Ribbon SBC Edge SWe Lite R11.0 on Azure Interop with Cisco UCM and Microsoft Teams Direct Routing : Interoperability Guide

Interoperable Vendors

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Document Overview

This document provides the configuration snapshot of the interoperability performed between Ribbon's SWe Edge on Azure with on-premise Cisco Unified Communications Manager (CUCM).

References

- For additional information on Cisco Unified Communications Manager, refer to https://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-installation-and-configuration-guides-list.html
- For additional information on Ribbon's SWe Edge, refer to Deploying an SBC SWe Edge from the Azure Marketplace

Alert
From Release 11.0.0 onwards, Ribbon "**SBC SWe Lite**" has been rebranded as "**SBC SWe Edge**".

About Ribbon SBC SWe Edge

The Ribbon Session Border Controller Software Edition Edge (SBC SWe Edge) provides best-in class communications security. The SBC SWe Edge dramatically simplifies the deployment of robust communications security services for SIP Trunking, Direct Routing and Cloud UC services. The SBC SWe Edge operates natively in the Azure and AWS Cloud as well as on virtual machine platforms including Microsoft Hyper-V, VMware and Linux KVM.

About Cisco Unified Communications Manager

Cisco Unified Communications Manager is a core call-control application of Cisco UCM. It provides enterprise-class call control, session management, voice, video, messaging, mobility and conferencing services in a way that is efficient, highly secure, scalable and reliable.

About Microsoft Teams Direct Routing

Microsoft Phone System Direct Routing allows the connection of a supported customer-provided Session Border Controller (SBC) to a Microsoft Phone System. Direct Routing enables using virtually any PSTN trunk with the Microsoft Phone System and configuring interoperability between customer-owned telephony equipment, such as a third-party private branch exchange (PBX), analog devices, and Microsoft Phone System.

Scope

This document provides configuration best practices for deploying Ribbon's SBC SWe Edge with Cisco Unified Communications Manager (CUCM). Note that these are configuration best practices and each customer may have unique needs and networks. Ribbon recommends that customers work with network design and deployment engineers to establish the network design that best meets their requirements.

Non-Goals

It is not the goal of this guide to provide detailed configurations that will meet the requirements of every customer. Use this guide as a starting point and build the SBC configurations in consultation with network design and deployment engineers.

Audience

This is a technical document intended for telecommunications engineers with the purpose of configuring both the Ribbon SBC and the third-party product. Navigating the third-party product as well as the Ribbon SBC SWe Edge GUI is required. Understanding the basic concepts of TLS/TCP/UDP, IP/Routing, and SIP/SRTP is also necessary to complete the configuration and any required troubleshooting.

Pre-Requisites

The following aspects are required before proceeding with the interop:

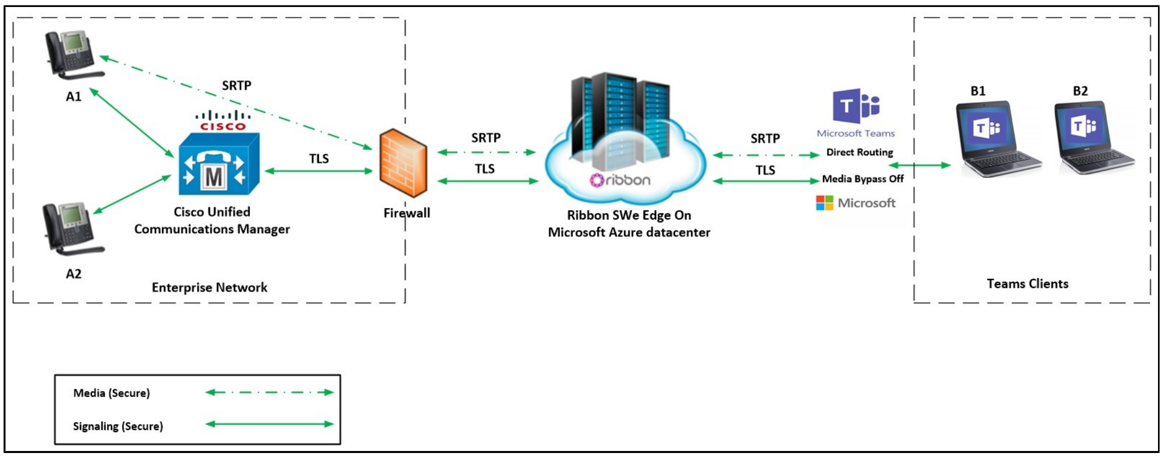
- Microsoft Azure subscription
- Ribbon SBC SWe Edge on Azure
- SBC SWe Edge License
  - This interop requires the acquisition and application of cloud SIP sessions, as documented at Cloud-Based SBC SWe Edge Deployment Licenses
- Public IP Addresses
- Service Provider SIP Trunk
- TLS Certificates for SBC SWe Edge
  - Refer to Working with Certificates

Product and Device Details

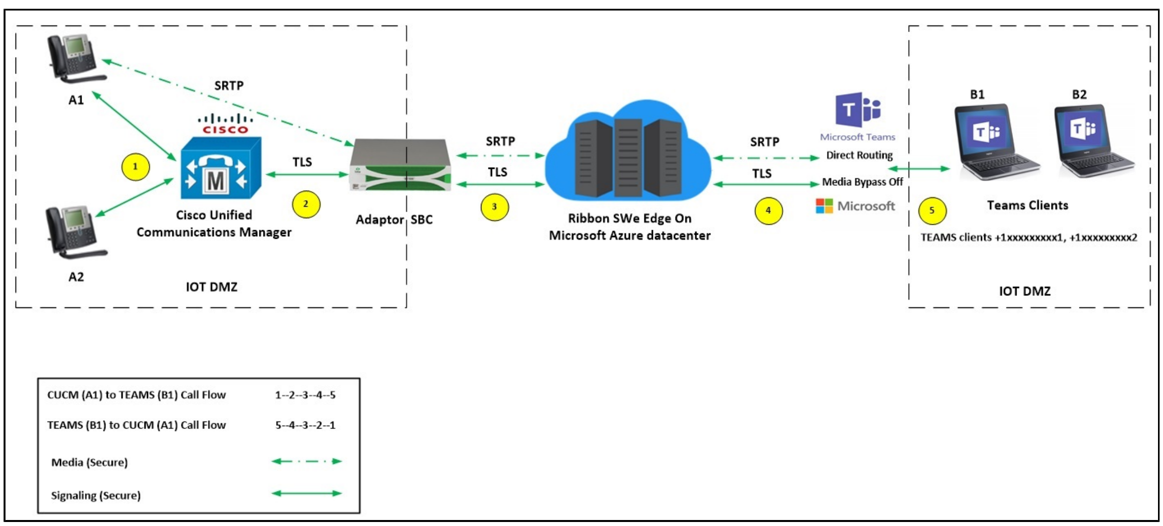
	Equipment/ Product	Software Version
Ribbon Communications	Ribbon SBC SWe Edge	11.0
Third-Party Products	Cisco Unified Communications Manager	12.5
Third-Party Phones	Cisco Jabber client	12.6.1.34405
Microsoft Corporation	Microsoft Teams Client Desktop app	1.4.00.19572
	Microsoft Teams Client Mobile app	1416
Administration and Debugging Tools	Wireshark	3.2.7
	LX Tool	2.1.0.6

Network Topology Diagram

Deployment Topology

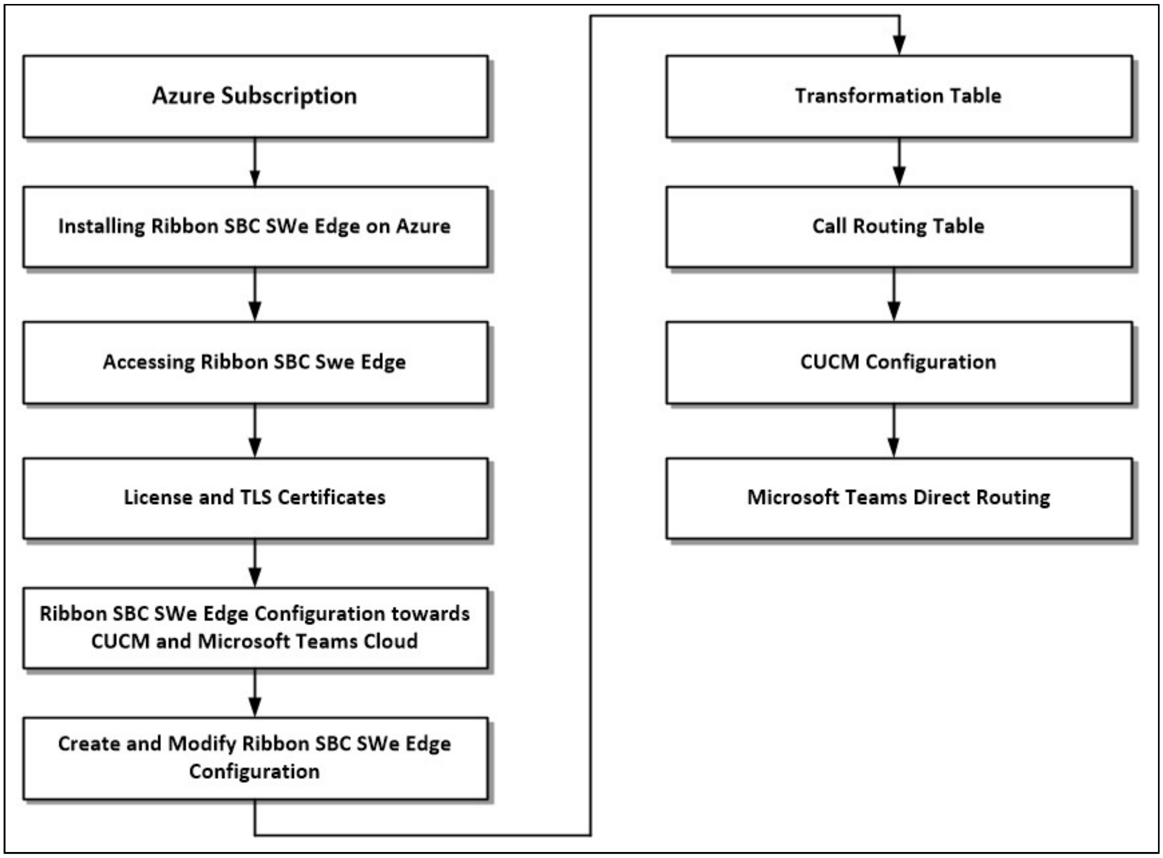


Interoperability Test Lab Topology (Call Flow Diagram)



Document Workflow

The sections in this document follow the sequence below. The reader is advised to complete each section for successful configuration.



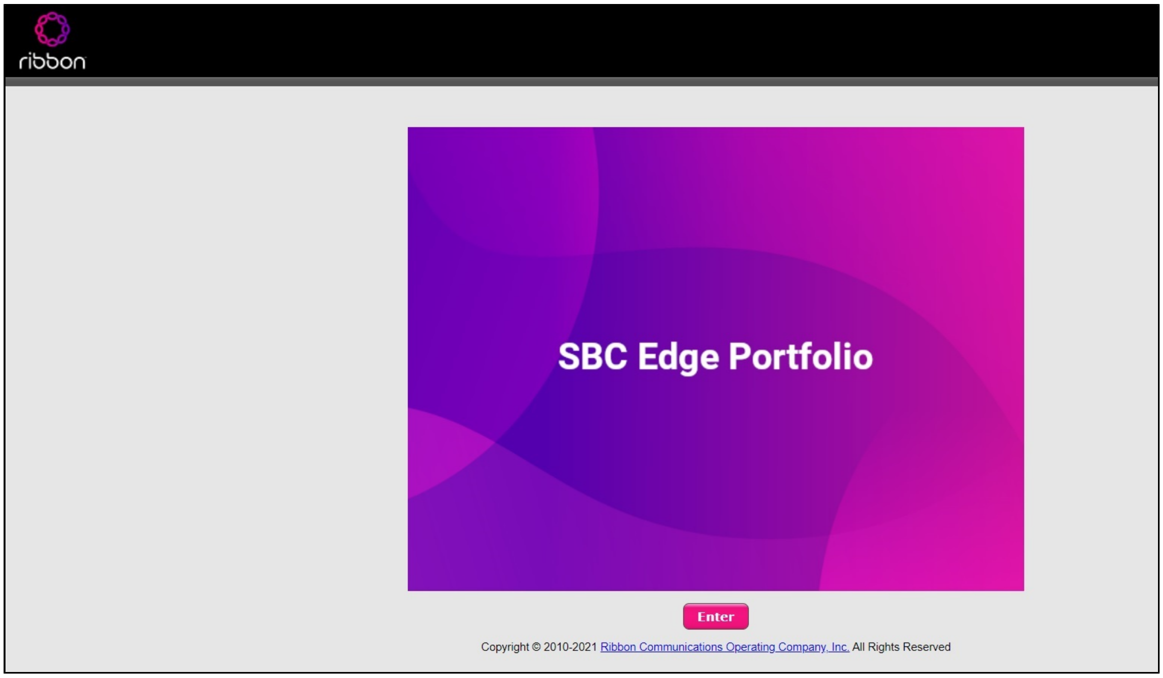
Section A: Ribbon SBC SWe Edge Configuration

Installing Ribbon SBC SWe Edge On Azure

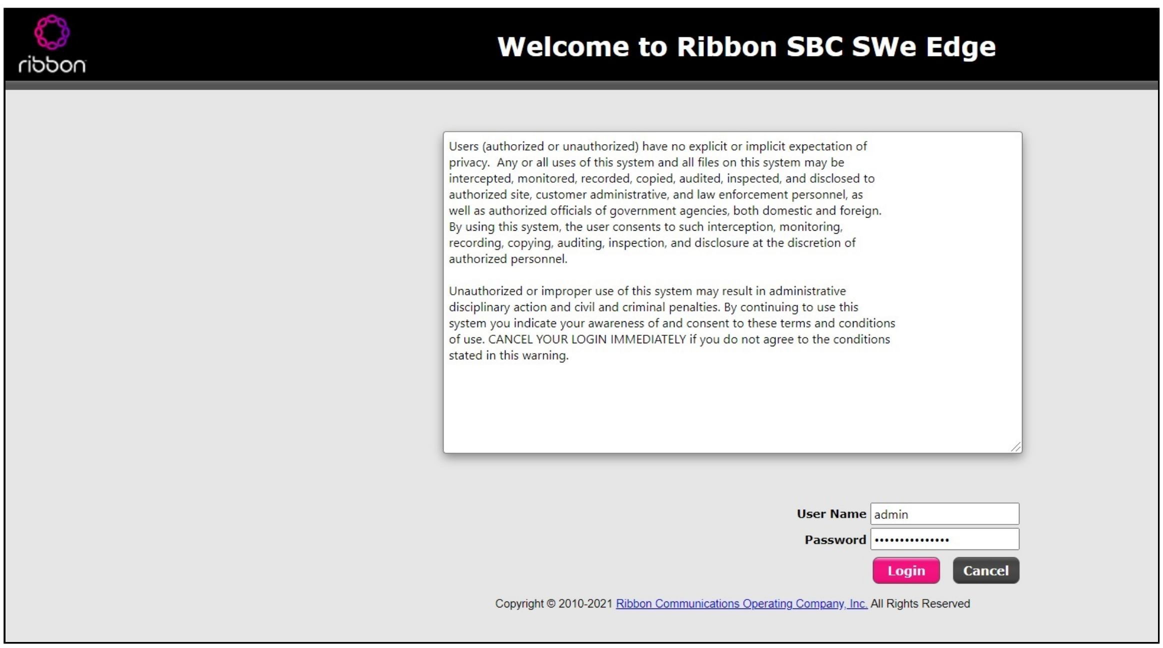
The SBC SWe Edge is available for deployment in Azure. It is created as a virtual machine (VM) hosted in Azure. To deploy an SBC SWe Edge instance, refer to Deploying an SBC SWe Edge from the Azure Marketplace.

Accessing Ribbon SBC SWe Edge

Open any browser and enter the SBC SWe Edge IP address.



Click **Enter** and log in with a valid User ID and Password.



View Networking Interfaces

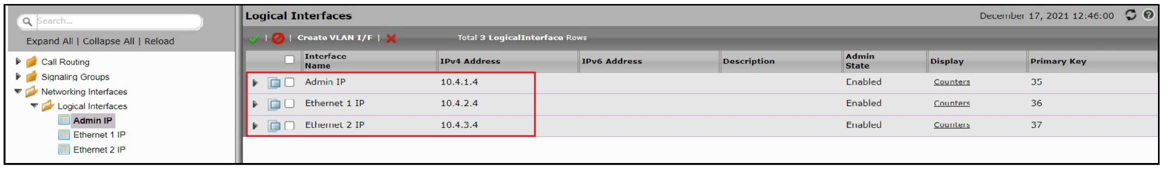
The SBC SWe Edge supports five system created logical interfaces (known as **Administrative IP**, **Ethernet 1 IP**, **Ethernet 2 IP**, **Ethernet 3 IP**, and **Ethernet 4 IP**). In addition to the system created logical interfaces, the Ribbon SBC SWe supports user-created VLAN logical sub-interfaces.

Administrative IP, Ethernet 1 IP and Ethernet 2 IP are used for this interop.

From the **Settings** tab, navigate to **Networking Interfaces > Logical Interfaces.**

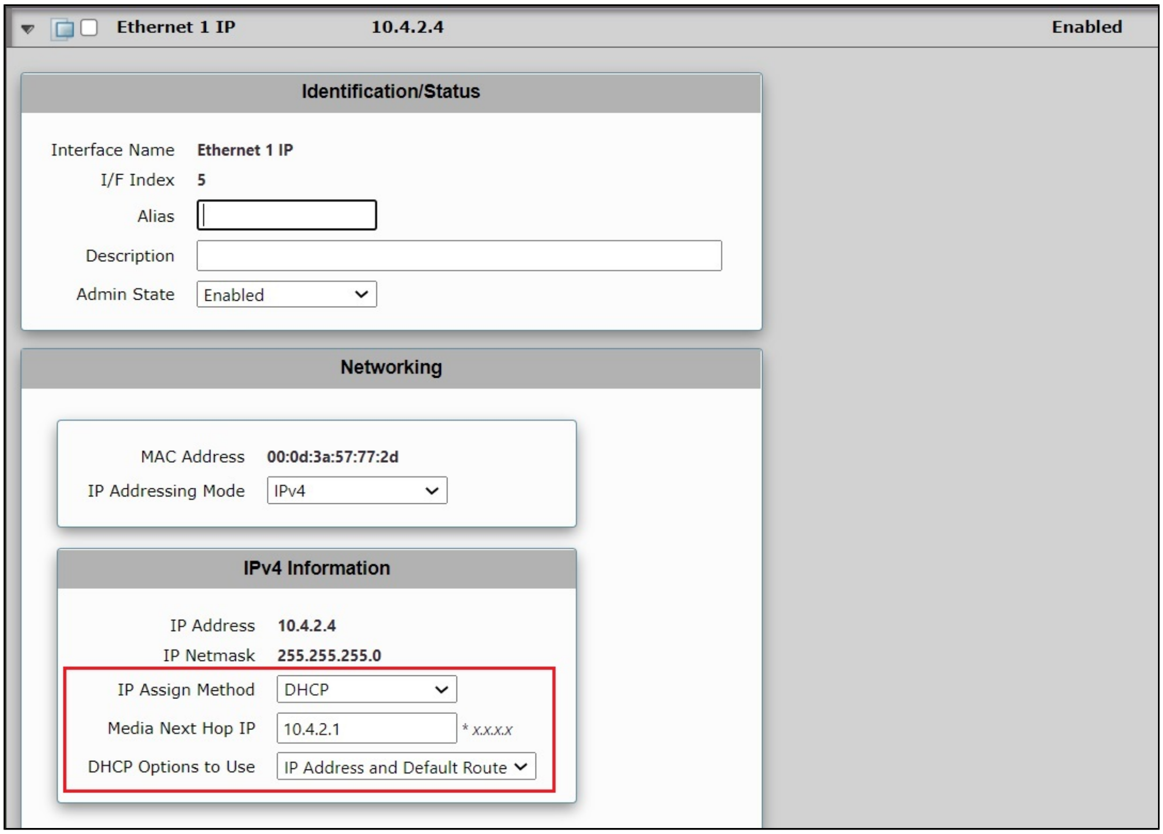
The SBC SWe Edge system supports a logical interface called the Admin IP (Administrative IP, also known as the Management IP). A Static IP or DHCP is used for running Initial Setup of the SBC SWe Edge system.

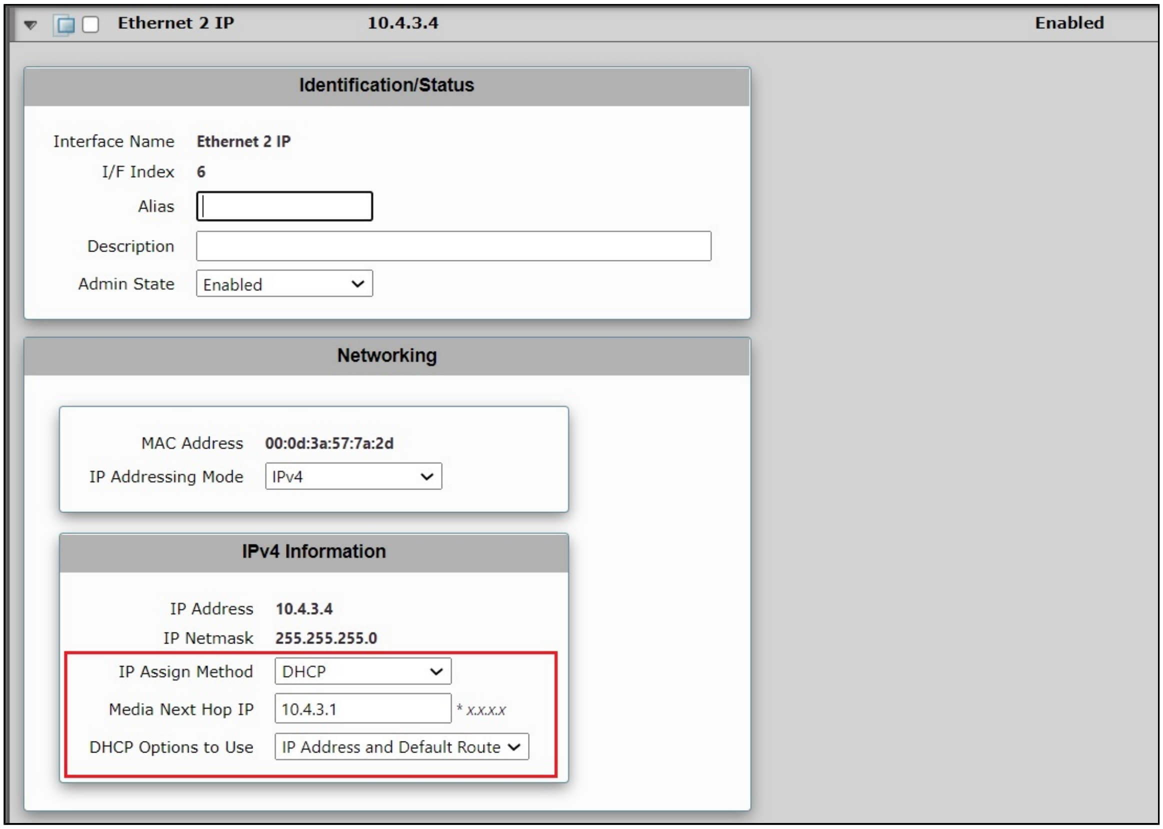
Administrative IP



Ethernet 1 IP and Ethernet 2

Ethernet 1 and 2 IP is assigned an IP address used for transporting all the VOIP media packets (for example, RTP, SRTP) and all protocol packets (for example, SIP, RTCP, TLS). DNS servers of the customer's network should map the SBC SWe Edge system hostname to this IP address. In the default software, **Ethernet 1 and 2 IP** is enabled and an IPv4 address is acquired through a connected DHCP server. This IP address is used for performing Initial Set up on the SBC SWe Edge.





Configure Static Routes

Static routes are used to create communication to remote networks. In a production environment, static routes are mainly configured for routing from a specific network to another network that you can only access through one point or one interface (single path access or default route).

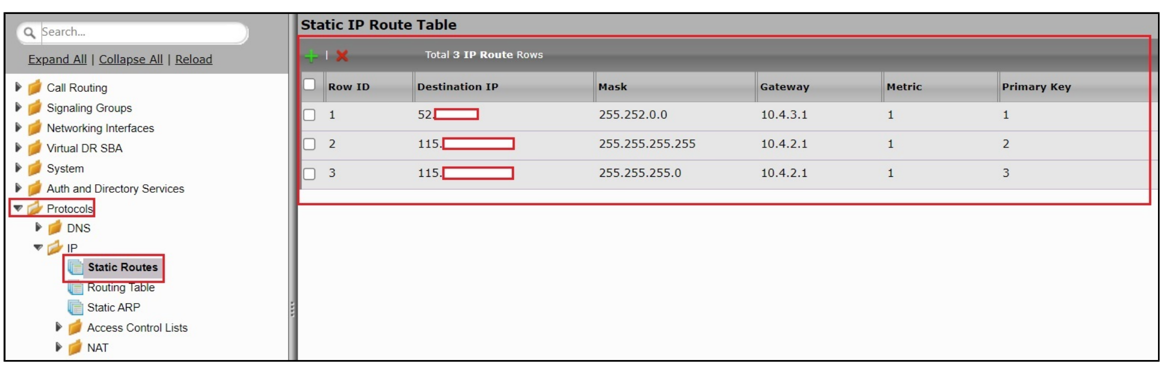
Derive the Private IP address and Gateway for each interface on Azure.

