Skype for Business 2015 using SIP trunk (TLS) to Cisco Unified Communications Manager Release 11.5. (1) SU5
# Table of Contents

**Introduction** ........................................................................................................................................... 4

- The following items were tested: .................................................................................................................. 4
- Listed below are the highlights of the integration issues: .............................................................................. 4
- Below are the key results: ............................................................................................................................. 5

**Network Topology** .................................................................................................................................... 5

**Limitations** ................................................................................................................................................ 5

**System Components** ................................................................................................................................... 6

- Hardware Requirements ................................................................................................................................. 6
- Software Requirements ................................................................................................................................... 7

**Features** ....................................................................................................................................................... 7

- Features Supported ....................................................................................................................................... 7
- Features Not Supported or Not Tested ........................................................................................................... 8

**Configuration** ................................................................................................................................................ 8

- Global Trunk Configuration Highlights: ........................................................................................................ 9
- Configuring Sequence and Tasks: .................................................................................................................. 9

**Configuring the Skype for Business** ......................................................................................................... 10

- Add Cisco UCM to Skype for Business Topology ......................................................................................... 10
- Trunk Configuration ...................................................................................................................................... 14
- Route Configuration ...................................................................................................................................... 18
- Voice Policy and PSTN Usage Configuration ............................................................................................... 20
- Dial Plan Configuration ................................................................................................................................. 22
- Call Park Range Configuration .................................................................................................................... 23
- Global Media Bypass Configuration .............................................................................................................. 24
- User Configuration ....................................................................................................................................... 25

**Client Configuration** .................................................................................................................................. 31

**Configuring the Cisco Unified Communications Manager** ........................................................................ 32

- SIP Trunk Security Profile for Trunk to Skype for Business ........................................................................ 32
- SIP Trunk Security Profile for Trunk to Unity Connection .......................................................................... 34
- SIP Profile .................................................................................................................................................... 35
- Media Resource Group ................................................................................................................................. 40
- Media Resource Group List .......................................................................................................................... 44
- Device Pool Configuration ............................................................................................................................. 46
Region Configuration ......................................................................................................................... 49
Normalization Script .......................................................................................................................... 50
SIP Trunk to Skype for Business Configuration .............................................................................. 58
SIP Trunk to Cisco Unity Connection Configuration ......................................................................... 63
Voice Mail Configuration .................................................................................................................... 67
Route Pattern to Skype for Business Extensions ............................................................................. 67
Route Pattern to invoke Jabber client with Remote Destination configured as Skype for Business
Extensions .............................................................................................................................................. 69
Route Pattern to Skype for Business Call Park range ..................................................................... 72
Route Pattern to Unity Connection Voice Mail .................................................................................. 74
Cisco UCM Extent and Connect ......................................................................................................... 76
UC service Configuration .................................................................................................................... 76
service Profile Configuration .............................................................................................................. 78
Cisco Unified CM IM Presence – CCMCIP Profile Configuration ...................................................... 80
SIP trunk to Cisco IM&Presence Trunk Configuration ...................................................................... 81
d user configuration ................................................................................................................................. 86
Remote Destination Configuration ....................................................................................................... 91
Cisco UCM CTI Remote Device Configuration ................................................................................ 92
Cisco Unity Connection ....................................................................................................................... 95
Telephony Integration – Add Phone System ....................................................................................... 95
Telephony Integration – Add Port Group ............................................................................................ 96
Telephony Integration – Add Ports ....................................................................................................... 98
User Configuration ............................................................................................................................... 98
Acronyms ............................................................................................................................................... 101
Introduction

This document describes the steps and configurations necessary for Cisco Unified Communications Manager (Cisco UCM) release 11.5.1 to interoperate with the Skype for Business 2015 on TLS using the following configuration:

The following items were tested:

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

Listed below are the highlights of the integration issues:

- Basic calls work from Cisco UCM to Skype for Business and vice versa. Testing has done with Cisco SIP phones only as SCCP phones do not support 80-bit crypto required by Skype for Business.
- Basic calls work from Cisco UCM to Skype for Business and vice versa using G711 ulaw and alaw
- Caller name and number is not updated correctly for basic calls and in the attended and early-attended transfer scenarios.
- Alerting name updates do not occur on Skype for Business.
- Video calls between the Cisco UCM and Skype for Business users were not tested.
• REFER support Skype for Business needs to be disabled to work Park scenario
• Skype for business does not support to update privacy:id sent by 18x and 2xx message sent from Cisco UCM

Below are the key results:

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

Network Topology

Limitations

These are the known limitations, caveats, or integration issues:
• Skype for Business do not support G729 codec. The trunk is tested with only G711 ulaw and alaw.
• 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 consider Privacy ID, sent by Cisco UCM during 180 Ringing or 200 OK when Connected Name/ID is restricted on Cisco UCM. Subsequently, Skype for Business does not support updating Connected Party display as Private.

• 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.

