCT
GUIDE

OpenScape Solution Set V11

Zoom Phone with OpenScape SBC and OpenScape Voice (Bring Your Own Carrier - BYOC)

Document Version 1.3

January 2025

Mitel
Powering connections

Contents

History of Changes 1

Issue	Date	Summary
1	10/2024	The first issue of the guide.
1.1	10/2024	Updated document for Unify OpenScape SBC/OpenScape Voice configurations.
1.2	11/2024	Updated access interface and routing configurations in Configuring Network settings on page 74. Added a table of other Zone Zoom IPs in Unify OpenScape SBC Configuration on page 70 for Unify OpenScape SBC/OpenScape Voice configurations.
1.3	01/2025	Updated entire document.

Introduction 2

This chapter contains the following sections:

This document outlines the process of connecting the **Unify OpenScape SBC** (OSSBC) and **OpenScape Voice** to **Zoom Phone** using Bring Your Own Carrier (BYOC)[1] and Bring Your Own PBX (BYOP)[2] configurations.

This integration provides a unified hybrid model that enables users to optimize the benefits of Zoom's cloud platform while maintaining connectivity with their on-premises telecom system (OSV) for telephony features. It is ideal for organizations that are currently using Zoom as a main collaboration tool and want to continue using their OSV system for call management and PSTN connectivity.

How it works:

The integration allows Zoom Phone to connect to the OSV system through a Generic SIP Trunk. OpenScape SBC and OpenScape Voice manage the communication between Zoom Phone and external networks, including the PSTN (Public Switched Telephone Network). OpenScape Voice 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 OpenScape SBC (OSSBC) links Zoom Phone and your on-premises infrastructure, ensuring smooth integration.

This solution provides secure traffic management, allowing users to retain their OSV system while benefiting from Zoom's cloud features. To support this integration, users must have OpenScape Voice and OpenScape SBC properly configured within their environment. Once OSV is configured, they can use the SBC to route calls, secure communication, and manage traffic between Zoom Phone and PSTN networks.

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

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

[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.

Product	Software Version
Unify OpenScape Voice	V10.3.31
OpenScape SBC	V11R1.0.0
Unify OpenScape Apps	V10 R3.10.1

2.1 Additional Support Information

In the current Unify product software implementation:

- OpenScape SBC with Unify OpenScape 4000 solution is supported.
- SBC standalone mode (without PBX) is currently supported.
- Domain-based Zoom multi-tenancy is supported.
- Comfort Noise generation is currently not supported by OpenScape SBC.
- The History-Info header is not currently supported by Unify OpenScape Voice and Unify OpenScape 4000.
- The OSEE environment with SBC-THIG and Zoom is currently not supported.

2.2 Network Topology

The block diagram below shows the Zoom Phone connection topology.

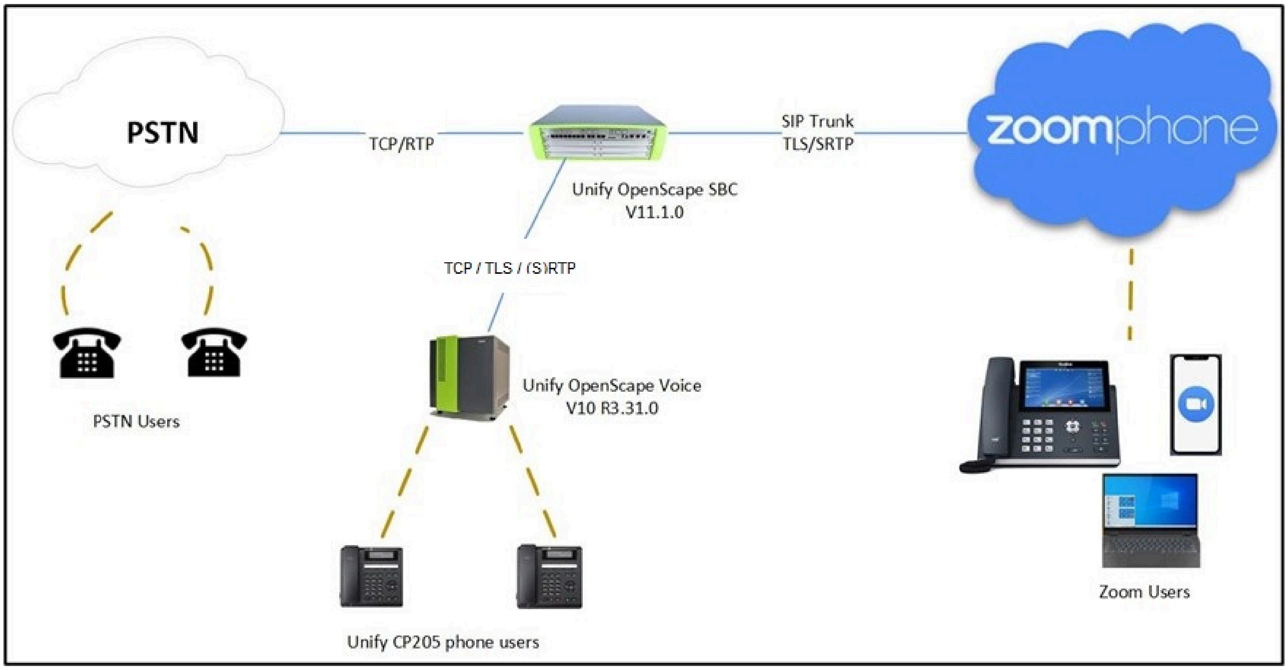


Figure 1: Network Topology Block Diagram

2.3 Related Documentation

For additional information on OpenScape SBC, refer to the following documents:

- OpenScape SBC V11 Administration Guide
- OpenScape SBC V11 Configuration Guide, Administration Documentation
- OpenScape SBC V11 Installation Guide
- OpenScape SBC V11 Security Checklist

For additional information on OpenScape Voice, refer to the following documents:

- OpenScape Voice V10 Administrator Guide
- OpenScape Voice V10 Service Manual, Service 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 the OpenScape 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.

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

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

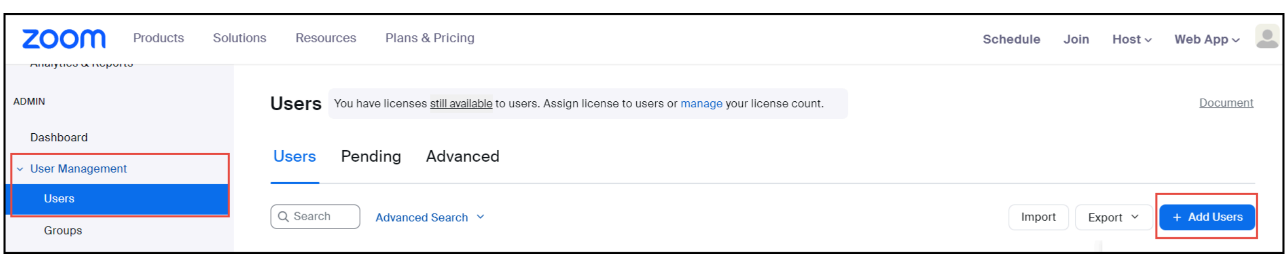
3.1 Adding Phone Users

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

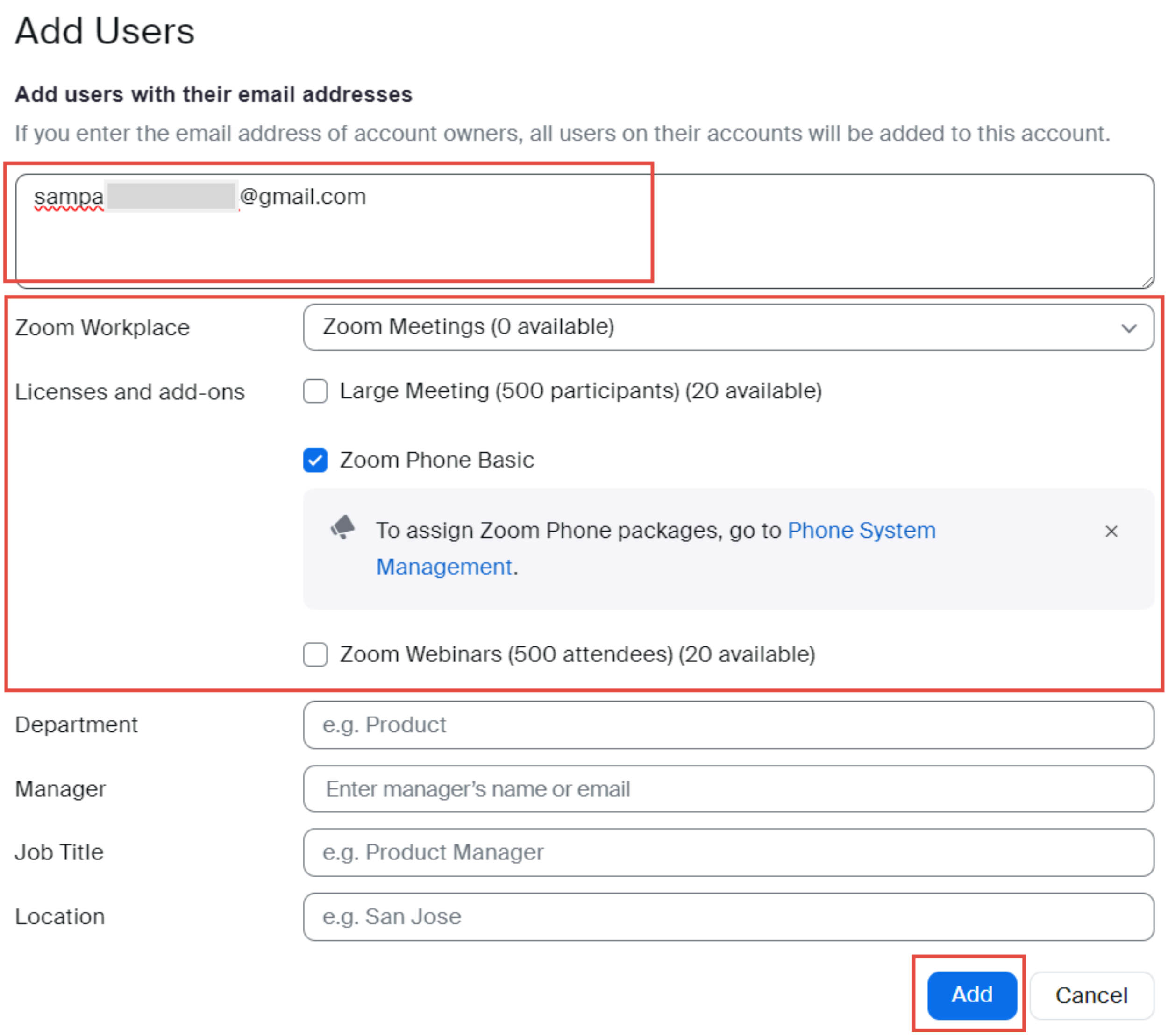
1. You have a Pro, Business, or Enterprise Zoom Phone account.
2. You are an administrator with the privilege to edit account settings.
3. You have completed the initial Zoom Phone setup. For more information, refer to Getting started with Zoom Phone (admin).

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**.



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

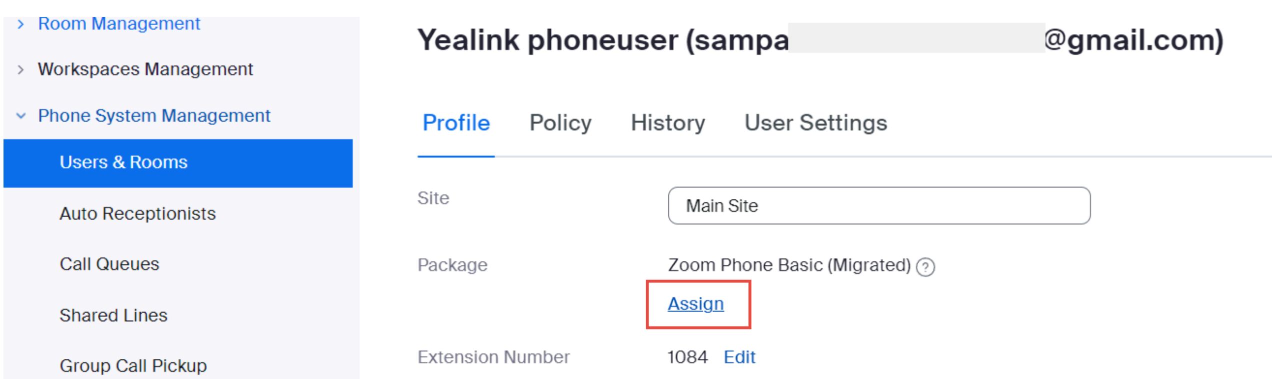
You can assign a calling plan to phone users to enable outbound calling.

Prerequisite

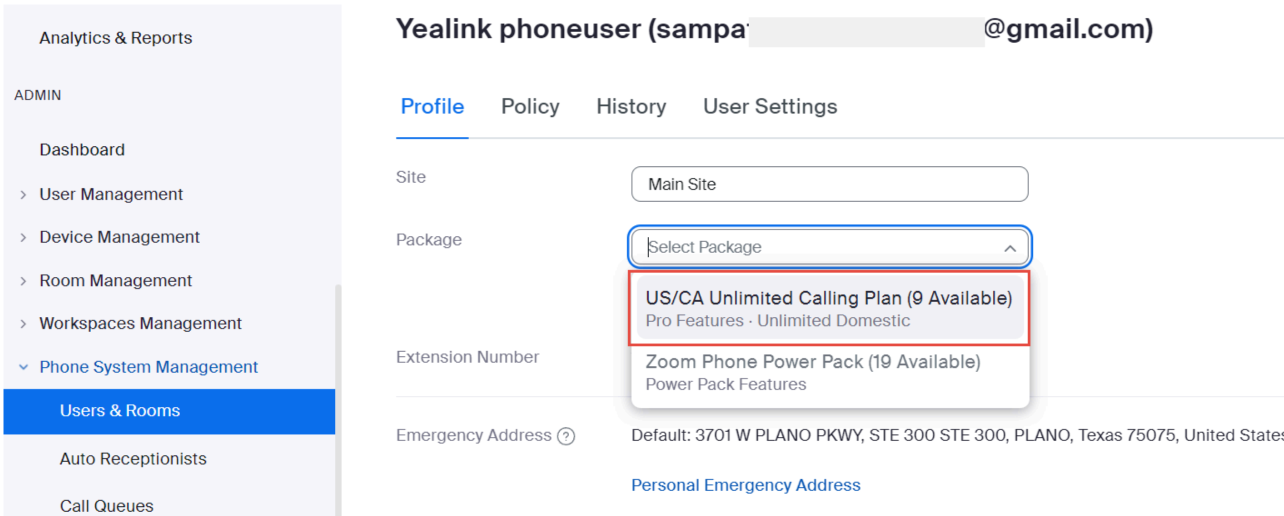
1. You are an administrator with the privilege to edit account settings.
2. You have assigned licenses to the phone users. For more information, refer to How to assign licenses.

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

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**.

3.2 Adding the OpenScape SBC

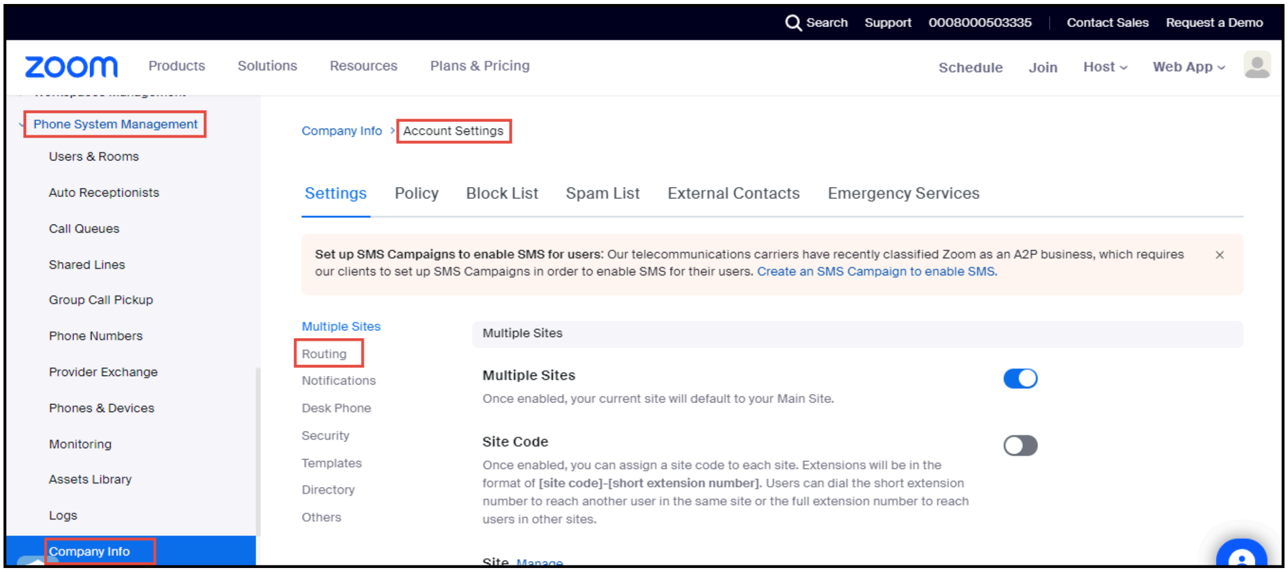
Follow the instructions below to add your OpenScape SBC in the Zoom Web Portal.

Prerequisites

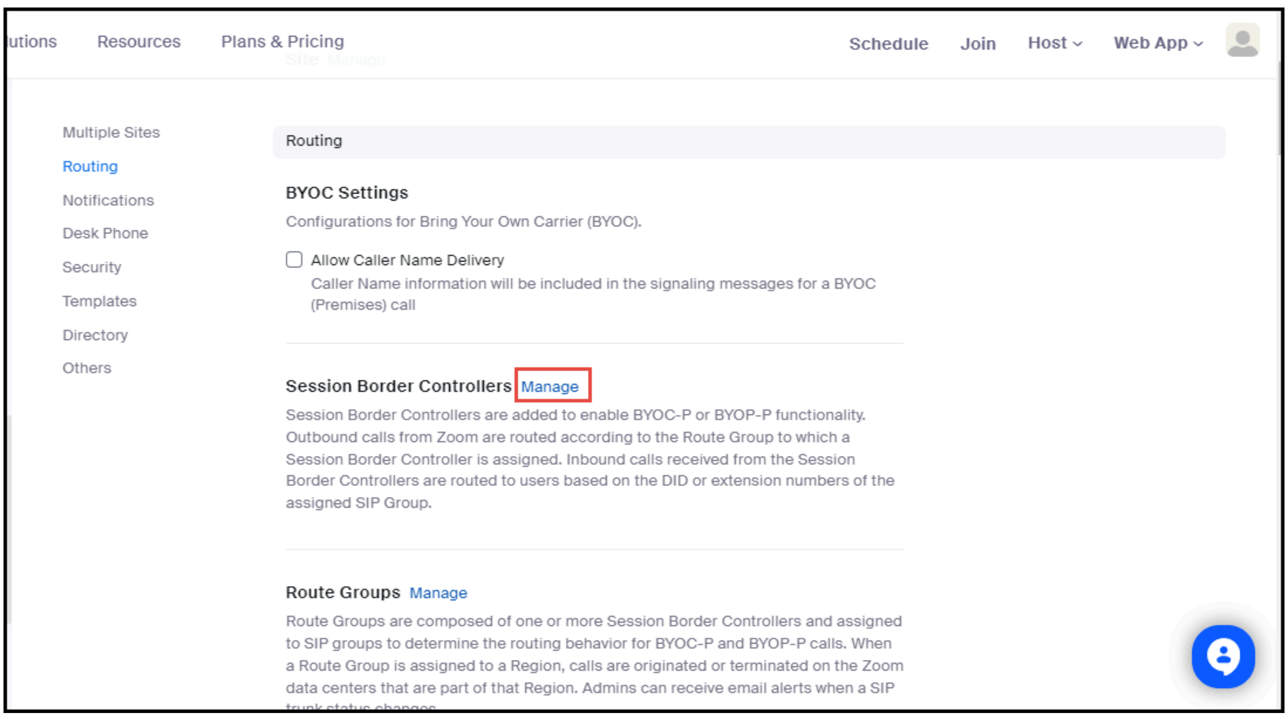
1. You are an administrator.
2. You have completed the initial Zoom Phone setup.
3. You have configured appropriate firewall rules for connectivity. For more information, refer to Zoom network firewall or proxy server settings.
4. You have a public IP address for SIP trunk connectivity.

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

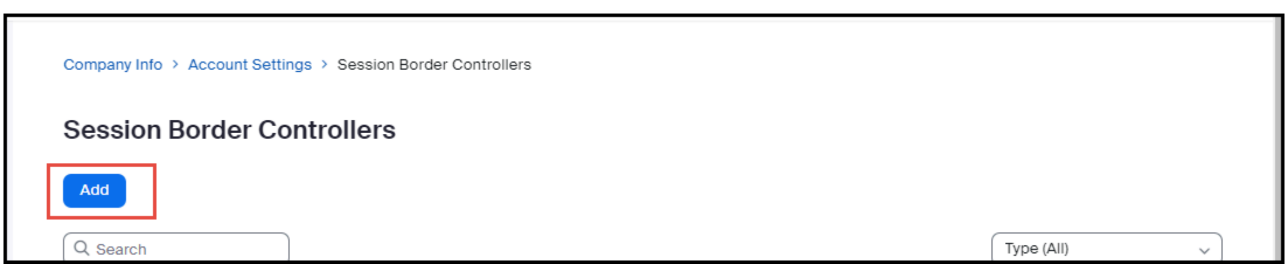
2. Navigate to **Phone System Management** > **Company Info** > **Account Settings** > **Routing.**



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



4. Click **Add.**



5. Configure the following:

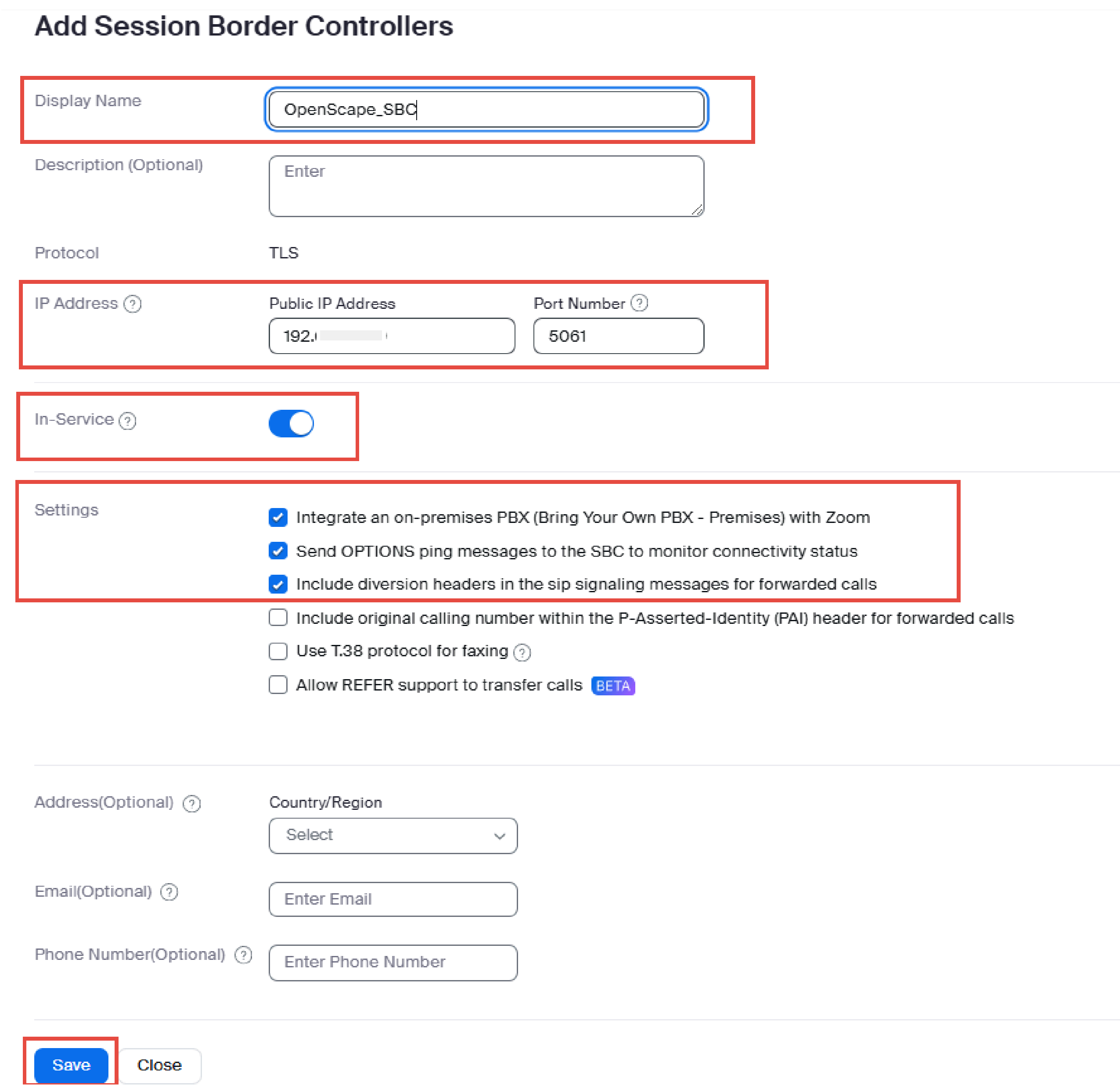
a. **Display Name:** Type the display name of your choice. For example, **OpenScape_SBC**.

b. **IP Address:** Enter the IP address of the OpenScape SBC 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 checkboxes**:**

- **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 OSSBC, 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.

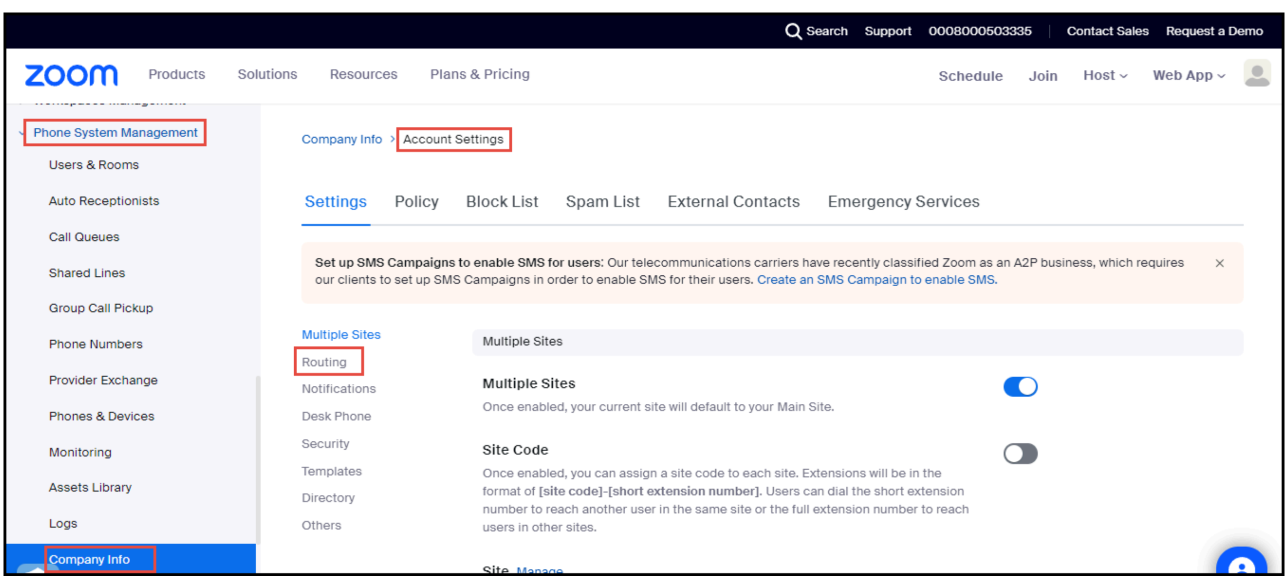
3.2.1 Configuring the Route Group

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.

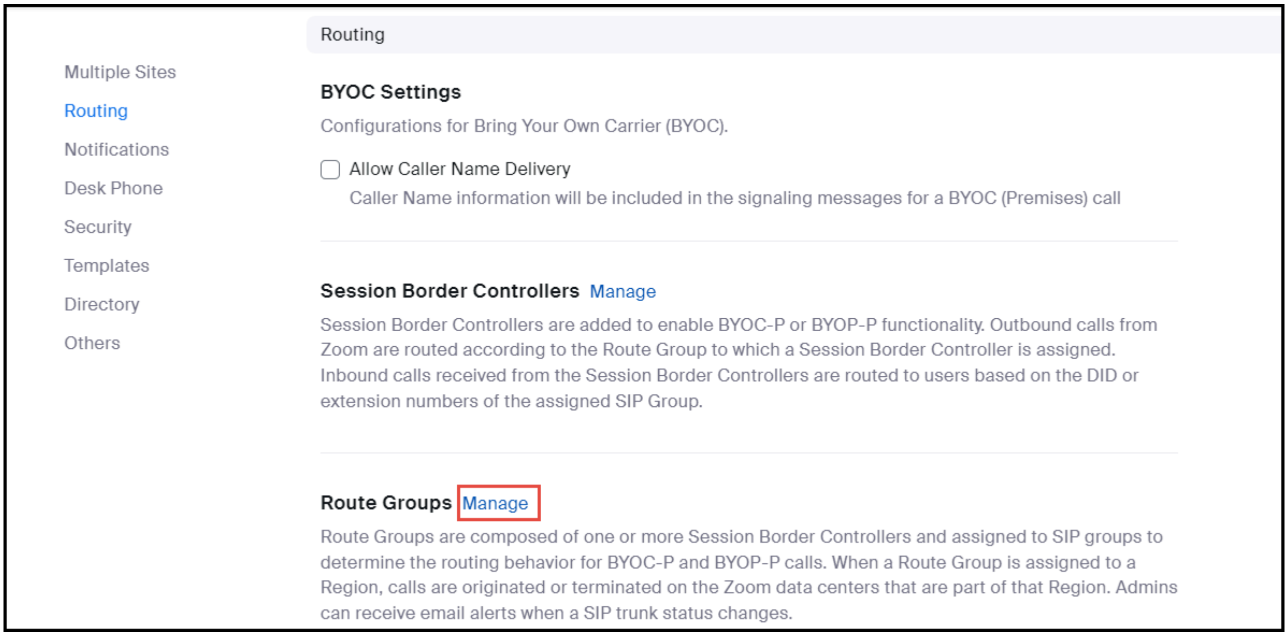
> **Note**:
> These configurations (Route Group, SIP Group, and Routing Rule) will take effect once phone numbers are added and assigned to the appropriate users. Until then, the routing logic will be in place, but calls will not be routed as expected.

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.**



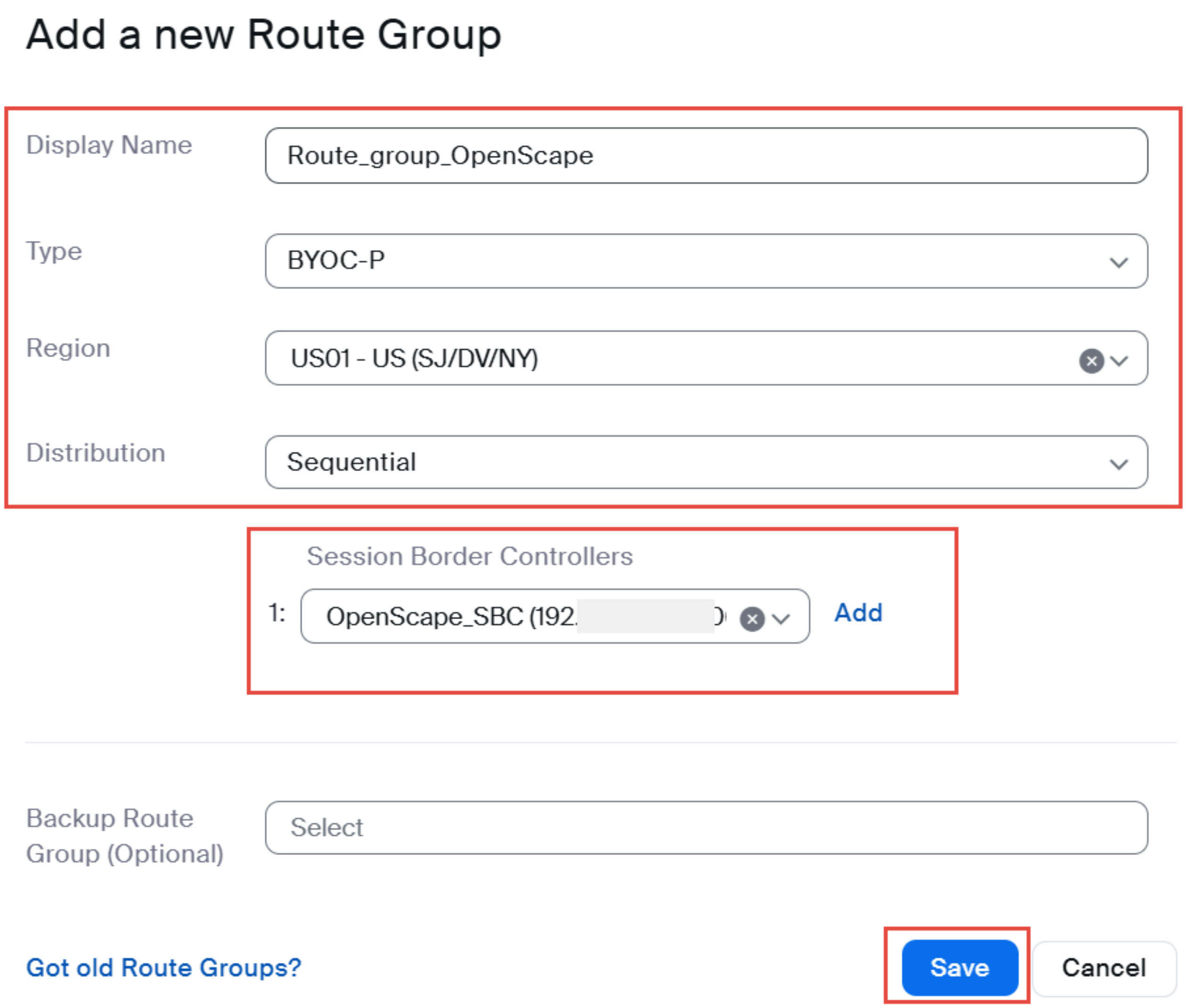
3. Click **Add.**

4. Configure the following:

   a. **Display Name:** Type the display name of your choice. For example, **Route_group_OpenScape**.
   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)**

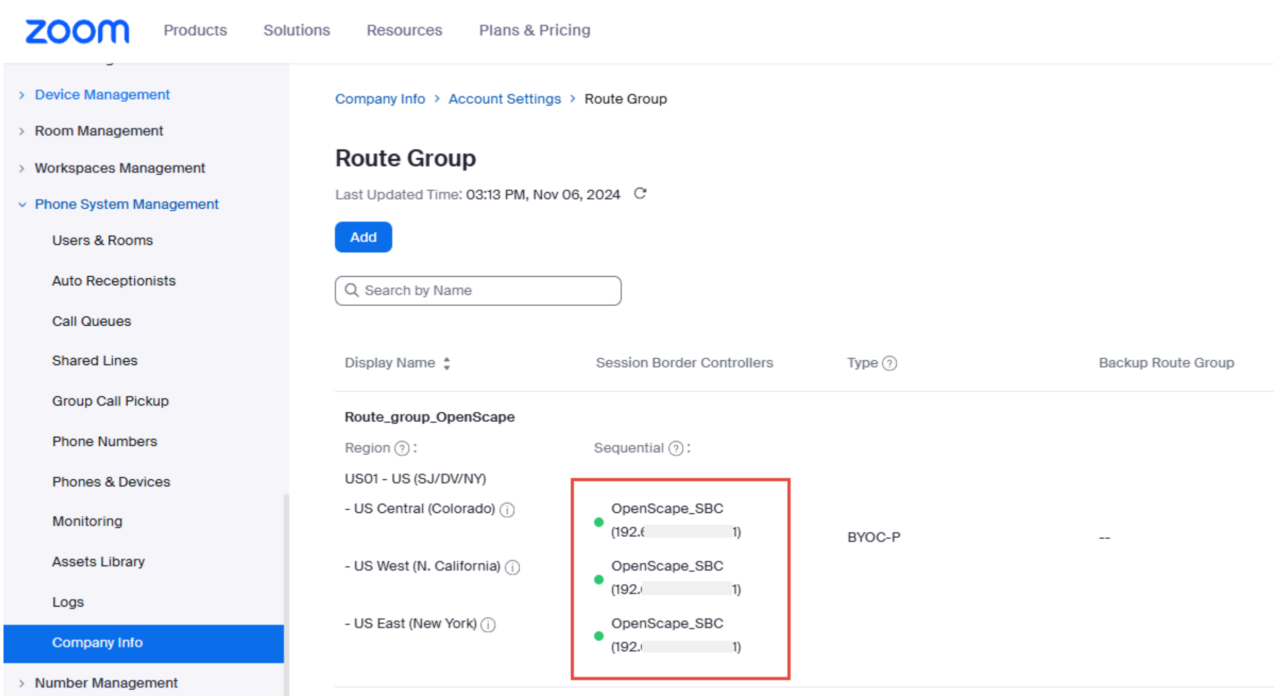
> **Note**:
> The format given above is an example.Choose the zone (SJ/DV/NY etc.) that is geographically closest to your SBC installation location.

d. From the **Distribution** drop-down menu, select **Sequential** and then from the **Session Border Controllers** drop-down menu, select the OpenScape_SBC that was created in Adding the OpenScape SBC on page 8.

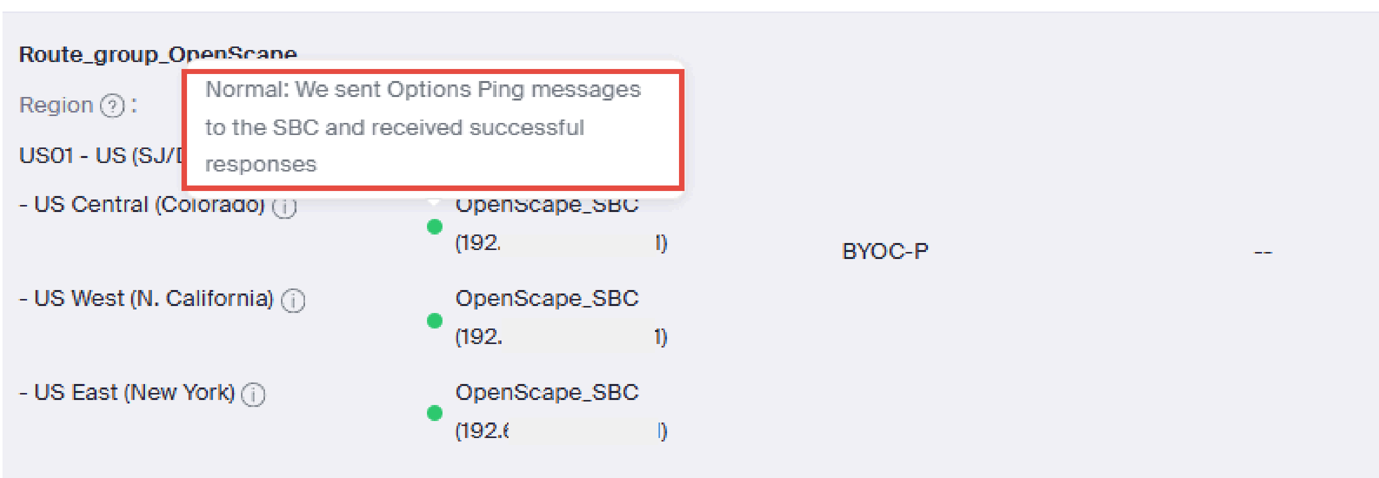


5. Click **Save**.

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



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

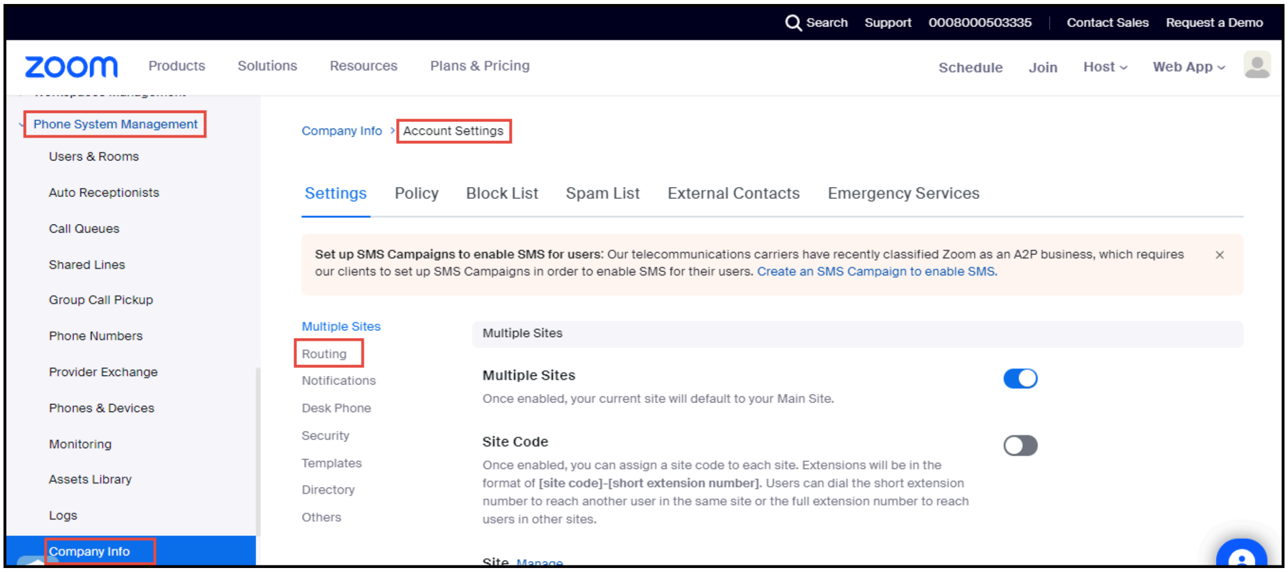


3.2.2 Configuring the SIP Group

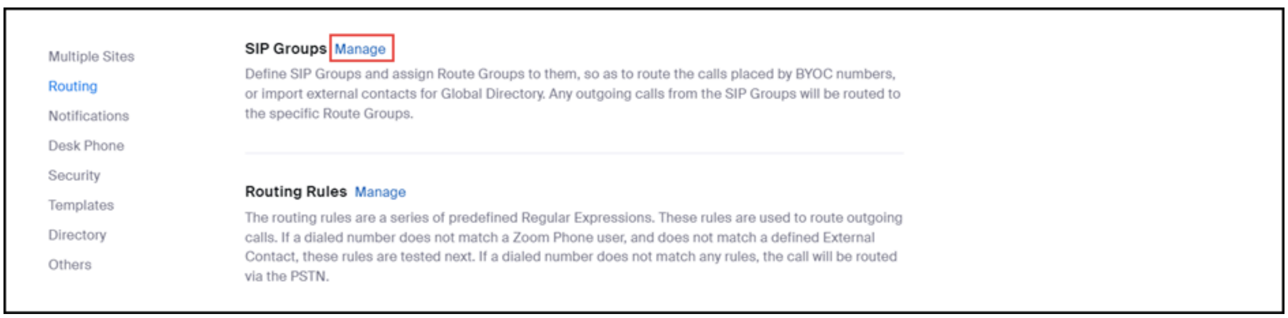
Follow the instructions below to configure SIP groups and assign Route Groups to them, in order to route calls placed by BYOC numbers. This step is mandatory for uploading the BYOC numbers.

To add a SIP Group:

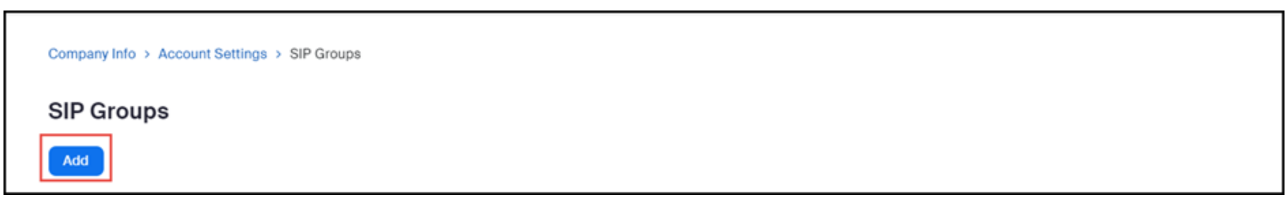
1. Navigate to **Phone System Management** > **Company Info** > **Account Settings** > **Routing**.