Destination IP
Specifies the destination IP address.

Mask
Specifies the network mask of the destination host or subnet. If the 'Destination IP Address' field and 'Mask' field are both 0.0.0.0, the static route is called the 'default static route'.

Gateway
Specifies the IP address of the next-hop router to use for a specific static route.

Metric
Specifies the cost of this route and therefore indirectly specifies the preference of the route. Lower values indicate more preferred routes. The typical value is 1 for most static routes, indicating that static routes are preferred to dynamic routes.

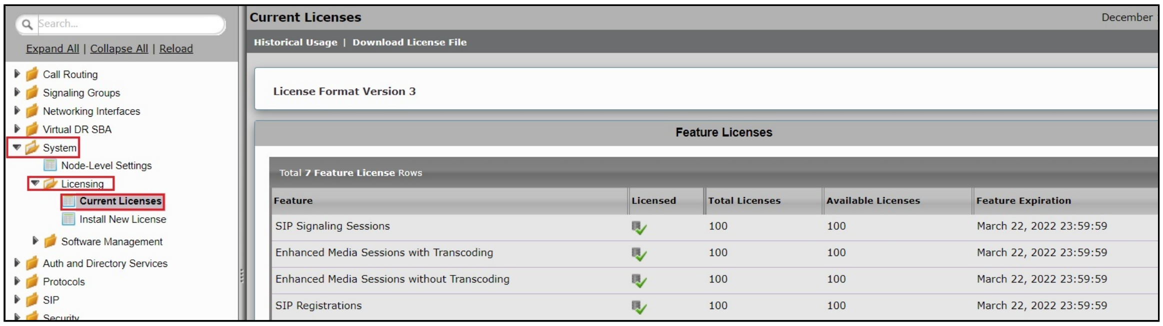


License and TLS Certificates

View License

This section describes how to view the status of each license along with a copy of the license keys installed on your SBC. The **Feature Licenses** panel enables you to verify whether a feature is licensed, along with the number of remaining licenses available for a given feature at run-time.

From the **Settings** tab, navigate to **System > Licensing > Current Licenses.**

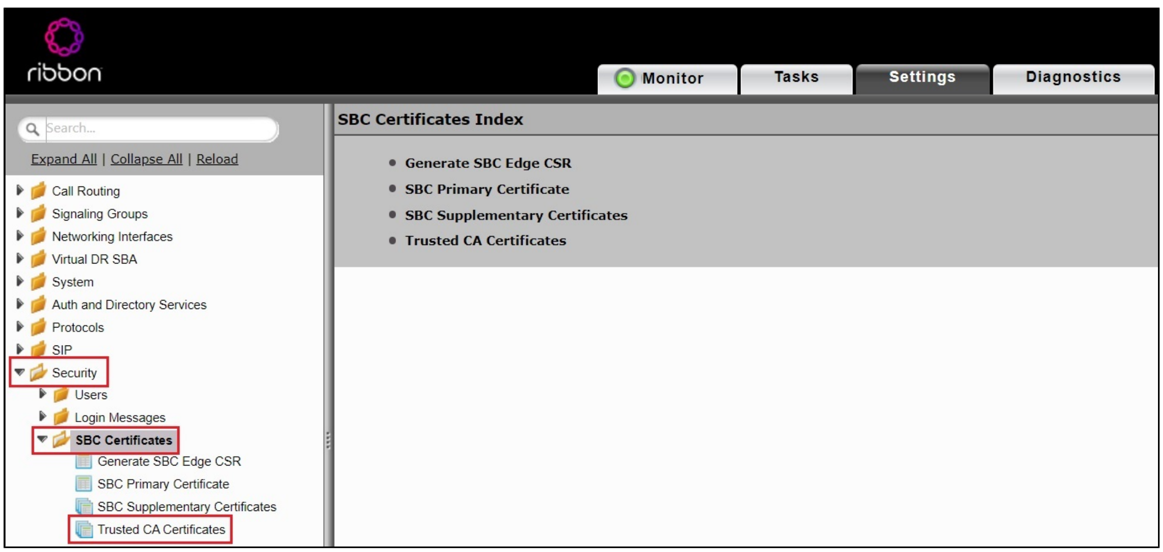


For more details on Licenses, refer to Cloud-Based SBC SWe Edge Deployment Licenses.

Import Trusted Root CA Certificates

A Trusted CA Certificate is a certificate issued by a trusted certificate authority. Trusted CA Certificates are imported to the SBC SWe Edge to establish its authenticity on the network.

From the **Settings** tab, navigate to **Security > SBC Certificates > Trusted CA Certificates.**



This section describes the process of importing Trusted Root CA Certificates, using either the File Upload or Copy and Paste methods.

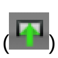
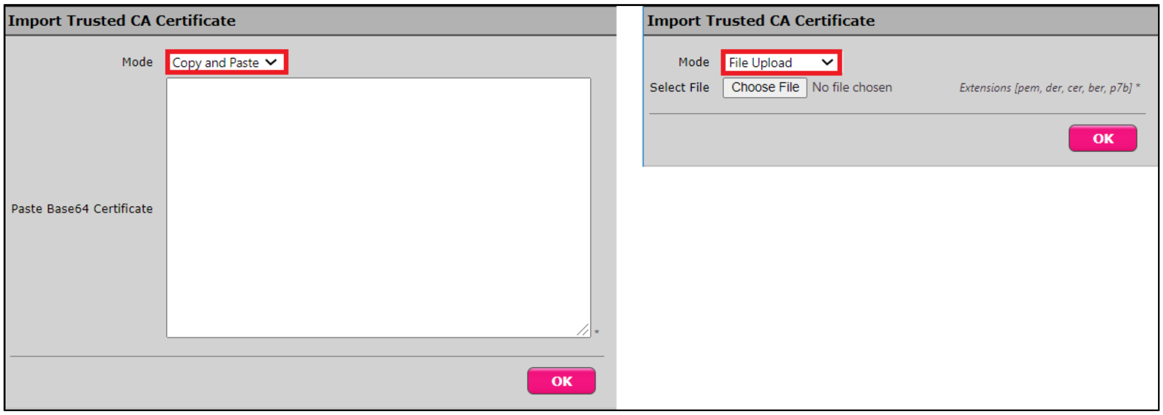
1. To import a Trusted CA Certificate, click the Import Trusted CA Certificate () Icon.
2. Select either Copy and Paste or File Upload from the **Mode** menu.
3. If you choose **File Upload**, use the **Select File** button to find the file.
4. Click **OK**.

Figure 1: Trusted Certificates 2



Follow the steps above to import the Service Provider's Root and Intermediate certificates of their Public CA.

For more details on Certificates, refer to Working with Certificates.

Note
When the **Verify Status** field in the Certificate panel indicates Expired or Expiring Soon, replace the Trusted CA Certificate. You must delete the old certificate before importing a new certificate successfully.

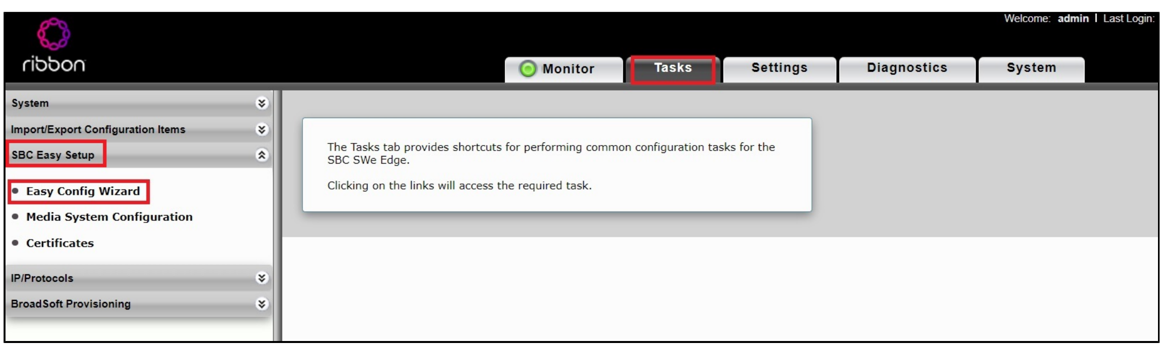
Warning
Most Certificate Vendors sign the SBC Edge certificate with an intermediate certificate authority. There is at least one, but there could be several intermediate CAs in the certificate chain. When importing the Trusted Root CA Certificates, import the root CA certificate and all Intermediate CA certificates. Failure to import all certificates in the chain causes the import of the SBC Edge certificate to fail. Refer to Unable To Get Local Issuer Certificate for more information.

Ribbon SWe Edge Configuration towards CUCM and Microsoft Teams Cloud

Access the Easy Configuration Wizard

1. In the WebUI, click the **Tasks** tab.
2. In the left navigation pane, navigate to **SBC Easy Setup > Easy Config Wizard.** The Easy Configuration screen opens.

The SBC Edge WebUI provides a built-in Easy Configuration wizard that allows you quickly and easily deploy the SBC for operation with provider endpoints (SIP trunk, ISDN PSTN trunk, or IP PBX trunk) and user endpoints (Microsoft Teams, Microsoft On Premises - Skype for Business /Lync, IP Phones, or ISDN PBX or IP PBX).



Navigating the Wizard

As the wizard runs, it directs you through three configuration steps:

Step 1: Set the following parameters to describe the topology for the telephony service provider and user ends of the scenario.

- **Application**: Click the drop-down arrow, then select the Service Provider and user endpoint types that the SBC is to connect to.
- **Scenario Description**: Type up to 32 characters to describe the connectivity scenario.
- **Telephone Country**: Click the drop-down arrow, then select the country in which the telephone services operate.
- **Emergency Services**: Choose **ELIN Identifier**, **E911/E112**, or **None** as the emergency services type.
- **SIP Sessions**: Type a number from 1-1200 to indicate the SIP sessions to allocate for the scenario.

Step 2: Configure the items required for the endpoints selected, fields display based on the endpoint selection in Step 1.

Step 3: The Easy Config validates the final parameters and displays a read-only summary of the configuration that the wizard will apply when you click **Finish** at Step 3. Before you click **Finish**, you can return to previous steps to make adjustments to the data summarized.

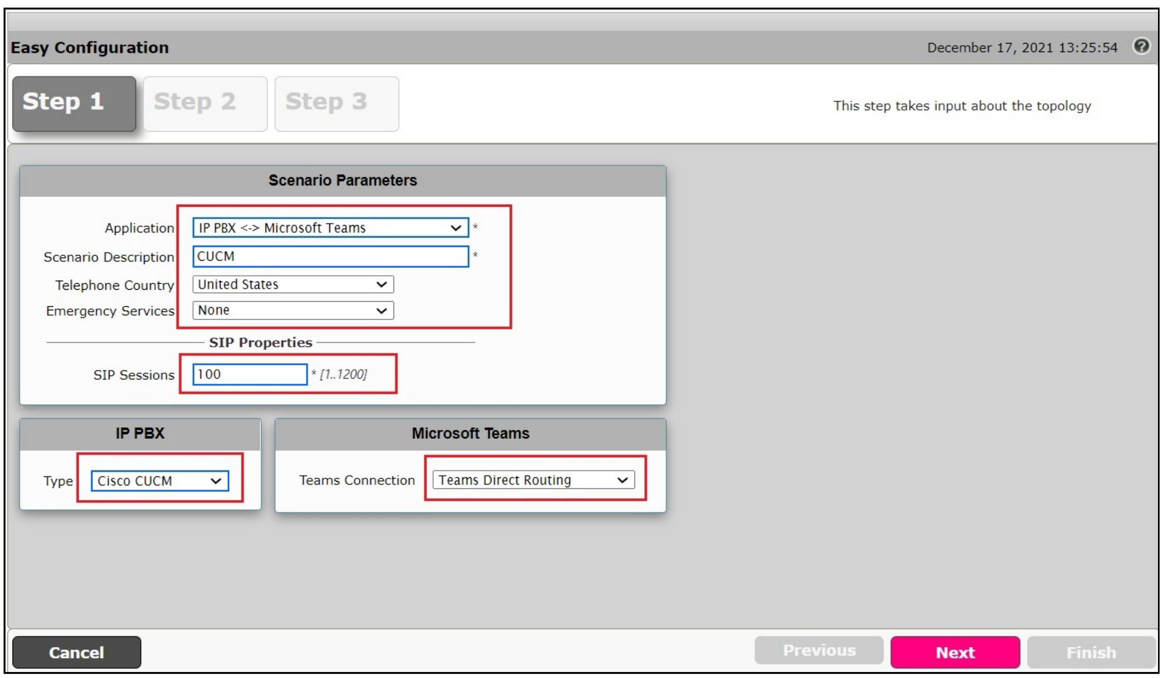
The wizard displays the following buttons for navigation:

- **Previous**: Moves back to the previous step.
- **Next**: Advances to the next step when the current step is validated and complete.
- **Finish**: Submits the data to the SBC.
- **Cancel**: Cancels the Easy Configuration data entered and redirects to the main WebUI.

Configure Ribbon SBC SWe Edge for CUCM and Microsoft Teams

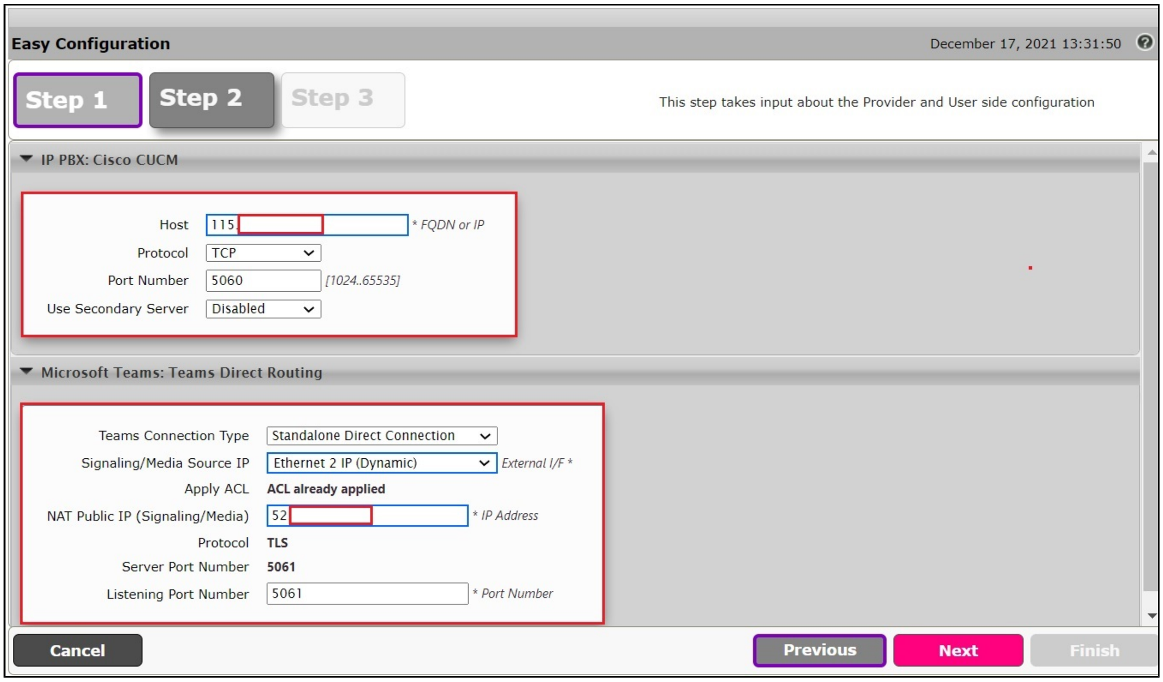
Step 1: Use the Multi-legged approach to configure IP PBX and Microsoft Teams.

1. Click the drop-down arrow on the **Application** and select IP PBX Microsoft Teams.
2. Provide the desired description.
3. Select **Telephone Country** as Unites States.
4. Choose from 1 to 1200 to allocate the SIP Sessions.
5. Select Cisco CUCM as **IP PBX Type.**
6. Select Teams Direct Routing as **Teams Connection.**
7. Click **Next**.



Step 2: After selecting the scenario in Step 1, the following template displays. Complete this step by performing the actions listed below:

1. Provide the Host IP address or FQDN for Cisco CUCM. The traffic is sent to these FQDNs/IP from the SBC SWe Edge.
2. Use **UDP/TCP** with port number 5060 for Service Provider SIP trunk configuration.
3. Select the Signaling/Media Source IP from drop down.
4. Provide the NAT Public IP (Signaling/Media).
5. Click **Next**.

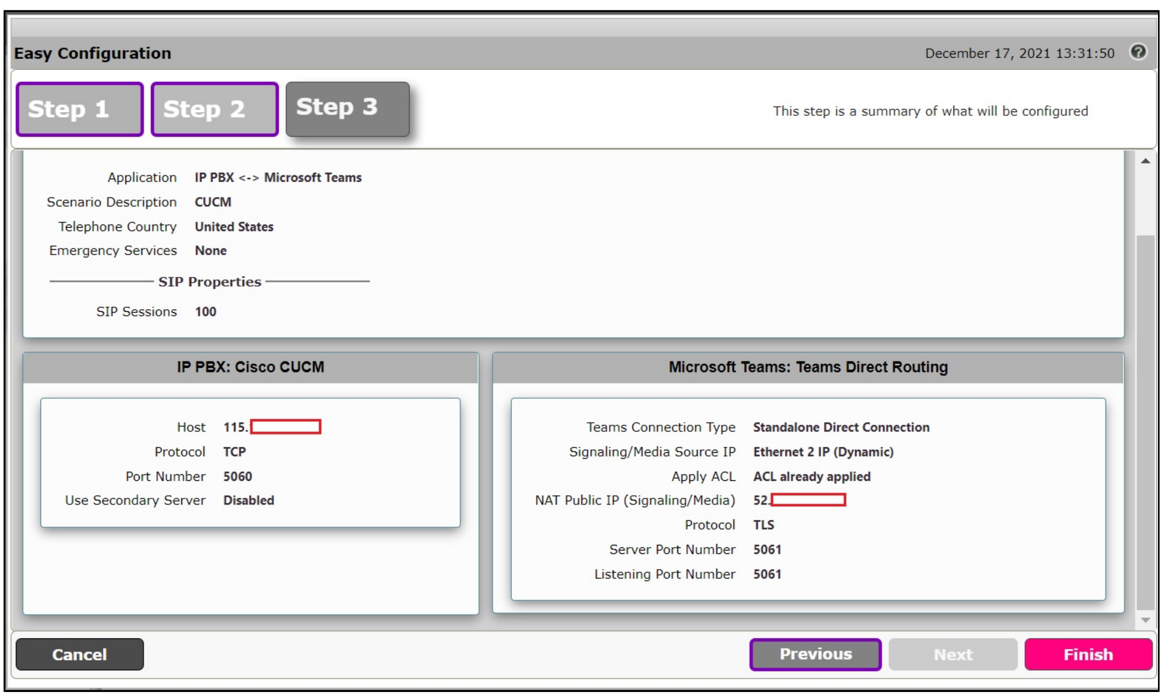


Note
While using "Easy Configuration Wizard" **TLS** protocol is not available by default for Cisco CUCM but can be configured later.

For more information about Microsoft Teams Direct Routing configuration , refer to the following: Connect SBC Edge to Microsoft Teams Direct Routing

Step 3: This step displays a read-only summary of the configuration.

1. Check if the information entered in the previous steps is correct. If the entered information is wrong, return to the previous step by clicking **Previous** and modify the required field.
2. Click **Finish** to complete the configuration.



- A pop up window appears once all the 3 steps are completed. Click **OK** to continue.
- Wait for the configuration to complete and click **OK** on the next window. This completes the configuration of Microsoft Teams and IP-PBX (CUCM) SIP Trunk on the SBC SWe Edge.

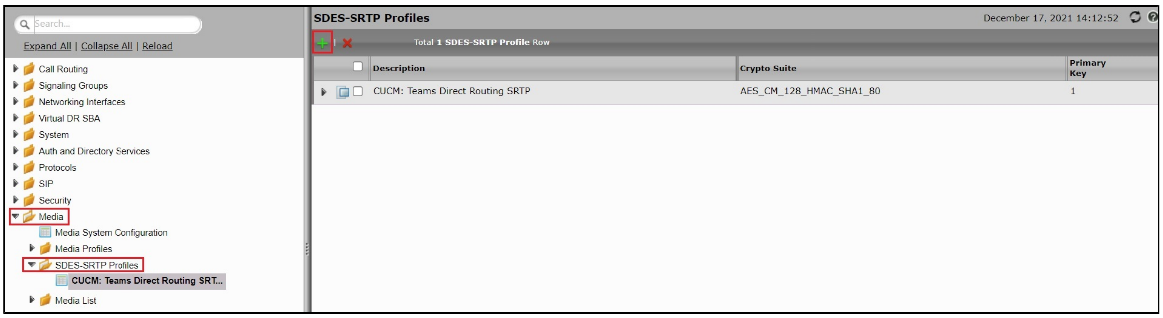
Create and Modify Ribbon SBC SWe Edge Configuration

The Easy Configuration Wizard does not currently set all applicable variables to the correct settings. This will be addressed in the subsequent SBC SWe Edge releases. Until then, follow the procedures below. This section describes the steps to configure the SBC SWe Edge with TLS /SRTP towards IP-PBX (CUCM) SIP Trunk. Ribbon strongly recommends encrypting the connection between the IP-PBX SIP Trunk and the SBC SWe Edge.

Create SRTP Profile

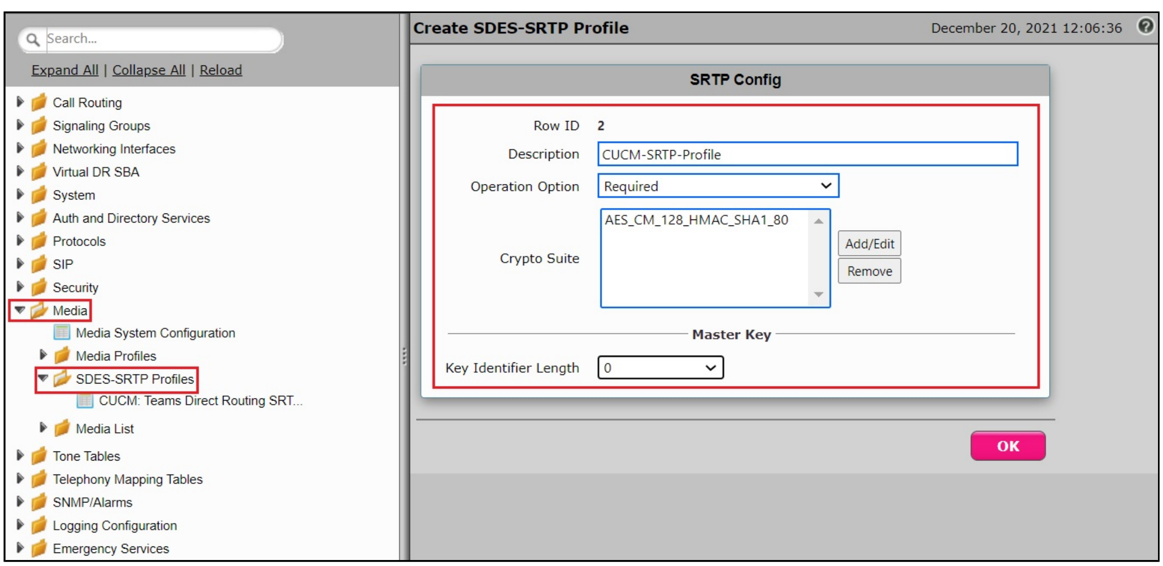
SDES-SRTP Profiles define a cryptographic context that is used in SRTP negotiation. SDES-SRTP Profiles required for enabling encryption and SRTP are applied to Media Lists. SDES-SRTP Profiles was previously named Media Crypto Profiles.

From the **Settings** tab, navigate to **Media > SDES-SRTP Profiles.** Click the icon to create a new SRTP profile.



Follow the steps below to complete the configuration:

1. Provide the desired description for the profile.
2. Set Operation Option as "Required". This setting permits call connections only if you can use encryption for the call. If the peer device does not support SRTP (Secure Real Time Protocol) for voice encryption over the IP network, the call setup will fail.
3. Attach the Crypto suite "AES_CM_128_HMAC_SHAI_80" - A crypto suite algorithm that uses the 128 bit AES-CM encryption key and a 80 bit HMAC_SHA1 message authentication tag length.
4. Key Identifier Length set to "0" - Set this value to **0** to disable the MKI in SDP.
5. Click **OK**.

