

# **Skype for Business 2015 using SIP trunk to Cisco Unified Communications Manager Release 10.5.2 SU3**

## Table of Contents

<b>Introduction</b> .....	4
The following items were tested: .....	4
Listed below are the highlights of the integration issues: .....	5
Below are the key results:.....	5
<b>Network Topology</b> .....	5
<b>Limitations</b> .....	6
<b>System Components</b> .....	6
Hardware Requirements.....	6
Software Requirements .....	7
<b>Features</b> .....	7
Features Supported .....	7
Features Not Supported or Not Tested .....	8
<b>Configuration</b> .....	8
Configuring Sequence and Tasks: .....	8
Configuring the Skype for Business.....	9
Add Cisco UCM to Skype for Business Topology.....	9
Skype for Business Trunk Configuration .....	13
Skype for Business Route Configuration.....	19
Skype for Business Voice Policy and PSTN Usage Configuration .....	21
Skype for Business Dial Plan Configuration .....	22
Skype for Business Call Park Range Configuration.....	28
Skype for Business Global Media Bypass Configuration .....	28
Skype for Business User Configuration .....	29
Skype for Business Client Configuration .....	35
<b>Configuring the Cisco Unified Communications Manager</b> .....	37
Cisco Unified Communications Manager SIP Trunk Security Profile for Trunk to Skype for Business ...	37
Cisco Unified Communications Manager SIP Trunk Security Profile for Trunk to Unity Connection .....	39

Cisco Unified Communications Manager SIP Profile .....	40
Cisco Unified Communications Manager Media Resource Group.....	44
Cisco Unified Communications Manager Media Resource Group List .....	47
Cisco Unified Communications Manager Device Pool Configuration .....	49
Cisco Unified Communications Manager Region Configuration.....	51
Cisco Unified Communications Manager Normalization Script.....	52
Cisco Unified Communications Manager SIP Trunk to Skype for Business Configuration .....	60
Cisco Unified Communications Manager SIP Trunk to Cisco Unity Connection Configuration .....	66
Cisco Unified Communications Manager Route Group .....	71
Cisco Unified Communications Manager Route List.....	72
Cisco Unified Communications Manager SIP Route Pattern .....	73
Cisco Unified Communications Manager Voice Mail Configuration.....	75
Cisco Unified Communications Manager Route Pattern to Skype for Business Extensions.....	75
Cisco Unified Communications Manager Route Pattern to invoke Jabber client with Remote Destination configured as Skype for Business Extensions .....	77
Cisco Unified Communications Manager Route Pattern to Skype for Business Call Park range .....	79
Cisco Unified Communications Manager Route Pattern to Unity Connection Voice Mail.....	81
<b>Cisco UCM Extent and Connect.....</b>	<b>83</b>
Cisco UCM UC service Configuration .....	83
Cisco UCM service Profile Configuration .....	84
Cisco Unified CM IM Presence – CCMCIP Profile Configuration.....	87
Cisco UCM – SIP trunk to Cisco IM&Presence Trunk Configuration .....	88
Cisco UCM end user configuration .....	92
Remote Destination Configuration .....	97
Cisco UCM CTI Remote Device Configuration.....	99
<b>Cisco Unity Connection .....</b>	<b>104</b>
Cisco Unity Connection Telephony Integration – Add Phone System .....	104
Cisco Unity Connection Telephony Integration – Add Port Group.....	104
Cisco Unity Connection Telephony Integration – Add Ports.....	107
Cisco Unity Connection User Configuration.....	107
<b>Acronyms.....</b>	<b>110</b>

## Introduction

This document describes the steps and configurations necessary for Cisco Unified Communications Manager (Cisco UCM) release 10.5.2 to interoperate with the Skype for Business 2015 using the following configuration:

On the Cisco UCM: MTP Enabled, PRACK Disabled and Early Offer SIP Profile.

On the Skype for Business: Media Bypass Enabled, Refer Enabled, Encryption support level Optional

### The following items were tested:

- Basic call between the two systems and verification of voice path, using both SIP and SCCP 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)

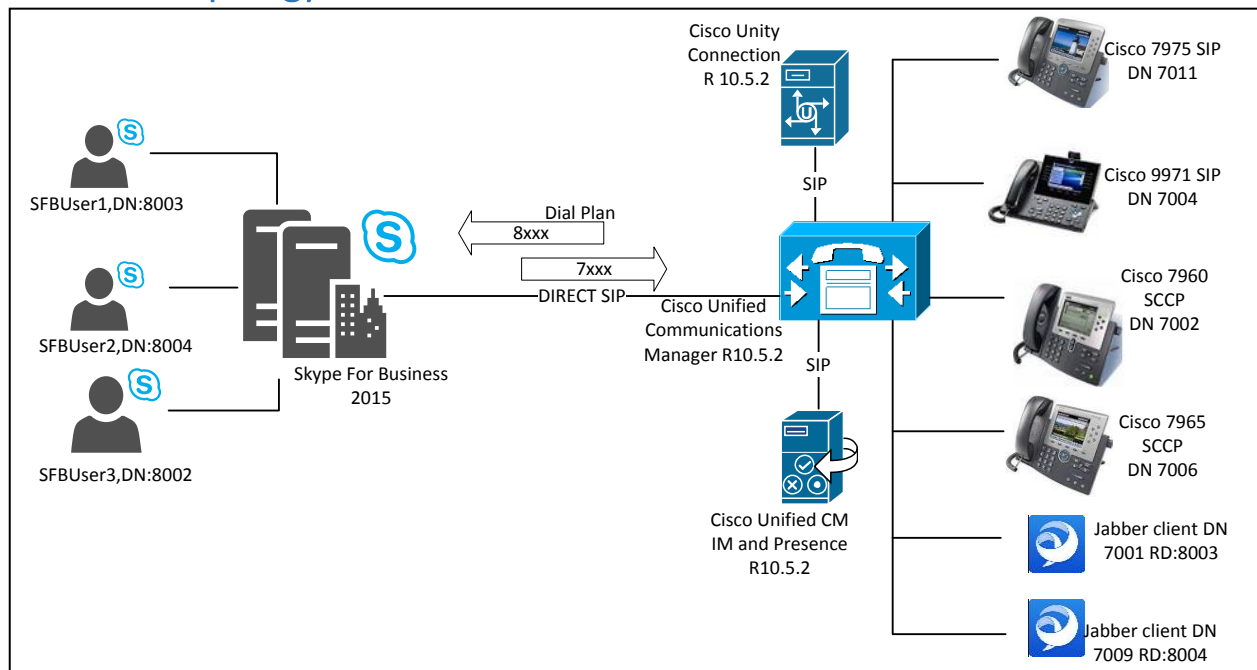
Listed below are the highlights of the integration issues:

- Basic calls work from Cisco UCM to Skype for Business and vice versa.
- Caller name and number is not updated correctly for basic calls and in the attended and early-attended transfer scenarios.
- Caller ID is updated to "Unknown Number" on Cisco UCM SIP phones and "External Call" on Cisco UCM SCCP Phones in transfer scenarios when a Skype for Business user initiates the transfer.
- Alerting name updates do not occur on Skype for Business.
- Video calls between the Cisco UCM and Skype for Business users were not tested.

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 was used for testing. This voicemail solution can provide centralized voicemail services, supporting both Skype for Business and Cisco end-users.

## Network Topology



## Limitations

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

- Skype for Business and Cisco UCM do not support overlap dialing modes on their SIP endpoints.
- Skype for Business does not support alerting name updates.
- Skype for Business does not update the caller ID (connected Name) for a basic or privacy enabled call from the Cisco UCM. Therefore, only the connected party number is displayed on the Cisco UCM Phone.
- Skype for Business does not update the CLID in transfer/conference scenarios. After the transfer/conference is complete, Cisco UCM sends mid call INVITE and UPDATE messages that contain PAI and RPI. However, Skype for Business does not update this information on its clients.
- In a transfer scenario, when Skype for Business initiates the call transfer, the caller ID of the initial Cisco UCM calling endpoint (transferee) is updated to “Unknown Number” if it is a SIP phone or “External Call” if it is an SCCP phone.
- In a call park scenario, when a Skype for Business client initiates the call park, the Cisco UCM endpoint that retrieves the parked call has its caller ID updated to “Unknown Number”.
- Skype for Business does not send PAID by default i.e. when restriction is not enabled. In an Extend & Connect scenario, this fails to initiate the Jabber client for call control. The incoming call to a Cisco UCM endpoint is therefore like a regular call without remote destination configuration.
  - This is currently a known issue on the Cisco UCM and is addressed by *“CSCuz48313 Tel URI / PAI support in CUCM”*.
  - As a workaround, the RD is configured with a “+” prefix and a route pattern to route a DN with a “+” prefix is also added.(Refer Cisco UCM configuration section - *Cisco Unified Communications Manager Route Pattern to invoke Jabber client with Remote Destination configured as Skype for Business Extensions*).
- Skype for Business does not support MWI notification from Cisco Unity Connection. It responds with a “405 Method Not Allowed” to a NOTIFY Message from the Cisco UCM that has MWI information.
- In a call forwarding scenario that involves multiple call forwards and a loop that terminates on a Cisco UCM or Skype for Business user, the calling party (Skype for Business client or Cisco UCM endpoint) hears a re-order tone when it calls the user on which the loop is formed.

## System Components

### Hardware Requirements

The following hardware was used

- Cisco UCS-C240-M3S VMWare Host

- Cisco 7960,7965 ,7975, 9951, and 9971 IP phones

## Software Requirements

The following software is required:

- Cisco UCSC-C240-M3S VMware vSphere Image Profile: ESXi-5.5.0-1331820-standard
- Cisco Unified Communications Manager release 10.5.2.13900-12
- Cisco Unified Communications Manager IM & P release 10.5.2.13900-12
- Cisco Unity Connection release 10.5.2.13900-12
- Cisco Jabber 11.6.0 Build 35037
- Skype for Business 2015 6.0.9319.0
- Skype for Business Client version : 15.0.4841.1000

## 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
- Alerting Name
- 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

- 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 3 PBXs.
- Scenarios involving Non SIP interfaces.

## 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.

### Configuring Sequence and Tasks:

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



## 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 and SCCP 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

## Cisco Unity Connection:

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

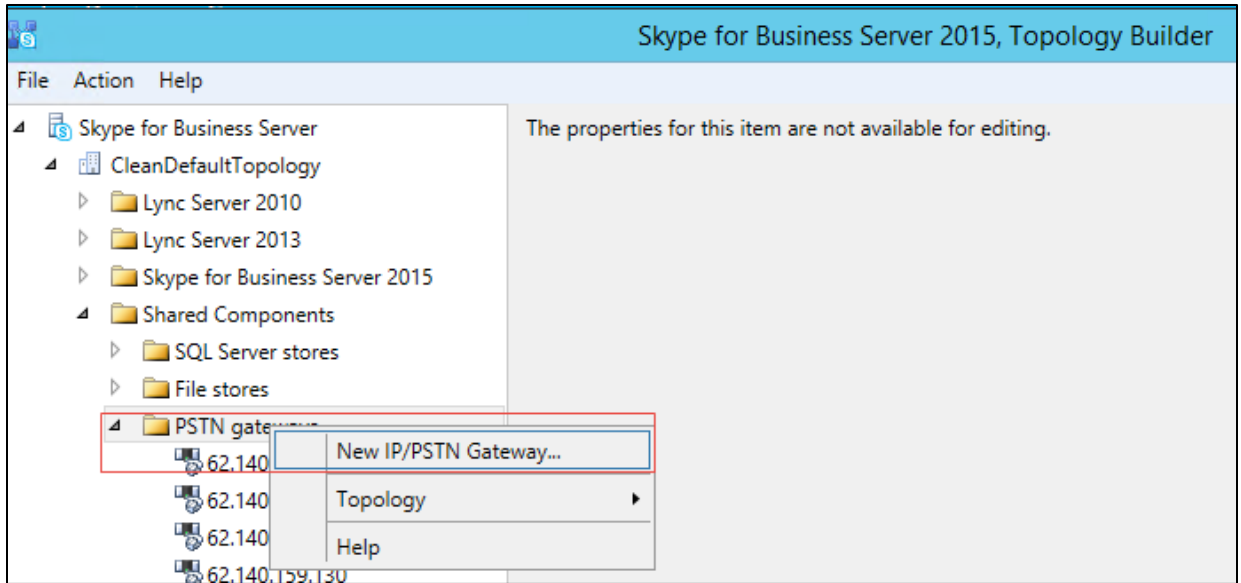
## Configuring the Skype for Business

### [Add Cisco UCM to Skype for Business Topology](#)

Run the Skype for Business 2015 Topology Builder as a user in the CSAdministrator group.