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



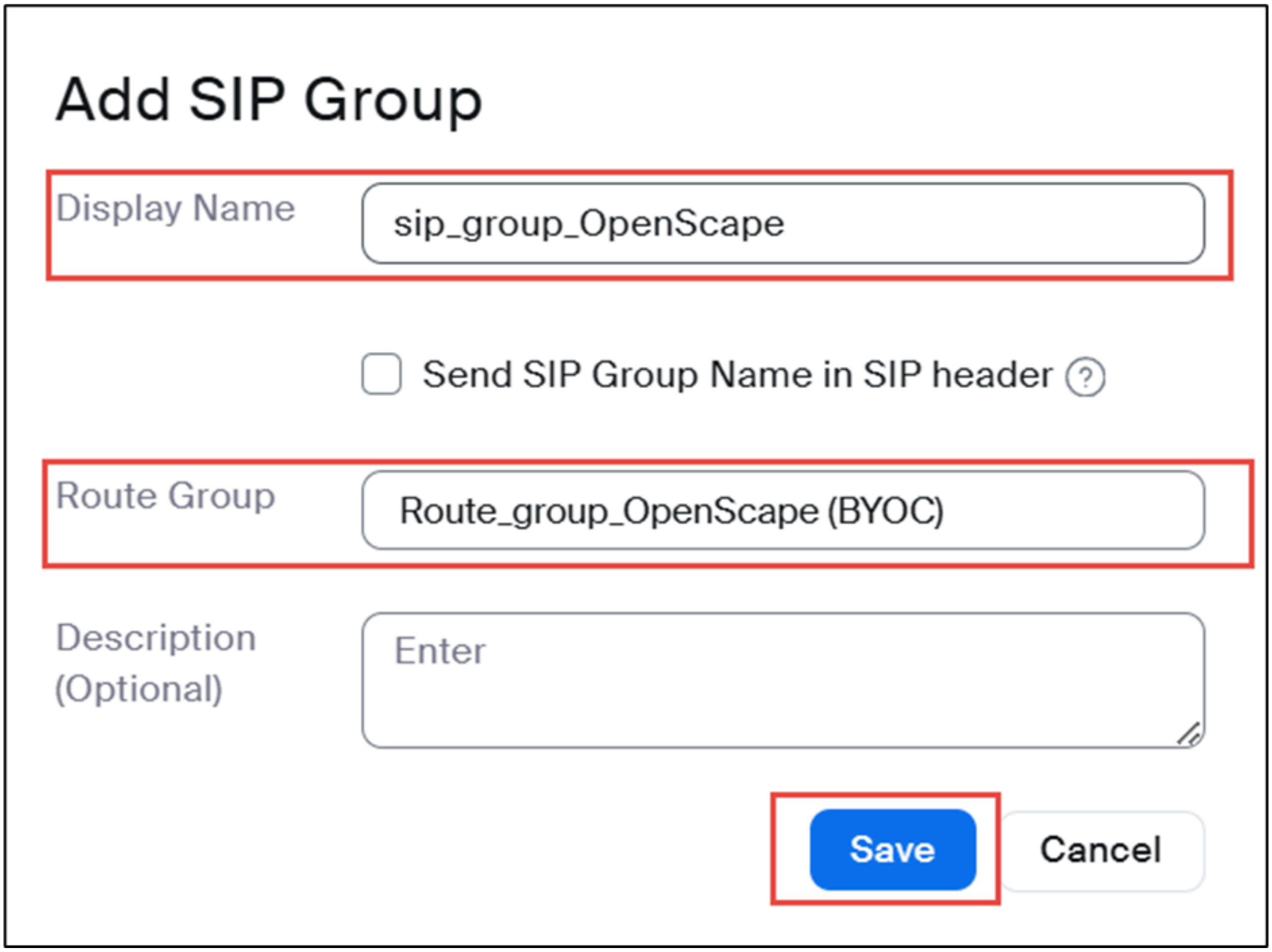
3. Click **Add.**



4. Configure the following:

   a. **Display Name:** Type the display name of your choice. For example, **sip_group_OpenScape**.

   b. From the **Route** drop-down menu**,** select the **Route_group_OpenScape (BYOC)** group, created in Configuring the Route Group on page 11.



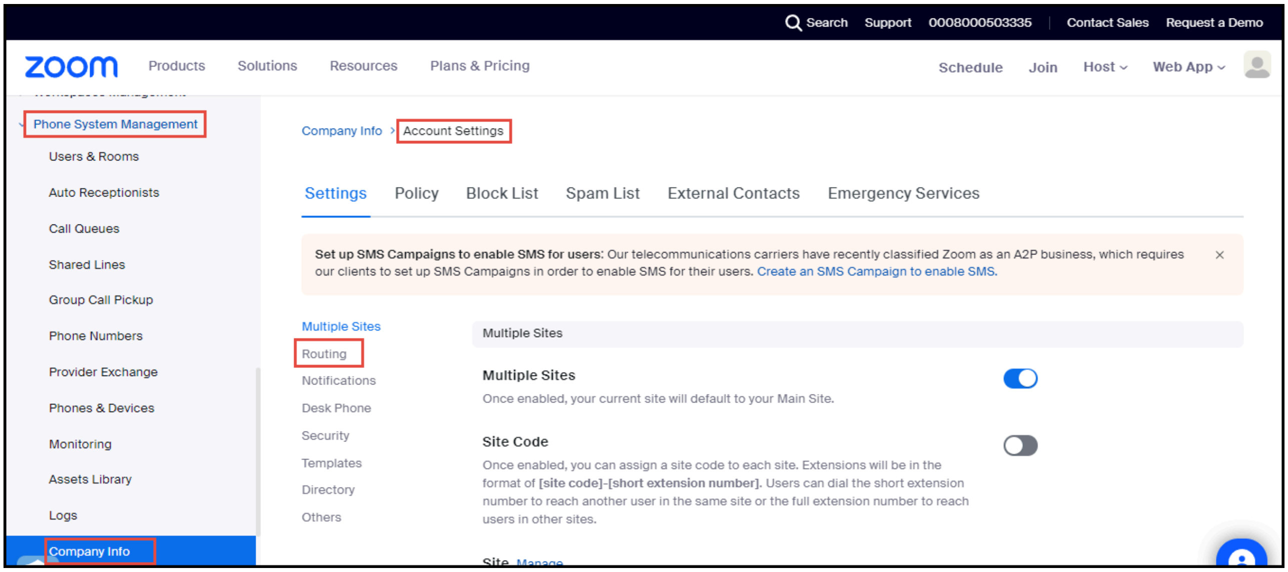
5. Click **Save.**

3.2.3 Configuring the Routing Rule

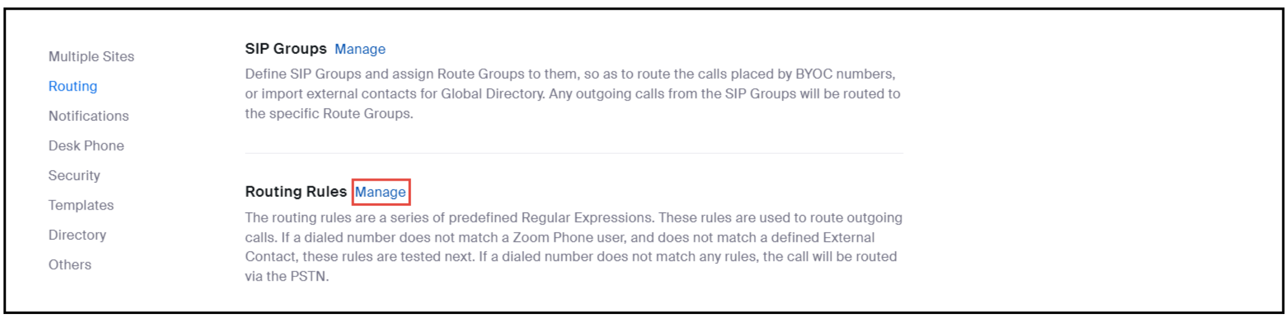
When configuring a **BYOC (Bring Your Own Carrier)** setup, you might create a routing rule to specify that calls from certain users or departments go through your OSSBC or network route.To add a Routing Rule for outbound calls:

Note:
Ensure that your Session Border Controller (OSSBC) 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.

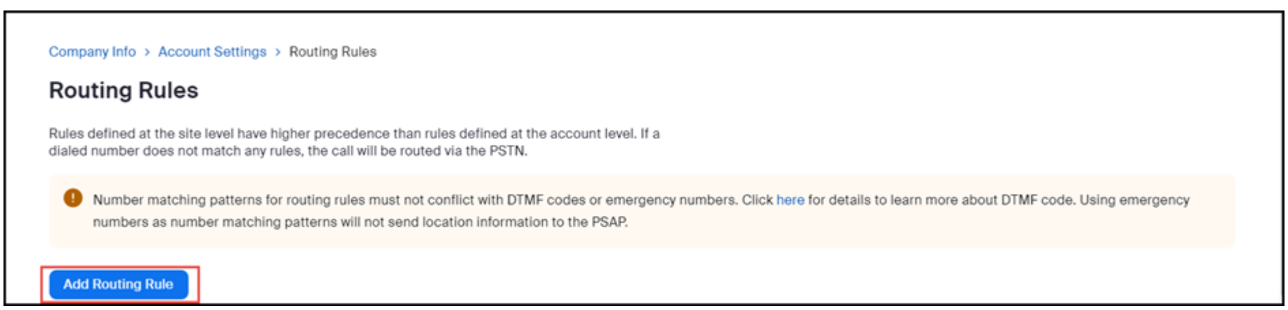
1. Navigate to **Phone System Management** > **Company Info** > **Account Settings** > **Routing**.



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

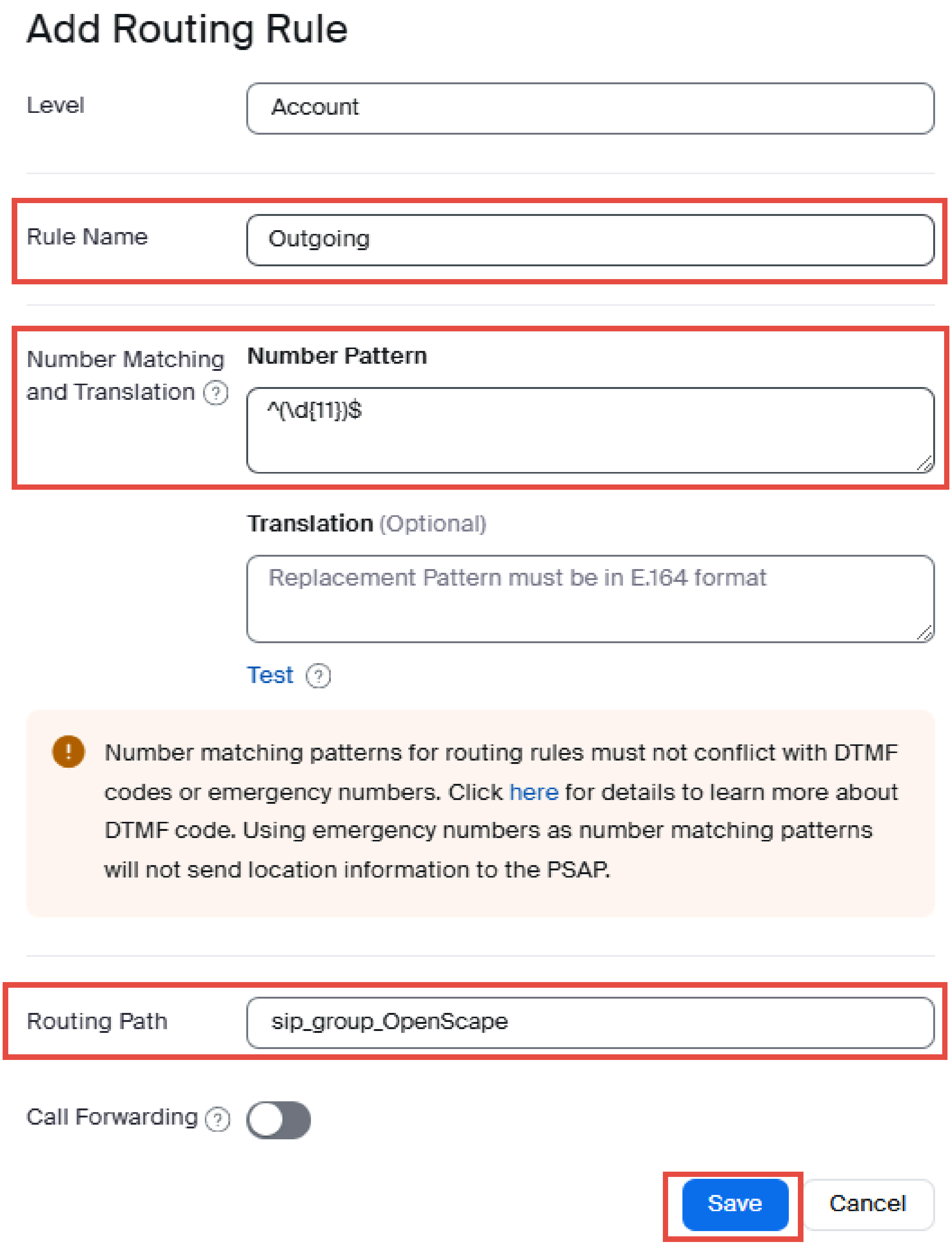


3. Click **Add Routing Rule** to add your rule**.**



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_OpenScape** routing path, created in 2.3 Adding SIP Group.


Add 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
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_OpenScape
Call Forwarding
Save
Cancel

5. Click **Save**.

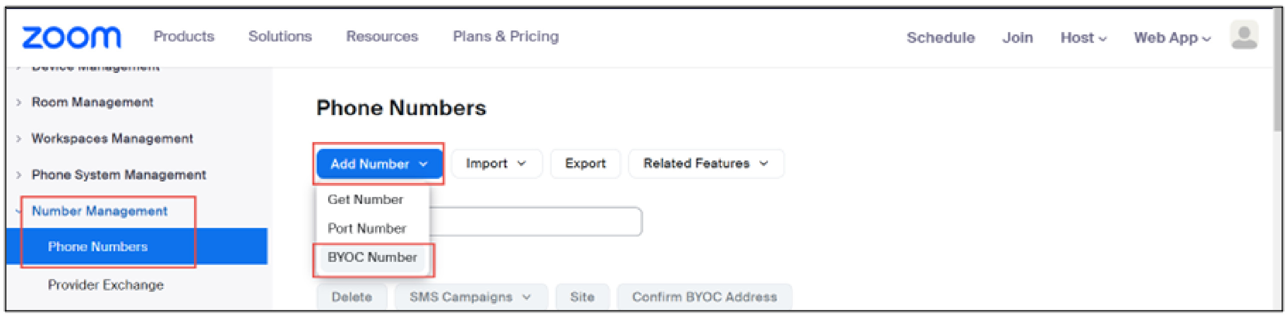
3.3 Adding BYOC Phone numbers

You can upload BYOC phone numbers.

Prerequisite

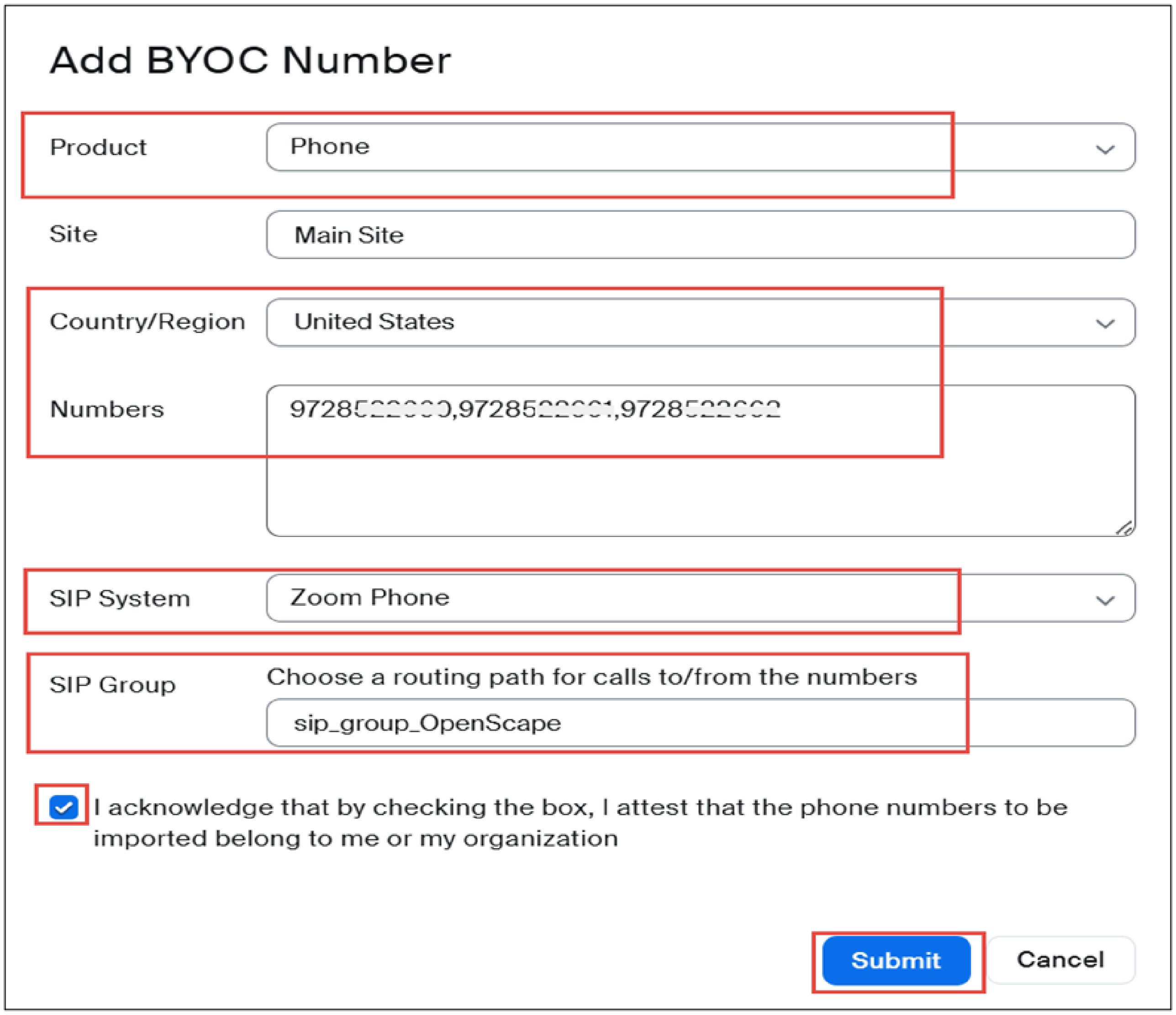
1. You are an administrator with the privilege to edit account settings.

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 created in Configuring the SIP Group on page 14.
   f. Check the acknowledgment box to consent.
   g. Click **Submit**.

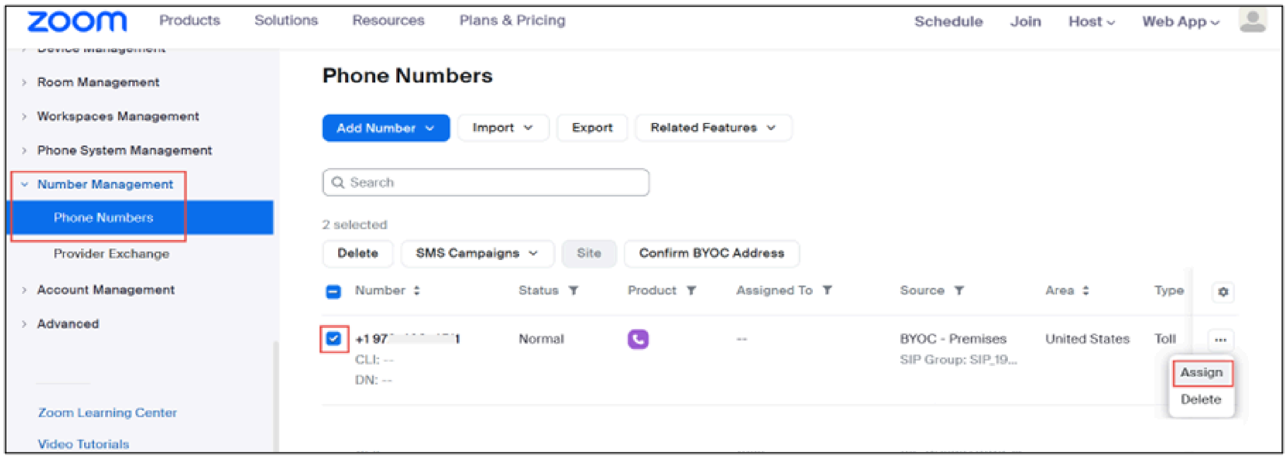


3.3.1 Assigning BYOC numbers

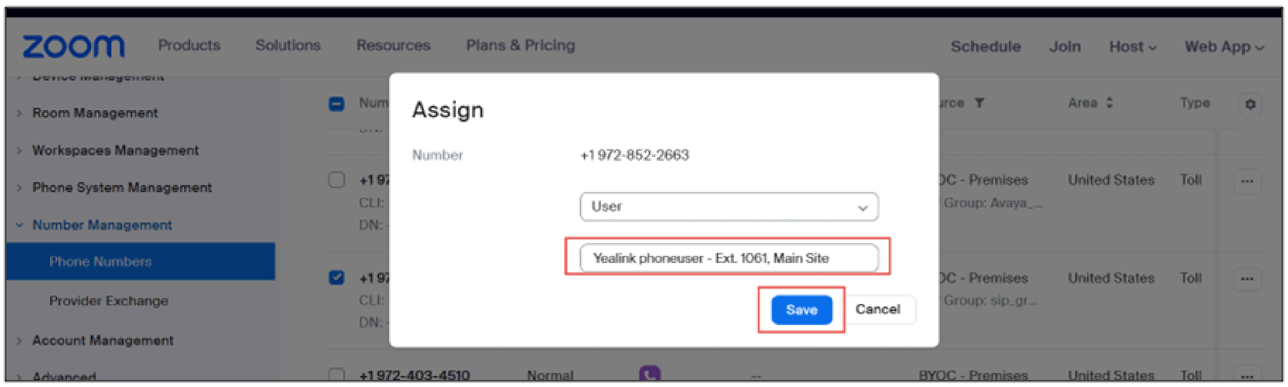
To assign Bring Your Own Carrier (BYOC) numbers to the Zoom phone users:

1. Navigate to **Phone System Management** > **Phone Numbers.**
2. Select the **phone number** that needs to be assigned to the Zoom phone user and click ⋯.

3. Click **Assign**.



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



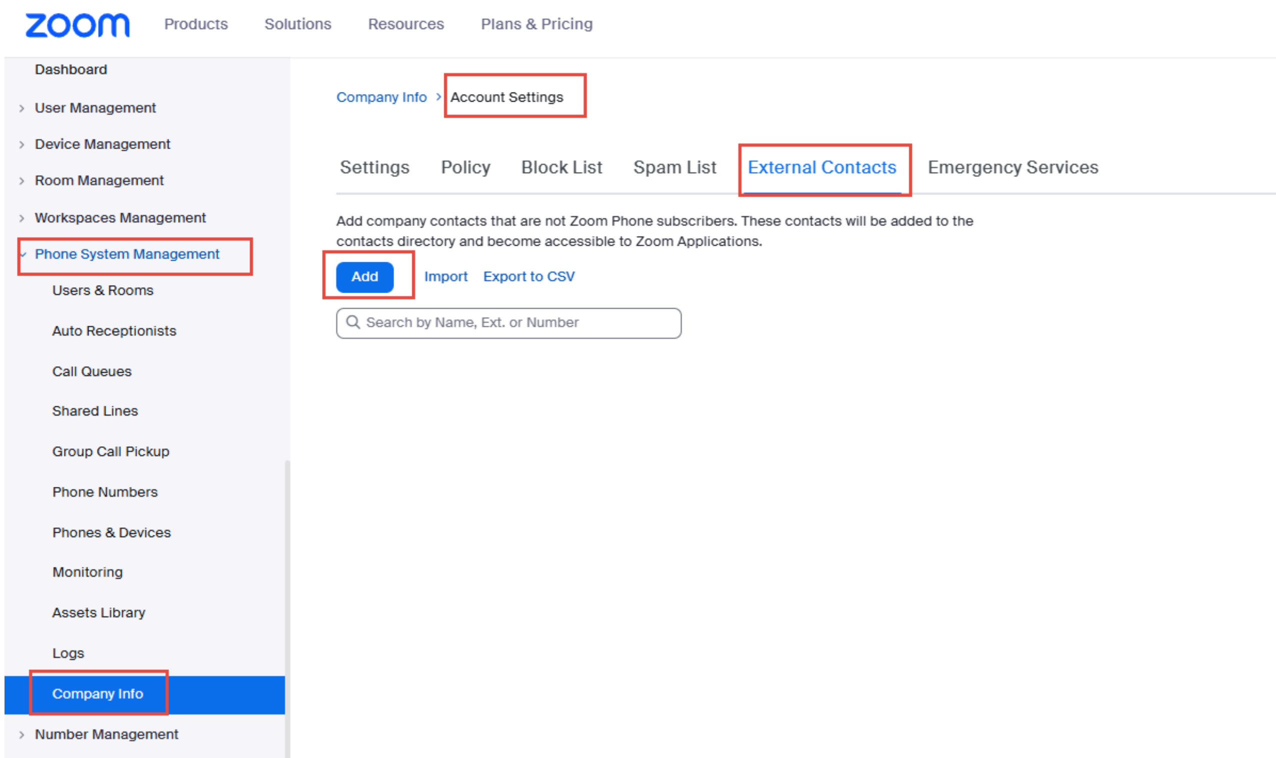
The phone number will be assigned to the selected user.

3.4 Adding BYOP numbers

Administrators can add OpenScape Voice users as external contacts, which will be added to the contacts directory and be accessible to Zoom applications. To add Bring Your Own PBX (BYOP) numbers:

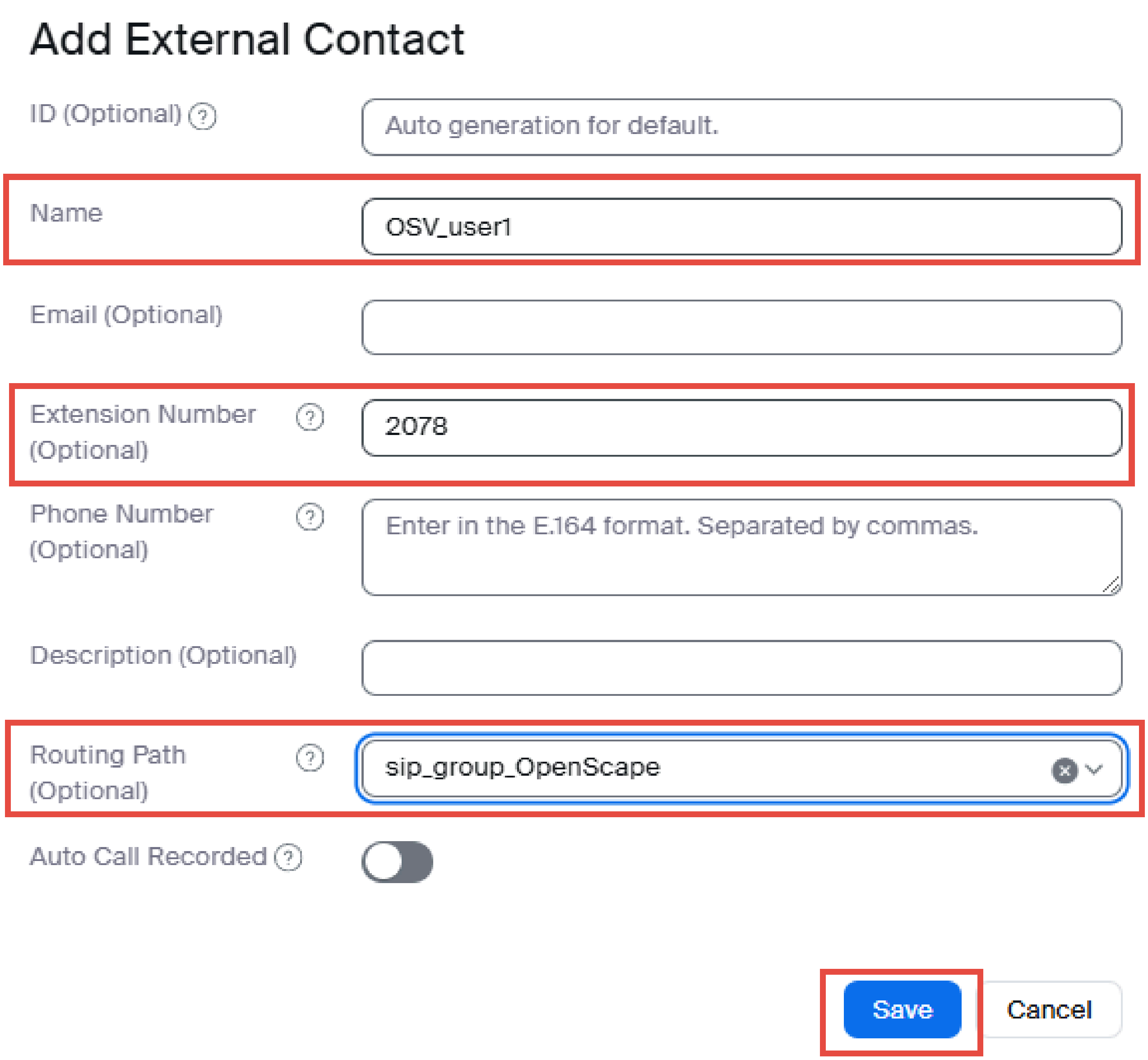
1. Navigate to **Phone System Management** > **Company Info** > **Account Settings** > **External Contacts.**

2. Click **Add.**


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

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

- **Name:** Type the name of the OpenScape Voice user. For example, **OSV_user1**.
- In the **Extension Number** field, enter the extension number of the OpenScape Voice user.
- From the **Routing path** drop-down menu, select the **SIP Group** created in **Configuring the SIP Group on page 14.**

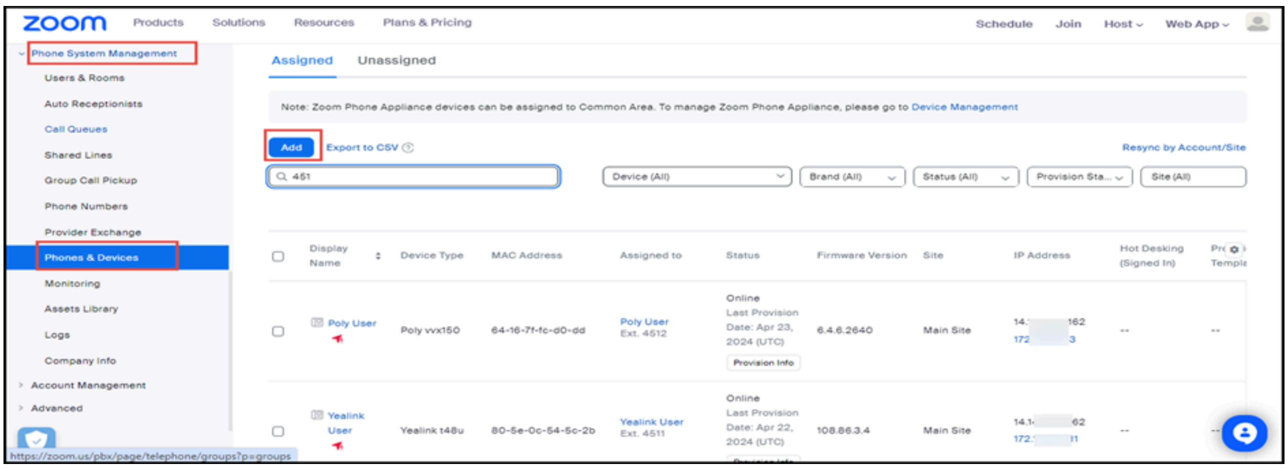


4. Click **Save**.

Provisioning Phones for Zoom Phone Users 4

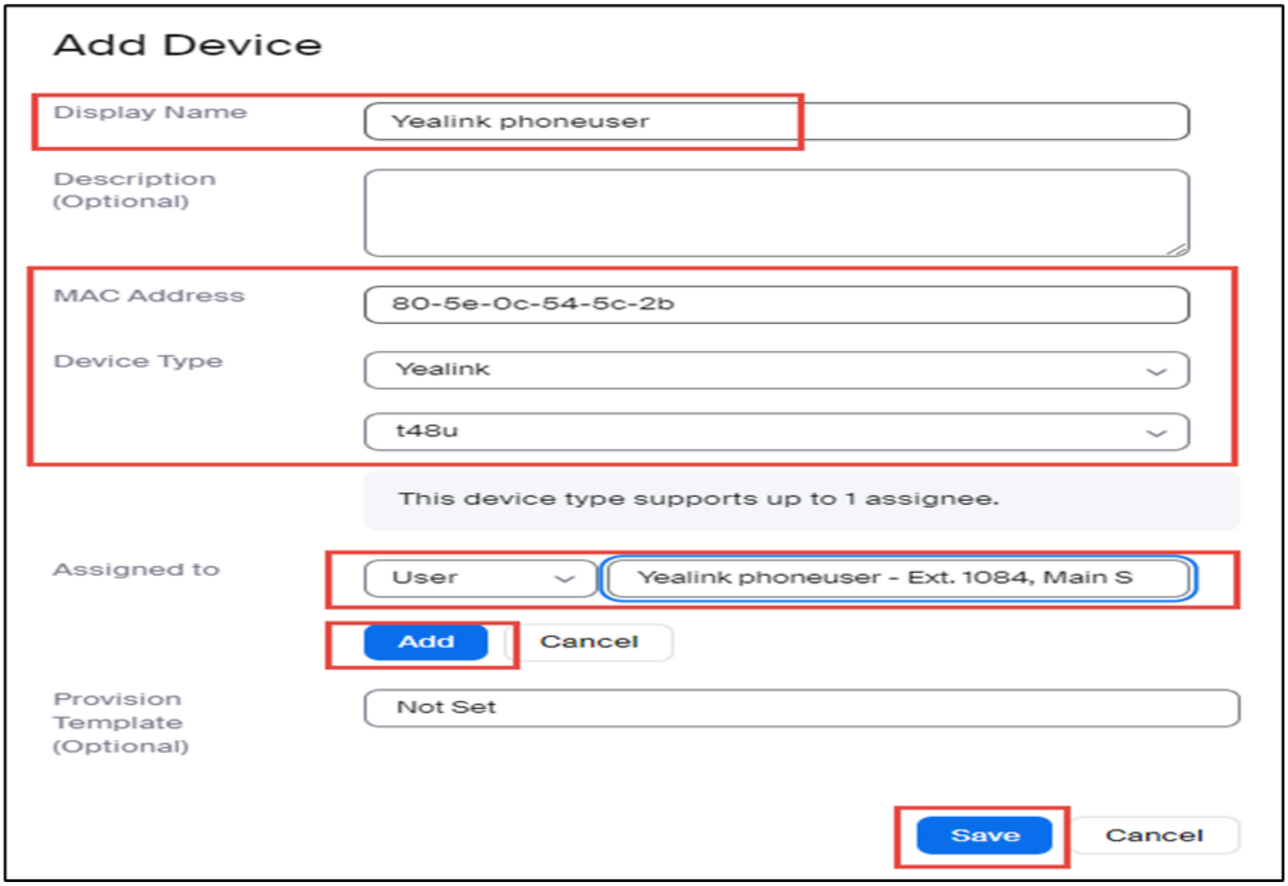
Follow the instructions below to provision Desk phones for Zoom Phone users. Zoom-certified vendor phone models are used for this test and will be available after provisioning.

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



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

   a. **Display Name:** Type the display name for the phone. For example, **Yealink phoneuser**.
   b. **MAC Address:** Enter the **MAC address** of the phone.
   c. **Device Type:** Select the device type. For example, **Yealink**.
   d. From the **Assigned to** drop-down menu, select the user to whom you want to assign the phone number and click **Add**.
   e. From the **Route** drop-down menu**,** select the **Route_group_OpenScape (BYOC)** group, created in Configuring the Route Group on page 11.
   f. Click **Save**.


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

Unify OpenScape Voice Configuration 5

This chapter contains the following sections:

- Configuring Endpoints
- Destinations and Routes Configuration
- Translation Configuration
- Configuring the SIP UA Forking
- Configuring Display Number Modification

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

In Unify OpenScape Voice, you must set up the connection to the OpenScape SBC and the signaling paths to Zoom Phone data centers and the SSP (PSTN provider).

Call routing must also be configured based on the numbering plan for Zoom users and PSTN subscribers.

As an example:

Items	Example
SBC IP	10.8.242.72 TCP 5060
Signaling path to Zoom destination 1 [3]	10.8.242.72 TCP 50001
Signaling path to Zoom destination 2	10.8.242.72 TCP 50002
Signaling path to Zoom destination 3	10.8.242.72 TCP 50003
Signaling path to PSTN provider: BCOM	10.8.242.72 TCP 50010
Zoom user number ranges (reachable from PSTN)	1972598xxxxx

[3] Please refer to the **Signaling Traffic** table under the **Premises Peering Firewall Requirements for Media and Signaling** section in the **Zoom Phone Bring Your Own Carrier- Premises (BYOC-P) Solution Reference Guide.**

5.1 Configuring Endpoints

An **Endpoint** is a network component, such as an originating or terminating device and in our case the OpenScape SBC. An endpoint can be a DN (Directory Number) that does not have a number associated with it yet. An **Endpoint Profile** enables the administrator to set parameters for that endpoint.

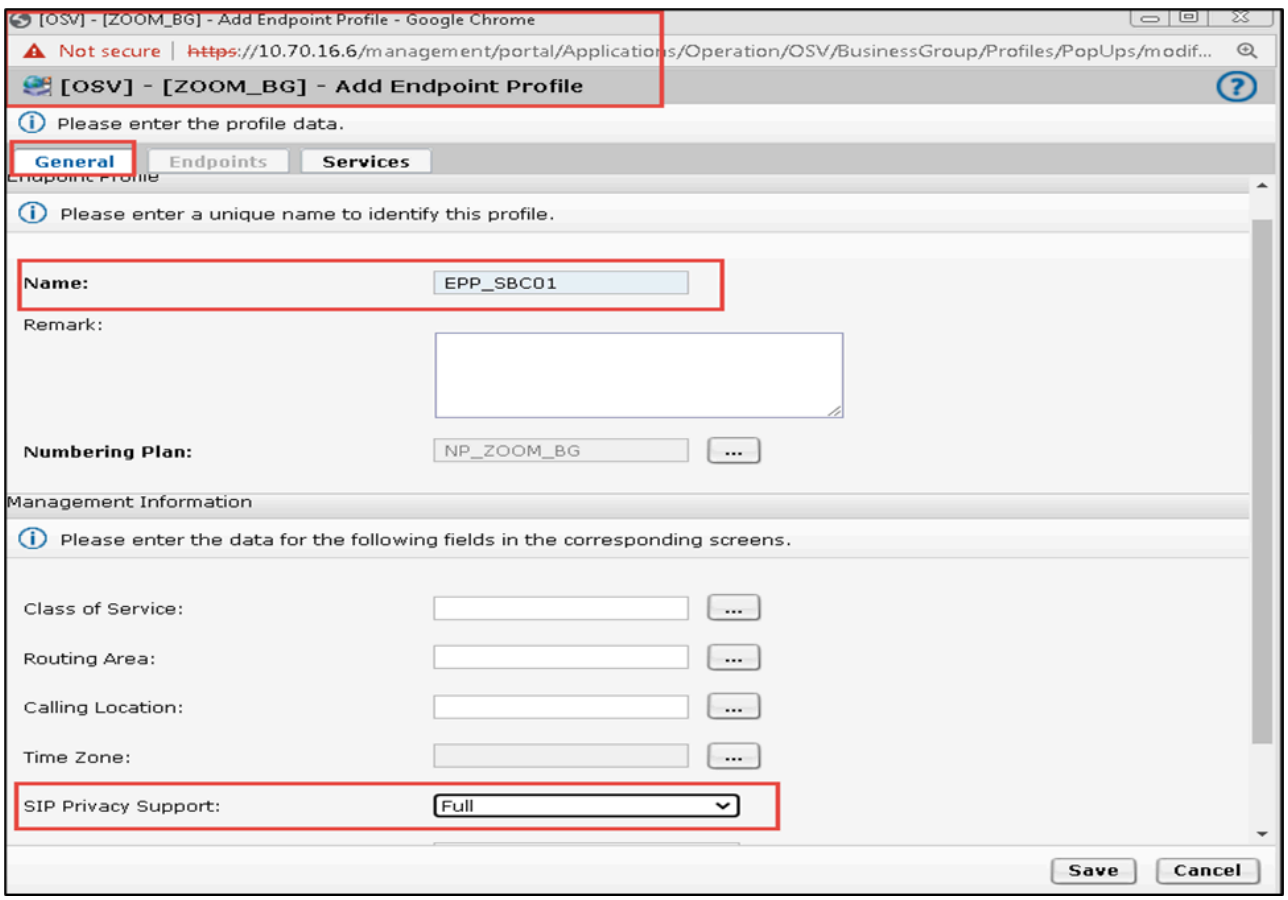
5.1.1 Configuring the OpenScape SBC Endpoint

To configure the OpenScape SBC Endpoint Profile:

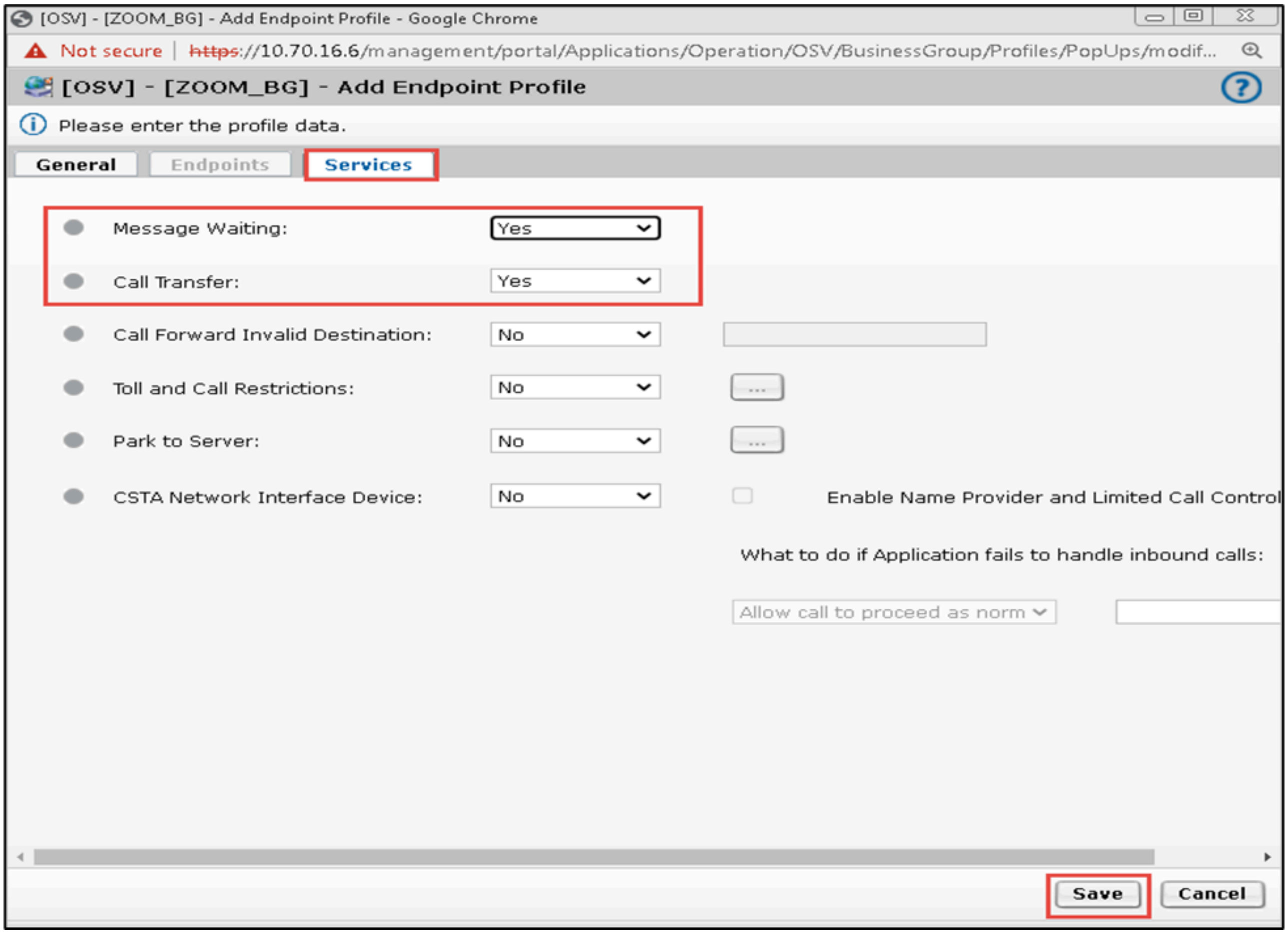
1. Navigate to **Unify OpenScape Common Management Platform** > **Configuration** > **Unify OpenScape Voice** > **Business Group** > **Business Group List.**
2. From the **Business Group List** drop-down menu, select your Business Group. For example, **Zoom_BG**.
3. In the selected Business Group, navigate to **Profiles** > **Endpoint** and click **Add**.
4. In the **Add Endpoint Profile** window, under the **General** tab, configure the following:

    a. **Name**: Enter the name of the endpoint profile. For example, **EPP_SBC01**.

    b. From the **SIP Privacy Support** drop-down menu, select **Full.**



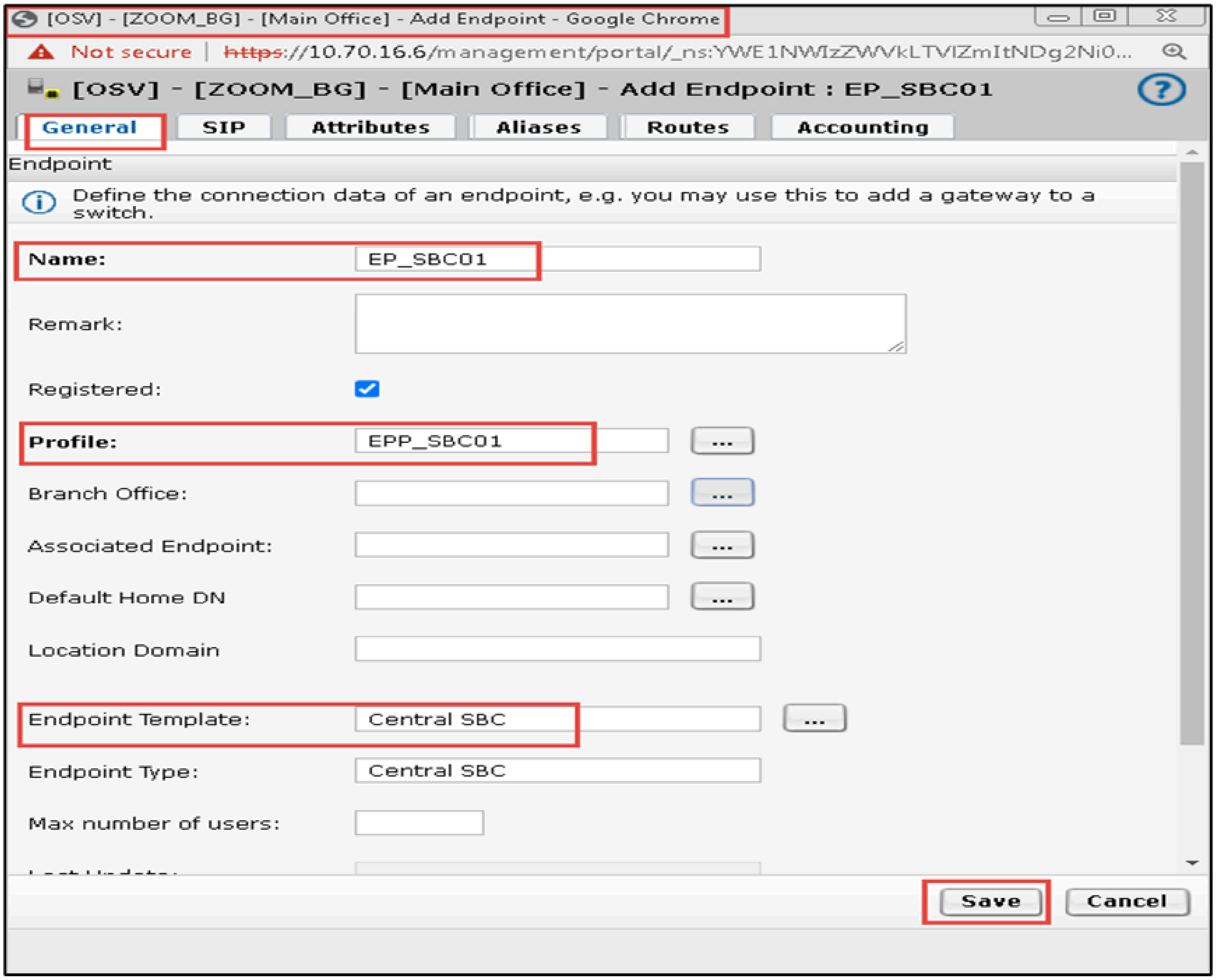
5. In the **Services** tab, enable the following required services by selecting **Yes** from the corresponding drop-down menus:
   - **Message Waiting**
   - **Call Transfer**



6. Click **Save**.
7. Navigate to **Unify OpenScape Common Management Platform** > **Configuration** > **Unify OpenScape Voice** > **Business Group** > **Members** > **Endpoints** to configure the **Endpoint**.
8. Click **Add**.