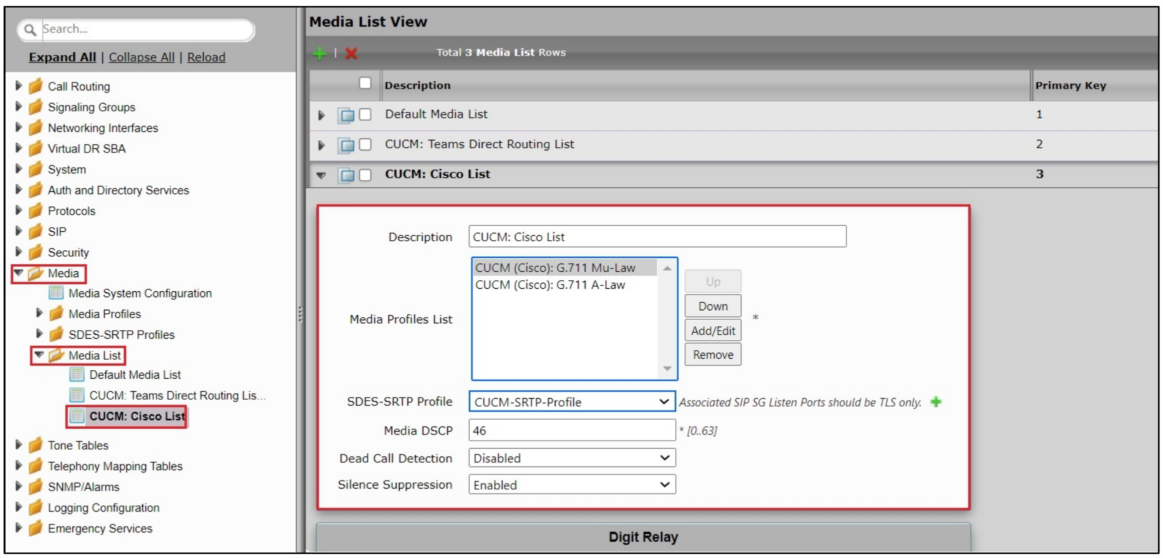


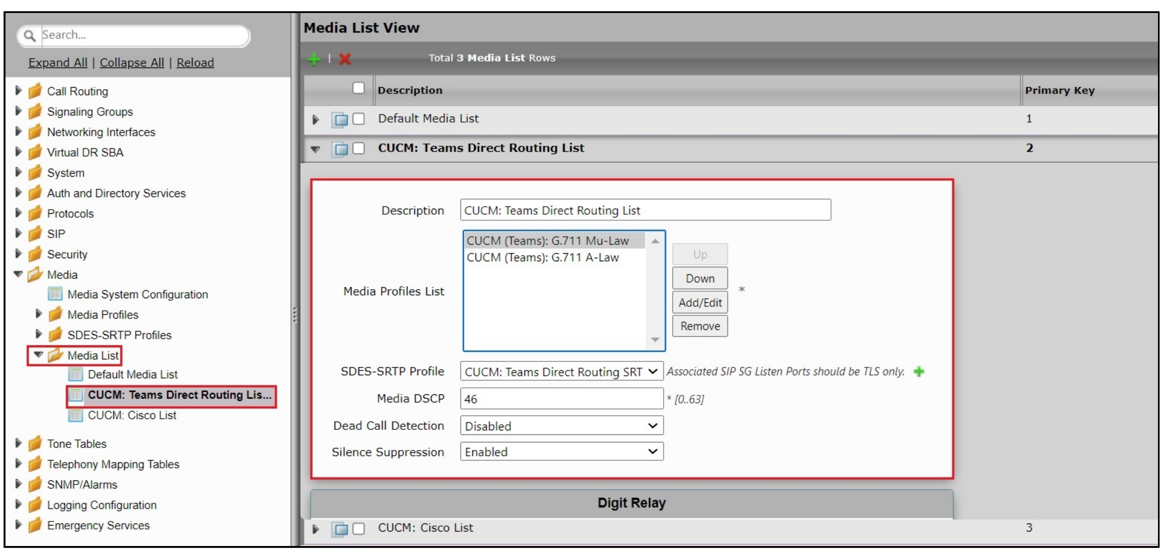
Warning
For SIP Trunk towards CUCM, If the SWe Edge SRTP profile is configured with "**Operation Option**" as "**Required**" and "**Crypto Suit**" as "**AES_CM_128_HMAC_SHA1_80**", call hold initiated from Cisco endpoint will fail. This is a known issue with Cisco CUCM. To overcome it, use "**AES_CM_128_HMAC_SHA1_32"** between CUCM and SWe Edge.

Update Media List

From the **Settings** tab, navigate to **Media > Media List,** Click the expand (▸) icon next to the entry.

1. Attach the SDES-SRTP profile (Specifies the profile for authentication/encryption protocols applied with this Media List) created in the previous step.
2. Select the required Media Profiles list.
3. Update the Media Profiles List as required for both Media List configurations.
4. Click **Apply**.





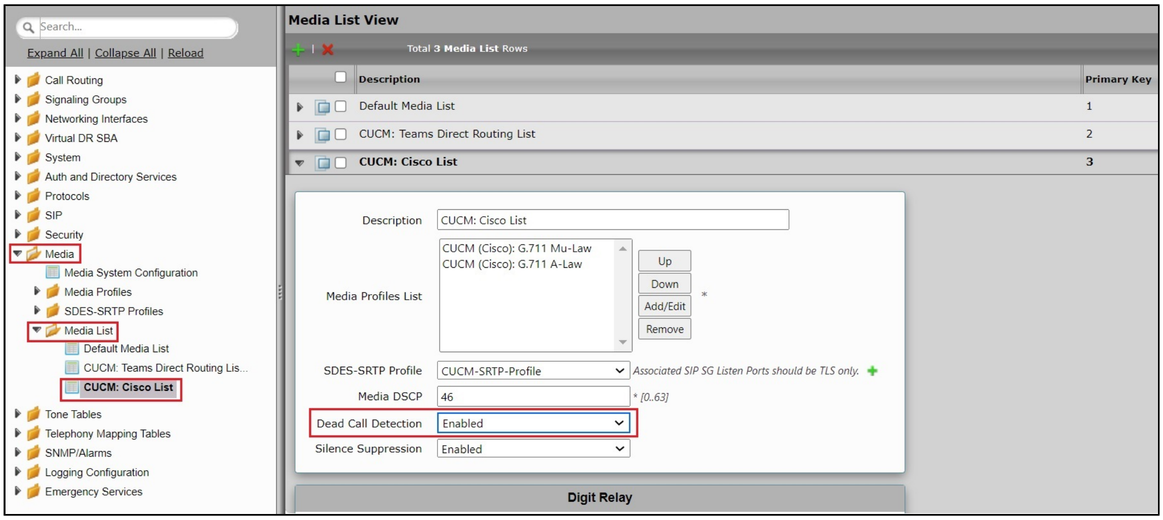
Enable Dead Call Detection

Specifies whether or not to use RTCP-based Dead Call Detection (DCD).

Dead Call Detection is accomplished by monitoring incoming RTCP packets. If this feature is enabled and no RTCP packets are received from the peer for 30 seconds, the call is considered "dead" and is disconnected.

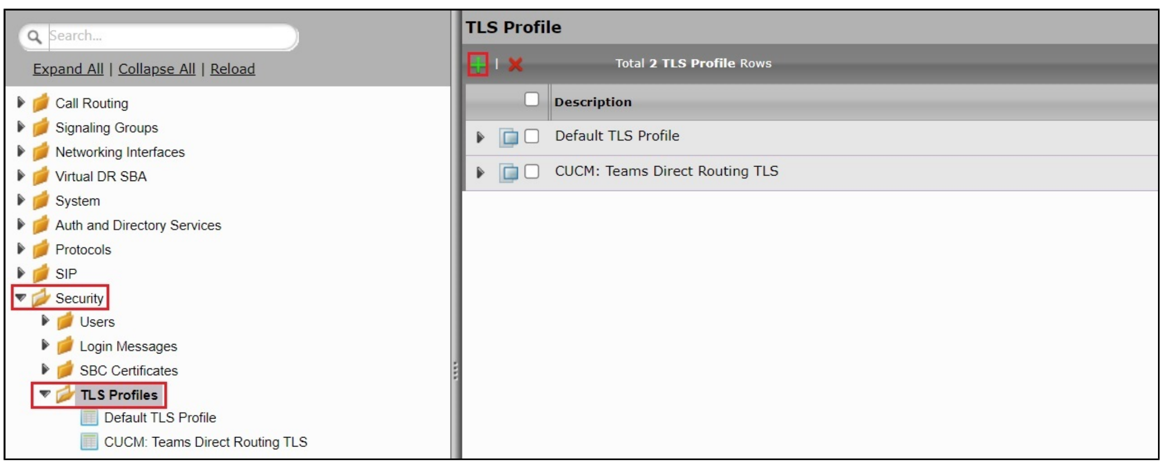
From the **Settings** tab, navigate to **Media > Media List.** Click the **expand** ( ) Icon next to the entry you wish to enable the feature.

- Enable DCD from the options provided in the drop-down.

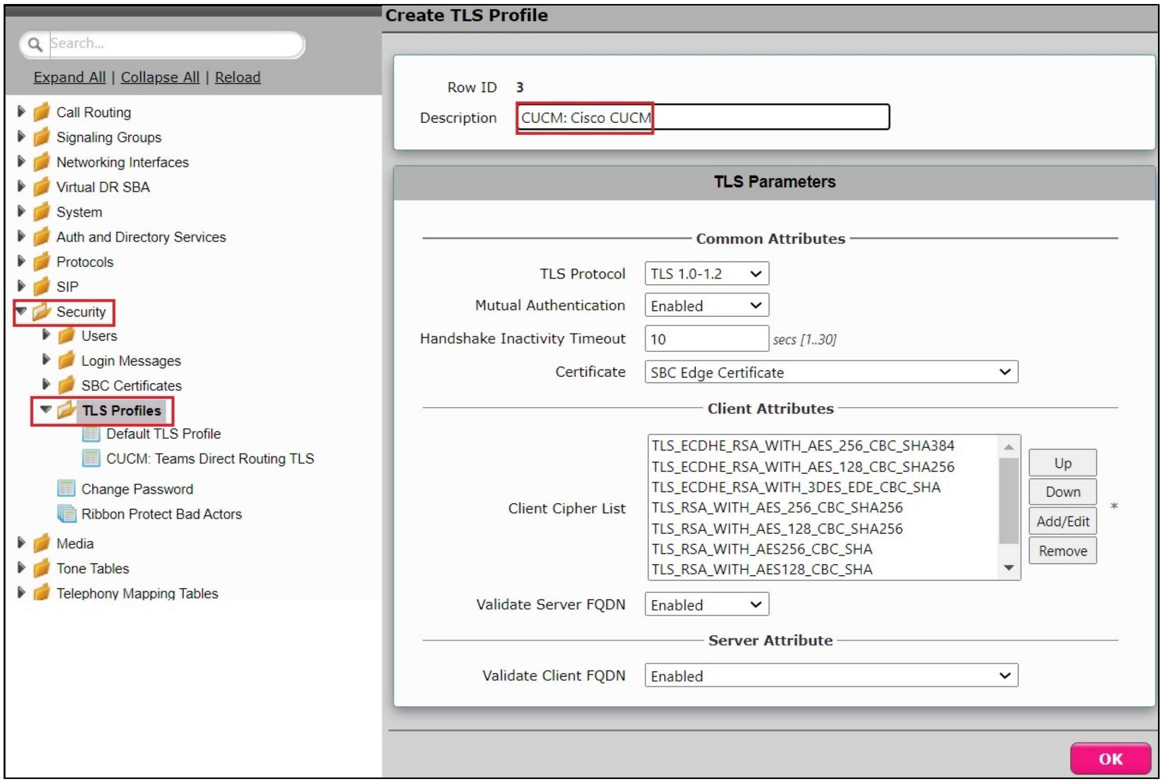


TLS Profile

From the **Security** tab, navigate to **TLS Profiles.** Click the icon to create a new TLS profile.



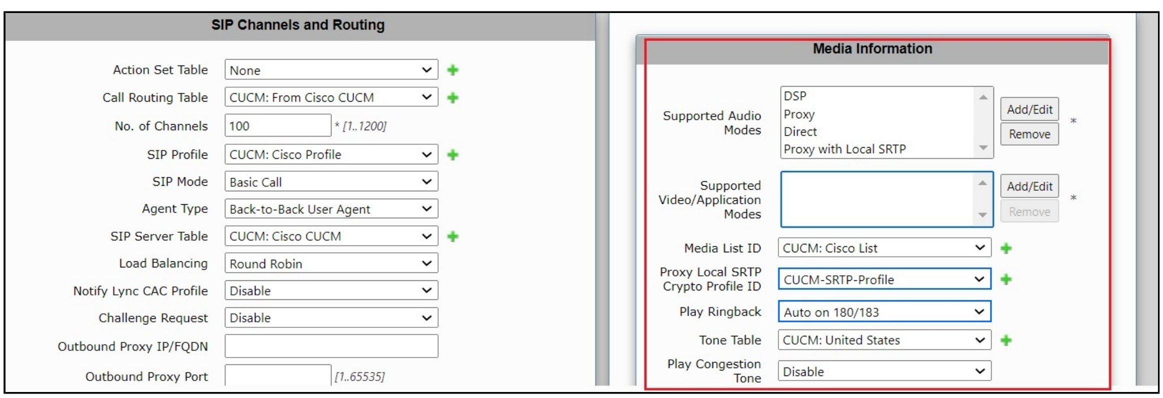
1. Provide the table's **Description** as desired.
2. Modify the Values as required.
3. Click **OK**.

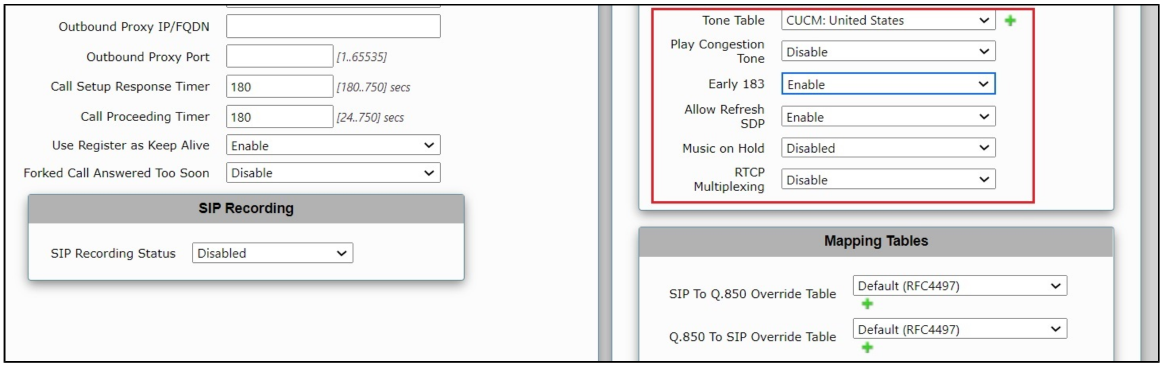


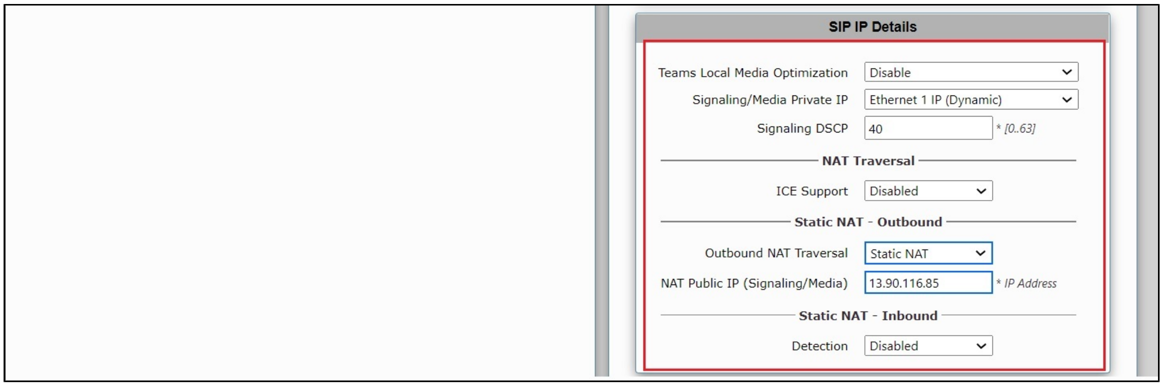
Update Signaling group

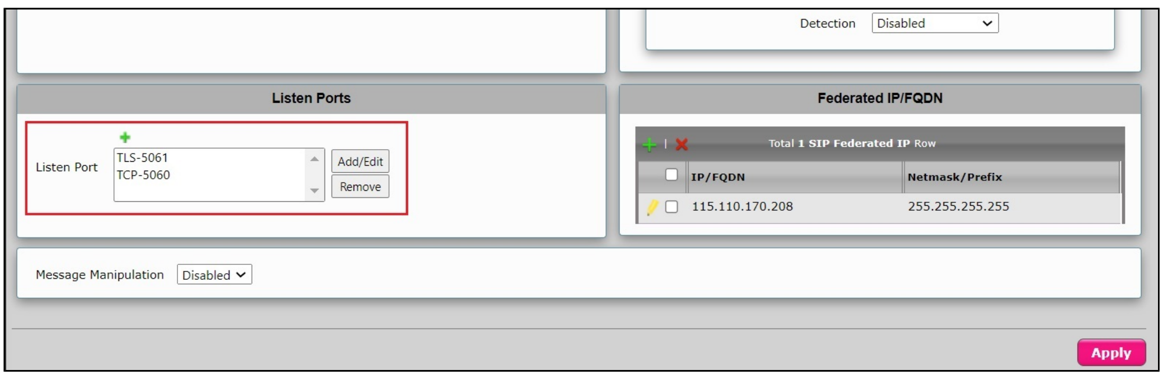
Change the settings on all the SGs as follows:

- Update the signaling group "**CUCM: Cisco CUCM**".
- Play Ringback - **Auto on 180/183** - Ringback is determined when processing 180 or 183.
- Early 183 - **Enable** - Specifies whether to send a SIP 183 response immediately after receiving an Invite message.
- Assign the interfaces for Signaling/Media Private IP to all the Signaling Groups accordingly.
- Enable Static NAT and map the respective IP addresses for both Signaling Groups.
- Add the listen port for TLS.









Note
You can configure SIP Trunk between Service provider and IP-PBX over UDP or TCP or TLS. Ribbon recommends use of TLS protocol to ensure security. Customers who do not wish to use TLS as preferred protocol can skip this section.

Update SIP Server Tables

SIP Server Tables contain information about the SIP devices connected to the SBC Edge. The entries in the tables provide information about the IP Addresses, ports, and protocols used to communicate with each server. The Table Entries also contain links to counters that are useful for troubleshooting.

From the **Settings** tab, navigate to **SIP > SIP Server Tables > CUCM: Cisco CUCM.** Click the expand ( ) icon next to the entry.

Keep Alive Frequency

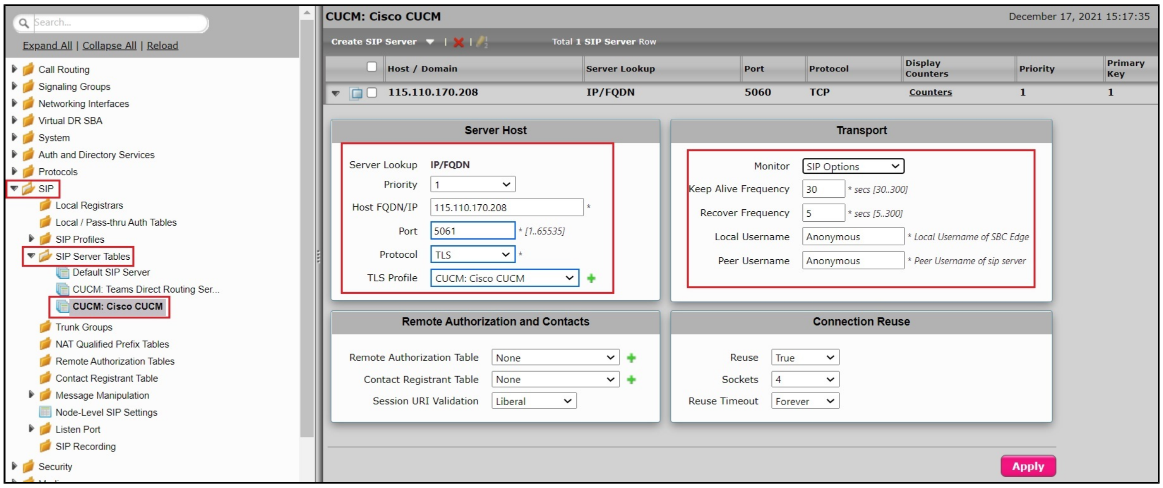
Specifies how often, in seconds, the SBC Edge queries the server with an OPTIONS message to determine the server's availability. Visible only when SIP Options is selected from the Monitor field. If the server does not respond, the SBC Edge marks the Signaling Group as down. When the server begins to respond to the OPTIONS messages again, it is marked as up. In this case, Keep Alive Frequency is set to 30 seconds.

Recover Frequency
Specifies frequency in seconds to check server to determine whether it has become available. Recovery Frequency is set to 5 seconds for this interop.

Local Username
Local user name of the SBC Edge system. Default entry: **Anonymous**. Visible only when **SIP Options** is selected from the **Monitor** field.

Peer Username
User name of the SIP Server. Visible only when **SIP Options** is selected from the **Monitor** field. The user can change Local and Peer Usernames according to their wishes.

1. Select Protocol **TLS** and Port 5061.
2. Attach the TLS Profile.



> **Note**
> Repeat the steps above to enable OPTIONS on other SIP Server Tables.

> **Note**
> During this interop the signaling group "**CUCM: Cisco CUCM**" Listen Port section is updated to TLS only. Update the signaling group accordingly.

> **Note**
> From the **System > Node-Level Settings** update the node level settings as required.

Transformation Table

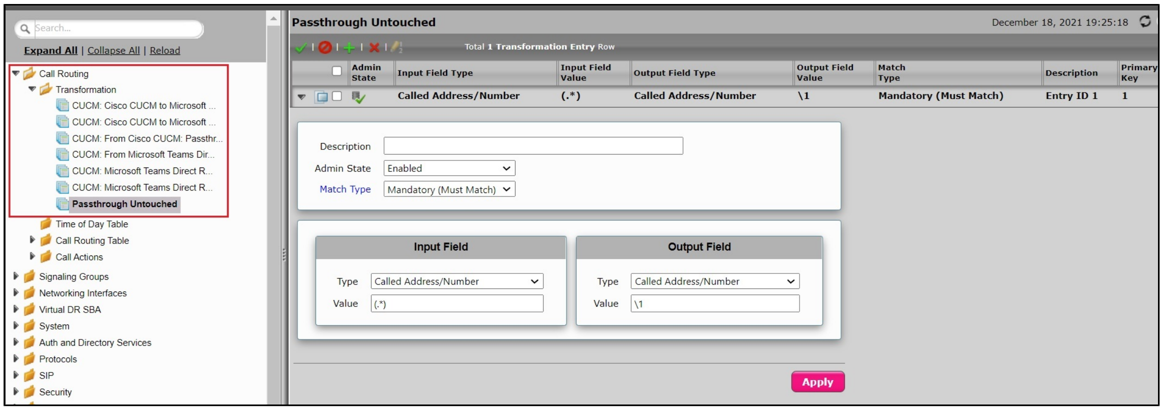
Transformation Tables facilitate the conversion of names, numbers and other fields when routing a call. They can, for example, convert a public PSTN number into a private extension number, or into a SIP address (URI). Every entry in a Call Routing Table requires a Transformation Table, and they are selected from there. In addition, Transformation tables are configurable as a reusable pool that Action sets can reference.

From the **Settings** tab, navigate to **Transformation.**

To Modify a Transformation Table

The Transformation Tables are created for Service Provider SIP Trunk through Easy Config Wizard. These are modified to allow specific patterns to reach the destination Signaling Group.

1. Click the **expand** ( ) icon next to the entry you wish to modify.
2. Modify the table's **Description** as desired.
3. Modify the Values from **Input field** and **Output field** as required.
4. Set the Match Type as **Optional (Match one)**.
5. Click **OK**.



Creating an Entry to a Message Transformation Table

For this interop, the entries are created based on the numbers associated with each endpoint. Users are free to select their own variables or Regular expressions.

1. Click the **Create( )** icon next to the table created in the previous step.
2. Provide the below details:

   Admin State:
   Enabled - The default state is Enabled.

   Match Type:
   Optional: Optional entries must match at least one of that Input Field type.
   When a call arrives at a Transformation Table, the incoming message contains a number of Informational Elements (IEs). These IEs include important call information such as: Called Address/Number, Called Extension, Calling Name, Redirecting Number and others. Each Informational Element is processed row by row in the Transformation Table.

   Value (Input/Output):
   Specifies the value to match against for the selected type. Depending on the type selected, values are free-form or selected from a menu.
3. Click **Apply**.

> **Note**
> For details on Transformation Table Entry configuration, refer to Creating and Modifying Entries to Transformation Tables. For call digit matching and manipulation through the use of regular expressions, refer to Creating Call Routing Logic with Regular Expressions.

> **Note**
> During this interop "Passthrough" transformation table only is used on both the sides.

