

# **Skype for Business 2019 using SIP trunk (TLS) to Cisco Unified Communications Manager Release 12.5.1**

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## Introduction

This document describes the steps and configurations necessary for Cisco Unified Communications Manager (Cisco UCM) release 12.5.1 to interoperate with the Skype for Business 2019 on TLS using the following configuration:

The following items were tested:

- Basic call between the two systems and verification of voice path, using both SIP and Legacy phones on the Cisco side, and SIP client on the Skype for Business side (Refer to limitation section for more info)
- CLIP/CLIR/CNIP/CNIR features: Calling party Name and Number delivery (allowed and restricted) (Refer to limitation section for more info)
- COLP/CONP/COLR/CONR features: Connected Name and Number delivery (allowed and restricted) (Refer to limitation section for more info)
- Call Transfer: Attended and Early attended (Refer to limitation section for more info)
- Alerting Name Identification (Refer to limitation section for more info)
- Call forwarding: Call Forward Unconditional(CFU), Call Forward Busy (CFB), and Call Forward No Answer (CFNA)
- Hold and Resume with Music on Hold
- Three-way conferencing (Refer to limitation section for more info)
- Voice messaging and MWI activation-deactivation (Refer to limitation section for more info)
- Extend and Connect (Refer to limitation section for more info)
- Call Park (Refer to limitation section for more info)

Listed below are the highlights of the integration issues:

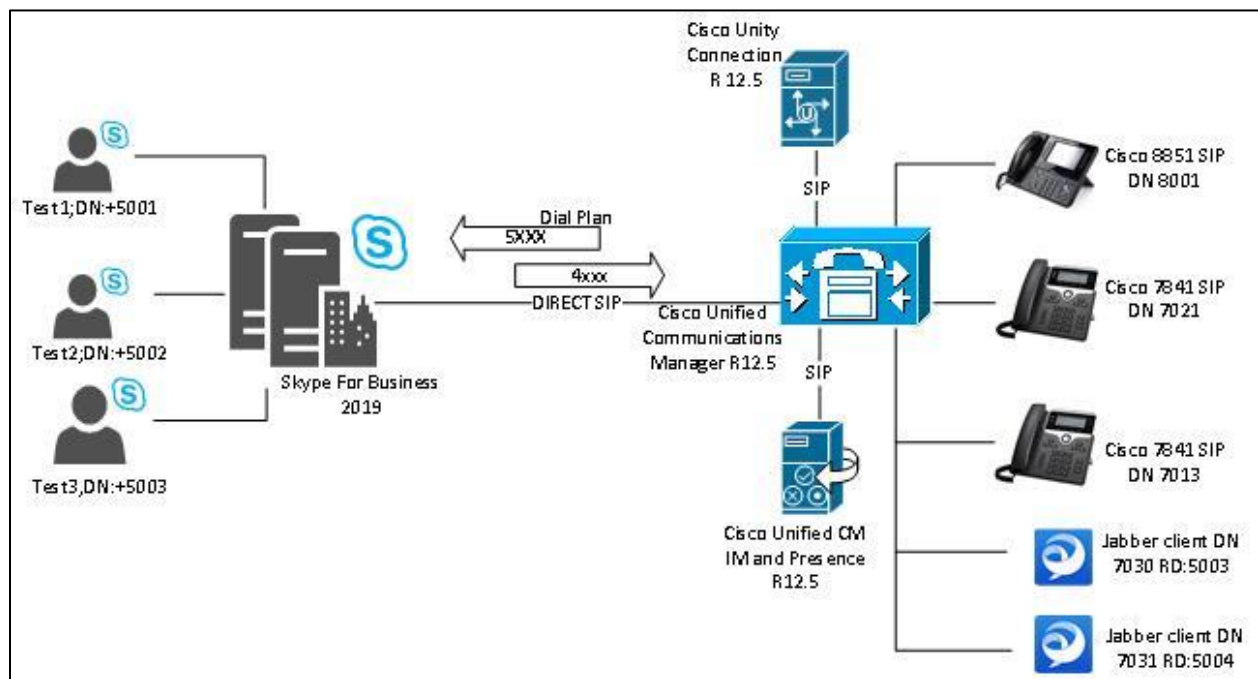
- Basic calls work from Cisco UCM to Skype for Business and vice versa. Only Cisco SIP phones were used on Cisco UCM side as SCCP phones do not support 80-bit crypto attribute required by Skype for Business
- Skype for Business drops the call on HOLD after 30 seconds where the call hold initiated from Skype for Business client.
- Basic calls work from Cisco UCM to Skype for Business and vice versa using G711 ulaw and alaw
- Caller name and number is not updated correctly for basic calls, attended and early-attended transfer scenarios
- Video calls between the Cisco UCM and Skype for Business users were not tested
- REFER support should be disabled in Skype for Business for the Call Park scenarios
- Skype for business does not consider privacy:id sent in 18x and 2xx message from Cisco UCM

- A call placed on hold by the Skype client is dropped after 30 seconds when Skype for Business Server 2019 version 7.0.2046.123 is used. The issue is resolved with the newer release of Skype for Business 2019 version 7.0.2046.151 and above.

Below are the key results:

- Basic call, Call Transfer, Call Forwarding, Conference Call, and Hold and Resume tested successfully with a few caveats and limitations.
- Centralized voicemail, using Unity Connection server integrated with Cisco UCM via SIP is used for testing. This voicemail solution can provide centralized voicemail services, supporting both Skype for Business users and Cisco end users.

## Network Topology



## Limitations

These are the known limitations, caveats, or integration issues:

- Skype for Business does not support G729 codec. Trunk tested with only G711 ulaw and alaw.
- Disable Media Bypass on Skype for Business. If enabled, the Hold INVITE sent by Cisco UCM (without SDP) being rejected with 488 on Skype for Business.
- Skype for Business and Cisco UCM does not support overlap dialing modes on their SIP endpoints

- Skype for Business does not consider Privacy: ID parameter sent by Cisco UCM during 180 Ringing or 200 OK when Connected Name/ID is restricted on Cisco UCM. Subsequently, Skype for Business does not support updating the Connected Party's display as Private.
- Skype for Business does not update the CLID in transfer/conference scenarios. After transfer/conference is completed, Cisco UCM sends mid call INVITE and UPDATE messages that contain PAI and RPI. Skype for Business does not update this information on their client display.
- Skype for Business sends incorrect number in history-info during forward scenarios. As a work around Skype for Business, users DN are configured with a prefix "+".
- Cisco UCM Remote Destination is configured with a prefix "+" and a Route Pattern to route a DN with a prefix '+' is added. (Refer Cisco UCM configuration section - Cisco Unified Communications Manager Route Pattern to invoke Jabber client with Remote Destination configured as Skype for Business Extension).
- Skype for Business does not support MWI notification from Cisco Unity Connection. It responds with a "405 Method Not Allowed" for a NOTIFY Message received from the Cisco UCM that has MWI information.
- In Multiple Call Forwarding scenario between Skype for Business Users and Cisco UCM Users, wherein both originator and terminator being Skype for Business Users, originator does not display the Caller ID of terminator.
- When Cisco UCM User completes the conference, the audio is on RTP. Pad-lock symbol on the Cisco phone disappears.
- When Skype for Business user makes a caller ID, restricted call to Cisco Phone A, caller ID is displayed as Private. However, when Cisco Phone A transfers the call to Cisco Phone B, the caller ID displayed on Cisco Phone B is Skype for Business user's caller ID instead of displaying as Private.

## System Components

### Hardware Requirements

The following hardware are used:

- Cisco UCS-C240-M3S VMWare Host
- Cisco 8851 and 7841 IP phones

## Software Requirements

The following software are used:

- Cisco UCSC-C240-M3S VMware vSphere Image Profile: ESXi-5.5.0-1331820-standard
- Cisco Unified Communications Manager release 12.5.1.11900-146
- Cisco Unified Communications Manager IM & Presense Service release 12.5.1.11900-117
- Cisco Unity Connection release 12.5.1.11900-57
- Cisco Jabber 12.6.1.34405 Build 284405
- Skype for Business Server 2019 version 7.0.2046.0
- Skype for Business Client version 16.0.11328.20390

## Features

This section lists supported and unsupported features. No deviation from the configuration presented in this document will be supported by Cisco. Please see the Limitations section for more information.

### Features Supported

- CLIP—calling line (number) identification presentation
- CLIR—calling line (number) identification restriction
- CNIP—calling Name identification presentation
- CNIR—calling Name identification restriction
- Attended call transfer
- Early attended call transfer
- CFU—call forwarding unconditional
- CFB—call forwarding busy
- CFNA—call forwarding no answer
- COLP—connected line (number) identification presentation
- COLR—connected line (number) identification restriction
- CONP—connected Name identification presentation
- CONR—connected Name identification restriction
- Hold and resume
- Conference call
- MWI—Message Waiting Indicator (only for Cisco Endpoints)

- Audio Codec Preference List
- Call Park/Pickup(see limitation section)
- Extend and Connect
- Shared Line on Cisco Endpoints

#### Features Not Supported or Not Tested

- G729 voice codec
- Call completion (callback, automatic callback)
- Shared Line on Skype for Business
- Message Waiting Indicator on Skype for Business Endpoints
- Blind transfer
- Video calls
- Scenarios that required third PBX.
- Scenarios involving Non-SIP interfaces
- Scenarios involving Cisco UCM SCCP Phones
- Scenarios involving Non-SIP interfaces
- Connected party restriction sends and receives on Skype for Business Server is not supported



## Configuration

The goal of this guide is to provide an overview of the integration between Cisco Unified Communication Manager and Skype for Business. The deployment will interconnect the UC systems using SIP. No PSTN connectivity has been tested with this integration. The following sections provide the required configurations for a successful integration.

### Global Trunk Configuration Highlights:

Setting	Value
Skype for Business Media Bypass	DISABLED
Skype for Business Encryption Support	OPTIONAL
Skype for Business REFER Support	DISABLED
Cisco UCM SIP Trunk MTP	DISABLED
Cisco UCM PRACK	ENABLED
Cisco UCM Early Offer	ENABLED
Transport type Cisco UCM to Skype for Business	TLS
Cisco UCM SRTP allowed	Enabled

### Configuring Sequence and Tasks:

#### Configuring the Skype for Business:

- Add Cisco UCM to Skype for Business Topology
- Trunk Configuration
- Route Configuration
- Voice Policy and PSTN Usage Configuration
- Dial Plan Configuration
- Call Park range Configuration
- Media Bypass Configuration
- User Configuration
- Client Configuration

#### Configuring the Cisco Unified Communications Manager:

- SIP trunk security profile
- SIP profile
- Media resource group and media resource group list
- Assign media resource group list (MRGL) in the default device pool
- Region configuration
- Normalization script

- SIP trunk to Skype for Business
- SIP Trunk to Cisco Unity Connection
- Assign User in Cisco Unity Connection
- SIP phones device configuration
- Route Group, Route List and SIP Route Pattern
- Voice Mail
- Route pattern to Skype for Business, Unity Connection and Skype for Business call Park Range
- Extend and Connect Feature and User configuration

Configuring the Cisco Unity Connection:

- Cisco Unity Connection Telephony Integration
- Cisco Unity Connection User Configuration

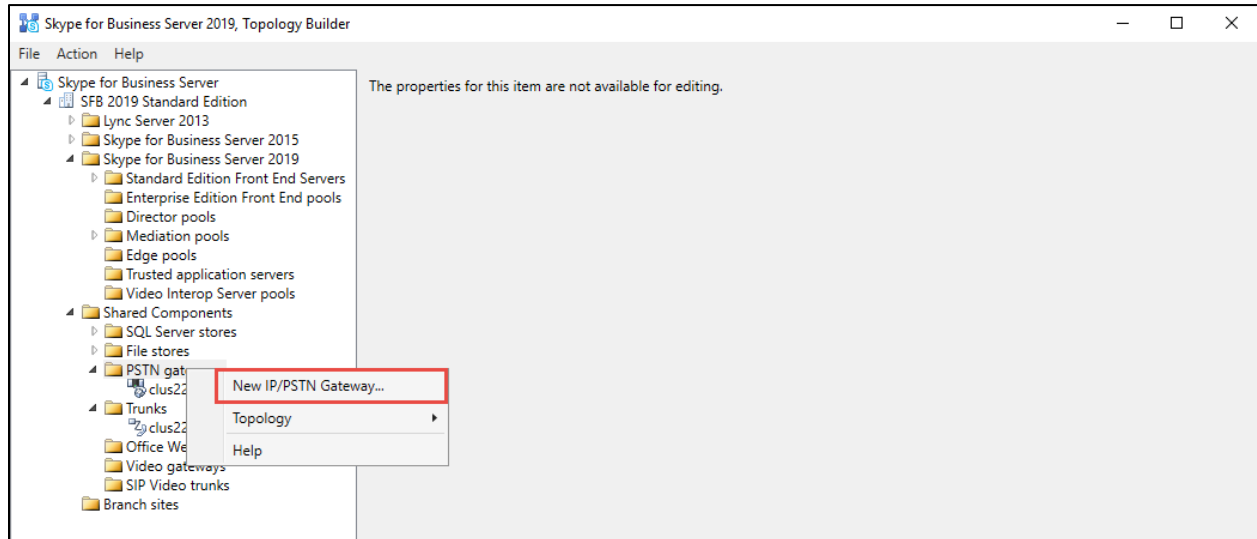
## Configuring the Skype for Business

### Add Cisco UCM to Skype for Business Topology

Run the Skype for Business 2019 Topology Builder as a user in the CSAdministrator group.

**Navigation:** Skype for Business Server→CleanDefaultTopology→Shared Components→PSTN gateways

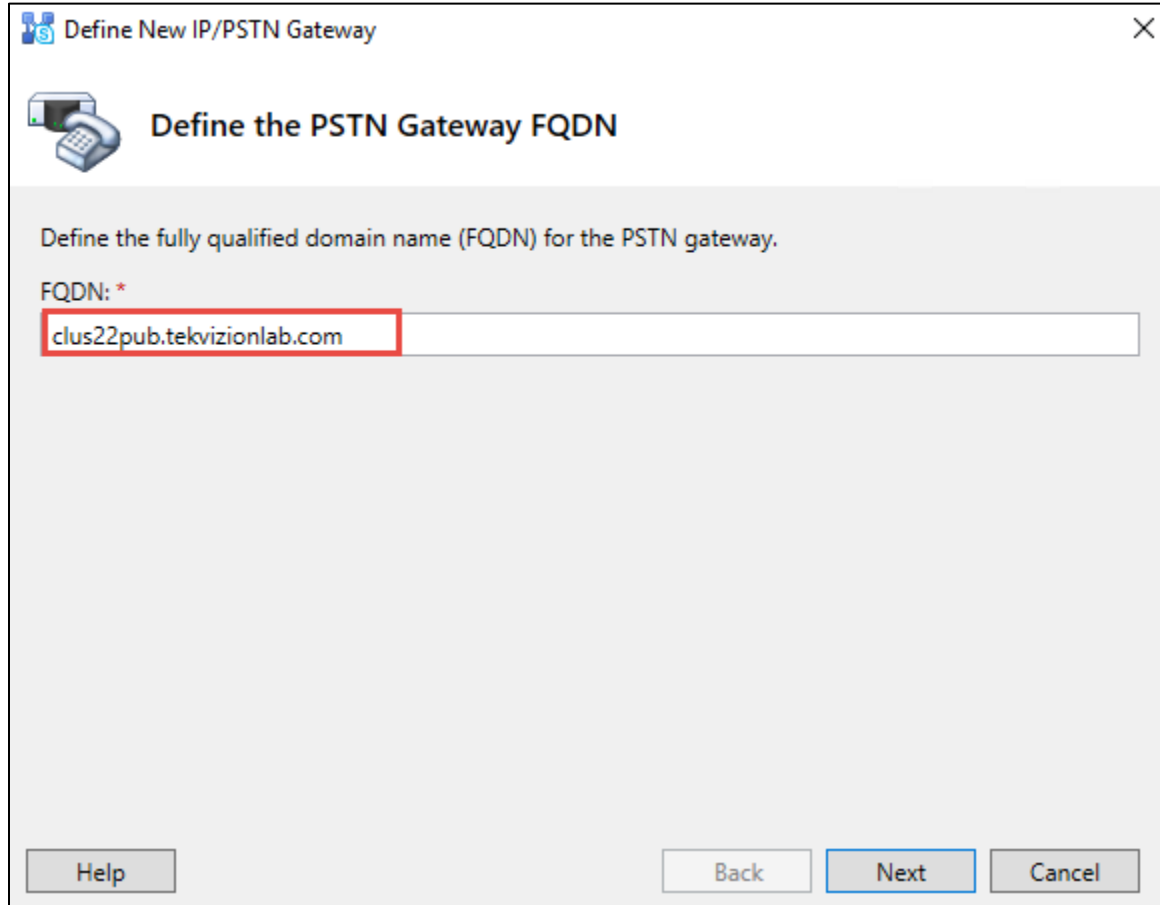
Right click and select “New IP/PSTN Gateway”



Set FQDN = <FQDN of the Cisco UCM>— clus22pub.tekvizionlabs.com is used in this test.

Click Next.

## Skype for Business – Add PSTN Gateway (Continued)



The image shows a Windows-style dialog box titled "Define New IP/PSTN Gateway". It features a close button (X) in the top right corner. Below the title bar, there is a header section with a telephone icon and the text "Define the PSTN Gateway FQDN". The main area of the dialog contains the instruction "Define the fully qualified domain name (FQDN) for the PSTN gateway." followed by a label "FQDN: \*". A text input field is provided, containing the value "clus22pub.tekvizionlab.com", which is highlighted with a red rectangular border. At the bottom of the dialog, there are four buttons: "Help", "Back", "Next" (which is highlighted with a blue border), and "Cancel".