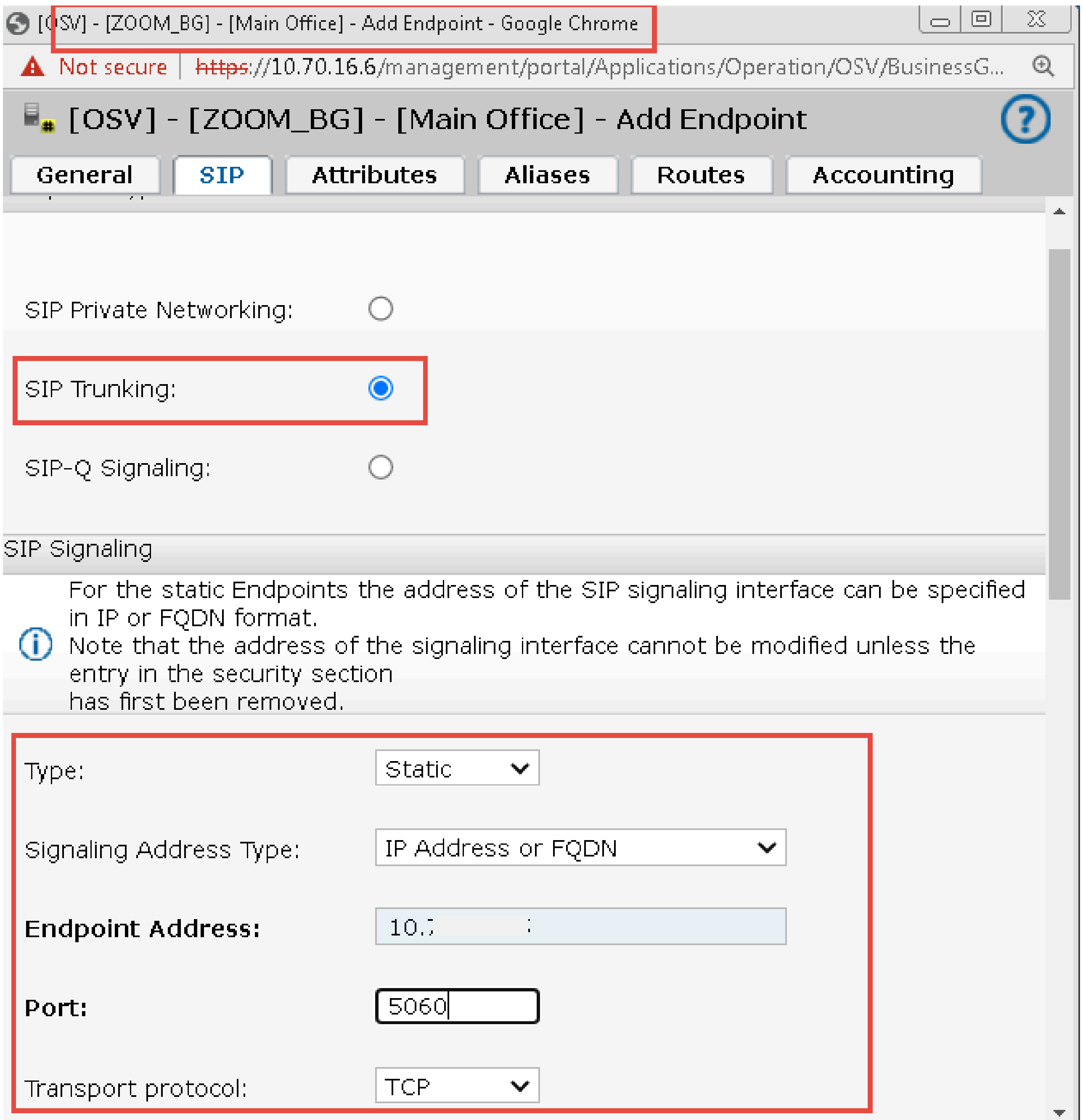
9. In the **General** tab, configure the following:

   a. **Name**: Enter the name of the SBC endpoint. For example, **EP_SBC01**
   b. **Profile**: Select the previously created endpoint profile. For example, **EPP_SBC01**.
   c. **Endpoint Template**: Select **Central SBC** (set of pre-configured endpoint attributes).
   d. Click **Save**.

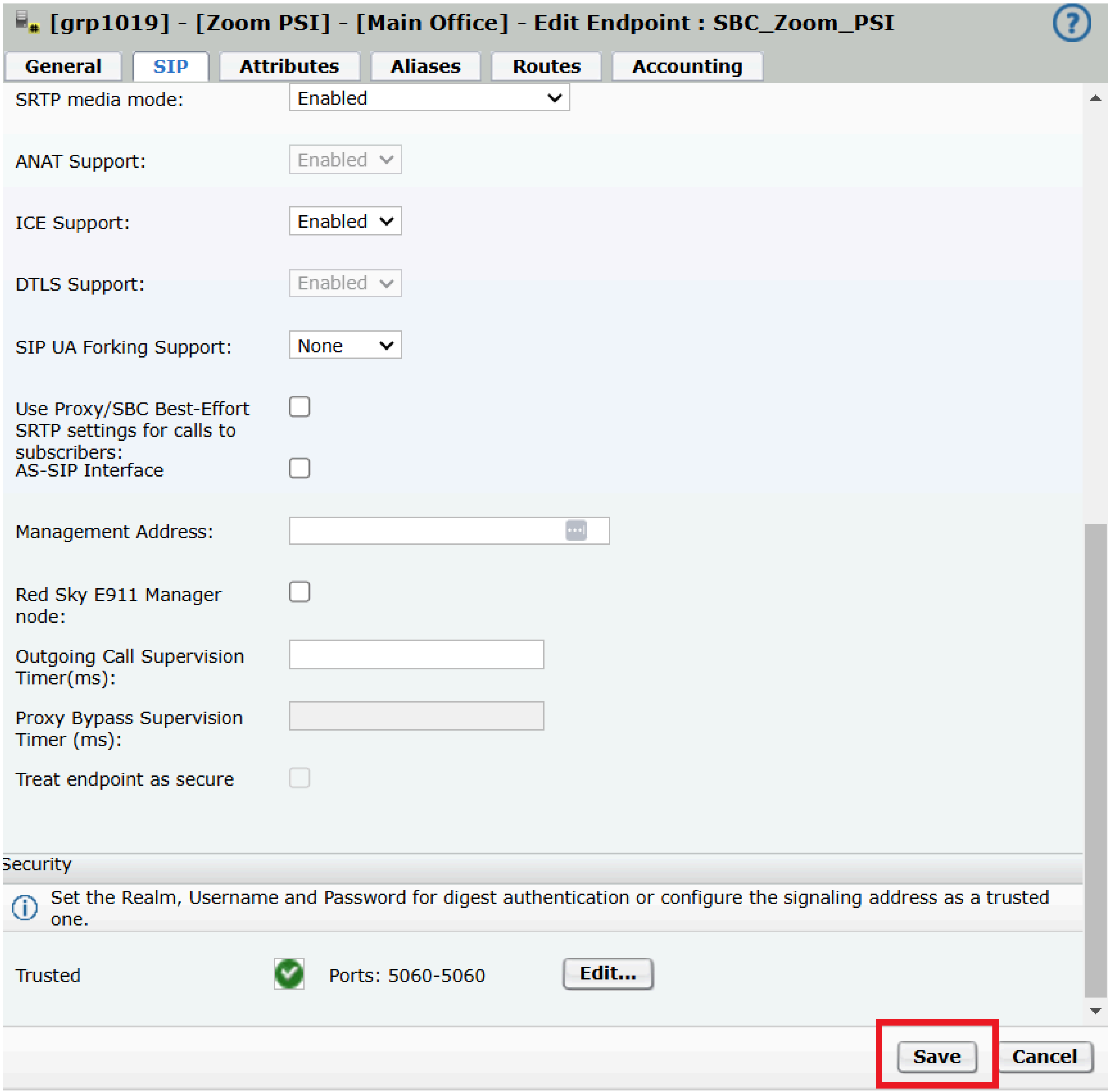


10. Select the **SIP** tab and configure the following:

    a. Select the **SIP Trunking** option to enable it.
    b. From the **Type** drop-down menu, select **Static** (it can be enabled only if the **SIP Proxy** attribute is enabled).
    c. From the **Signaling Address Type** drop-down menu**, select IP Address or FQDN** (route the calls via proxy).
    d. **Endpoint Address:** Enter the SBC address.
    e. **Port**: Enter the port number.
    f. From the **Transport protocol** drop-down menu, select **TCP.**

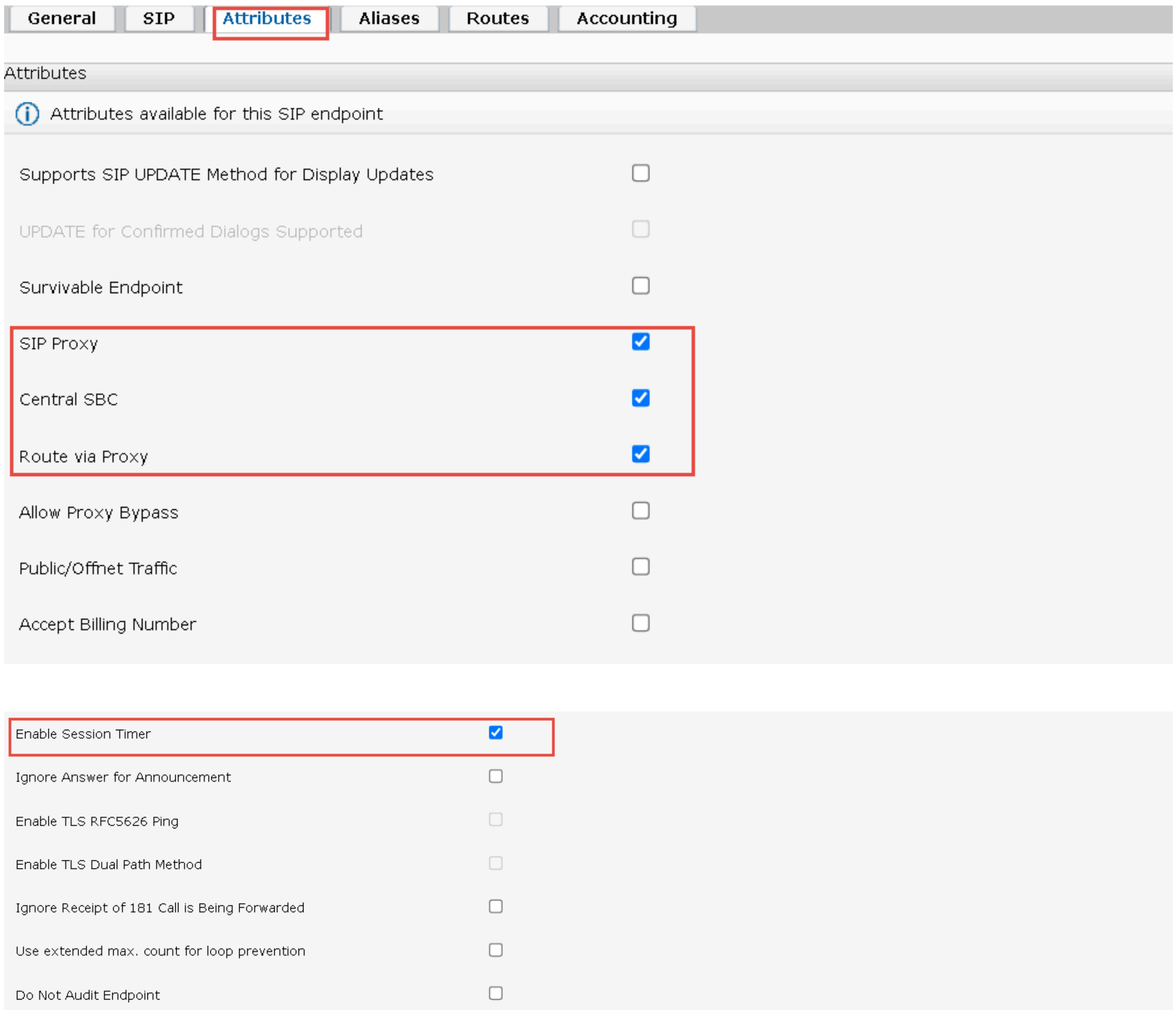


11. Locate the **Security** section, click **Edit**, and add the primary SIP port (5060) of the SBC. Click **Save**.

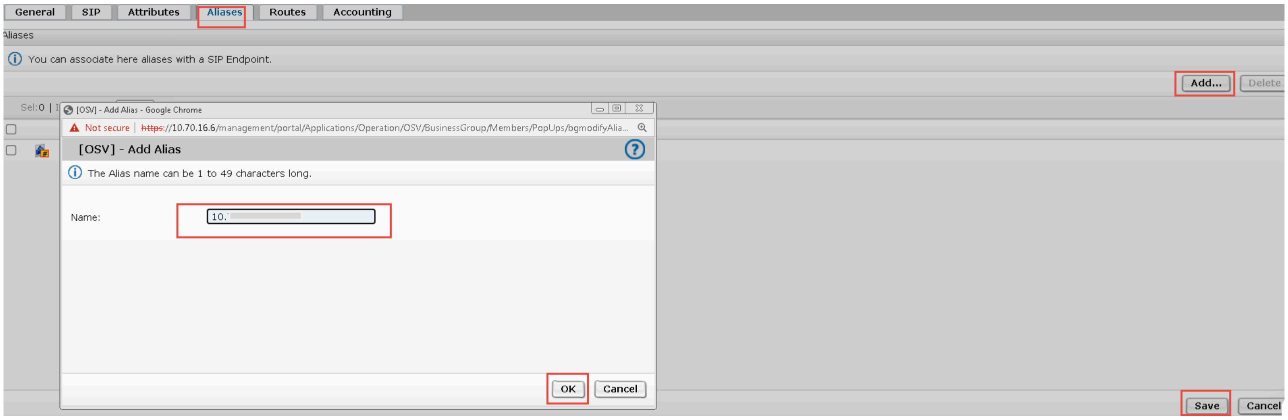


12. The **Attributes** tab is populated automatically since the "Central SBC" template was selected in the **General** tab. Ensure that the following are selected:

- **SIP Proxy**
- **Central SBC**
- **Route via Proxy**
- **Enable Session Timer**



13. Select the **Aliases** tab and click **Add** to enter the SBC LAN interface for incoming SIP traffic.



14. Click **OK** and then click **Save**.

5.1.2 Configuring the Zoom Phone Endpoint

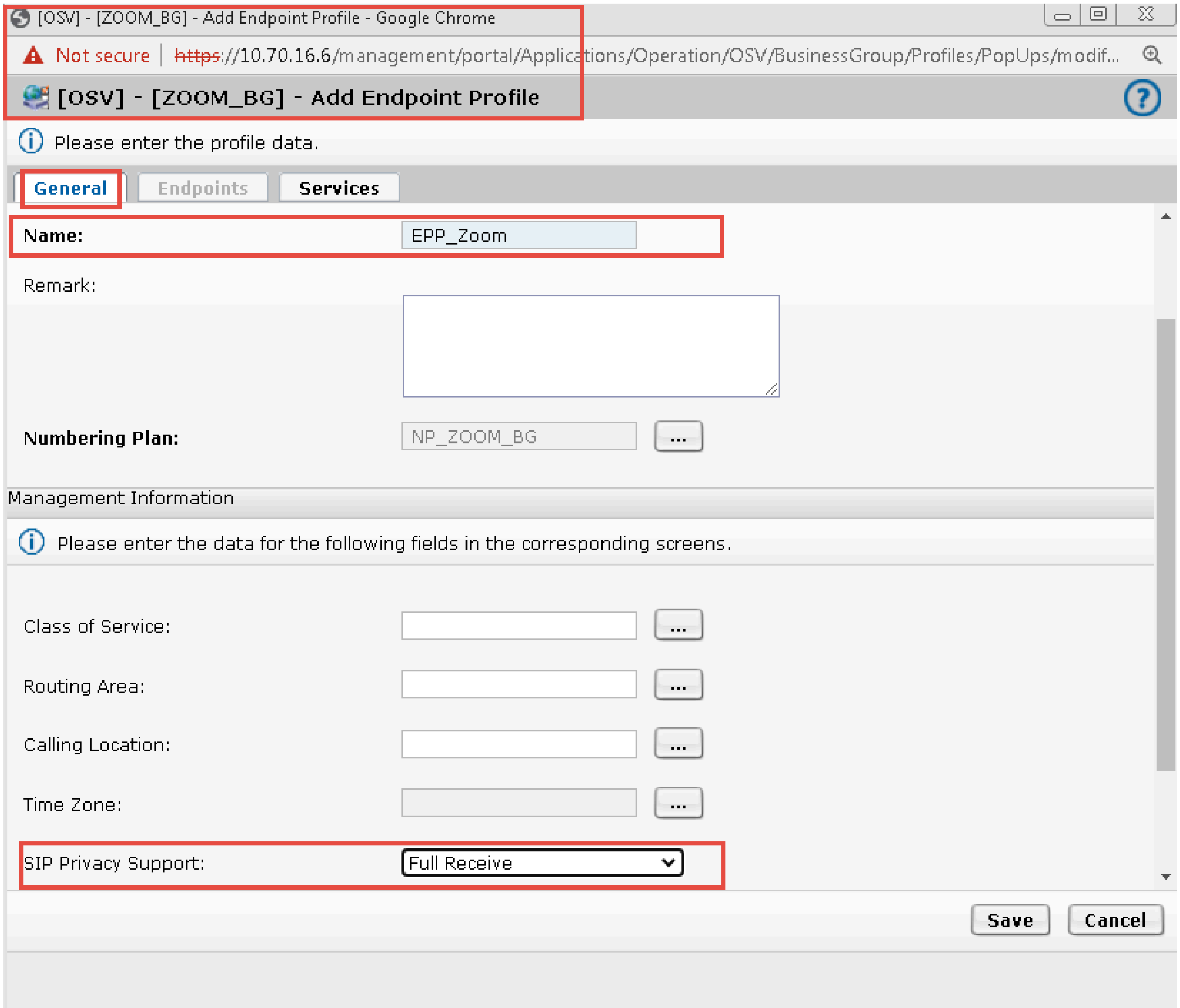
Prerequisite

To configure **SIP UA Forking Support** for the Zoom Phone Endpoint, you must enable the SIP UA Forking Support option. To do this, follow the instructions in Configuring the SIP UA Forking on page 64.

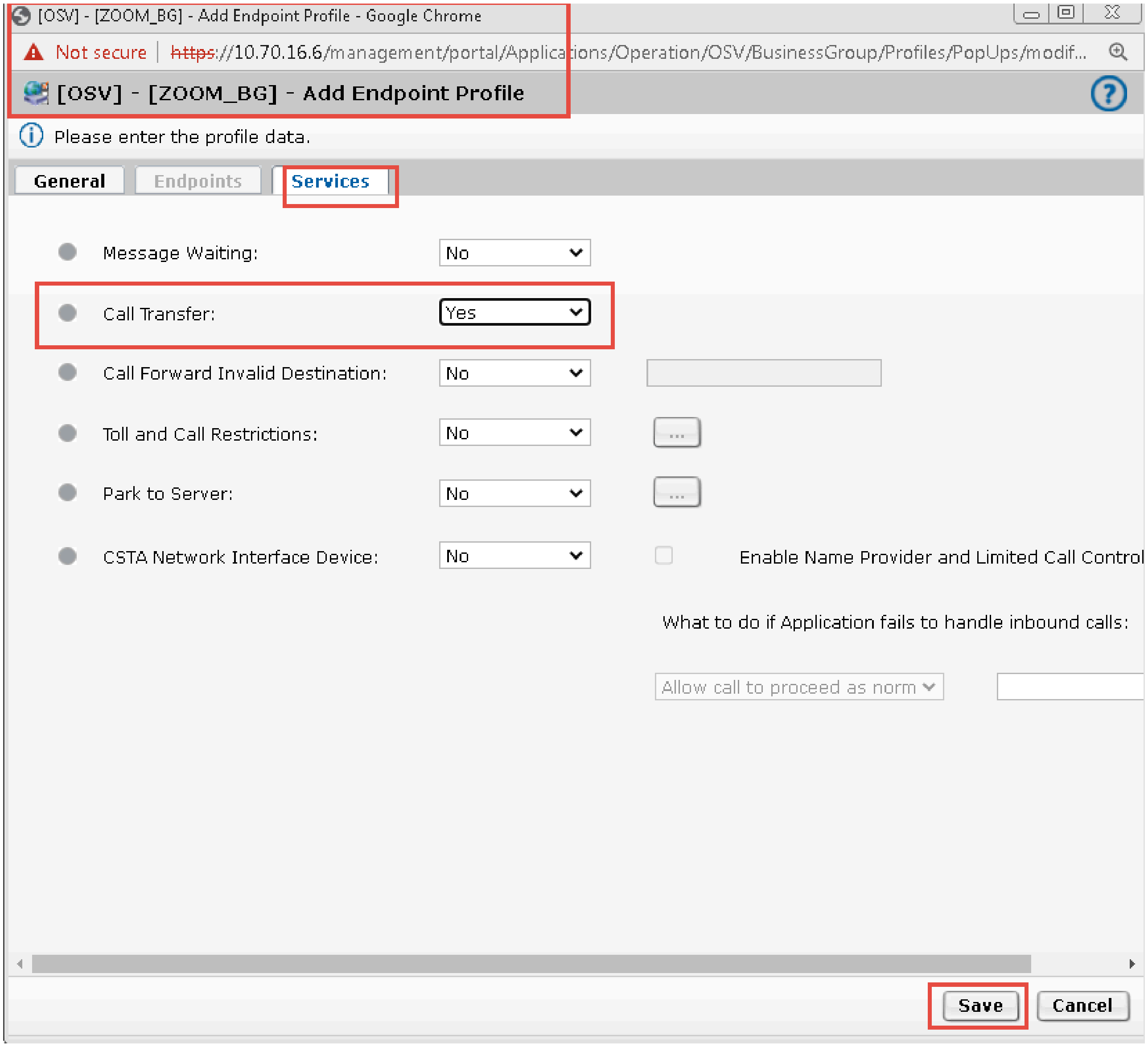
1. Navigate to **Unify OpenScape Common Management Platform** > **Configuration** > **Unify OpenScape Voice** > **Business Group** > **Profiles** > **Endpoint** to configure the Zoom Endpoint Profile.
2. Click **Add**.

3. In the **Add Endpoint Profile** window, under the **General** tab, configure the following:

- **Name**: Enter the name of the endpoint profile. For example, **EPP_Zoom**.
- From the **SIP Privacy Support** drop-down menu, select **Full Receive.**



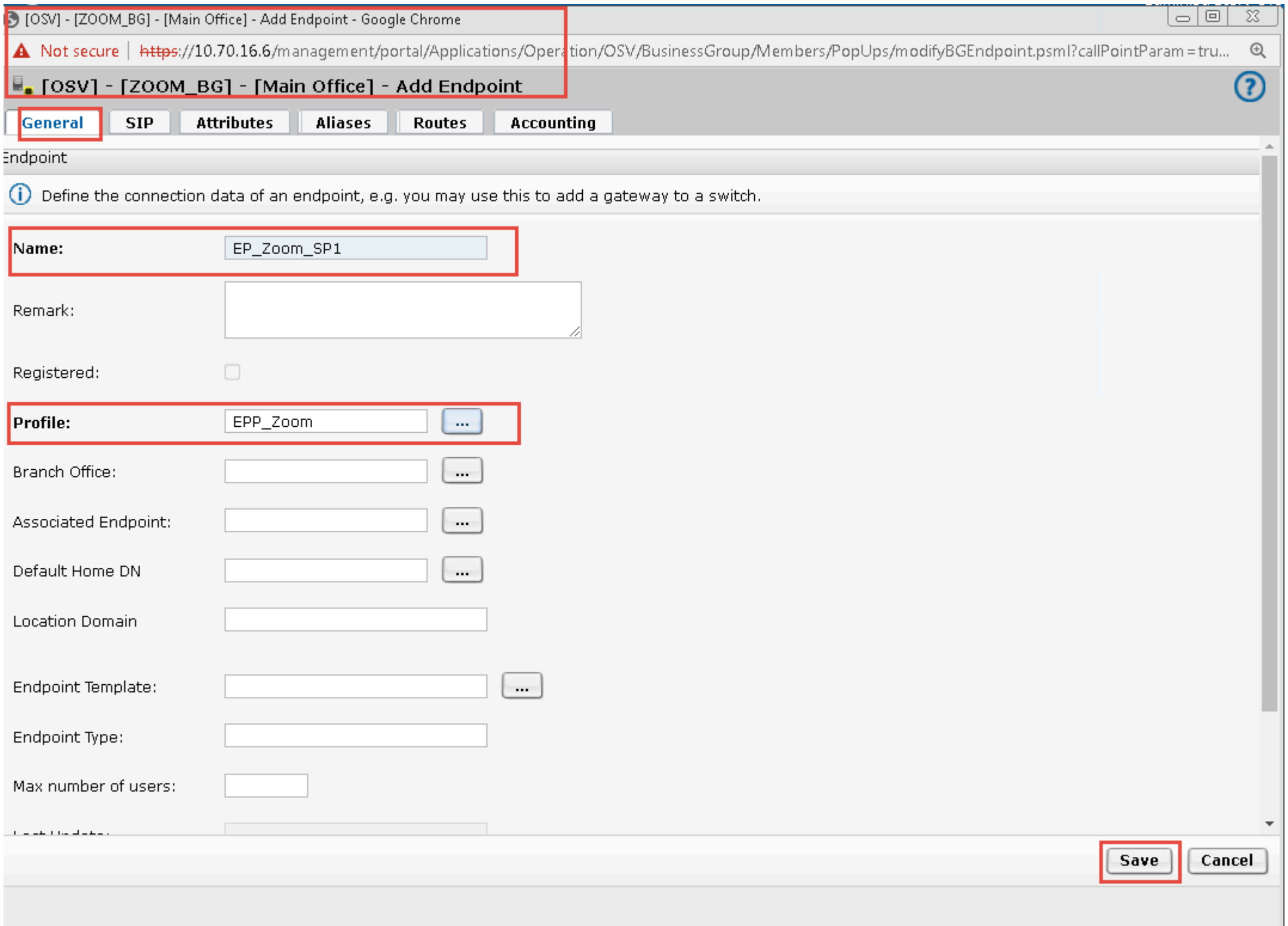
4. In the **Services** tab, enable the **Call Transfer** option, by selecting **Yes** from the drop-down menu.



5. Click **Save.**
6. In the Unify OpenScape Common Management Platform, navigate to **Configuration > OpenScape Voice > BusinessGroup > Members > Endpoints** and click **Add.**

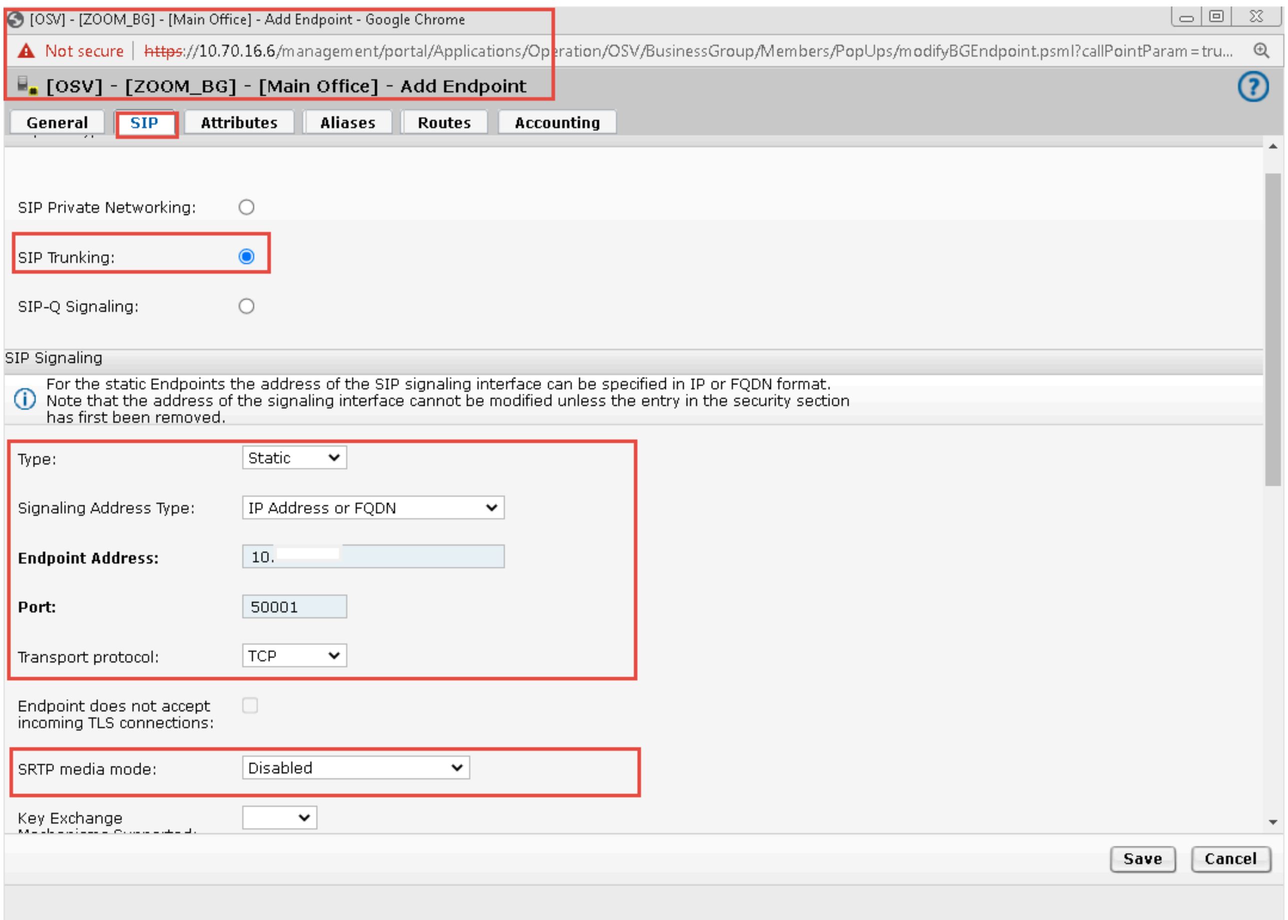
7. In the **Add Endpoint** pop-up, under the **General** tab, configure the following:

   a. **Name**: Enter the name of the Zoom endpoint. For example, **EP_Zoom_SP1**.
   b. **Profile**: Select the previously created Zoom endpoint profile. For example, **EPP_Zoom**.

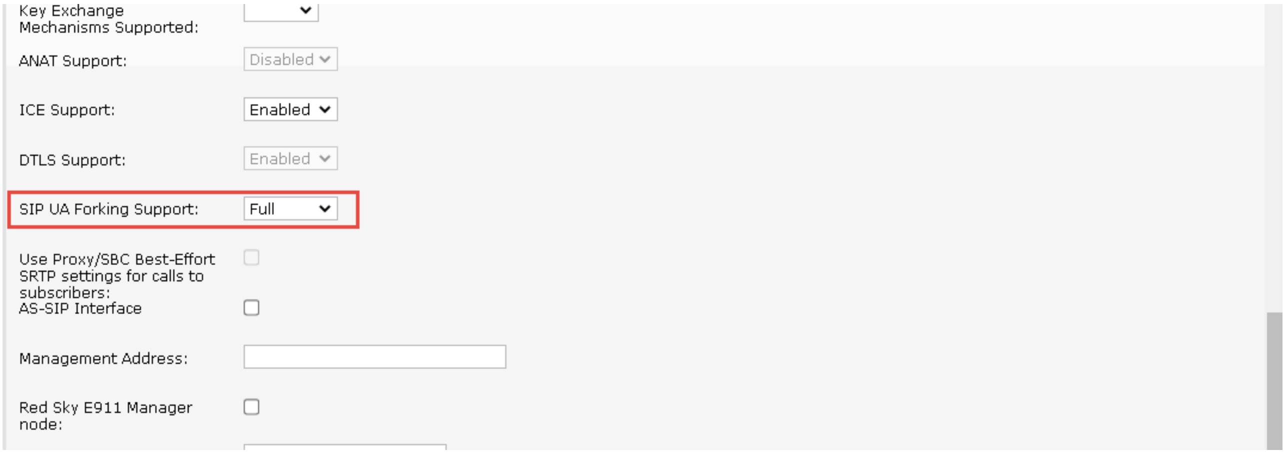


8. Select the **SIP** tab and configure the following:

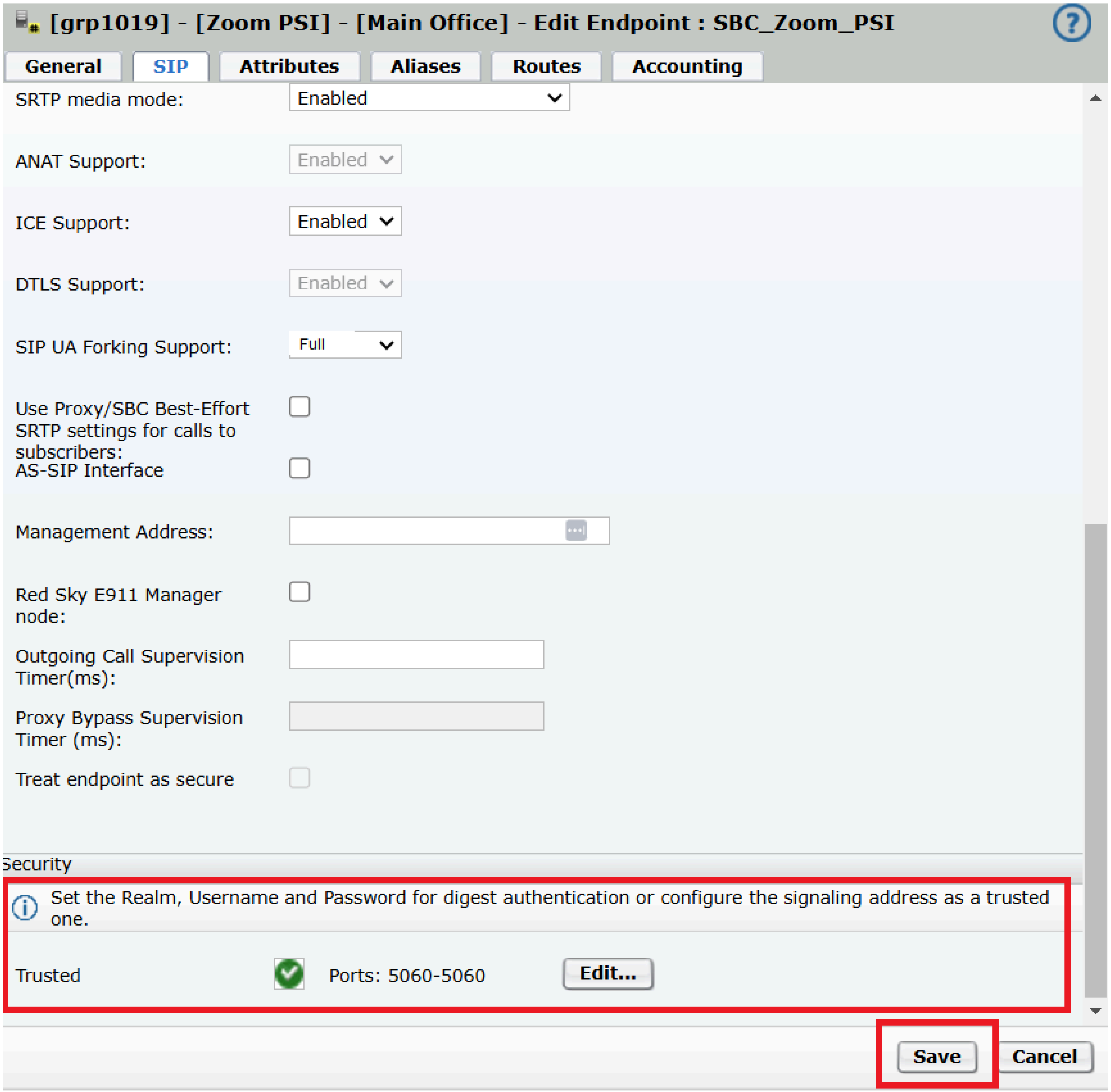
   a. Select the **SIP Trunking** option to enable it.
   b. From the **Type** drop-down menu, select **Static** (it can be enabled only if the **SIP Proxy** attribute is enabled).
   c. From the **Signaling Address Type** drop-down menu**, select IP Address or FQDN** (route the calls via proxy).
   d. **Endpoint Address:** Enter the SBC address.
   e. **Port**: Enter the port number for Zoom trunk.
   f. From the **Transport protocol** drop-down menu, select **TCP**.
   g. From the **SRTP media mode** drop-down menu, select **Disabled.**



   h. From the **SIP UA Forking Support** drop-down menu, select **Full**:

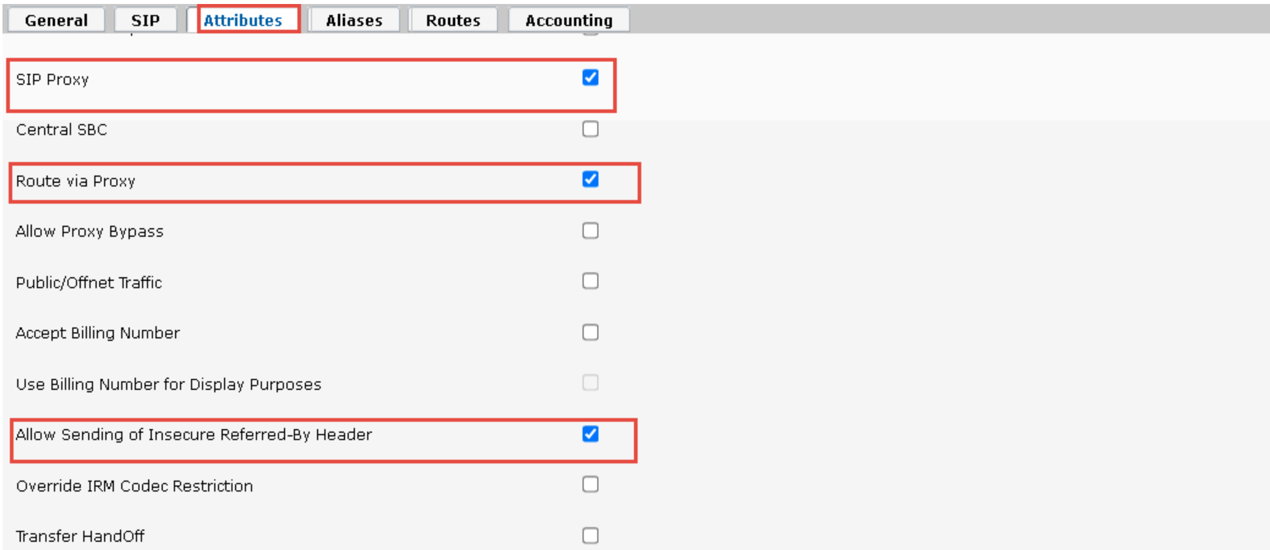

Key Exchange Mechanisms Supported:
ANAT Support:
Disabled
ICE Support:
Enabled
DTLS Support:
Enabled
SIP UA Forking Support:
Full
Use Proxy/SBC Best-Effort SRTP settings for calls to subscribers:
AS-SIP Interface
Management Address:
Red Sky E911 Manager node:

9. Locate the **Security** section, click **Edit,** and add the primary SIP port (5060) of the SBC. Click **Save**.



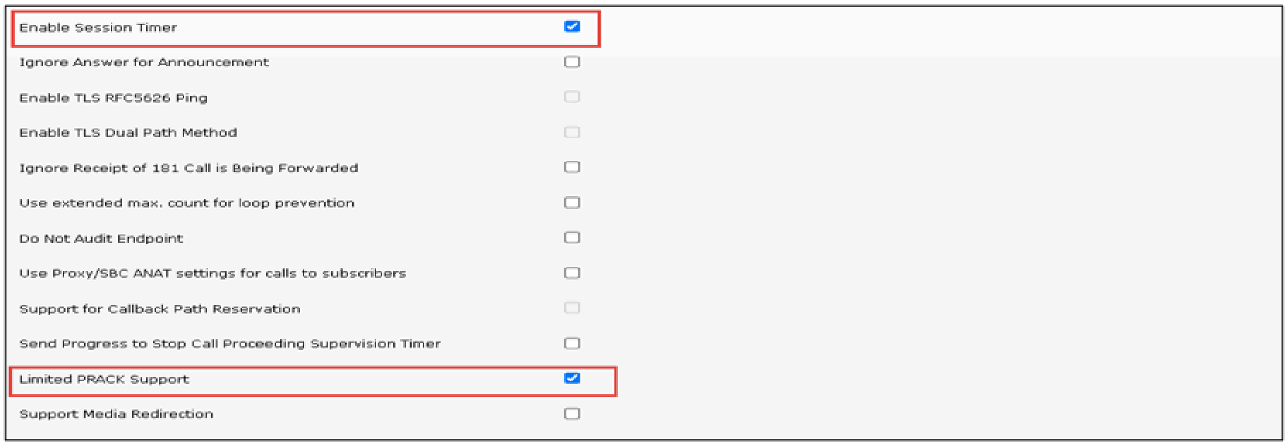
10. In the **Attributes** tab, select the following parameters to activate them:

- **SIP Proxy**
- **Route via Proxy**
- **Allow sending of Insecure Referred-By Header**



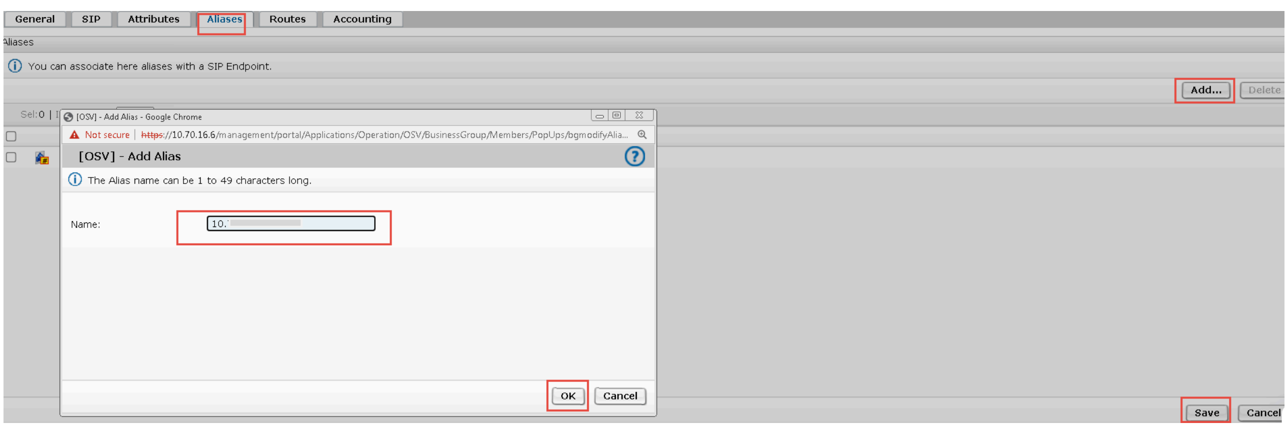
- **Do not send Invite without SDP**
- **Send International Numbers in Global number format (GNF)**

- **Enable session timer**
- **Limited PRACK support**



- **Support Replaces Header**
- **Ignore Receipt/Do not send privacy header**
- **Enable REFER Notifications**

11. Select the **Aliases** tab, click **Add** and enter the **SBC LAN interface** with port number for incoming SIP traffic.