Call Routing Table

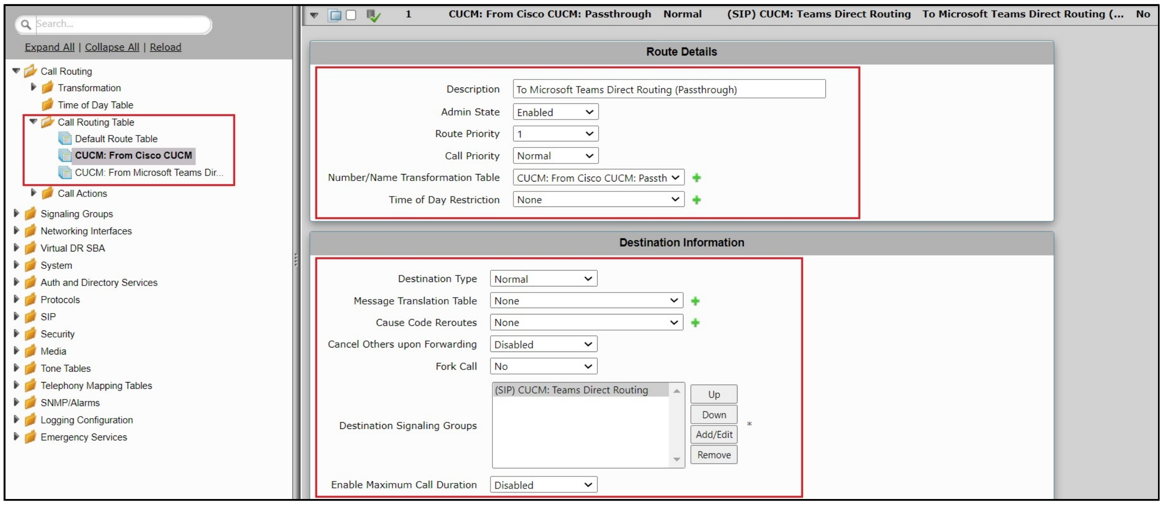
Call Routing allows carrying of calls between Signaling Groups. Routes are defined by Call Routing Tables, which allow for flexible configuration where calls are carried, and how they are translated.

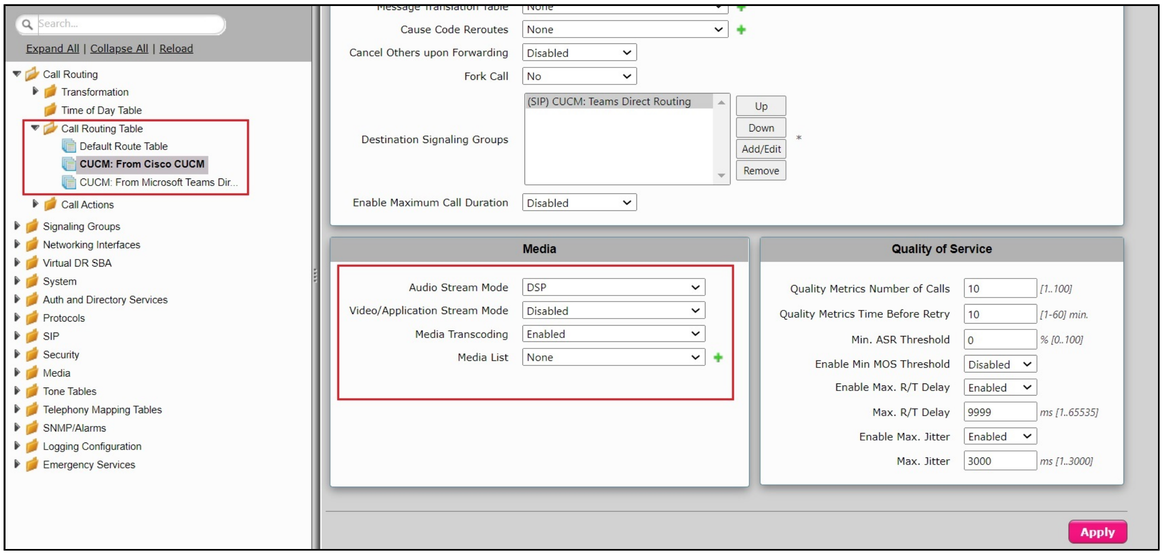
From the **Settings** tab, navigate to **Call Routing** > **Call Routing Table.**

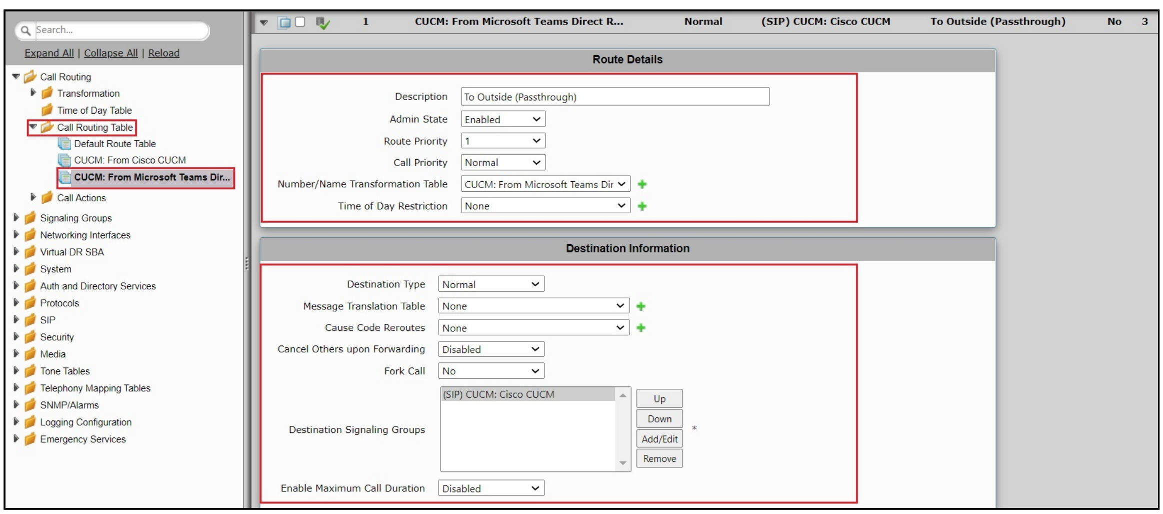
The Call Routing Tables are created to route the calls between IP-PBX (CUCM) -Service Provider through Easy Config Wizard. The user is allowed to modify these tables as per the requirement.

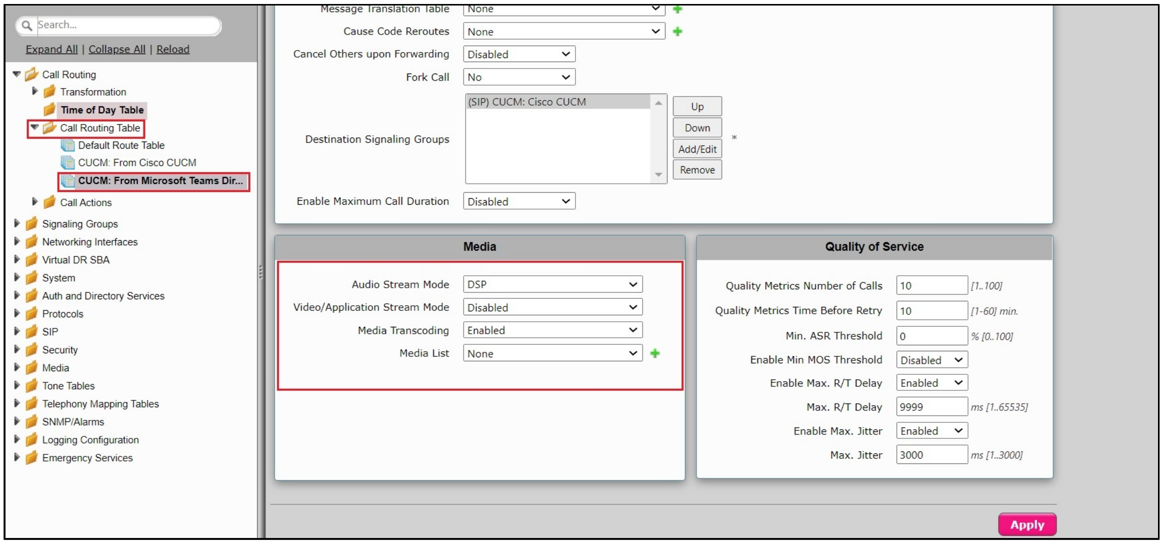
Modifying an Entry to a Call Routing Table

1. Click the **expand** ( ) icon next to the entry you wish to modify.
2. Edit the entry properties as required.


Search...
Expand All | Collapse All | Reload
Call Routing
Transformation
Time of Day Table
Call Routing Table
Default Route Table
CUCM: From Cisco CUCM
CUCM: From Microsoft Teams Dir...
Call Actions
Signaling Groups
Networking Interfaces
Virtual DR SBA
System
Auth and Directory Services
Protocols
SIP
Security
Media
Tone Tables
Telephony Mapping Tables
SNMP/Alarms
Logging Configuration
Emergency Services
1 CUCM: From Cisco CUCM: Passthrough Normal (SIP) CUCM: Teams Direct Routing To Microsoft Teams Direct Routing (... No
Route Details
Description To Microsoft Teams Direct Routing (Passthrough)
Admin State Enabled
Route Priority 1
Call Priority Normal
Number/Name Transformation Table CUCM: From Cisco CUCM: Passth
Time of Day Restriction None
Destination Information
Destination Type Normal
Message Translation Table None
Cause Code Reroutes None
Cancel Others upon Forwarding Disabled
Fork Call No
Destination Signaling Groups (SIP) CUCM: Teams Direct Routing
Up
Down
Add/Edit
Remove
Enable Maximum Call Duration Disabled


Search...
Expand All | Collapse All | Reload
Call Routing
Transformation
Time of Day Table
Call Routing Table
Default Route Table
CUCM: From Cisco CUCM
CUCM: From Microsoft Teams Dir...
Call Actions
Signaling Groups
Networking Interfaces
Virtual DR SBA
System
Auth and Directory Services
Protocols
SIP
Security
Media
Tone Tables
Telephony Mapping Tables
SNMP/Alarms
Logging Configuration
Emergency Services
Cause Code Reroutes None
Cancel Others upon Forwarding Disabled
Fork Call No
Destination Signaling Groups (SIP) CUCM: Teams Direct Routing
Up
Down
Add/Edit
Remove
Enable Maximum Call Duration Disabled
Media
Audio Stream Mode DSP
Video/Application Stream Mode Disabled
Media Transcoding Enabled
Media List None
Quality of Service
Quality Metrics Number of Calls 10 [1..100]
Quality Metrics Time Before Retry 10 [1-60] min.
Min. ASR Threshold 0 % [0..100]
Enable Min MOS Threshold Disabled
Enable Max. R/T Delay Enabled
Max. R/T Delay 9999 ms [1..65535]
Enable Max. Jitter Enabled
Max. Jitter 3000 ms [1..3000]
Apply


Search...
Expand All | Collapse All | Reload
Call Routing
Transformation
Time of Day Table
Call Routing Table
Default Route Table
CUCM: From Cisco CUCM
CUCM: From Microsoft Teams Dir...
Call Actions
Signaling Groups
Networking Interfaces
Virtual DR SBA
System
Auth and Directory Services
Protocols
SIP
Security
Media
Tone Tables
Telephony Mapping Tables
SNMP/Alarms
Logging Configuration
Emergency Services
1 CUCM: From Microsoft Teams Direct R... Normal (SIP) CUCM: Cisco CUCM To Outside (Passthrough) No 3
Route Details
Description To Outside (Passthrough)
Admin State Enabled
Route Priority 1
Call Priority Normal
Number/Name Transformation Table CUCM: From Microsoft Teams Dir
Time of Day Restriction None
Destination Information
Destination Type Normal
Message Translation Table None
Cause Code Reroutes None
Cancel Others upon Forwarding Disabled
Fork Call No
Destination Signaling Groups (SIP) CUCM: Cisco CUCM
Up
Down
Add/Edit
Remove
Enable Maximum Call Duration Disabled



Creating an Entry to a Call Routing Table

Call Routing Tables are one of the central connection points of the system, linking Transformation Tables, Message Translations, Cause Code Reroute Tables, Media Lists and the three types of Signaling Groups (ISDN, SIP and CAS).

In the SBC Edge, call routing occurs between **Signaling Groups**.

In order to route any call to or from a call system connected to the SBC, you must first configure a Signaling Group to represent that device or system. The following list illustrates the hierarchical relationships of the various Telephony routing components of a SBC call system:

- Signaling Group describes the source call and points to a routing definition known as a Call Route Table
- Call Route Table contains one or more Call Route Entries
- Call Route Entries points to the destination Signaling Group(s)

Each call routing entry describes how to route the call and also points to a Transformation Table that defines the conversion of names, numbers and other fields when routing a call.

To create an entry:

1. Click the **Create Routing Entry** (+) icon.
2. Set the following fields:

   Admin State:
   Enabled - Enables the call route entry for routing the call, displays in configuration header as .

   Route Priority:
   Priority of the route from 1 (highest) to 10 (lowest). Higher priority routes are matched against before lower priority routes regardless of the order of the routes in the table.

   Number/Name Transformation Table:
   Specifies the Transformation Table to use for this routing entry. This drop down list is populated from the entries in the Transformation Table.

   Destination Signaling Groups:
   Specifies the Signaling Groups used as the destination of calls. The first operational Signaling Group from the list is chosen to place the call. Click the Add/Edit button to select the destination signaling group.

   Audio Stream Mode:
   DSP (default entry): The SBC uses DSP resources for media handling (transcoding) but it does not facilitate the capabilities/features between endpoints that are not supported within the SBC (codec/capability mismatch). When DSP is configured, the Signaling Groups enabled to support DSP are attempted in order.

   Media Transcoding:
   Enabled: Enable Transcoding on SIP-to-SIP calls.
3. Click **Apply**.

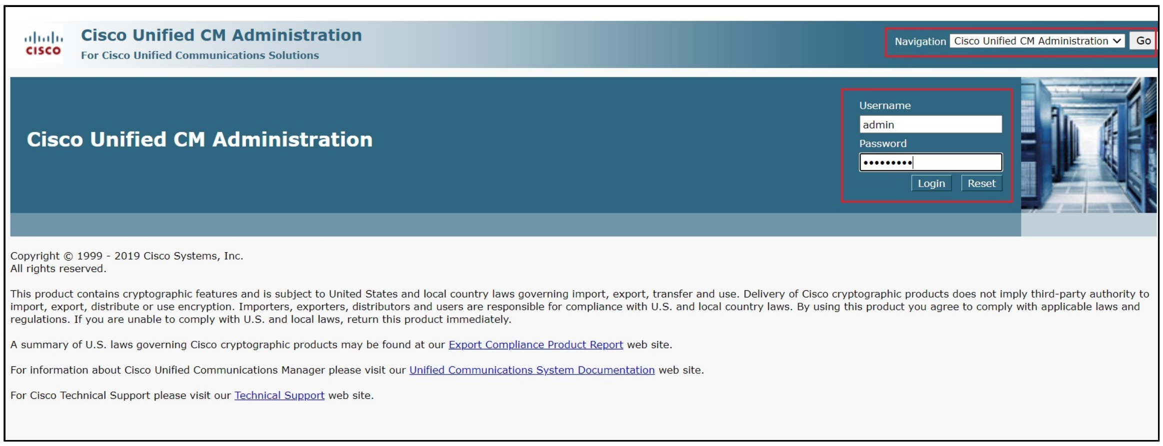
> **Note**
> During this interop only "Passthrough" transformation table is used for call routing and removed other transformation entries.

> **Warning**
> In Call routing table "**Audio Stream Mode"** by default is DSP mode. It is recommended to use the default DSP mode configuration.

Section B: CUCM Configuration

Accessing CUCM (Cisco Unified CM Administration)

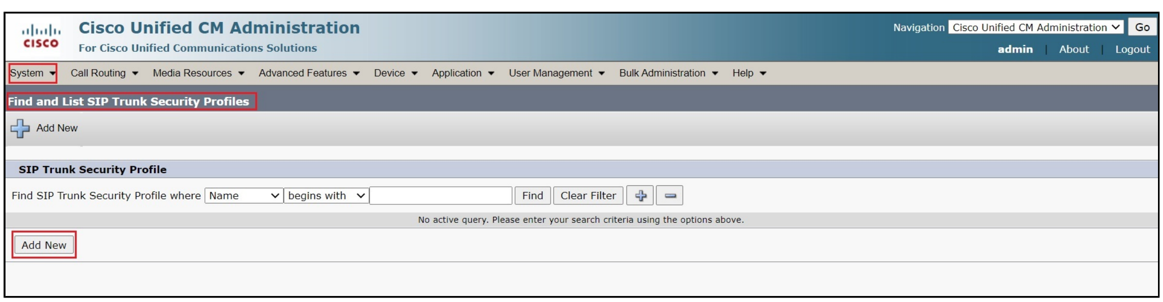
1. Open Browse and enter the CUCM IP Address.
2. Select **Cisco Unified CM Administration** from the Navigation drop-down.
3. Provide the credentials and click **Login**.



Configure SIP Trunk Security Profile

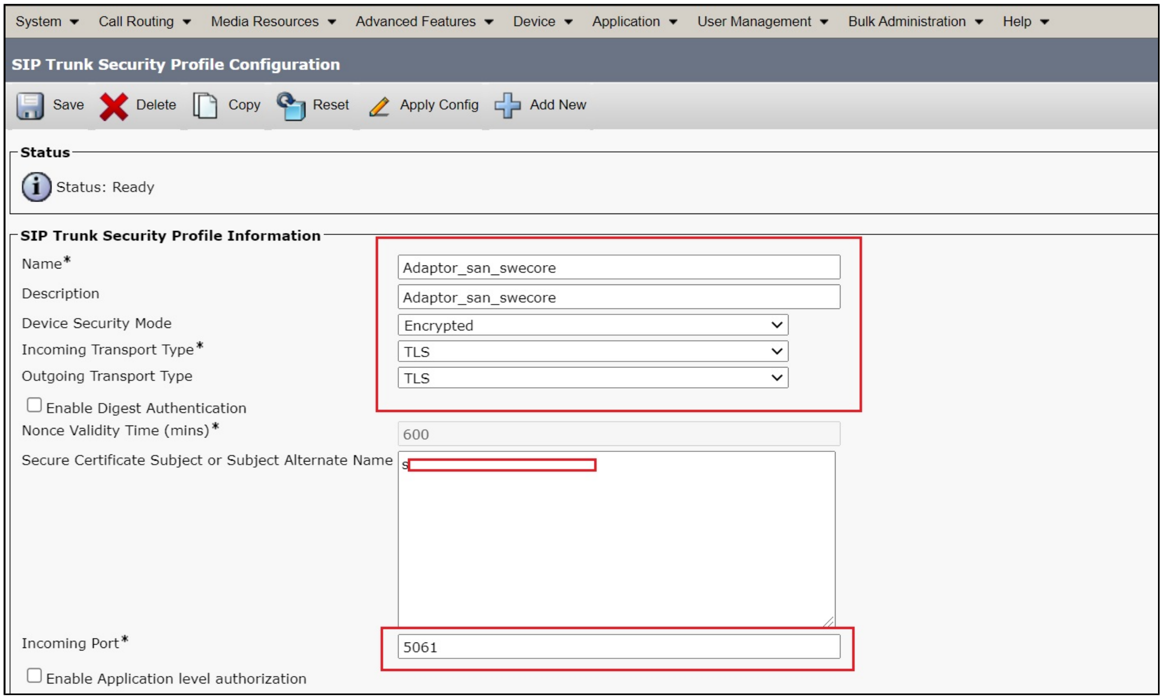
Unified Communications Manager Administration groups security-related settings for the SIP trunk to allow you to assign a single security profile to multiple SIP trunks. Security-related settings include device security mode, digest authentication, and incoming/outgoing transport type settings.

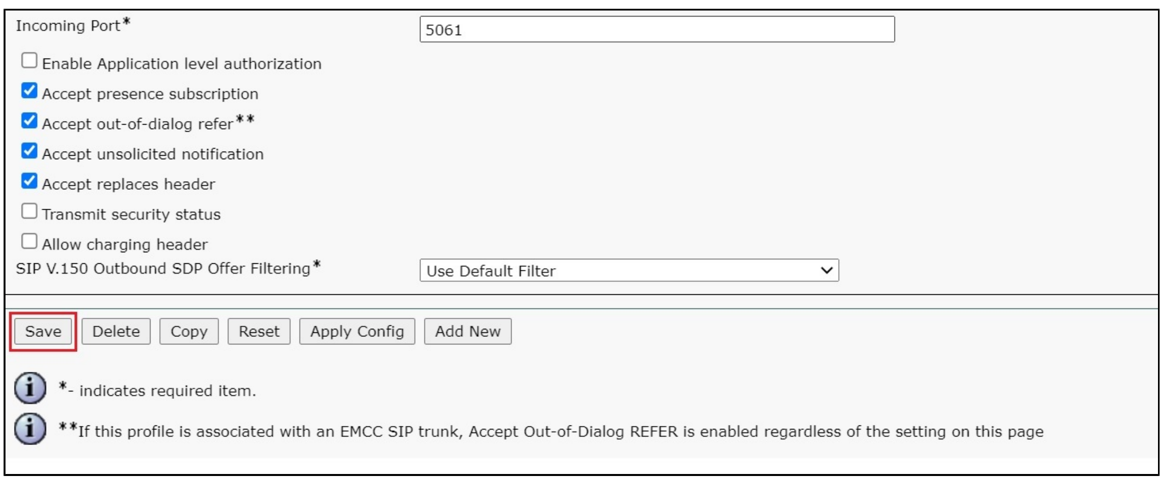
- From Cisco Unified CM Administration, navigate to **System > Security > SIP Trunk Security Profile.**
- Click **Add New**.



- Provide the desired Name and Description.
- Choose **Secure** from Device Security Mode.
- From Incoming Transport Type, select **TLS.**
  - When Device Security Mode is Non Secure, TCP+UDP specifies the transport type.
- Select Outgoing Transport Type as **TLS**.
- Click **Save**.

> **Note**
> Customers are free to choose any transport medium depends on their requirements. Ribbon strongly recommends use of secure TLS protocol.



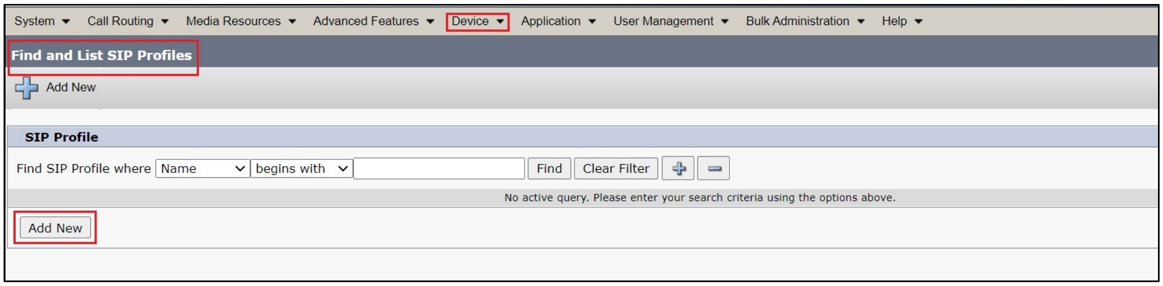


> **Note**
> For more information on regarding CSR and Certificate generation for CUCM, refer to https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-border-element/215412-configure-sip-tls-between-cucm-cube-cube.html

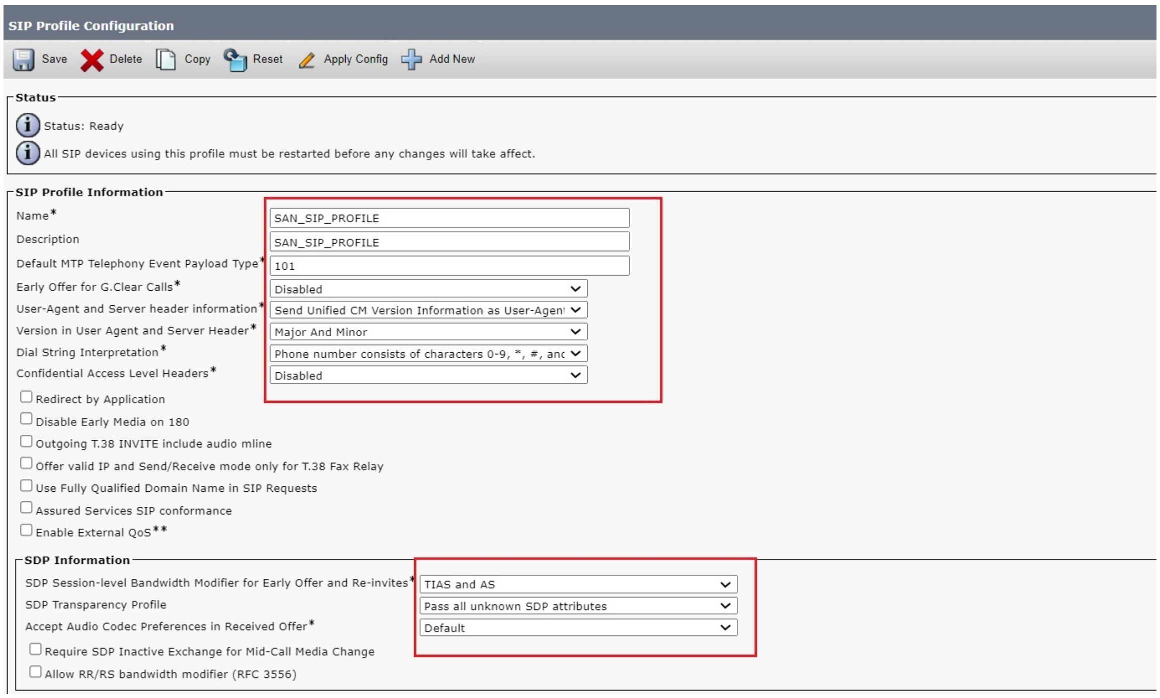
Configure SIP Profiles

A SIP profile comprises the set of SIP attributes that are associated with SIP trunks and SIP endpoints. SIP profiles include information such as name, description, timing, retry, call pickup URI, and so on. The profiles contain some standard entries that you cannot delete or change.