Define New IP/PSTN Gateway

Define the PSTN Gateway FQDN

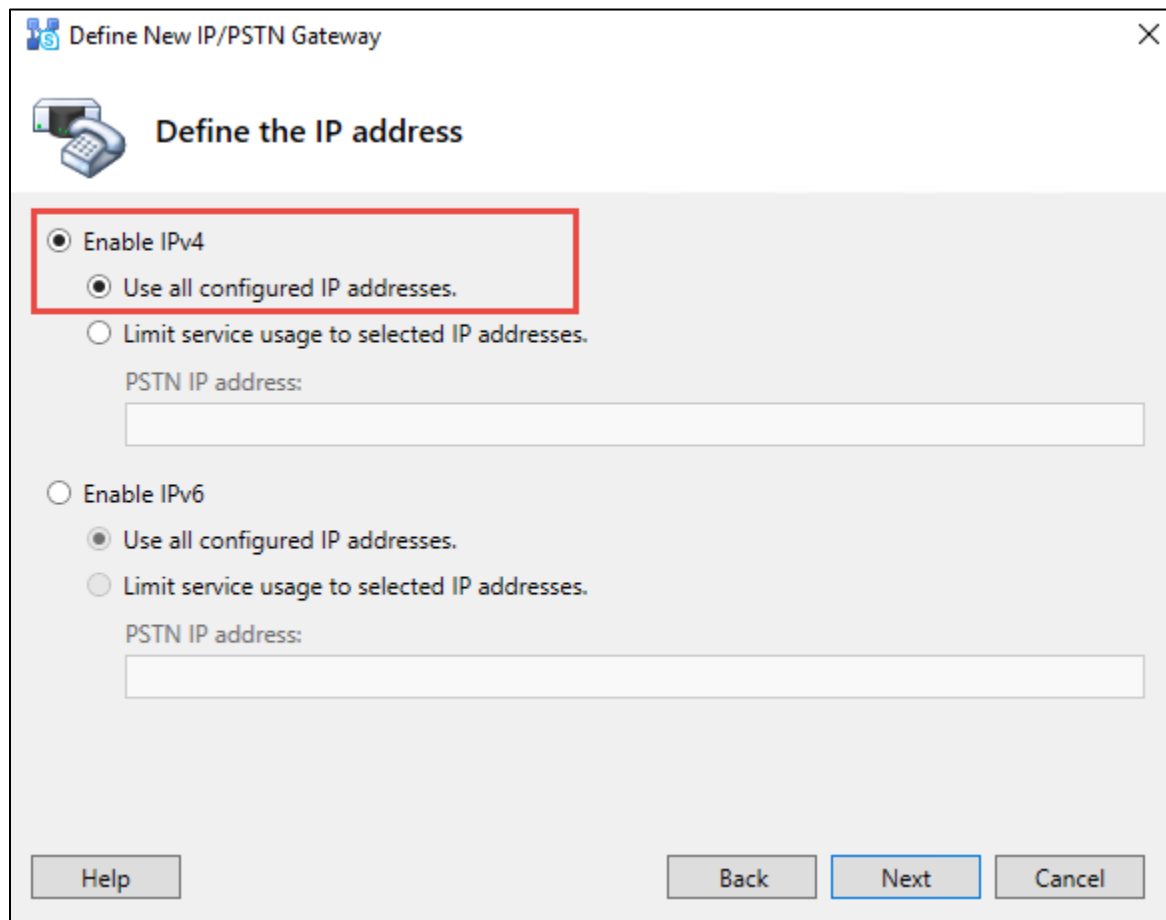
Define the fully qualified domain name (FQDN) for the PSTN gateway.

FQDN: \*

clus22pub.tekvizionlab.com

Help Back Next Cancel

Check the Enable IPv4 and Use all configured IP addresses radio button  
Click Next.



Define New IP/PSTN Gateway

Define the IP address

☒ Enable IPv4  
☒ Use all configured IP addresses.  
☐ Limit service usage to selected IP addresses.  
PSTN IP address:

☐ Enable IPv6  
☒ Use all configured IP addresses.  
☐ Limit service usage to selected IP addresses.  
PSTN IP address:

Help Back Next Cancel

Set **Trunk Name** = FQDN of the Cisco UCM – **clus22pub.tekvizionlabs.com** is used for this test

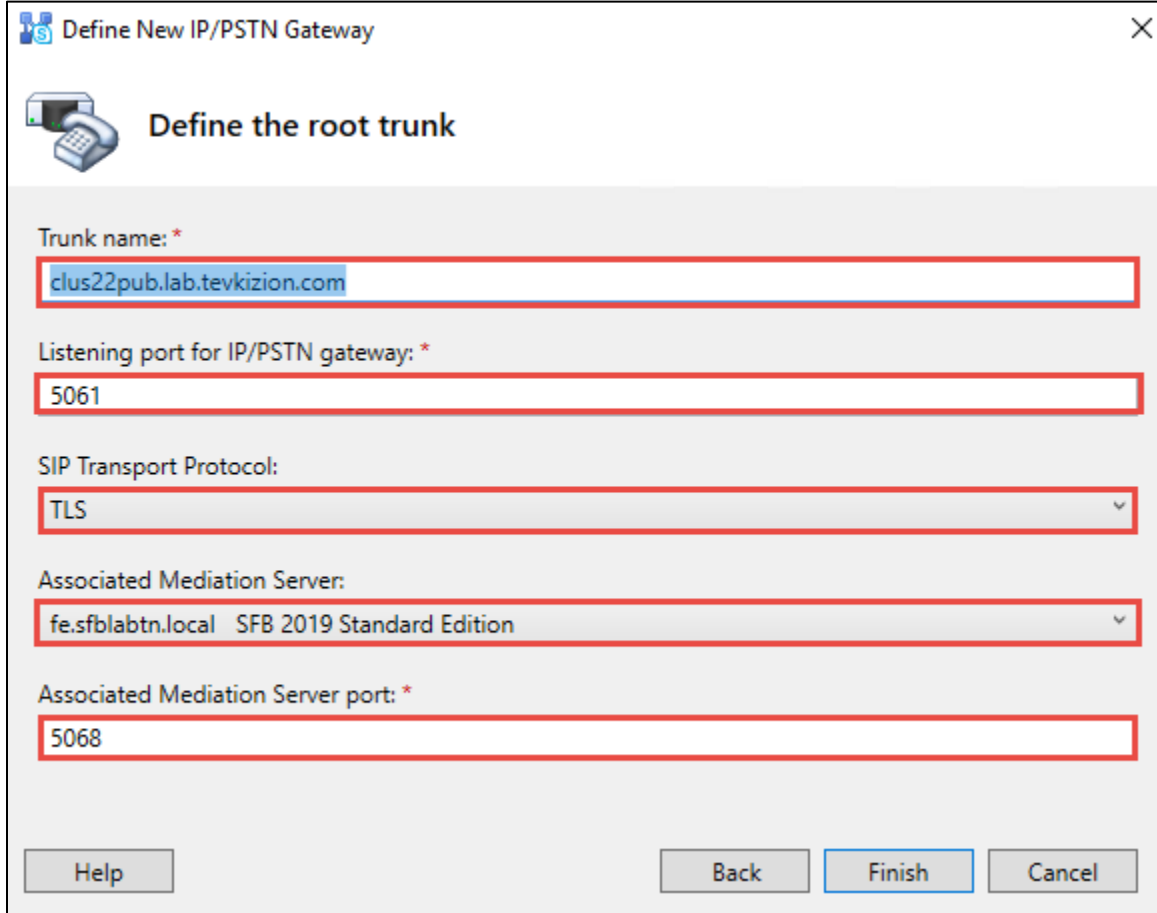
Set **Listening port for IP/PSTN gateway** = The **Listening port** should match the **Incoming Port** setting in the CISCO UCM's **SIP Trunk Security Profile** – **5061** is used for this test

Set **SIP Transport Protocol** = TLS


Set **Associate Mediation Server**: Assign this PSTN gateway to the Front End co-located mediation server – **fe.sfablabtn.local** is used for this test.

Click Finish.

## Skype for Business – Add PSTN Gateway (Continued)



Define New IP/PSTN Gateway

 Define the root trunk

Trunk name: \*

clus22pub.lab.tevkizion.com

Listening port for IP/PSTN gateway: \*

5061

SIP Transport Protocol:

TLS

Associated Mediation Server:

fe.sfbtabtn.local SFB 2019 Standard Edition

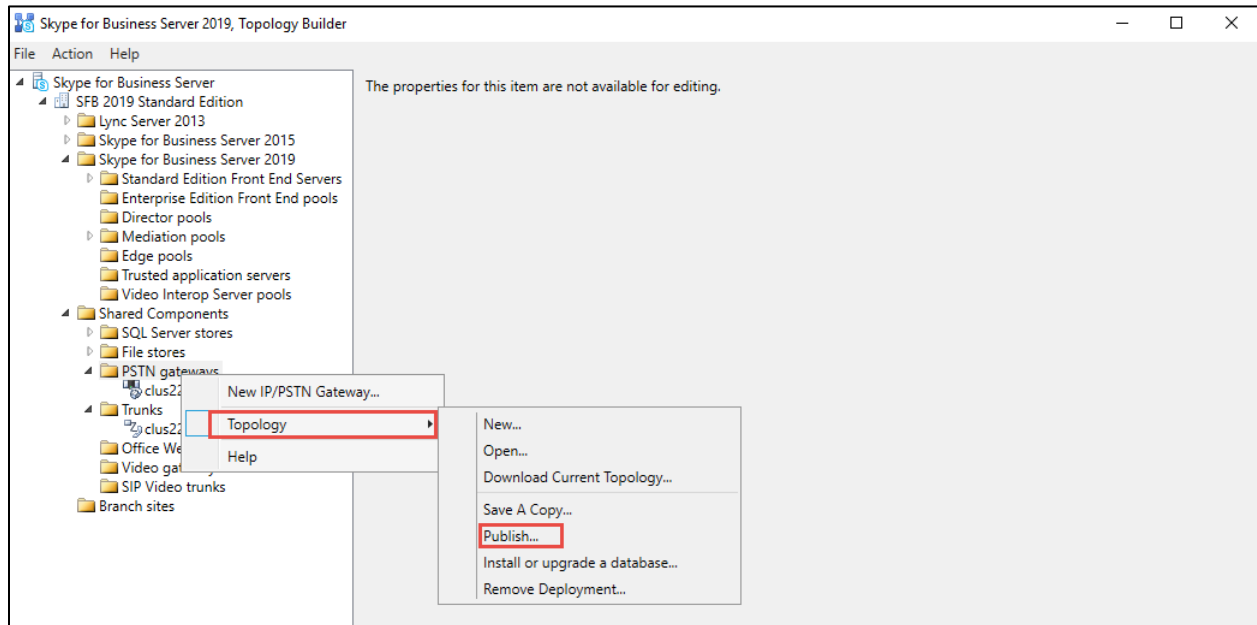
Associated Mediation Server port: \*

5068

Help Back Finish Cancel

Publish the topology so these new configurations take effect.

## Skype for Business – Add PSTN Gateway (Continued)

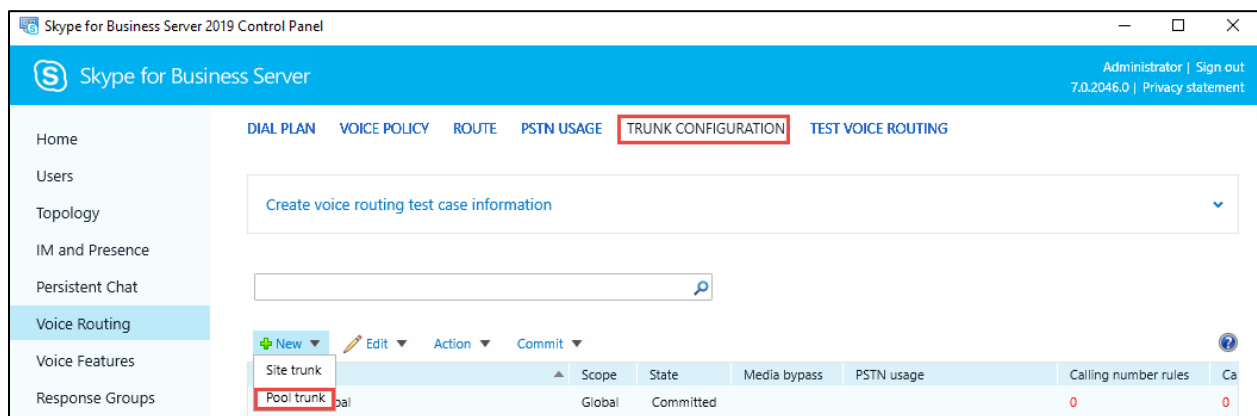


## Trunk Configuration

Open the Skype for Business 2019 Control Panel.

**Navigation:** Voice Routing -> Trunk Configuration

Select New → Pool Trunk



Set **Service** = Trunk to Cisco UCM that was created earlier as a PSTN gateway in the topology builder – clus22pub.tekvizionlabs.com is used for the test.

Set **Maximum early dialogs supported** = 23

Set **Encryption support level** = **Optional**

Set **Refer Support** = **None**

Uncheck **Enable media bypass**

Check **Centralized media processing**

Uncheck **Enable RTP latching**

Check **Enable forward call history**

Uncheck **Enable forward P-Asserted-Identity data\*** [Note: this is checked when test scenarios that involve restrict ID need to be executed]

Uncheck **Enable outbound routing failover timer**

Skype for Business Server 2019 Control Panel

Administrator | Sign out  
7.0.2046.0 | Privacy statement

Home  
Users  
Topology  
IM and Presence  
Persistent Chat  
**Voice Routing**  
Voice Features  
Response Groups  
Conferencing  
Clients  
Federation and External Access  
Monitoring and Archiving  
Security  
Network Configuration

DIAL PLAN VOICE POLICY ROUTE PSTN USAGE TRUNK CONFIGURATION TEST VOICE ROUTING

Create voice routing test case information

Edit Trunk Configuration - PstnGateway:clus22pub.tekvizionlab...

OK Cancel

Scope: Pool  
Name: \*  
PstnGateway:clus22pub.tekvizionlabs.com  
Description:  
Maximum early dialogs supported: 23  
Encryption support level: Optional  
Refer support: None  
☐ Enable media bypass  
☒ Centralized media processing



## Skype for Business –Trunk Configuration (Continued)

The screenshot shows the Skype for Business Server 2019 Control Panel interface. The left sidebar contains a navigation menu with the following items: Home, Users, Topology, IM and Presence, Persistent Chat, Voice Routing (highlighted), Voice Features, Response Groups, Conferencing, Clients, Federation and External Access, Monitoring and Archiving, Security, and Network Configuration. The top navigation bar includes the title 'Skype for Business Server' and the user 'Administrator | Sign out 7.0.2046.0 | Privacy statement'. The main content area is titled 'Edit Trunk Configuration - PstnGateway:clus22pub.tekvizionlab...' and contains several configuration options: 'Create voice routing test case information' (a link), 'OK' and 'Cancel' buttons, and a list of checkboxes: 'Enable RTP latching', 'Enable forward call history', 'Enable forward P-Asserted-Identity data' (highlighted with a red box), and 'Enable outbound routing failover timer' (checked). Below these is a section for 'Associated PSTN Usages' with a 'Select...' button, a 'Remove' button, and up/down arrows. A table with two columns, 'PSTN usage record' and 'Associated routes', is shown below the arrows.

Skype for Business Server 2019 Control Panel

Administrator | Sign out 7.0.2046.0 | Privacy statement

DIAL PLAN VOICE POLICY ROUTE PSTN USAGE TRUNK CONFIGURATION TEST VOICE ROUTING

Create voice routing test case information

Edit Trunk Configuration - PstnGateway:clus22pub.tekvizionlab...

OK Cancel

☐ Enable RTP latching

☐ Enable forward call history

☐ Enable forward P-Asserted-Identity data

☒ Enable outbound routing failover timer

Associated PSTN Usages

Select... Remove ↑ ↓

PSTN usage record	Associated routes
-------------------	-------------------

## Skype for Business –Trunk Configuration (Continued)

The screenshot displays the Skype for Business Server 2019 Control Panel interface. The top navigation bar includes the Skype logo, the text "Skype for Business Server", and the user role "Administrator" with a "Sign out" link. The version "7.0.2046.0" and a "Privacy statement" link are also present. The left sidebar lists various configuration areas: Home, Users, Topology, IM and Presence, Persistent Chat, Voice Routing (highlighted), Voice Features, Response Groups, Conferencing, Clients, Federation and External Access, Monitoring and Archiving, Security, and Network Configuration. The main content area is titled "Edit Trunk Configuration - PstnGateway:clus22pub.tekvizionlab...". It features a "Create voice routing test case information" button and "OK" and "Cancel" buttons. Below these are two sections for translation rules: "Calling number translation rules" and "Called number translation rules". Each section includes a toolbar with "New", "Copy", "Paste", "Select...", "Show details...", and "Remove" options, along with up and down arrows for reordering. Each section contains a table with columns for "Translation rule", "State", "Pattern to match", and "Translation pattern".

Skype for Business Server 2019 Control Panel

Administrator | Sign out  
7.0.2046.0 | Privacy statement

DIAL PLAN VOICE POLICY ROUTE PSTN USAGE TRUNK CONFIGURATION TEST VOICE ROUTING

Create voice routing test case information

Edit Trunk Configuration - PstnGateway:clus22pub.tekvizionlab...

OK Cancel

**Calling number translation rules**

New Copy Paste Select... Show details... Remove Up Down

Translation rule	State	Pattern to match	Translation pattern
------------------	-------	------------------	---------------------

**Called number translation rules**

New Copy Paste Select... Show details... Remove Up Down

Translation rule	State	Pattern to match	Translation pattern
------------------	-------	------------------	---------------------

## Route Configuration

**Navigation:** Voice Routing -> Route

Click New

Set Name = enter a name to identify this Route. CiscoRoute is used for this test.

Add associated trunks = select the trunk configured earlier – PstnGateway: clus22pub.tekvizionlabs.com

The screenshot displays the 'Skype for Business Server 2019 Control Panel' interface. The left-hand navigation pane lists various system components, with 'Voice Routing' currently selected. The main content area features a top navigation bar with tabs for 'DIAL PLAN', 'VOICE POLICY', 'ROUTE', 'PSTN USAGE', 'TRUNK CONFIGURATION', and 'TEST VOICE ROUTING'. Below this, a 'Create voice routing test case information' link is visible. The primary section is titled 'Edit Voice Route - CiscoRoute', which includes 'OK' and 'Cancel' buttons. The configuration form contains a 'Scope:' section with a 'Name:' field (containing 'CiscoRoute') and a 'Description:' field. Below these is a 'Build a Pattern to Match' section, which includes instructions to add starting digits and a list of 'Starting digits for numbers that you want to allow'. This list currently contains one entry, 'Type a valid number and then click Add.', with 'Add', 'Exceptions', and 'Remove' buttons associated with it.

## Skype for Business –Route Configuration (Continued)

Skype for Business Server 2019 Control Panel

Administrator | Sign out  
7.0.2046.0 | Privacy statement

**Skype for Business Server**

DIAL PLAN VOICE POLICY ROUTE PSTN USAGE TRUNK CONFIGURATION TEST VOICE ROUTING

Home  
Users  
Topology  
IM and Presence  
Persistent Chat  
**Voice Routing**  
Voice Features  
Response Groups  
Conferencing  
Clients  
Federation and External Access  
Monitoring and Archiving  
Security  
Network Configuration

Create voice routing test case information

Edit Voice Route - CiscoRoute

OK Cancel

**Match this pattern:**

.

Edit Reset ?

☐ **Suppress caller ID**

**Alternate caller ID:**

**Associated trunks:**

PstnGateway:clus22pub.tekvizi...

Add... Remove

## Skype for Business –Route Configuration (Continued)

The screenshot displays the 'Skype for Business Server 2019 Control Panel' interface. The top navigation bar includes the Skype logo, the text 'Skype for Business Server', and user information: 'Administrator | Sign out 7.0.2046.0 | Privacy statement'. A secondary navigation bar contains tabs: 'DIAL PLAN', 'VOICE POLICY', 'ROUTE', 'PSTN USAGE', 'TRUNK CONFIGURATION', and 'TEST VOICE ROUTING'. The left sidebar lists various configuration areas, with 'Voice Routing' currently selected. The main content area is titled 'Edit Voice Route - CiscoRoute' and features 'OK' and 'Cancel' buttons. Below this is a large empty box with a 'REMOVE' button. The 'Associated PSTN Usages' section includes a 'Select...' button, a 'Remove' button, and up/down arrow icons. A table lists the associated PSTN usages:

PSTN usage record	Associated voice policies
CiscoPSTNUsage	Cisco

At the bottom, there is a 'Translated number to test:' label, an input field, and a 'Go' button.