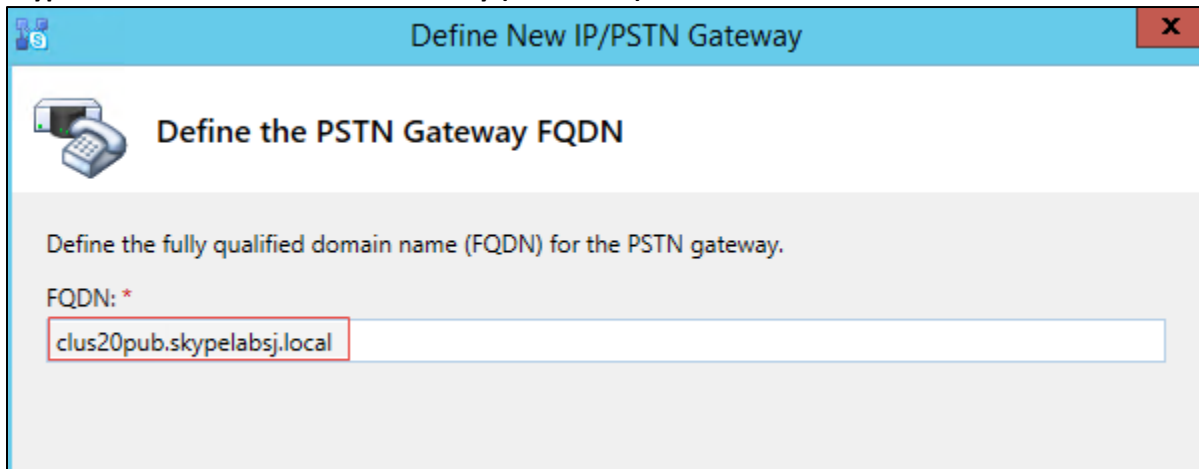
**Navigation:** Skype for Business Server → CleanDefaultTopology → Shared Components → PSTN gateways

Right click and select “New IP/PSTN Gateway”



Set FQDN = <FQDN of the Cisco UCM>- clus20pub.skypelabsj.local is used in this test.  
Click Next.

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



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

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

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 – clus20pub.skypelabsj.local 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** – 5060 is used for this test

Set **SIP Transport Protocol** = TCP

Set **Associate Mediation Server**: Assign this PSTN gateway to the Front End co-located mediation server – fe01.skypelabsj.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: \*

clus20pub.skypelabsj.local

Listening port for IP/PSTN gateway: \*

5060

SIP Transport Protocol:

TCP

Associated Mediation Server:

FE01.skypelabsj.local CleanDefaultTopology

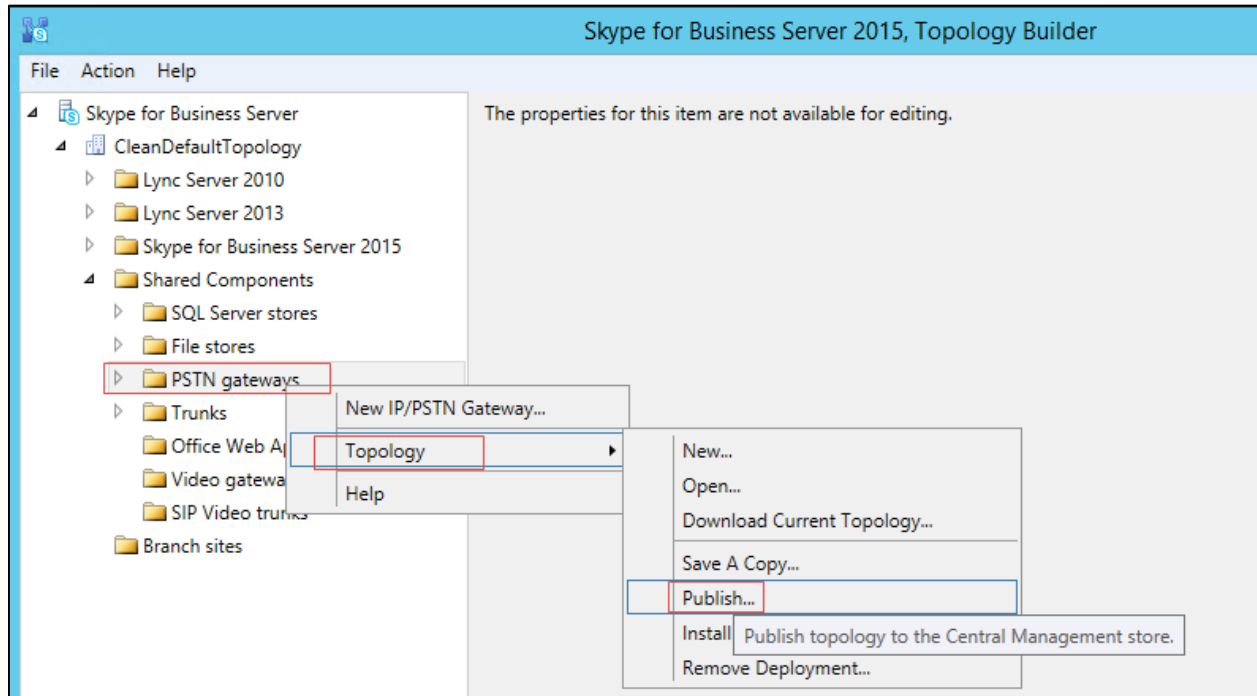
Associated Mediation Server port: \*

5060

Help Back Finish Cancel

Publish the topology so these new configurations take effect.

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

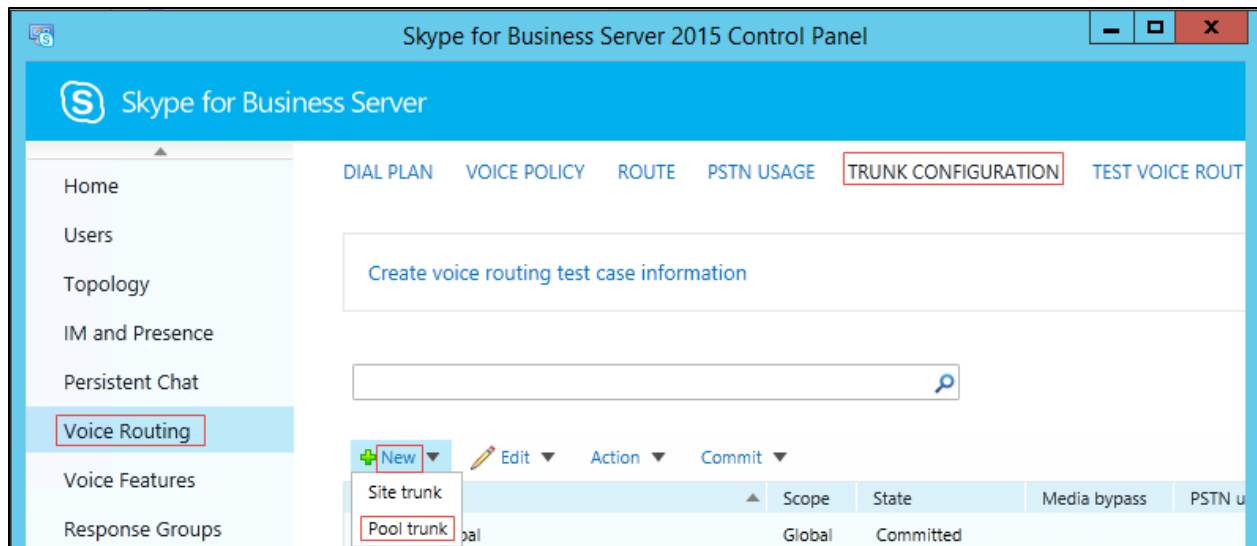


### Skype for Business Trunk Configuration

Open the Skype for Business 2015 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 – clus20pub.skypelabsj.local is used for the test.

Set **Maximum early dialogs supported** = 20

Set **Encryption support level** = Optional

Set **Refer Support** = Enable sending refer to the gateway

Check **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]

Check **Enable outbound routing failover timer**

**Skype for Business –Trunk Configuration (Continued)**

- 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

Create voice routing test case information

Edit Trunk Configuration - PstnGateway:clus20pub.skypelabsj.lo...

OK Cancel

Scope: Pool

Name: \*

PstnGateway:clus20pub.skypelabsj.local

Description:

Maximum early dialogs supported:

20

Encryption support level:

Optional

Refer support:

Enable sending refer to the gateway

Enable media bypass

Centralized media processing

Enable RTP latching

Enable forward call history

Enable forward P-Asserted-Identity data

Enable outbound routing failover timer



## Skype for Business –Trunk Configuration (Continued)

Edit Trunk Configuration - PstnGateway:clus20pub.skypelabsj.lo...

✓ OK ✗ Cancel

^ Associated PSTN Usages

Select... Remove ↑ ↓

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

Translated number to test:

^ Associated translation rules

Calling number translation rules



+ New Copy Paste Select... Show details... Remove ↑ ↓

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

Add a Translation rule under Called number translation rules – **CUCMExtn** was created in this test. This is used to remove the “+” that is added by SFB during a transfer to a Cisco UCM extension. If SFB attempts a transfer to a Cisco UCM extension without this rule, the transfer fails because the extension is not recognized by Cisco UCM.

## Skype for Business –Trunk Configuration- Translation Rule

Edit Trunk Configuration ▶ Edit Called Number Translation Rule - CUCMExtn

**Name: \***

**Description:**

**Build a Translation Rule**

Fill in the fields that you want to use, or create the rule manually by clicking Edit.

**Starting digits:**

**Length:**

**Digits to remove:**

**Digits to add:**

**Pattern to match: \***

**Translation rule: \***

## Skype for Business –Trunk Configuration (Continued)

Edit Trunk Configuration - PstnGateway:clus20pub.skypelabsj.lo...

✓ OK ✗ Cancel

**Called number translation rules**

+ New Copy Paste Select... Show details... Remove ↑ ↓

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

**Phone number to test:**

Go ?

Calling number  Called number

## Skype for Business Route Configuration

**Navigation:** Voice Routing -> Route

Click New

Set Name = enter a name to identify this Route. SFB-Cisco is used for this test.

Add Associated trunks = select the trunk configured earlier – PstnGateway:clus20pub.skypelabsj.local

- 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 - SFB-Cisco

OK  Cancel

Scope:

Name: \*

SFB-Cisco

Description:

Build a Pattern to Match

Add the starting digits that you want this route to handle, or create the expression manually by clicking Edit.

Starting digits for numbers that you want to allow:

Type a valid number and then click Add.