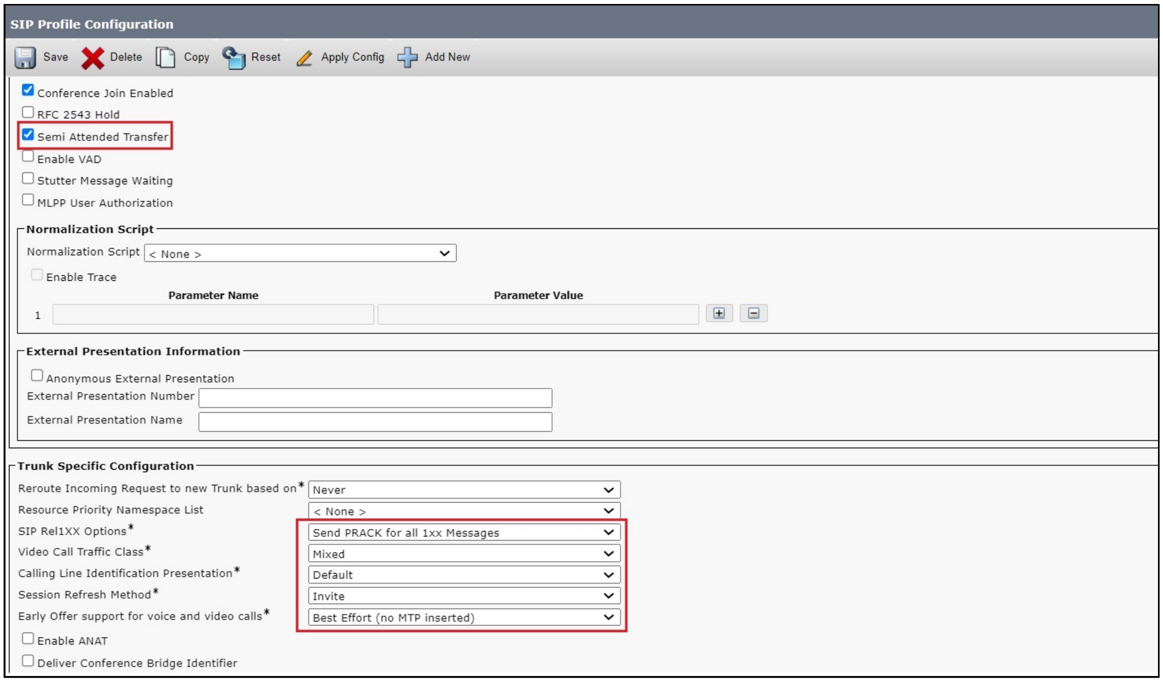
- From Cisco Unified CM Administration, navigate to **Device > Device Settings > SIP Profile.**
- Click **Add New**.



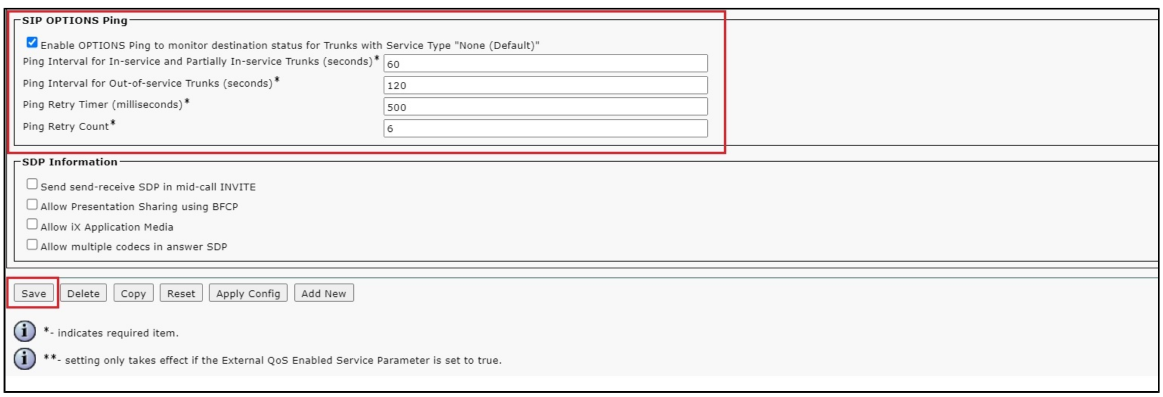
- Enter a name to identify the SIP profile.
- Provide description to identify the purpose of the SIP profile.



- From SIP Rel1XX Options drop-down, choose **Send PRACK for all 1xx Messages**.
- From Early Offer support for voice and video calls drop-down, choose Best Effort (no MTP inserted).
  - Provide Early Offer for the outbound call only when caller side's media port, IP and codec information is available.
  - Provide Delayed Offer for the outbound call when caller side's media port, IP and codec information is not available. No MTP is inserted to provide Early Offer in this case.

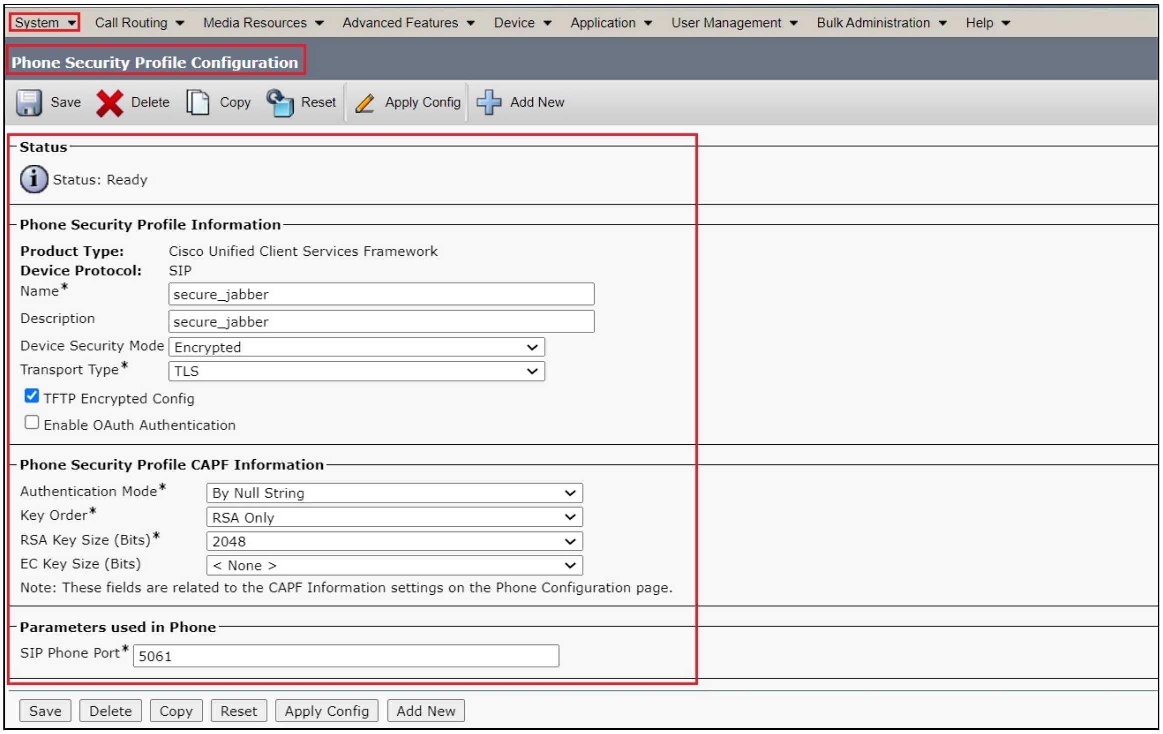


- Enable **SIP OPTIONS Ping**.
  - SIP OPTIONS are requests to the configured destination address on the SIP trunk.
- Click **Save**.



Configure Phone Security Profiles

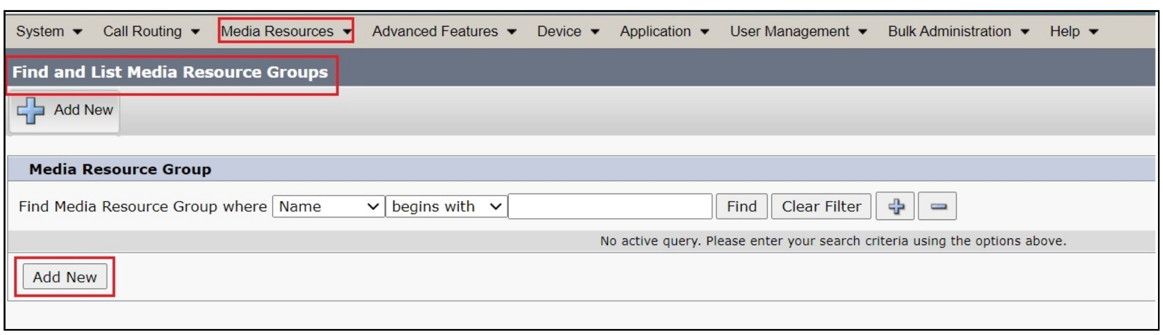
- From Cisco Unified CM Administration, navigate to **System> Security > Phone Security Profile.**
- Click **Add New**.
- Provide the required details.
- Click **Save**.



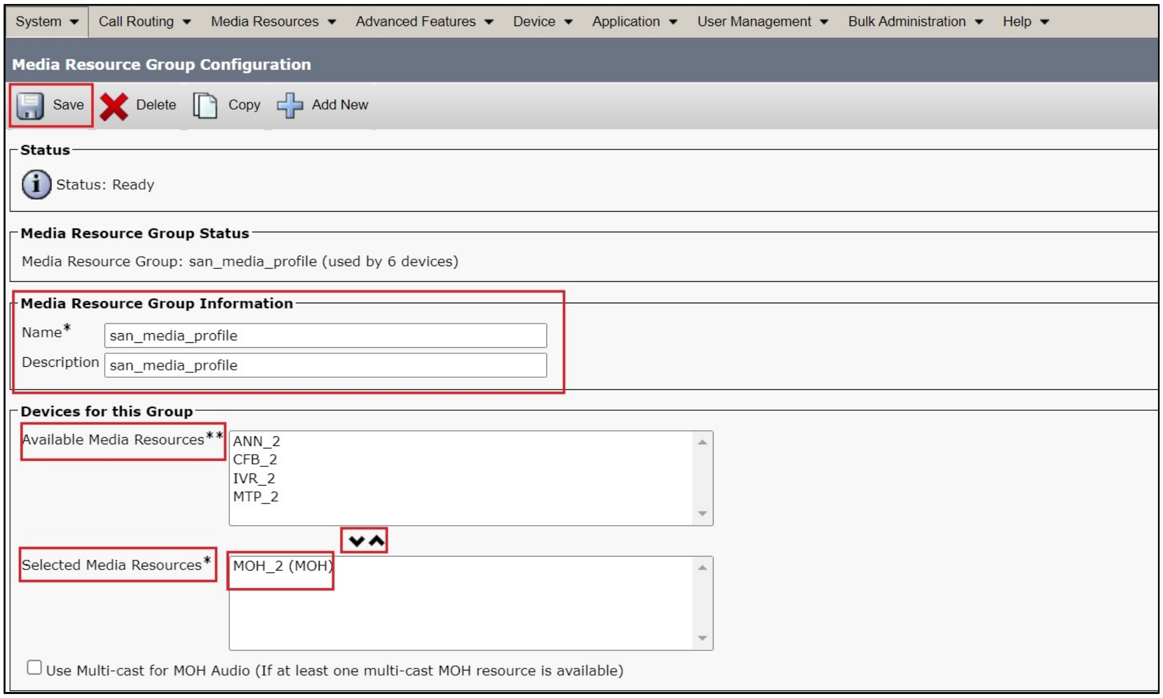
Configure Media Resource Group

Media resource management comprises working with media resource groups and media resource group lists. Media resource management provides a mechanism for managing media resources, so all Cisco Unified Communications Managers within a cluster can share them. Media resources provide conferencing, transcoding, media termination, annunciator, and music on hold services.

- From Cisco Unified CM Administration, navigate to **Media Resources > Media Resource Group.**
- Click **Add New**.



- Enter a unique name in this required field to identify the media resource group.
- Enter a description for the media resource group.
- To add a media resource for this media resource group, choose one (MoH_2 in this case) from the available Media Resources list and click the down arrow. After a media resource is added, its name moves to the Selected Media Resources pane.

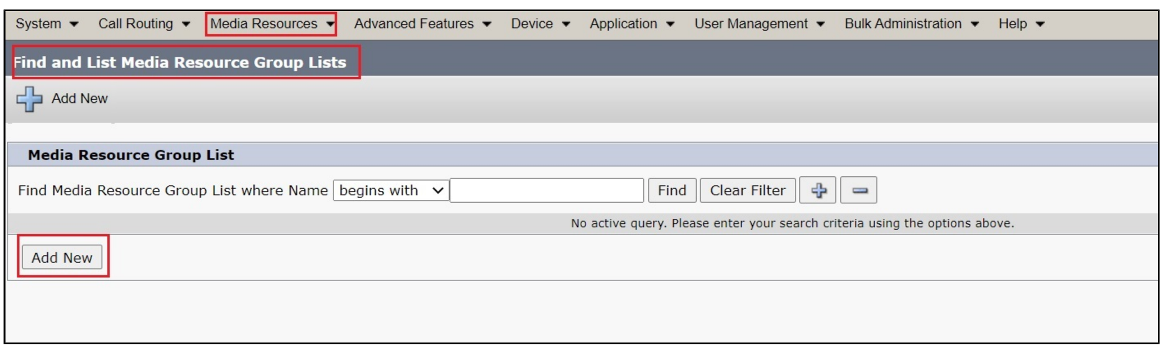


- Click **Save**.

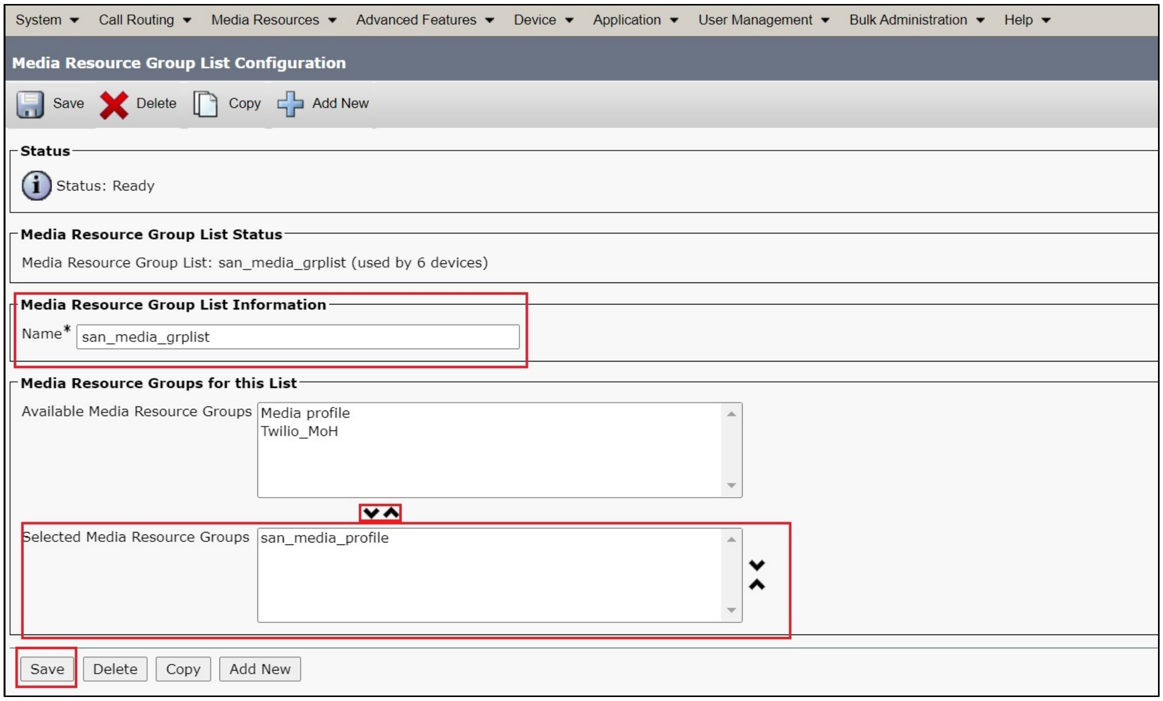
Configure Media Resource Group List

A Media Resource Group List provides a prioritized grouping of media resource groups. An application selects the required media resource, such as a music on hold server, from among the available media resources according to the priority order that is defined in a Media Resource Group List.

- From Cisco Unified CM Administration, navigate to **Media Resources > Media Resource Group List** menu path to configure media resource group lists.
- Click **Add New**.



- Enter a unique name in this required field to identify the Media Resource Group List.
- Choose the Media Resource Group created in the previous step from the Available Media Resource Groups list and click the down arrow that is located between the two panes. After a media resource group is added, its name moves to the Selected Media Resource Groups pane.

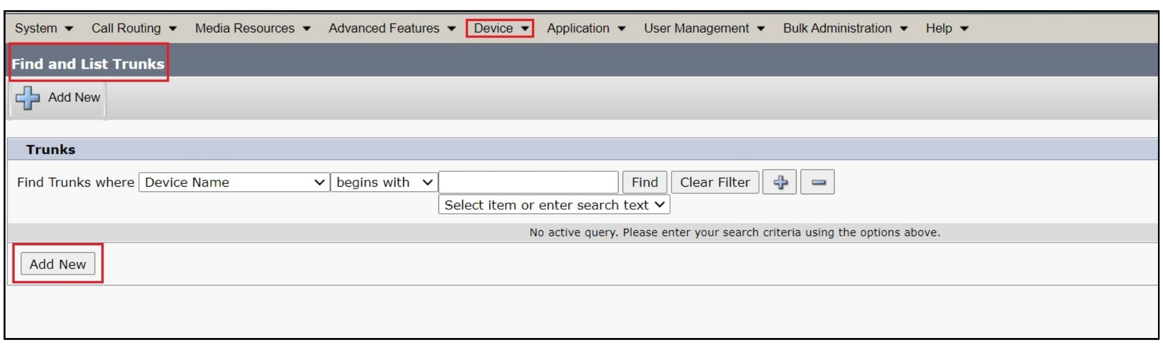


- Click **Save**.

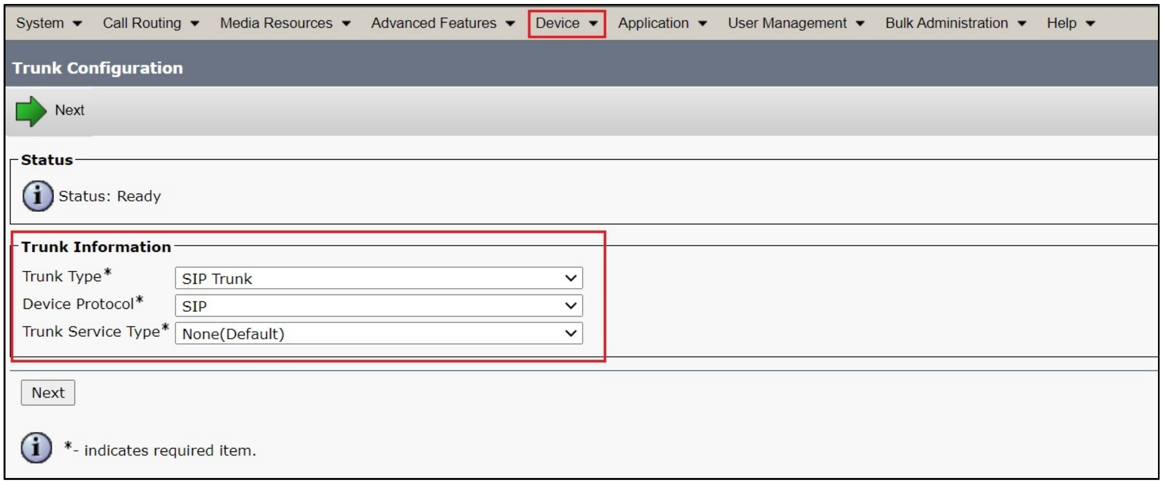
Trunk Configuration

Use a trunk device to configure a logical route to a SIP network.

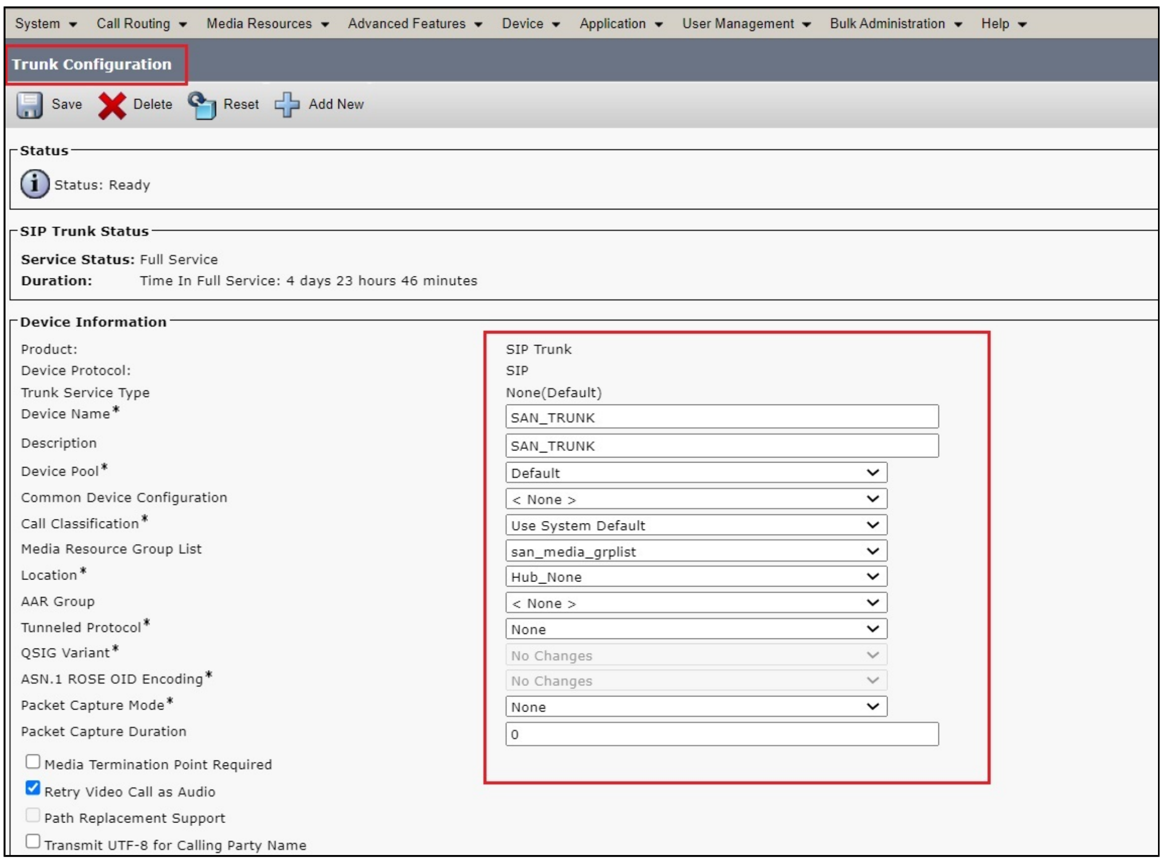
- From Cisco Unified CM Administration, choose **Device > Trunk.**
- Click **Add New**.