12. Click **OK** and then click **Save**.
13. Repeat the same procedure to create the endpoints for the remaining Zoom IPs, assigning the respective port numbers:

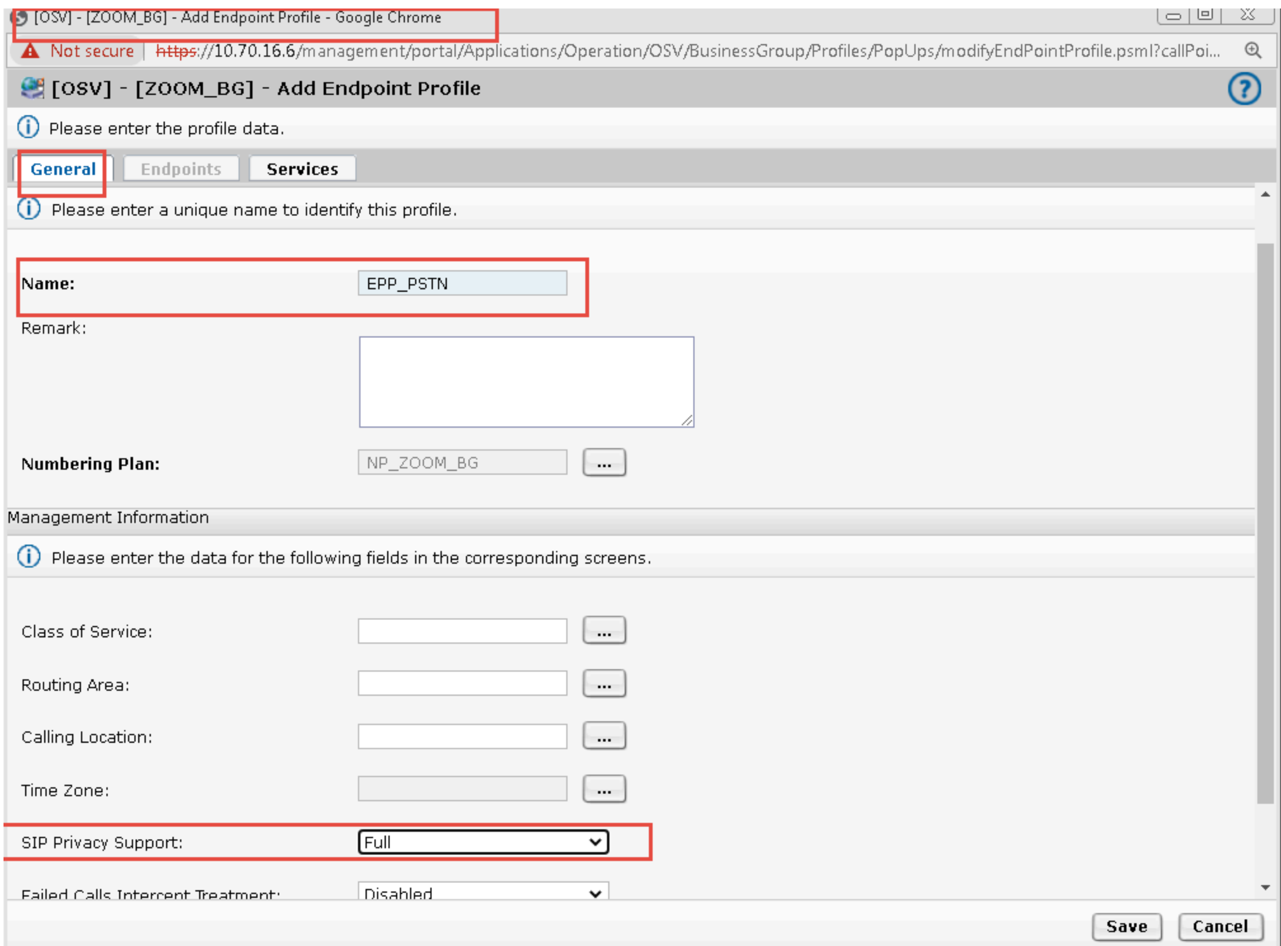
- EP_Zoom_SP2 with port 50002
- EP_Zoom_SP3 with port 50003

5.1.3 Configuring the PSTN Endpoint

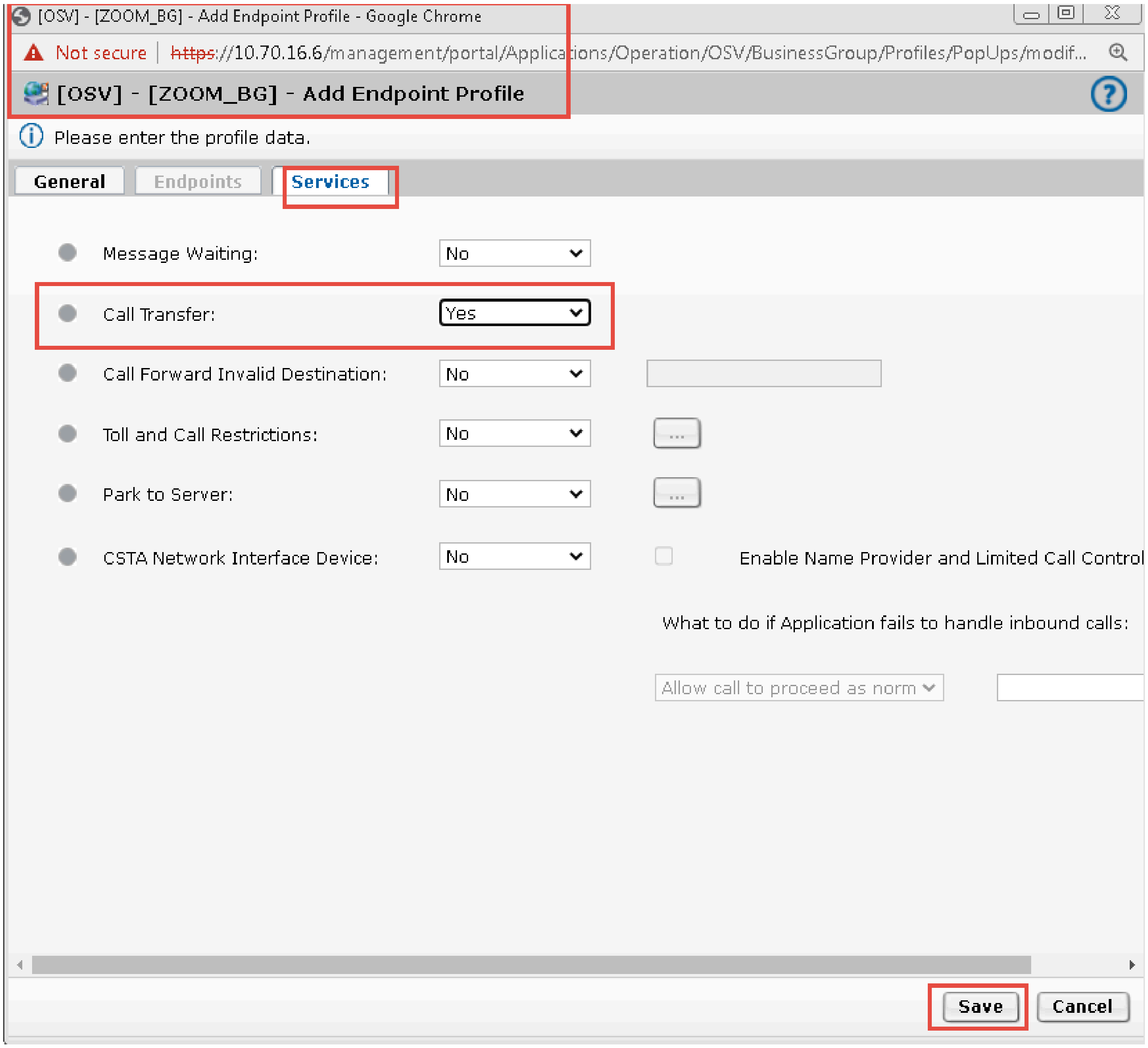
Note:
The configuration below is an example. The actual configuration steps depend on your provider's requirements.

1. Navigate **to Unify OpenScape Common Management Platform** > **Configuration** > **Unify OpenScape Voice > Business Group** > **Profiles** > **Endpoint** to configure the PSTN Endpoint Profile.
2. Click **Add.**
3. In the **Add Endpoint Profile** pop-up, under the **General** tab, configure the following:

   a. **Name**: Enter the name of the endpoint profile. For example, **EPP_PSTN.**
   b. From the **SIP Privacy Support** drop-down menu, select **Full.**



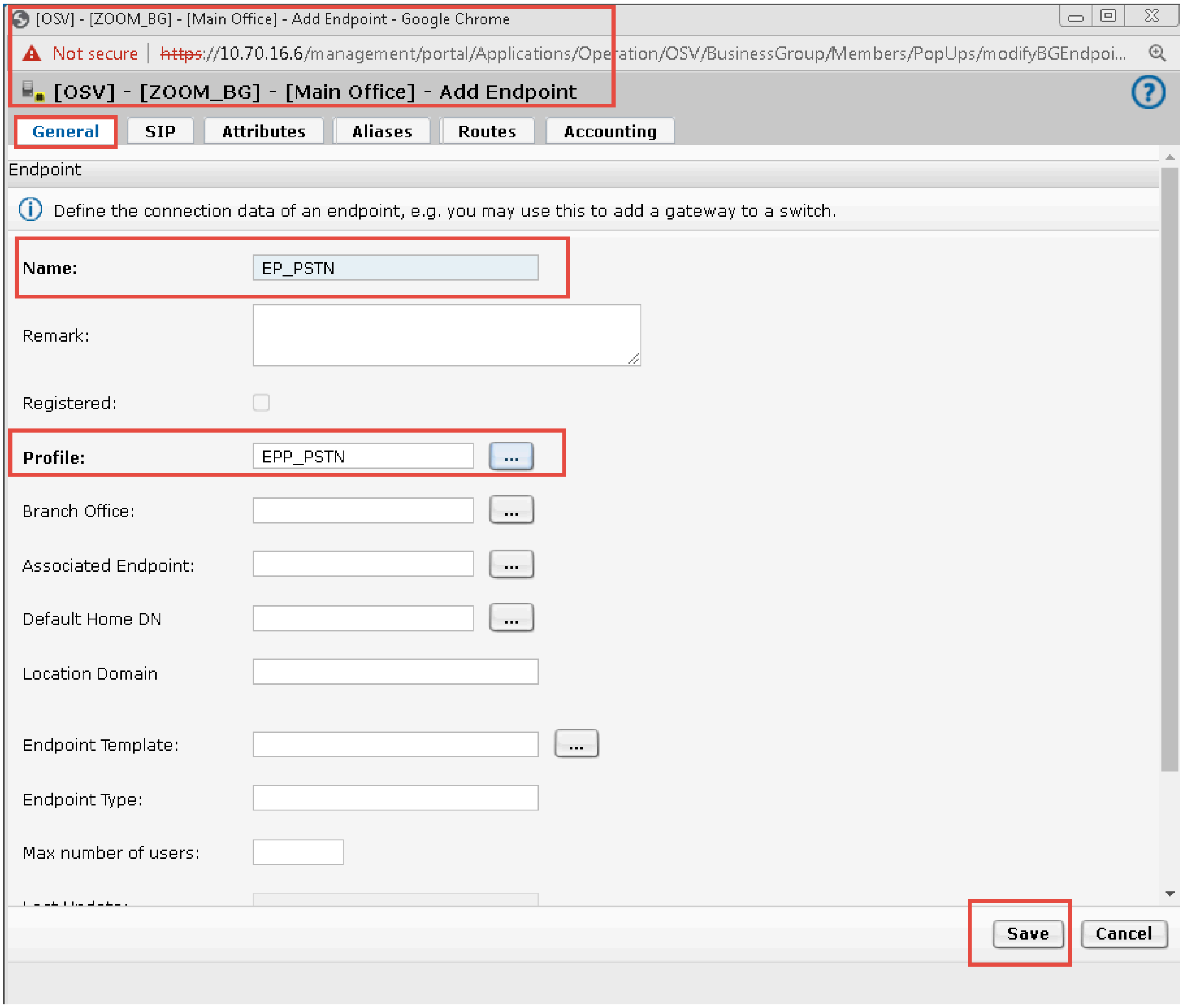
4. In the **Services** tab, enable the **Call Transfer**, by selecting **Yes** from the drop-down menu.



5. Click **Save.**
6. To add Endpoints: In the Unify OpenScape Common Management Platform, navigate to **Configuration > OpenScape Voice > Business Group > Members > Endpoints**
7. Click **Add.**

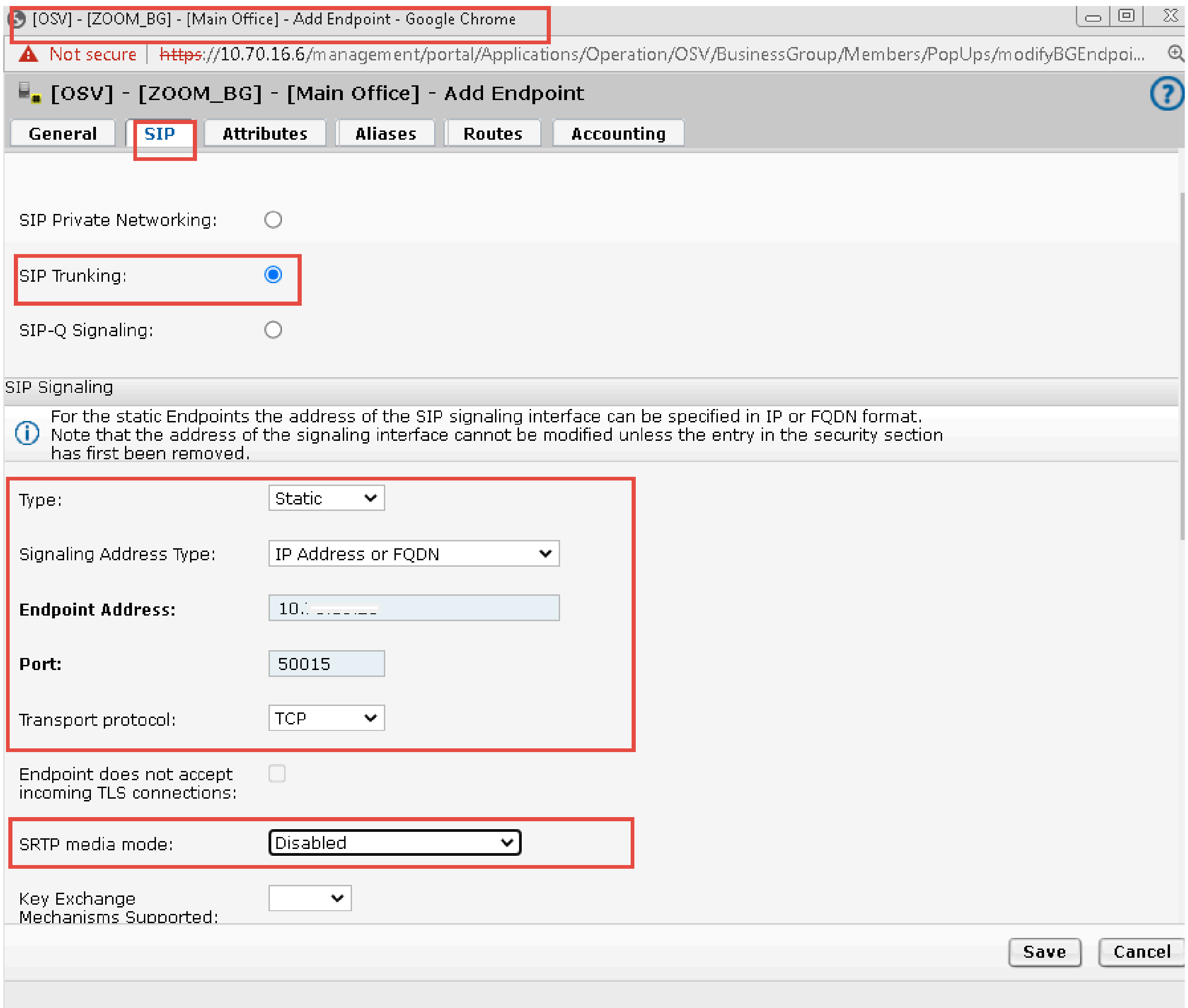
8. In the **Add Endpoint** pop-up, under the **General** tab, configure the following:

   a. **Name**: Enter the name of the PSTN endpoint. For example, **EP_PSTN**.
   b. **Profile**: Select the previously created PSTN endpoint profile. For example, **EPP_PSTN.**



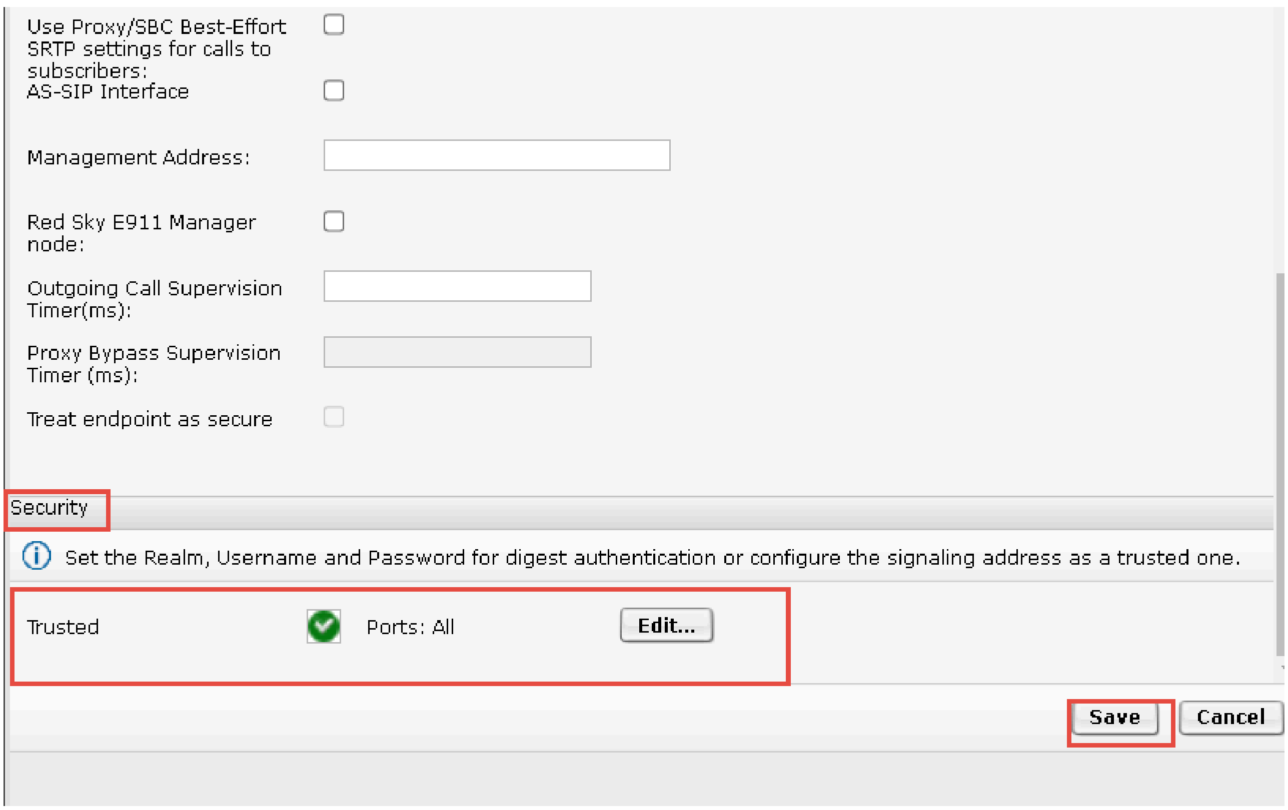
9. Select the **SIP** tab and configure the following:

   a. Select the **SIP Trunking** option to enable it.
   b. From the **Type** drop-down menu, select **Static** (it can be enabled only if the **SIP Proxy** attribute is enabled).
   c. From the **Signaling Address Type** drop-down menu**, select IP Address or FQDN** (route the calls via proxy).
   d. **Endpoint Address:** Enter the SBC address.
   e. **Port**: Enter the port number for PSTN trunk.
   f. From the **Transport protocol** drop-down menu, select **TCP**.
   g. From the **SRTP media mode** drop-down menu, select **Disabled.**



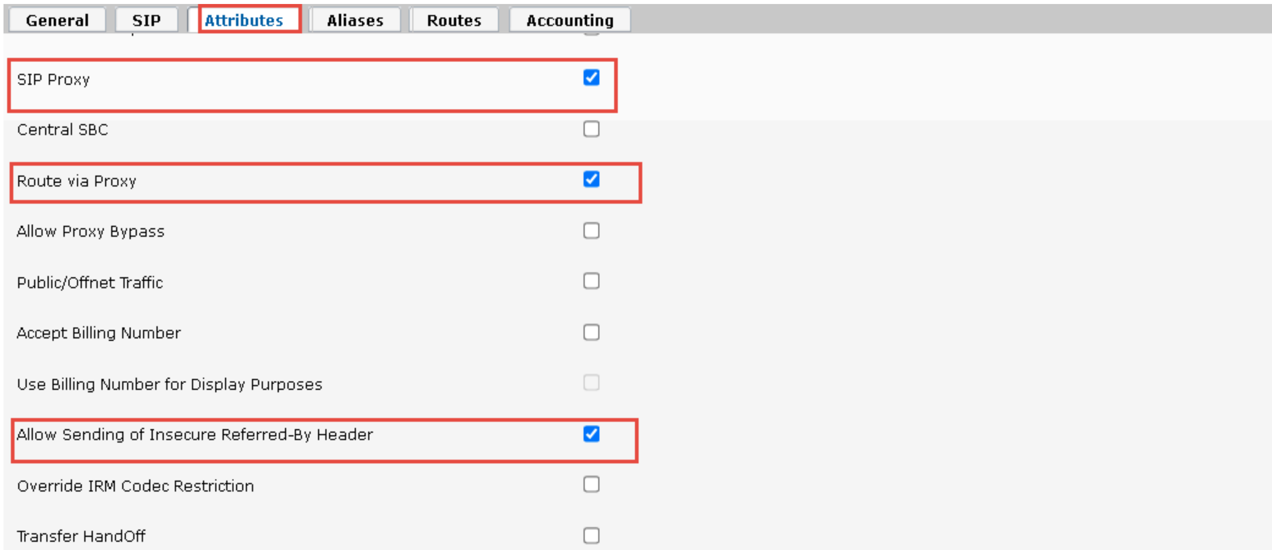
   h. From the **SIP UA Forking Support** drop-down menu, select **None**.

10. Locate the **Security** section, click **Edit,** and add all the ports. Click **Save**.

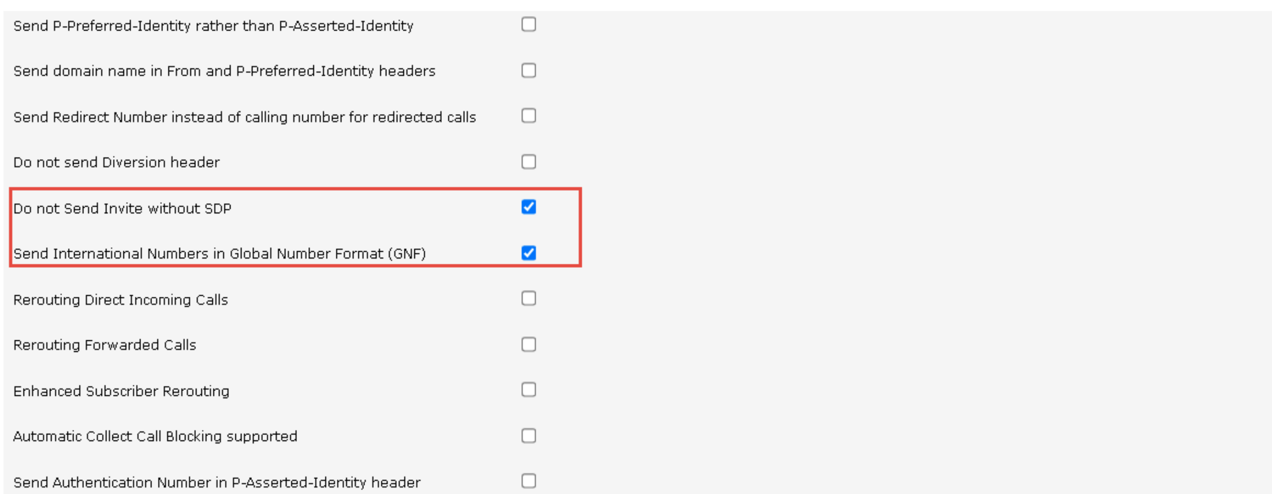


11. In the **Attributes** tab, select the following parameters to activate them:

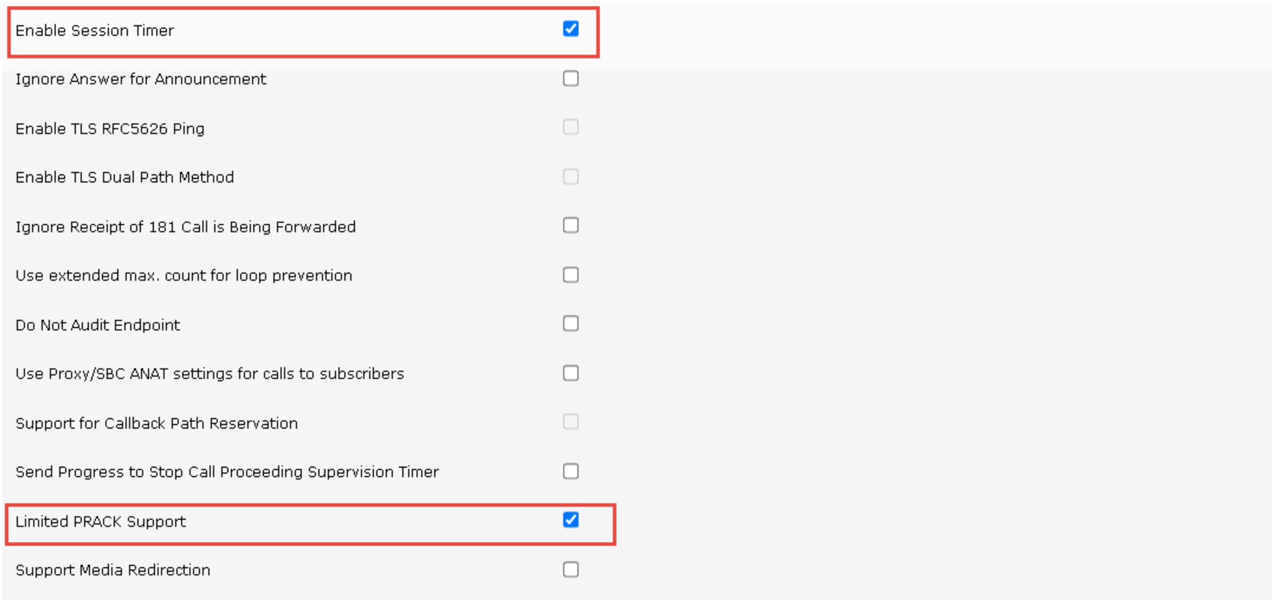
- **SIP Proxy**
- **Route via Proxy**
- **Allow sending of Insecure Referred-By Header**



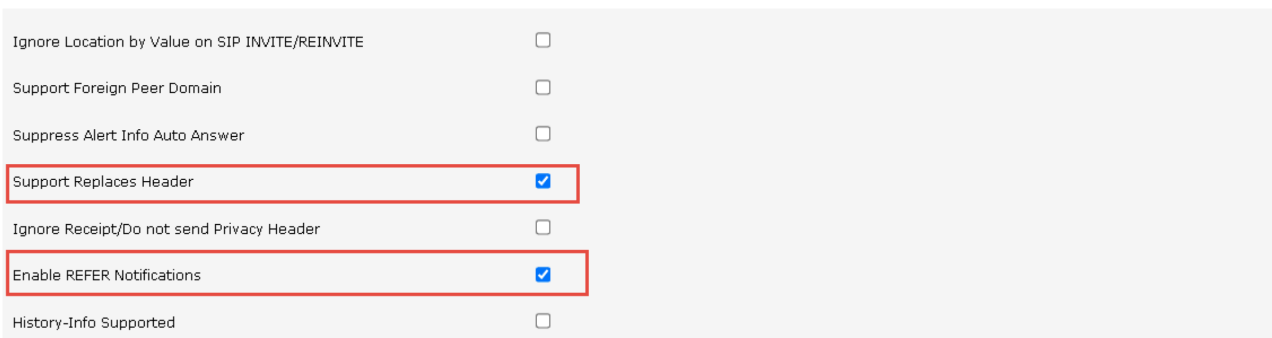
- **Do not send Invite without SDP**
- **Send International Numbers in Global number format (GNF)**



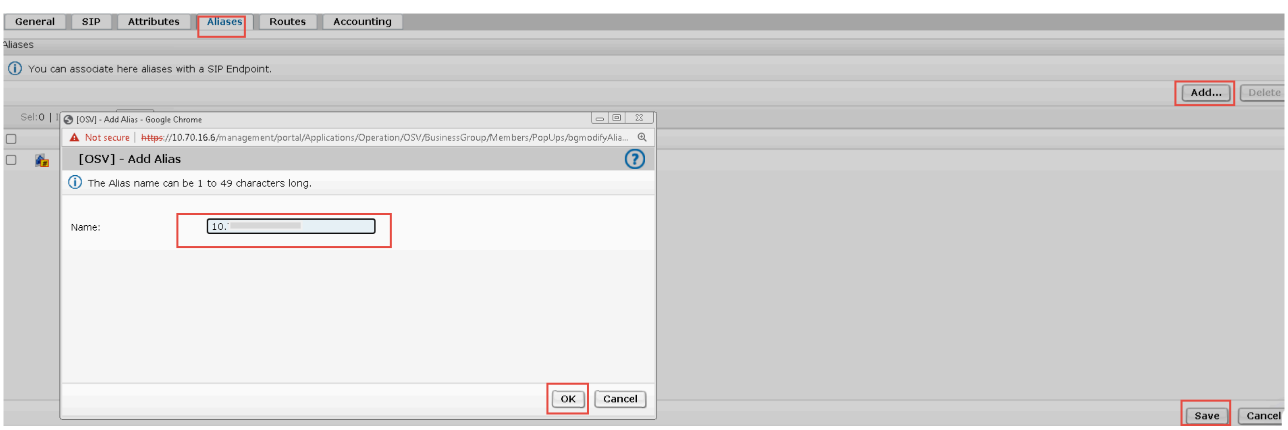
- **Enable session timer**
- **Limited PRACK support**



- **Support Replaces Header**
- **Enable REFER Notifications**



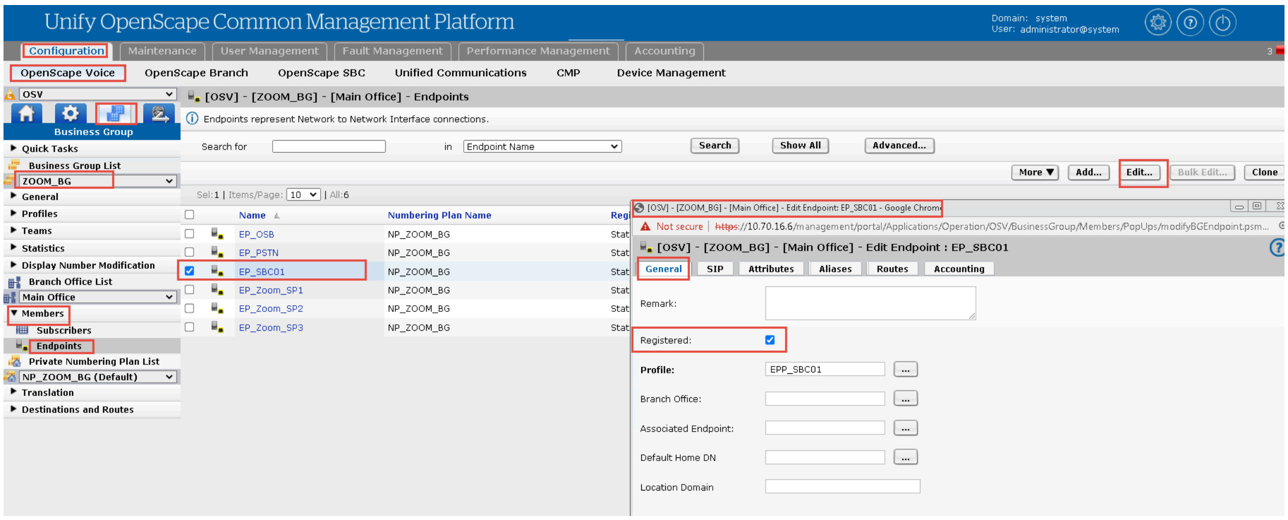
12. Select the **Aliases** tab, click **Add** and enter the **SBC LAN interface** with port number for the incoming SIP traffic.



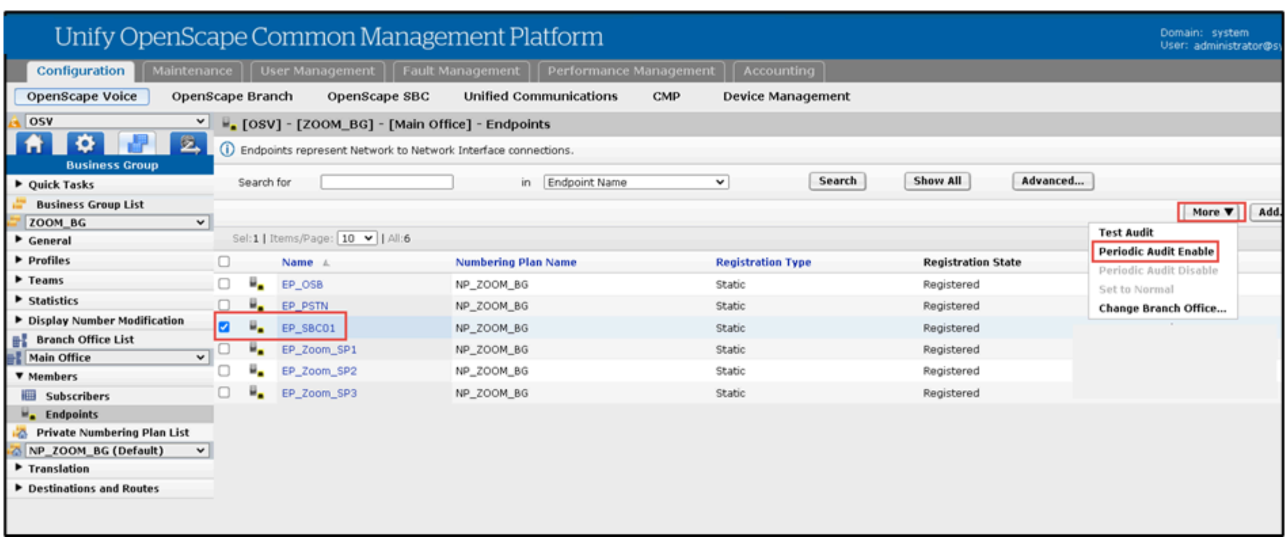
13. Click **OK** and then click **Save.**

5.1.4 Endpoint Overview

1. Navigate to the **Unify OpenScape Common Management Platform** > **Configuration** > **Unify OpenScape Voice** > **Business Group** > **Members** > **Endpoints** window. A list of all the configured endpoints in Unify OpenScape Voice is displayed.
2. Select an endpoint and click **Edit.**
3. In the **Edit Endpoint** pop-up, under the **General** tab, check the **Registered** checkbox.



4. Enable the **Registered** option for all the created endpoints.
5. To activate the sending of SIP OPTIONS messages for all the created endpoints, select an endpoint and click the **More** drop-down menu to expand it.



6. Select the **Periodic Audit Enable** option to enable it and route the traffic to the accessible Zoom endpoint(s).

The overview of the created endpoints on the Common Management Platform window is displayed as below:

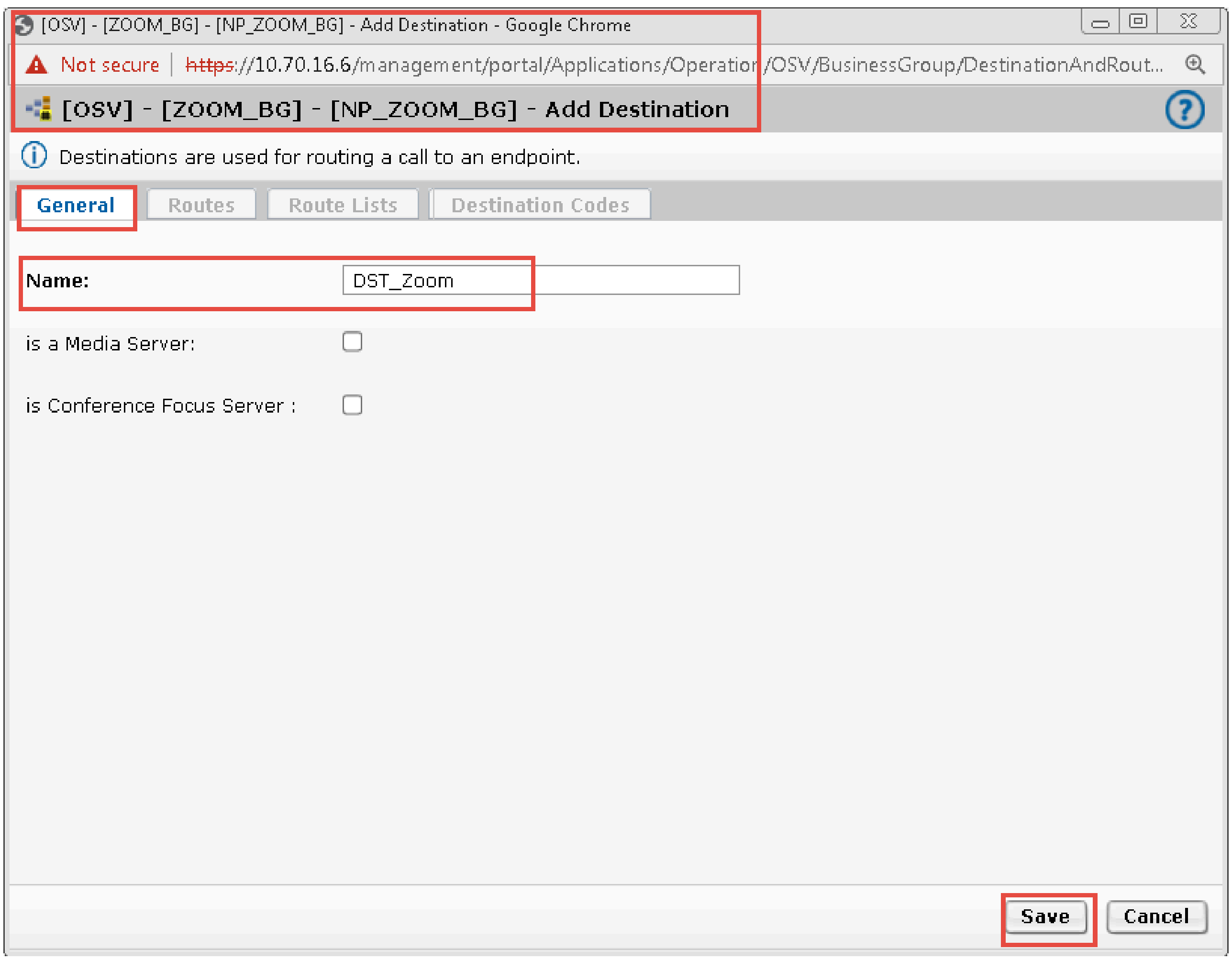
5.2 Destinations and Routes Configuration

Destinations are logical targets for off-net or on-net routing. When a destination is created, its name is bound to the numbering plan where it is made. Destinations are used to route a call to an endpoint representing a gateway.

Each **Route** is a collection of groups or addresses providing a destination path.

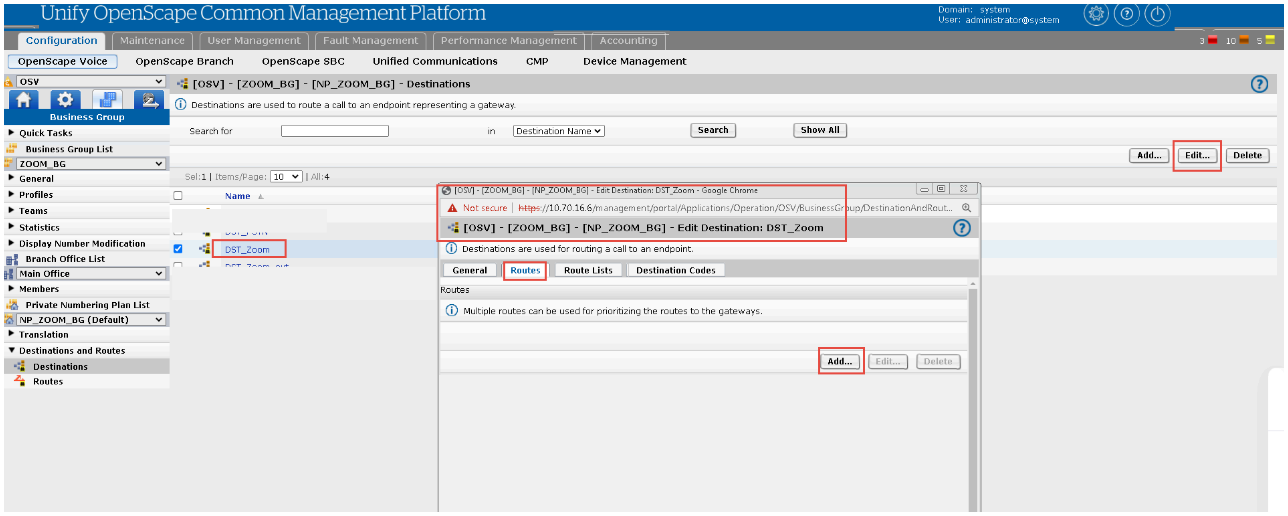
5.2.1 Configuring the Zoom Destination

1. Navigate to **Unify OpenScape Common Management Platform** > **Configuration** > **Unify OpenScape Voice** > **Business Group** > **Destinations and Routes** > **Destinations**.
2. Click **Add**.
3. In the **Add Destination** pop-up**,** under the **General** tab, enter the name of the Zoom destination. For example, DST_Zoom.



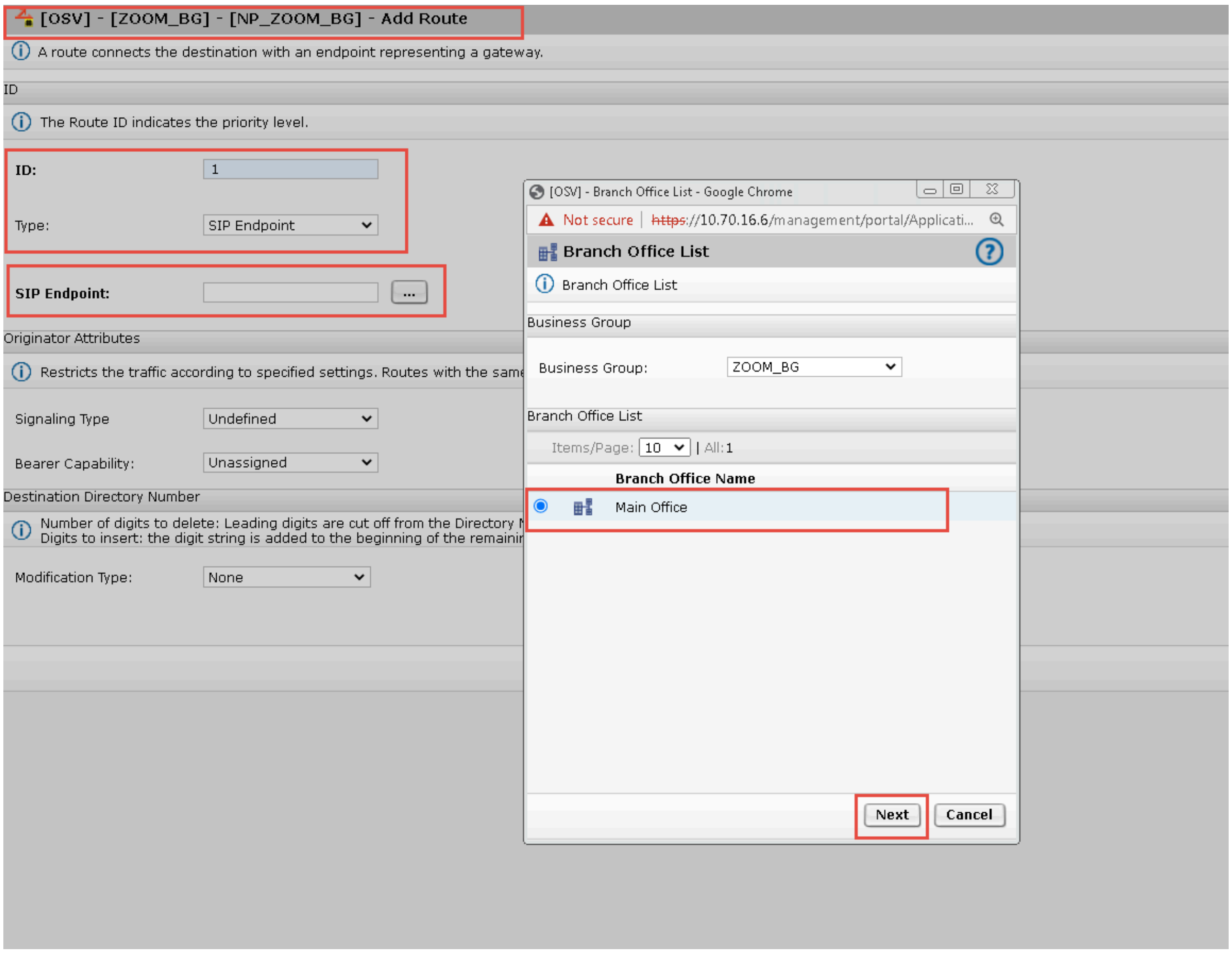
4. Click **Save.**
5. Select the destination you created in the previous step and click **Edit**.