## Voice Policy and PSTN Usage Configuration

**Navigation:** Voice Routing -> Voice Policy

Click New

Set Name = enter a name to identify this voice policy – Cisco is used in this test.

Set Calling Features:

- Check Enable call forwarding
- Check Enable delegation
- Check Enable call transfer
- Check Enable call park
- Check Enable simultaneous ringing of phones
- Check Enable team call
- Check Enable PSTN reroute
- Uncheck Enable bandwidth policy override
- Uncheck Enable malicious call tracing
- Enable Busy options
- Uncheck Enable busy options for Federation

Set Associated PSTN usages:

- Click New
- Set Name: enter a name to identify the PSTN Usage record – CiscoPSTNUsage is used in the test.
- Set Associated Routes = select the route created earlier= CiscoRoute

Skype for Business Server 2019 Control Panel

Administrator | Sign out  
7.0.2048.0 | Privacy statement

Home  
Users  
Topology  
IM and Presence  
Persistent Chat  
Voice Routing  
Voice Features  
Response Groups  
Conferencing  
Clients  
Federation and External Access  
Monitoring and Archiving  
Security  
Network Configuration

DIAL PLAN VOICE POLICY ROUTE PSTN USAGE TRUNK CONFIGURATION TEST VOICE ROUTING

Create voice routing test case information

Edit Voice Policy - Cisco

OK Cancel

Scope: User  
Name: \*  
Cisco  
Description:  

Calling Features

☒ Enable call forwarding

☒ Enable delegation

☒ Enable call transfer

☒ Enable call park

☒ Enable simultaneous ringing of phones

☒ Enable team call

☒ Enable PSTN reroute

☐ Enable bandwidth policy override

☐ Enable malicious call tracing

☒ Enable busy options

☐ Enable busy options for Federation

Associated PSTN Usages

New Select... Show details... Remove

PSTN usage record	Associated routes
CiscoPSTNUsage	CiscoRoute

Page 23 of 102

## Dial Plan Configuration

**Navigation:** Voice Routing-> Dial Plan

Default Dial plan used for this topology

Skype for Business Server 2019 Control Panel

Administrator | Sign out  
7.0.2048.0 | Privacy statement

DIAL PLAN VOICE POLICY ROUTE PSTN USAGE TRUNK CONFIGURATION TEST VOICE ROUTING

Home  
Users  
Topology  
IM and Presence  
Persistent Chat  
Voice Routing  
Voice Features  
Response Groups  
Conferencing  
Clients  
Federation and External Access  
Monitoring and Archiving  
Security  
Network Configuration

Create voice routing test case information

Edit Dial Plan - Global

OK Cancel

Scope: Global

Name: \*  
Global

Simple name: \*  
DefaultProfile

Description:

Dial-in conferencing region:

External access prefix:

Associated Normalization Rules

New Copy Paste Select... Show details... Remove

Normalization rule	State	Pattern to match	Translation pattern
Keep All	Committed	^(\d+)\$	\$1



## Call Park Range Configuration

**Navigation:** Voice Features -> Call Park

Click New.

Set **Name** = enter text to identify the call Park Range – Orbit range is used in the test.

Set **Number Range** = 4500 to 4599 is used in the test.

Set **FQDN of destination server**= select the desired server – fe.sflabtn.local is used in the test

Skype for Business Server 2019 Control Panel

Skype for Business Server 7.0.

Home  
Users  
Topology  
IM and Presence  
Persistent Chat  
Voice Routing  
**Voice Features**  
Response Groups  
Conferencing

CALL PARK UNASSIGNED NUMBER

Edit Call Park Number Range - Cisco call park

Commit Cancel

**Name: \***  
Cisco call park

**Number range: \***  
4500 - 4599

**FQDN of destination server: \***  
fe.sflabtn.local

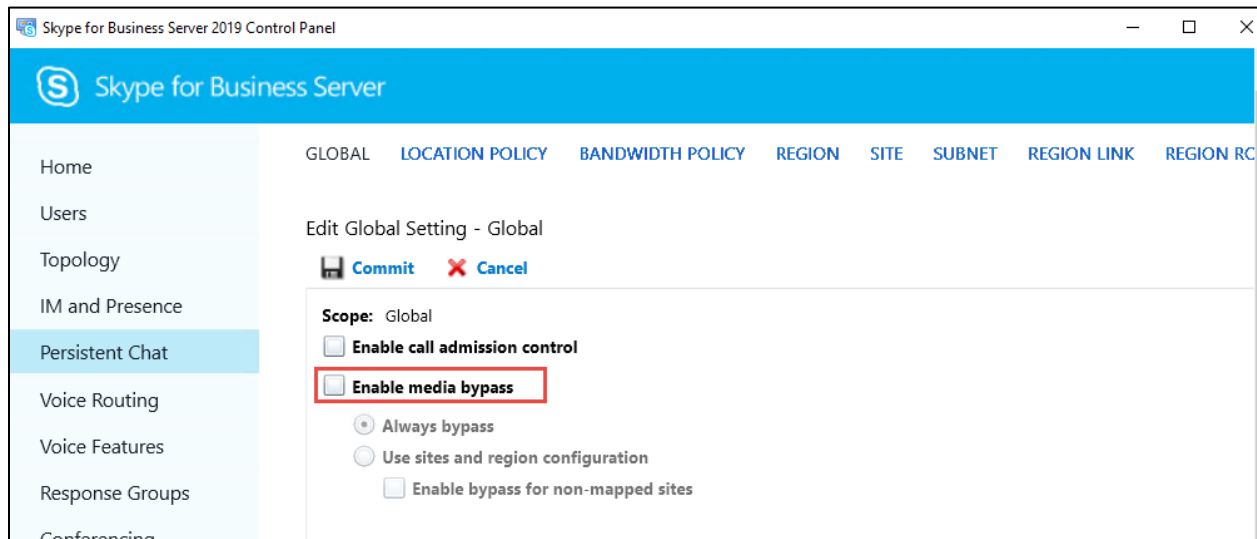
## Global Media Bypass Configuration

**Navigation:** Network Configuration -> Global

Edit Global Setting –

- Uncheck Enable media bypass

Commit the configuration.

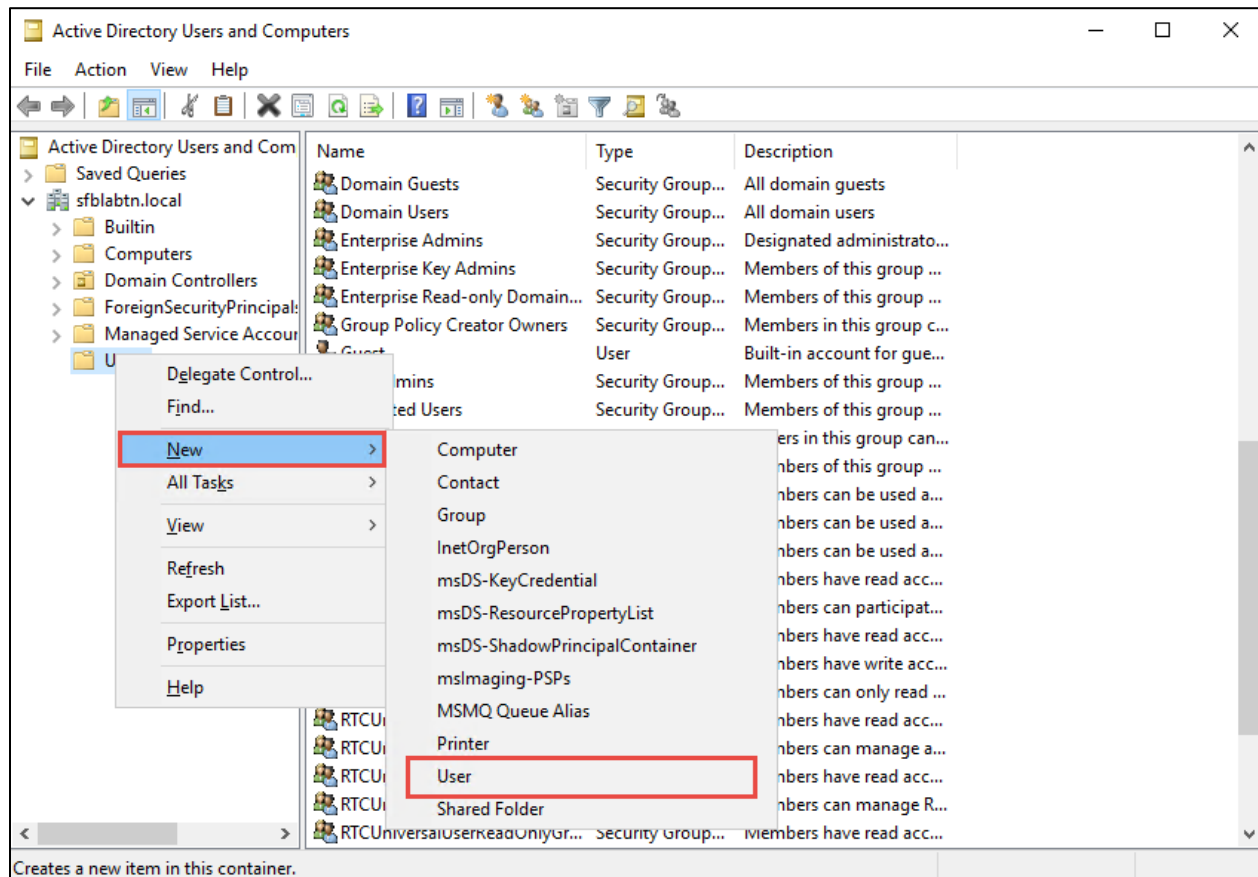


## User Configuration

Login to the Skype for Business Active Directory

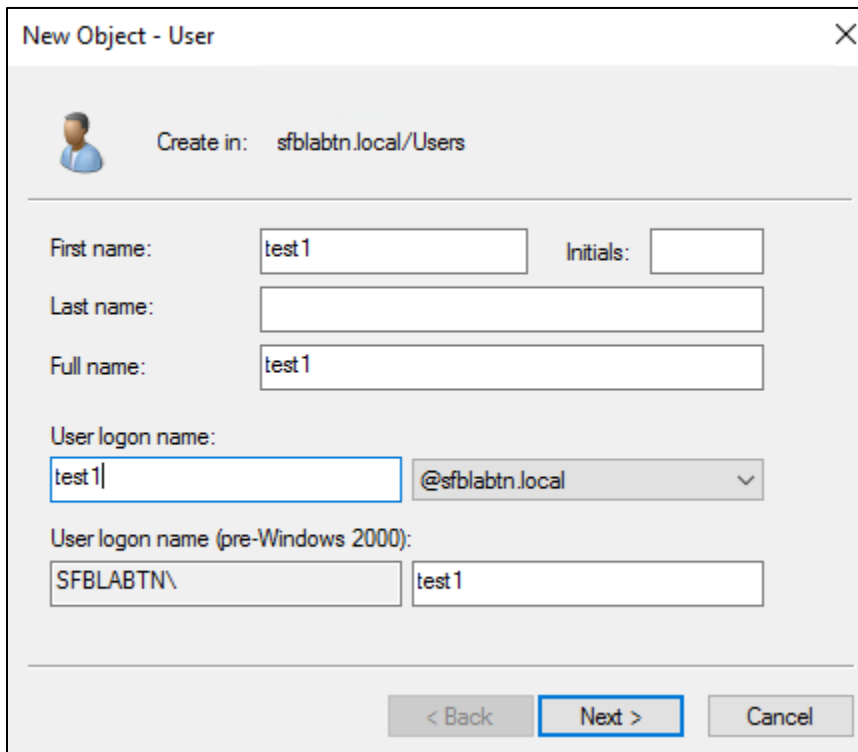
Navigation: Active Directory Users and Computers → Users

Add a New User



## Skype for Business – New User configuration (continued)

Follow the screenshots below to add a new user:




The screenshot shows the 'New Object - User' dialog box. At the top, it says 'Create in: sfblabtn.local/Users'. Below this, there are several input fields:

- First name:** test1
- Initials:** (empty)
- Last name:** (empty)
- Full name:** test1
- User logon name:** test1 (with a dropdown menu showing @sfblabtn.local)
- User logon name (pre-Windows 2000):** SFBLABTN\ (with a dropdown menu showing test1)

At the bottom, there are three buttons: '< Back', 'Next >' (highlighted with a blue border), and 'Cancel'.

## Skype for Business – New User configuration (continued)

New Object - User

 Create in: sfblabtn.local/Users

Password:

••••••••

Confirm password:

••••••••

☐ User must change password at next logon

☒ User cannot change password

☒ Password never expires

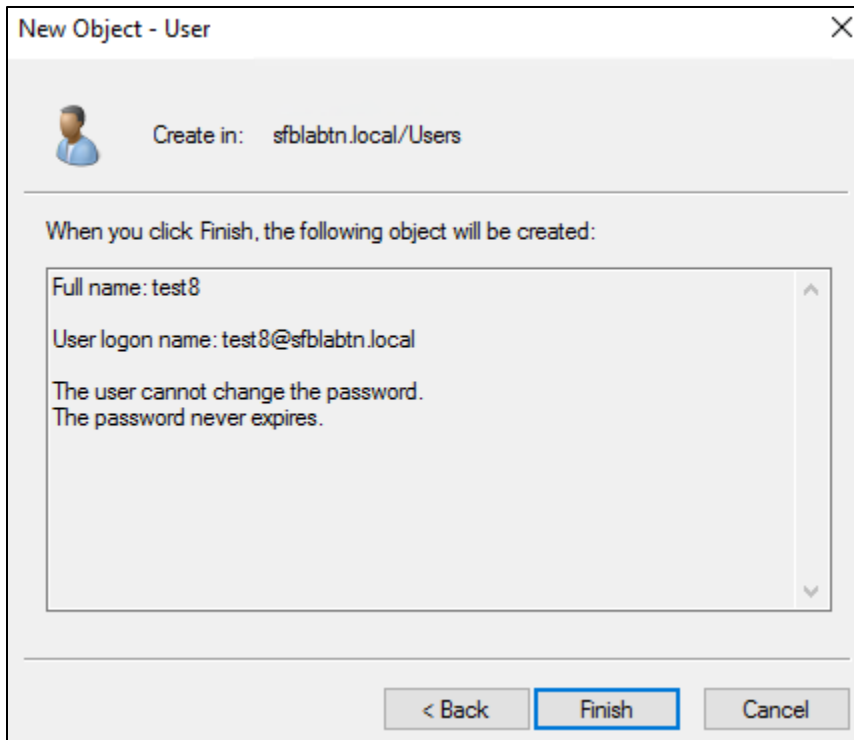
☐ Account is disabled

< Back

Next >

Cancel

## Skype for Business – New User configuration (continued)



The screenshot shows a dialog box titled "New Object - User" with a close button (X) in the top right corner. Below the title bar, there is a user icon and the text "Create in: sfblabtn.local/Users". A horizontal line separates this from the main content area, which starts with the text "When you click Finish, the following object will be created:". Below this text is a scrollable list box containing the following information:

- Full name: test8
- User logon name: test8@sfblabtn.local
- The user cannot change the password.
- The password never expires.

At the bottom of the dialog box, there are three buttons: "< Back", "Finish" (which is highlighted with a blue border), and "Cancel".

Once the user is created, login to the Skype for Business 2019 Control Panel

Navigation: Users → Enable users

Click on the Add button and find the new user created earlier.

## Skype for Business – New User configuration (continued)

Skype for Business Server 2019 Control Panel

Skype for Business Server

Home  
Users  
Topology  
IM and Presence  
Persistent Chat  
Voice Routing  
Voice Features  
Response Groups  
Conferencing  
Clients  
Federation and External Access  
Monitoring and Archiving  
Security

USER SEARCH

New Skype for Business Server User

Enable Cancel

**Users:**

Display name	Status
--------------	--------

Add...  
Remove

**Assign users to a pool: \***

**Generate user's SIP URI:**

☒ Use user's email address  
☐ Use the user principal name (UPN)  
☐ Use the following format:  
    <FirstName>.<LastName> @

☐ Use the following format:

Set Assign users to a **pool**= **fe.sfbtabtn.local** from drop down menu

Set Generate user's **SIP URI**: Specify a SIP URI: sip:test1@sfbsp.local .This is used in this test

Set **Telephony**=**Enterprise Voice**

Set Line **URI**: = **tel: +5001** is used for the test. This is the DN for the user.

Set Dial **plan policy** = **Automatic** (as configured earlier)

Set **Voice policy**= **Cisco** (as configured earlier)

Click Enable.

## Skype for Business – New User configuration (continued)

Skype for Business Server 2019 Control Panel

Administrator | Sign out  
7.0.2046.0 | Privacy statement

Home

Users

Topology

IM and Presence

Persistent Chat

Voice Routing

Voice Features

Response Groups

Conferencing

Clients

Federation and External Access



Monitoring and Archiving

Security

Network Configuration

USER SEARCH

Edit Skype for Business Server User - test1

 Commit  Cancel

**Display name:**  
test1

☒ Enabled for Skype for Business Server

**SIP address: \***  
sip:test1 @ sfb-labtn.local

**Registrar pool:**  
fe.sfb-labtn.local ?

**Telephony:**  
Enterprise Voice ?

**Line URI:**  
tel:+5001 ?

**Dial plan policy:**  
<Automatic> View...

**Voice policy:**  
Cisco View...



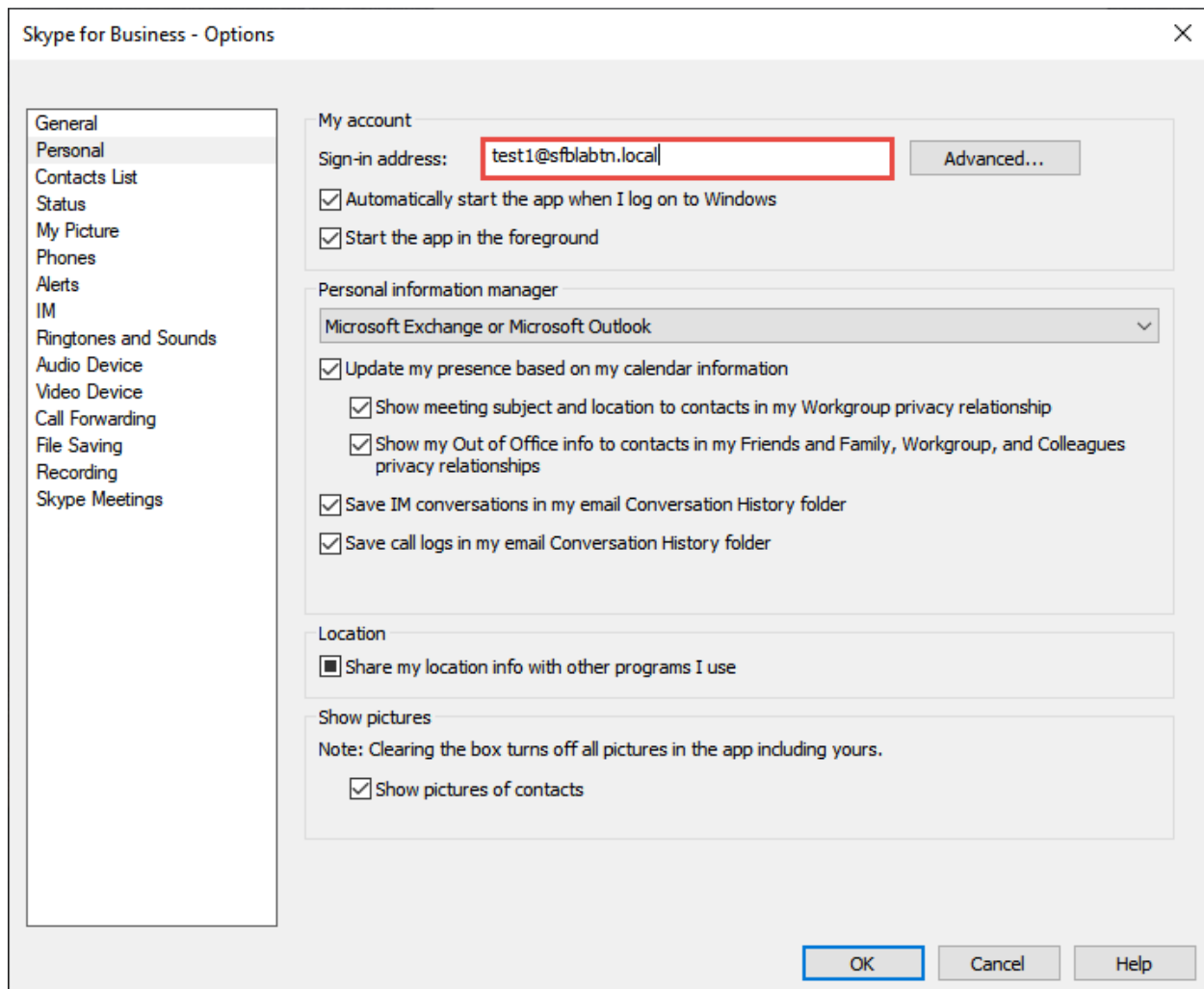
## Client Configuration

Download the latest version of the Skype for Business client and launch the same.