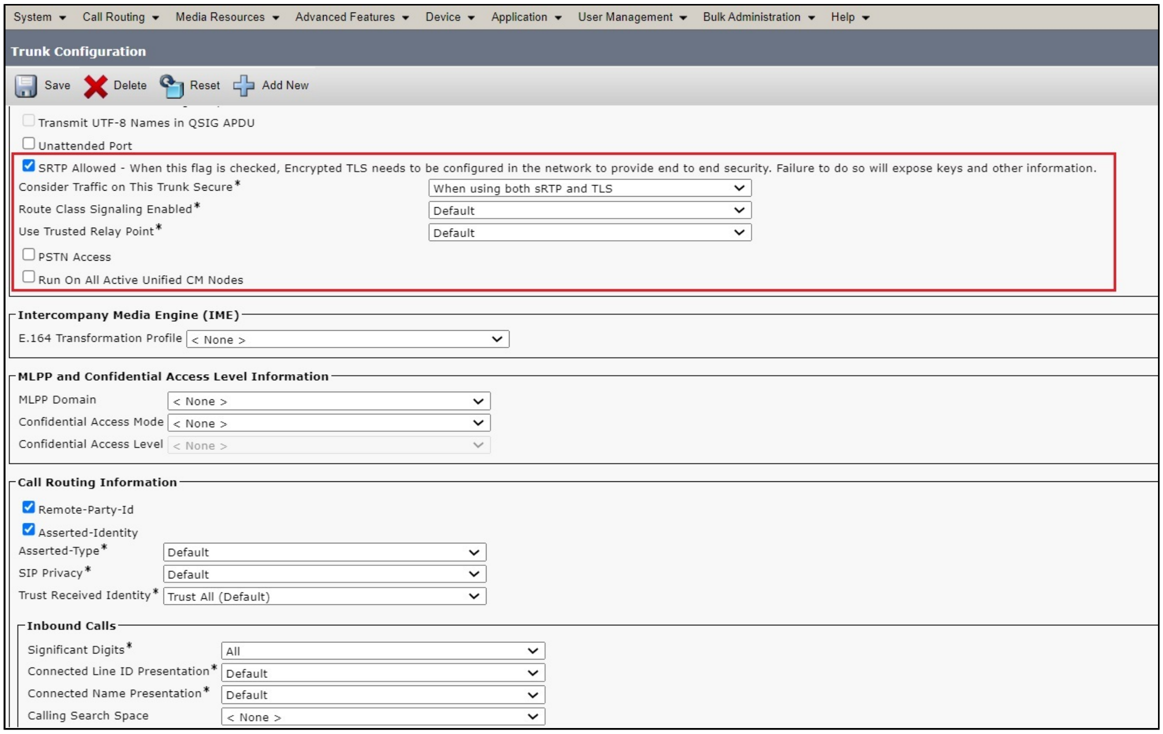
- From the Trunk Type drop-down list, choose **SIP Trunk**.
- Choose **SIP** from Device Protocol drop-down.
- From Trunk Service Type, select the default value (None).
- Click **Next**.

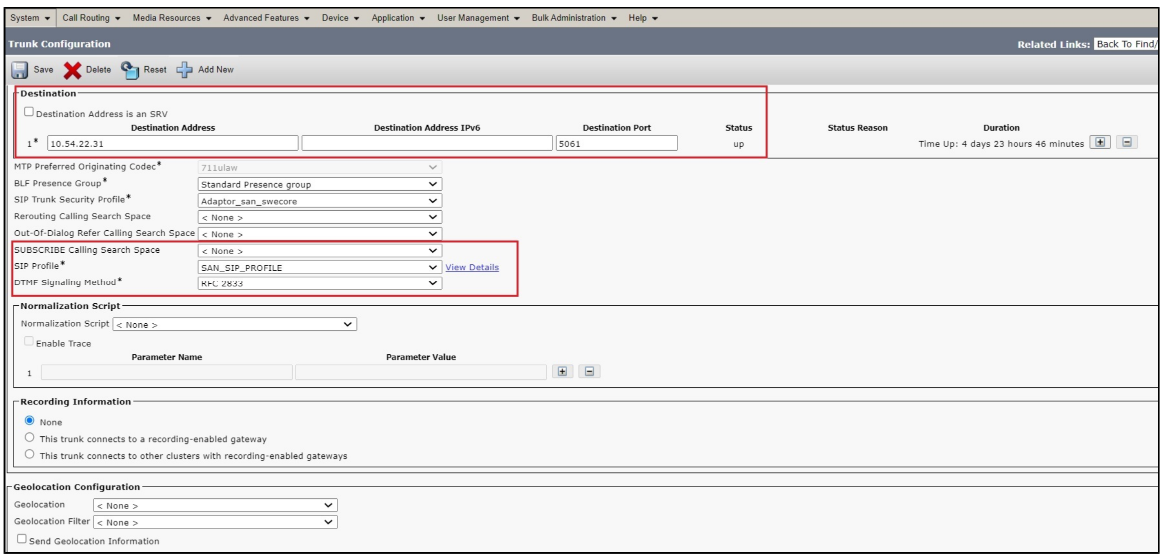


- Enter a unique identifier for the trunk.
- Enter a descriptive name for the trunk.
- Choose the Default Device Pool.
- Choose the Media Resource Group List created in the previous step.



- Provide the destination address.
  - The Destination Address represents the remote SIP peer with that this trunk will communicate.
  - SIP trunks only accept incoming requests from the configured Destination Address and the incoming port that is specified in the SIP Trunk Security Profile that is associated with this trunk.
- Choose the **SRTP Allowed** (only when SIP Trunk profile is created as TLS)
- Choose the **SIP Trunk Security Profile** created to apply to the SIP trunk.
- Select the **SIP Profile** created from the list.
- Choose **RFC 2833** as DTMF Signaling Method.
- Click **Save**.





- Click **Save**
- Click the **Reset** button.

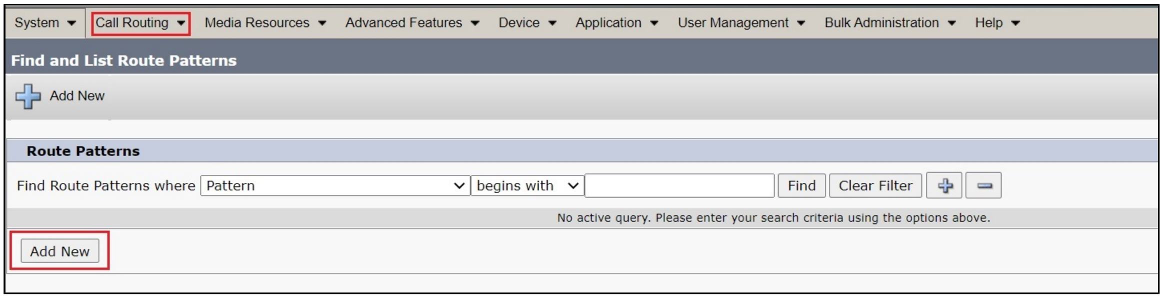
- Reset, Restart and Close the window. Refresh the SIP trunk page and wait until the Server status changes from Unknown to Full Service.

> **Note**
> Resetting/restarting a SIP device does not physically reset/restart the hardware, it only reinitializes the configuration that is loaded by Cisco Unified Communications Manager.
>
> For SIP trunks, Restart and Reset behave the same way, so all active calls will disconnect when either choice is pressed.

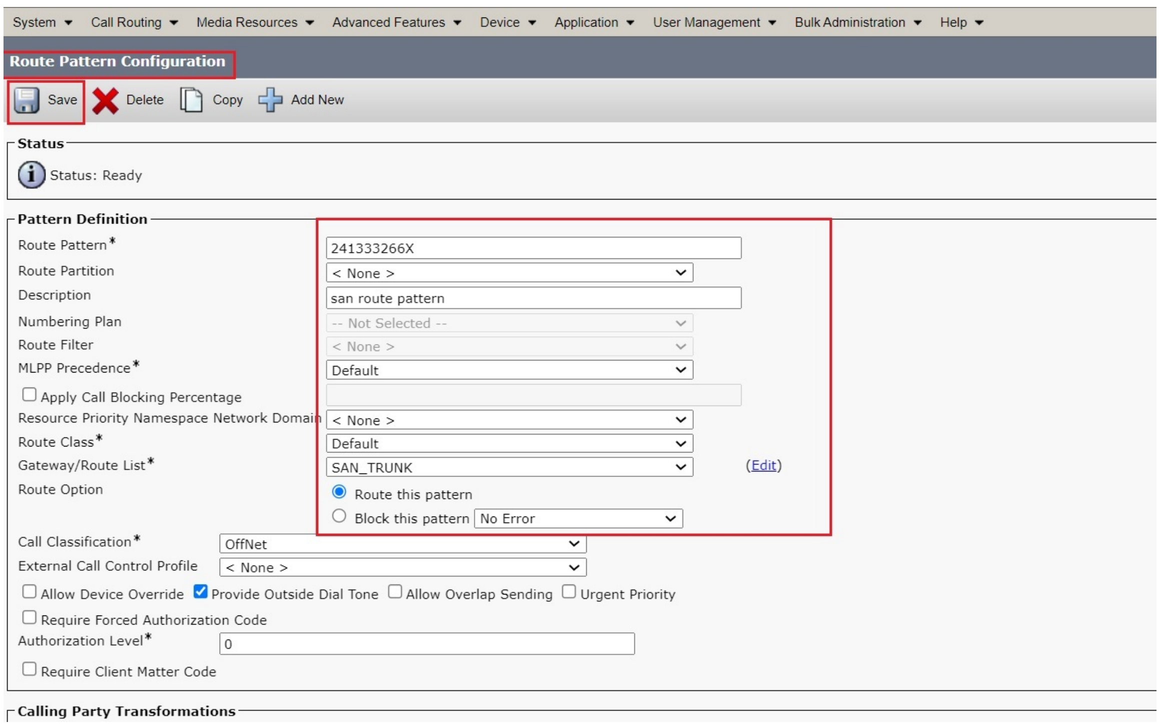
Configure Call Routing

A route pattern comprises a string of digits (an address) and a set of associated digit manipulations that route calls to a route list or a gateway. Route patterns provide flexibility in network design. They work in conjunction with route filters and route lists to direct calls to specific devices and to include, exclude, or modify specific digit patterns.

- In Cisco Unified Communications Manager Administration, use the **Call Routing > Route/Hunt > Route Pattern** menu path to configure route patterns.
- Click **Add New**.



- Enter the route pattern, including numbers and wildcards (do not use spaces); for example, for NANP, enter 9.@ for typical local access or 8XXX for a typical private network numbering plan. Valid characters include the uppercase characters A, B, C, and D and \+, which represents the international escape character +.
- Configure the Route Pattern as below.
- Choose SIP Trunk created from the gateway or route list drop-down to add the route pattern.

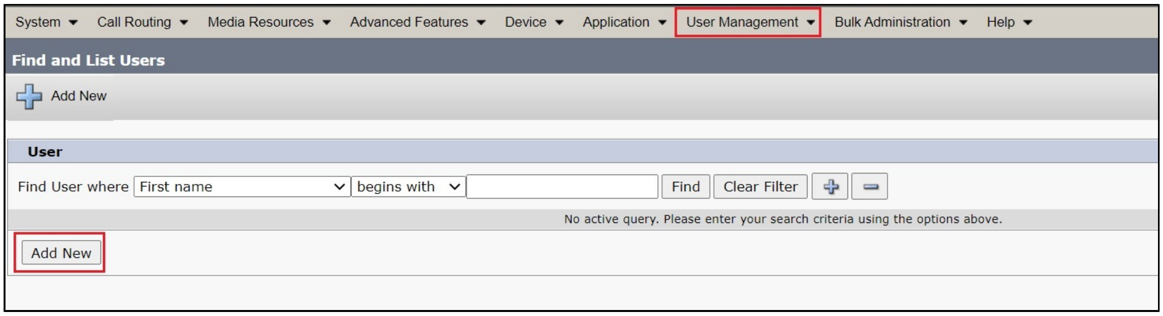


- Click **Save**.

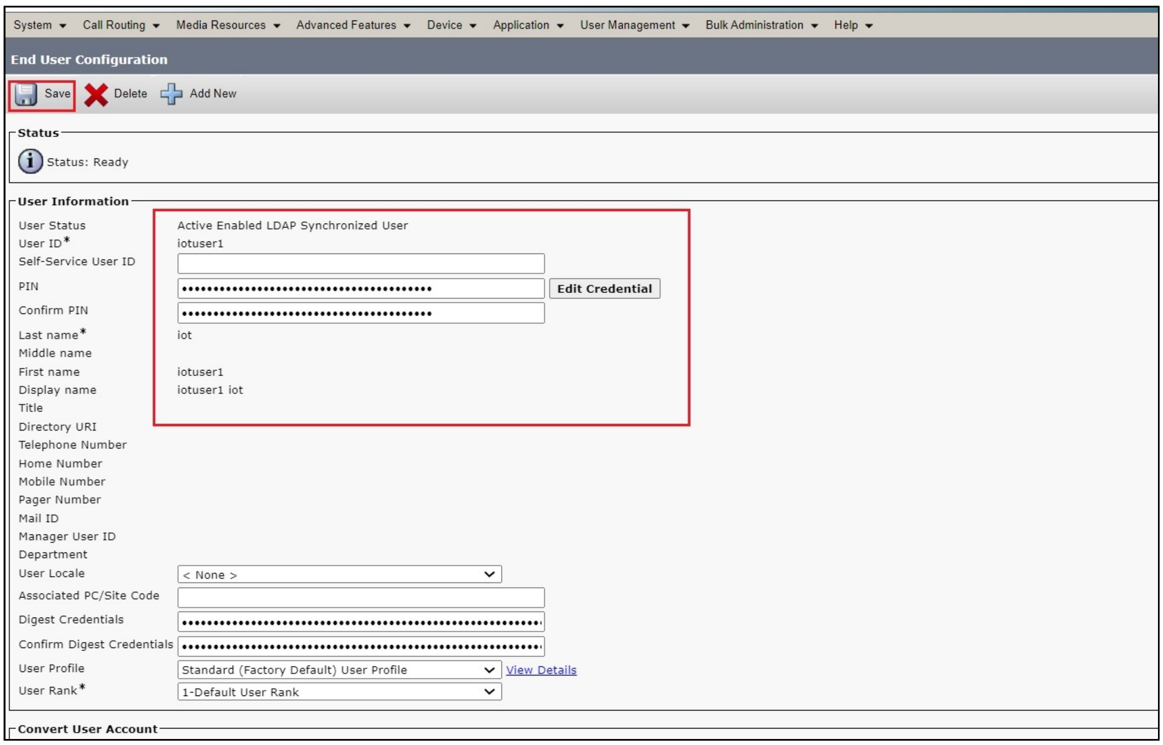
Configure End Users

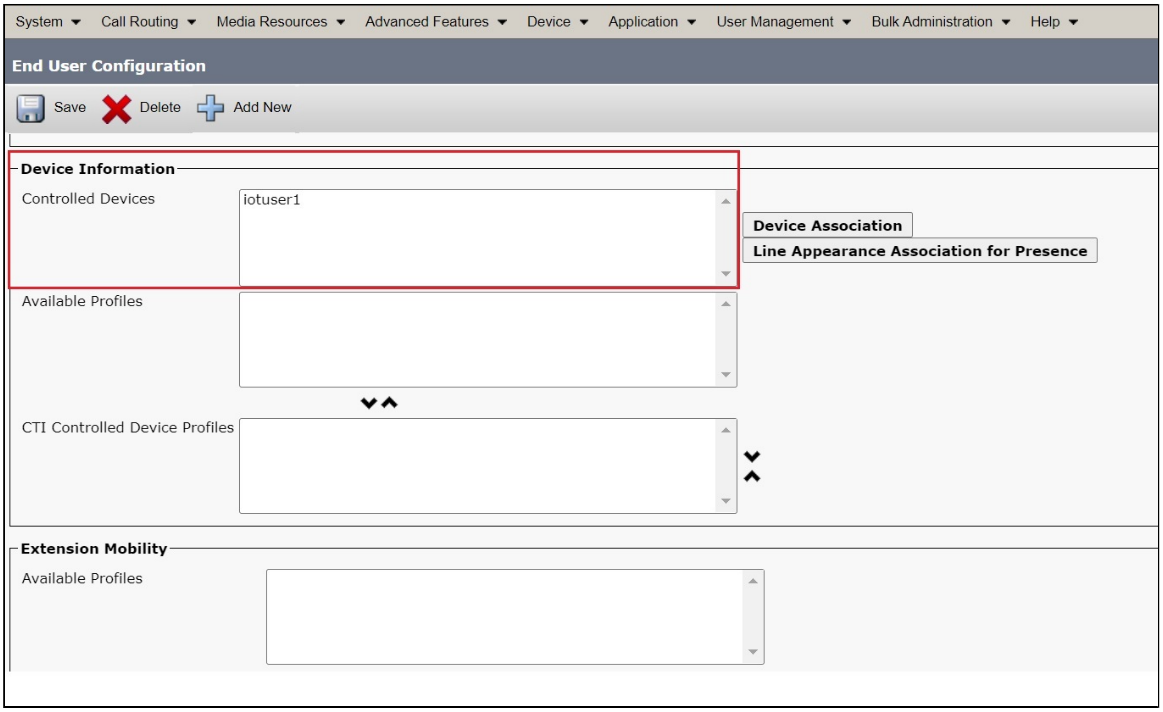
The End User Configuration window allows you to add, search, display, and maintain information about Unified Communications Manager end users. End users can control phones after you associate a phone in the End User Configuration window.

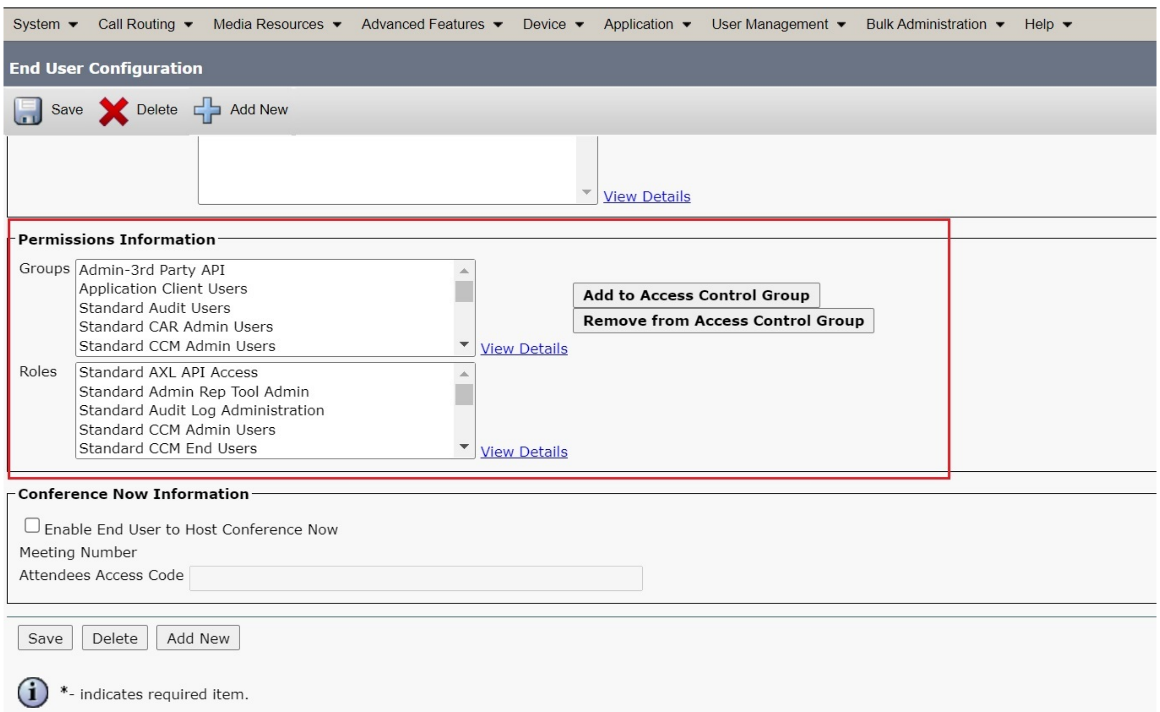
- In Cisco Unified CM Administration, use the **User Management > End User** menu path to configure end users.
- Click **Add New**.



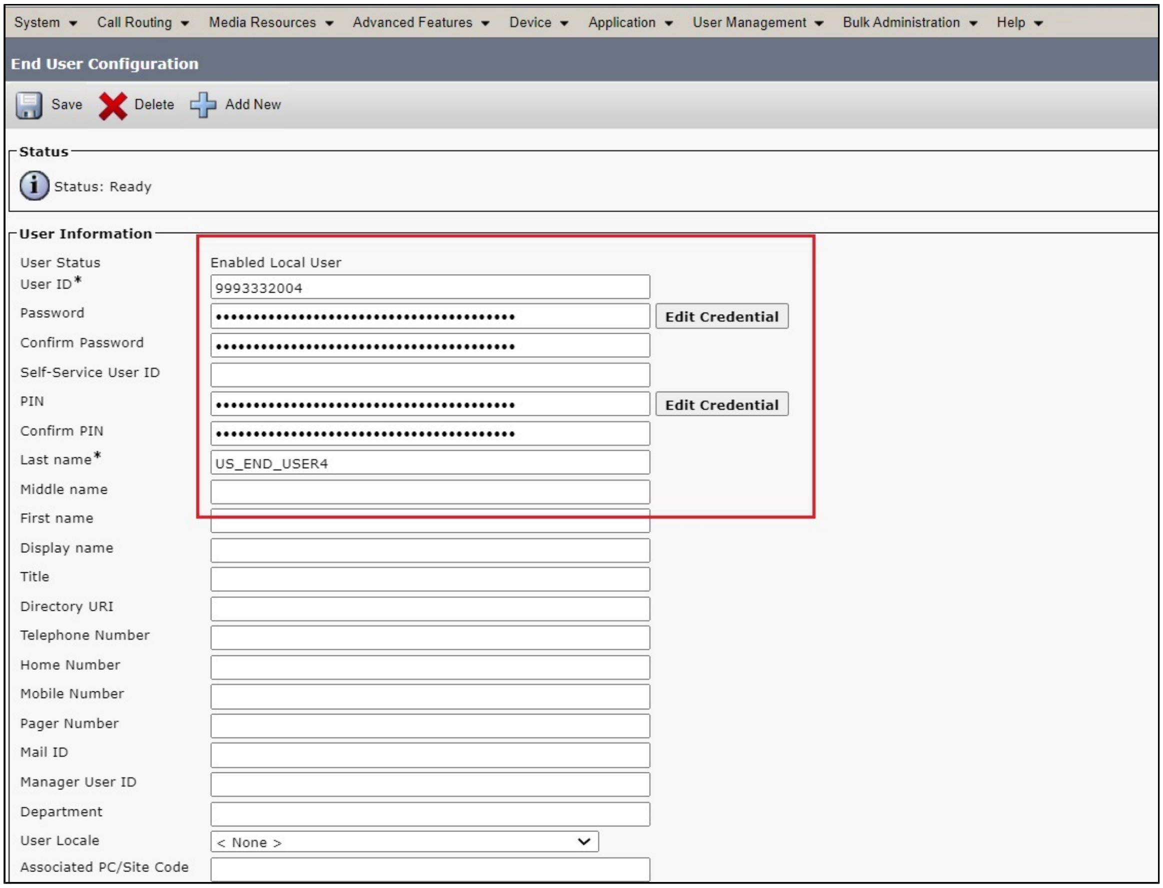
- We have two examples taken to configure End Users (Cisco Jabber and Cisco DX650).
- Enter the unique end user identification name.
- Enter alphanumeric or special characters for the end user password and confirm the same.
- Enter numeric characters for the end user PIN and confirm.
- Enter the end user last name.
- For Digest Credentials, enter a string of alphanumeric characters and confirm.
- For Cisco Jabber as below.