Match this pattern: \*

\*.\*

Suppress caller ID

Alternate caller ID:

## Skype for Business Voice Policy and PSTN Usage Configuration

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

Click New

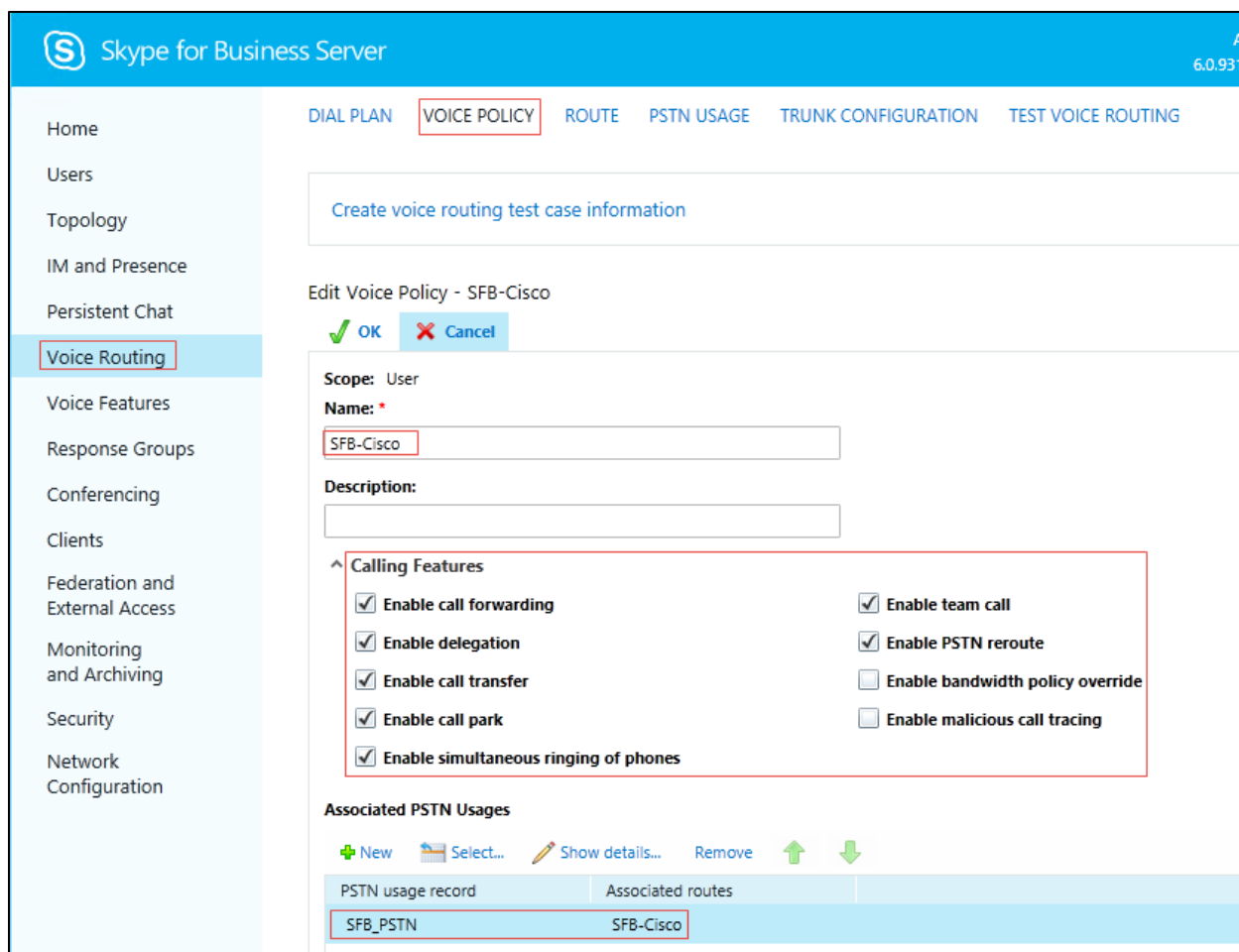
Set Name = enter a name to identify this voice policy – SFB-Cisco is used in this test.

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

Set Associated PSTN usages:

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



## Skype for Business Dial Plan Configuration

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

Add a new User dial plan and a new Pool dial plan.

User dial plan:

Set Name = enter text to identify this dial plan – *cucm* is used in this test.

A user dial plan with a normalization rule was configured for this test:

- CUCM 4 Digit: To reach the 4 digit extensions from Cisco UCM – This allows 4 digits to be dialed and not undergo any normalization.

- 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 - cucm

OK Cancel

Scope: User

Name: \*

Simple name: \*

Description:

Dial-in conferencing region:

External access prefix:


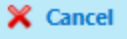
Associated Normalization Rules

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

Normalization rule	State	Pattern to match	Translation pattern
CUCM 4digit	Committed	^\d{4}\$	\$1

## Skype for Business – User Dial Plan-Normalization Rule

Edit Dial Plan ▶ Edit Normalization Rule - CUCM 4digit

**Name: \***

**Description:**

**Build a Normalization Rule**  
Fill in the fields that you want to use, or create the rule manually by clicking Edit.

**Starting digits:**


**Length:**  
Exactly

**Digits to remove:**

**Digits to add:**

**Pattern to match: \***

**Translation rule: \***



Pool dial plan:

Select Service: PstnGateway:clus20pub.skypelabsj.local is selected

Set Simple Name= enter text to identify this pool dial plan. PstnGateway\_clus20pub.skypelabsj.local is used in this test



Associated Normalization Rules→New

Set Name: enter text to identify this rule – Call pick up From CUCM was created in this test

This is to accept the call park range dialed by Cisco UCM users to retrieve a call parked by the SFB client.

### **Skype for Business – Pool Dial Plan-Normalization Rule 1**

Edit Dial Plan ▶ Edit Normalization Rule - Call pick up From CUCM

**Name: \***

Call pick up From CUCM

**Description:**

**Build a Normalization Rule**

Fill in the fields that you want to use, or create the rule manually by clicking Edit.

**Starting digits:**

**Length:**

Exactly

**Digits to remove:**

0

**Digits to add:**

**Pattern to match: \***

`^\d{3}$`

**Translation rule: \***

`$1`

**Internal extension**



**Dialed number to test:**

Add another normalization rule as below:

This is used by the client to dial out to internal extensions and to the external PBX i.e. Cisco UCM

## Skype for Business – Pool Dial Plan-Normalization Rule 2

Edit Dial Plan ▶ Edit Normalization Rule - Keep All

 OK  Cancel

**Name: \***

**Description:**

**Build a Normalization Rule**

Fill in the fields that you want to use, or create the rule manually by clicking Edit.

**Starting digits:**


**Length:**  
At least


**Digits to remove:**

**Digits to add:**  
+

**Pattern to match: \***

**Translation rule: \***



Internal extension 

## Skype for Business Call Park Range Configuration

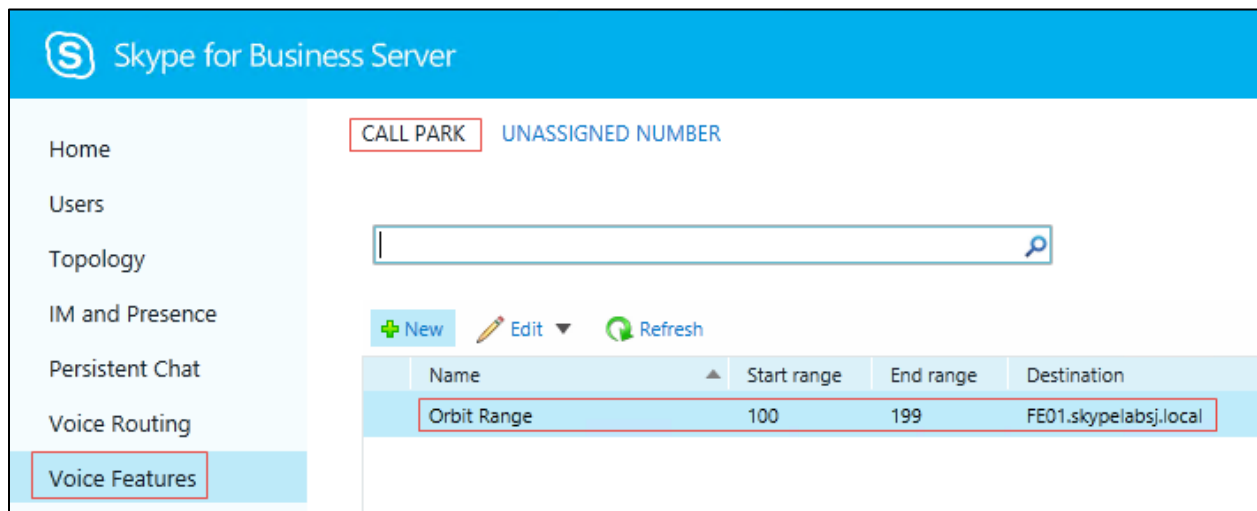
**Navigation:** Voice Features -> Call Park

Click New.

Set Name = enter text to identify this call park range – Orbit range is used in the test.

Set Number Range = 100 to 199 is used in the test.

Set FQDN of destination server= select the desired server - FE01.skypelabsj.local is used in the test



Skype for Business Server

CALL PARK UNASSIGNED NUMBER

Home  
Users  
Topology  
IM and Presence  
Persistent Chat  
Voice Routing  
Voice Features

+ New Edit Refresh

Name	Start range	End range	Destination
Orbit Range	100	199	FE01.skypelabsj.local

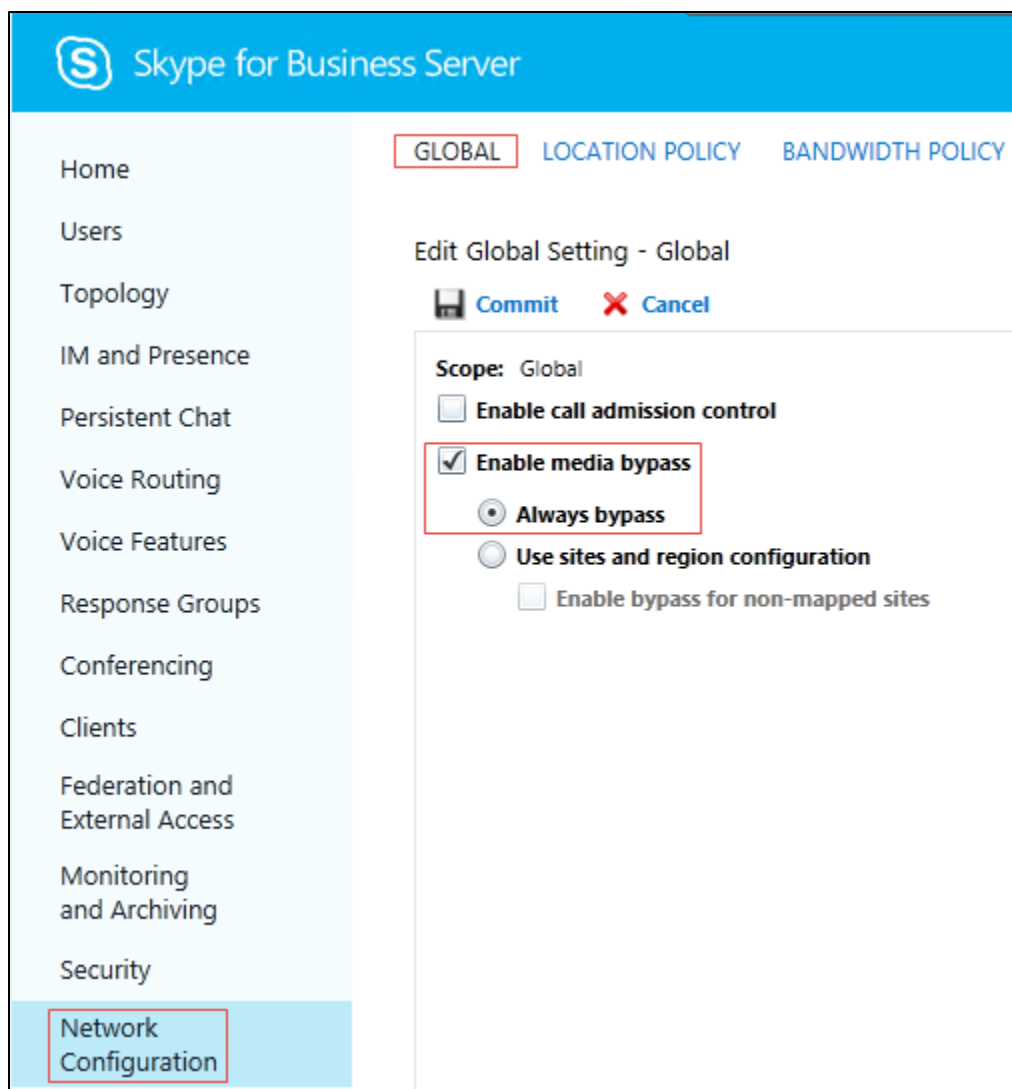
## Skype for Business Global Media Bypass Configuration

**Navigation:** Network Configuration -> Global

Edit Global Setting –

- Check Enable media bypass
- Check Always bypass

Commit the configuration.