Navigation: Settings→Tools→Options→Personal→MyAccount

Set Sign-in-address= enter the sip uri of the user configured in username@domain format.

[test1@sfblabtn.local](mailto:test1@sfblabtn.local) is used for example.



Click Advanced. Select Manual Configuration.

Set Internal Server Name= Enter the FQDN of the domain (fe.sfblabtn.local is used for example)

Advanced Connection Settings ✕

Select the method you want to use to configure your connection settings.

☐ Automatic configuration

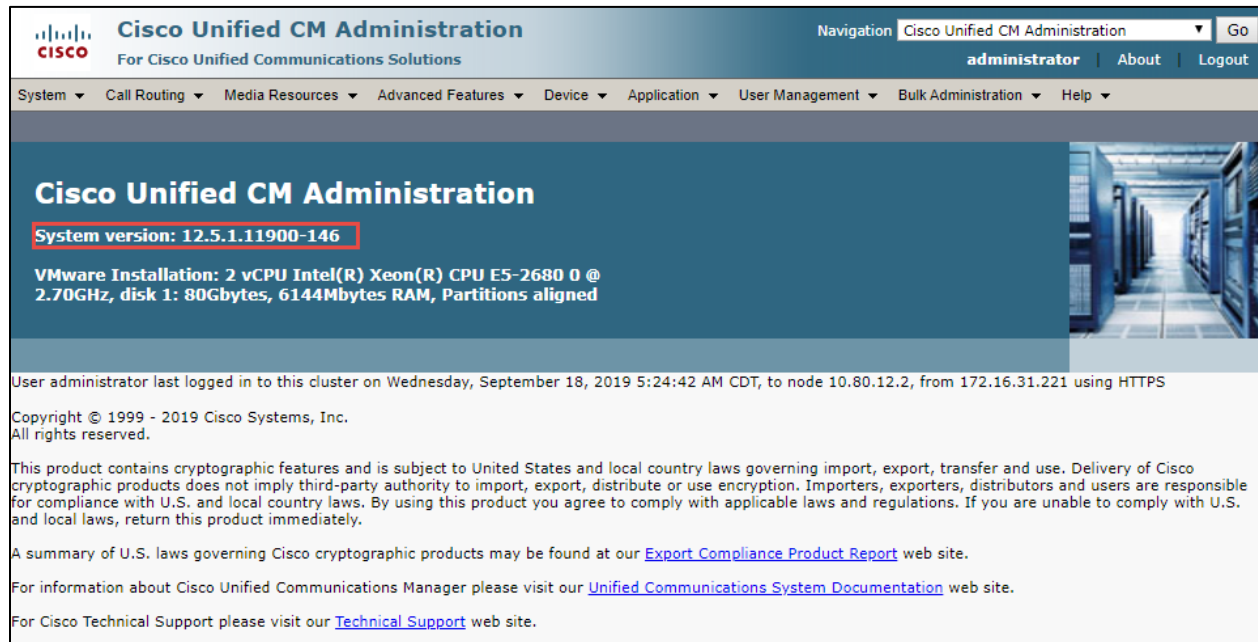
☒ Manual configuration

Internal server name:

External server name:

# Configuring the Cisco Unified Communications Manager

## Cisco Unified Communications Manager Software Version



## SIP Trunk Security Profile for Trunk to Skype for Business

**Navigation:** System → Security → SIP trunk security profile

Set **Name\*** = **SFB\_SECURITY\_PROFILE**. This is used for the test.

Set **Device Security mode** = **Encrypted**

Set **Incoming Transport Type** = **TLS**

Set **Outgoing Transport Type** = **TLS**

Set **X.509 Subject Name** = **fe.sflabtn.local**

Set **Incoming Port** = **5061**

Check **Accept Presence Subscription**

Check **Accept out of dialog refer**

Check **Accept unsolicited notification**

Check **Accept Replaces header**

Check **Transmit security status**

All other values are default.

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

### SIP Trunk Security Profile Configuration

Related Links: Back To Find/List

Save | Delete | Copy | Reset | Apply Config | Add New

**Status**  
Status: Ready

**SIP Trunk Security Profile Information**

Name\* SFB\_SECURITY\_PROFILE

Description SFB\_SECURITY\_PROFILE

Device Security Mode Encrypted

Incoming Transport Type\* TLS

Outgoing Transport Type TLS

☐ Enable Digest Authentication

Nonce Validity Time (mins)\* 600

Secure Certificate Subject or Subject Alternate Name fe.sfbtabtn.local

Incoming Port\* 5061

☐ Enable Application level authorization

☒ Accept presence subscription

☒ Accept out-of-dialog refer\*\*

☒ Accept unsolicited notification

☒ Accept replaces header

☒ Transmit security status

☐ Allow charging header

SIP V.150 Outbound SDP Offer Filtering\* Use Default Filter

## SIP Trunk Security Profile for Trunk to Unity Connection

**Navigation:** System → Security → SIP trunk security profile

Set **Name\*** = TLS\_CUC. This is used for the test.

Set Device **Security mode** = Encrypted

Set **Incoming Transport Type** = TLS

Set **Outgoing Transport Type** = TLS

Set **X.509 Subject Name** = clus22cuc.tekvizionlabs.com

Set **Incoming Port** = 5061

check **Accept Presence Subscription**

Check **Accept out-of-dialog refer\*\***

Check **Accept unsolicited notification**

Check **Accept Replaces header**

## Check Transmit security status

All other values are default.

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

### SIP Trunk Security Profile Configuration

Related Links: Back To Find/List | Go

Save | Delete | Copy | Reset | Apply Config | Add New

**Status**  
Status: Ready

**SIP Trunk Security Profile Information**

Name*	TLS_CUC
Description	TLS_CUC
Device Security Mode	Encrypted
Incoming Transport Type*	TLS
Outgoing Transport Type	TLS
<input type="checkbox"/> Enable Digest Authentication	
Nonce Validity Time (mins)*	600
Secure Certificate Subject or Subject Alternate Name	clus22cuc.tekvizionlabs.com
Incoming Port*	5061
<input type="checkbox"/> Enable Application level authorization	
<input checked="" type="checkbox"/> Accept presence subscription	
<input checked="" type="checkbox"/> Accept out-of-dialog refer**	
<input checked="" type="checkbox"/> Accept unsolicited notification	
<input checked="" type="checkbox"/> Accept replaces header	
<input checked="" type="checkbox"/> Transmit security status	
<input type="checkbox"/> Allow charging header	
SIP V.150 Outbound SDP Offer Filtering*	Use Default Filter

## SIP Profile

**Navigation:** Device → Device Settings → SIP Profile

Set **Name\*** = **SFB\_SIP\_PROFILE** - Standard SIP Profile. This is used for this test.

Set **SIP Rel1XX Options** = **Send PRACK for all 1xx messages**

Set **Early Offer support for voice and video calls** = **Best Effort (no MTP inserted)**

Check **Enable OPTIONS Ping** to monitor Destination status for Trunks with Service Type "None (Default)"

All other values are default.

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go  
administrator | Search Documentation | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help

**SIP Profile Configuration** Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

**Status**

- Status: Ready
- All SIP devices using this profile must be restarted before any changes will take affect.

**SIP Profile Information**

Name*	SFB_SIP_PROFILE
Description	SFB_SIP_PROFILE
Default MTP Telephony Event Payload Type*	101
Early Offer for G.Clear Calls*	Disabled
User-Agent and Server header information*	Send Unified CM Version Information as User-Agent
Version in User Agent and Server Header*	Major And Minor
Dial String Interpretation*	Phone number consists of characters 0-9, *, #, and
Confidential Access Level Headers*	Disabled

☐ Redirect by Application

☐ Disable Early Media on 180

☐ Outgoing T.38 INVITE include audio mline

☐ Offer valid IP and Send/Receive mode only for T.38 Fax Relay







☐ Use Fully Qualified Domain Name in SIP Requests

☐ Assured Services SIP conformance

☐ Enable External QoS\*\*

## Cisco Unified Communications Manager SIP Profile (Continued)

**SIP Profile Configuration**Related Links: [Back To Find/List](#) [Go](#)

 Save  Delete  Copy  Reset  Apply Config  Add New

**SDP Information**

SDP Session-level Bandwidth Modifier for Early Offer and Re-invites\*

SDP Transparency Profile

Accept Audio Codec Preferences in Received Offer\*

☐ Require SDP Inactive Exchange for Mid-Call Media Change

☐ Allow RR/RS bandwidth modifier (RFC 3556)

TIAS and AS

Pass all unknown SDP attributes

Default

**Parameters used in Phone**

Timer Invite Expires (seconds)\*

180

Timer Register Delta (seconds)\*

5

Timer Register Expires (seconds)\*

3600

Timer T1 (msec)\*

500

Timer T2 (msec)\*

4000

Retry INVITE\*

6

Retry Non-INVITE\*

10

Media Port Ranges

☒ Common Port Range for Audio and Video

☐ Separate Port Ranges for Audio and Video

Start Media Port\*

16384

Stop Media Port\*

32766

DSCP for Audio Calls

Use System Default







DSCP for Video Calls

Use System Default

## Cisco Unified Communications Manager SIP Profile (Continued)

**SIP Profile Configuration**

Related Links: [Back To Find/List](#) [Go](#)







 Save  Delete  Copy  Reset  Apply Config  Add New

DSCP for Audio Portion of Video Calls	<div>Use System Default</div>
DSCP for TelePresence Calls	<div>Use System Default</div>
DSCP for Audio Portion of TelePresence Calls	<div>Use System Default</div>
Call Pickup URI*	<div>x-cisco-serviceuri-pickup</div>
Call Pickup Group Other URI*	<div>x-cisco-serviceuri-opickup</div>
Call Pickup Group URI*	<div>x-cisco-serviceuri-gpickup</div>
Meet Me Service URI*	<div>x-cisco-serviceuri-meetme</div>
User Info*	<div>None</div>
DTMF DB Level*	<div>Nominal</div>
Call Hold Ring Back*	<div>Off</div>
Anonymous Call Block*	<div>Off</div>
Caller ID Blocking*	<div>Off</div>
Do Not Disturb Control*	<div>User</div>
Telnet Level for 7940 and 7960*	<div>Disabled</div>
Resource Priority Namespace	<div>&lt; None &gt;</div>
Timer Keep Alive Expires (seconds)*	<div>120</div>
Timer Subscribe Expires (seconds)*	<div>120</div>
Timer Subscribe Delta (seconds)*	<div>5</div>
Maximum Redirections*	<div>70</div>
Off Hook To First Digit Timer (milliseconds)*	<div>15000</div>
Call Forward URI*	<div>x-cisco-serviceuri-cfwdall</div>



## Cisco Unified Communications Manager SIP Profile (Continued)

**SIP Profile Configuration**Related Links: [Back To Find/List](#) [Go](#)

 Save  Delete  Copy  Reset  Apply Config  Add New



Speed Dial (Abbreviated Dial) URI\*

☒ Conference Join Enabled  
☐ RFC 2543 Hold  
☒ Semi Attended Transfer  
☐ Enable VAD  
☐ Stutter Message Waiting  
☐ MLPP User Authorization

**Normalization Script**

Normalization Script

☐ Enable Trace

	Parameter Name	Parameter Value	
1	<input type="text"/>	<input type="text"/>	 

**Incoming Requests FROM URI Settings**

Caller ID DN

Caller Name

**Trunk Specific Configuration**

Reroute Incoming Request to new Trunk based on\*

Resource Priority Namespace List

SIP Rel1XX Options\*

## Cisco Unified Communications Manager SIP Profile (Continued)

**SIP Profile Configuration** Related Links: [Back To Find/List](#) [Go](#)

Save Delete Copy Reset Apply Config Add New

Session Refresh Method\* Invite

Early Offer support for voice and video calls\* Best Effort (no MTP inserted)

☐ Enable ANAT

☐ Deliver Conference Bridge Identifier

☐ Allow Passthrough of Configured Line Device Caller Information

☐ Reject Anonymous Incoming Calls

☐ Reject Anonymous Outgoing Calls

☐ Send ILS Learned Destination Route String

☐ Connect Inbound Call before Playing Queuing Announcement

**SIP OPTIONS Ping**

☒ Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"

Ping Interval for In-service and Partially In-service Trunks (seconds)\* 60

Ping Interval for Out-of-service Trunks (seconds)\* 120

Ping Retry Timer (milliseconds)\* 500

Ping Retry Count\* 6

**SDP Information**

☐ Send send-receive SDP in mid-call INVITE

☐ Allow Presentation Sharing using BFCP

☐ Allow iX Application Media

☒ Allow multiple codecs in answer SDP


## Media Resource Group

**Navigation Path:** Media Resources → Media Resource Group; Add New  
**Media Resource Group MRG**

Set **Name\*** = **MRG\_SW\_MTP**, This is used for this test.

Set all resources in the Selected Media Resources\* Box.





All other values are default.



**Cisco Unified CM Administration**  
 For Cisco Unified Communications Solutions

Navigation **Cisco Unified CM Administration** Go  
**administrator** | [Search Documentation](#) | [About](#) | [Logout](#)

System ▾ | Call Routing ▾ | Media Resources ▾ | Advanced Features ▾ | Device ▾ | Application ▾ | User Management ▾ | Bulk Administration ▾  
 Help ▾

**Media Resource Group Configuration**
Related Links: [Back To Find/List](#) Go

 Save
  Delete
  Copy
  Add New

**Status**  
 Status: Ready

**Media Resource Group Status**  
 Media Resource Group: MRG\_SW\_MTP (used by 0 devices)

**Media Resource Group Information**  
 Name\* MRG\_SW\_MTP  
 Description With SW MTP

**Devices for this Group**  
 Available Media Resources\*\*   

▼ ▲

 Selected Media Resources\*
 

ANN\_2 (ANN)  
 ANN\_3 (ANN)  
 CFB\_2 (CFB)  
 CFB\_3 (CFB)  
 IVR\_2 (IVR)

☐ Use Multi-cast for MOH Audio (If at least one multi-cast MOH resource is available)

Save Delete Copy Add New


### Resource Group for MRG\_SW\_NoMTP

Set **Name\***= **MRG\_SW\_NoMTP**. This is used for the test.

Set **Available Media Resources** = **MTP\_2, MTP\_3 and MTP\_4**

Set other resources in the **Selected Media Resources\***

All other values are default.



Cisco Unified CM Administration

For Cisco Unified Communications Solutions

Navigation

Cisco Unified CM Administration

Go

administrator | [Search Documentation](#) | [About](#) | [Logout](#)

System

Call Routing

Media Resources

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Help

Media Resource Group Configuration

Related Links: [Back To Find/List](#) Go

 Save  Delete  Copy  Add New

Status

 Status: Ready

Media Resource Group Status

Media Resource Group: MRG\_SW\_NoMTP (used by 3 devices)

Media Resource Group Information

Name\*

MRG\_SW\_NoMTP

Description

With Out MTP

Devices for this Group

Available Media Resources\*\*

MTP\_2  
MTP\_3  
MTP\_4

Selected Media Resources\*

CFB\_3 (CFB)  
IVR\_2 (IVR)  
IVR\_3 (IVR)  
MOH\_2 (MOH)  
MOH\_3 (MOH)

☐ Use Multi-cast for MOH Audio (If at least one multi-cast MOH resource is available)


Save

Delete

Copy

Add New

## Cisco Unified Communications Manager Media Resource Group Configuration


**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go  
**administrator** | [Search Documentation](#) | [About](#) | [Logout](#)

[System](#) ▾ [Call Routing](#) ▾ [Media Resources](#) ▾ [Advanced Features](#) ▾ [Device](#) ▾ [Application](#) ▾ [User Management](#) ▾ [Bulk Administration](#) ▾  
[Help](#) ▾



**Find and List Media Resource Groups**

[+](#) Add New [Select All](#) [Clear All](#) [Delete Selected](#)

**Status**  
 5 records found

**Media Resource Group** (1 - 5 of 5) **Rows per Page** 50 ▾

Find Media Resource Group where Name ▾ begins with ▾  [Find](#) [Clear Filter](#) [+](#) [-](#)

<input type="checkbox"/>	Name ^	Description	Multi-cast	Copy
<input type="checkbox"/>	<a href="#">MRG_SW_MTP</a>	With SW MTP	false	
<input type="checkbox"/>	<a href="#">MRG_SW_NoMTP</a>	With Out MTP	false	

[Add New](#) [Select All](#) [Clear All](#) [Delete Selected](#)

## Media Resource Group List

Navigation Path: Media Resources→ Media Resource Group List

Add New

Set **Name\***= **MRGL\_SW\_MTP**. This is used for this test.

Set **Available Media Resources** = **MRG\_NO-MTP**

Set **Selected Media Resource Groups**= **MRGL\_SW\_MTP**

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and a navigation dropdown menu set to "Cisco Unified CM Administration". Below this is a secondary navigation bar with links for "administrator", "Search Documentation", "About", and "Logout". A main navigation bar contains various system categories like "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", and "Bulk Administration".