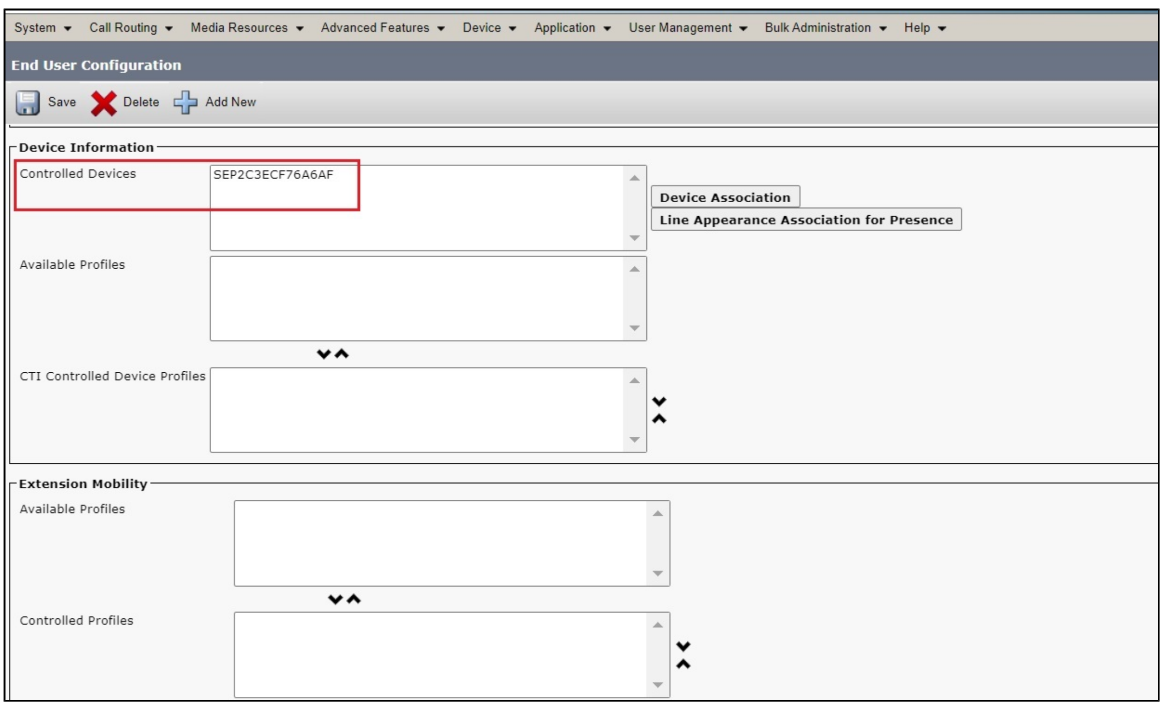


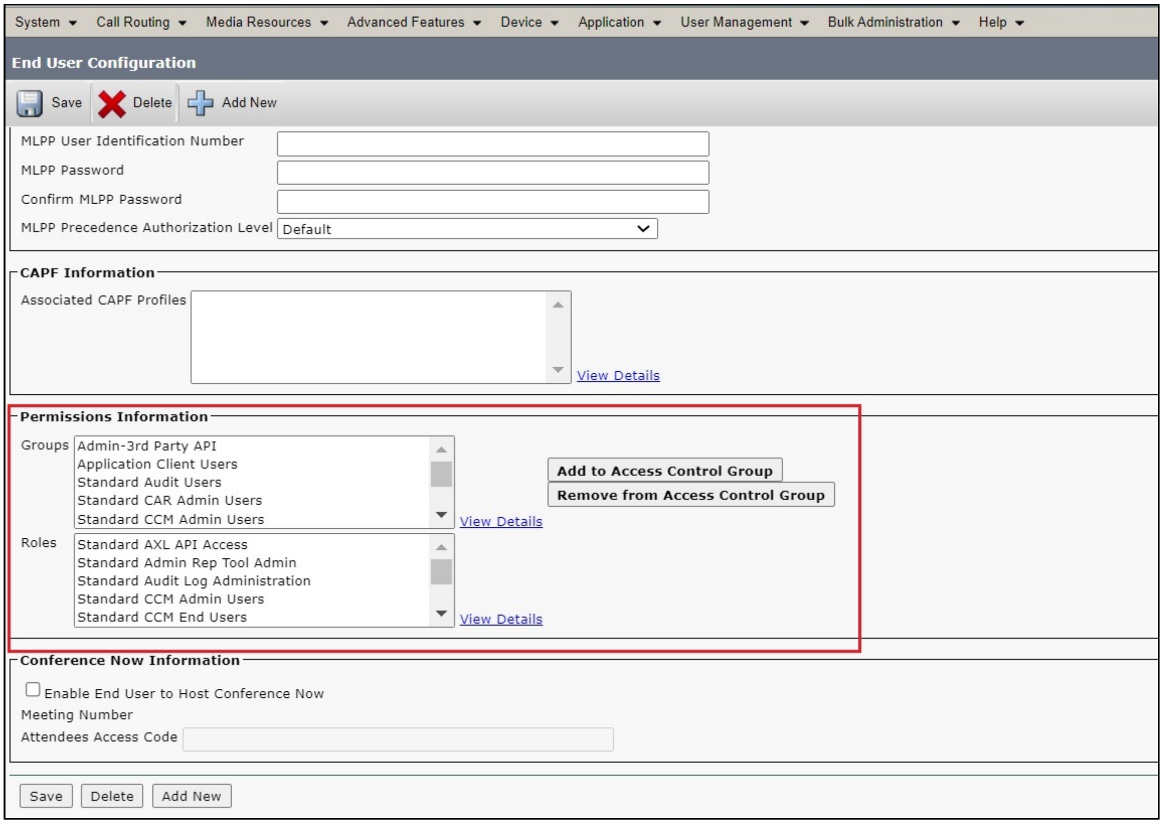




- For Cisco DX650 as below.

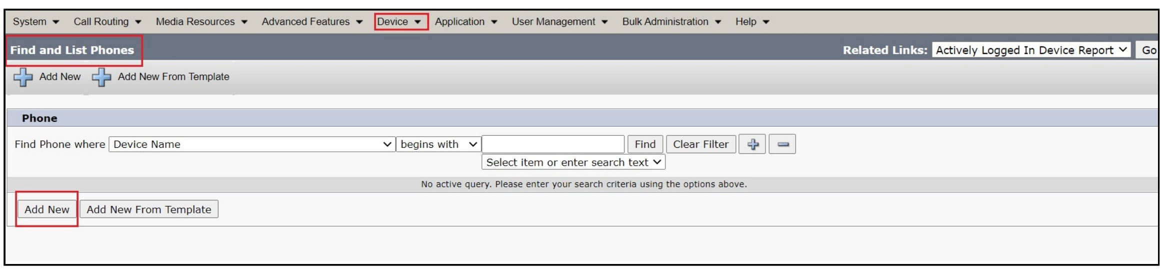

System
Call Routing
Media Resources
Advanced Features
Device
Application
User Management
Bulk Administration
Help
End User Configuration
Save
Delete
Add New
Status
Status: Ready
User Information
User Status
Enabled Local User
User ID*
9993332004
Password
Edit Credential
Confirm Password
Self-Service User ID
PIN
Edit Credential
Confirm PIN
Last name*
US_END_USER4
Middle name
First name
Display name
Title
Directory URI
Telephone Number
Home Number
Mobile Number
Pager Number
Mail ID
Manager User ID
Department
User Locale
< None >
Associated PC/Site Code


System
Call Routing
Media Resources
Advanced Features
Device
Application
User Management
Bulk Administration
Help
End User Configuration
Save
Delete
Add New
Device Information
Controlled Devices
SEP2C3ECF76A6AF
Device Association
Line Appearance Association for Presence
Available Profiles
CTI Controlled Device Profiles
Extension Mobility
Available Profiles
Controlled Profiles

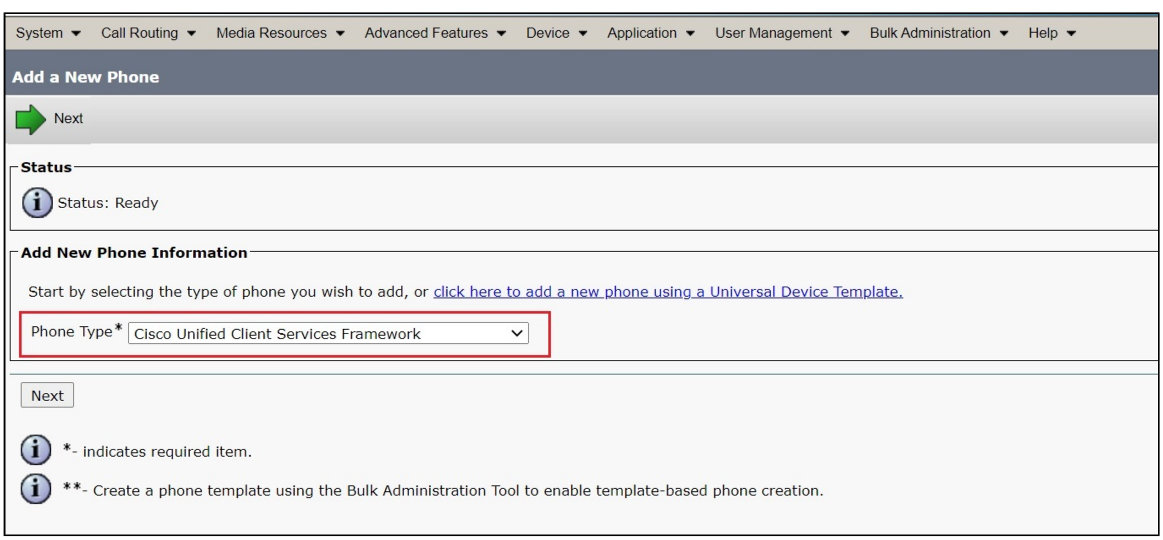


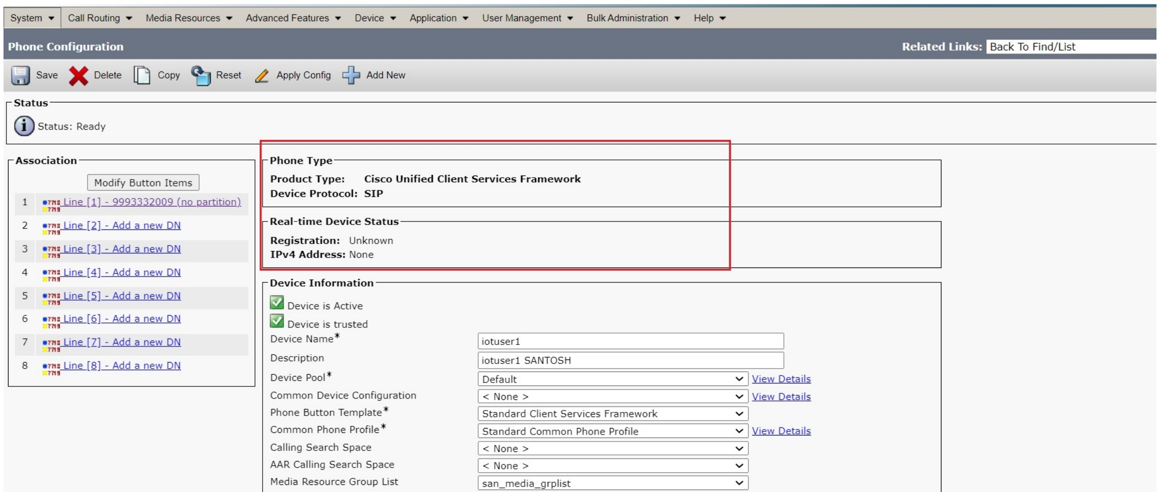
Phone Setup

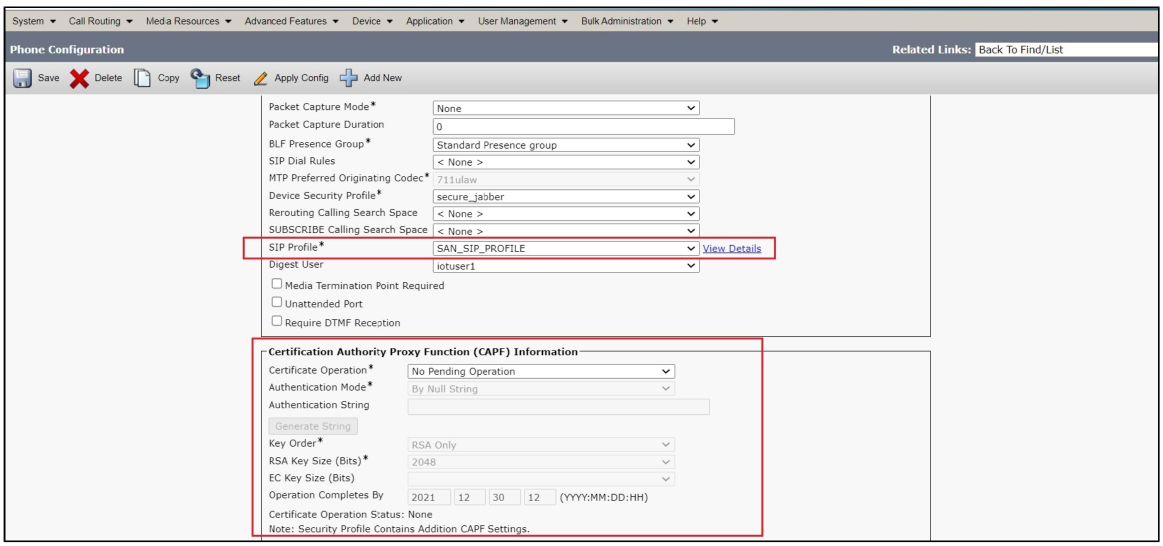
- In Cisco Unified Communications Manager Administration, use the **Device > Phone** menu path to configure phones.
- Click **Add New**.

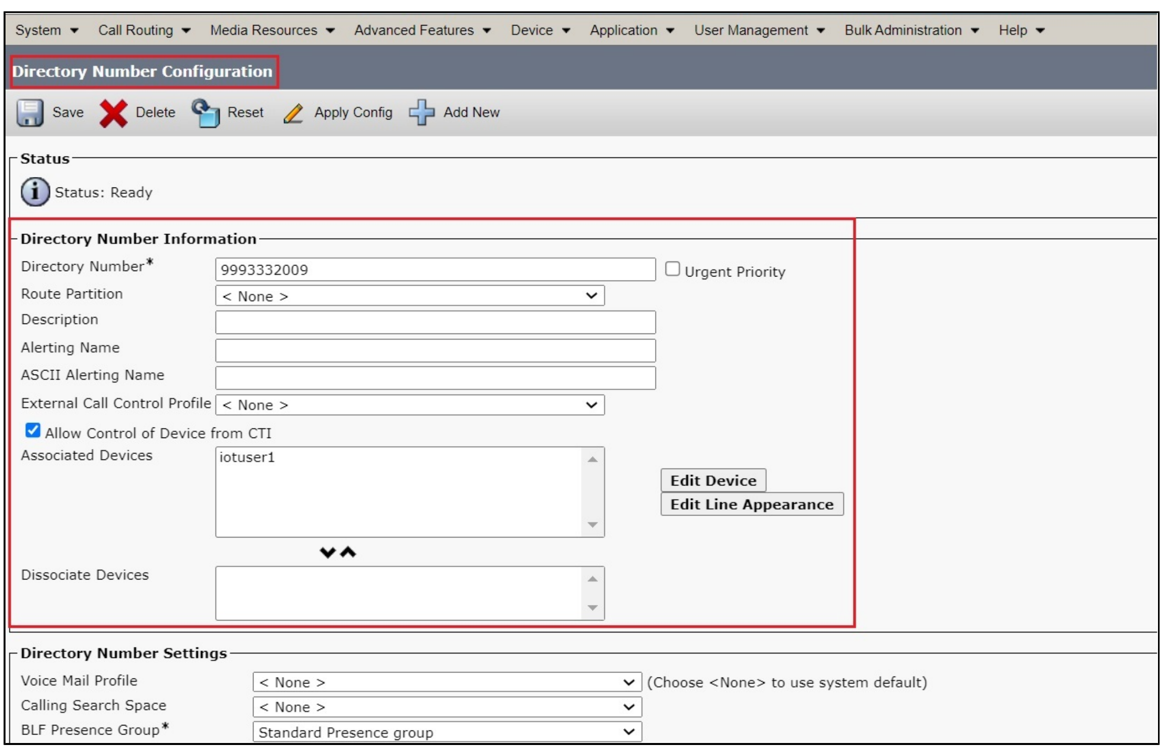


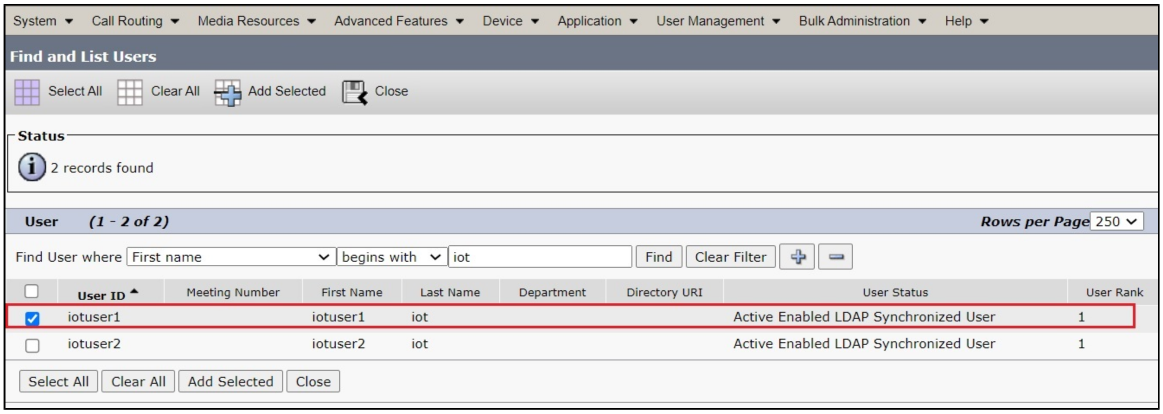
- From the Phone Type drop-down, choose Third-party AS-SIP Endpoint.
- Click **Next**.




System
Call Routing
Media Resources
Advanced Features
Device
Application
User Management
Bulk Administration
Help
Phone Configuration
Related Links: Back To Find/List
Save
Delete
Copy
Reset
Apply Config
Add New
Status
Status: Ready
Association
Modify Button Items
1 Line [1] - 9993332009 (no partition)
2 Line [2] - Add a new DN
3 Line [3] - Add a new DN
4 Line [4] - Add a new DN
5 Line [5] - Add a new DN
6 Line [6] - Add a new DN
7 Line [7] - Add a new DN
8 Line [8] - Add a new DN
Phone Type
Product Type: Cisco Unified Client Services Framework
Device Protocol: SIP
Real-time Device Status
Registration: Unknown
IPv4 Address: None
Device Information
Device is Active
Device is trusted
Device Name* iotuser1
Description iotuser1 SANTOSH
Device Pool* Default View Details
Common Device Configuration < None > View Details
Phone Button Template* Standard Client Services Framework
Common Phone Profile* Standard Common Phone Profile View Details
Calling Search Space < None >
AAR Calling Search Space < None >
Media Resource Group List san_media_grplist


System
Call Routing
Media Resources
Advanced Features
Device
Application
User Management
Bulk Administration
Help
Phone Configuration
Related Links: Back To Find/List
Save
Delete
Copy
Reset
Apply Config
Add New
Packet Capture Mode* None
Packet Capture Duration 0
BLF Presence Group* Standard Presence group
SIP Dial Rules < None >
MTP Preferred Originating Codec* 711ulaw
Device Security Profile* secure_jabber
Rerouting Calling Search Space < None >
SUBSCRIBE Calling Search Space < None >
SIP Profile* SAN_SIP_PROFILE View Details
Digest User iotuser1
Media Termination Point Required
Unattended Port
Require DTMF Reception
Certification Authority Proxy Function (CAPF) Information
Certificate Operation* No Pending Operation
Authentication Mode* By Null String
Authentication String
Generate String
Key Order* RSA Only
RSA Key Size (Bits)* 2048
EC Key Size (Bits)
Operation Completes By 2021 12 30 12 (YYYY:MM:DD:HH)
Certificate Operation Status: None
Note: Security Profile Contains Addition CAPF Settings.

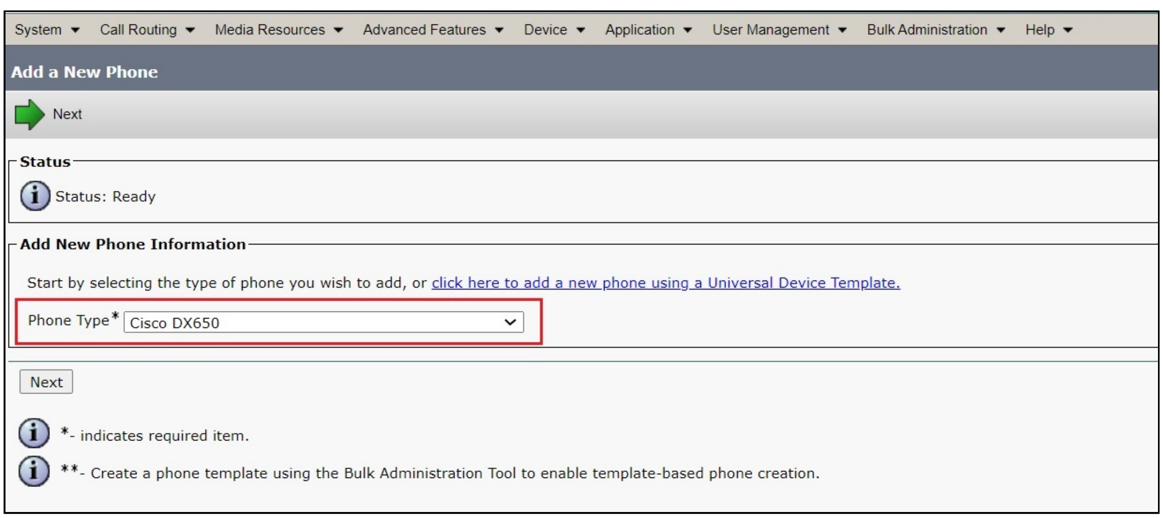

System
Call Routing
Media Resources
Advanced Features
Device
Application
User Management
Bulk Administration
Help
Directory Number Configuration
Save
Delete
Reset
Apply Config
Add New
Status
Status: Ready
Directory Number Information
Directory Number* 9993332009
Urgent Priority
Route Partition < None >
Description
Alerting Name
ASCII Alerting Name
External Call Control Profile < None >
Allow Control of Device from CTI
Associated Devices iotuser1
Edit Device
Edit Line Appearance
Dissociate Devices
Directory Number Settings
Voice Mail Profile < None > (Choose <None> to use system default)
Calling Search Space < None >
BLF Presence Group* Standard Presence group

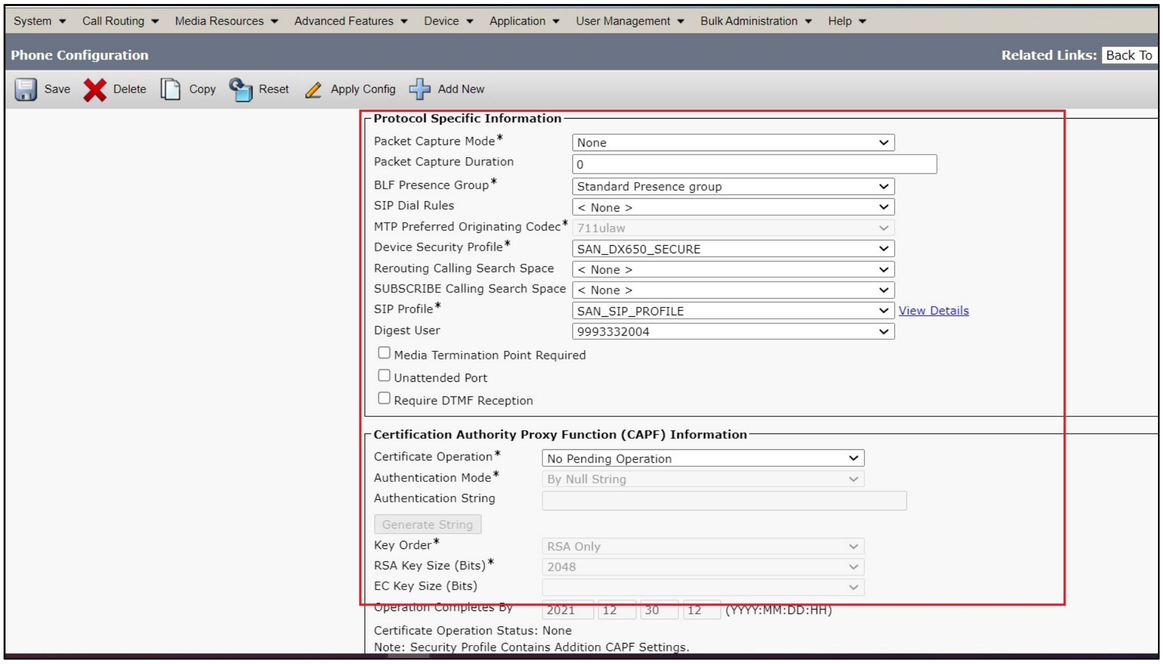


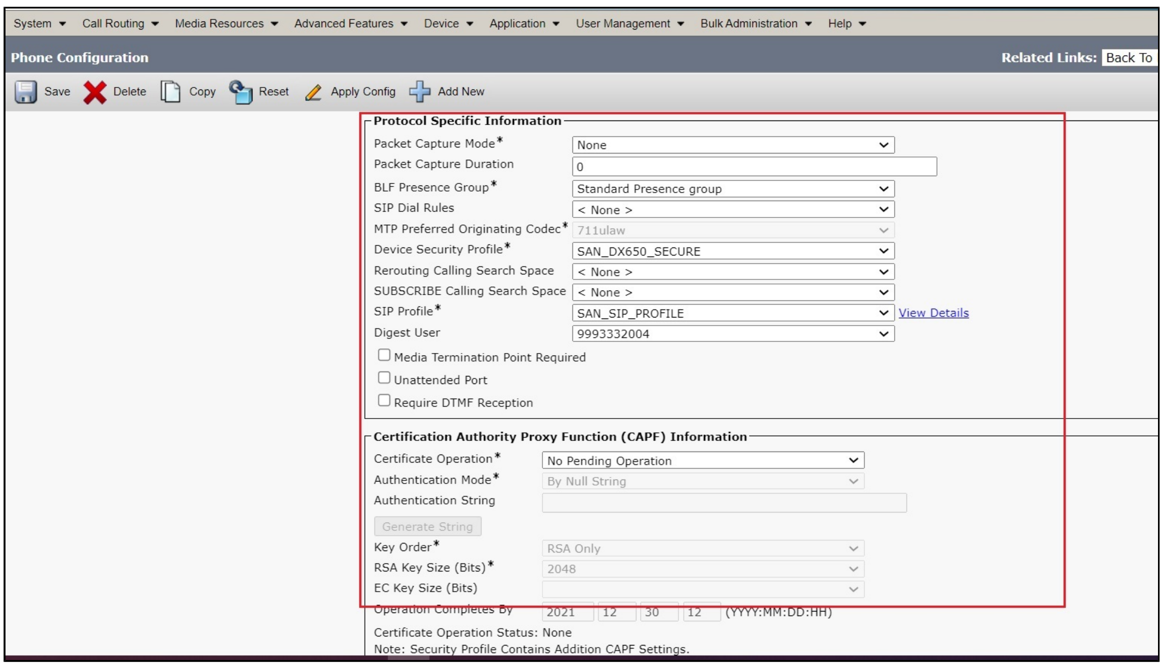
- Choose Device Trust Mode as **Not Trusted**, if third part end point is selected for phone button template.
- Enter the Media Access Control (MAC) address that identifies Cisco Unified IP Phones. Ensure that the value comprises 12 hexadecimal characters.
- Choose **Default** Device pool.
  - A Device pool defines sets of common characteristics for devices, such as region, date/time group, and soft key template.
- Choose **Cisco Unified Client Services Framework** for Jabber clients **or Cisco DX650** for DX650 phones from the phone button template drop-down.
  - The phone button template determines the configuration of buttons on a phone and identifies which feature (line, speed dial, and so on) is used for each button.
- Associate the Media Resource Group List created.
- Choose the user ID of the assigned phone user.