6. In the **Edit Destination** pop-up, select the **Routes** tab and click **Add.**



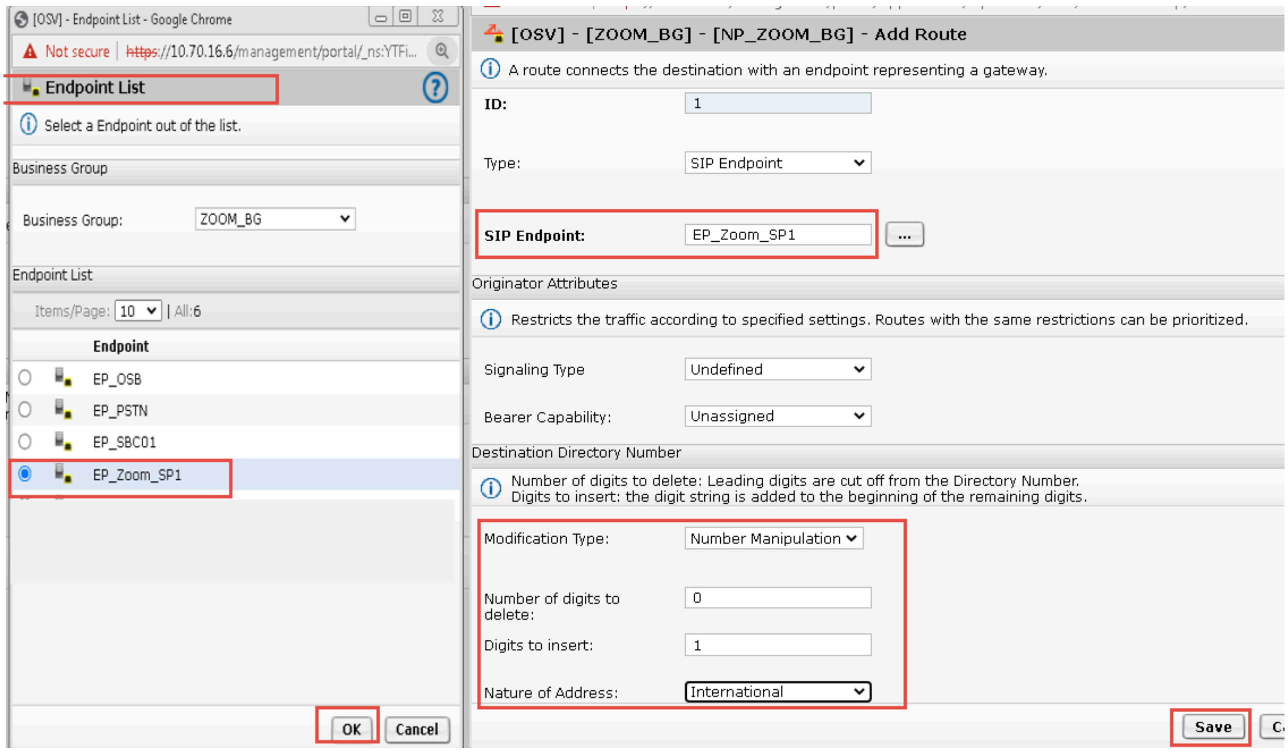
7. In the **Add Route** pop-up, configure the following:

   a. **ID**: Enter the priority level of this route ID as 1 (if there are multiple routes to a destination and route prioritization is selected, the route with the lowest numbered route ID has the highest priority and will be selected first).
   b. From the **Type** drop-down menu, select **SIP Endpoint**.
   c. Click the three-dot icon at the right side of the **SIP Endpoint.**
   d. In the **Branch Office List** pop-up, select **Main Office** and click **Next.**

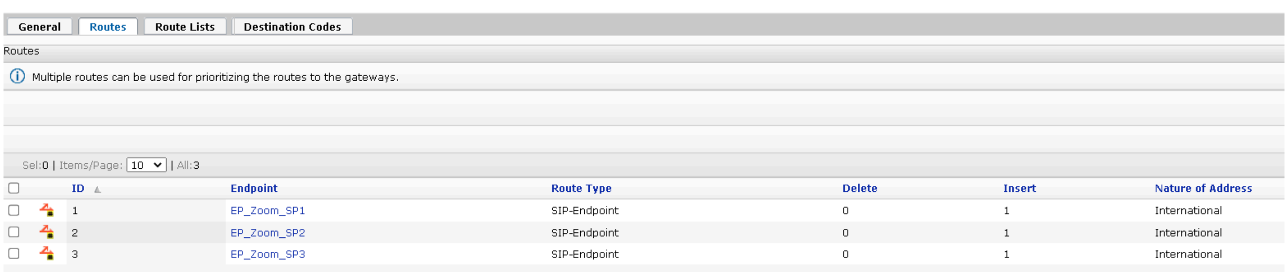


8. In the **Endpoint list** pop-up, select the **Zoom endpoint** and click **OK,** which applies in the SIP endpoint field**.** Configure the destination directory number settings as below:

   a. From the **Modification Type** drop-down menu**,** select **Number Manipulation.**
   b. From the **Number of digits to delete drop-down menu,** select the number of digits to cut off from the directory number.
   c. From the **Digits to insert** drop-down menu, enter the digit string which gets added to the beginning of the remaining digits.
   d. From the **Nature of Address** drop-down menu**,** select **International.**
   e. Click **Save.**

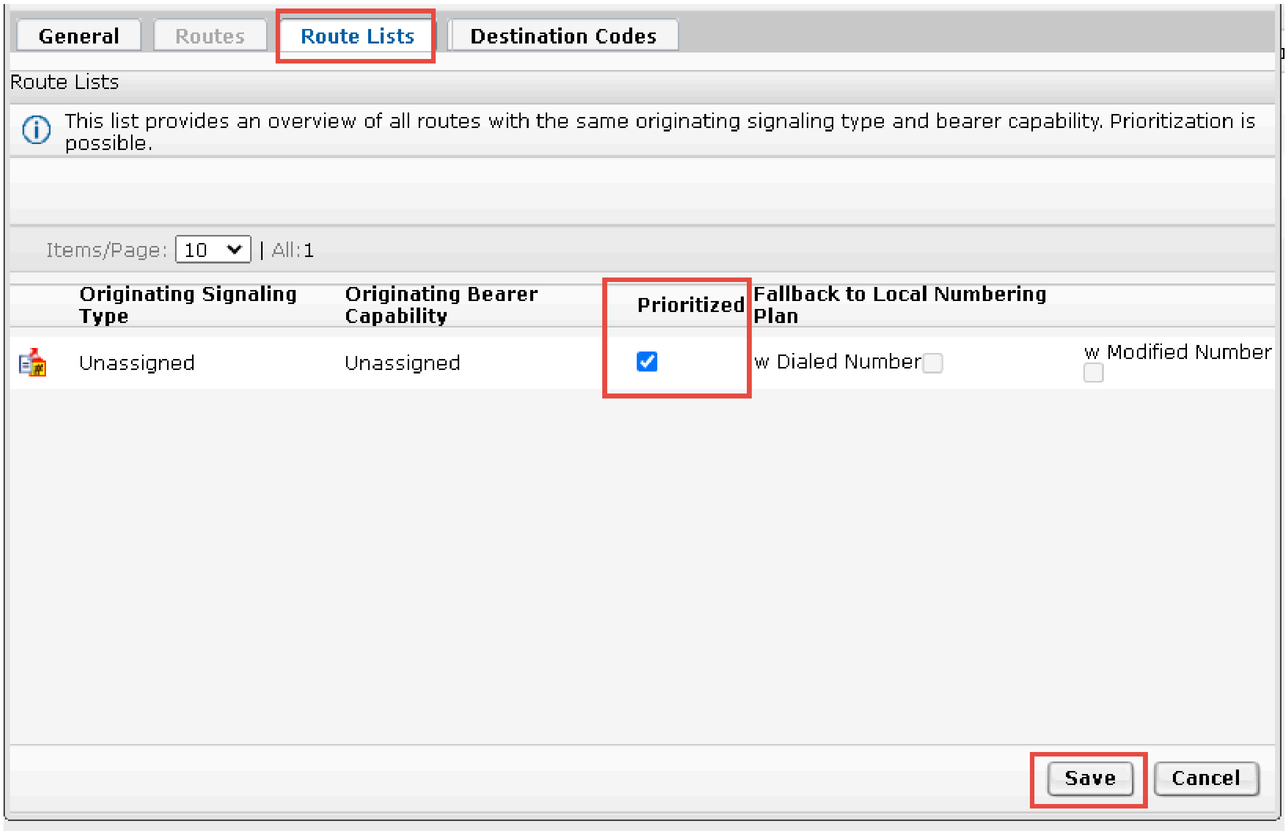


9. Repeat the same procedure for the remaining Zoom endpoints, assigning a different ID and priority level per endpoint, as shown in the example below:



ID	Endpoint	Route Type	Delete	Insert	Nature of Address
1	EP_Zoom_SP1	SIP-Endpoint	0	1	International
2	EP_Zoom_SP2	SIP-Endpoint	0	1	International
3	EP_Zoom_SP3	SIP-Endpoint	0	1	International

10. In the **Route Lists** tab, select the **Prioritized** flag to enable the Zoom route prioritization, as shown below:



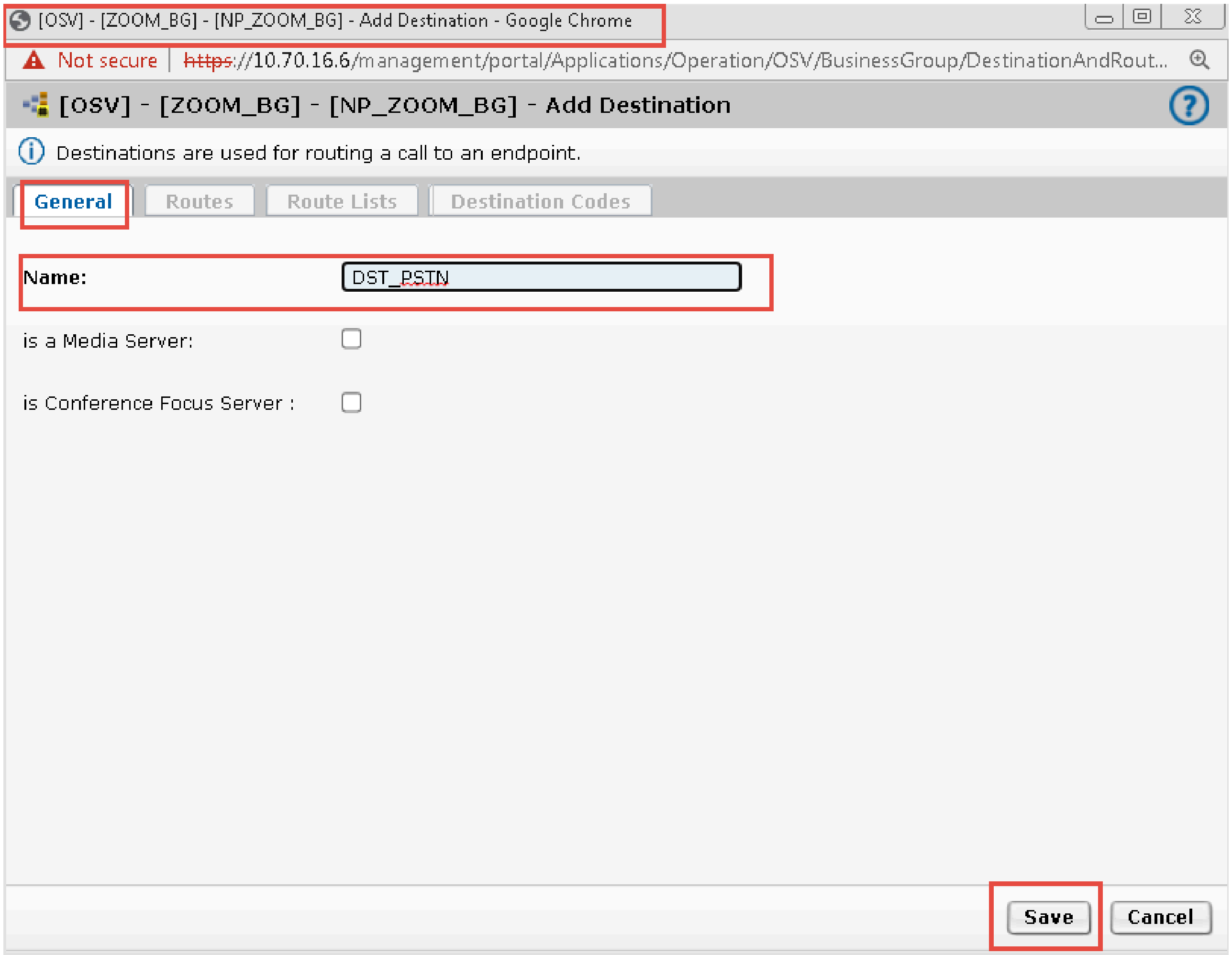
11. Click **Save.**

5.2.2 Configuring the PSTN Destination

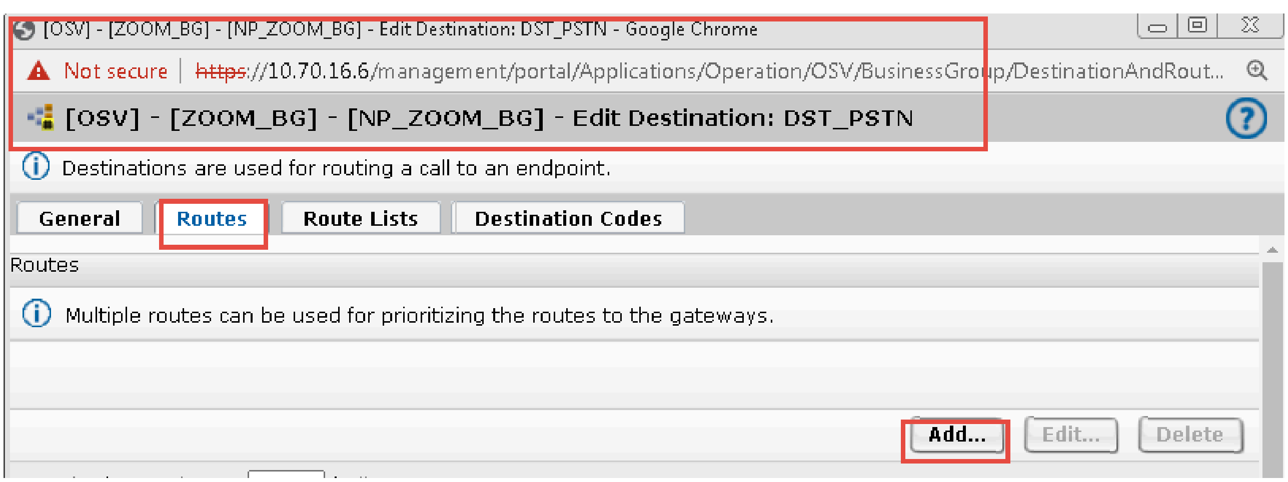
Note:
The configuration below is an example. The actual configuration steps depend on your provider's requirements.

1. Navigate to **Unify OpenScape Common Management Platform** > **Configuration** > **Unify OpenScape Voice** > **Business Group** > **Destinations and Routes** > **Destinations**.
2. Click **Add.**

3. In the **Add Destination** pop-up, under the **General** tab, enter the name of the Zoom destination. For example, DST_PSTN.

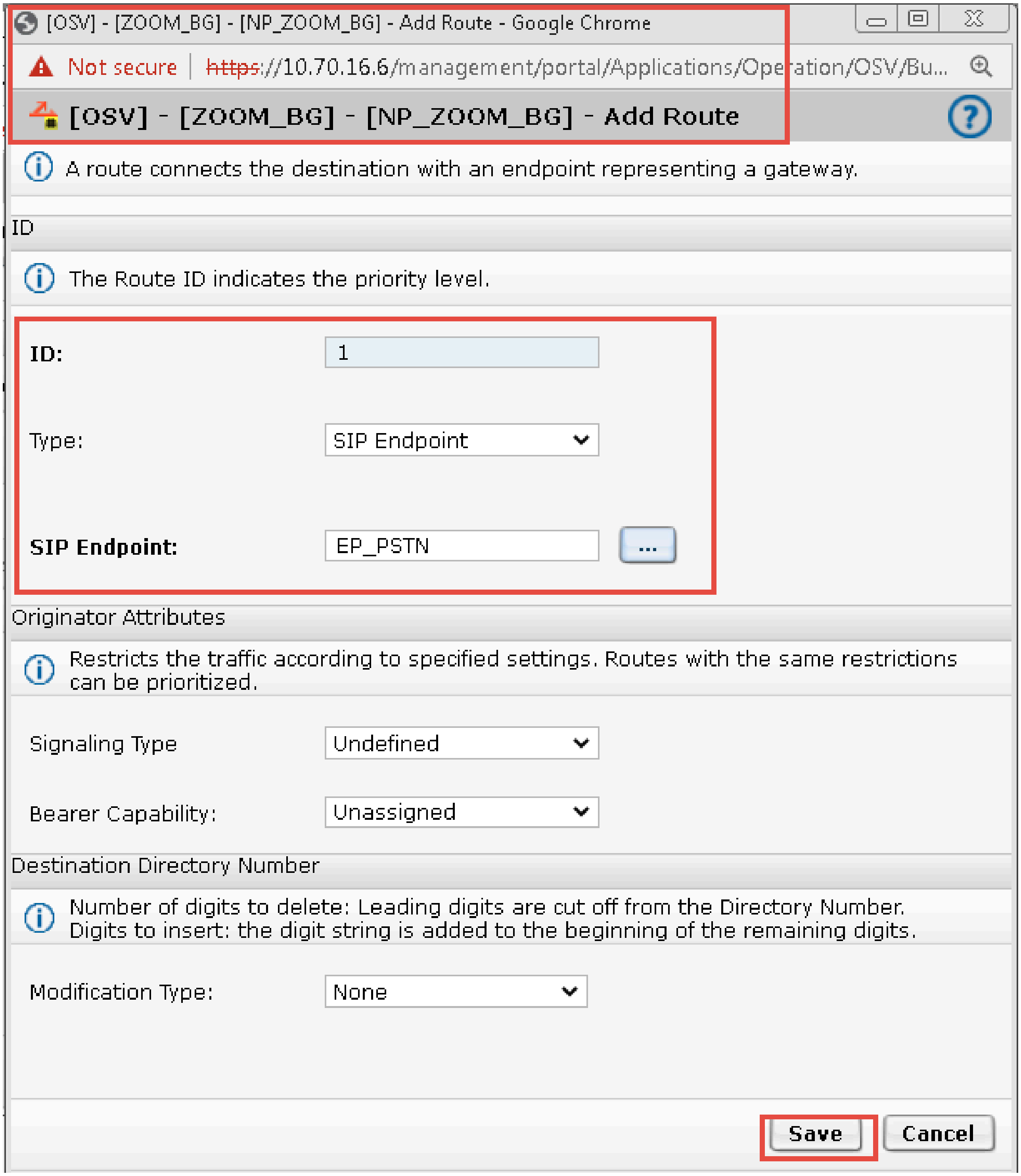


4. Click **Save.**
5. Select the destination you created in the previous step and click **Edit**.
6. In the **Edit Destination** window, select the **Routes** tab and click **Add.**



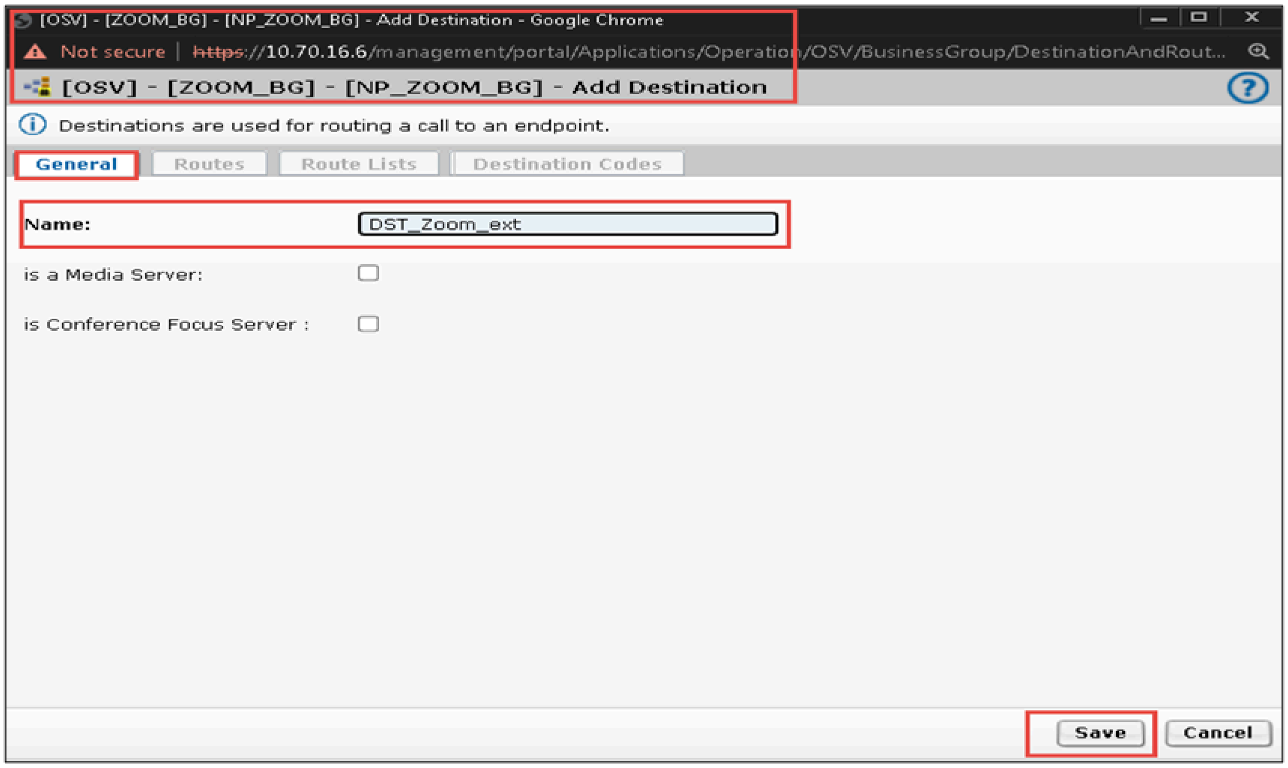
7. In the **Add Route** pop-up, configure the following:

   a. **ID**: Enter the priority level of this route ID as 1 (if there are multiple routes to a destination and route prioritization is selected, the route with the lowest numbered route ID has the highest priority and will be selected first).
   b. From the **Type** drop-down menu, select **SIP Endpoint**.
   c. In the **SIP Endpoint** field, enter the **PSTN endpoint.** For example**, EP_PSTN**.

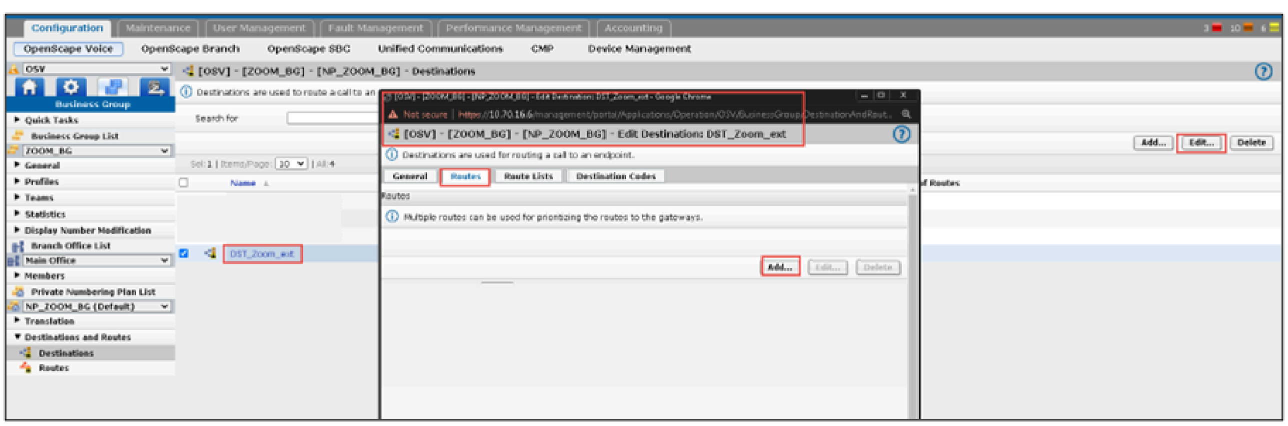


5.2.3 Configuring the OpenScape OSV extension Destination

1. Navigate to **Unify OpenScape Common Management Platform** > **Configuration** > **Unify OpenScape Voice** > **Business Group** > **Destinations and Routes** > **Destinations**.
2. Click **Add.**
3. In the **Add Destination** pop-up**,** under the **General** tab, enter the name of the Zoom destination. For example, DST_Zoom_ext.



4. Click **Save.**
5. Select the destination you created in the previous step and click **Edit**.
6. In the **Edit Destination** window, select the **Routes** tab and click **Add.**

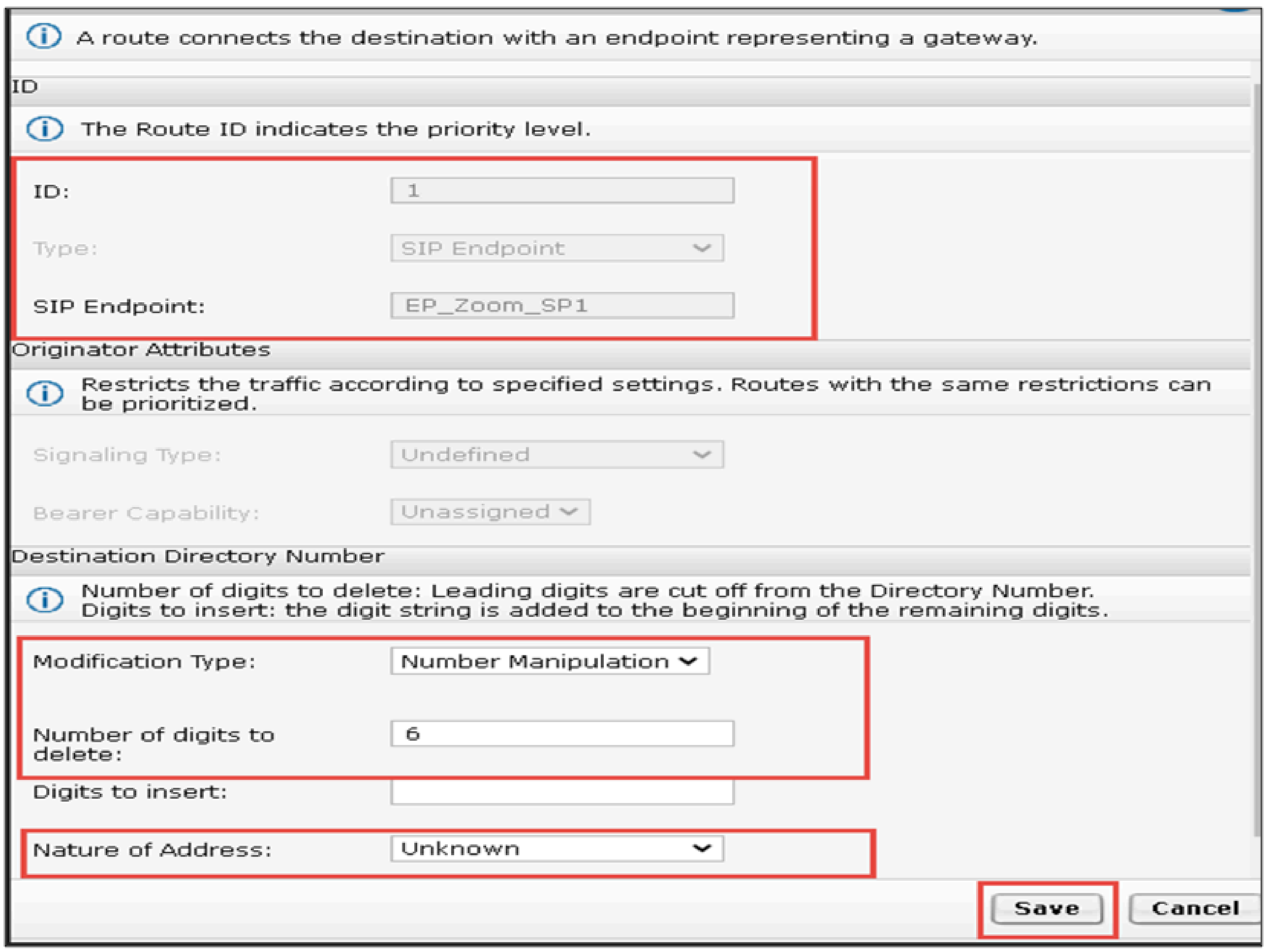


7. In the **Add Route** pop-up, configure the following:

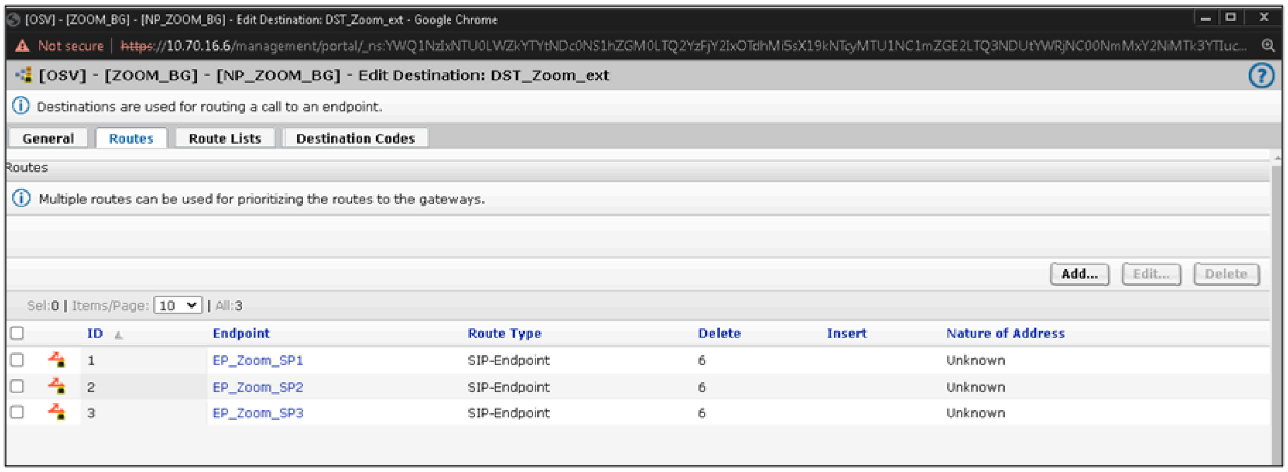
   a. **ID**: Enter the priority level of this route ID as 1 (if there are multiple routes to a destination and route prioritization is selected, the route with the lowest numbered route ID has the highest priority and will be selected first).
   b. From the **Type** drop-down menu, select **SIP Endpoint**.
   c. In the **SIP Endpoint** field, enter the **Zoom endpoint.** For example**, EP_Zoom_SP1**.
   d. Configure the destination directory number settings as below:

      i. From the **Modification Type** drop-down menu, select **Number Manipulation**.
      ii. In the **Number of digits to delete** field, enter the number of digits to cut off from the directory number.
      iii. From the **Number of Address** drop-down menu, select **Unknown**.

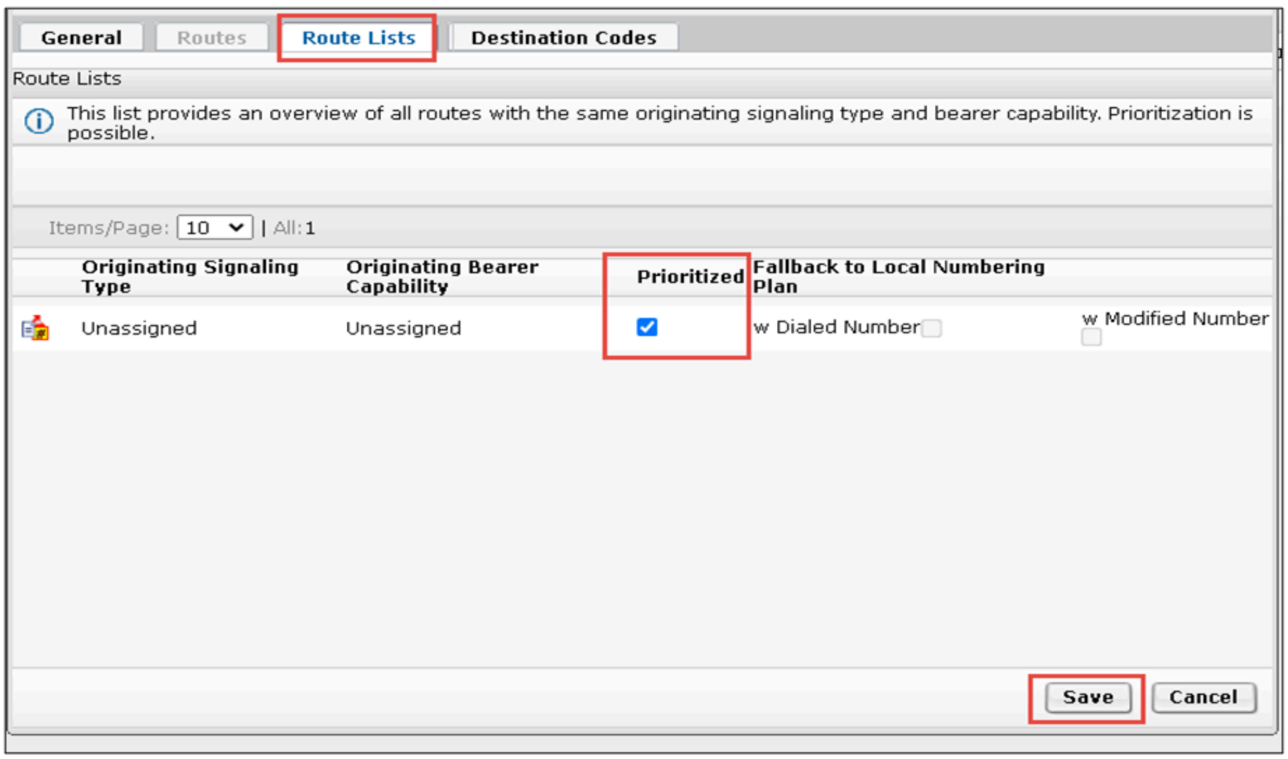
   e. Click **Save**.



8. Repeat the same procedure for the remaining Zoom endpoints, assigning a different ID and priority level per endpoint, as shown in the example below:



9. In the **Route Lists** tab, enable the **Prioritized** flag and click **Save**.



5.3 Translation Configuration

With **Translation,** the administrator configures the routing of outgoing calls based on the dialed digits from OS Voice subscribers. A call can only be routed if the dialed digits match a PAC (Prefix Access Code).

The **Destination Code** feature provides destination codes for basic telephone service. A destination code will be applied to a call if the dialed or modified (via PAC) digits and the nature of the address match.

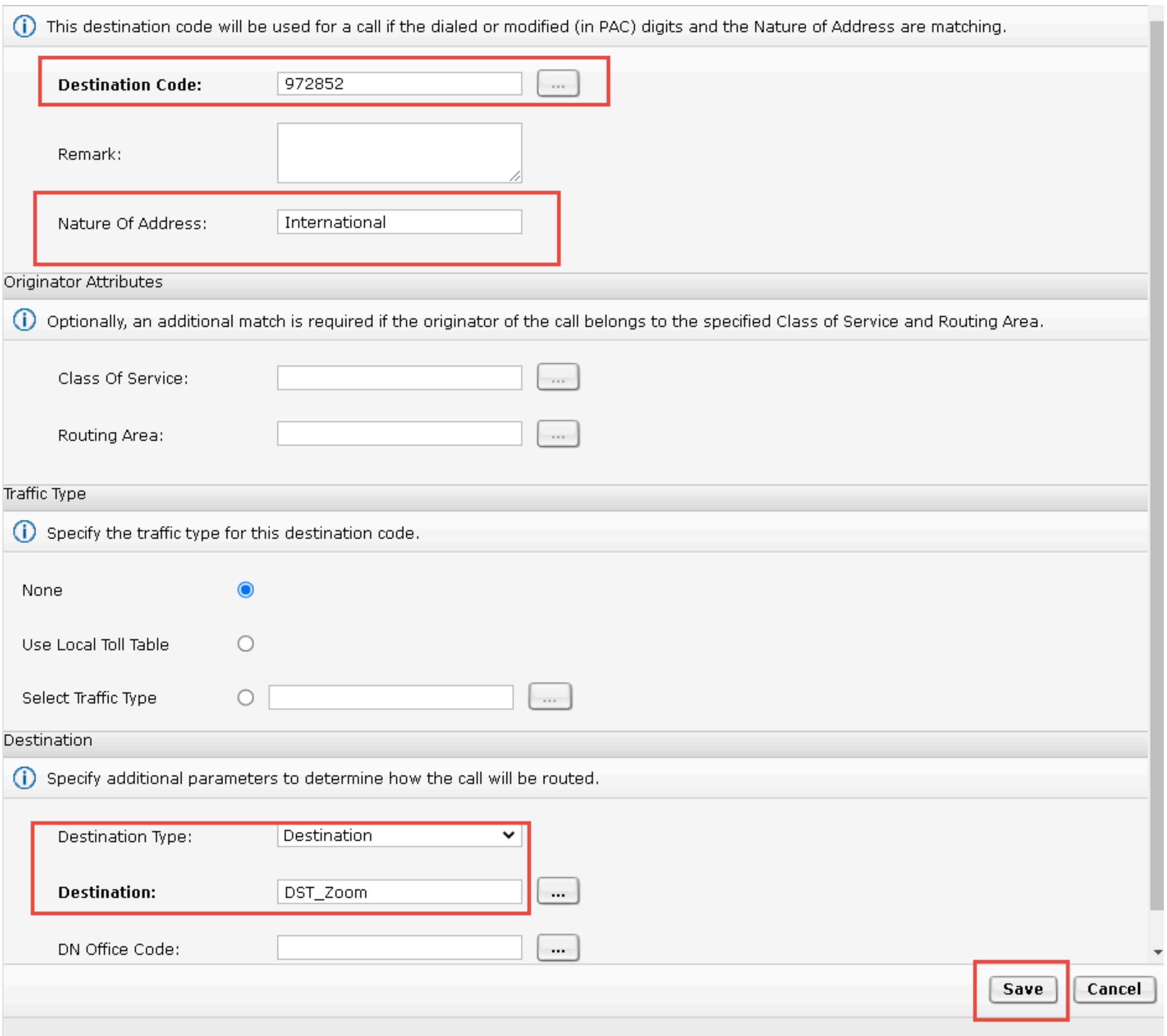
5.3.1 Configuring the Zoom Numbers Routing

1. Navigate to **OpenScape Common Management Platform** > **Configuration** > **OpenScape Voice** > **Business Group** > **Translation** > **Prefix Access Codes**.
2. Click **Add.**
3. In the **Add Prefix Access Code** pop-up, configure the following parameters:

   a. **Prefix Access code:** Enter the starting digits of Zoom users.
   b. **Minimum length:** Enter the minimum expected length of Zoom numbers.
   c. **Maximum length:** Enter the maximum expected length of Zoom numbers.
   d. **Digit Position:** Configure as 0, which implies not removing any digits from the dialed number before sending it to the destination.
   e. **Prefix Type:** Configure the off-net access to permit access to remote destinations.
   f. From the **Nature of Address** drop-down menu, select **International**.
   g. From the **Destination Type** drop-down menu, select **None**. The resulting digits will be processed in the user's numbering plans destination codes table.
   h. Click **Save.**
4. Navigate to **OpenScape Common Management Platform** > **Configuration** > **Unify OpenScape Voice** > **Business Group** > **Translation** > **Destination Codes**.
5. Click **Add**.

6. In the **Add Destination Code** pop-up, configure the following:

   a. **Destination Code**: Select the previously created Prefix Access Code (PAC).
   b. From the **Nature of Address** drop-down menu, select **International**.
   c. From the **Destination Type** drop-down menu, select **Destination**.
   d. **Destination:** Select the destination of Zoom. For example, DST_Zoom.
   e. Click **Save**.

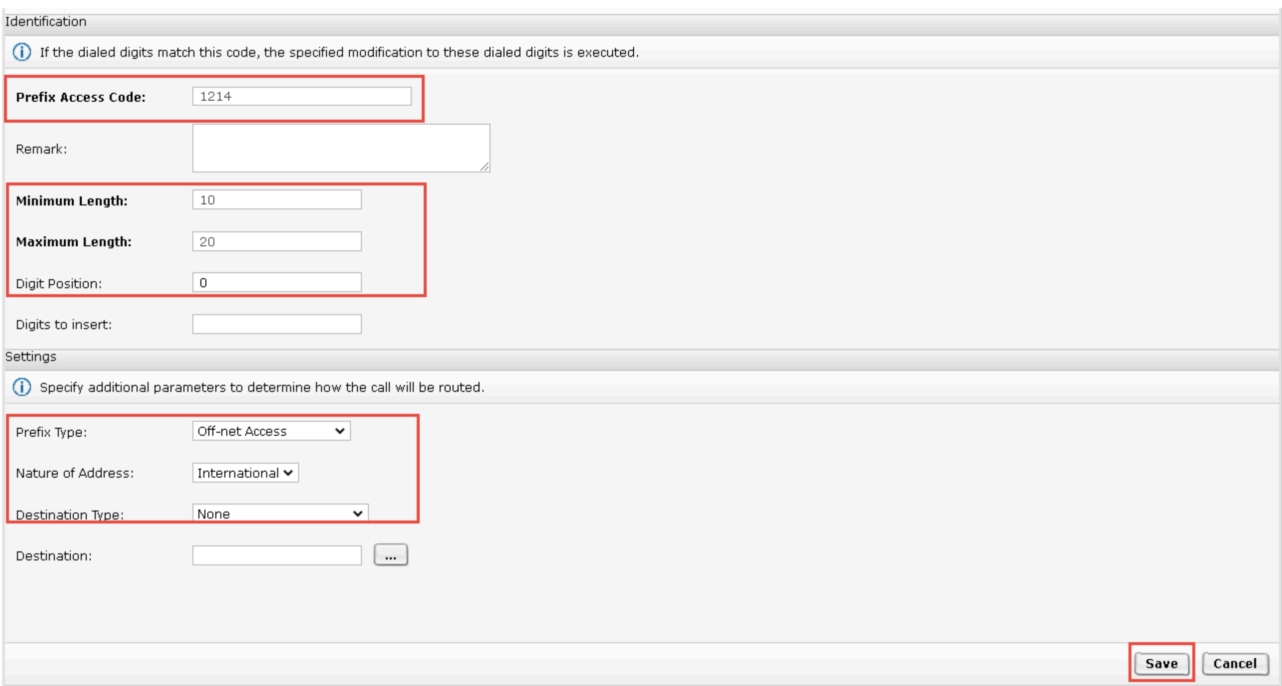


5.3.2 Configuring the PSTN Numbers Routing

Note:
The configuration below is an example. The actual configuration steps depend on your provider's requirements.

1. Navigate to **OpenScape Common Management Platform** > **Configuration** > **OpenScape Voice** > **Business Group** > **Translation** > **Prefix Access Codes**.
2. Click **Add.**
3. In the **Add Prefix Access Code** pop-up, configure the following:

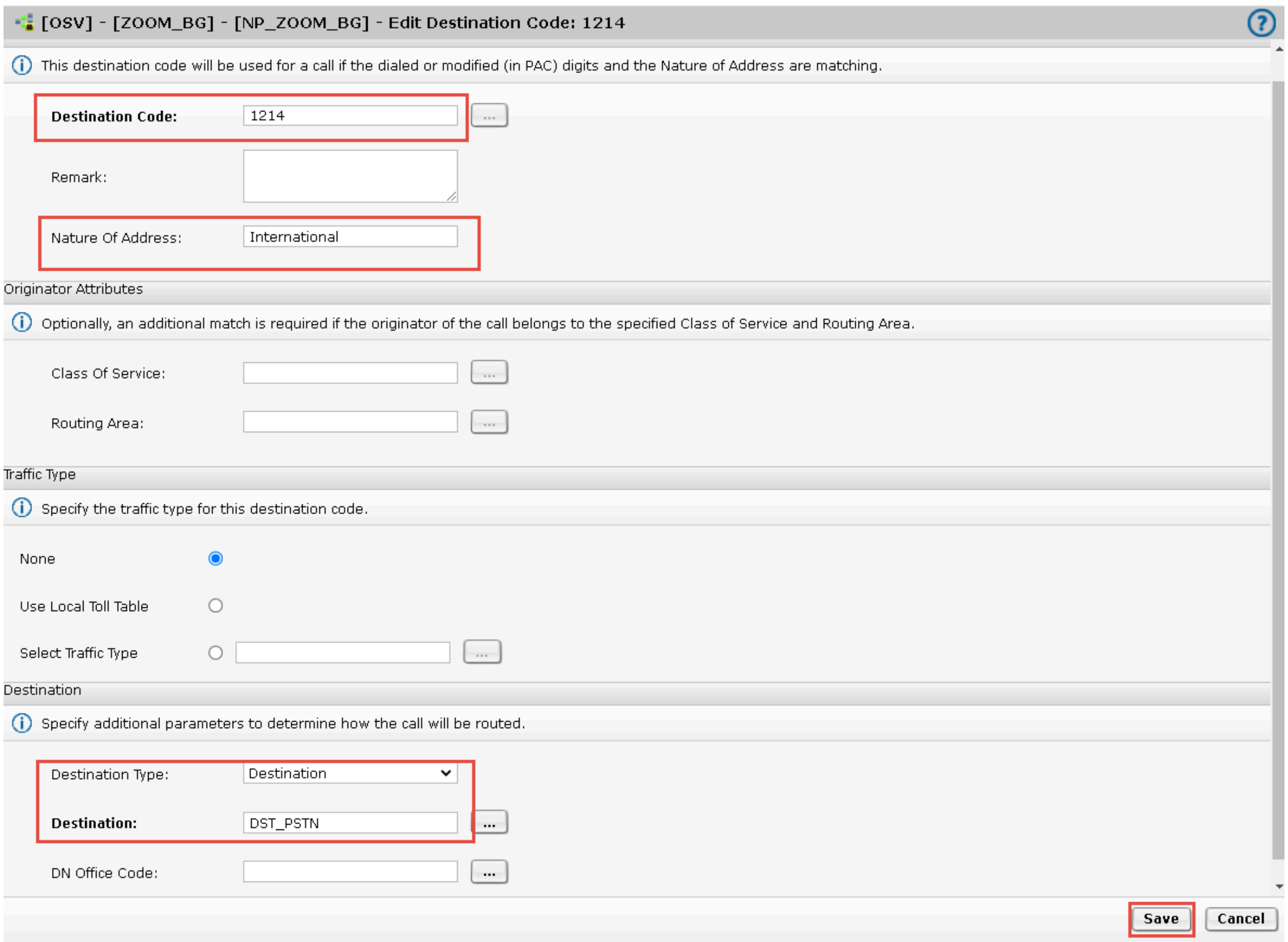
   a. **Prefix Access code:** Enter the starting digits of the PSTN users.
   b. **Minimum length:** Enter the minimum expected length of the PSTN numbers.
   c. **Maximum length:** Enter the maximum expected length of the PSTN numbers.
   d. **Digit Position:** Configured as 0, which implies not removing any digits from the dialed number before sending it to the destination.
   e. From the **Prefix Type** drop-down menu, select **Off-net access** to permit access to remote destinations.
   f. From the **Nature of Address** drop-down menu, select **International**.
   g. From the **Destination Type** drop-down menu**,** select **None** so that the resulting digits are processed in the user's numbering plan destination codes table.
4. Click **Save**.