The main content area is titled "Media Resource Group List Configuration". It features a "Related Links" section with a "Back To Find/List" link. Below this is a toolbar with icons for "Save", "Delete", "Copy", and "Add New".

The configuration section is divided into several panels:

- Status:** Shows "Status: Ready".
- Media Resource Group List Status:** Displays "Media Resource Group List: MRGL\_SW\_MTP (used by 0 devices)".
- Media Resource Group List Information:** Contains a "Name\*" field with the value "MRGL\_SW\_MTP".
- Media Resource Groups for this List:** This section contains two dropdown menus. The "Available Media Resource Groups" dropdown has "MRG\_NO-MTP" selected. The "Selected Media Resource Groups" dropdown has "MRGL\_SW\_MTP" selected. Arrows indicate the relationship between these groups.


At the bottom of the configuration section, there are buttons for "Save", "Delete", "Copy", and "Add New". A footer note states: "\*- indicates required item."

Add new

Set **Name\***= **MRGL\_SW\_NoMTP**. This is used for the test

Set **Available Media Resources** **MRG\_SW\_MTP**

Set **Selected Media Resource Groups**= **MRGL\_SW\_NoMTP**



Cisco Unified CM Administration

For Cisco Unified Communications Solutions

Navigation

Cisco Unified CM Administration





Go

administrator | [Search Documentation](#) | [About](#) | [Logout](#)


System ▾ | Call Routing ▾ | Media Resources ▾ | Advanced Features ▾ | Device ▾ | Application ▾ | User Management ▾ | Bulk Administration ▾ | Help ▾

Media Resource Group List Configuration

Related Links: [Back To Find/List](#) Go

 Save  Delete  Copy  Add New

Status

 Status: Ready

Media Resource Group List Status

Media Resource Group List: MRGL\_SW\_NoMTP (used by 3 devices)

Media Resource Group List Information

Name\*

Media Resource Groups for this List

Available Media Resource Groups

MRG SW MTP

Selected Media Resource Groups

MRG\_SW\_NoMTP


↕ ↕

Save

Delete

Copy

Add New

 \*- indicates required item.

## Cisco Unified Communications Manager Media Resource Group List Configuration

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

### Find and List Media Resource Group Lists

+ Add New | Select All | Clear All | Delete Selected

**Status**  
5 records found

**Media Resource Group List (1 - 5 of 5)** Rows per Page 50

Find Media Resource Group List where Name begins with [ ] Find Clear Filter + -

	Name ^	Copy
<input type="checkbox"/>	MRGL SW MTP	
<input type="checkbox"/>	MRGL SW NoMTP	

+ Add New | Select All | Clear All | Delete Selected

### Device Pool Configuration

Device Pool - **G711\_Pool** is created in this test.

**Navigation Path:** System → Device Pool

Add New.

Set **Device Pool Name\***= **G711\_pool**. This is used in the test.

Set **Cisco Unified Communications Manager Group\***= **Default**


Set **Date/Time Group\*** = **CMLocal**

Set **Region\*** = **G711 Region**. This is used in this example

Set **Media Resource Group List** =**MRGL\_NO\_MTP**. This is used in this example.

All other values are default.










**Cisco Unified CM Administration**  
 For Cisco Unified Communications Solutions

Navigation **Cisco Unified CM Administration** Go  
**administrator** | [Search Documentation](#) | [About](#) | [Logout](#)

System ▾ | [Call Routing](#) ▾ | [Media Resources](#) ▾ | [Advanced Features](#) ▾ | [Device](#) ▾ | [Application](#) ▾ | [User Management](#) ▾ | [Bulk Administration](#) ▾ | [Home](#)

**Device Pool Configuration**
Related Links: [Back To Find/List](#) Go

 Save
  Delete
  Copy
  Reset
  Apply Config
  Add New

---

**Device Pool Information**

Device Pool: G711\_pool (10 members\*\*)

---

**Device Pool Settings**

Device Pool Name*	G711_pool
Cisco Unified Communications Manager Group*	Default ▾
Calling Search Space for Auto-registration	< None > ▾
Adjunct CSS	< None > ▾
Reverted Call Focus Priority	Default ▾
Intercompany Media Services Enrolled Group	< None > ▾







---

**Roaming Sensitive Settings**

Date/Time Group*	CMLocal ▾
Region*	G711 Region ▾
Media Resource Group List	MRGL_NO_MTP ▾
Location	Hub_None ▾
Network Locale	< None > ▾
SRST Reference*	Disable ▾
Connection Monitor Duration***	<input type="text"/>
Single Button Barge*	Default ▾
Join Across Lines*	Default ▾
Physical Location	< None > ▾
Device Mobility Group	< None > ▾

## Cisco Unified Communications Manager Device Pool Configuration (Continued)

**Device Pool Configuration**Related Links: [Back To Find/List](#) [Go](#)

 Save  Delete  Copy  Reset  Apply Config  Add New

Wireless LAN Profile Group < None > [View Details](#)

**Local Route Group Settings**

Standard Local Route Group < None >

**Device Mobility Related Information\*\*\*\***

Device Mobility Calling Search Space < None >

AAR Calling Search Space < None >

AAR Group < None >

Calling Party Transformation CSS < None >

Called Party Transformation CSS < None >

**Geolocation Configuration**

Geolocation < None >

Geolocation Filter < None >

**Call Routing Information**

**Incoming Calling Party Settings**

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

[Clear Prefix Settings](#) [Default Prefix Settings](#)

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	<input type="text" value="Default"/>	<input type="text"/>	<span>&lt; None &gt;</span>

## Cisco Unified Communications Manager Device Pool Configuration (Continued)

Device Pool Configuration

Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

International Number

Default

< None >

Unknown Number

Default

< None >

Subscriber Number

Default

< None >

Incoming Called Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	Default	0	< None >
International Number	Default	0	< None >
Unknown Number	Default	0	< None >
Subscriber Number	Default	0	< None >

Phone Settings

Caller ID For Calls From This Phone

Calling Party Transformation CSS < None >

Connected Party Settings

Connected Party Transformation CSS < None >

Redirecting Party Settings

Redirecting Party Transformation CSS < None >

Save Delete Copy Reset Apply Config Add New

i

\*- indicates required item.

i

\*\*Number of devices that have to be reset when this device pool is updated. To see a detailed list of these devices and other dependencies, click on Dependency Records.

i

\*\*\*Leave the field blank or enter -1 to use the configuration from the enterprise parameter.

i

\*\*\*\*These five parameters will overwrite device level settings when device is roaming and in the same device mobility group.

## Region Configuration

**Navigation Path:** System → Region Information → Region

Add New

G711 Region is created in this test.

Set **Name\***= **G711 Region**. This is used in this example

Set **Region**= **G711 Region**. This is used in this example

Set **Audio Codec Preference List**= **G711\_Preferred Codec List**

Set **Maximum Audio Bit Rate**= **64 Kbps (G7.22, G7.11)**. This is used in this example

Set **Region**=**Default**. This is used in this example

Set **Audio Codec Preference List**= **G711\_Preferred Codec List**. This is used in this example

Set **Maximum Audio Bit Rate**= **64 Kbps (G722, G7.11)**. This is used in this example

All other values are default

The screenshot displays the Cisco Unified CM Administration interface. The top navigation bar includes the Cisco logo, the title 'Cisco Unified CM Administration', and a navigation dropdown menu. Below the navigation bar, there is a breadcrumb trail: System > Call Routing > Media Resources > Advanced Features > Device > Application > User Management > Bulk Administration > Help. The main content area is titled 'Region Configuration' and includes a 'Related Links' section with a 'Back To Find/List' link. Below this, there are icons for Save, Delete, Reset, Apply Config, and Add New. The 'Region Information' section shows a form with 'Name\*' set to 'G711 Region'. The 'Region Relationships' section contains a table with the following data:

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
Default	G711_Preferred Codec List	64 kbps (G.722, G.711)	Use System Default (384 kbps)	Use System Default (2000000000 kbps)
G711 Region	G711_Preferred Codec List	64 kbps (G.722, G.711)	Use System Default (384 kbps)	Use System Default (2000000000 kbps)

Below the table, there is a note: 'NOTE: Regions not displayed'. The table also includes a row for 'Use System Default' for each column.

## Normalization Script

**Navigation:** Device->Device Settings->SIP Normalization Script

Add New

Set Name = enter text here to identify the normalization script for use on trunk. lync\_interop\_updated is used in this test.

Set Content = add script content.

Note: "lync\_interop" was the originally provided script by Cisco Support for Cisco UCM-Skype for Business TLS integration. However, with the script activated, the call from Skype for Business to Cisco UCM was not established on SRTP still. Cisco UCM sent AVP (that is, chose RTP instead of SRTP) when Skype for Business sent a=tcap: 1 RTP/SAVP through INVITE SDP. The script was updated to fix this issue and "lync\_interop\_updated" is the updated script.

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go

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**SIP Normalization Script Configuration** Related Links: Back To Find/List Go

Save Delete Reset Add New Import File

Status: Ready

**SIP Normalization Script Info**

Name\* lync\_interop\_updated

Description

Content\*

```
--[[
Description:
Provides interoperability for Microsoft Lync

Handle Below Scenarios

1. Add user=phone for all outbound Invite messages because it is
mandatory for Lync

2. Change the CT=Line values to 1000 , Moderate bandwidth in all
outgoing messages from CUCM to Lync

3. There is Remote ringback hear issue
There is issue with PRACK enabled on CUCM and media bypass enabled on
Lync. Enabling media bypass on Lync allows the rtp from lync
]]--
```

Script Execution Error Recovery Action\* Message Rollback Only

System Resource Error Recovery Action\* Disable Script

Memory Threshold\* 50 kilobytes

Lua Instruction Threshold\* 1000 instructions

Save Delete Reset Add New Import File

## Normalization Script

--[[

### Description:

Provides interoperability for Microsoft Lync

Handle Below Scenarios

1. Add user=phone for all outbound Invite messages because it is mandatory for Lync
2. Change the CT=Line values to 1000 , Moderate bandwidth in all outgoing messages from CUCM to Lync
3. There is Remote ringback hear issue  
There is issue with PRACK enabled on CUCM and media bypass enabled on Lync. Enabling media bypass on Lync allows the rtp from lync endpoint to flow through CUCM directly instead of flowing through Mediation server. The problem with PRACK enabled is that Lync endpoint is now not able to answer the incoming call. Looking into the traces, it appears that even though Lync sent updated connection information in 183 w/sdp, the call manager is still sending rtp to the mediation server which seems to be incorrect" So In this scenario CUCM expects 180 Ringing not 183 Session progress.  
So added the Script to convert 183 Session Progress to 180 Ringing.
4. There is incoming Invite from Lync and in From Header there is "user=phone" which cause CUCM to send malformed data in to different layers which cause call failure. So this is work around for that Scenario.
5. Script modify the AS header which from outgoing messages because call forward fails due to bandwidth negotiation value is A=64 is not supported
6. Script convert the History info to diversion Header since call forward to unity Is not supported.
7. Transfer Scenario: Referred-By in Incoming Invite is converted to Diversion Header.

### Script Parameters:

Release: 9.1(2) , 10.0.(1)

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All rights reserved.

```

--]]

M = {}

M.allowHeaders = {"History-Info"}

trace.enable()

local function getDisplayName (i_header)
    local position_of_uri=string.find(i_header, "<")
    if position_of_uri <= 2
    then
        display_name=nil
    else
        -- save display name which arrives in quotes
        local display_name_tmp = string.sub(i_header,1, (position_of_uri - 1))
        -- now remove the quotes
        display_name_tmp = string.gsub(display_name_tmp,"","")
        -- now remove the space
        display_name = string.gsub(display_name_tmp,' ','')
    end
    return display_name
end

local function modify_CT_bandwidth(msg)
    local sdp = msg:getSdp()
    if sdp
    then
        local b_CT_line = sdp:getLine("b=CT","64")
        if not b_CT_line
        then
            local b_CT_line = sdp:getLine("b=CT","0")
            if not b_CT_line
            then
                return
            end
            b_CT_line = b_CT_line:gsub("0", "1000")
            sdp = sdp:modifyLine("b=CT", "0", b_CT_line)
            msg:setSdp(sdp)
            return
        end
        b_CT_line = b_CT_line:gsub("64", "1000")
        sdp = sdp:modifyLine("b=CT", "64", b_CT_line)
        msg:setSdp(sdp)
    end
end

local function remove_AS_bandwidth(msg)

```

```

local sdp = msg:getSdp()
if sdp
then
    local b_AS_line = sdp:getLine("b=AS", "64")
    if b_AS_line
    then
        sdp = sdp:removeLine("b=AS", "64")
        msg:setSdp(sdp)
    end
end
end

local function process_outbound_request(msg)

    local method, ruri, ver = msg:getRequestLine()

    if string.find(ruri, "@")
    then
        local uri = ruri .. ";user=phone"
        msg:setRequestUri(uri)
    end

    modify_CT_bandwidth(msg)
    remove_AS_bandwidth(msg)

end

local function process_outbound_message(msg)
    modify_CT_bandwidth(msg)
    remove_AS_bandwidth(msg)
end

local function process_inbound_progress(msg)

    msg:setResponseCode(180, "Ringing")
    local sdp = msg:getSdp()

    if sdp then
        sdp = sdp:removeMediaDescription("audio")
        msg:setSdp(sdp)
    end

    local req = msg:getHeader("Require")
    local reqHeader = req
    if req
    then
        msg:removeHeader("Require")
    end
end

```



```

end

local rseq = msg:getHeader("Rseq")
local rseqPresnt = rseq
if rseq
then
    seqVal = msg:getHeaderValues("Rseq")
    msg:removeHeader("Rseq")
end

local sdp = msg:getSdp()
if sdp
then
    msg:removeUnreliableSdp()
end

if reqHeader
then
    msg:addHeader("Require", "100rel")
end

if rseqPresnt
then
    msg:addHeader("RSeq", seqVal[1])
end
end

-- Future reference for changing cause values in diversion header scenario
-- local HiCauseToDiversion = { }
-- HiCauseToDiversion["302"] = "unconditional"
-- HiCauseToDiversion["486"] = "user-busy"
-- HiCauseToDiversion["408"] = "no-answer"
-- HiCauseToDiversion["480"] = "deflection"
-- HiCauseToDiversion["487"] = "deflection"
-- HiCauseToDiversion["503"] = "unavailable"
-- HiCauseToDiversion["404"] = "unknown"

function convertHIToDiversion(msg)
    local historyInfos = msg:getHeaderValues("History-Info")

    for i, hi in ipairs(historyInfos)
    do
        hi = string.gsub(hi, "%%3B", ";")
        hi = string.gsub(hi, "%%3D", "=")
        hi = string.gsub(hi, "%%22", "\"")
        hi = string.gsub(hi, "%%20", " ")
    end
end

```

```
-- MS format: <sip:+19728522619@med02.lyncabsj.local;user=phone>;index=1;ms-retarget-  
reason=forwarding
```

```
local uri, index, reason = string.match (hi, "<(sip:.*@.*>;index=(.*)reason=(.*)")
```

```
trace.format("hi: uri '%s', reason '%s'", uri or "nil", reason or "nil")
```

```
if uri
```

```
then
```

```
    local diversion = string.format("<%s>", uri)
```

```
    if reason
```

```
    then
```

```
        diversion = string.format("<%s>;reason=\"unconditional\"", uri)
```

```
    end
```

```
    msg:addHeader("Diversion", diversion)
```

```
end
```

```
end
```

```
end
```

```
function convertReferredByToDiversion(msg)
```

```
    local refInfo = msg:getHeader("Referred-By")
```

```
    if refInfo
```

```
    then
```

```
        local diversion = string.format("%s;reason=\"unconditional\"", refInfo)
```

```
        msg:addHeader("Diversion", diversion)
```

```
    end
```

```
end
```

```
local function replaceHistoryHeader(msg)
```

```
    local hist = msg:getHeader("History-Info")
```

```
    if hist
```

```
    then
```

```
        convertHIToDiversion(msg)
```

```
        local di = msg:getHeader("Diversion")
```

```
        if di
```

```
        then
```

```
            msg:removeHeader("History-Info")
```

```
        end
```

```
    end
```

```
end
```

```

local function replaceReferredByHeader(msg)

    local refby = msg:getHeader("Referred-By")
    if refby
    then
        convertReferredByToDiversion(msg)
    end

end

local function modifyUserFrom(msg)
-- get a data from "From" header and replace
    local removeUser= ""
    local value = msg:getHeader("From")

    if value
    then
        value = value:gsub(";user=phone", removeUser)
        if value
        then
            msg:modifyHeader("From", value)
        end
    end
end

local function process_inbound_request(msg)
    modifyUserFrom(msg)
    replaceHistoryHeader(msg)
    replaceReferredByHeader(msg)
    removecryptoline(msg)
    local sdp = msg:getSdp()
    if sdp
    then

        local tcap = sdp:getLine("a=tcap:", "RTP/SAVP")
    if
    tcap
    then
        local a_m_line = sdp:getLine("m=audio", "RTP/AVP")
        a_m_line = a_m_line:gsub("AVP", "SAVP")
        sdp = sdp:modifyLine("m=audio", "RTP/AVP", a_m_line)
        end
        sdp=sdp:removeLine("a=crypto:", "|2^31|")
        msg:setSdp(sdp)
    end
end

```

```

function process_inbound_any_response(msg)
    msg:addHeader("SUPPORTED","X-cisco-srtp-fallback")
    local sdp = msg:getSdp()

    if sdp
    then
        trace.format("Inbound SDP")
        local tcap = sdp:getLine("a=tcap:", "RTP/SAVP")
        if tcap
        then
            local a_m_line = sdp:getLine("m=audio", "RTP/AVP")
            a_m_line = a_m_line:gsub("AVP", "SAVP")
            sdp = sdp:modifyLine("m=audio", "RTP/AVP", a_m_line)
        end
        trace.format("before removing crypto")
        sdp=sdp:removeLine("a=crypto:", "|2^31|")
        trace.format("after removing crypto")
        msg:setSdp(sdp)
    end
end

function removecryptoline(msg)
    local sdp = msg:getSdp()
    if sdp
    then
        trace.format("removecryptoline before removing crypto")
        sdp=sdp:removeLine("a=crypto:", "|2^31|")
        trace.format("removecryptoline after removing crypto")
        msg:setSdp(sdp)
    end
end

function process_inbound_any_request(msg)
    msg:addHeader("SUPPORTED","X-cisco-srtp-fallback")
end

M.outbound_INVITE      = process_outbound_request
M.outbound_ACK         = process_outbound_message
M.outbound_200_INVITE  = process_outbound_message
M.outbound_18X_INVITE  = process_outbound_message

M.inbound_183_INVITE   = process_inbound_progress
M.inbound_INVITE       = process_inbound_request
M.inbound_ANY_ANY      = process_inbound_any_response
M.inbound_ANY          = process_inbound_any_request

```

return M

## SIP Trunk to Skype for Business Configuration

**Navigation:** Device → Trunk

Set **Device Name\*** = **SFB\_2019**. This is used for the test

Set **Device Pool\*** = **G711\_pool**. This is used for the test

Set **Call Classification\*** = **Use System Default**. This is used for the test

Set **Media Resource Group List** = **MRGL\_SW\_NoMTP**. This is used for the test

Uncheck Media Termination Point Required

Check Run On All Active Unified CM Nodes

Check Redirecting Diversion Header Delivery – Inbound

Set **Destination Address** = **172.16.31.99** [FQDN of Skype for Business Mediaation Server] this is used in the test

Set **SIP Trunk Security Profile\*** = **SFB\_SECURITY\_PROFILE**


Set **SIP Profile\*** = **SFB\_SIP\_PROFILE** – Standard SIP Profile

Set **DTMF Signaling Method\*** = **No Preference**

Set **Normalization Script** = **lync\_interop\_updated**

All other values are default.