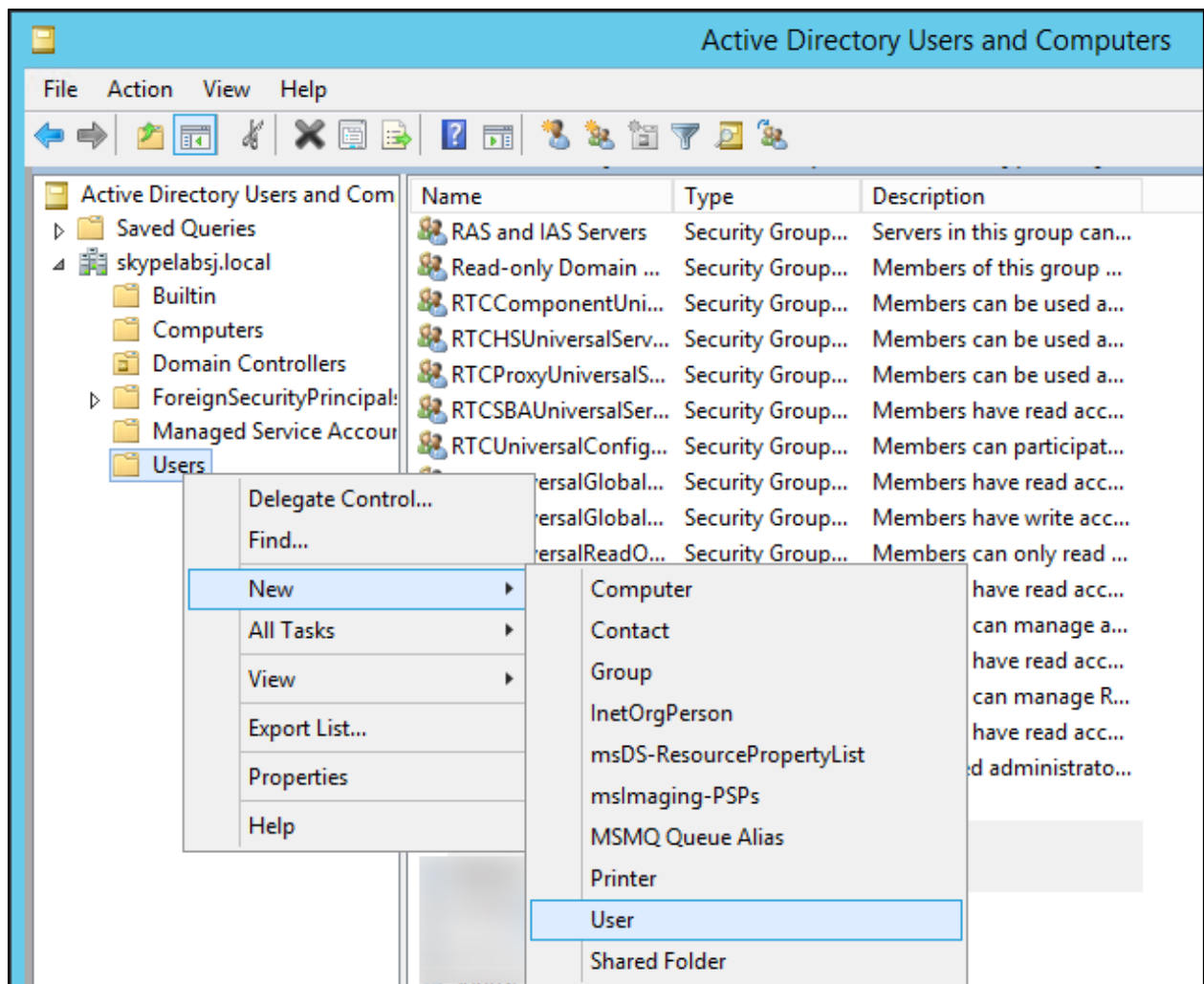


## Skype for Business User Configuration

Login to the Skype for Business Active Directory

Navigation: Active Directory Users and Computers → Users

Add a New User



Follow the screenshots below to add a new user:

Skype for Business – New User configuration (continued)

New Object - User

Create in: skypeLABSJ.local/Users

First name: SFBUser1 Initials: |

Last name:

Full name: SFBUser1


User logon name: user1 @skypeLABSJ.local

User logon name (pre-Windows 2000): SKYPELABSJ\ user1

< Back Next > Cancel

Skype for Business – New User configuration (continued)

New Object - User ✕

 Create in: skypeLABSj.local/Users

---

Password:

Confirm password:

User must change password at next logon

User cannot change password

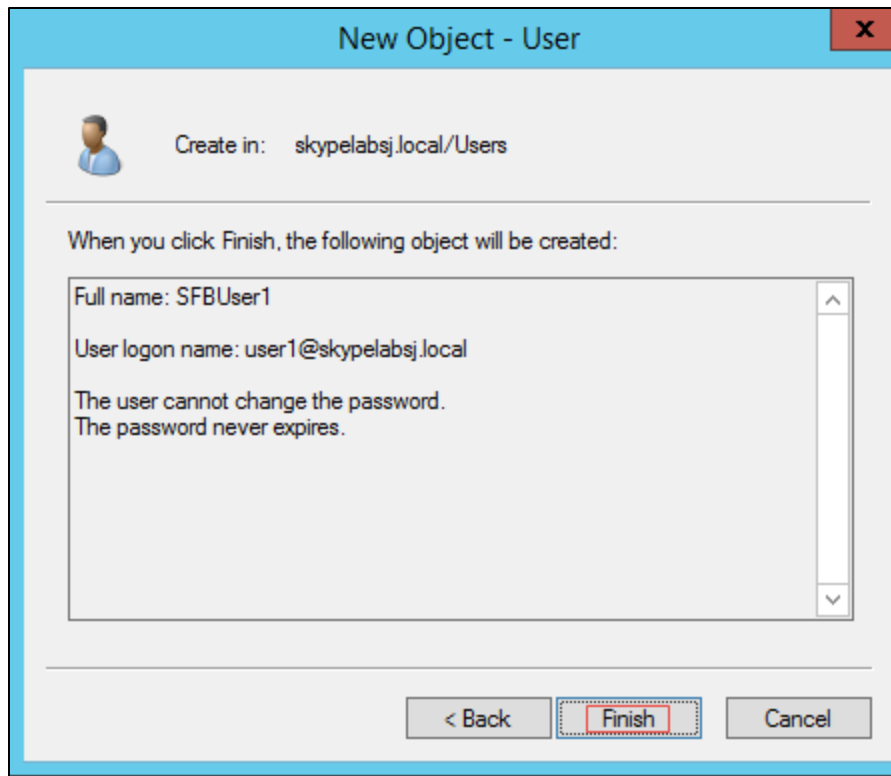
Password never expires

Account is disabled

---

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





Once the user is created, login to the Skype for Business 2015 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 2015 Control Panel

Skype for Business Server

Home

**Users**

Topology

IM and Presence

Persistent Chat

Voice Routing

Voice Features

Response Groups

Conferencing

Clients

USER SEARCH

New Skype for Business Server User

**Enable** **Cancel**

**Users:**

Display name	Status	
		<b>Add...</b>
		<b>Remove</b>

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

**Generate user's SIP URI:**

Use user's email address

Set Assign users to a pool= FE01.skypelabsj.local from drop down menu

Set Generate user's SIP URI: Specify a SIP URI: sip:SFBUser1@skypelabsj.local .This is used in this test

Set Telephony=Enterprise Voice

Set Line URI: = <tel:+8003> is used for the test. This is the DN for the user.

Set Dial plan policy = cucm (as configured earlier)

Set Voice policy= SFB-Cisco (as configured earlier)

Click Enable.

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

**Display name:**  
SFBUser1

**Enabled for Skype for Business Server**

**SIP address: \***  
sip:SFBUser1 @ skype1absj.local

**Registrar pool:**  
FE01.skype1absj.local

**Telephony:**  
Enterprise Voice

**Line URI:**  
tel:+8003

**Dial plan policy:**  
cucm [View...](#)

**Voice policy:**  
SFB-Cisco [View...](#)

**Conferencing policy:**  
<Automatic> [View...](#)

**Client version policy:**  
<Automatic> [View...](#)

**PIN policy:**  
<Automatic> [View...](#)

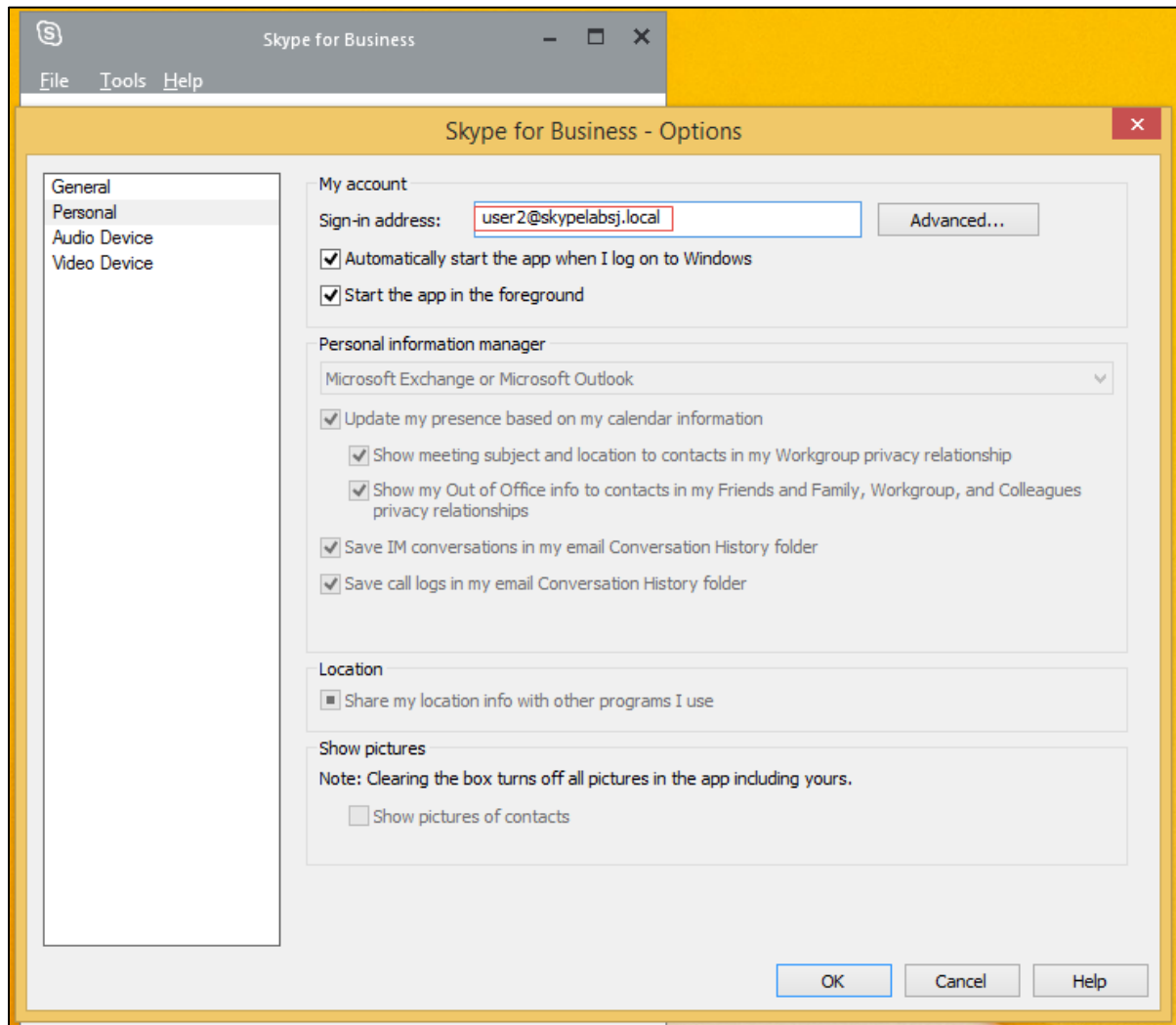
**External access policy:**  
<Automatic> [View...](#)

### Skype for Business Client Configuration

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

Navigation: Settings→Tools→Options→Personal→MyAccount

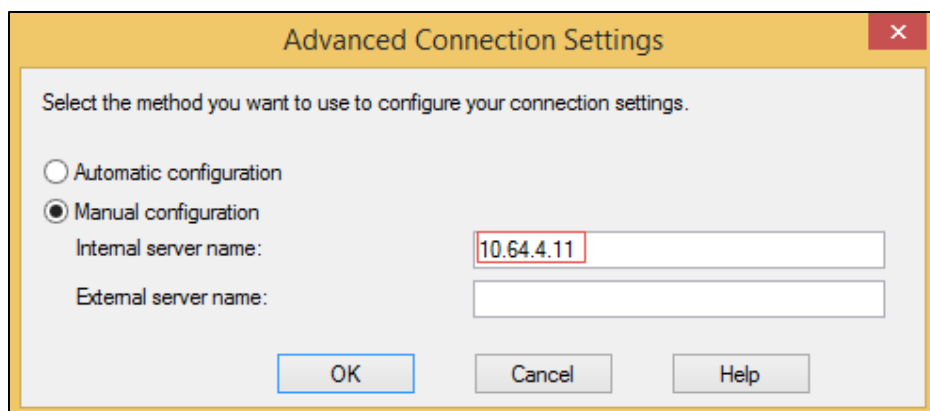
Set Sign-in-address= enter the sip uri of the user configured in username@domain format. [user2@skypelabsj.local](mailto:user2@skypelabsj.local) is used for example.