5. Navigate to **OpenScape Common Management Platform** > **Configuration** > **Unify OpenScape Voice** > **Business Group** > **Translation** > **Destination Codes**.
6. Click **Add.**

7. In the **Add Destination Code** pop-up, configure the following:

   a. **Destination code:** Select the previously created Prefix access code.
   b. From the **Nature of Address** drop-down menu, select **International**.
   c. From the **Destination Type** drop-down menu, select **Destination**.
   d. **Destination:** Select the destination of PSTN.
8. Click **Save**.



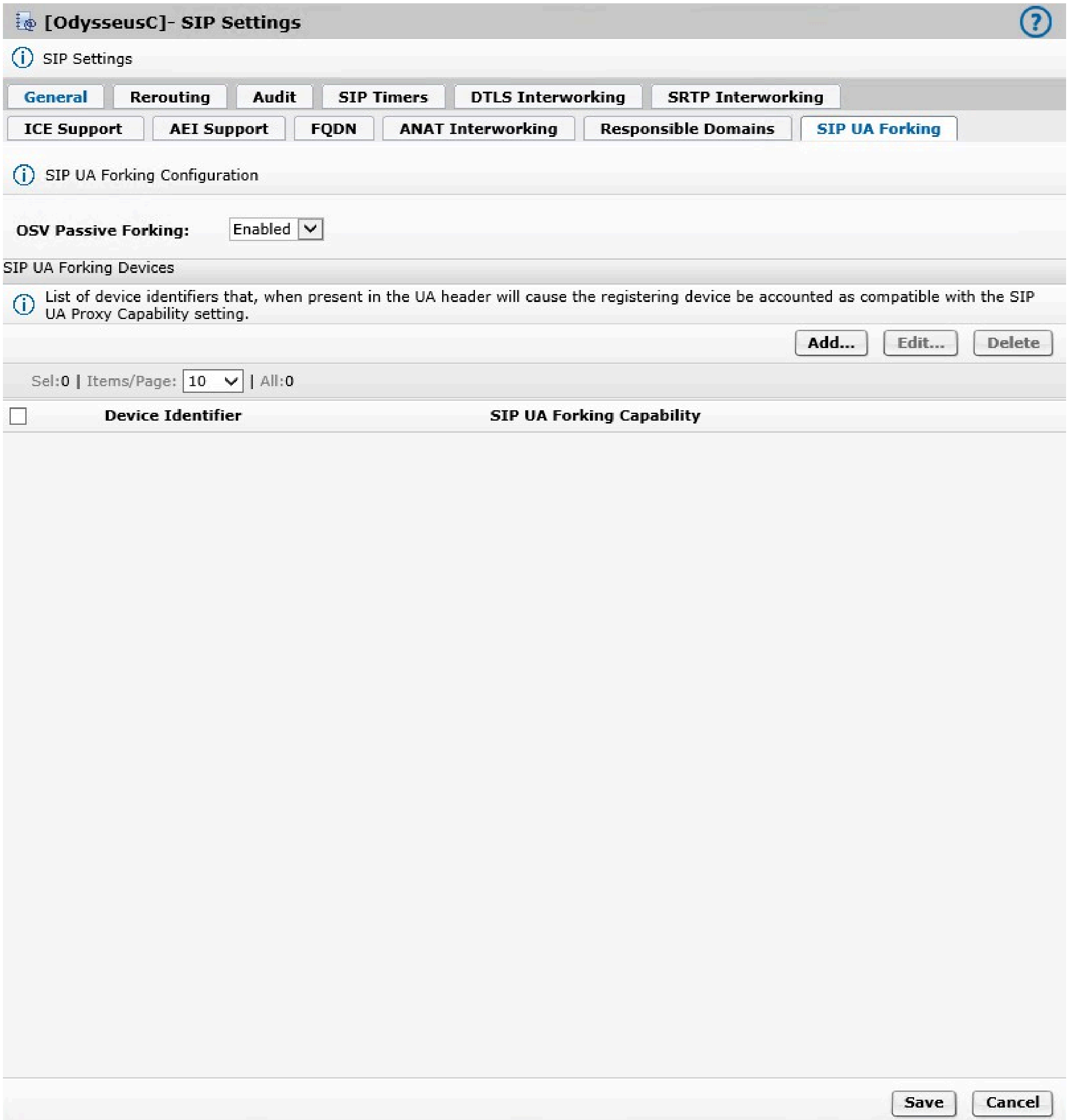
5.4 Configuring the SIP UA Forking

OSV Passive Forking (UAC) provides an interworking function that merges multiple Zoom downstream early dialogs into a single upstream SIP dialog. This functionality shields upstream SIP clients (SIP UAC) establishing sessions with the Zoom network from being exposed to the full RFC 3261/RFC3264 forking SIP Proxy server behavior of the Zoom Phone System. The SIP UA Forking tab enables the feature and lists all devices configured with their respective SIP forking capabilities.

To activate the OSV Passive Forking feature:

1. Navigate to **OpenScape Common Management Platform > Configuration > OSV > Administration > Signaling Management > SIP**.

2. In the **SIP UA Forking** tab, from the **OSV Passive Forking** drop-down menu, select **Enabled** and click **Save**.



5.5 Configuring Display Number Modification

In case the FROM of an INVITE message needs to be manipulated from, for example, the SBC to the SSP, the header manipulation occurs in the OpenScape Voice **Display Number Modification** configuration.

Prerequisite

1. You have created an Office Code to OpenScape Voice. For example:
   - Country Code: 1
   - Area Code: 972
   - Local Office Code: 598
   - Directory Number Start:xxx (starting extension)

To configure the Display Number Modification, follow the instructions below:

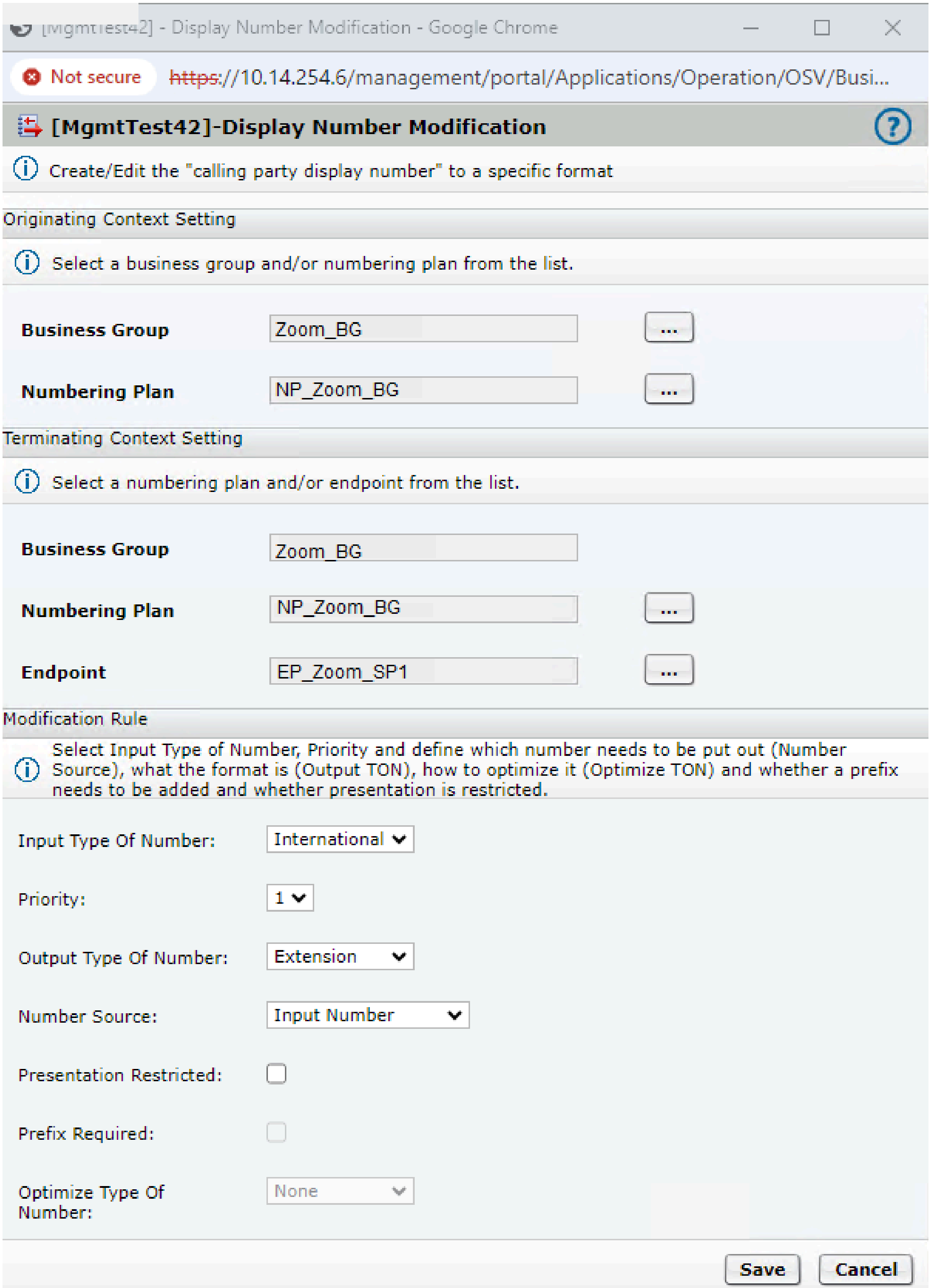
> **Note**:
> The number modification configuration below is an example. The actual configuration steps are dependent on the requirements of your provider.

1. Navigate to **OpenScape Common Management Platform** > **Configuration** > **OSV** > **Business Group** > **Display Number Modification** > **Modifications**.
2. Click **Add**.

3. Add the office code by entering the following:

- **Endpoint**: EP_Zoom_SP1
- **Input Type Of Number**: International
- **Priority:** 1 (highest priority)
- **Output Type Of Number**: Extension
- **Number Source:**

  Input Number (defines the input format of the "presenter number" when it comes into the OpenScape Voice.


[MgmtTest42] - Display Number Modification - Google Chrome
Not secure
https://10.14.254.6/management/portal/Applications/Operation/OSV/Busi...
[MgmtTest42]-Display Number Modification
Create/Edit the "calling party display number" to a specific format
Originating Context Setting
Select a business group and/or numbering plan from the list.
Business Group
Zoom_BG
Numbering Plan
NP_Zoom_BG
Terminating Context Setting
Select a numbering plan and/or endpoint from the list.
Business Group
Zoom_BG
Numbering Plan
NP_Zoom_BG
Endpoint
EP_Zoom_SP1
Modification Rule
Select Input Type of Number, Priority and define which number needs to be put out (Number Source), what the format is (Output TON), how to optimize it (Optimize TON) and whether a prefix needs to be added and whether presentation is restricted.
Input Type Of Number:
International
Priority:
1
Output Type Of Number:
Extension
Number Source:
Input Number
Presentation Restricted:
Prefix Required:
Optimize Type Of Number:
None
Save
Cancel

4. Repeat the steps for the remaining Zoom endpoints.
5. Click**Save**.

Unify OpenScape SBC Configuration 6

This chapter contains the following sections:

This chapter outlines the configuration of OpenScape SBC for interworking with Zoom Direct Routing. Once OSV is configured, you can use the SBC to route calls, secure communication, and manage traffic to Zoom Phone and PSTN networks.

Prerequisite

1. You have obtained a public certificate issued by one of the Zoom-supported CAs. Refer to the Configuring Certifications section.

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

The OpenScape Session Border Controller is a highly scalable SBC solution supporting broad SIP interoperability, advanced media handling and robust security. OpenScape SBC enables enterprises to deliver voice services like SIP trunking and unified communications. OpenScape SBC performs interoperability, security, management, and control capabilities to support SIP trunking applications. The OpenScape SBC is not a B2BUA and has limited SIP message manipulation capabilities. The OpenScape Voice IP-PBX provides call routing, enhanced SIP message manipulation, and number modification facilities. Thus, the SIP signaling for incoming and outgoing calls to Zoom clients will always pass through the OS Voice.

The OpenScape SBC will be configured with the connection to OS Voice, SSP (BCOM) and Zoom Phone System (remote) endpoints.

As an example:

Table 1: Zoom IPs Table

Items	Example
SBC Core (LAN) IP	10.8.242.72
SBC Access (WAN) IP	195.97.14.76

Items	Example
SBC Public FQDN	sbc01.athdrlabs.xyz
OS Voice node 1 (SIP Signaling) IP	10.8.242.16 TCP 5060
OS Voice node 2 (SIP Signaling) IP	10.8.242.26 TCP 5060
Zoom IP 1 SIP trunk	162.12.233.59 (see note) TLS 5061 (LAN port for OS Voice trunk 50001)
Zoom IP 2 SIP trunk	162.12.232.59 (see note) TLS 5061 (LAN port for OS Voice trunk 50002)
Zoom IP 3 SIP trunk	162.12.235.59 (see note) TLS 5061 (LAN port for OS Voice trunk 50003)
SSP (BCOM) SIP trunk	Remote URL: sip.bcom.nl Default Home DN: 31850080990 (LAN port for OS Voice trunk 50010)

Important:
The Zoom IP address example is valid for the North America region. Please check the Zoom site for the current IP Addresses.

Whether routine or not, Zoom Phone Direct Routing's specific OSSBC configuration will be omitted. Unify OpenScape SBC installation and administration documentation can be found on the Unify customer documentation site.

Table 2: Signaling Traffic IPs

Traffic T ype	Protocol	Source	Destination Ports	Destination IPs	Region
				162.12.233.59	

Traffic Type	Protocol	Source	Destination Ports	Destination IPs	Region
				162.12.232.59	North America
				162.12.235.85	
				64.211.144.247	LATAM
				149.137.69.247	
				213.19.144.198	EMEA
				213.244.140.198	
Signaling	TLS	Customer SBC	5061	103.122.166.248 103.122.167.248	Australia
				149.137.41.246	APAC
				207.226.132.198	
				209.9.211.198	HK
				101.36.167.237	HK2
				149.137.25.246	Japan
				207.226.132.198	

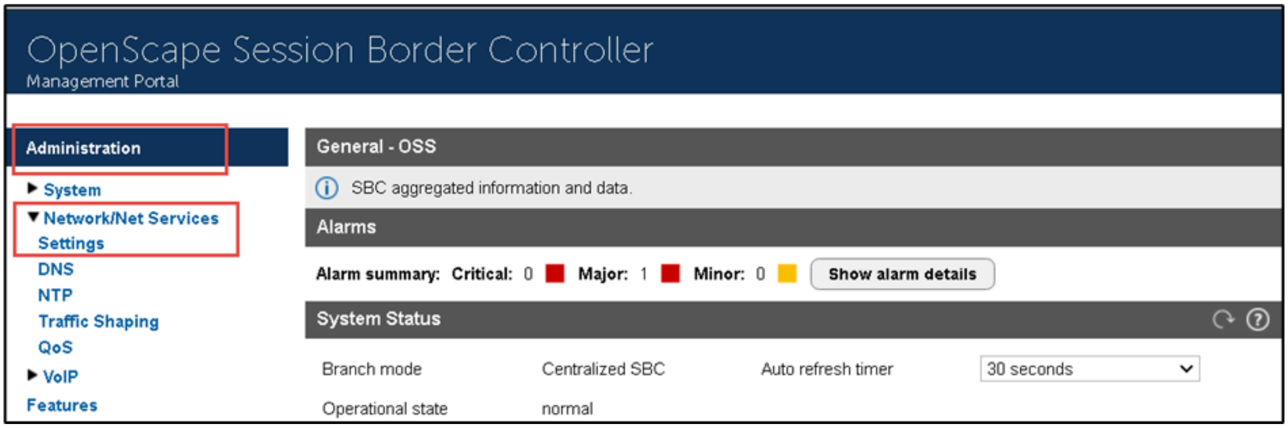
Table 3: Media Traffic IPs

Traffic Type	Protocol	Source	Destination Ports	Destination IPs	Region
				162.12.232.0/24	

Traffic T ype	Protocol	Source	Destination Ports	Destination IPs	Region
				162.12.233.0/24	North America
				162.12.235.0/24	
				64.211.144.0/24	LATAM
				149.137.69.0/24	
				213.19.144.128/25	EMEA
				213.244.140.0/24	
Media	UDP/SRTP	Customer SBC	20000-64000	103.122.166.0/24 103.122.167.0/24	Australia
				149.137.41.0/24	APAC
				207.226.132.0/24	
				209.9.211.192/26	HK
				101.36.167.0/24	
				207.226.132.0/24	Japan
				149.137.25.0/24	

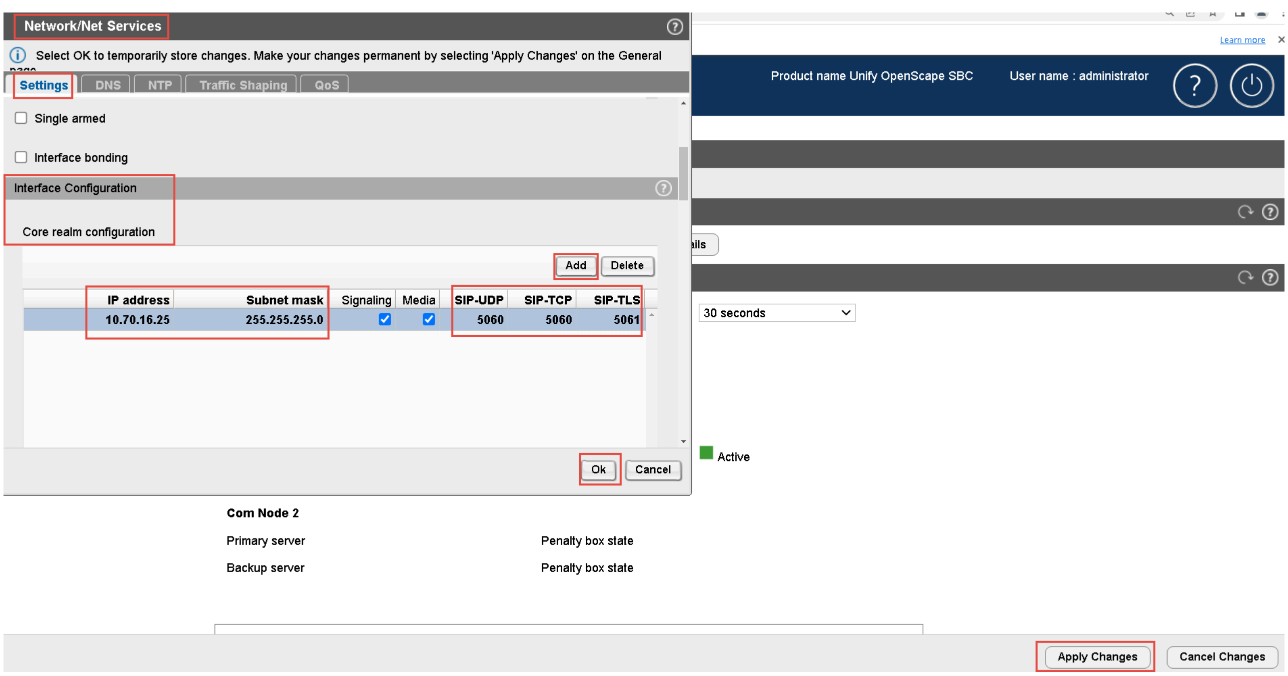
6.1 Configuring Network settings

1. Navigate to **Administration** > **Network/Net Services** > **Settings**.



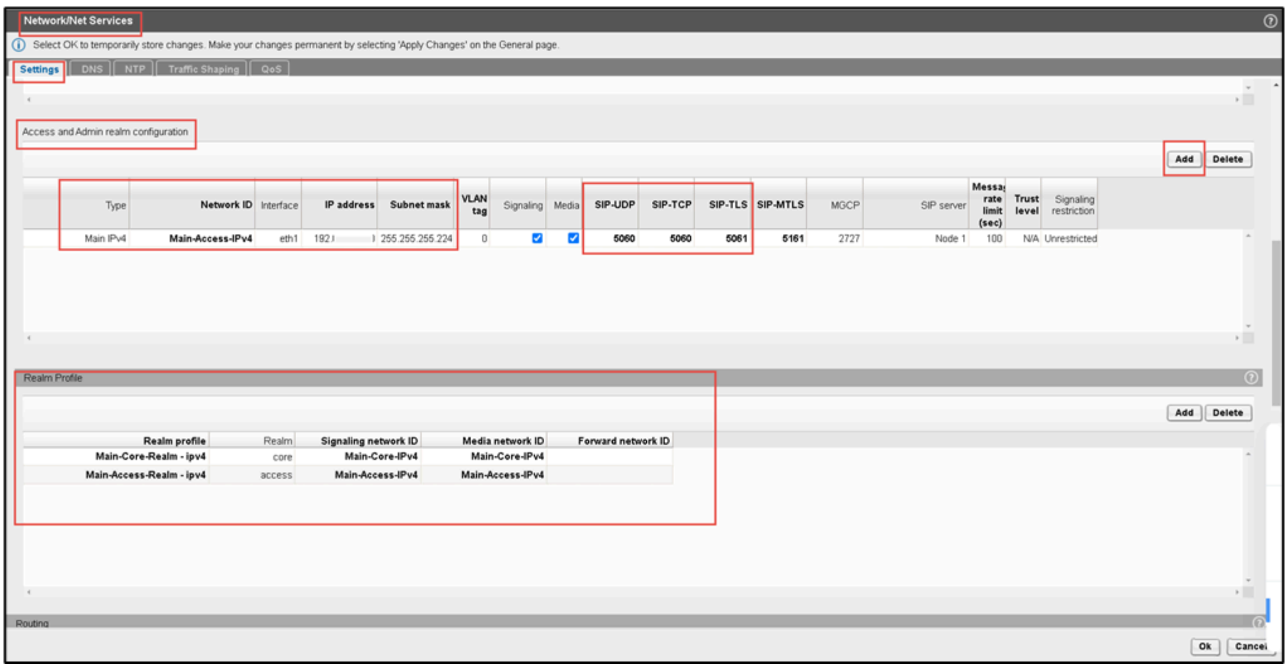
2. In the **Network/Net Services** window, under the **Settings** tab, locate the **Interface Configuration** > **Core Realm Configuration** area and click **Add**. Configure the following:

   a. **IP address**: Enter the SBC IP address.
   b. **Subnet mask:** Enter the subnet mask value.
   c. Select the appropriate interface for core Realm (for example, **eth0**).
   d. **SIP-UDP:** Configure port number as 5060.
   e. **SIP-TCP:** Configure port number as 5060.
   f. **SIP-TLS:** Configure port number as 5061.
   g. Click **Ok,** then click **Apply Changes** on the SBC Main page.



3. In the **Network/Net Services** window, under the **Settings** tab, locate the **Access and Admin realm configuration** area.
4. Click **Add**. In the **Network/Net Services** pop-up, configure the following:

    a. **Type:** Select Type as Main IPV4.
    b. **Network-ID:** Configure network ID as Main-Access-IPv4.
    c. Select the appropriate Interface for core Realm (for example, **eth1**).
    d. **IP address**: Enter the SBC IP address associated with the public side of the network.
    e. **Subnet mask:** Enter the subnet mask value.
    f. **SIP-UDP:** Configure port number as 5060.
    g. **SIP-TCP:** Configure port number as 5060.
    h. **SIP-TLS:** Configure port number as 5061.
    i. Map the **realm profile** for **core** and **access** interface as shown in the below screenshot.
    j. Click **Ok.**
    k. Click **Apply Changes** on the SBC Main page.

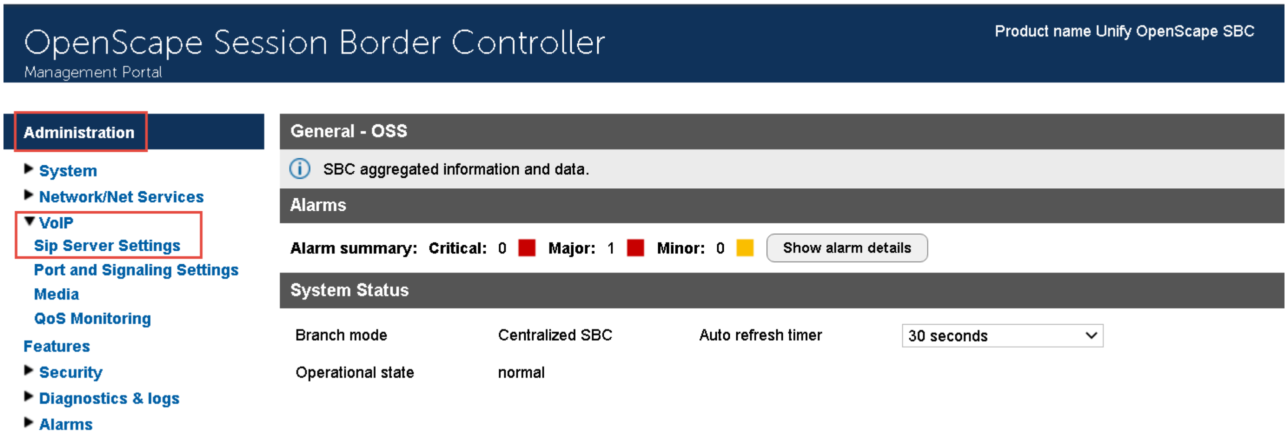


5. In the **Network/Net Services** pop-up, under the **Settings** tab, locate the **Routing** area to configure the default gateway address.
6. In the **Routing Configuration** section, click **Add** and add the static routes for core and access interface.
7. Click **Ok**.
8. Click **Apply Changes**.

6.2 Configuring SIP Server

The SIP connectivity to OpenScape Voice is configured in the **OSSBC Management Portal** > **VOIP** window.

1. Navigate to **Administration** > **VoIP** > **SIP Server Settings**.



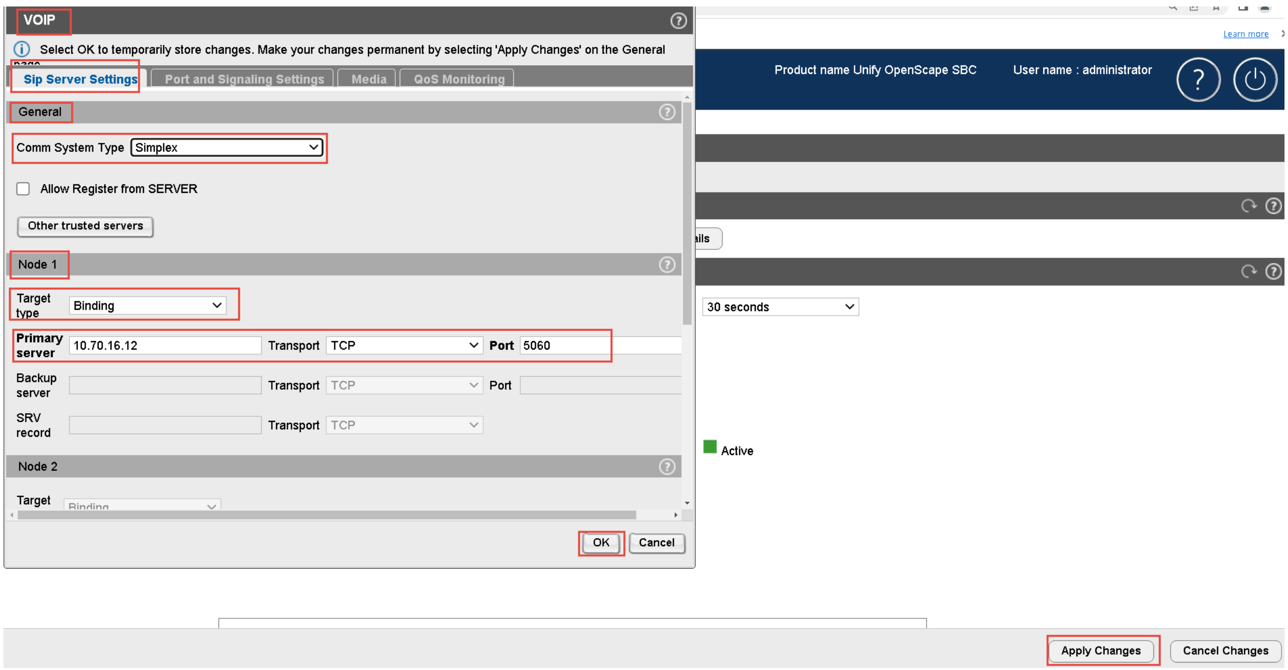
2. In the **Sip Server Settings** tab, enter the following:

   a. Under **General,** from the **Comm System Type** drop-down menu, select **Simplex**.

> **Note**:
> The **Simplex** option is available for OSV deployed as a Single Server . If your OpenScape Voice is deployed as a a Dual-Node (Redundant), select one of the other options based on the OSV deployment: Collocated, Active-Standby, or Clustered.

b. Under the **Node 1** section:

- From the **Target type** drop-down menu, select **Binding.**
- **Primary Server**: Enter the OpenScape Voice **SIP Signaling IP address**.
- **Transport**: **TCP** (for both OS Voice Node 1 and Node 2)
- **Port**: **5060** (listening port for both **OS Voice Node 1** and **Node 2**).



3. Click **OK**.
4. Click **Apply Changes.**

> **Note**:
> The OS Voice SIP Signaling Manager addresses for UDP/TCP/TLS can be found in OS Voice node's **node.cfg** file located in /etc/hiq8000 folder (parameters "sipsm1_vip" for **OS Voice Node1** and "sipsm2_vip" for **OS Voice**). Alternatively, the OS Voice SIPSM IP addresses can be found from CMP.

6.3 Configuring Certificates

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

The certificate must have the SBC FQDN as the subject field's common name (CN). Certificates with a wildcard in the certificate's **Subject Alternate Name** field, conforming to RFC2818, are also supported.

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 OpenScape SBC 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 OSSBC (for example, "certificate.pem").
- OSSBC server certificate private key used for the CSR to CA (for example, "privatekey.pem").

The files mentionedabove must be uploaded to OpenScape SBC for the TLS connection with the Zoom Phone System interface.

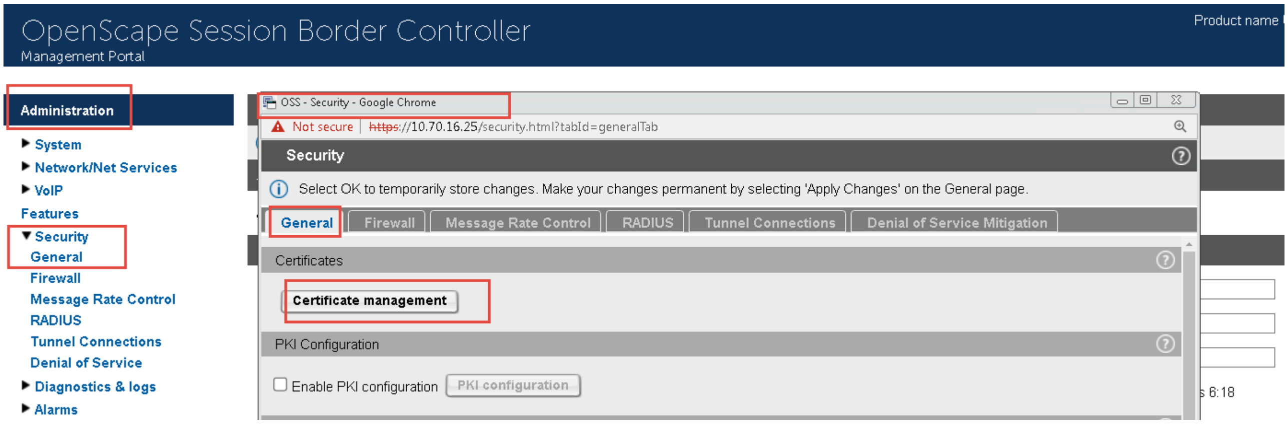
Prerequisite

1. Adequate administrative permissions.
2. Adequate knowledge of TLS certificate handling.
3. At least one OpenScape SBC is configured and in operation.
4. The connection to the OpenScape Voice system is up.

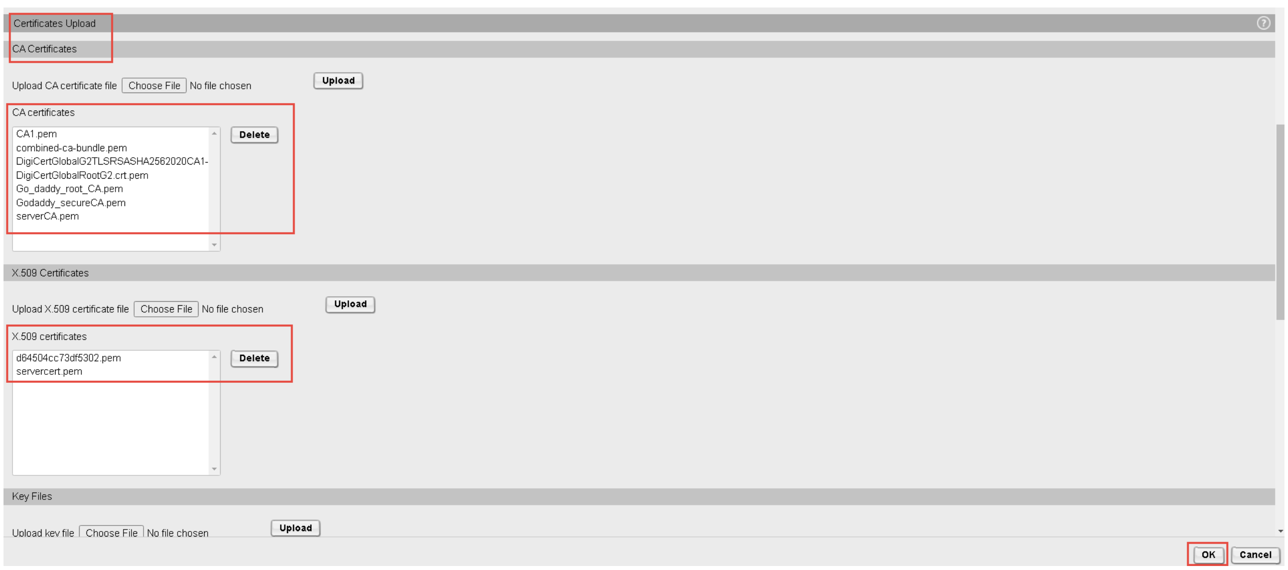
1. Navigate to OpenScape SBC **Management Portal** > **Security** > **General.**

2. In the **Security** pop-up, under the **Certificates** section, click **Certificate Management**.

   The **Certificate Management** window appears with the **General Configuration** tab displayed as default.

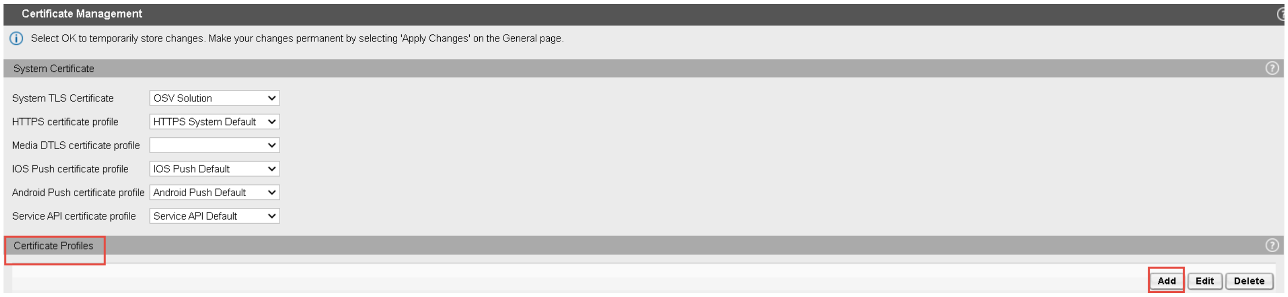


3. Under the **CA Certificate** area, click **Choose File** and browse to select the **CA certificates**. Click **Upload**.
4. Under the X.509 Certificate area, click **Choose File** and browse to select the **X.509** certificates. Click **Upload**.



5. Under the **Key Files** section, click **Choose File** and browse to select the OSSBC server certificate private key. Click **Upload**.

6. To create the Zoom certificate profile: In the **Certificate Management** pop-up, under the **Certificate profiles** area, click **Add**.

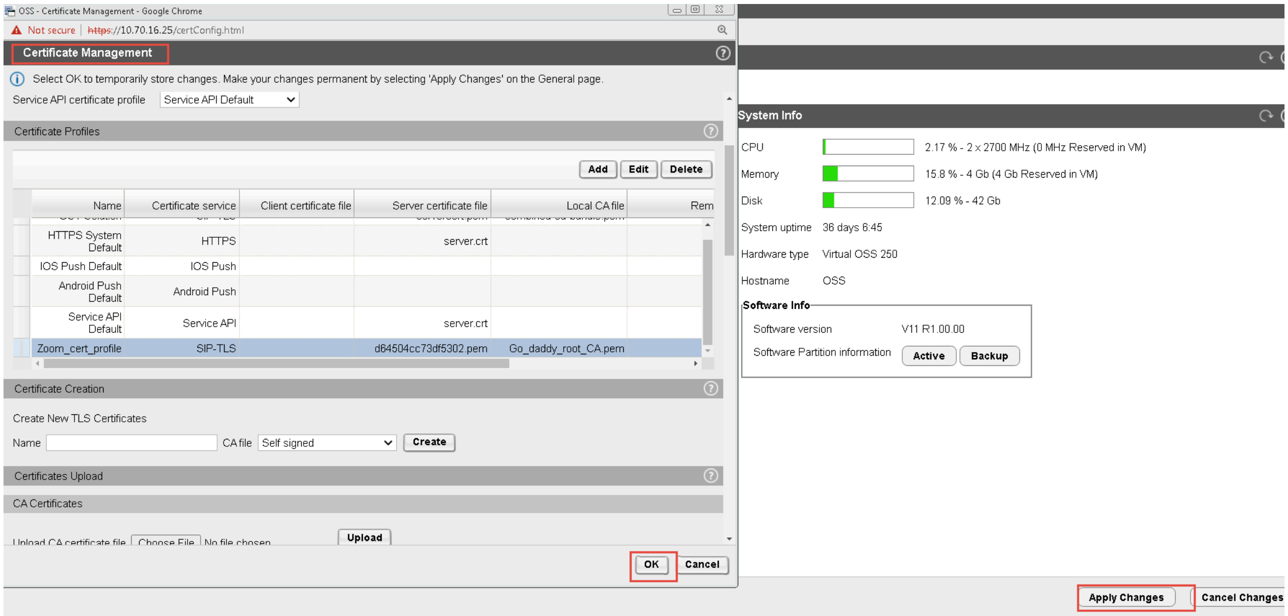


7. Configure the following parameters:

   a. **Certificate profile name**: Enter the name of the Zoom certificate profile.
   b. From the **Certificate Service** drop-down menu, select **SIP-TLS**.
   c. From the **Local server certificate file** drop-down menu, select the certificate to be used when establishing a TLS connection as a server.
   d. From **the Local CA file** drop-down menu, select the CA certificate.
   e. From the **Local key file** drop-down menu, select the key file that contains the private key.
   f. From the **TLS version** drop-down menu, select **TLS1.2**.

8. Click **OK**.
9. Click **OK** in the **Certificate Management** window and in the **Security** window.

10. Click **Apply Changes** on the OpenScape SBC main page.



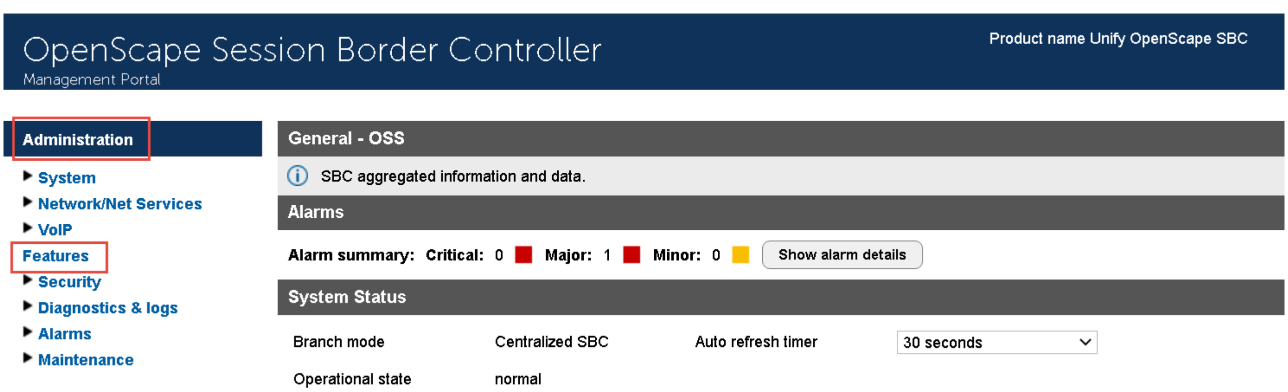
6.4 Configuring Media Profiles

In the **Media Profiles** settings, various SDP messages and audio (RTP) traffic parameters can be configured for the OpenScape SBC SIP endpoints to Zoom Phone System, SSP (PSTN provider), and Unify OpenScape Voice.

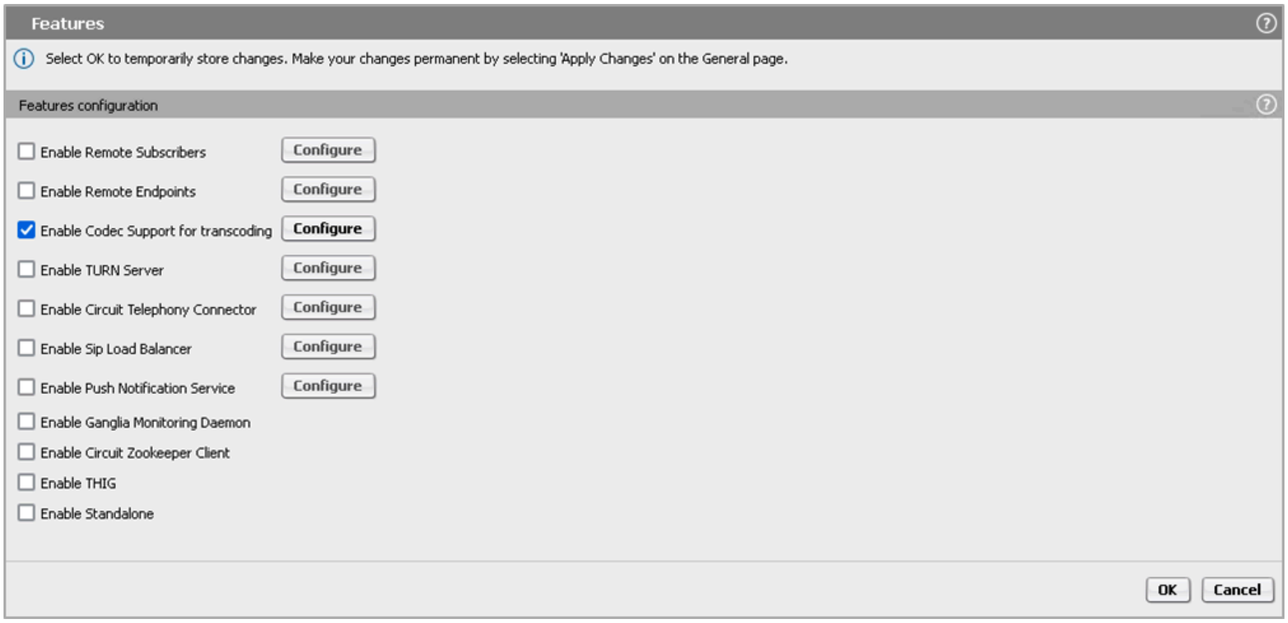
6.4.1 Configuring the Codec Manipulation Options

In case transcoding or certain codec prioritization for audio is required for the OSSBC – Zoom Phone System and OSSBC – SSP media profiles for the corresponding SIP trunks, it is required to enable the codec configuration options first for the media profile setup.

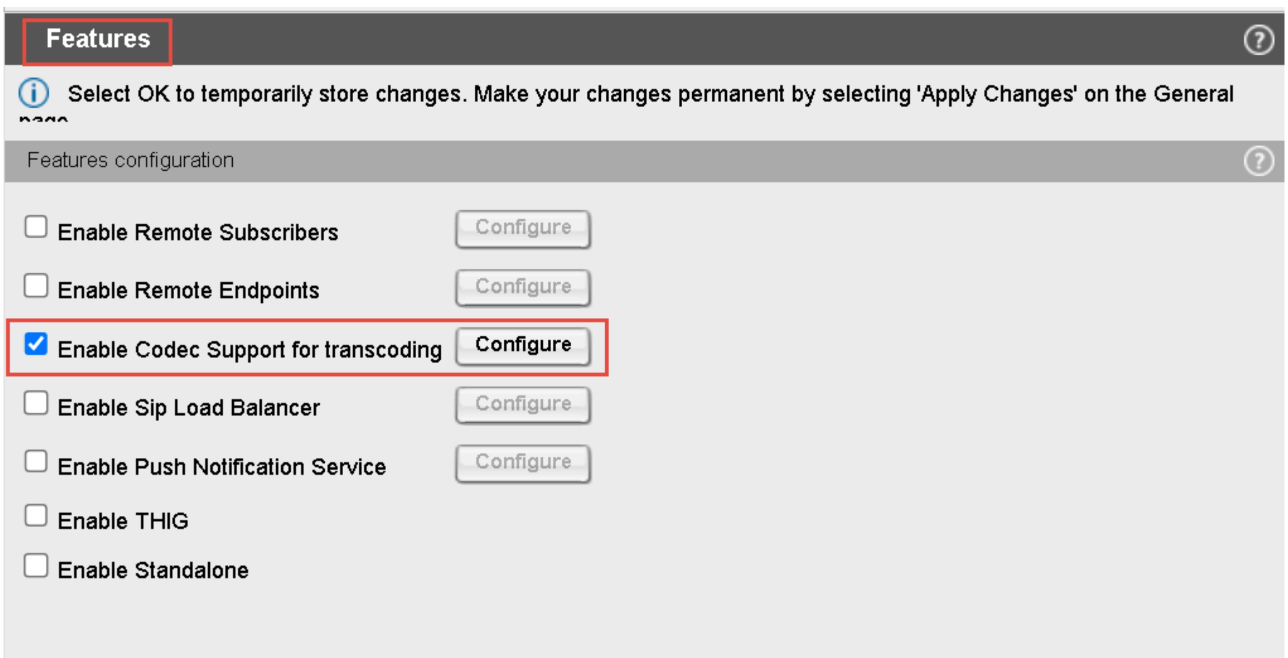
1. Navigate to the **OpenScape SBC Management Portal** > **Features** window.