> **Note**
> CUCM supports auto registration of Cisco endpoints, refer to the following link for more details:
>
> https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmcfg/CUCM_BK_C95ABA82_00_admin-guide-100/CUCM_BK_C95ABA82_00_admin-guide-100_chapter_011010.html

- Choose the security profile to apply to the device. Customer can choose to have a Non-Secure SIP Profile if they are using a Non-Secure SIP Trunk.
- Associate the SIP Profile created before.
  - SIP profiles provide specific SIP information for the phone such as registration and keep-alive timers, media ports, and do not disturb control.
- Choose an end user that you want to associate with the phone for this setting that is used with digest authentication (SIP security).
- Update the CAPF information.
- Click **Save**.
- For DX650 select as below.




System
Call Routing
Media Resources
Advanced Features
Device
Application
User Management
Bulk Administration
Help
Phone Configuration
Related Links: Back To
Save
Delete
Copy
Reset
Apply Config
Add New
Protocol Specific Information
Packet Capture Mode* None
Packet Capture Duration 0
BLF Presence Group* Standard Presence group
SIP Dial Rules < None >
MTP Preferred Originating Codec* 711ulaw
Device Security Profile* SAN_DX650_SECURE
Rerouting Calling Search Space < None >
SUBSCRIBE Calling Search Space < None >
SIP Profile* SAN_SIP_PROFILE View Details
Digest User 9993332004
Media Termination Point Required
Unattended Port
Require DTMF Reception
Certification Authority Proxy Function (CAPF) Information
Certificate Operation* No Pending Operation
Authentication Mode* By Null String
Authentication String
Generate String
Key Order* RSA Only
RSA Key Size (Bits)* 2048
EC Key Size (Bits)
Operation Completes By 2021 12 30 12 (YYYY:MM:DD:HH)
Certificate Operation Status: None
Note: Security Profile Contains Addition CAPF Settings.


System
Call Routing
Media Resources
Advanced Features
Device
Application
User Management
Bulk Administration
Help
Phone Configuration
Related Links: Back To
Save
Delete
Copy
Reset
Apply Config
Add New
Protocol Specific Information
Packet Capture Mode* None
Packet Capture Duration 0
BLF Presence Group* Standard Presence group
SIP Dial Rules < None >
MTP Preferred Originating Codec* 711ulaw
Device Security Profile* SAN_DX650_SECURE
Rerouting Calling Search Space < None >
SUBSCRIBE Calling Search Space < None >
SIP Profile* SAN_SIP_PROFILE View Details
Digest User 9993332004
Media Termination Point Required
Unattended Port
Require DTMF Reception
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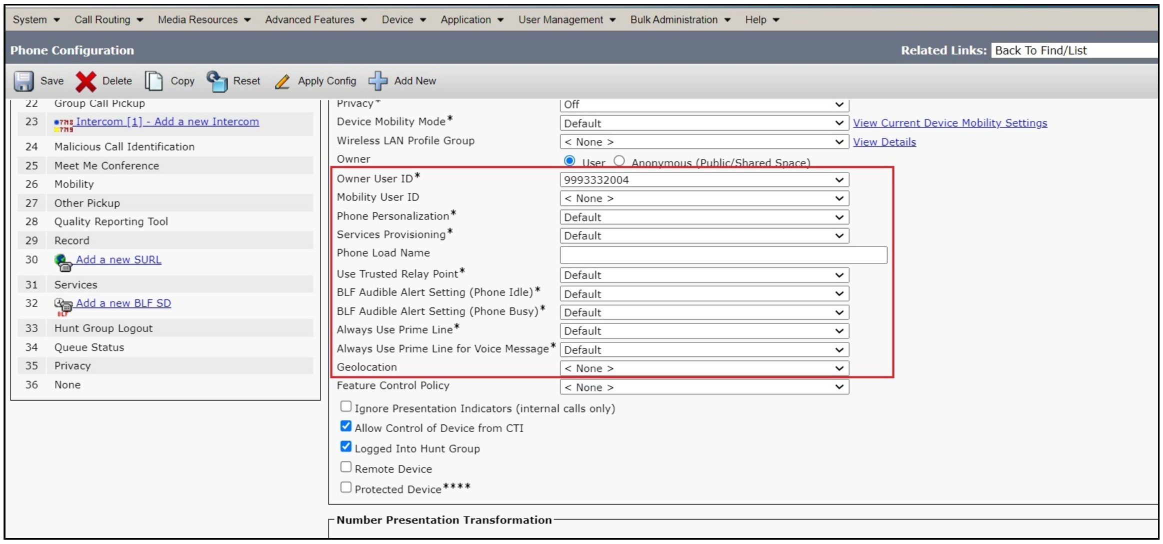
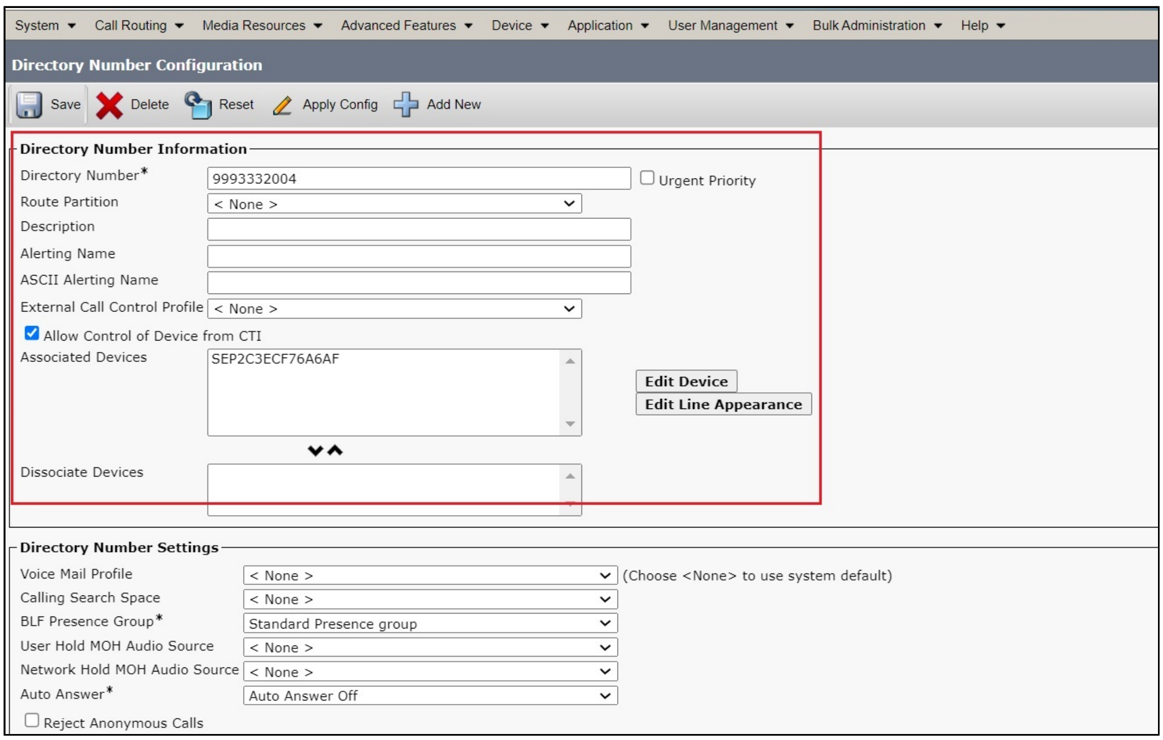
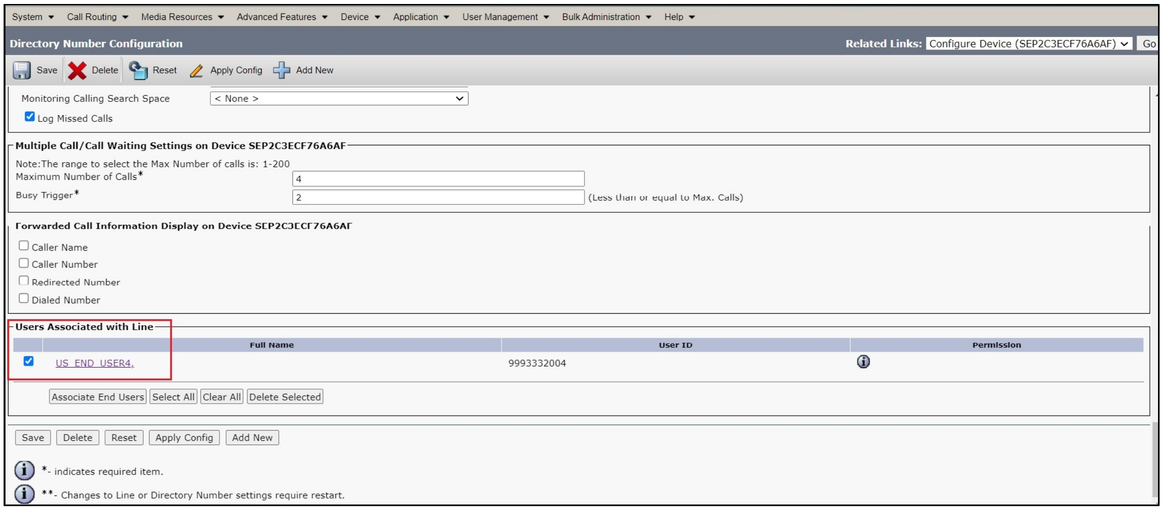
Key Order* RSA Only
RSA Key Size (Bits)* 2048
EC Key Size (Bits)
Operation Completes By 2021 12 30 12 (YYYY:MM:DD:HH)
Certificate Operation Status: None
Note: Security Profile Contains Addition CAPF Settings.



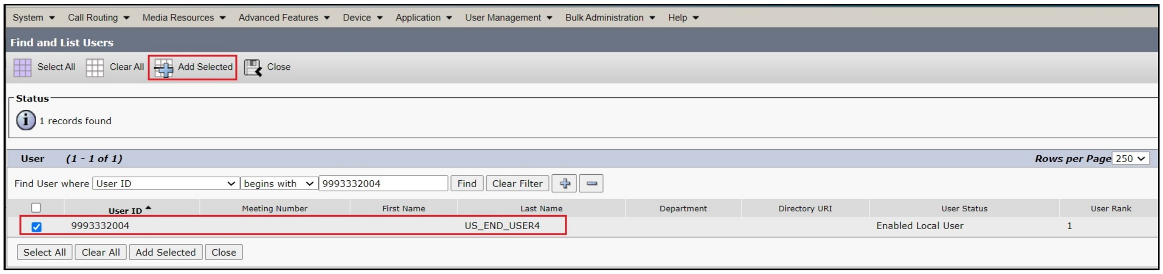
- Click this link to add a remote destination to associate with this device. The Remote Destination Configuration window displays, which allows you to add a new remote destination to associate with this device.



- Add the Directory number.
- Click **Save**.



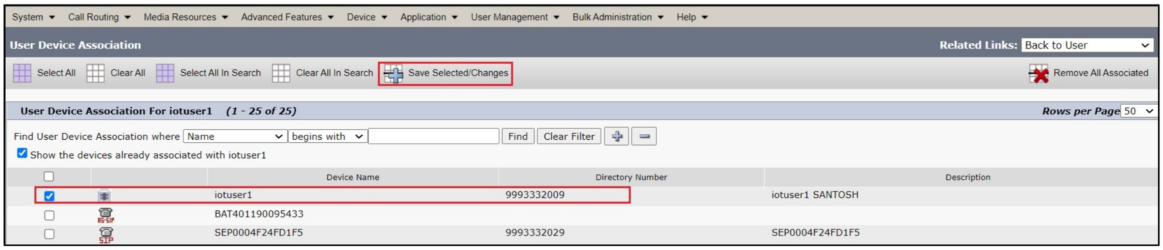
- Click the **Associate End User** button.
- Select the end user created from the list and click **Add Selected**.

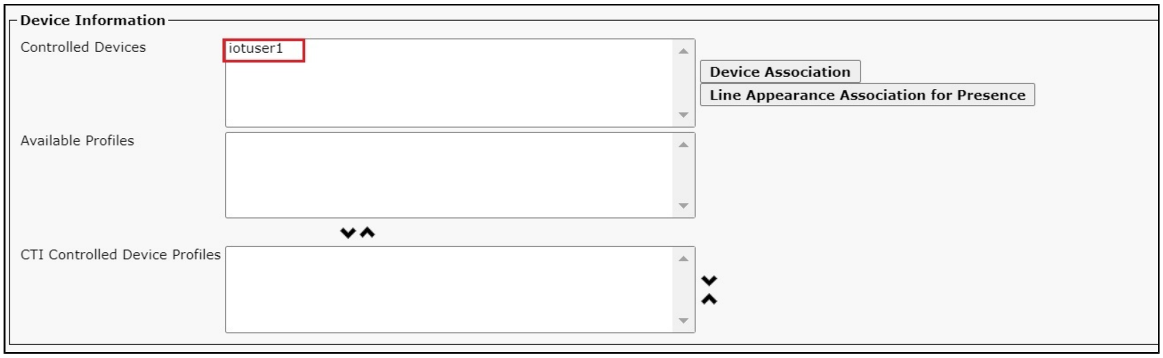


- After the above step, the user association is completed.
- Save the configuration.
- Click **Apply Config** followed by the Reset button.
- Reset, Restart and Close the window.

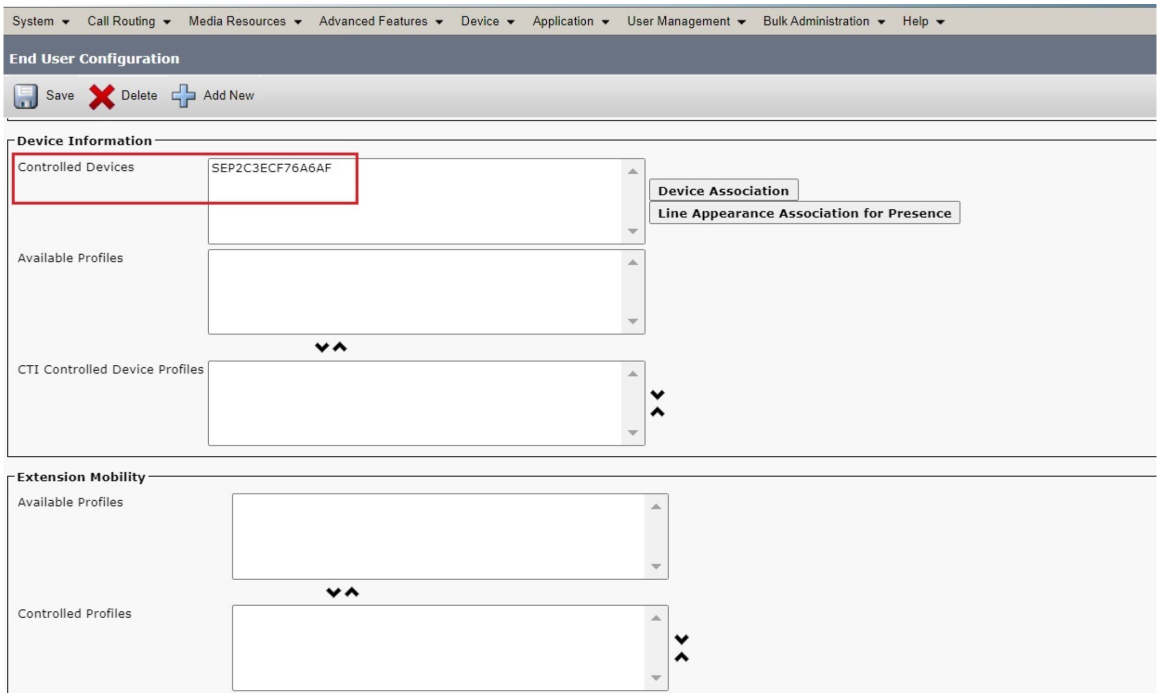
Device Association

- Navigate back to **User Management >** End User.
- In the Device Information field, click **Device Association**. This will display all the available devices.
- Select the device created in the previous step and save.
- After selecting the appropriate device, it will appear in the Controlled Devices pane.





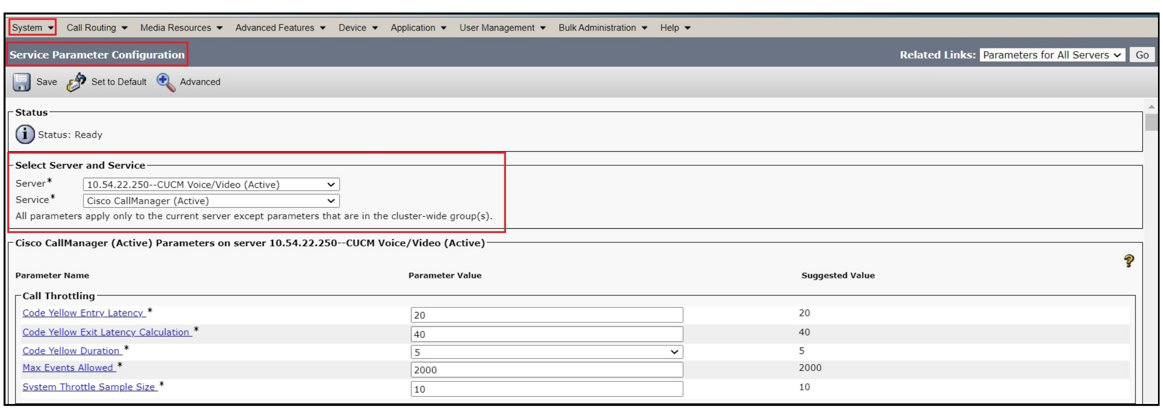
- For DX650 as below.



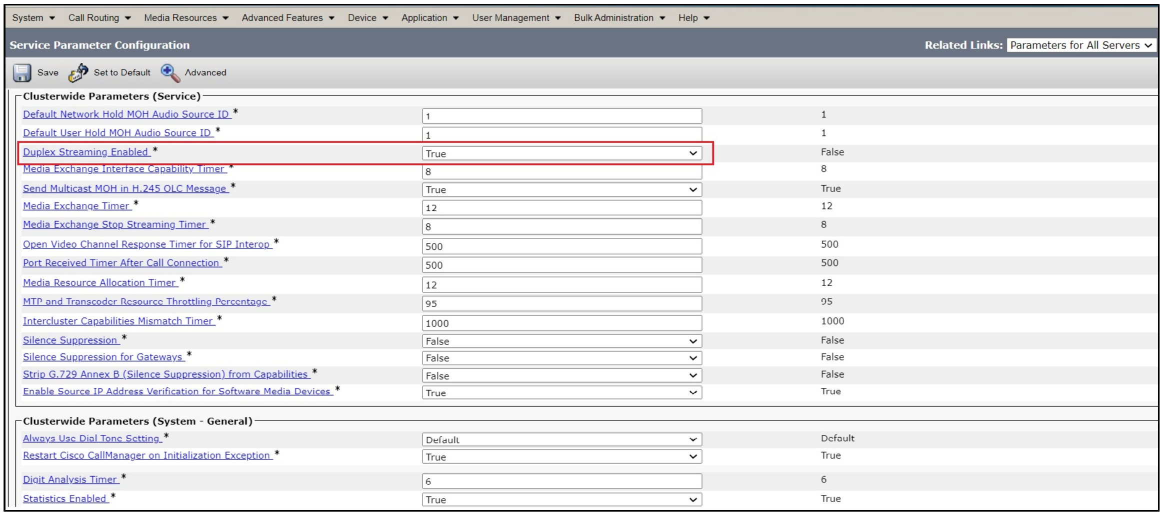
Enable MoH

In Cisco Unified Communications Manager Administration, use the **System > Service Parameters** menu path to configure service parameters.

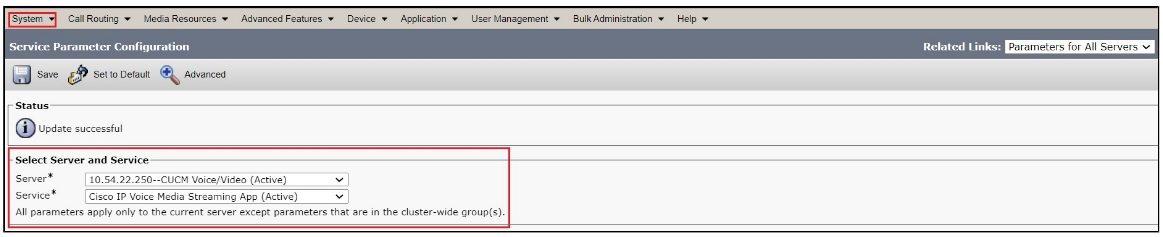
- In the Server drop-down list box in the Service Parameter Configuration window, choose the CUCM server being used. In this case, active means that you provisioned the server in Cisco Unified Communications Manager Administration.
- From Service drop-down select Cisco CallManager. The service displays as active in the Service Parameters Configuration window.

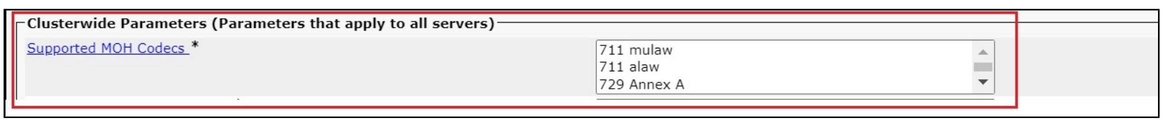


- Set the Duplex Streaming Enabled flag to True. This parameter determines whether Music On Hold (MOH) and Annunciator use duplex streaming.
- Click **Save**.



- From Service drop-down select Cisco IP Voice Streaming App. The service displays as active in the Service Parameters Configuration window.
- Set the Supported MOH Codecs.
- Click **Save**.





Section C: Microsoft Teams Direct Routing

For Microsoft Teams related configurations and queries, please contact the Microsoft technical support team, for details visit: https://support.microsoft.com/contactus

For detailed information about Microsoft Teams direct routing products and solutions, please visit:

- https://docs.microsoft.com/en-us/microsoftteams/cloud-voice-landing-page
- https://docs.microsoft.com/en-us/microsoftteams/direct-routing-configure

> **Note**
> This interop was performed with Media-Bypass OFF configuration on Microsoft Teams Direct Routing.

Supplementary Services & Features Coverage

The following checklist depicts the set of services/features covered through the configuration defined in this Interop Guide.

Sr. No.	Supplementary Services/ Features	Coverage
01.	OPTIONS validation	✔
02.	Call Setup and Termination over TLS	✔
03.	Ringing and Local Ringback Tone	✔

04.	Remote Ringback Tone Handling	✔
05.	Cancel Call, No Answer, Busy and Call Rejection	✔
06.	Basic Call with different codecs	✔
07.	DTMF	✔
08.	Anonymous Calls	✔
09.	Call Hold and Resume	✔
10.	Call Forward - Unconditional, Busy and No Answer	✔
11.	Call Transfer (Blind/Unattended)	✔
12.	Call Transfer (Attended)	✔
13.	Call Conference	✘
14.	Meet Me Conference	✘
15.	4xx/5xx Response Handling	✔
16.	Long Duration Calls	✔
17.	Early and Late Media	✔
18.	Simultaneous Ringing	✔
19.	Transcode Calls	✔

Legend

Supported	✔
Not Supported	✘

Caveats

- Meet Me and Adhoc conference could not be tested due to unavailability of hardware transcoder within the lab environment. Lab has CUCM software conference bridge that does not support sRTP. Customers using non-secure trunk and media will not face this issue. For more details visit https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/security/11_0_1/secugd/CUCM_BK_C1A78C1D_00_cucm-security-guide-1101/secure_conference_resources_setup.pdf
- Media packet Loss issue is observed between the end points for basic call scenarios that is a known issue to Ribbon. This issue has been addressed and will be fixed in the upcoming SBC release.
- Media packet Loss issue is observed between the TEAMS end points for call forward and call transfer scenarios that is a known issue to Ribbon. This issue has been addressed and will be fixed in the upcoming SBC release.
- Proxy with sRTP relay mode for comfort noise and RTCP passthrough scenarios issue is observed that is a known issue to Ribbon. This issue has been addressed and will be fixed in the upcoming SBC release.

Support

For any support related queries about this guide, please contact your local Ribbon representative, or use the details below:

- Sales and Support: 1-833-742-2661
- Other Queries: 1-877-412-8867
- Website: https://ribboncommunications.com/about-us

References

For detailed information about Ribbon products and solutions, please visit: https://ribboncommunications.com/products

For additional information on Cisco Unified Communications Manager, please visit: https://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-installation-and-configuration-guides-list.html

For additional information on Ribbon SBC SWe Edge on Azure, please visit: Deploying an SBC SWe Edge from the Azure Marketplace.

Conclusion

This Interoperability Guide describes successful configuration of interop involving Ribbon SBC SWe Edge on Azure, Cisco Unified Communications Manager and Microsoft Teams.

All features and capabilities tested are detailed within this document - any limitations, notes or observations are also recorded in order to provide the reader with an accurate understanding of what has been covered and what has not.

Configuration guidance is provided to enable the reader to replicate the same base setup - additional configuration changes are possibly required to suit the exact deployment environment.