Click Advanced. Select Manual Configuration.

Set Internal Server Name= Enter the IP address of the Skype for Business Front End Pool

**Skype for Business – Client configuration (continued)**



## Configuring the Cisco Unified Communications Manager

Cisco Unified Communications Manager Software Version

## Cisco Unified Communications Manager SIP Trunk Security Profile for Trunk to Skype for Business

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

Set Name\*= SFB-Non-secure. This is used for the test.

- Set Device Security mode = Non Secure
- Set Incoming Transport Type = TCP+UDP
- Set Outgoing Transport Type = TCP
- Check Accept Presence Subscription
- Check Accept out of dialog refer
- Check Accept unsolicited notification
- Check Accept Replaces header
- 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 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*	SFB-Non-secure
Description	
Device Security Mode	Non Secure
Incoming Transport Type*	TCP+UDP
Outgoing Transport Type	TCP

Enable Digest Authentication  
Nonce Validity Time (mins)\*: 600  
X.509 Subject Name:   
Incoming Port\*: 5060

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

## Cisco Unified Communications Manager SIP Trunk Security Profile for Trunk to Unity Connection

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

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

Set Device Security mode = Non Secure

Set Incoming Transport Type = TCP+UDP

Set Outgoing Transport Type = TCP

Check Accept Presence Subscription

Check Accept unsolicited notification

Check Accept Replaces header

All other values are default.

### SIP Trunk Security Profile Configuration

Save Delete Copy Reset Apply Config Add New

#### SIP Trunk Security Profile Information

Name*	UnityConnectionTrunkSecurityProfile
Description	UnityConnectionTrunkSecurityProfile
Device Security Mode	Non Secure
Incoming Transport Type*	TCP+UDP
Outgoing Transport Type	TCP
<input type="checkbox"/> Enable Digest Authentication	
Nonce Validity Time (mins)*	600
X.509 Subject Name	
Incoming Port*	5060
<input type="checkbox"/> Enable Application level authorization	
<input checked="" type="checkbox"/> Accept presence subscription	
<input 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

## Cisco Unified Communications Manager SIP Profile

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

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

Set Description = this text is used to identify this SIP Profile.

Set SIP Rel1XX Options = Disabled

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.

The screenshot shows the Cisco Unified CM Administration interface for SIP Profile Configuration. The page title is "SIP Profile Configuration" and it includes a navigation menu with options like System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The main content area is divided into two sections: "Status" and "SIP Profile Information".

**Status**

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

**SIP Profile Information**

Name*	SFB - Standard SIP Profile
Description	Default 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

Use Fully Qualified Domain Name in SIP Requests

Assured Services SIP conformance



## 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\* TIAS and AS

SDP Transparency Profile Pass all unknown SDP attributes

Accept Audio Codec Preferences in Received Offer\* Default

Require SDP Inactive Exchange for Mid-Call Media Change

Allow RR/RS bandwidth modifier (RFC 3556)

**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
Start Media Port*	16384
Stop Media Port*	32766
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup

## Cisco Unified Communications Manager SIP Profile (Continued)

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

Call Pickup Group URI*	<input type="text" value="x-cisco-serviceuri-gpickup"/>
Meet Me Service URI*	<input type="text" value="x-cisco-serviceuri-meetme"/>
User Info*	<input type="text" value="None"/>
DTMF DB Level*	<input type="text" value="Nominal"/>
Call Hold Ring Back*	<input type="text" value="Off"/>
Anonymous Call Block*	<input type="text" value="Off"/>
Caller ID Blocking*	<input type="text" value="Off"/>
Do Not Disturb Control*	<input type="text" value="User"/>
Telnet Level for 7940 and 7960*	<input type="text" value="Disabled"/>
Resource Priority Namespace	<input type="text" value="&lt; None &gt;"/>
Timer Keep Alive Expires (seconds)*	<input type="text" value="120"/>
Timer Subscribe Expires (seconds)*	<input type="text" value="120"/>
Timer Subscribe Delta (seconds)*	<input type="text" value="5"/>
Maximum Redirections*	<input type="text" value="70"/>
Off Hook To First Digit Timer (milliseconds)*	<input type="text" value="15000"/>
Call Forward URI*	<input type="text" value="x-cisco-serviceuri-cfwdall"/>
Speed Dial (Abbreviated Dial) URI*	<input type="text" value="x-cisco-serviceuri-abbrdial"/>

Conference Join Enabled  
 RFC 2543 Hold  
 Semi Attended Transfer

## Cisco Unified Communications Manager SIP Profile (Continued)

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

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"/>	<input type="button" value="+"/> <input type="button" value="-"/>

**Incoming Requests FROM URI Settings**

Caller ID DN:   
Caller Name:

**Trunk Specific Configuration**

Reroute Incoming Request to new Trunk based on\*:

RSVP Over SIP\*:

Resource Priority Namespace List:

Fall back to local RSVP

SIP Rel1XX Options\*:

Video Call Traffic Class\*:

Calling Line Identification Presentation\*:

## Cisco Unified Communications Manager SIP Profile (Continued)

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

Session Refresh Method\*

Early Offer support for voice and video calls\*

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

**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)\*

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

Ping Retry Timer (milliseconds)\*

Ping Retry Count\*

**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

## Cisco Unified Communications Manager Media Resource Group

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

### Media Resource Group MRG

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

Set Description = this text is used to identify this Media Resource Group.

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

All other values are default.

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

---

**Status**

Status: Ready

---

**Media Resource Group Status**

Media Resource Group: MRG (used by 4 devices)

---

**Media Resource Group Information**

Name\*

Description

---

**Devices for this Group**

Available Media Resources\*\*

▼ ▲

Selected Media Resources\*

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

---

### Resource Group for MRG\_NoMTP

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

Set Description = this text is used to identify this Media Resource Group.

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

Set other resources in the Selected Media Resources\*

All other values are default.

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

**Status**

Status: Ready

**Media Resource Group Status**

Media Resource Group: MRG\_NoMTP (used by 26 devices)

**Media Resource Group Information**

Name\*

Description

**Devices for this Group**

Available Media Resources\*\*

- MTP\_2 (MTP)
- MTP\_4 (MTP)
- MTP\_3 (MTP)

v v

Selected Media Resources\*

- ANN\_2 (ANN)
- ANN\_3 (ANN)
- ANN\_4 (ANN)
- CFB\_2 (CFB)
- CFB\_3 (CFB)

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

### Cisco Unified Communications Manager Media Resource Group Configuration

**Cisco Unified CM Administration** Navigation [Cisco Unified CM Administration](#)

**For Cisco Unified Communications Solutions** 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**

**Status**

2 records found

**Media Resource Group (1 - 2 of 2)** Rows per Page 50 ▾

Find Media Resource Group where Name ▾ begins with ▾

<input type="checkbox"/>	Name ^	Description	Multi-cast	Copy
<input type="checkbox"/>	<a href="#">MRG</a>		false	<input type="button" value="Copy"/>
<input type="checkbox"/>	<a href="#">MRG_NoMTP</a>		false	<input type="button" value="Copy"/>

## Cisco Unified Communications Manager Media Resource Group List

Navigation Path: Media Resources → Media Resource Group List

Add New

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

Set Description = this text is used to identify this Media Resource Group List.

Set Available Media Resources = MRGL

Set Selected Media Resource Groups = MRG

The screenshot shows the 'Media Resource Group List Configuration' page. At the top, there is a title bar with 'Media Resource Group List Configuration' and 'Related Links: Back To Find/List Go'. Below the title bar is a toolbar with icons for Save, Delete, Copy, and Add New. The main content area is divided into several sections: 'Status' (Status: Ready), 'Media Resource Group List Status' (Media Resource Group List: MRGL (used by 4 devices)), 'Media Resource Group List Information' (Name\*: MRGL), and 'Media Resource Groups for this List'. The 'Media Resource Groups for this List' section contains two dropdown menus: 'Available Media Resource Groups' (MRG\_NoMTP) and 'Selected Media Resource Groups' (MRG). At the bottom of the page, there are buttons for Save, Delete, Copy, and Add New.

Add new

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

Set Description = this text is used to identify this Media Resource Group List

Set Available Media Resources MRG

Set Selected Media Resource Groups = MRGL\_NoMTP

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

**Status**

Status: Ready

**Media Resource Group List Status**

Media Resource Group List: MRGL\_NoMTP (used by 26 devices)

**Media Resource Group List Information**

Name\*

**Media Resource Groups for this List**

Available Media Resource Groups

▼ ▲

Selected Media Resource Groups

▼ ▲

### Cisco Unified Communications Manager Media Resource Group List Configuration

**Cisco Unified CM Administration** Navigation [Cisco Unified CM Administration](#)

**For Cisco Unified Communications Solutions** 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**

**Status**

2 records found

**Media Resource Group List (1 - 2 of 2)** Rows per Page 50 ▾

Find Media Resource Group List where Name

<input type="checkbox"/>	Name ^	Copy
<input type="checkbox"/>	<a href="#">MRGL</a>	
<input type="checkbox"/>	<a href="#">MRGL_NoMTP</a>	



## Cisco Unified Communications Manager Device Pool Configuration

Device Pool - **G711 Preferred** is created in this test.

**Navigation Path:** System → Device Pool

Add New.

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

Set Cisco Unified Communications Manager Group\* = Default

Set Date/Time Group\* = CMLocal

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

Set Media Resource Group List = MRGL. This is used in this example.

All other values are default.

The screenshot displays the Cisco Unified CM Administration interface for configuring a Device Pool. The page title is "Device Pool Configuration" and the device pool name is "G711 Preferred (29 members\*\*)".

**Device Pool Settings:**

- Device Pool Name\*: G711 Preferred
- 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 Preferred
- Media Resource Group List: MRGL
- Location: < None >
- Network Locale: < None >
- SRST Reference\*: Disable
- Connection Monitor Duration\*\*\*: [Empty field]
- Single Button Barge\*: Default
- Join Across Lines\*: Default
- Physical Location: < None >
- Device Mobility Group: < None >
- Wireless LAN Profile Group: < None >

There is a "View Details" link at the bottom right of the Roaming Sensitive Settings section.

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

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

Save
 Delete
 Copy
 Reset
 Apply Config
 Add New

**Local Route Group Settings**

Standard Local Route Group

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

Device Mobility Calling Search Space

AAR Calling Search Space

AAR Group

Calling Party Transformation CSS

Called Party Transformation CSS

**Geolocation Configuration**

Geolocation

Geolocation Filter

**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.

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

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

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

---

**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.

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value=" &lt; None &gt;"/>
International Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value=" &lt; None &gt;"/>
Unknown Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value=" &lt; None &gt;"/>
Subscriber Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value=" &lt; None &gt;"/>

---

**Phone Settings**

**Caller ID For Calls From This Phone**

Calling Party Transformation CSS

---

**Connected Party Settings**

Connected Party Transformation CSS

---

**Redirecting Party Settings**

Redirecting Party Transformation CSS

---

## Cisco Unified Communications Manager Region Configuration

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

Add New

G711 Preferred is created in this test.