2. Check the **Enable Codec Support for transcoding** checkbox.

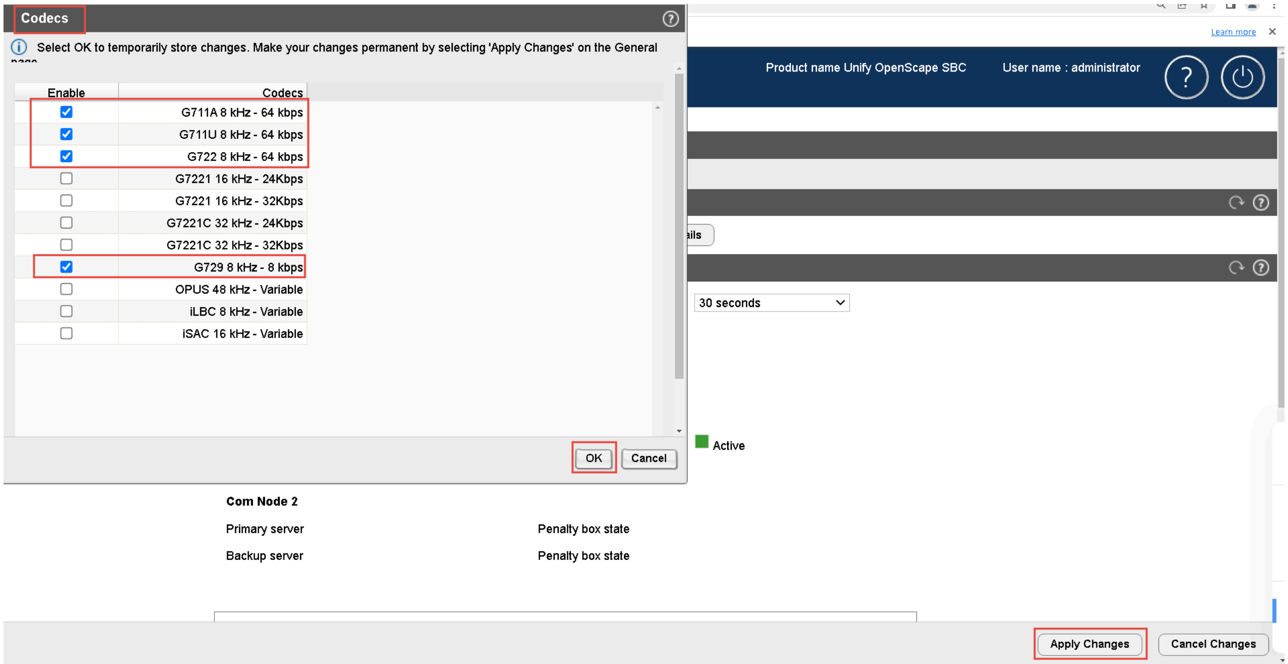


3. In the **Features** pop-up, check the **Enable Codec Support for transcoding** checkbox and click **Configure**.



4. In the **Codecs** window, you can enable the codecs to be available for the media profiles (for example, transcoding and prioritization). Select the following checkboxes:

   a. G711A 8 kHz - 64 kbps
   b. G711U 8 kHz - 64 kbps
   c. G722 8 kHz - 64 kbps
   d. G729 8 kHz - 64 kbps



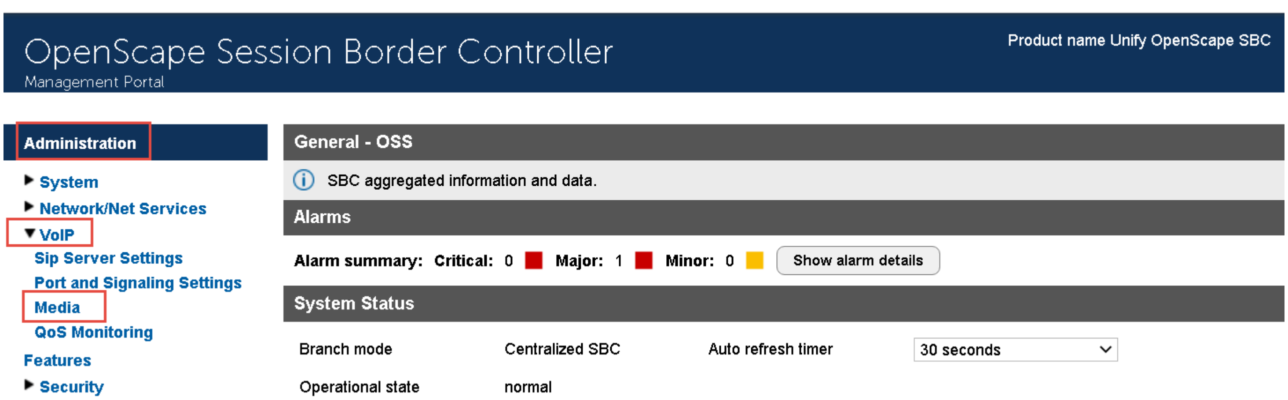
1. Click **OK.**
2. Click **Apply Changes**.

6.4.2 Configuring the Zoom Media Profile

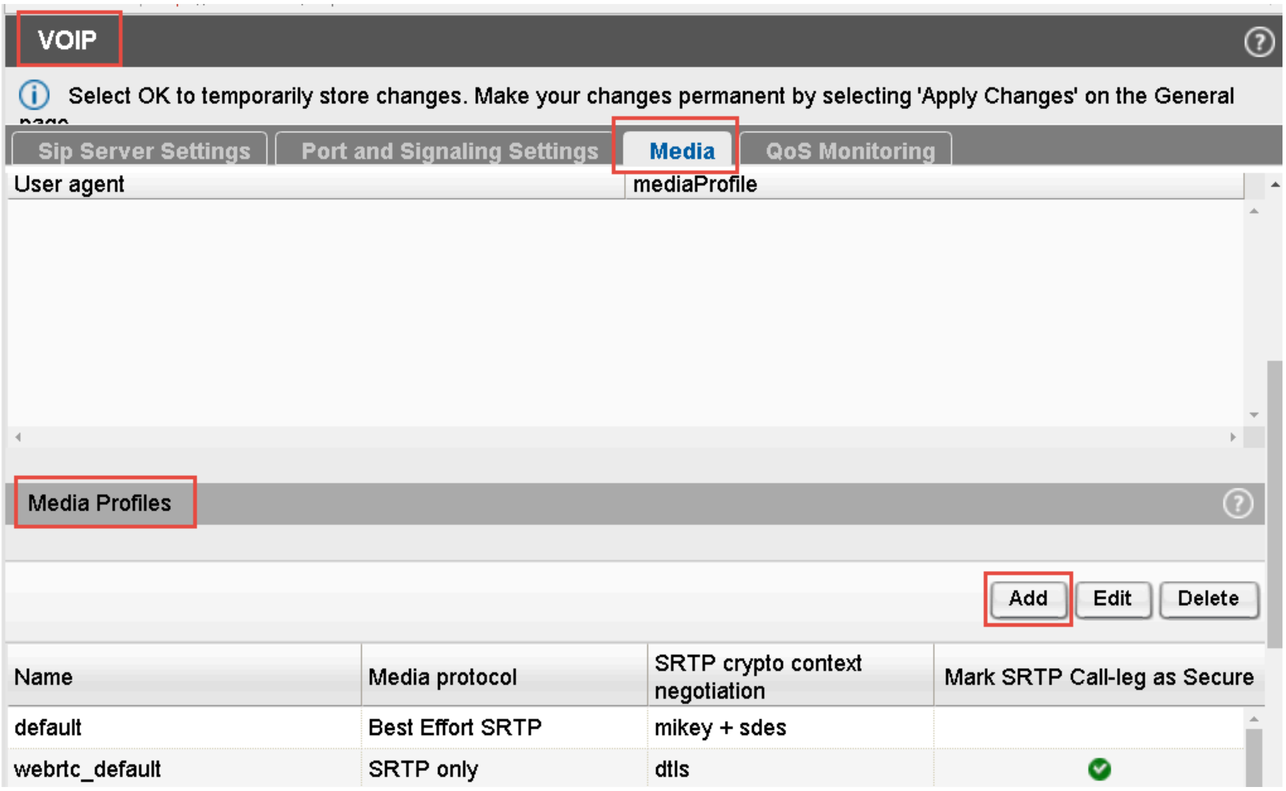
The communication between the SBC and the Zoom Phone System is secured with SRTP.

In the example presented in the current sub-section, the PSTN is supposed to not support G.711, and transcoding to G.711 is required for calls between PSTN subscribers and Zoom clients on OSSBC —Zoom Phone System SIP trunks.

1. **Navigate** to **OSSBC Management Portal** > **VOIP** > **Media**.



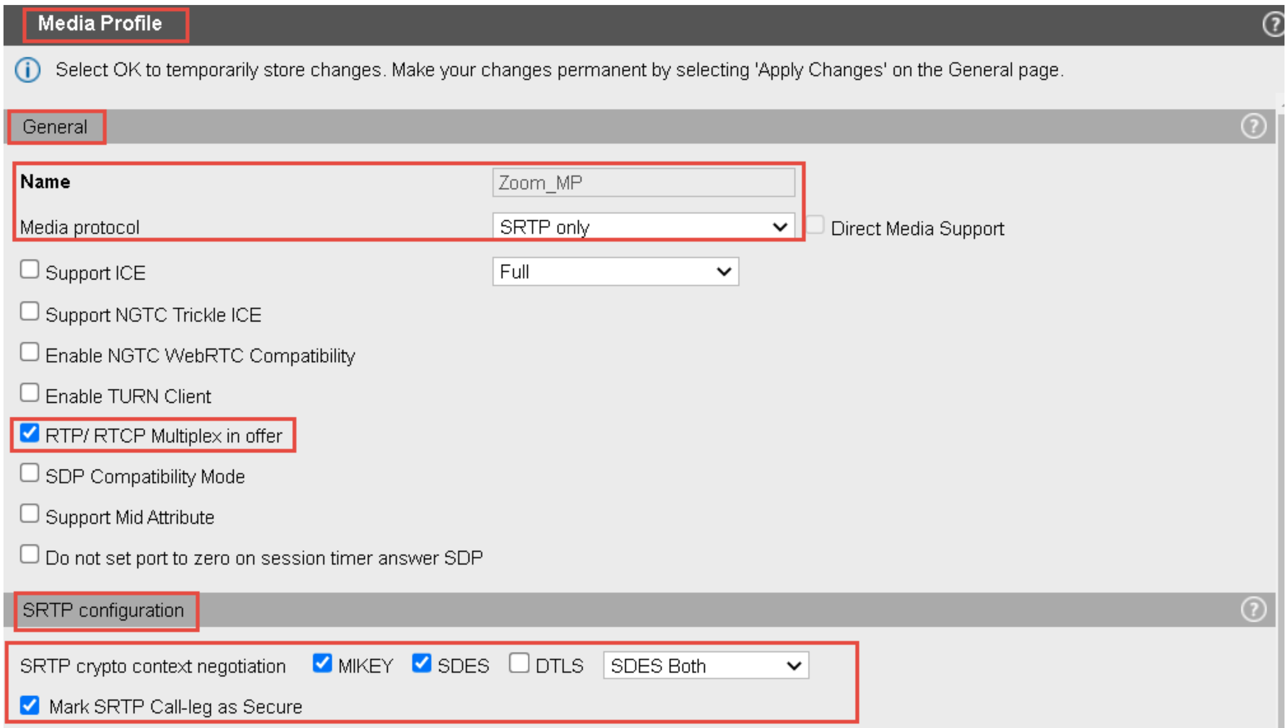
2. In the **VOIP** pop-up, go to the **Media** tab.



3. Locate the **Media Profiles** area and click **Add.**

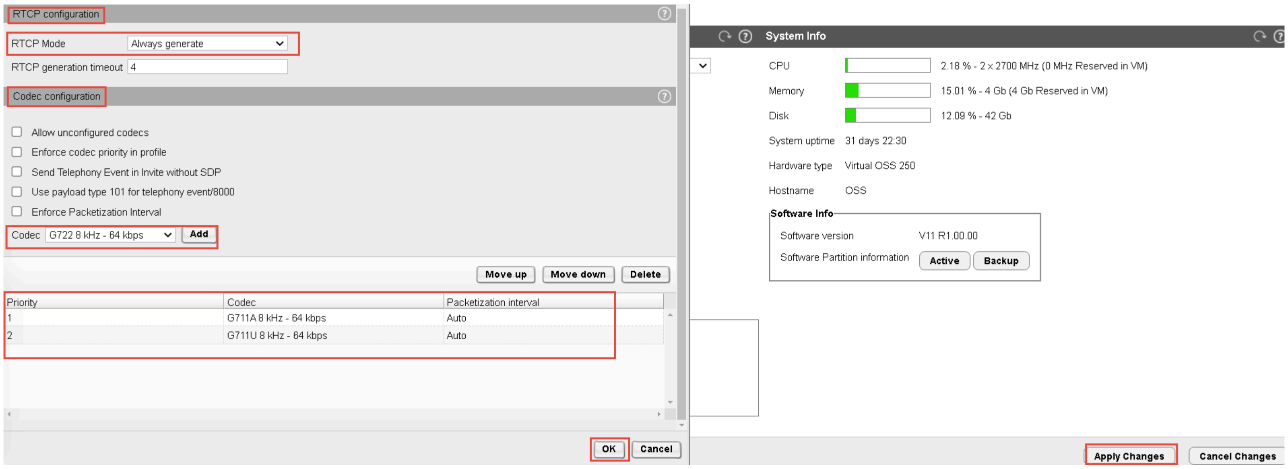
4. In the **Media Profile** pop-up**,** under the **General** area**,** create the media profile for OSSBC - Zoom connections by entering the following:

    - **Name:** Type the media profile name**.** For example, Zoom_MP.
    - From the **Media protocol** drop-down menu, select **SRTP only.**
    - Check the **RTP/RTCP Multiplex in offer** checkbox.
    - Under the **SRTP configuration** area, check the following checkboxes:
        - **SDES**
        - **MIKEY**
        - **Mark SRTP Call-leg as Secure**



5. Under the **RTCP configuration** area, from the **RTCP Mode** drop-down menu, select **Generate Always**.

6. Under the **Codec Configuration** area, select the required codecs and click **Add** to add them for transcoding (with priority). For example G711A, and G711U, as shown below:



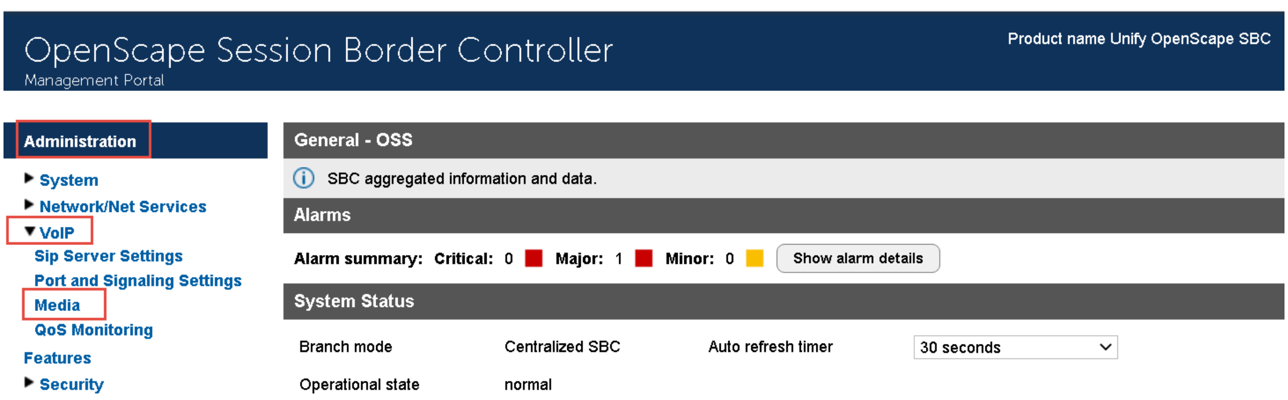
7. Click **OK** to return to the **Media** window.
8. Click **OK** on the **VoIP** window.
9. Click **Apply Changes.**

6.4.3 Configuring the PSTN Media Profile

> **Note**:
> The configuration below is an example. The actual configuration steps depend on your provider's requirements.

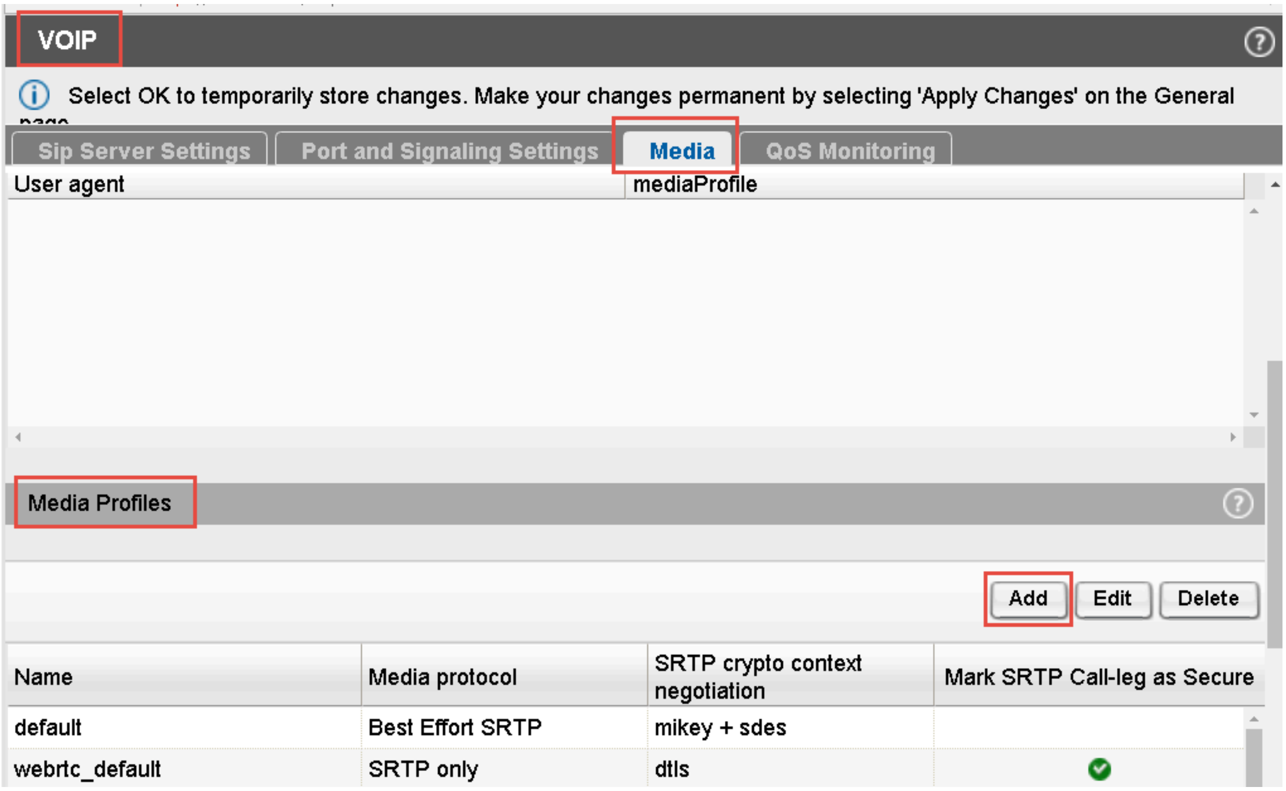
In the current sub-section, as an example, it is supposed that for calls between Zoom clients and PSTN subscribers, certain codecs need to be prioritized on OSSBC – SSP (BCOM) SIP trunk.

1. Navigate to the **Unify OpenSape SBC Management Portal** > **VOIP** > **Media** window.



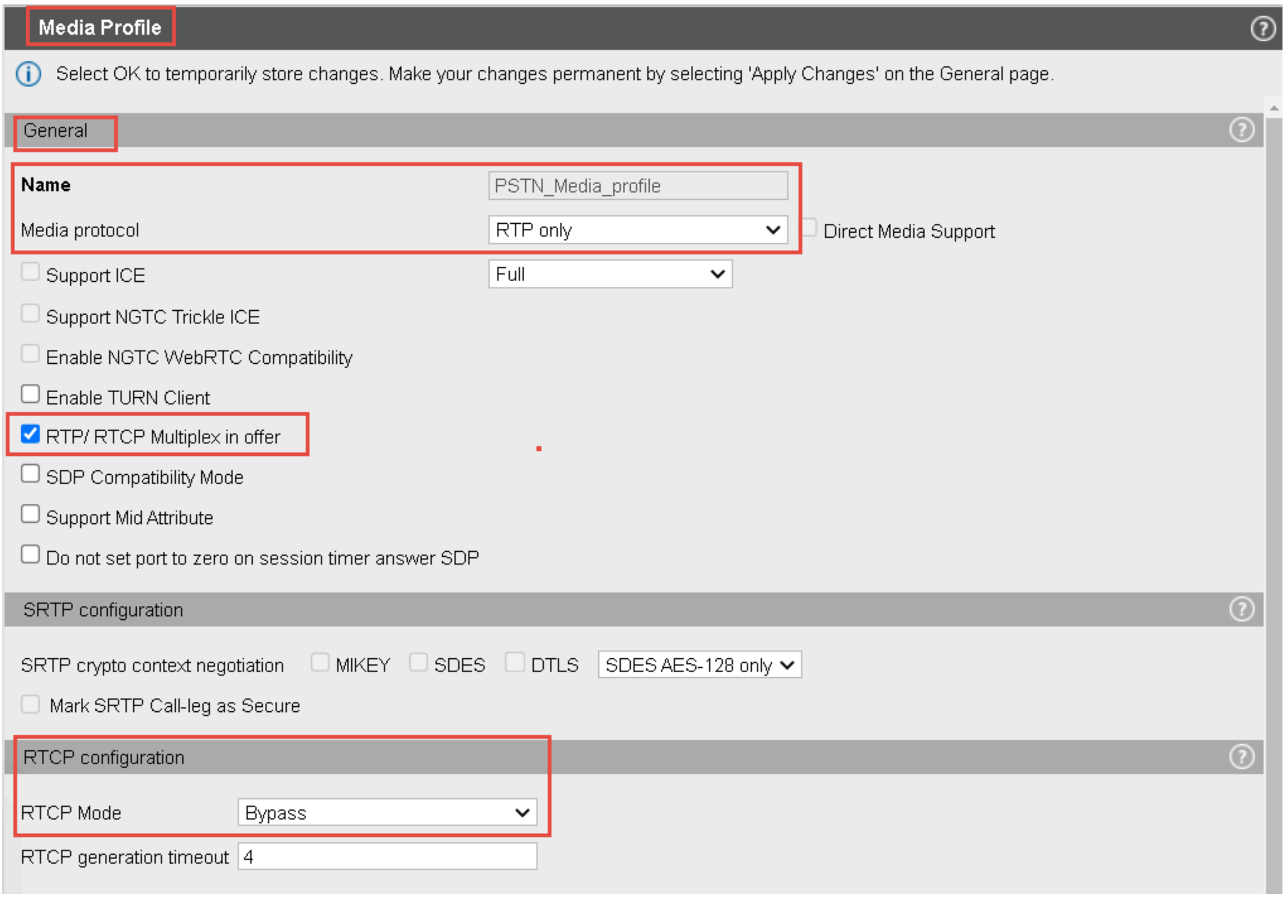
2. In the **VOIP** pop-up, go to the **Media** tab.

3. Locate the **Media Profiles** area and click **Add** to create the media profile for OSSBC to PSTN service provider trunk.

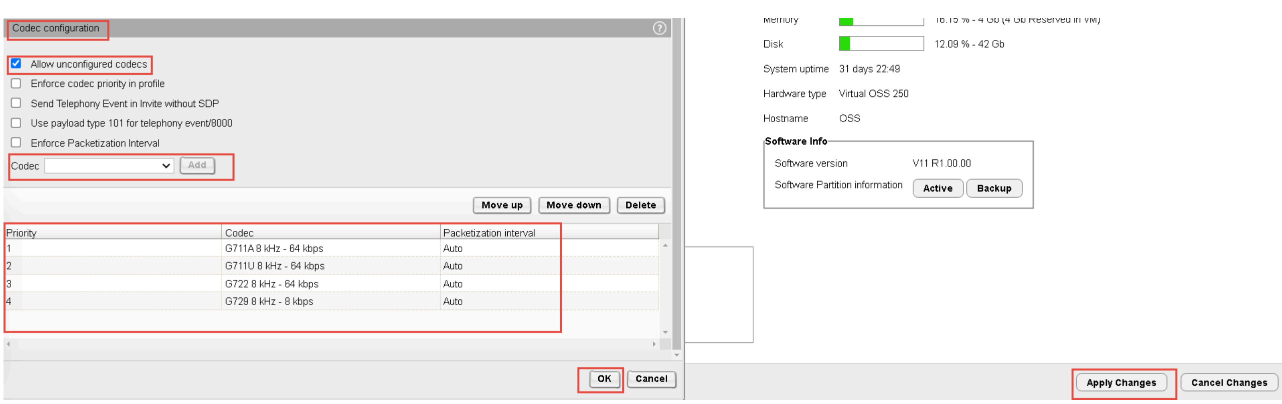


4. In the **Media profile** pop-up, locate the **General** section and configure the following:

- **Name**: Enter the name of the media profile.
- From the **Media protocol** drop-down menu, select **RTP only**.
- Check the **RTP/RTCP Multiplex in offer** checkbox.
- Under the RTCP configuration area, from the **RTCP Mode** drop-down menu, select **Bypass.**

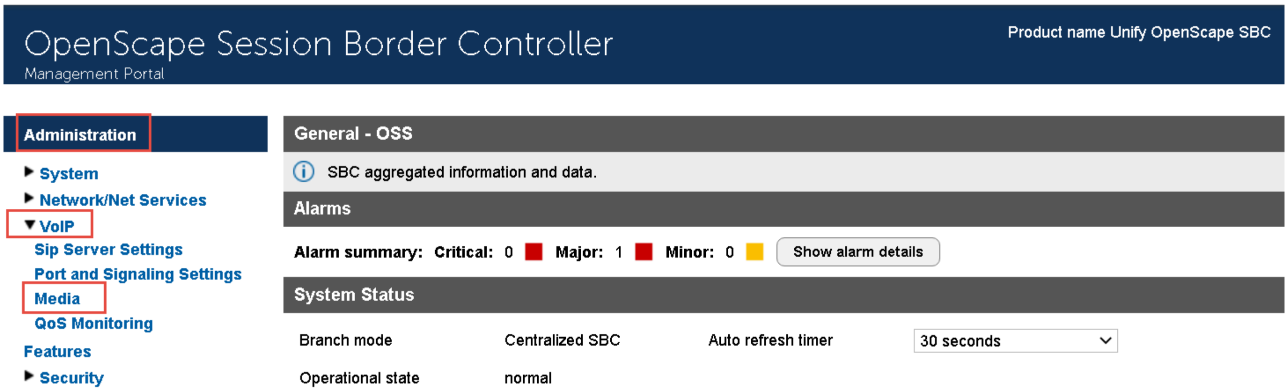


5. In the **Codec configuration**, select the **Allow unconfigured codecs** option, then select the required codecs from the drop-down menu. Click **Add** to add them.
6. Click **OK.**
7. Click **Apply Changes** on the SBC main page.

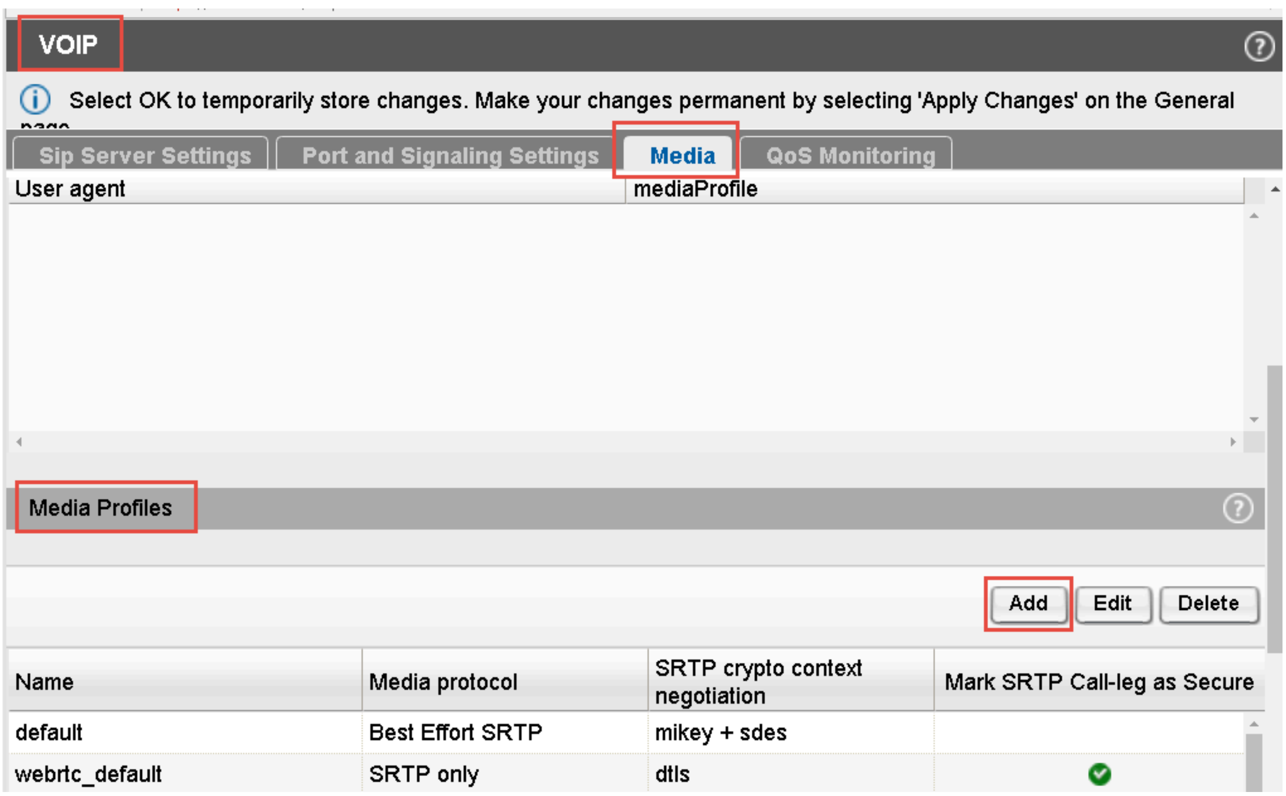


6.4.4 Configuring the Unify OpenScape Voice Media Profile

1. Navigate to the **OSSBC Management Portal** > **VOIP** > **Media** window.



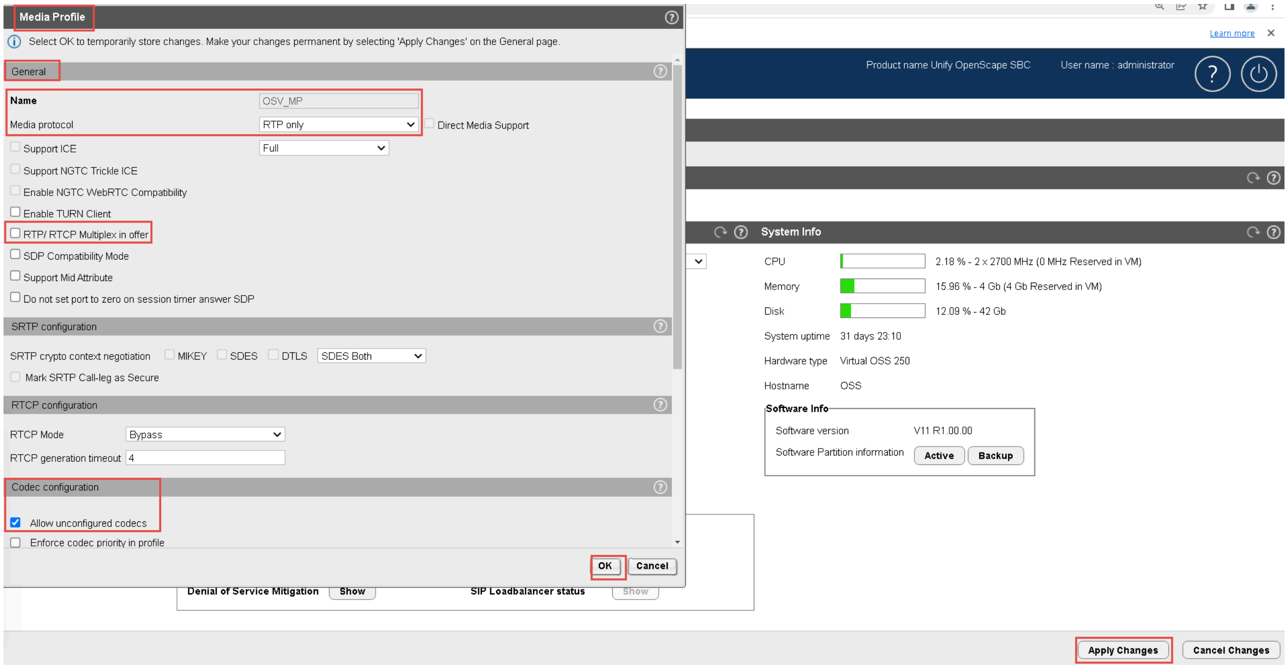
2. In the **VOIP** pop-up, go to the **Media** tab.
3. In the **Media Profiles** area, click **Add** to create the media profile for OSSBC – OS Voice connection.



4. In the **Media profile** pop-up, locate the **General** section and configure the following:

- **Name**: Enter the name of the media profile.
- From the **Media protocol** drop-down menu, select **RTP only**.
- Check the **RTP/RTCP Multiplex in offer** checkbox.

5. In the **Codec configuration**, select the **Allow unconfigured codecs** option, then select the required codecs from the drop-down menu. Click **Add** to add them.

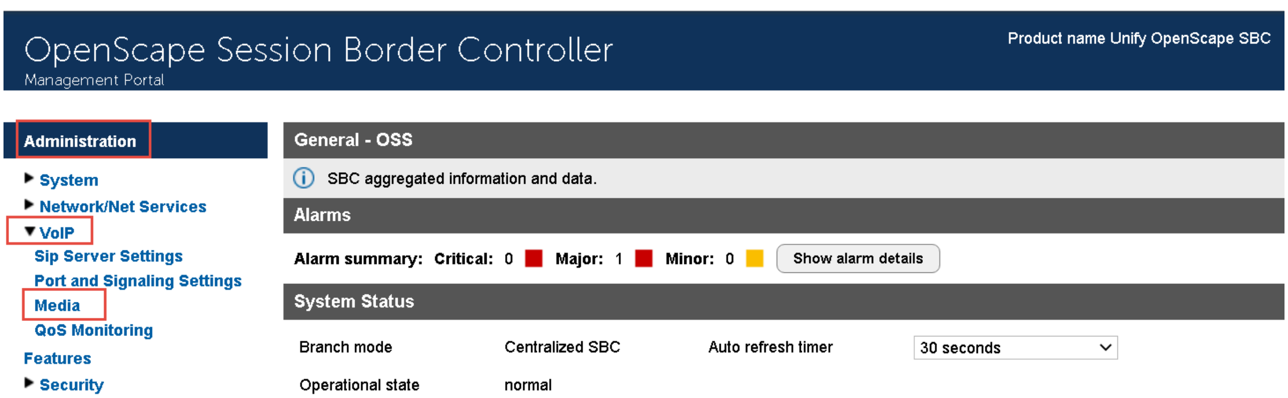


6. Click **OK.**
7. Click **Apply Changes** on the SBC main page.

6.4.5 General Media Settings

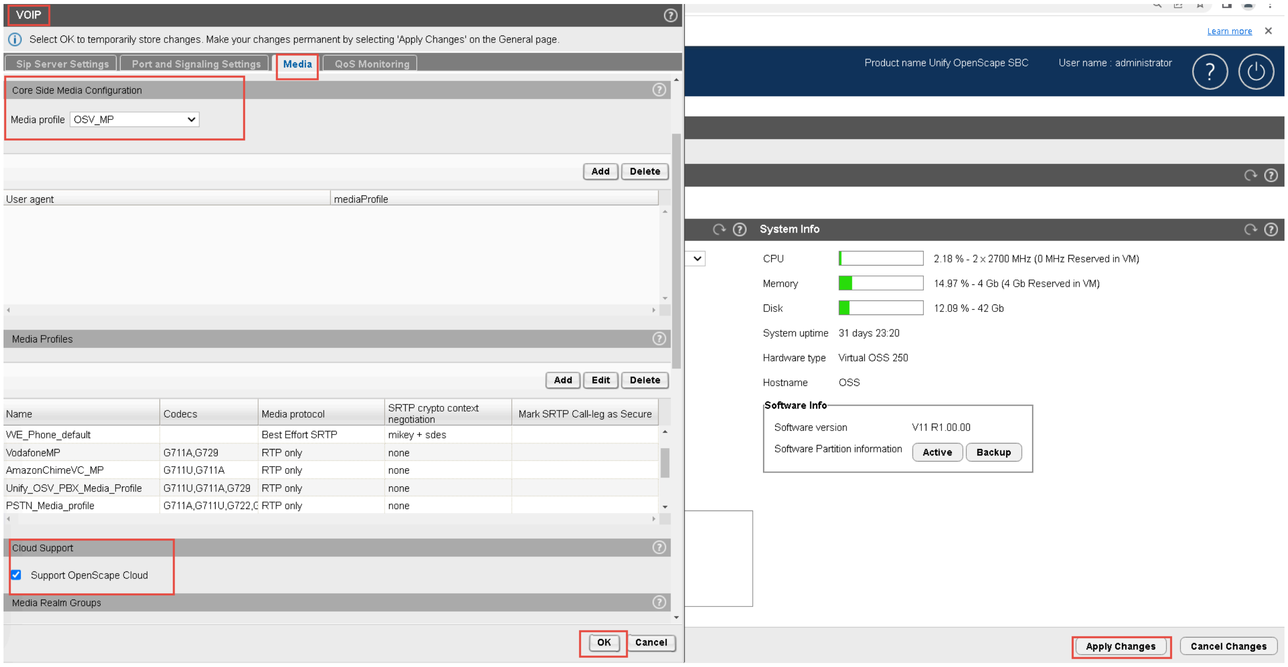
After creating the media profiles, configure the General media settings.

1. Navigate to the **OSSBC Management Portal** > **VOIP** > **Media** window.



2. In the **VOIP** pop-up, go to the **Media** tab.
3. Locate the **Core Side Media Configuration** area and select the previously created OpenScape Voice media profile from the **Media Profile** drop-down menu, used for the OSSBC – OS Voice SIP trunk.

4. Check the **Support OpenScape Cloud** checkbox to enable this option.



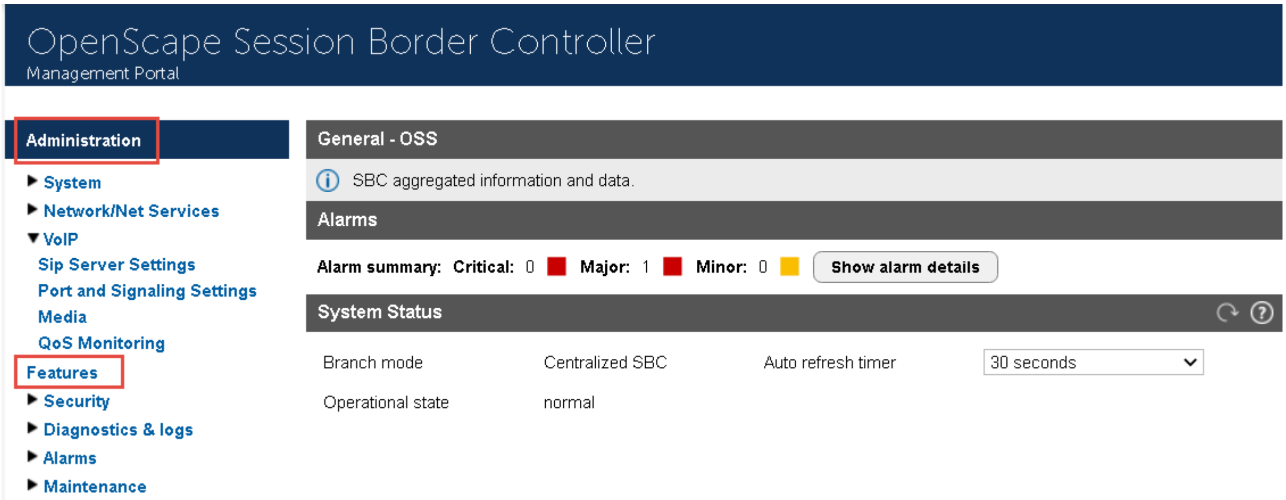
5. Click **OK** and then click **Apply Changes** on the SBC main page.
6. Click **OK**.
7. Click **Apply Changes.**

6.5 Configuring Remote Endpoints

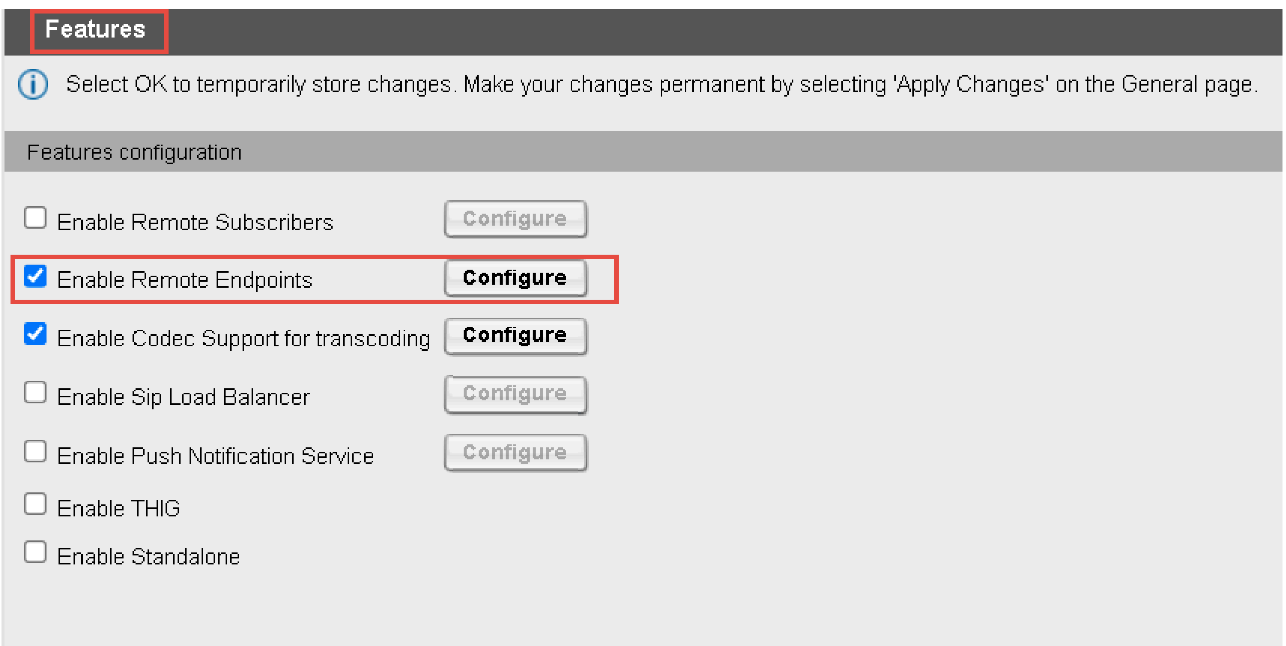
In the **Remote Endpoint** configuration, you can set up the OpenScape SBC with Zoom Phone System and the PSTN (BCOM SSP) SIP trunks.

6.5.1 Configuring the Zoom Remote Endpoints

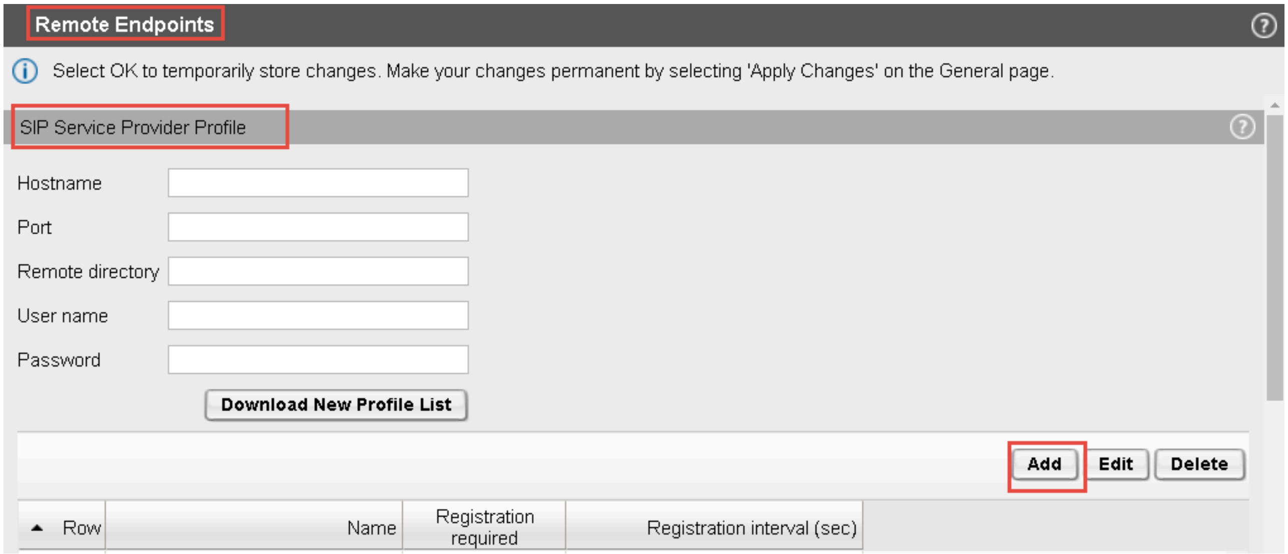
1. Navigate to the **Administration > Features** window.