## Cisco Unified Communications Manager SIP Trunk to Skype for Business Configuration (Continued)

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions


Navigation Cisco Unified CM Administration Go

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**Trunk Configuration** Related Links: Back To Find/List Go

Save Delete Reset Add New

**Status**  
 Status: Ready

**SIP Trunk Status**  
**Service Status:** Full Service  
**Duration:** Time In Full Service: 1 day 12 hours 25 minutes

**Device Information**

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	SFB_2019
Description	SFB_2019
Device Pool*	G711_pool
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	MRGL_SW_No_MTP
Location*	Hub_None
AAR Group	< None >
Tunneled Protocol*	None
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes
Packet Capture Mode*	None
Packet Capture Duration	0
<input type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Path Replacement Support	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Transmit UTF-8 Names in QSIG APDU	
<input type="checkbox"/> Unattended Port	
<input checked="" type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.	
Consider Traffic on This Trunk Secure*	When using both sRTP and TLS
Route Class Signaling Enabled*	Default
Use Trusted Relay Point*	Default
<input type="checkbox"/> PSTN Access	
<input checked="" type="checkbox"/> Run On All Active Unified CM Nodes	

## Cisco Unified Communications Manager SIP Trunk to Skype for Business Configuration (Continued)

**Trunk Configuration**
Related Links: [Back To Find/List](#) [Go](#)

Save
 Delete
 Reset
 Add New

**Intercompany Media Engine (IME)**

E.164 Transformation Profile < None >

**MLPP and Confidential Access Level Information**

MLPP Domain < None >  
 Confidential Access Mode < None >  
 Confidential Access Level < None >

**Call Routing Information**

☒ Remote-Party-Id  
☒ Asserted-Identity  
 Asserted-Type\* Default  
 SIP Privacy\* Default  
 Trust Received Identity\* Trust All (Default)

**Inbound Calls**

Significant Digits\* All  
 Connected Line ID Presentation\* Default  
 Connected Name Presentation\* Default  
 Calling Search Space < None >  
 AAR Calling Search Space < None >  
 Prefix DN   
☒ Redirecting Diversion Header Delivery - Inbound

**Incoming Calling Party Settings**

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings Default Prefix Settings


Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<span>&lt; None &gt;</span>	<input checked="" type="checkbox"/>

**Incoming Called Party Settings**

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<span>&lt; None &gt;</span>	<input checked="" type="checkbox"/>


**Cisco Unified CM Administration**  
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System ▾ | Call Routing ▾ | Media Resources ▾ | Advanced Features ▾ | Device ▾ | Application ▾ | User Management ▾ | Bulk Administration ▾ | Help ▾

**Trunk Configuration**
Related Links: Back To Find/List ▾ Go

Save Delete Reset Add New

**Connected Party Settings**

Connected Party Transformation CSS < None > ▾  
☒ Use Device Pool Connected Party Transformation CSS

**Outbound Calls**

Called Party Transformation CSS < None > ▾  
☒ Use Device Pool Called Party Transformation CSS  
 Calling Party Transformation CSS < None > ▾  
☒ Use Device Pool Calling Party Transformation CSS  
 Calling Party Selection\* Originator ▾  
 Calling Line ID Presentation\* Default ▾  
 Calling Name Presentation\* Default ▾  
 Calling and Connected Party Info Format\* Deliver DN only in connected party ▾  
☐ Redirecting Diversion Header Delivery - Outbound  
 Redirecting Party Transformation CSS < None > ▾  
☒ Use Device Pool Redirecting Party Transformation CSS

**Presentation Information**

☐ Anonymous Presentation  
 Presentation Number   
 Presentation Name   
☐ Send Presentation Name and Number only in the FROM header and not in the other identity headers



## Cisco Unified Communications Manager SIP Trunk to Skype for Business Configuration (Continued)

**Trunk Configuration**Related Links: [Back To Find/List](#) [Go](#)

Save Delete Reset Add New

**SIP Information**

**Destination**

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1*	172.16.31.99		5068

MTP Preferred Originating Codec\*  
BLF Presence Group\*  
SIP Trunk Security Profile\*  
Rerouting Calling Search Space  
Out-Of-Dialog Refer Calling Search Space  
SUBSCRIBE Calling Search Space  
SIP Profile\*  
DTMF Signaling Method\*

711ulaw  
Standard Presence group  
SFB\_SECURITY\_PROFILE  
< None >  
< None >  
< None >  
SFB\_SIP\_PROFILE  
No Preference

[View Details](#)

**Normalization Script**

Normalization Script Lync\_interop\_updated

☐ Enable Trace

	Parameter Name	Parameter Value
1		

**Recording Information**

☒ None  
☐ This trunk connects to a recording-enabled gateway  
☐ This trunk connects to other clusters with recording-enabled gateways

## SIP Trunk to Cisco Unity Connection Configuration

**Navigation:** Device → Trunk

Set **Device Name\*** = **Clus22unity**. This is used for the test.

Set **Device Pool\*** = **Default**

Check Run On All Active Unified CM Nodes

Check Redirecting Diversion Header Delivery – Inbound

Check Redirecting Diversion Header Delivery – Outbound

Set **Destination Address** = **10.80.12.6**. This is used for the test.

Set **SIP Trunk Security Profile\*** = **TLS\_CUC**

Set **SIP Profile\*** = **Standard SIP Profile - OPTIONS**

Set **DTMF Signaling Method \*** = **No Preference**





All other values are default

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and a navigation menu with options like "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", "Bulk Administration", and "Help". The "Device" menu is expanded, showing "SIP Trunk" as the selected option. Below the navigation bar, the "Trunk Configuration" section is visible, with a "Related Links" box containing "Back To Find/List". The main configuration area is divided into two sections: "SIP Trunk Status" and "Device Information". The "SIP Trunk Status" section shows "Service Status: No Service" and "Duration: Time not in Full Service: 1 day 2 hours 19 minutes". The "Device Information" section contains a table of configuration parameters. The "Device Name\*" field is highlighted with a red box and contains the value "Clus22unity". The "Device Pool\*" field is also highlighted with a red box and contains the value "Default". Other fields include "Product: SIP Trunk", "Device Protocol: SIP", "Trunk Service Type: None(Default)", "Description: Clus22unity", "Common Device Configuration: < None >", "Call Classification\*: Use System Default", "Media Resource Group List: < None >", "Location\*: Hub\_None", "AAR Group: < None >", "Tunneled Protocol\*: None", "QSIG Variant\*: No Changes", "ASN.1 ROSE OID Encoding\*: No Changes", "Packet Capture Mode\*: None", and "Packet Capture Duration: 0".

Parameter	Value
Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	Clus22unity
Description	Clus22unity
Device Pool*	Default
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	< None >
Location*	Hub_None
AAR Group	< None >
Tunneled Protocol*	None
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes
Packet Capture Mode*	None
Packet Capture Duration	0

## Cisco Unified Communications Manager SIP Trunk to Cisco Unity Connection Configuration (Continued)

**Trunk Configuration**Related Links: [Back To Find/List](#) [Go](#)

 Save  Delete  Reset  Add New

☐ Media Termination Point Required

☒ Retry Video Call as Audio

☐ Path Replacement Support

☐ Transmit UTF-8 for Calling Party Name

☐ Transmit UTF-8 Names in QSIG APDU

☐ Unattended Port

☒ SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.

Consider Traffic on This Trunk Secure\*

When using both sRTP and TLS

Route Class Signaling Enabled\*

Default

Use Trusted Relay Point\*

Default

☐ PSTN Access

☐ Run On All Active Unified CM Nodes

**Intercompany Media Engine (IME)**

E.164 Transformation Profile [< None >](#)

**MLPP and Confidential Access Level Information**

MLPP Domain [< None >](#)

Confidential Access Mode [< None >](#)

Confidential Access Level [< None >](#)

## Cisco Unified Communications Manager SIP Trunk to Cisco Unity Connection Configuration (Continued)

**Trunk Configuration**
Related Links: [Back To Find/List](#) [Go](#)

Save
 Delete
 Reset
 Add New

**Call Routing Information**

- ☒ Remote-Party-Id
- ☒ Asserted-Identity
- Asserted-Type\*
- SIP Privacy\*

**Inbound Calls**

- Significant Digits\*
- Connected Line ID Presentation\*
- Connected Name Presentation\*
- Calling Search Space
- AAR Calling Search Space
- Prefix DN
- ☒ Redirecting Diversion Header Delivery - Inbound

**Incoming Calling Party Settings**

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value=" &lt; None &gt;"/>	<input checked="" type="checkbox"/>

## Cisco Unified Communications Manager SIP Trunk to Cisco Unity Connection Configuration (Continued)

**Trunk Configuration**
Related Links: [Back To Find/List](#) [Go](#)

Save
 Delete
 Reset
 Add New

**Incoming Called Party Settings**

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

[Clear Prefix Settings](#)
[Default Prefix Settings](#)

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

**Connected Party Settings**

Connected Party Transformation CSS [< None >](#)

☒ Use Device Pool Connected Party Transformation CSS

**Outbound Calls**

Called Party Transformation CSS [< None >](#)

☒ Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS [< None >](#)

☒ Use Device Pool Calling Party Transformation CSS

Calling Party Selection\* [Originator](#)

Calling Line ID Presentation\* [Default](#)

Calling Name Presentation\* [Default](#)

Calling and Connected Party Info Format\* [Deliver DN only in connected party](#)

☒ Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS [< None >](#)

**Trunk Configuration**
Related Links: [Back To Find/List](#) [Go](#)

Save
 Delete
 Reset
 Add New

Redirecting Party Transformation CSS [< None >](#)

☒ Use Device Pool Redirecting Party Transformation CSS

**Presentation Information**

☐ Anonymous Presentation

Presentation Number

Presentation Name

☐ Send Presentation Name and Number only in the FROM header and not in the other identity headers

**SIP Information**

**Destination**

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1 *	10.80.12.6		5061

MTP Preferred Originating Codec\* [711ulaw](#)

BLF Presence Group\* [Standard Presence group](#)

SIP Trunk Security Profile\* [TLS\\_CUC](#)

Rerouting Calling Search Space [< None >](#)

Out-Of-Dialog Refer Calling Search Space [< None >](#)

SUBSCRIBE Calling Search Space [< None >](#)

SIP Profile\* [Standard SIP Profile - OPTIONS](#) [View Details](#)

DTMF Signaling Method\* [No Preference](#)

## Voice Mail Configuration

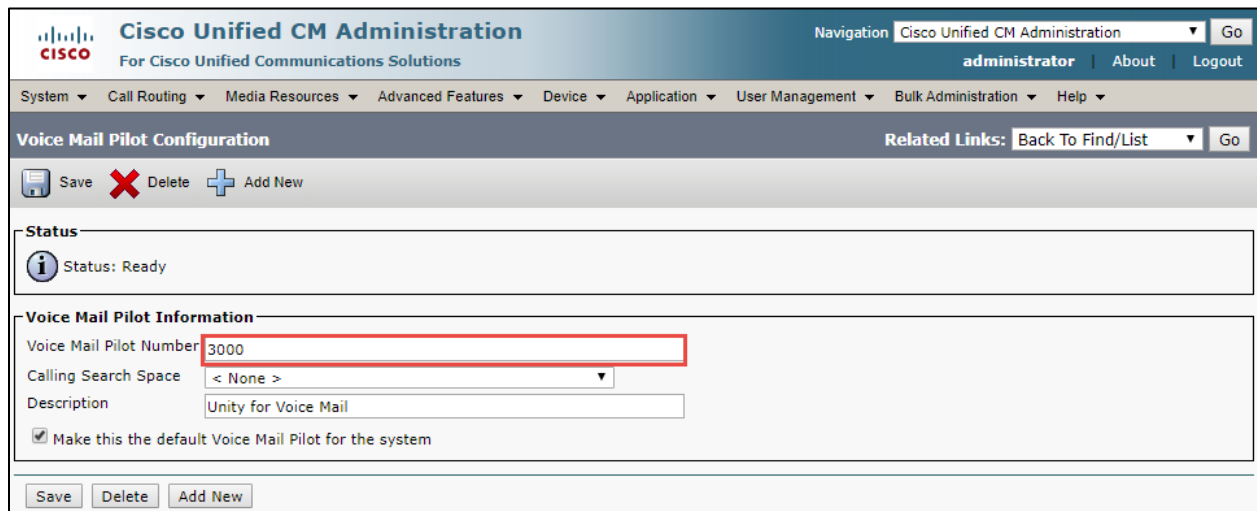
Configure Voice Mail Pilot:

Navigation: Advanced Features → Voice Mail → Voice Mail Pilot

Add new

Set **Voice Mail Pilot Number = 3000** .This is used for the test

Set **Description = Unity for Voice Mail** .This text is used to identify this SIP Profile



The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and a navigation menu. The main content area is titled "Voice Mail Pilot Configuration". Below the title, there are icons for Save, Delete, and Add New. The "Status" section shows "Status: Ready". The "Voice Mail Pilot Information" section contains the following fields:

- Voice Mail Pilot Number: 3000 (highlighted with a red box)
- Calling Search Space: < None >
- Description: Unity for Voice Mail
- ☒ Make this the default Voice Mail Pilot for the system

At the bottom of the form, there are buttons for Save, Delete, and Add New.

## Route Pattern to Skype for Business Extensions

**Navigation:** Call Routing → Route/Hunt → Route Pattern

Add New

Set **Route Pattern\* = 5XXX**. This is used to route to the Skype for Business in this test

Set **Gateway/Route List\* = SFB\_2019**. This is used for the test

Set **Calling Line ID Presentation= Default**


Set **Calling Name Presentation= Default**

Set **Connected Line ID Presentation\*= Default**

Set **Calling Name Presentation\* = Default**

Prefix digit Outgoing Calls (Out Going) = +, Skype user configured + as prefix

All other values are default.


**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

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**Route Pattern Configuration**
Related Links: Back To Find/List Go

Save Delete Copy Add New

**Status**  
*i* Status: Ready

**Pattern Definition**

Route Pattern\* 5XXX

Route Partition < None >

Description SFB 2019

Numbering Plan -- Not Selected --

Route Filter < None >

MLPP Precedence\* Default

☐ Apply Call Blocking Percentage

Resource Priority Namespace Network Domain < None >

Route Class\* Default

Gateway/Route List\* SFB\_2019 [Edit](#)

Route Option  
☒ Route this pattern  
☐ Block this pattern No Error

Call Classification\* OffNet

External Call Control Profile < None >

☐ Allow Device Override ☒ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority





☐ Require Forced Authorization Code

Authorization Level\* 0

☐ Require Client Matter Code

## Route Pattern Configuration for 2xxx (Continued)

**Route Pattern Configuration** Related Links: [Back To Find/List](#) ▼ [Go](#)

 Save  Delete  Copy  Add New

**Calling Party Transformations**

☒ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation\* Default ▼

Calling Name Presentation\* Default ▼

Calling Party Number Type\* Cisco CallManager ▼

Calling Party Numbering Plan\* Cisco CallManager ▼

**Connected Party Transformations**

Connected Line ID Presentation\* Default ▼

Connected Name Presentation\* Default ▼

**Called Party Transformations**

Discard Digits < None > ▼

Called Party Transform Mask

Prefix Digits (Outgoing Calls) +

Called Party Number Type\* Cisco CallManager ▼

Called Party Numbering Plan\* Cisco CallManager ▼

**ISDN Network-Specific Facilities Information Element**

Network Service Protocol -- Not Selected -- ▼

Carrier Identification Code

Network Service	Service Parameter Name	Service Parameter Value
<span>-- Not Selected -- ▼</span>	<span>&lt; Not Exist &gt;</span>	<input type="text"/>

Save Delete Copy Add New

Route Pattern to invoke Jabber client with Remote Destination configured as Skype for Business Extensions

**Navigation:** Call Routing → Route/Hunt → Route Pattern

Set **Route Pattern\*** = \+5XXX. This is used to route to the Skype for Business when using the Extend and Connect functionality in this test

Set **Gateway/Route List\*** = SFB\_2019. This is used for the test

Set **Calling Line ID Presentation**= Default


Set **Calling Name Presentation**= Default

Set **Connected Line ID Presentation\***= Default

Set **Calling Name Presentation\*** = Default

All other values are default








**Cisco Unified CM Administration**  
 For Cisco Unified Communications Solutions


Navigation Cisco Unified CM Administration Go

administrator | [About](#) | [Logout](#)

System ▾ | Call Routing ▾ | Media Resources ▾ | Advanced Features ▾ | Device ▾ | Application ▾ | User Management ▾ | Bulk Administration ▾ | Help ▾

**Route Pattern Configuration**
Related Links: [Back To Find/List](#) Go

 Save
  Delete
  Copy
  Add New





**Status**  
 Status: Ready

**Pattern Definition**

Route Pattern *	<input type="text" value="+5XXX"/>
Route Partition	< None >
Description	SFB 2019 - Jabber
Numbering Plan	-- Not Selected --
Route Filter	< None >
MLPP Precedence *	Default
<input type="checkbox"/> Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class *	Default
Gateway/Route List *	SFB_2019 <a href="#">(Edit)</a>
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern <span>No Error</span>
Call Classification *	OffNet
External Call Control Profile	< None >
<input type="checkbox"/> Allow Device Override <input checked="" type="checkbox"/> Provide Outside Dial Tone <input type="checkbox"/> Allow Overlap Sending <input type="checkbox"/> Urgent Priority	
<input type="checkbox"/> Require Forced Authorization Code	
Authorization Level *	0
<input type="checkbox"/> Require Client Matter Code	

## Cisco Unified Communications Manager Route pattern Configuration (Continued)

**Route Pattern Configuration**Related Links: [Back To Find/List](#) [Go](#)

 Save  Delete  Copy  Add New

**Calling Party Transformations**

☒ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation\*

Calling Name Presentation\*

Calling Party Number Type\*

Calling Party Numbering Plan\*

**Connected Party Transformations**

Connected Line ID Presentation\*

Connected Name Presentation\*

**Called Party Transformations**

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type\*

Called Party Numbering Plan\*

**ISDN Network-Specific Facilities Information Element**

Network Service Protocol

Carrier Identification Code

Network Service	Service Parameter Name	Service Parameter Value
<input type="text" value="-- Not Selected --"/>	<input type="text" value=" &lt; Not Exist &gt;"/>	<input type="text"/>

SaveDeleteCopyAdd New

## Route Pattern to Skype for Business Call Park range

The Skype for Business Call Park range configured is 1000-1999. The following route pattern "45XX" is therefore configured to enable a parked call to be retrieved from Cisco UCM.