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

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

Set Audio Codec Preference List= G711 Preferred

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 G729. 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

**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

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

Save Delete Reset Apply Config Add New

**Region Information**  
Name\* G711 Preferred

**Region Relationships**

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 G729	64 kbps (G.722, G.711)	Use System Default (384 kbps)	Use System Default (2000000000 kbps)
G711 Preferred	G711 G729	64 kbps (G.722, G.711)	Use System Default (384 kbps)	Use System Default (2000000000 kbps)
NOTE: Regions not displayed	Use System Default	Use System Default	Use System Default	Use System Default

## Cisco Unified Communications Manager Normalization Script

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

Add New

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

Set Content = add script content.

Note: The only part of script used for this test was converting the History-Info to Diversion since call forward to Unity Connection fails without the Diversion header since it doesn't support History-Info.

SIP Normalization Script Configuration Related Links: [Back To Find/List](#)

**Status**

Status: Ready

**SIP Normalization Script Info**

Name\*

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\*

System Resource Error Recovery Action\*

Memory Threshold\*  kilobytes

Lua Instruction Threshold\*  instructions

## Normalization Script

```
--[[

Description:
Provides interoperability for 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 end
point 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)

Copyright (c) 2009-2011 Cisco Systems, Inc. All rights reserved.

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

    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", " ")

        -- MS format: <sip:+19728522619@med02.lync labsj.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

```

```

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)
end

function process_inbound_any_response(msg)
  msg:addHeader("SUPPORTED","X-cisco-srtp-fallback")
  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_request(msg)

```

```

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

```

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

**Navigation:** Device → Trunk

Set Device Name\* = SFB-FE01-CUCM. This is used for the test

Set Description = this text is used to identify this Trunk Group

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

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

Set Media Resource Group List = MRGL\_MTP. This is used for the test

Check Media Termination Point Required

Check Run On All Active Unified CM Nodes

Check Redirecting Diversion Header Delivery – Inbound

Set Destination Address = FE01.skypelabsj.local. [FQDN of Skype for Business Front End]This is used in the test

Set SIP Trunk Security Profile\*= SFB-Non-secure

Set SIP Profile\*= SFB – Standard SIP Profile

Set DTMF Signaling Method\*= RFC 2833

Set Normalization Script = CiscoScriptForSFB

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 For Cisco Unified Communications Solutions", and a navigation menu with "Cisco Unified CM Administration" selected. Below the navigation bar, there are tabs for "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", "Bulk Administration", and "Help". The main content area is titled "Trunk Configuration" and includes a "Related Links" section with "Back To Find/List". Below this, there are icons for "Save", "Delete", "Reset", and "Add New".

The configuration details are organized into sections:

- Status:** Status: Ready
- SIP Trunk Status:** Service Status: Full Service; Duration: Time In Full Service: 0 day 0 hour 2 minutes
- Device Information:** Product: SIP Trunk; Device Protocol: SIP; Trunk Service Type: None(Default); Device Name\*: SFB-FE01-CUCM; Description: (empty); Device Pool\*: G711 Preferred; Common Device Configuration: < None >; Call Classification\*: Use System Default; Media Resource Group List: MRGL; Location\*: Hub\_None; AAR Group: < None >; Tunneled Protocol\*: None; QSIG Variant\*: No Changes; ASN.1 ROSE OID Encoding\*: No Changes; Packet Capture Mode\*: None

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

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

Packet Capture Duration

**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\*

Route Class Signaling Enabled\*

Use Trusted Relay Point\*

PSTN Access

**Run On All Active Unified CM Nodes**

---

**Intercompany Media Engine (IME)**

E.164 Transformation Profile

---

**MLPP and Confidential Access Level Information**

MLPP Domain

Confidential Access Mode

Confidential Access Level

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

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

**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

**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

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

**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	Default	0	< None >	<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	Default	0	< None >	<input checked="" type="checkbox"/>

**Connected Party Settings**

Connected Party Transformation CSS < None >

Use Device Pool Connected Party Transformation CSS



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

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

---

**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\*

Calling and Connected Party Info Format\*

Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS

Use Device Pool Redirecting Party Transformation CSS

---

**Caller Information**

Caller ID DN

Caller Name

Maintain Original Caller ID DN and Caller Name in Identity Headers

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

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

---

**SIP Information**

Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port	Status
1*	FE01.skypelabsj.local		5060	up

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\*  [View Details](#)  
DTMF Signaling Method\*

**Normalization Script**

Normalization Script

Enable Trace

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

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

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

**Recording Information**

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

**Geolocation Configuration**

Geolocation   
Geolocation Filter   
 Send Geolocation Information

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

## Cisco Unified Communications Manager SIP Trunk to Cisco Unity Connection Configuration

**Navigation:** Device → Trunk

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

Set Description = this text is used to identify this Trunk Group.

Set Device Pool\* = G711 Preferred

Check Run On All Active Unified CM Nodes

Check Redirecting Diversion Header Delivery – Inbound

Check Redirecting Diversion Header Delivery – Outbound

Set Destination Address = 10.80.10.5. This is used for the test.

Set SIP Trunk Security Profile\*= UnityConnectionTrunkSecurityProfile

Set SIP Profile\*= SFB - Standard SIP Profile

DTMF Signaling Method \*= RFC 2833

All other values are default

The screenshot shows the Cisco Unified CM Administration interface for Trunk Configuration. The page includes a navigation bar at the top with the Cisco logo and the text "Cisco Unified CM Administration For Cisco Unified Communications Solutions". The user is logged in as "administrator". The main content area is titled "Trunk Configuration" and includes a "Related Links" section with a "Back To Find/List" button. Below this, there are icons for "Save", "Delete", "Reset", and "Add New". The configuration is organized into sections: "Status" (Ready), "SIP Trunk Status" (Full Service, 0 day 1 hour 12 minutes), and "Device Information". The Device Information section contains a table of configuration fields:

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	UnityConnection
Description	UnityConnection
Device Pool*	G711 Preferred
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](#)

Packet Capture Duration

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\*

Route Class Signaling Enabled\*

Use Trusted Relay Point\*

PSTN Access

Run On All Active Unified CM Nodes

---

**Intercompany Media Engine (IME)**

E.164 Transformation Profile

---

**MLPP and Confidential Access Level Information**

MLPP Domain

Confidential Access Mode

Confidential Access Level

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

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

---

**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](#)

---

**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.

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 >

Use Device Pool Redirecting Party Transformation CSS

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

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

---

**Caller Information**

Caller ID DN:

Caller Name:

Maintain Original Caller ID DN and Caller Name in Identity Headers

---

**SIP Information**

**Destination**

Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1 *	10.80.10.5		5060

MTP Preferred Originating Codec\*: 711ulaw

BLF Presence Group\*: Standard Presence group

**SIP Trunk Security Profile\***: UnityConnectionTrunkSecurityProfile

Rerouting Calling Search Space: < None >

Out-Of-Dialog Refer Calling Search Space: < None >

SUBSCRIBE Calling Search Space: < None >

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

DTMF Signaling Method\*: RFC 2833

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

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

---

**Normalization Script**

Normalization Script

Enable Trace

	Parameter Name	Parameter Value	
1	<input type="text"/>	<input type="text"/>	<input type="button" value="+"/> <input type="button" value="−"/>

---

**Recording Information**

None

This trunk connects to a recording-enabled gateway

This trunk connects to other clusters with recording-enabled gateways

---

**Geolocation Configuration**

Geolocation

Geolocation Filter

Send Geolocation Information

---

## Cisco Unified Communications Manager Route Group

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

Add New

SFB-CUCM was configured in this test

Set Route Group Name = SFB-CUCM

Set Distribution Algorithm = Circular

Select SFB-FE-01-CUCM from Available Devices and click the Add to Route Group

**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

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

Save

**Route Group Information**

Route Group Name\* SFB-CUCM  
Distribution Algorithm\* Circular

**Route Group Member Information**

**Find Devices to Add to Route Group**

Device Name contains [ ] Find

Available Devices\*\*  
SFB-FE01-CUCM

Port(s) None Available

Add to Route Group

**Current Route Group Members**

Selected Devices (ordered by priority)\* SFB-FE01-CUCM (All Ports) [Reverse Order of Selected Devices]

Removed Devices\*\*\*

## Cisco Unified Communications Manager Route List

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

Add New

SFB-CUCM\_Route List was created for this test.

Set Name: SFB-CUCM\_Route List

Set Cisco Unified Communications Manager Group = Default

Click on Add Route Group

Set Route Group\* = SFB-CUCM-[NON-QSIG]



The screenshot displays the Cisco Unified CM Administration web interface. At the top, the navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and the user role "administrator". Below this is a menu bar with options like System, Call Routing, Media Resources, etc. The main content area is titled "Route List Configuration" and contains three sections:

- Route List Information:** Shows registration details (Registered with Cisco Unified Communications Manager clus20pub, IPv4 Address: 10.80.10.2), a checked "Device is trusted" box, and a "Name\*" field containing "SFB-CUCM\_Route List". Other fields include "Description" and "Cisco Unified Communications Manager Group\*" set to "Default". There are checkboxes for "Enable this Route List" and "Run On All Active Unified CM Nodes".
- Route List Member Information:** Features a "Selected Groups\*\*" list with "SFB-CUCM" selected, a "Removed Groups\*\*\*" list, and an "Add Route Group" button.
- Route List Details:** Shows a small icon and the text "SFB-CUCM".

## Cisco Unified Communications Manager SIP Route Pattern

**Navigation:** Call Routing → SIP Route Pattern

Add New

Set IPv4 Pattern\* = fe01.skypelabsj.local. This is the FQDN of the Skype for Business Front End server.

Set SIP Trunk/Route List\* = SFB\_CUCM\_Route List

**Cisco Unified CM Administration** 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 Route Pattern Configuration

Related Links: Back To Find/List Go

Save Delete Copy Add New

**Pattern Definition**

Pattern Usage: Domain Routing

IPv4 Pattern\*: fe01.skypelabsj.local

IPv6 Pattern:

Description:

Route Partition: < None >

SIP Trunk/Route List\*: SFB-CUCM\_Route List (Edit)

Block Pattern

**Calling Party Transformations**

Use Calling Party's External Phone Mask

Calling Party Transformation Mask:

Prefix Digits (Outgoing Calls):

Calling Line ID Presentation\*: Default

Calling Line Name Presentation\*: Default

**Connected Party Transformations**

Connected Line ID Presentation\*: Default

Connected Line Name Presentation\*: Default

Save Delete Copy Add New

In a similar way, add SIP Route Patterns for all the servers that comprise the Skype for Business environment.

In the test, the following SIP Route Patterns were configured:

**SIP Route Pattern (1 - 4 of 4)** Rows per Page: 50

Find SIP Route Pattern where IPv4 Pattern begins with Find Clear Filter

<input type="checkbox"/>	IPv4 Pattern ^	IPv6 Pattern	Description	Route Partition	Copy
<input type="checkbox"/>	<a href="#">fe01.skypelabsj.local</a>				
<input type="checkbox"/>	<a href="#">med01.skypelabsj.local</a>				
<input type="checkbox"/>	<a href="#">med02.skypelabsj.local</a>				
<input type="checkbox"/>	<a href="#">medpool.skypelabsj.local</a>				

## Cisco Unified Communications Manager Voice Mail Configuration

Configure Voice Mail Pilot:

Navigation: Advanced Features → Voice Mail → Voice Mail Pilot

Add new

Set Voice Mail Pilot Number = 7000 .This is used for the test

Set Description = Unity Connection VM .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 search bar. Below the navigation bar is a menu with options like System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled 'Voice Mail Pilot Configuration' and includes a 'Save' button, a status section showing 'Status: Ready', and a 'Voice Mail Pilot Information' section. The 'Voice Mail Pilot Information' section contains three input fields: 'Voice Mail Pilot Number' with the value '7000', 'Calling Search Space' with the value '< None >', and 'Description' with the value 'Unity Connection VM'. There is also a checkbox labeled 'Make this the default Voice Mail Pilot for the system' which is unchecked. A 'Save' button is located at the bottom of the form.