2. In the **Features** pop-up, check the **Enable Remote Endpoints** checkbox and click **Configure**.

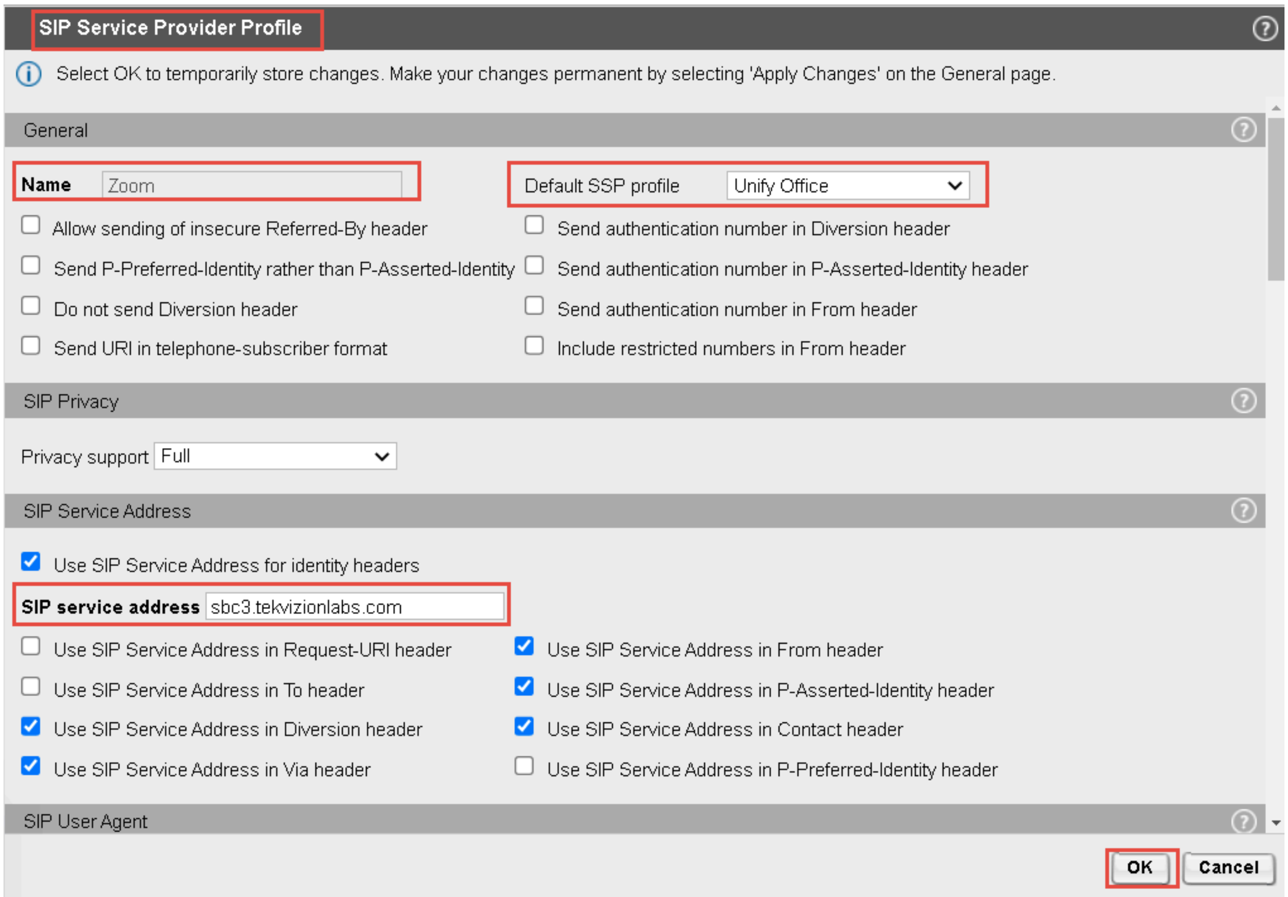


3. In the **"Remote Endpoints"** pop-up, locate the **"SIP Service Provider Profile"** area and click **Add** to add the endpoint profile for the OSSBC – Zoom Phone System endpoint.

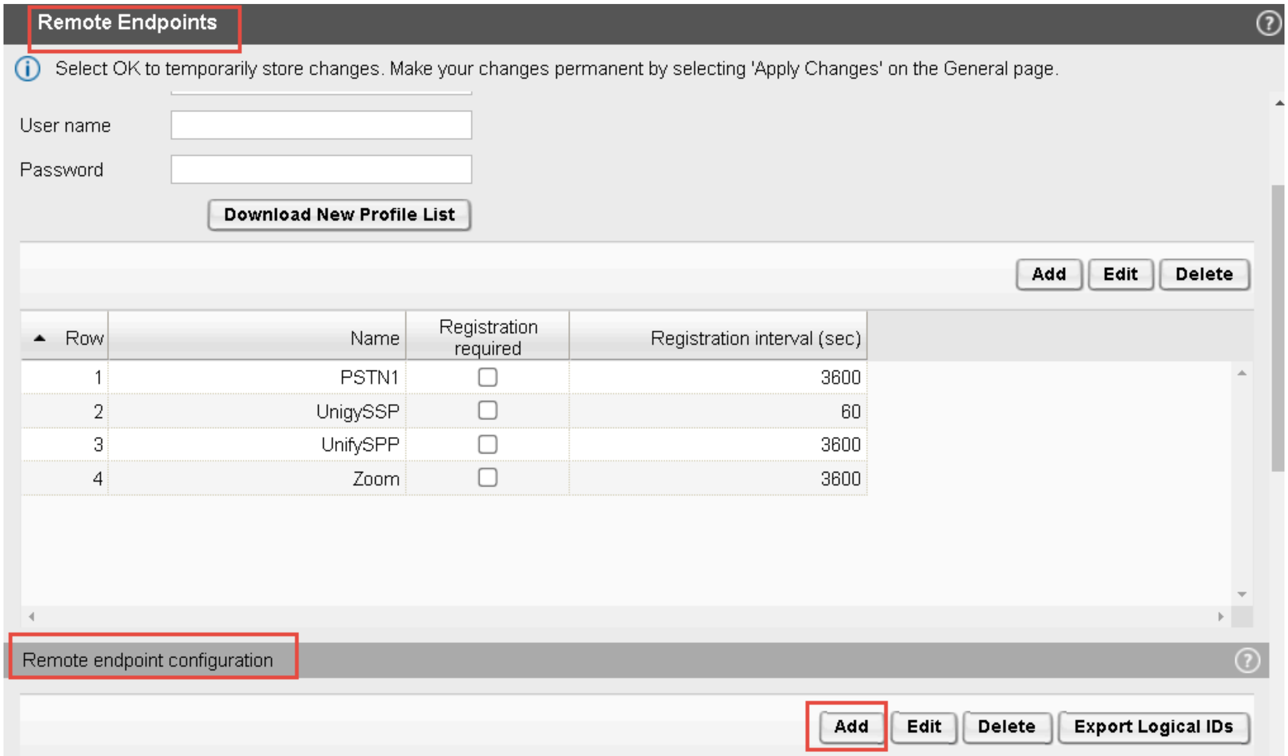


4. In the **SIP Service Provider** pop-up, configure the following:

   a. **Name**: Enter the name of the SIP Service Provider profile. For example, Zoom.
   b. From the **Default SSP Profile drop-down** menu**,** select **Unify Office**.
   c. **SIP service address**: Enter the SBC's public FQDN and click **OK** to return to the **Remote endpoints** window.

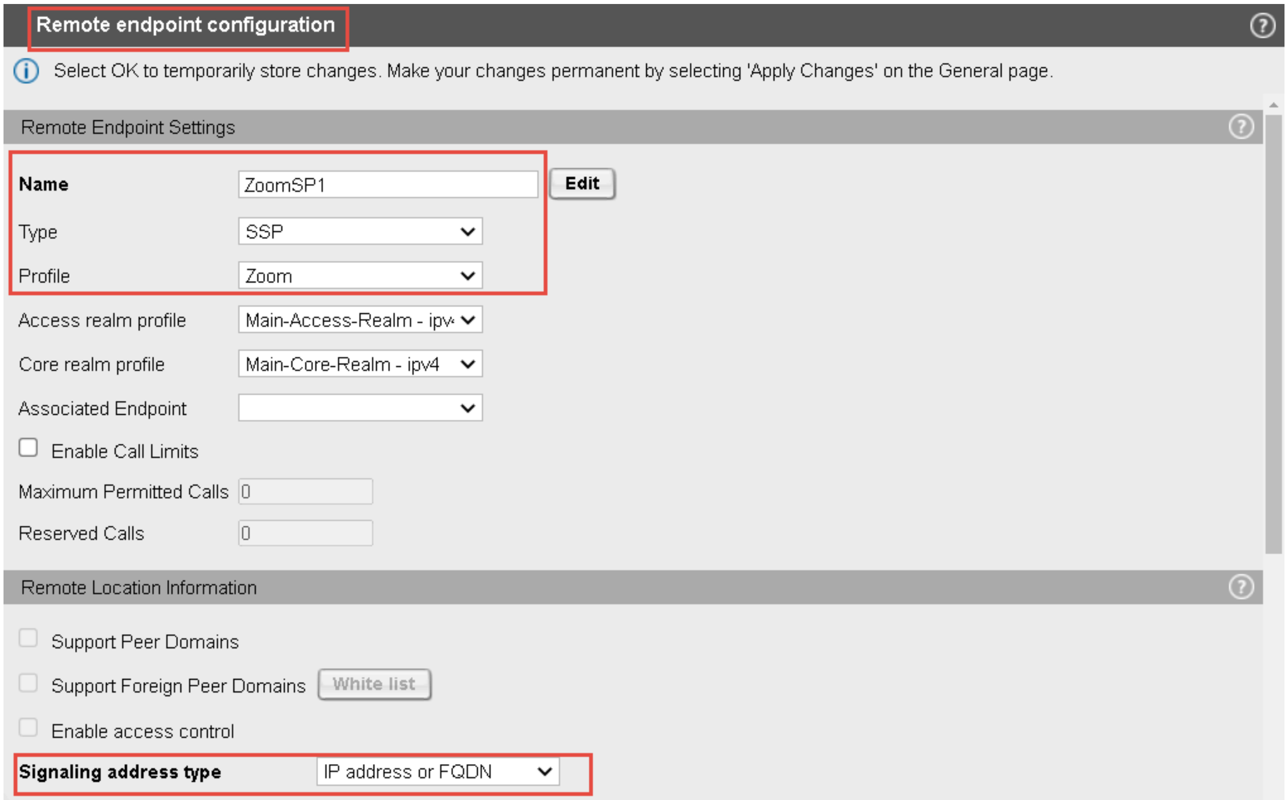


5. In the **Remote endpoints** window, locate the **Remote endpoint configuration** area, and click **Add.**



6. In the **Remote endpoint configuration**pop-up, configure the following:

   a. **Name:** Enter the name of the remote endpoint. For example, ZoomSP1.
   b. From the **Type drop-down** menu**,** select **SSP**.
   c. From the **Profile** drop-down menu, select **Zoom**.
   d. From the **Signaling address** type drop-down menu, select **IP address or FQDN**.

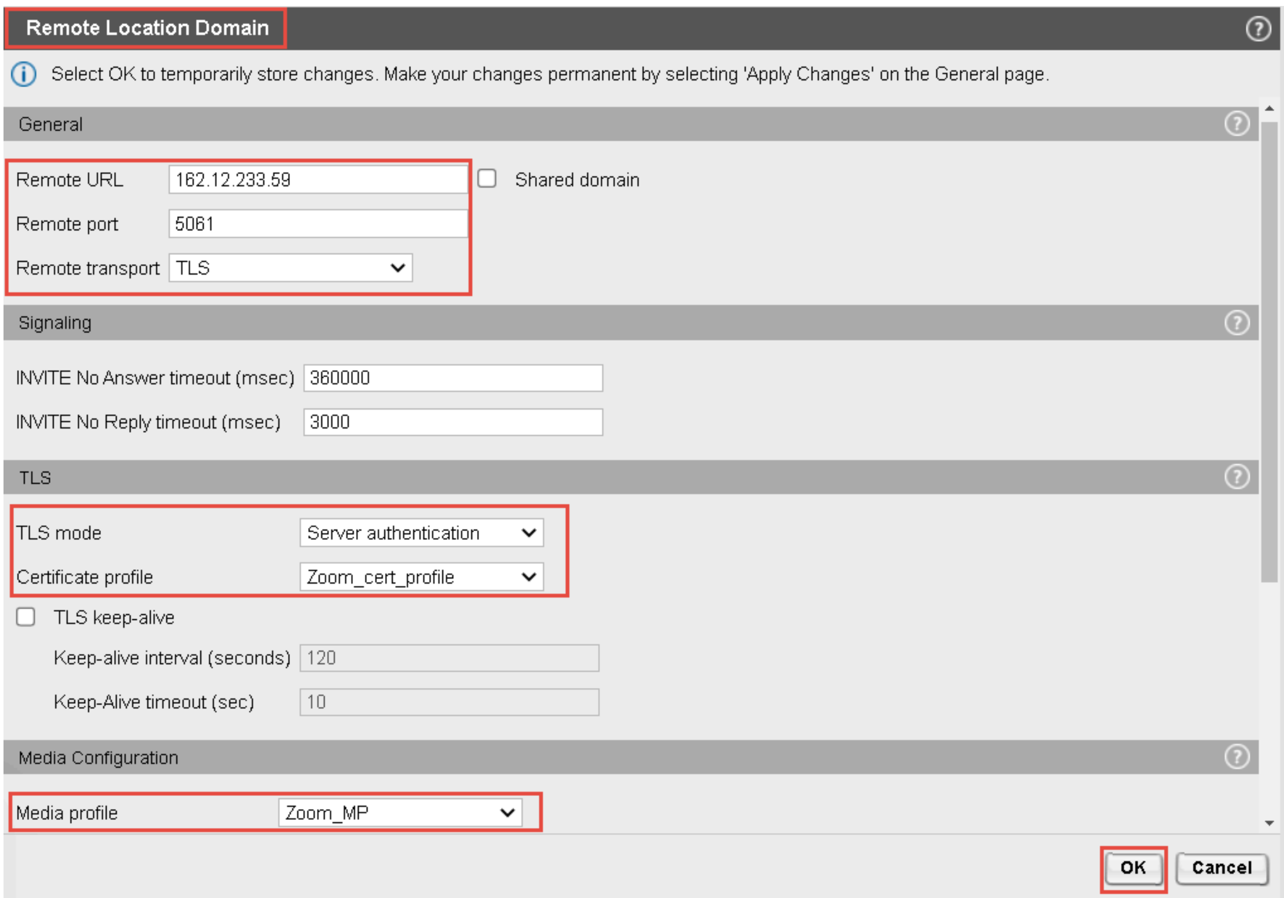


7. Locate the **Remote Location domain** area and click **Add** to add the IP address.

8. In the **Remote Location Domain** window, configure the following:

   a. **Remote URL:** Enter the Zoom IP address (see the Zoom IPs Table under Chapter 3 Unify OpenScape SBC Configuration on page 70).
   b. **Remote port**: Enter the port number (**5061**).
   c. Locate the **TLS** area, and from the **TLS mode** drop-down menu**,** select **Server authentication.**

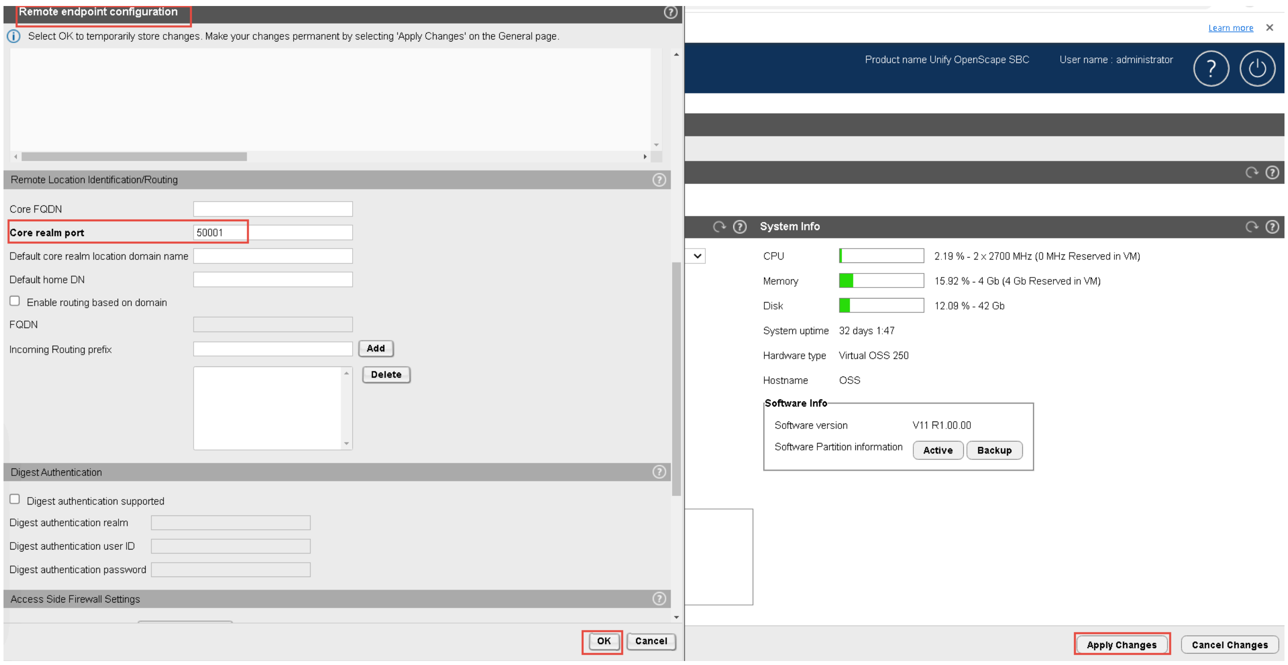
      (or Mutual authentication in case MTLS is required)
   d. From the **Remote transport** drop-down menu, select TLS.
   e. From the **Certificate profile** drop-down menu, select **Zoom_Cert_Profile.**
   f. Locate the **Media Configuration** area, and from the **Media profile** drop-down menu, select the **Zoom** media profile.



9. Click **OK**.
10. In the **Remote endpoint configuration** window, locate the **Remote Location Identification Routing** area.
11. In the **Core realm port** field, enter the core realm value as **50001**.

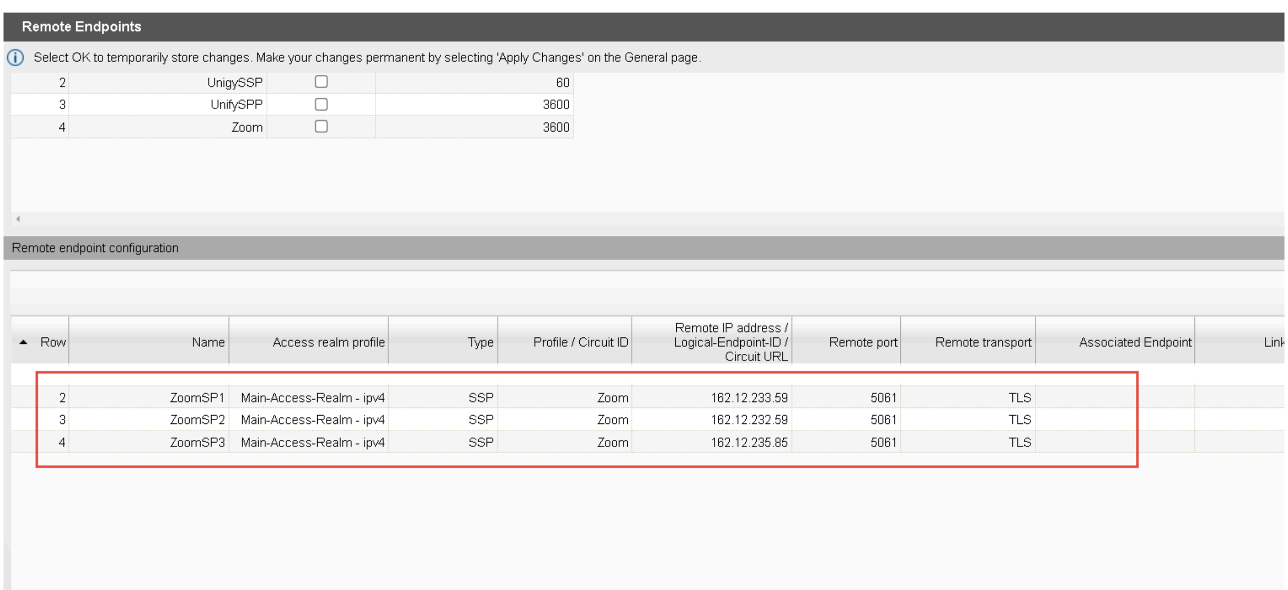
Important:

The value for each Endpoint of Zoom should be unique. Add **50002** for the second Zoom endpoint, **50003** for the third, and so on.



12. Repeat the configurations in the **Remote endpoint configuration** window for the remaining Zoom IPs.
13. Click **OK**.
14. Click **Apply changes**.

The **Remote Endpoints** window should look like the figure below:



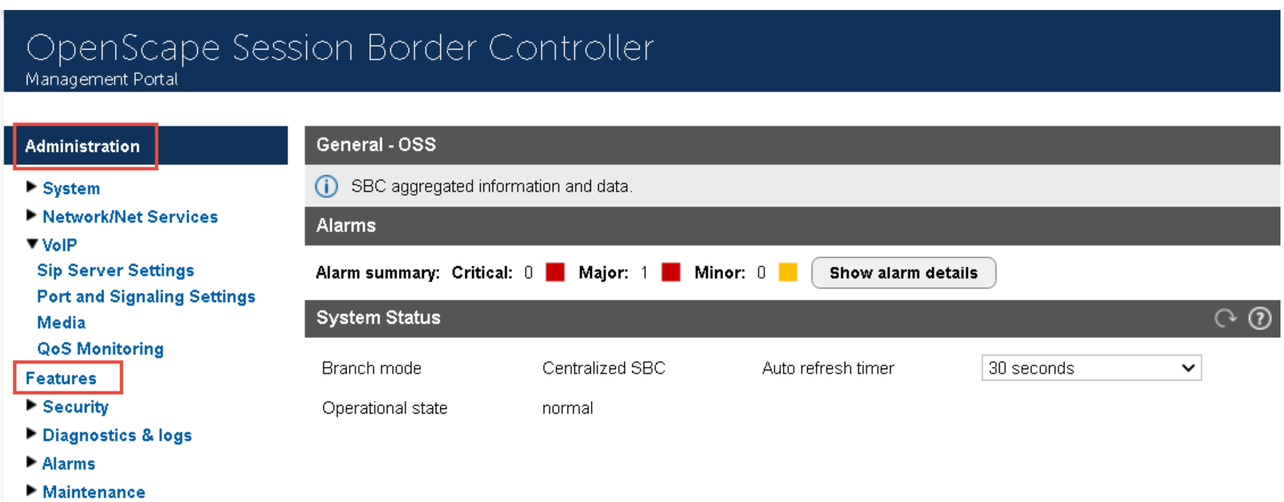
> **Note**:
> See the Tables in Unify OpenScape SBC Configuration on page 70

> **Note**:
> Please refer to the Signaling Traffic table under the Premises Peering Firewall Requirements for Media and Signaling section in the **Zoom Phone Bring Your Own Carrier- Premises (BYOC-P) Solution Reference Guide**.

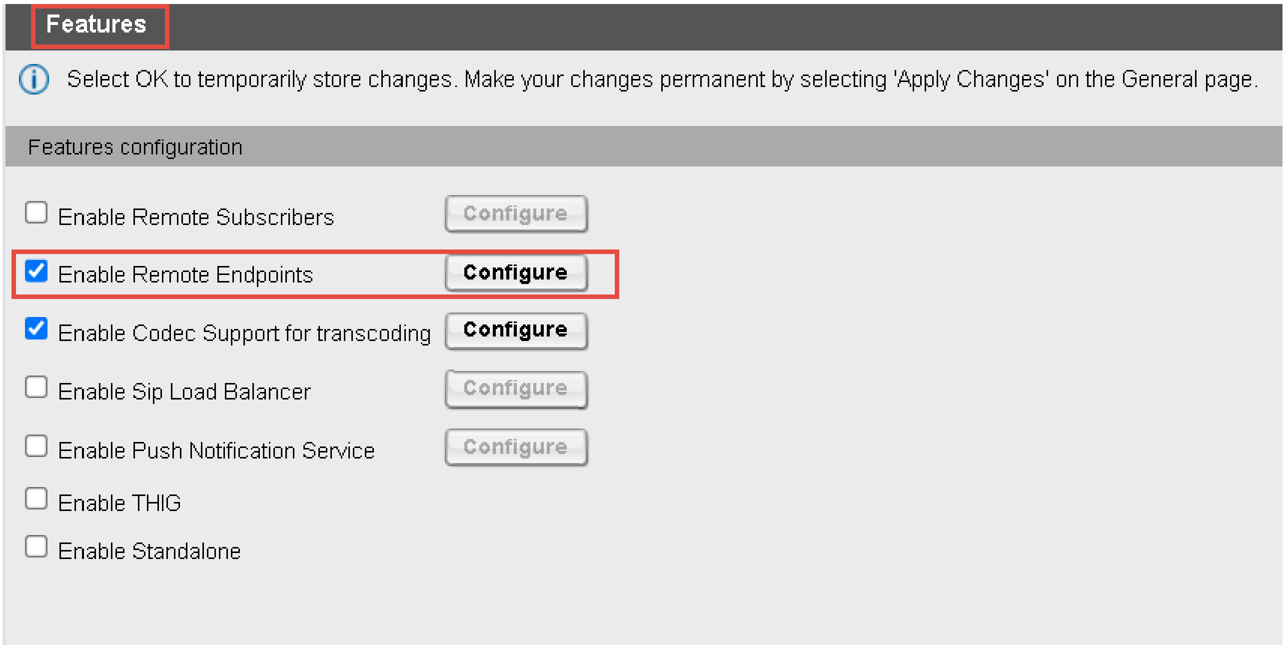
6.5.2 Configuring the PSTN Remote Endpoint

> **Note**:
> The configuration below is an example. The actual configuration steps depend on your provider's requirements.

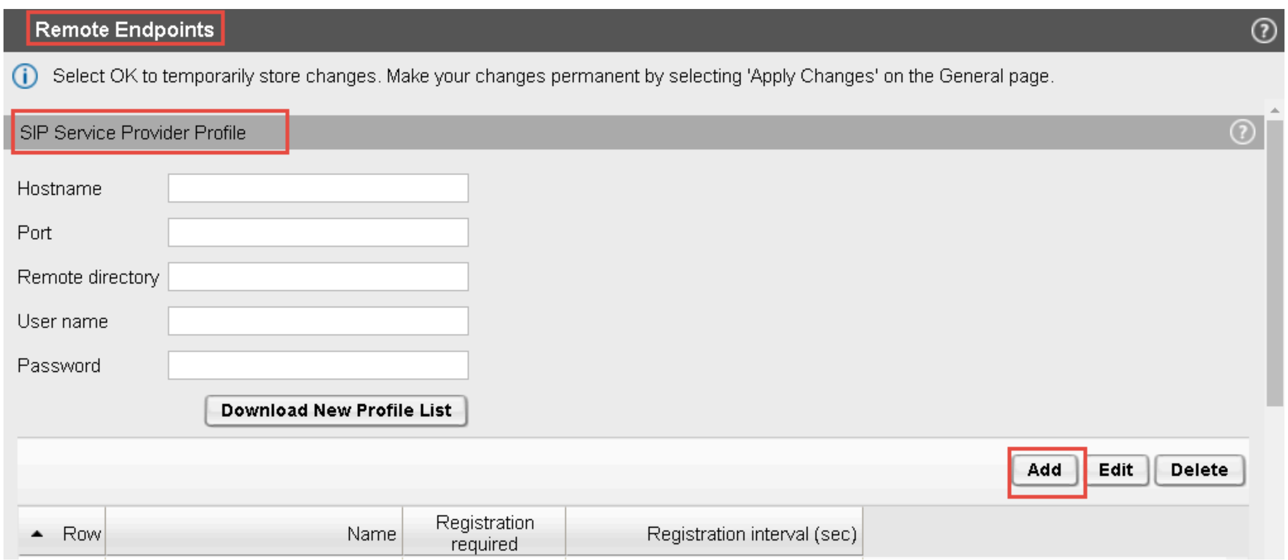
1. Navigate to the **Administration > Features** window.



2. In the **Features** pop-up, check the **Enable Remote Endpoints** checkbox and click **Configure**.

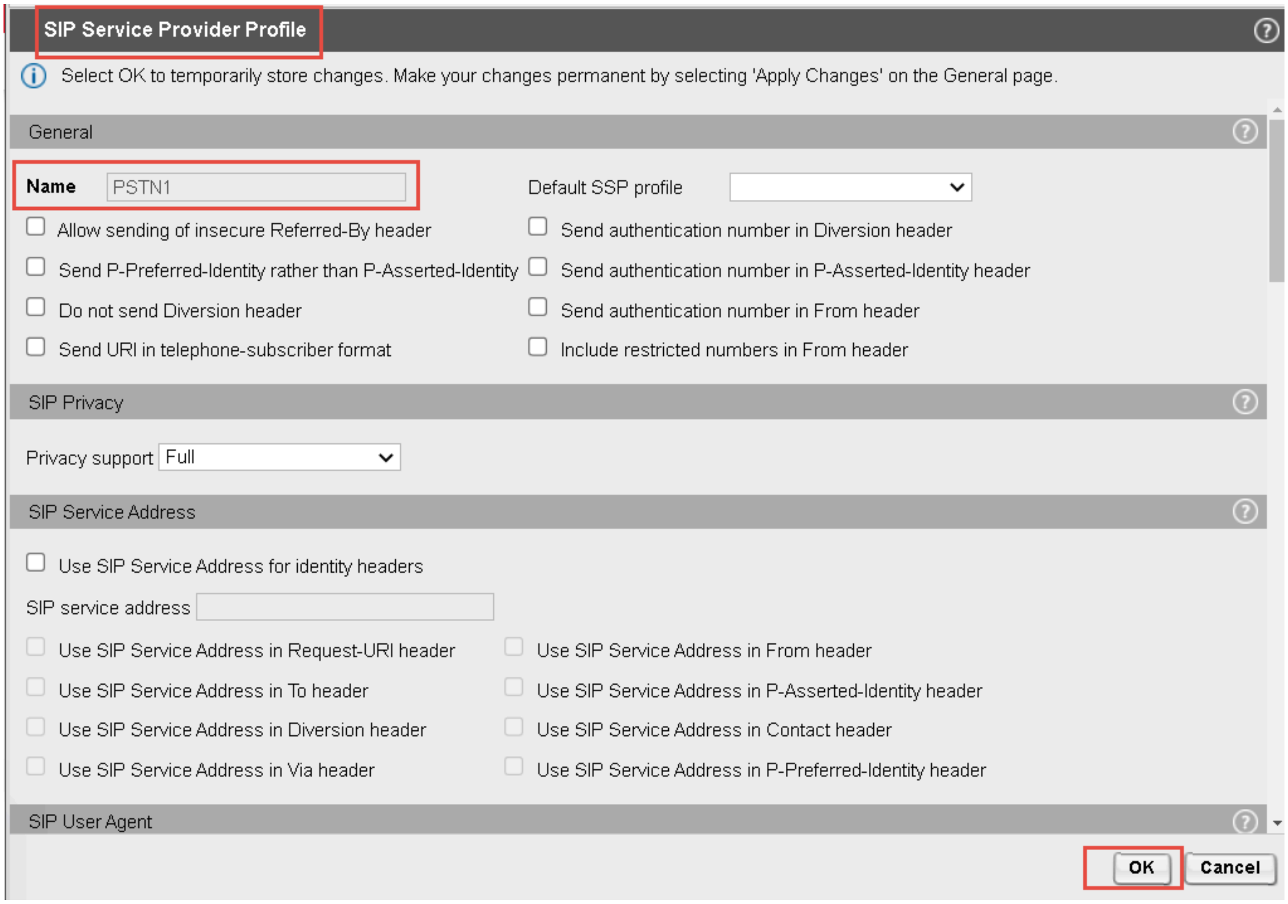


3. In the **Remote Endpoints** window, click **Add** in the **SIP Service Provider Profile** area to add the endpoint profile for the OSSBC – SSP (BCOM) endpoint.

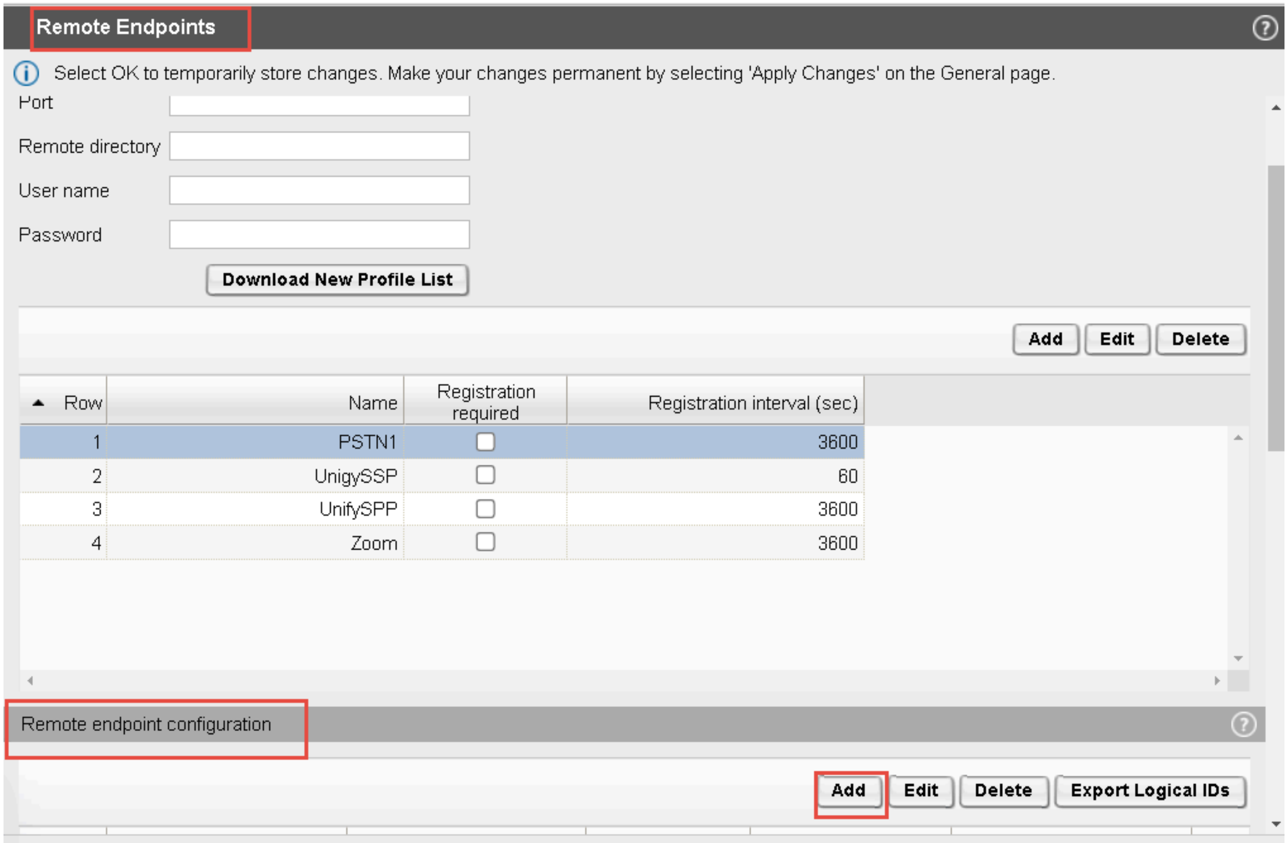


4. In the **SIP Service Provider Profile** window, enter the following:

- **Name**: Enter the name of the profile. For example, **PSTN1.**
- Click **OK** to return to the **Remote endpoints** window.

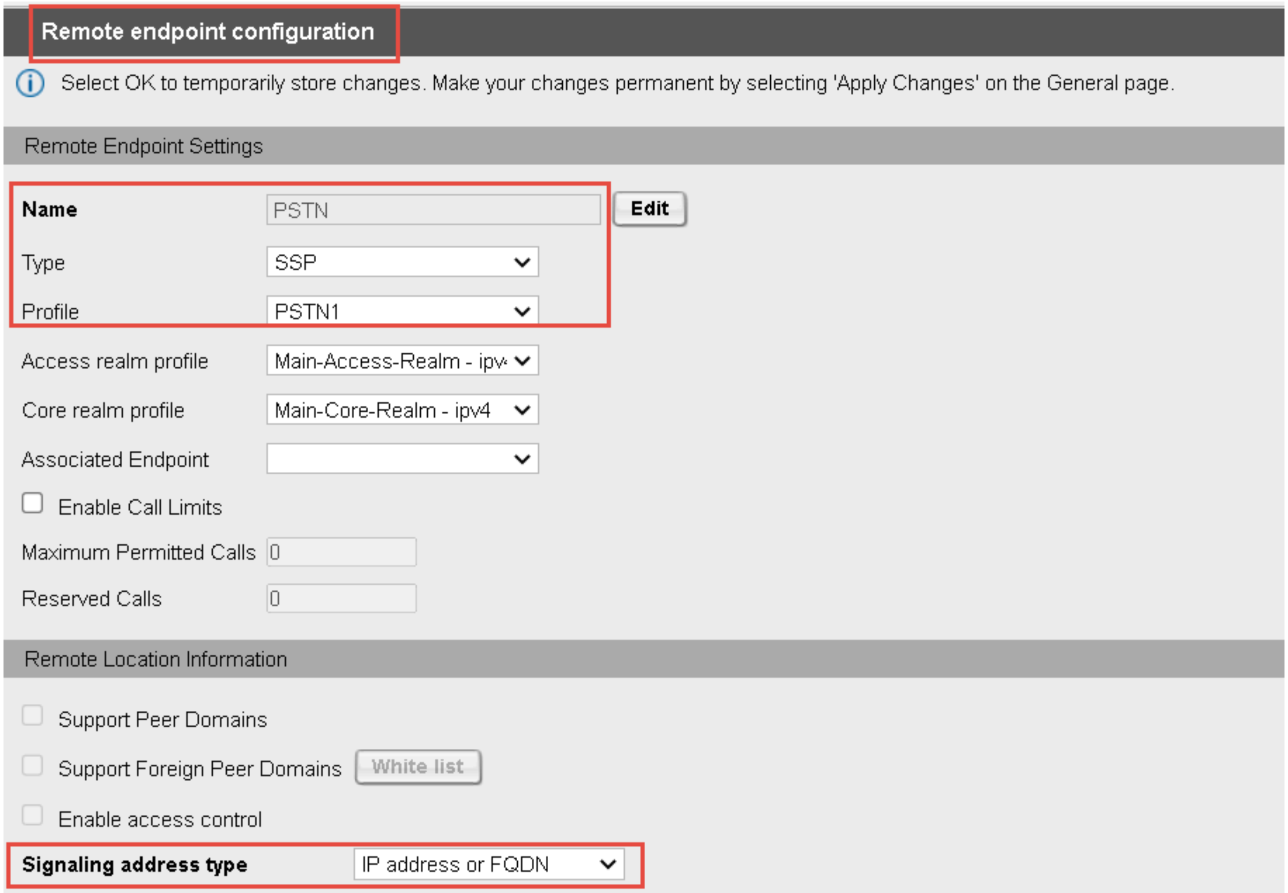


5. Locate the **Remote endpoint configuration** area and click **Add**.



6. In the **Remote Endpoint configuration** window, configure the following parameters:

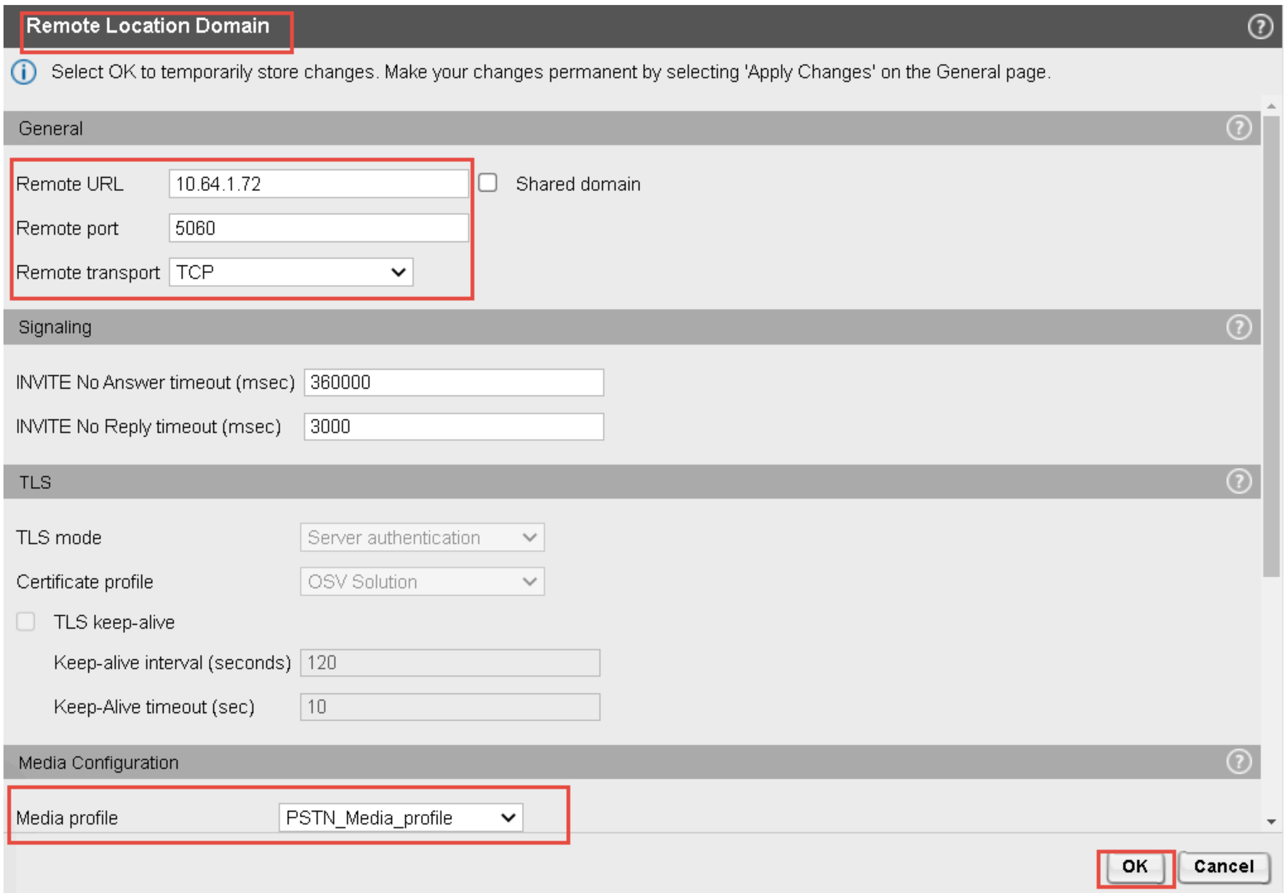
- **Name**: Enter the name of the remote endpoint. For example, PSTN.
- **From the Type drop**-down menu, select **SSP**.
- **From the Profile drop**-down menu, select the PSTN SIP service provider profile. For example, PSTN1.
- From the **Signaling address type** drop-down menu, select **IP address or FQDN**.



7. Locate the **Remote Location domain** area and click **Add** to add the Zoom IP address.

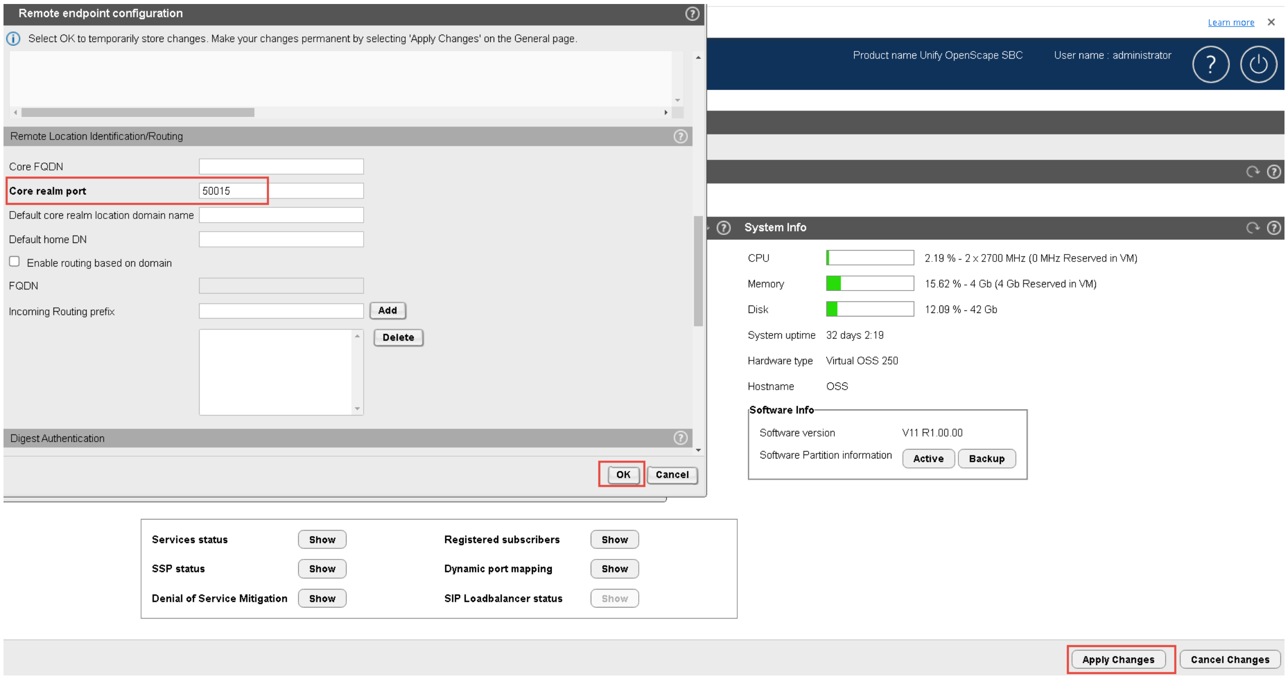
8. In the **Remote Location Domain** window, enter the following:

   a. **Remote URL:** Enter the PSTN IP address
   b. **Remote port**: Enter the port number provided by the PSTN provider (for example, **5061**)
   c. From the **Remote transport** drop-down menu, select the transport protocol provided by the PSTN provider. For example **TCP**.
   d. Locate the **Media Configuration** area, and from the **Media profile** drop-down menu, select the **PSTN** media profile.
   e. Click **OK**.



9. In the **Remote endpoint configuration** window, locate the **Remote Location IdentificationRouting** area.

10. In the **Core realm port** field, enter the core realm value as **50015**.



11. Click **OK** to return to the Remote Endpoints window.
12. Click **OK** on all open windows.
13. Click **Apply Changes.**

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