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

**Route Pattern Configuration** | Related Links: Back To Find/List | Go

Save | Delete | Copy | Add New

**Status**  
Status: Ready

**Pattern Definition**

Route Pattern\* 45XX

Route Partition < None >

Description SFB 2019 - Call Park

Numbering Plan -- Not Selected --

Route Filter < None >

MLPP Precedence\* Default

☐ Apply Call Blocking Percentage

Resource Priority Namespace Network Domain < None >

Route Class\* Default

Gateway/Route List\* SFB\_2019 (Edit)

Route Option  
☒ Route this pattern  
☐ Block this pattern No Error

Call Classification\* OffNet

External Call Control Profile < None >

☐ Allow Device Override ☒ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority





☐ Require Forced Authorization Code

Authorization Level\* 0

☐ Require Client Matter Code

## Cisco Unified Communications Manager Call Park Range Configuration (Continued)

**Route Pattern Configuration**Related Links: [Back To Find/List](#) [Go](#)

 Save  Delete  Copy  Add New

**Calling Party Transformations**

☒ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation\* Default

Calling Name Presentation\* Default

Calling Party Number Type\* Cisco CallManager

Calling Party Numbering Plan\* Cisco CallManager

**Connected Party Transformations**

Connected Line ID Presentation\* Default

Connected Name Presentation\* Default

**Called Party Transformations**

Discard Digits < None >

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type\* Cisco CallManager

Called Party Numbering Plan\* Cisco CallManager

**ISDN Network-Specific Facilities Information Element**

Network Service Protocol -- Not Selected --

Carrier Identification Code

Network Service	Service Parameter Name	Service Parameter Value
<span>-- Not Selected --</span>	<span>&lt; Not Exist &gt;</span>	<input type="text"/>

Save Delete Copy Add New

## Route Pattern to Unity Connection Voice Mail

A route pattern 3000 (which is the voice mail pilot), is configured to reach Unity Connection Voice Mail.

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and a navigation menu with options like "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", "Bulk Administration", and "Help". The user is logged in as "administrator".

The main section is titled "Route Pattern Configuration". It includes a "Related Links" section with a "Back To Find/List" link. Below this is a toolbar with icons for "Save", "Delete", "Copy", and "Add New".





The "Status" section shows "Status: Ready".

The "Pattern Definition" section contains the following fields:

- Route Pattern\***: 3000
- Route Partition**: < None >
- Description**: (empty text box)
- Numbering Plan**: -- Not Selected --
- Route Filter**: < None >
- MLPP Precedence\***: Default
- Apply Call Blocking Percentage**: (unchecked checkbox)
- Resource Priority Namespace Network Domain**: < None >
- Route Class\***: Default
- Gateway/Route List\***: Clus22unity (with an "Edit" link)
- Route Option**:
  - ☒ Route this pattern
  - ☐ Block this pattern (with a "No Error" dropdown)
- Call Classification\***: OffNet
- External Call Control Profile**: < None >
- Allow Device Override**: (unchecked checkbox)
- Provide Outside Dial Tone**: (checked checkbox)
- Allow Overlap Sending**: (unchecked checkbox)
- Urgent Priority**: (unchecked checkbox)
- Require Forced Authorization Code**: (unchecked checkbox)
- Authorization Level\***: 0
- Require Client Matter Code**: (unchecked checkbox)

## Cisco Unified Communications Manager Route Pattern Configuration (Continued)

**Route Pattern Configuration**Related Links: [Back To Find/List](#) [Go](#)

 Save  Delete  Copy  Add New

**Calling Party Transformations**

☐ Use Calling Party's External Phone Number Mask  
Calling Party Transform Mask   
Prefix Digits (Outgoing Calls)   
Calling Line ID Presentation\* 

Default ▼

  
Calling Name Presentation\* 

Default ▼

  
Calling Party Number Type\* 

Cisco CallManager ▼

  
Calling Party Numbering Plan\* 

Cisco CallManager ▼

**Connected Party Transformations**

Connected Line ID Presentation\* 

Default ▼

  
Connected Name Presentation\* 

Default ▼

**Called Party Transformations**

Discard Digits 

< None > ▼

  
Called Party Transform Mask   
Prefix Digits (Outgoing Calls)   
Called Party Number Type\* 

Cisco CallManager ▼

  
Called Party Numbering Plan\* 

Cisco CallManager ▼

**ISDN Network-Specific Facilities Information Element**

Network Service Protocol 

-- Not Selected -- ▼

  
Carrier Identification Code   

Network Service	Service Parameter Name	Service Parameter Value
<div>-- Not Selected -- ▼</div>	<div>&lt; Not Exist &gt;</div>	<input type="text"/>

Save Delete Copy Add New

## Cisco UCM Extent and Connect

Extend and Connect is a feature that allows administrators to rapidly deploy UC Computer Telephony Integration (CTI) applications, which interoperate with any endpoint. With Extend and Connect, users can leverage the benefits of UC applications from any location using any device. This feature also allows Interoperability between newer UC solutions and legacy systems, so customers can migrate to newer UC Solutions over time as existing hardware is deprecated.

### UC service Configuration

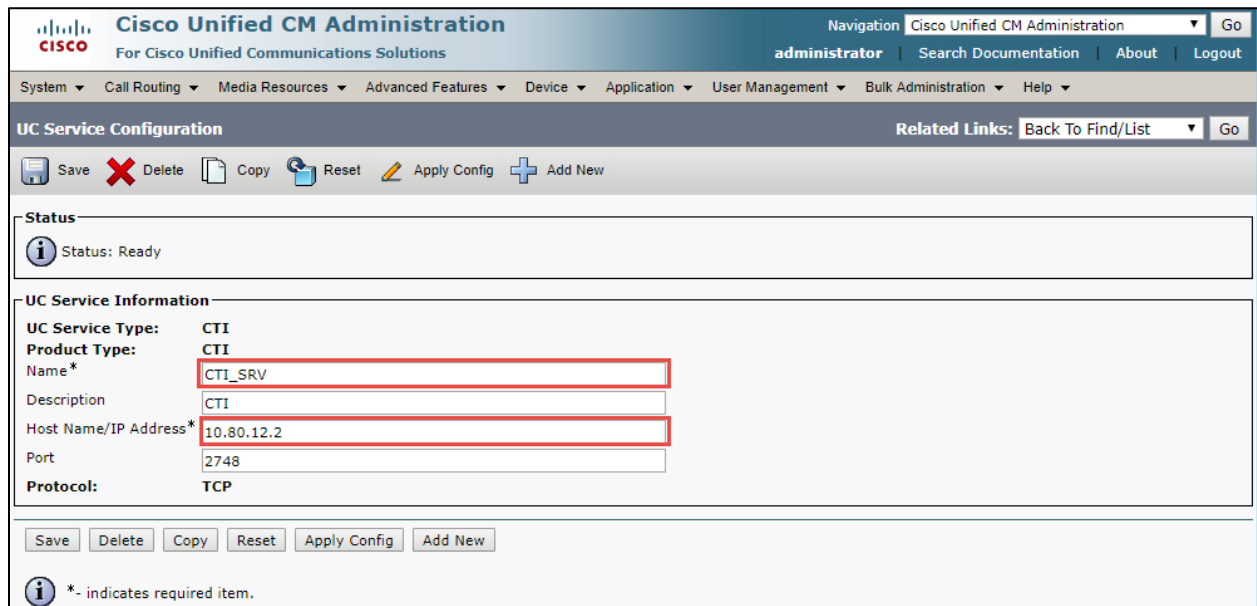
**Navigation Path:** User Management → User setting → UC Service

Add New

Select Service Type as CTI

Set **Name** = **CTI\_SRV**

Set **Host Name/IP Address\*** = **10.80.12.2**; this is the Cisco UCM publisher IP.



**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go  
administrator | Search Documentation | About | Logout

System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help

**UC Service Configuration** Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

**Status**  
Status: Ready

**UC Service Information**

UC Service Type: CTI  
Product Type: CTI  
Name\*: CTI\_SRV  
Description: CTI  
Host Name/IP Address\*: 10.80.12.2  
Port: 2748  
Protocol: TCP

Save Delete Copy Reset Apply Config Add New


\*- indicates required item.

In the same manner, a UC Service is configured for the subscriber also.

A UC service of Type IM and Presence is configured with the IP of the Presence server.

## Service Profile Configuration





**Navigation:** User Management → User setting → Service Profile


**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation **Cisco Unified CM Administration**   
**administrator** | [Search Documentation](#) | [About](#) | [Logout](#)

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

**Service Profile Configuration** Related Links: [Back To Find/List](#)

 Save  Delete  Copy  Add New

**Status**  
 Status: Ready

**Service Profile Information**  
Name\*   
Description   
☐ Make this the default service profile for the system





**Voicemail Profile**  
Primary   
Secondary   
Tertiary   
[Credentials source for voicemail service](#)\*

**MailStore Profile**  
Primary   
Secondary   
Tertiary   
[Inbox Folder](#)\*   
[Trash Folder](#)\*   
[Polling Interval \(in seconds\)](#)\*   
☒ [Allow dual folder mode](#)



## Cisco UCM service profile Configuration (Continued)

**Service Profile Configuration**Related Links: [Back To Find/List](#) [Go](#)

 Save  Delete  Copy  Add New

**Conferencing Profile**

Primary <None>

Secondary <None>

Tertiary <None>

Server Certificate Verification Any

[Credentials source for web conference service](#)\* Not set

**Directory Profile**

Primary <None>

Secondary <None>

Tertiary <None>

☒ [Use UDS for Contact Resolution](#)

☒ [Use Logged On User Credential](#)

[Username](#)

[Password](#)

[Search Base 1](#)

[Search Base 2](#)

[Search Base 3](#)

☒ [Recursive Search on All Search Bases](#)

[Search Timeout \(seconds\)\\*](#)

[Base Filter \(Only used for Advance Directory\)](#)

[Predictive Search Filter \(Only used for Advance Directory\)](#)

## Cisco UCM service profile Configuration (Continued)

<a href="#">Search Timeout (seconds)*</a>	5
<a href="#">Base Filter (Only used for Advance Directory)</a>	
<a href="#">Predictive Search Filter (Only used for Advance Directory)</a>	
<input type="checkbox"/> <a href="#">Allow Jabber to Search and Add Security Groups</a>	

---

**IM and Presence Profile**

Primary	IMP_IR ▼
Secondary	<None> ▼
Tertiary	<None> ▼

---

**CTI Profile**


Primary	CTI_SRV ▼
Secondary	<None> ▼
Tertiary	<None> ▼

---

**Video Conference Scheduling Portal Profile**

Primary	<None> ▼
Secondary	<None> ▼
Tertiary	<None> ▼

---

 \*- indicates required item.

## Cisco Unified CM IM Presence – CCMCIP Profile Configuration

**Navigation Path:** Application → CCMCIP Profile

Set Name \*: remotedesk. This is used in this example.

Set **Primary CCMCIP Host** \*: 10.80.12.2.Cisco Publisher IP. This is used in this test.

Set **Backup CCMCIP Host** \*: 10.80.12.3.Cisco Publisher IP. This is used in this test.

Add Users to Profile: jabber3.This is used in this test.

**Cisco Unified CM IM and Presence Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM IM and Presence Administration | Go

1 | administrator | Search | Logout | About

System | Presence | Messaging | Application | Bulk Administration | Diagnostics | Help

**CCMCIP Profile Configuration** Related Links: Back To Find/List | Go

Save Delete Add New

**Status**  
Status: Ready

**CCMCIP Profile Settings**

Name\* remotedesk

Description

Primary CCMCIP Host\* 10.80.12.2

Backup CCMCIP Host\* 10.80.12.3

Server Certificate Verification\* Any Certificate

☐ Make this the default CCMCIP Profile for the system.

**Users in Profile**

	User ID	Firstname	Lastname	Department
<input type="checkbox"/>	jabber3		jabber3	

Add Users to Profile Select All Clear All Delete Selected Rows per Page 50

Save Delete Add New

## SIP trunk to Cisco IM&Presence Trunk Configuration

**Navigation Path:** Device → Trunk

Set **Device Name\***=IMPTTrunk. This is used for the test.

Set **Device Pool\*** = Default. This is used for the test.

Set **Media Resource Group List** = MRGL\_SW\_NoMTP. This is used for the test.


Set **Destination Address** = 10.80.12.5. This is used in this example.

Set **SIP Trunk Security Profile\***= Non-Secure SIP Trunk Profile.

Set **SIP Profile\***= Standard SIP Profile.

Set **DTMF Signaling Method\***= No Preference.

All other values are default.


**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go  
**administrator** | [Search Documentation](#) | [About](#) | [Logout](#)

System ▾ | Call Routing ▾ | Media Resources ▾ | Advanced Features ▾ | Device ▾ | Application ▾ | User Management ▾ | Bulk Administration ▾  
Help ▾

**Trunk Configuration**
Related Links: Back To Find/List Go

Save Delete Reset Add New


**Status**  
*i* Status: Ready

**SIP Trunk Status**  
**Service Status:** Unknown  
**Duration:** Unknown

**Device Information**

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	IMPTrunk
Description	
Device Pool*	Default
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	MRGL_SW_NoMTP
Location*	Hub_None
AAR Group	< None >
Tunneled Protocol*	None
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes
Packet Capture Mode*	None
Packet Capture Duration	0





## Cisco Unified Communications Manager SIP Trunk to CUP Configuration (Continued)

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go  
**administrator** | [Search Documentation](#) | [About](#) | [Logout](#)

System ▾ | Call Routing ▾ | Media Resources ▾ | Advanced Features ▾ | Device ▾ | Application ▾ | User Management ▾ | Bulk Administration ▾ | Help ▾

**Trunk Configuration** Related Links: Back To Find/List Go

 Save  Delete  Reset  Add New

☐ Media Termination Point Required

☒ Retry Video Call as Audio

☐ Path Replacement Support

☐ Transmit UTF-8 for Calling Party Name

☐ Transmit UTF-8 Names in QSIG APDU

☐ Unattended Port

☐ SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.

Consider Traffic on This Trunk Secure\* When using both sRTP and TLS ▾

Route Class Signaling Enabled\* Default ▾

Use Trusted Relay Point\* Default ▾

☐ PSTN Access

☐ Run On All Active Unified CM Nodes

**Intercompany Media Engine (IME)**

E.164 Transformation Profile < None > ▾

**MLPP and Confidential Access Level Information**

MLPP Domain < None > ▾

Confidential Access Mode < None > ▾





Confidential Access Level < None > ▾

**Call Routing Information**

☒ Remote-Party-Id

## Cisco UCM SIP Trunk to CUP Configuration (Continued)

**Trunk Configuration**Related Links: [Back To Find/List](#) [Go](#)

 Save  Delete  Reset  Add New

**Incoming Called Party Settings**

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

[Clear Prefix Settings](#) [Default Prefix Settings](#)

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input data-bbox="808 625 1263 651" type="text" value=" &lt; None &gt; "/>	<input checked="" type="checkbox"/>

**Connected Party Settings**

Connected Party Transformation CSS

☒ Use Device Pool Connected Party Transformation CSS

**Outbound Calls**

Called Party Transformation CSS

☒ Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS

☒ Use Device Pool Calling Party Transformation CSS

Calling Party Selection\*

Calling Line ID Presentation\*

Calling Name Presentation\*

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

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Cisco Unified CM Administration
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**Trunk Configuration**
Related Links:
Back To Find/List
Go

Save
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 Reset
 Add New

☐ Redirecting Diversion Header Delivery - Outbound  
Redirecting Party Transformation CSS
< None >
☒ Use Device Pool Redirecting Party Transformation CSS

**Presentation Information**

☐ Anonymous Presentation  
Presentation Number  
Presentation Name  
☐ Send Presentation Name and Number only in the FROM header and not in the other identity headers

**SIP Information**

**Destination**
☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1 *	10.80.12.5		5060

MTP Preferred Originating Codec\*
711ulaw
BLF Presence Group\*
Standard Presence group
SIP Trunk Security Profile\*
Non Secure SIP Trunk Profile
Rerouting Calling Search Space
< None >
Out-Of-Dialog Refer Calling Search Space
< None >
SUBSCRIBE Calling Search Space
< None >
SIP Profile\*
Standard SIP Profile
DTMF Signaling Method\*
No Preference

**Normalization Script**

Normalization Script
< None >
☐ Enable Trace

	Parameter Name	Parameter Value
1		

**Recording Information**

☒ None
☐ This trunk connects to a recording-enabled gateway
☐ This trunk connects to other clusters with recording-enabled gateways

**Geolocation Configuration**

Geolocation
< None >
Geolocation Filter
< None >
☐ Send Geolocation Information

Save
Delete
Reset
Add New

*\* - indicates required item.*  
*\*\* - Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.*

## End user configuration

Add user to Cisco UCM

**Navigation:** User Management → End user

Set **User ID\*** = **jabber3**. This is used for the test.

Set **Last Name** = **jabber3**. This is used for the test.

Check Home Cluster.

Click the Device Association

Select CT11 from User Device Association screen




The screenshot shows the 'End User Configuration' page in the Cisco Unified CM Administration interface. The page has a navigation bar at the top with 'Cisco Unified CM Administration' and 'For Cisco Unified Communications Solutions'. Below the navigation bar, there are tabs for 'System', 'Call Routing', 'Media Resources', 'Advanced Features', 'Device', 'Application', 'User Management', 'Bulk Administration', and 'Help'. The 'User Management' tab is selected. The page title is 'End User Configuration'. There are buttons for 'Save', 'Delete', and 'Add New'. A 'Status' section shows 'Status: Ready'. The 'User Information' section contains fields for 'User Status' (Enabled Local User), 'User ID\*' (jabber3), 'Password', 'Confirm Password', 'Self-Service User ID' (7030), 'PIN', 'Confirm PIN', 'Last name\*' (jabber3), 'Middle name', 'First name', 'Display name', 'Title', 'Directory URI' (jabber3@tekvisionlabs.com), 'Telephone Number', 'Home Number', and 'Mobile Number'. There are 'Edit Credential' buttons next to the 'Password' and 'PIN' fields. The fields for 'User ID\*', 'Self-Service User ID', and 'Last name\*' are highlighted with red boxes.

User Information	
User Status	Enabled Local User
User ID*	jabber3
Password	.....
Confirm Password	.....
Self-Service User ID	7030
PIN	.....
Confirm PIN	.....
Last name*	jabber3
Middle name	
First name	
Display name	
Title	
Directory URI	jabber3@tekvisionlabs.com
Telephone Number	
Home Number	
Mobile Number	



## Cisco UCM end user Configuration (Continued)

**End User Configuration**Related Links: [Back to Find List Users](#) [Go](#)

 Save  Delete  Add New

Pager Number

Mail ID

Manager User ID

Department

User Locale

Associated PC/Site Code

Digest Credentials

Confirm Digest Credentials

User Profile

User Rank\*

< None >

Use System Default( "Standard (Factory Default) Us

1-Default User Rank

[View Details](#)

**Service Settings**

☒ Home Cluster

☒ Enable User for Unified CM IM and Presence (Configure IM and Presence in the associated UC Service Profile)

☒ Include meeting information in presence(Requires Exchange Presence Gateway to be configured on CUCM IM and Presence server)

[Presence Viewer for User](#)

UC Service Profile

Jabber\_SVC\_Profile

[View Details](#)

## Cisco UCM end user Configuration (Continued)

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

**End User Configuration** | Related Links: Back to Find List Users | Go

Save | Delete | Add New

**Device Information**

Controlled Devices: CTIRDjabber3

Available Profiles:

CTI Controlled Device Profiles:

**Extension Mobility**

Available Profiles:

Controlled Profiles:

Default Profile: -- Not Selected --

BLF Presence Group\*: Standard Presence group

SUBSCRIBE Calling Search Space: < None >

Maximum Login Time (HHH:MM):

☒ Allow Control of Device from CTI

☐ Enable Extension Mobility Cross Cluster

## Cisco UCM end user Configuration (Continued)

Check Allow Control of Device from CTI

Select the Primary Extension for this user. 7030 is used for this example.