## Cisco Unified Communications Manager Route Pattern to Skype for Business Extensions

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

Add New

Set Route Pattern\* = 8XXX. This is used to route to the Skype for Business in this test

Set Description = this text is used to identify this Route Pattern

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

Uncheck Provide Outside Dial Tone

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 | Search Documentation | 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

**Pattern Definition**

Route Pattern*	8XXX
Route Partition	< None >
Description	
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-CUCM_Route List <a href="#">(Edit)</a>
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error
Call Classification*	OnNet
External Call Control Profile	< None >
<input type="checkbox"/> Allow Device Override <input 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	

## Route Pattern Configuration for 8xxx (Continued)

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

---

**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"/>

Cisco Unified Communications Manager Route Pattern to invoke Jabber client with Remote Destination configured as Skype for Business Extensions

Set Route Pattern\* = \+.8XXX. This is used to route to the Skype for Business when using the Extend and Connect functionality in this test

Set Description = this text is used to identify this Route Pattern

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

Uncheck Provide Outside Dial Tone

Set Calling Line ID Presentation= Default

Set Calling Name Presentation= Default

Set Connected Line ID Presentation\*= Default

Set Calling Name Presentation\* = Default

Discard Digits = PreDot

All other values are default

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

Save  Delete  Copy  Add New

**Status**  
Status: Ready

**Pattern Definition**

Route Pattern*	\+.8XXX
Route Partition	< None >
Description	To enable routing to SFB via 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-CUCM_Route List <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	

<b>Calling Party Transformations</b>		
<input type="checkbox"/>	Use Calling Party's External Phone Number Mask	
Calling Party Transform Mask	<input type="text"/>	
Prefix Digits (Outgoing Calls)	<input type="text"/>	
Calling Line ID Presentation*	Default	
Calling Name Presentation*	Default	
Calling Party Number Type*	Cisco CallManager	
Calling Party Numbering Plan*	Cisco CallManager	
<b>Connected Party Transformations</b>		
Connected Line ID Presentation*	Default	
Connected Name Presentation*	Default	
<b>Called Party Transformations</b>		
Discard Digits	PreDot	
Called Party Transform Mask	<input type="text"/>	
Prefix Digits (Outgoing Calls)	<input type="text"/>	
Called Party Number Type*	Cisco CallManager	
Called Party Numbering Plan*	Cisco CallManager	
<b>ISDN Network-Specific Facilities Information Element</b>		
Network Service Protocol	-- Not Selected --	
Carrier Identification Code	<input type="text"/>	
Network Service	Service Parameter Name	Service Parameter Value

## Cisco Unified Communications Manager Route Pattern to Skype for Business Call Park range

The Skype for Business Call Park range configured is 100-199 .The following route pattern “1XX” is therefore configured to enable a parked call to be retrieved from Cisco UCM.

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

**Pattern Definition**

Route Pattern*	1XX	
Route Partition	< None >	
Description	call park pickup on SFB	
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-CUCM_Route List	<a href="#">(Edit)</a>
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern	No Error
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		



## Route Pattern Configuration for 1XX (Continued)

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

Calling Name Presentation\* Default v

Calling Party Number Type\* Cisco CallManager v

Calling Party Numbering Plan\* Cisco CallManager v

**Connected Party Transformations**

Connected Line ID Presentation\* Default v

Connected Name Presentation\* Default v

**Called Party Transformations**

Discard Digits < None > v

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type\* Cisco CallManager v

Called Party Numbering Plan\* Cisco CallManager v

**ISDN Network-Specific Facilities Information Element**

Network Service Protocol -- Not Selected -- v

Carrier Identification Code

Network Service	Service Parameter Name	Service Parameter Value
-- Not Selected --	< Not Exist >	

## Cisco Unified Communications Manager Route Pattern to Unity Connection Voice Mail

A route pattern 7000 (which is the voice mail pilot), is configured to reach Unity Connection Voice Mail.

**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

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

Save Delete Copy Add New

**Pattern Definition**

Route Pattern*	7000	
Route Partition	< None >	
Description	Voice mail to unity Connection	
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*	UnityConnection	<a href="#">(Edit)</a>
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error	
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		

**Calling Party Transformations**

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

## Route Pattern Configuration for 7000 (Continued)

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

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
-- Not Selected --	< Not Exist >	

## 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.

### Cisco UCM 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\* = 10.80.10.2; this is the Cisco UCM publisher IP.

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

**Status**  
*i* Status: Ready

**UC Service Information**

**UC Service Type:** CTI  
**Product Type:** CTI  
**Name\*** CTI\_SRV  
 Description  
**Host Name/IP Address\*** 10.80.10.2  
 Port 2748  
**Protocol:** TCP

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.

**Cisco Unified CM Administration** Navigation [Cisco Unified CM Administration](#)

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

**Find and List UC Services**

**Status**  
*i* 3 records found

**UC Service (1 - 3 of 3)** Rows per Page 50

Find UC Service where

<input type="checkbox"/>	Name ^	UC Service Type	Product Type	Host/IP Address	Port	Protocol
<input type="checkbox"/>	<a href="#">CTI_SRV</a>	CTI	CTI	10.80.10.2	2748	TCP
<input type="checkbox"/>	<a href="#">CTI_SUB1</a>	CTI	CTI	10.80.10.3	2748	TCP
<input type="checkbox"/>	<a href="#">IMP_SRV</a>	IM and Presence	Unified CM (IM and Presence)	10.80.10.6		

## Cisco UCM service Profile Configuration

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

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

---

**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](#)

---

**Conferencing Profile**

Primary: 
  
 Secondary: 
  
 Tertiary: 
  
 Server Certificate Verification: 
  
[Credentials source for web conference service\\*](#):

---

**Directory Profile**

Primary: 
  
 Secondary: 
  
 Tertiary:

[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 \(seconds\)\\*](#):

**Cisco UCM service profile Configuration (Continued)**

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

Save
 Delete
 Copy
 Add New

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

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

**IM and Presence Profile**

Primary IMP\_SRV ▼

Secondary <None> ▼

Tertiary <None> ▼

**CTI Profile**

Primary CTI\_SRV ▼

Secondary CTI\_SUB1 ▼

Tertiary <None> ▼

**Video Conference Scheduling Portal Profile**

Primary <None> ▼

Secondary <None> ▼

Tertiary <None> ▼

## 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 \*: 10.80.10.2.Cisco Publisher IP. This is used in this test.

Set Backup CCMCIP Host \*: 10.80.10.3.Cisco Publisher IP. This is used in this test.

Add Users to Profile: user1, user 2 and user3 .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

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\* 10.80.10.2

Backup CCMCIP Host\* 10.80.10.3

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"/>	user1	Jabber	user1	
<input type="checkbox"/>	user2		user2	
<input type="checkbox"/>	user3	SFB	user3	

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

## Cisco UCM – SIP trunk to Cisco IM&Presence Trunk Configuration

**Navigation Path:** Device → Trunk

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

Set Description = this text is used to identify this Trunk Group.

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

Set Media Resource Group List = MRGL. This is used for the test.

Set Destination Address = 10.80.10.6. 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.



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

**Device Information**

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	IMPTrunk
Description	
Device Pool*	G711 Preferred
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	MRGL
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

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

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

ASN.1 ROSE OID Encoding*	No Changes
Packet Capture Mode*	None
Packet Capture Duration	0

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 >

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

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

Save ✖ Delete ↺ Reset + Add New

---

**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"/>

---

**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.

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 UCM SIP Trunk to CUP Configuration (Continued)

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

**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\*:

Calling and Connected Party Info Format\*:

Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS:

Use Device Pool Redirecting Party Transformation CSS

---

**Caller Information**

Caller ID DN:

Caller Name:

Maintain Original Caller ID DN and Caller Name in Identity Headers

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

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

**SIP Information**

**Destination**

Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port	Statu
1*	10.80.10.6		5060	N/A

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\*:  [View Details](#)

DTMF Signaling Method\*:

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

<b>Normalization Script</b>	
Normalization Script	< None >
<input type="checkbox"/> Enable Trace	
<b>Parameter Name</b>	<b>Parameter Value</b>
1	
<input type="button" value="+"/> <input type="button" value="-"/>	
<b>Recording Information</b>	
<input checked="" type="radio"/> None <input type="radio"/> This trunk connects to a recording-enabled gateway <input type="radio"/> This trunk connects to other clusters with recording-enabled gateways	
<b>Geolocation Configuration</b>	
Geolocation	< None >
Geolocation Filter	< None >
<input type="checkbox"/> Send Geolocation Information	
<input type="button" value="Save"/> <input type="button" value="Delete"/> <input type="button" value="Reset"/> <input type="button" value="Add New"/>	

## Cisco UCM end user configuration

Add user to Cisco UCM

**Navigation:** User Management → End user

Set User ID\* = user1. This is used for the test.

Set Last Name = user1. This is used for the test.

Check Home Cluster.

Click the Device Association

Select CT11 from User Device Association screen

**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 ▾

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

**User Information**

User Status: Enabled Local User

User ID\*

Password

Confirm Password

Self-Service User ID

PIN

Confirm PIN

Last name\*

Middle name

First name

Title

Directory URI

Telephone Number

Home Number

Mobile Number

Pager Number

Mail ID

Manager User ID

Department

User Locale

## Cisco UCM end user Configuration (Continued)

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

Digest Credentials

Confirm Digest Credentials

User Profile

---

**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  [View Details](#)

---

**Device Information**

Controlled Devices

Available Profiles

v ^

CTI Controlled Device Profiles

## Cisco UCM end user Configuration (Continued)

The screenshot displays the Cisco Unified CM Administration web interface. At the top, the 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 navigation menu includes System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The current page is "End User Configuration", with a "Related Links" section containing "Back to Find List Users".

Below the navigation, there are action buttons: Save, Delete, and Add New. The "Device Information" section is expanded, showing a table with the following data:

Controlled Devices	Device Association
CTI1	Line Appearance Association for Presence

Below the table, there are sections for "Available Profiles" and "CTI Controlled Device Profiles", each with a list box and up/down arrows. The "Extension Mobility" section is also visible, showing an "Available Profiles" list box.

## Cisco UCM end user Configuration (Continued)

Check Allow Control of Device from CTI  
Select the Primary Extension for this user.5007 is used for this example.

**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 ▾

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

Save [X] Delete [X] Add New [X]

**Extension Mobility**

Available Profiles [ ]

Controlled Profiles [ ]

Default Profile: -- Not Selected --

BLF Presence Group\*: Standard Presence group

SUBSCRIBE Calling Search Space: < None >

Allow Control of Device from CTI

Enable Extension Mobility Cross Cluster

**Directory Number Associations**

Primary Extension: 5007

Check Enable Mobility

**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](#)

Add the following permissions for Standard Users:



- Standard CCM End-Users
- Standard CTI Enabled
- Standard CCMUSER Administration

**Permissions Information**

<b>Groups</b>	<ul style="list-style-type: none"> <li>Standard CCM End Users</li> <li>Standard CTI Enabled</li> </ul>	<input type="button" value="Add to Access Control Group"/> <input type="button" value="Remove from Access Control Group"/>
	<a href="#">View Details</a>	
<b>Roles</b>	<ul style="list-style-type: none"> <li>Standard CCM End Users</li> <li>Standard CCMUSER Administration</li> <li>Standard CTI Enabled</li> </ul>	
	<a href="#">View Details</a>	

\*- indicates required item.

## Remote Destination Configuration

**Navigation:** Device → Remote Destination

Add New

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

Set Destination Number\*= +8004. This is used for the test. [8004 is a Skype for Business extension]

Check Enable Extend and Connect.

Set CTI Remote Device = CTI1

**Remote Destination Information**

Name: JabberRD

Destination Number\*: +8004

Owner User ID\*: user1

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 Phone is pressed.

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

CTI Remote Device\*: CTI1

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

Save Delete Copy Reset Apply Config Add New

Ignore Presentation Indicators (internal calls only)

**Number Presentation Transformation**

**Caller ID For Calls From This Phone**

Calling Party Transformation CSS: < None >

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

**Remote Number**

Calling Party Transformation CSS: < None >

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

**Protocol Specific Information**

BLF Presence Group\*: Standard Presence group

SUBSCRIBE Calling Search Space: < None >

Rerouting Calling Search Space: < None >

**Associated Remote Destinations**

Route calls to all remote destinations when client is not connected

Name	Destination Number
JabberRD	+8004

[Add a New Remote Destination](#)

Two Remote Destinations were configured for this test:

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

Remote

Find Destination where: Name begins with Find Clear Filter

	Name ^	Destination Number	Remote Destination Profile	Dual Mode Phone	IMS-Integrated Mobile	CTI Remote Device	Copy
<input type="checkbox"/>	Jabber8003	+8003				CTIRDuser1	
<input type="checkbox"/>	JabberRD	+8004				CTI1	

## 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. user1 is used in this test.

Set Device Pool: G711 Preferred

Save.