Check Enable Mobility.

Add the following permissions for Standard Users:

- Standard CCM End-Users
- Standard CTI Enabled
- Standard CTI Allow Control of All Devices
- Standard CCMUSER Administration

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go

administrator | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

**End User Configuration** Related Links: [Back to Find List Users](#) Go

Save Delete Add New

**Directory Number Associations**

Primary Extension 7030

**Mobility Information**

☒ **Enable Mobility**

☒ Enable Mobile Voice Access

Maximum Wait Time for Desk Pickup\* 10000

Remote Destination Limit\* 4

Remote Destination Profiles [View Details](#)

**Multilevel Precedence and Preemption Authorization**

MLPP User Identification Number

MLPP Password

Confirm MLPP Password

MLPP Precedence Authorization Level Default

**CAPF Information**

Associated CAPF Profiles [View Details](#)

**Permissions Information**

Groups Standard CCM End Users  
Standard CTI Allow Control of All Devices  
Standard CTI Enabled [View Details](#)

Roles Standard CCM End Users  
Standard CCMUSER Administration  
Standard CTI Allow Control of All Devices  
Standard CTI Enabled [View Details](#)

**Conference Now Information**

☐ Enable End User to Host Conference Now

Meeting Number 7030

Attendees Access Code

Save Delete Add New

## Remote Destination Configuration

**Navigation:** Device→Remote Destination

Add New

Set **name** = **Jabber RD** .This is used for the test

Set **Destination Number\***= **+5002**. This is used for the test. [7030 is a Skype for Business extension]

Check Enable Extend and Connect.

## Set CTI Remote Device = CTIRD1

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

### Remote Destination Configuration

Related Links: Back To Find/List | Go

Save | Delete | Copy | Add New

**Status**  
Status: Ready

**CTI Remote Device**

Line	Line Association
Line [1] - 7030 (no partition)	<input checked="" type="checkbox"/>

**Remote Destination Information**

Name: JabberRD

Destination\*: +5002

Mobility User ID\*: jabber3

☐ Enable Unified Mobility features

Remote Destination Profile\*: -- Not Selected --

Single Number Reach Voicemail Policy\*: Use System Default

☒ Enable Single Number Reach  
Ring this phone and my business phone at the same time when my business line(s) is dialed.

☒ Enable Move to Mobile  
If this is a mobile phone, transfer active calls to this phone when the mobility button on your Cisco IP Ph

☒ Enable Extend and Connect  
Allow this phone to be controlled by CTI applications (e.g. Jabber)

CTI Remote Device\*: CTIRDjabber3

**Timer Information**

Wait\* 4.0 seconds before ringing this phone when my business line is dialed.\*

Prevent this call from going straight to this phone's voicemail by using a time delay of\* 1.5 seconds to det

straight to voicemail.\*

Stop ringing this phone after\* 19.0 seconds to avoid connecting to this phone's voicemail.\*

The CTI Remote Device configuration is updated with the remote destination:

Remote Destination Configuration
Related Links: Back To Find/List Go

Save Delete Copy Add New

When Single Number Reach is Enabled

Ring Schedule

☒ All the time  
☐ As specified below

☐ Monday ☐ No Office Hours to ☐ No Office Hours  
All Day  
☐ Tuesday ☐ No Office Hours to ☐ No Office Hours  
All Day  
☐ Wednesday ☐ No Office Hours to ☐ No Office Hours  
All Day  
☐ Thursday ☐ No Office Hours to ☐ No Office Hours  
All Day  
☐ Friday ☐ No Office Hours to ☐ No Office Hours  
All Day  
☐ Saturday ☐ No Office Hours to ☐ No Office Hours  
All Day  
☐ Sunday ☐ No Office Hours to ☐ No Office Hours  
All Day  
Time Zone\* (GMT) Etc/GMT

When receiving a call during the above ring schedule

☒ Always ring this destination  
☐ Ring this destination only if caller is in -- Not Selected -- View Details  
☐ Do not ring this destination if caller is in -- Not Selected -- View Details

Two Remote Destinations were configured for this test:

Cisco Unified CM Administration
Navigation Cisco Unified CM Administration Go
administrator About Logout

System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help

Find and List Remote Destinations

Add New Select All Clear All Delete Selected

Status
2 records found

Remote Destination (1 - 2 of 2) Rows per Page 50

Find Remote Destination where Destination begins with +5 Find Clear Filter

	Name ^	Destination	Remote Destination Profile	Dual Mode Phone	IMS-Integrated Mobile	CTI Remote Device/Cisco Spark Remote Device	Copy
<input type="checkbox"/>	JabberRD	+5002				CTIRDjabber3	
<input type="checkbox"/>	Jabber RD2	+5006				CTIRDjabber4	

Add New Select All Clear All Delete Selected

## Cisco UCM CTI Remote Device Configuration

**Navigation:** Device → Phone

Add New.

Select **Phone Type \*** = **CTI Remote Device**

The CTI Remote Device type represents the user's remote device(s).

Select the desired Owner User ID. jabber3 is used in this test.

Set Device Pool: Default

Save.

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and a navigation menu with options like System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The "Device" menu is selected, leading to the "Phone Configuration" page. The page header shows "Related Links: Back To Find/List" and a "Go" button. Below the header, there are icons for Save, Delete, Copy, Reset, Apply Config, and Add New. The main content area is divided into several sections: "Status" (Ready), "Association" (two lines: Line [1] - 7030 (no partition) and Line [2] - Add a new DN), "Phone Type" (Product Type: CTI Remote Device), "Real-time Device Status" (Registration: Registered with Cisco Unified Communications Manager clus22pub, IPv4 Address:), and "Device Information". The "Device Information" section contains a list of fields: Device is Active (checked), Device is not trusted (unchecked), Active Remote Destination (+5002), Owner User ID\* (jabber3), Device Name\* (CTIRDjabber3), Description, Device Pool\* (Default), Calling Search Space (< None >), User Hold MOH Audio Source (1-SampleAudioSource), Network Hold MOH Audio Source (1-SampleAudioSource), Location\* (Hub\_None), User Locale (< None >), Network Locale (< None >), Mobility User ID (jabber3), and a checkbox for Ignore Presentation Indicators (internal calls only). A red box highlights the "Owner User ID\*" and "Device Name\*" fields.

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

**Phone Configuration** | Related Links: Back To Find/List | Go

Save | Delete | Copy | Reset | Apply Config | Add New

**Status**  
Status: Ready

**Association**

1	Line [1] - 7030 (no partition)
2	Line [2] - Add a new DN

**Phone Type**  
Product Type: CTI Remote Device

**Real-time Device Status**  
Registration: Registered with Cisco Unified Communications Manager clus22pub  
IPv4 Address:

**Device Information**

<input checked="" type="checkbox"/> Device is Active	
<input type="checkbox"/> Device is not trusted	
Active Remote Destination	+5002
Owner User ID*	jabber3
Device Name*	CTIRDjabber3
Description	
Device Pool*	Default <a href="#">View Details</a>
Calling Search Space	< None >
User Hold MOH Audio Source	1-SampleAudioSource
Network Hold MOH Audio Source	1-SampleAudioSource
Location*	Hub_None
User Locale	< None >
Network Locale	< None >
Mobility User ID	jabber3
<input type="checkbox"/> Ignore Presentation Indicators (internal calls only)	

## Cisco UCM CTI Remote Device Configuration (Continued)

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and the subtitle "For Cisco Unified Communications Solutions". The user is logged in as "administrator". The main menu shows various categories: System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The "Phone Configuration" page is active, with a "Related Links" section containing a "Back To Find/List" link. The page features a toolbar with icons for Save, Delete, Copy, Reset, Apply Config, and Add New. The configuration area is divided into several sections: "Number Presentation Transformation" (containing "Caller ID For Calls From This Phone" and "Remote Number" sub-sections), "Protocol Specific Information" (containing BLF Presence Group, SUBSCRIBE Calling Search Space, and Rerouting Calling Search Space), "Associated Remote Destinations" (containing a checkbox for routing calls and a table of destinations), and "Do Not Disturb" (containing a checkbox for Do Not Disturb and a DND Option dropdown). The "Associated Remote Destinations" table shows a single entry with the name "JabberRD" and the destination "+5002".

**Number Presentation Transformation**

**Caller ID For Calls From This Phone**

Calling Party Transformation CSS:

☒ Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)

**Remote Number**

Calling Party Transformation CSS:

☒ Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)

**Protocol Specific Information**

BLF Presence Group\*:

SUBSCRIBE Calling Search Space:

Rerouting Calling Search Space:

**Associated Remote Destinations**

☐ Route calls to all remote destinations when client is not connected

Name	Destination
JabberRD	+5002

[Add a New Remote Destination](#)

**Do Not Disturb**

☐ Do Not Disturb

DND Option\*:

Add a DN to this device.

DN +5002 was configured for this test.

# Cisco Unity Connection

## Telephony Integration – Add Phone System

**Navigation:** Telephony Integrations → Phone system

Add New

Set Phone **System Name\*** = **SFB\_CUCM**. This Name used for this test

Check Use Same Port for Enabling and Disabling MWIs

The screenshot displays the Cisco Unity Connection Administration web interface. The left sidebar shows the navigation tree with 'Phone System' selected under 'Telephony Integrations'. The main content area is titled 'Phone System Basics (SFB\_CUCM)'. It includes a 'Phone System' section with a text field for 'Phone System Name\*' containing 'SFB\_CUCM'. Below this is the 'Message Waiting Indicators' section, where the checkbox 'Use Same Port for Enabling and Disabling MWIs' is checked and highlighted with a red box. Other options in this section include 'Send Message Counts', 'Force All MWIs Off for this Phone System', and a 'Run' button for synchronizing MWIs. The 'Call Loop Detection by Using DTMF' section has checkboxes for supervised transfers and forwarded message notification calls, with a DTMF tone dropdown set to 'A' and a guard time of 2500 milliseconds. The 'Call Loop Detection by Using Extension' section has a checked checkbox for forwarded message notification calls. The 'Phone View Settings' section includes checkboxes for enabling phone view and fields for CTI phone access username and password. The 'Outgoing Call Restrictions' section has radio buttons for enabling outgoing calls, disabling all outgoing calls immediately, or disabling all outgoing calls between specific times, with dropdowns for beginning and ending times.

## Telephony Integration – Add Port Group

**Navigation:** Telephony Integration → Port Group or from previous Screen, Related Links “Add Port Group”  
Go

Set **Phone System** = **SFB\_CUCM**

Set **Create From** – **Port group**

Set **Type** = **SIP**

Set Display Name\* = **SFB\_CUCM-1**. This Name used for this example.



Set Ipv4 Address or Host Name = 10.80.12.2 [This is the Cisco UCM publisher IP]

Check Register with SIP server

Set **Security Profile = 5061/TLS**

Set **Security Mode = Encrypted**

Check Secure RTP

Check Enable Message Waiting Indicators

**Cisco Unity Connection Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unity Connection Administration ▼ Go  
administrator | Search Documentation | About | Sign Out

**Port Group Basics (SFB\_CUCM-1)**

Port Group Edit Refresh Help  
Save Delete Previous Next

**Port Group**  
Display Name\* SFB\_CUCM-1  
Integration Method SIP  
Reset Status Reset Not Required Reset

**Session Initiation Protocol (SIP) Settings**  
☒ Register with SIP Server  
☐ Authenticate with SIP Server  
Authentication Username  
Authentication Password  
Contact Line Name  
SIP Security Profile 5061/TLS ▼  
☒ Enable Next Generation Encryption  
☒ Secure RTP

**Advertised Codec Settings**  
Change Advertising  
Display Name Packet Size  
G.711 mu-law 20 ▼  
Change Advertising

**Message Waiting Indicator Settings**  
☒ Enable Message Waiting Indicators  
Delay between Requests 0 milliseconds  
Maximum Concurrent Requests 0  
Retries After Successful Attempt 5 milliseconds  
Retry Interval After Successful Attempt 5 milliseconds  
Save Delete Previous Next  
Fields marked with an asterisk (\*) are required.

Click Save.

## Telephony Integration – Add Ports

Set **Number of Ports** = 4

Set **Phone System** = SFB\_CUCM-1-001

Set **Port Group** = SFB\_CUCM-1

Set **Server** = clus22cuc.lab.tekvizion.com (which is the FQDN of Cisco Unity Server)

The screenshot displays the Cisco Unity Connection Administration web interface. The left sidebar shows a navigation tree with 'Telephony Integrations' expanded, and 'Port' selected. The main content area is titled 'Port Basics (SFB\_CUCM-1-001)'. It includes a 'Phone System Port' section with the following fields: 'Enabled' (checked), 'Port Name' (SFB\_CUCM-1-001), 'Phone System' (SFB\_CUCM), 'Port Group' (SFB\_CUCM-1), and 'Server' (clus22cuc.lab.tekvizionlabs.com). A 'Restart' button is next to the Port Name field. Below this is the 'Port Behavior' section with checkboxes for 'Answer Calls', 'Perform Message Notification', 'Send MWI Requests (may also be disabled by the port group)', and 'Allow TRAP Connections', all of which are checked. Navigation buttons (Save, Delete, Previous, Next) are present at the top and bottom of the configuration area.

## SIP Certificate

**Navigation:** Telephony Integrations → Security → SIP Certificate

Add New

Set **Display Name\*** = **clus22cuc.tekvizionlabs.com**. This Name used for this test

Set **Subject Name\*** = **clus22cuc.tekvizionlabs.com**. Subject Name must match the X.509 Subject Name in SIP trunk security profile in Cisco UCM

The screenshot displays the Cisco Unity Connection Administration web interface. The left sidebar shows the navigation tree with 'Security' > 'SIP Certificate' selected. The main content area is titled 'Edit SIP Certificate (clus22cuc.tekvizionlabs.com)'. It contains a form with the following fields:

- Display Name\***: clus22cuc.tekvizionlabs.com
- Subject Name\***: clus22cuc.tekvizionlabs.com
- Subject**: CN=clus22cuc.tekvizionlabs.com, OU=ECSBU, O=Cisco Systems Inc.
- Issuer**: CN=CiscoUnity-057b0739-0df9-4f44-9024-e9b2dfe56652
- Valid From**: Mon Sep 16 23:55:14 CDT 2019
- Valid Until**: Mon Sep 14 02:44:15 CDT 2026
- Version**: 3
- Subject Alternative Name**: [6, clus22cuc.tekvizionlabs.com]
- Serial Number**: 467a4324e3d24f5aa2c78f7c7f27a02d
- Certificate Text**: A large text area containing the PEM-formatted certificate data, starting with '-----BEGIN CERTIFICATE-----' and ending with '-----END CERTIFICATE-----'.

At the bottom of the form, there is a 'Generate New' button and a set of navigation buttons: 'Save', 'Delete', 'Previous', and 'Next'.

## User Configuration

**Navigation:** Cisco Unity Connection → Users → Users

Set **Alias\*** = **+5001** (This is used for the test)

Set **First Name** = **Test** (This is used to identify the User)

Set **Extension\*** = **+5001** (This is user's extension number)

Set **Partition** = **clus22cucpartition**

Set **Phone System** = **SFB\_CUCM**

**Cisco Unity Connection Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unity Connection Administration | administrator | Search Documentation | About | Sign Out

**Cisco Unity Connection**

- Users
  - Users
  - Import Users
  - Synch Users
- Class of Service
  - Class of Service
  - Class of Service Membership
- Templates
  - User Templates
  - Call Handler Templates
  - Contact Templates
  - Notification Templates
- Contacts
  - Contacts
- Distribution Lists
  - System Distribution Lists
- Call Management
  - System Call Handlers
  - Directory Handlers
  - Interview Handlers
  - Custom Recordings
- Call Routing
- Message Storage
  - Mailbox Stores
  - Mailbox Stores Membership
  - Mailbox Quotas
  - Message Aging
- Networking

**Edit User Basics (+5001)**

Search Users | Edit User Basics (+5001) | Related Links | Bulk Edit By CSV | Go

User Edit Refresh Help

Save Delete Previous Next

**Name**

Alias\* +5001

First Name Test

Last Name

Display Name +5001

SMTP Address +5001 @clus22cuc.lab.tekvizion.com

Initials

Title

Employee ID

**LDAP Integration Status**

☐ Integrate with LDAP Directory

☒ Do Not Integrate with LDAP Directory

**Phone**

Extension\* +5001

Cross-Server Transfer Extension or URI +5001

Outgoing Fax Number

Outgoing Fax Server --- Not Selected ---

# Cisco Unity Connection Administration

For Cisco Unified Communications Solutions

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administrator

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Cisco Unity Connection

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Users

Import Users

Synch Users

Class of Service

Class of Service

Class of Service Membership

Templates

User Templates

Call Handler Templates

Contact Templates

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Message Storage

Mailbox Stores

Mailbox Stores Membership

Mailbox Quotas

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Partition

clus22cuc Partition

Search Scope

clus22cuc Search Space

Phone System

SFB\_CUCM

Class of Service

Voice Mail User COS

Active Schedule

Weekdays

View

☐ Set for Self-enrollment at Next Sign-In

☒ List in Directory

☒ Send Non-Delivery Receipts on Failed Message Delivery

☐ Skip PIN When Calling From a Known Extension

☐ Use Short Calendar Caching Poll Interval

Recorded Name

↑

↓

+5001

Number or URI

▶

■

●

30

Volume

◀

1x

▶

Speed

Location

Address

Building

City

State

Postal Code

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## Acronyms

Acronym	Definition
CCNR	Call Completion on No Reply
CFB	Call Forwarding on Busy
CFNA	Call Forwarding No Answer
CFU	Call Forwarding Unconditional
Cisco UCM	Cisco Unified Communications Manager
CLIP	Calling Line (Number) Identification Presentation
CLIR	Calling Line (Number) Identification Restriction
CNIP	Calling Name Identification Presentation
CNIR	Calling Name Identification Restriction
COLP	Connected Line (Number) Identification Presentation
COLR	Connected Line (Number) Identification Restriction
CONP	Connected Name Identification Presentation
CONR	Connected Name Identification Restriction
CT	Call Transfer
CUP	Cisco Unified Presence
DNS	Domain Name Server
EXT	Extension
FQDN	Fully Qualified Domain Name
MRGL	Media Resource Group List
MTP	Media Termination Point
MWI	Message Waiting Indicator
PBX	Private Branch Exchange
PSTN	Public Switched Telephone Network
RTP	Real Time Protocol
SCCP	Skinny Client Control Protocol
SFB	Skype for Business
SIP	Session Initiated Protocol
UDP	Uniform Dial Plan
VM	Voice Mail