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

**Association**

1	<a href="#">Line [1] - 7009 (no partition)</a>
2	<a href="#">Line [2] - Add a new DN</a>

**Phone Type**

**Product Type:** CTI Remote Device

**Real-time Device Status**

**Registration:** Registered with Cisco Unified Communications Manager clus20pub  
**IPv4 Address:**

**Device Information**

Device is Active  
 Device is not trusted

Active Remote Destination: 8004

Owner User ID*	<input type="text" value="user1"/>
Device Name*	<input type="text" value="CTI1"/>
Description	<input type="text" value="CTI1"/>
Device Pool*	<input type="text" value="G711 Preferred"/> <a href="#">View Details</a>
Calling Search Space	<input type="text" value="&lt; None &gt;"/>
User Hold MOH Audio Source	<input type="text" value="&lt; None &gt;"/>
Network Hold MOH Audio Source	<input type="text" value="&lt; None &gt;"/>
Location*	<input type="text" value="Hub_None"/>
User Locale	<input type="text" value="English, United States"/>
Network Locale	<input type="text" value="United States"/>

Ignore Presentation Indicators (internal calls only)

## Cisco UCM CTI Remote Device Configuration (Continued)

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

**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

[Add a New Remote Destination](#)

**Do Not Disturb**

Do Not Disturb

DND Option\*

Add a DN to this device.

DN 7009 was configured for this test.

## Cisco UCM CTI Remote Device DN Configuration

Directory Number Configuration Related Links: [Configure Device \(CTI1\)](#)

### Directory Number Information

Directory Number*	7009	<input type="checkbox"/> Urgent Priority
Route Partition	< None >	
Description		
Alerting Name	Jabber_Lync8004	
ASCII Alerting Name	Jabber_Lync8004	
External Call Control Profile	< None >	
<input checked="" type="checkbox"/> Allow Control of Device from CTI		
Associated Devices	CT11	<input type="button" value="Edit Device"/> <input type="button" value="Edit Line Appearance"/>
Dissociate Devices		

### Directory Number Settings

Voice Mail Profile	< None >	(Choose <None> to use system default)
Calling Search Space	< None >	
BLF Presence Group*	Standard Presence group	
User Hold MOH Audio Source	< None >	
Network Hold MOH Audio Source	< None >	
<input type="checkbox"/> Reject Anonymous Calls		

**Cisco UCM CTI Remote Device DN Configuration (Continued)**

Directory Number Configuration Related Links: [Configure Device \(CTI1\)](#)

---

**Line 1 on Device CTI1**

Display (Caller ID) 
Display text for a line appearance is intended for displaying text such as a name instead of a directory number for calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.

ASCII Display (Caller ID)

Line Text Label

External Phone Number Mask

Recording Option \*

Recording Profile

Recording Media Source \*

Monitoring Calling Search Space

---

**Multiple Call/Call Waiting Settings on Device CTI1**

Note: The range to select the Max Number of calls is: 1-200

Maximum Number of Calls\*

Busy Trigger\*  (Less than or equal to Max. Calls)

---

**Forwarded Call Information Display on Device CTI1**

Caller Name  
 Caller Number  
 Redirected Number  
 Dialed Number

# Cisco Unity Connection

## 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

The screenshot displays the Cisco Unity Connection Administration web interface. The left sidebar shows a navigation tree with 'Phone System' selected under 'Telephony Integrations'. The main content area is titled 'Phone System' and contains the following configuration options:

- Phone System Name\*:** SFB\_CUCM
- Default TRAP Phone System
- Message Waiting Indicators:**
  - Send Message Counts
  - Use Same Port for Enabling and Disabling MWIs
  - Force All MWIs Off for this Phone System
  - Synchronize All MWIs on This Phone System
- Call Loop Detection by Using DTMF:**
  - Enable for Supervised Transfers
  - Enable for Forwarded Message Notification Calls (by Using DTMF)
  - DTMF Tone To Use: A
  - Guard Time: 2500 milliseconds
- Call Loop Detection by Using Extension:**
  - Enable for Forwarded Message Notification Calls (by Using Extension)
- Phone View Settings:**
  - Enable Phone View
  - CTI Phone Access Username: [text field]
  - CTI Phone Access Password: [text field]
- Outgoing Call Restrictions:**
  - Enable outgoing calls
  - Disable all outgoing calls immediately
  - Disable all outgoing calls between
    - Beginning Time: 12:00 AM
    - Ending Time: 12:00 AM

At the bottom of the form are buttons for 'Save', 'Delete', 'Previous', and 'Next'.

## Cisco Unity Connection 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 Type = SIP

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

Set Ipv4 Address or Host Name = 10.80.10.2 [This is the Cisco UCM publisher IP]



Check Register with SIP server

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

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

**Cisco Unity Connection**

- Users
- Class of Service
- Templates
- Contacts
- Distribution Lists
- Call Management
- Message Storage
- Networking
- Unified Messaging
- Video
- Dial Plan
- System Settings
- Telephony Integrations
  - Phone System
  - Port Group
  - Port
  - Speech Connect Port
  - Trunk
  - Security
- Tools

**New Port Group**

Search Port Groups New Port Group  
Related Links Check Telephony Configuration Go

Port Group Reset Help

Save

**New Port Group**

Phone System SFB\_CUCM  
Create From  Port Group Type SIP  Port Group

**Port Group Description**

Display Name\* SFB\_CUCM-1

Authenticate with SIP Server

Authentication Username  
Authentication Password  
Contact Line Name  
SIP Security Profile 5060  
SIP Transport Protocol TCP

**Primary Server Settings**

IPv4 Address or Host Name 10.80.10.2  
IPv6 Address or Host Name  
Port 5060

Save

Click Save.

### Cisco UCM Unity Connection Port Group Configuration (Continued)

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

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

**Cisco Unity Connection**

- Users
- Class of Service
- Templates
- Contacts
- Distribution Lists
- Call Management
- Message Storage
- Networking
- Unified Messaging
- Video
- Dial Plan
- System Settings
- Telephony Integrations
  - Phone System
  - Port Group**
  - Port
  - Speech Connect Port
  - Trunk
  - Security
- Tools

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

Search Port Groups Port Group Basics (SFB\_CUCM-1)  
Related Links Add Ports Go

Port Group Edit Refresh Help

Save Delete Previous Next

**Status**

The phone system cannot take calls if it has no ports. Use the Related Links to add ports.

**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 5060

SIP Transport Protocol TCP

### Cisco UCM Unity Connection Port Group Configuration (Continued)

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

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

**Cisco Unity Connection**

- Users
- Class of Service
- Templates
- Contacts
- Distribution Lists
- Call Management
- Message Storage
- Networking
- Unified Messaging
- Video
- Dial Plan
- System Settings
- Telephony Integrations
  - Phone System
  - Port Group**
  - Port
  - Speech Connect Port
  - Trunk
  - Security

**Advertised Codec Settings**

Change Advertising

Display Name	Packet Size
G.711 mu-law	20
G.729	20

Change Advertising

**Message Waiting Indicator Settings**

Enable Message Waiting Indicators

Delay between Requests 0 milliseconds

Maximum Concurrent Requests 0

Retries After Successful Attempt 0

Retry Interval After Successful Attempt 5 milliseconds

Save Delete Previous Next

## Cisco Unity Connection Telephony Integration – Add Ports

The screenshot shows the Cisco Unity Connection Administration interface. The main content area is titled 'New Port' and includes a navigation menu on the left. The 'New Phone System Port' section is highlighted with a red box and contains the following configuration options:

- Enabled
- Number of Ports:
- Phone System:
- Port Group:
- Server:

The 'Port Behavior' section includes the following options:

- Answer Calls
- Perform Message Notification
- Send MWI Requests (may also be disabled by the port group)
- Allow TRAP Connections

A status warning at the top of the page reads: "Because it has no port groups, PhoneSystem is not listed in the Phone system field." There are 'Save' buttons at the bottom of both the 'New Phone System Port' and 'Port Behavior' sections.

## Cisco Unity Connection User Configuration

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

Set Alias\* = 4001. This is used for the test.

Set First Name = this text is used to identify this User.

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

Save.

Set Phone System= SFB\_CUCM. This is used in this example.

All other values are default.

## Cisco Unity Connection User Configuration (Continued)

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

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

**Cisco Unity Connection**

- Users
  - Users
  - Import Users
  - Synch Users
- Class of Service
- Templates
- Contacts
- Distribution Lists
- Call Management
- Message Storage
- Networking
- Unified Messaging
- Video
- Dial Plan
- System Settings
- Telephony Integrations
  - Phone System
  - Port Group
  - Port
  - Speech Connect Port
  - Trunk
  - Security
- Tools

**New User** Search Users New User  
Related Links Bulk Edit By CSV Go

User Reset Help

Save

**New User from Template**

User Type User With Mailbox  
Based on Template\* voicemailusertemplate

**Name**

Alias\* 8004  
First Name User1  
Last Name SFB  
Display Name SFB, User1  
SMTP Address @clus20unity.lab.tekvizion.com

**Mailbox Store**

Mailbox Store Unity Messaging Database -1

**Phone**

Extension\* 8004  
Cross-Server Transfer Extension or URI  
Outgoing Fax Number  
Corporate Email Address

Save

## Cisco Unity Connection User Configuration (Continued)

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

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

**Cisco Unity Connection**

- Users
  - Users
  - Import Users
  - Synch Users
- Class of Service
- Templates
- Contacts
- Distribution Lists
- Call Management
- Message Storage
- Networking
- Unified Messaging
- Video
- Dial Plan
- System Settings
- Telephony Integrations
  - Phone System
  - Port Group
  - Port
  - Speech Connect Port
  - Trunk
- Security
- Tools

**Created User(s)**

**Name**

Alias\* 8004

First Name User1

Last Name SFB

Display Name SFB, User1

SMTP Address 8004 @clus20unity.lab.tekvizion.com

Initials

Title

Employee ID

**LDAP Integration Status**

Integrate with LDAP Directory

Do Not Integrate with LDAP Directory

**Phone**

Extension\* 8004

Cross-Server Transfer Extension or URI

Outgoing Fax Number

Outgoing Fax Server --- Not Selected ---

Partition clus20unity Partition

Search Scope clus20unity Search Space

Phone System SFB\_CUCM

Class of Service Voice Mail User COS

Active Schedule Weekdays

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

## Cisco Unity Connection User Configuration (Continued)

All values are default.

The screenshot shows the Cisco Unity Connection Administration interface. The left sidebar contains a navigation tree with the following items: Cisco Unity Connection, Users, Import Users, Synch Users, Class of Service, Class of Service Membership, Templates, User Templates, Call Handler Templates, Contact Templates, Notification Templates, 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, Legacy Links, Branch Management, and HTTP(S) Links. The main content area is titled 'Location' and contains the following fields: Address, Building, City, State, Postal Code, Country (United States), Use System Default Time Zone (checked), Time Zone ((GMT-06:00) America/Chicago), Language (Use System Default Language), English(United States), Department, Manager, Billing ID, Corporate Email Address, Generate SMTP Proxy Address From Corporate Email Address (unchecked), Directory URI, Corporate Phone Number, and buttons for Save, Delete, Previous, and Next. A note at the bottom states: 'Fields marked with an asterisk (\*) are required.'

Similarly, create a user that has a Cisco extension.

## 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

<b>Acronym</b>	<b>Definition</b>
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

Cisco and the Cisco logo are trademarks or registered trademarks of Cisco and/or its affiliates in the U.S. and other countries. To view a list of Cisco trademarks, go to this URL: [www.cisco.com/go/trademarks](http://www.cisco.com/go/trademarks). Third-party trademarks mentioned are the property of their respective owners. The use of the word partner does not imply a partnership relationship between Cisco and any other company. (1110R)