• Cisco CUM user parks a call with Skype for Business user. When another Skype for Business user retrieves that call, there is one-way audio. The Skype for Business user, who is retrieved from parking, could not hear the other user.

• Skype for Business does not send PAI by default i.e. when restriction is not enabled.

• Skype for Business sends incorrect details DN in history-info as a work around, DN configured in Skype for Business as prefix with "+".

• Cisco UCM Remote Destination 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 Multiple Call Forwarding scenario between Skype for Business Users and Cisco UCM Users, wherein both originator and terminator being Skype for Business Users, originator does not display the Caller ID of terminator.

• When Cisco UCM User completes the conference, the audio is on RTP. Pad-lock symbol on the Cisco phone disappears.

System Components

Hardware Requirements

The following hardware are used:

- Cisco UCS-C240-M3S VMWare Host
- Cisco 8945, 8841, 7841, and 9971 IP phones
Software Requirements

The following software are used:

- Cisco UCSC-C240-M3S VMware vSphere Image Profile: ESXi-5.5.0-1331820-standard
- Cisco Unified Communications Manager release 11.5.1.15900-18
- Cisco Unified Communications Manager IM & P release 11.5.1.15900-33
- Cisco Unity Connection release 11.5.1.15900-18
- Cisco Jabber 12.5.1.2706 Build 277406
- Skype for Business 2015 6.0.9319.534
- Skype for Business Client version : 15.0.5111.1000

Features

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

Features Supported

- CLIP—calling line (number) identification presentation
- CLIR—calling line (number) identification restriction
- CNIP—calling Name identification presentation
- CNIR—calling Name identification restriction
- Attended call transfer
- Early attended call transfer
- CFU—call forwarding unconditional
- CFB—call forwarding busy
- CFNA—call forwarding no answer
- COLP—connected line (number) identification presentation
- COLR—connected line (number) identification restriction
- CONP—connected Name identification presentation
- CONR—connected Name identification restriction
• Hold and resume
• Conference call
• MWI—Message Waiting Indicator (only for Cisco Endpoints)
• Audio Codec Preference List
• Call Park/Pickup (see limitation section)
• Extend and Connect
• Shared Line on Cisco Endpoints

Features Not Supported or Not Tested

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

Configuration

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

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Skype for Business Media Bypass</td>
<td>DISABLED</td>
</tr>
<tr>
<td>Skype for Business Encryption Support</td>
<td>OPTIONAL</td>
</tr>
<tr>
<td>Skype for Business REFER Support</td>
<td>DISABLED</td>
</tr>
<tr>
<td>Cisco UCM SIP Trunk MTP</td>
<td>DISABLED</td>
</tr>
<tr>
<td>Cisco UCM PRACK</td>
<td>ENABLED</td>
</tr>
<tr>
<td>Cisco UCM Early Offer</td>
<td>ENABLED</td>
</tr>
<tr>
<td>Transport type Cisco UCM to Skype for Business</td>
<td>TLS</td>
</tr>
<tr>
<td>Cisco UCM SRTP allowed</td>
<td>Enabled</td>
</tr>
</tbody>
</table>

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 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>–clus21pub.sbsp.local is used in this test.

Click Next.
Check the Enable IPv4 and Use all configured IP addresses radio button
Click Next.
Skype for Business – Add PSTN Gateway (Continued)

Set Trunk Name = FQDN of the Cisco UCM – clus21.pub.sfbsp.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 – 5061 is used for this test
Set SIP Transport Protocol = TLS
Set Associate Mediation Server: Assign this PSTN gateway to the Front End co-located mediation server – fe01.sfbsp.local is used for this test.
Click Finish.
Publish the topology so these new configurations take effect.
Skype for Business – Add PSTN Gateway (Continued)

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 – clus21pub.lab.tekvizion.com is used for the test.

Set **Maximum early dialogs supported** = 20
Set **Encryption support level** = Optional
Set **Refer Support** = None
Uncheck **Enable media bypass**
Check **Centralized media processing**
Uncheck **Enable RTP latching**
Check **Enable forward call history**
Uncheck **Enable forward P-Asserted-Identity data** [Note: this is checked when test scenarios that involve restrict ID need to be executed]
Uncheck **Enable outbound routing failover timer**
Skype for Business – Trunk Configuration (Continued)
**Route Configuration**

**Navigation:** Voice Routing -> Route

Click New

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

Add associated trunks = select the trunk configured earlier – PstnGateway: clus21pub.lab.tekvizion.com
Voice Policy and PSTN Usage Configuration

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

Click New

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

Set Calling Features:

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

Set Associated PSTN usages:

- Click New

  - Set Name: enter a name to identify the PSTN Usage record – CiscoPSTNUsage is used in the test.

  - Set Associated Routes = select the route created earlier= CiscoRoute
Voice Policy - Cisco

Create voice routing test case information

Edit Voice Policy - Cisco

OK   Cancel

Sym: User
Name: Cisco
Description:

Calling Features
- Enable call forwarding
- Enable delegation
- Enable call transfer
- Enable call park
- Enable simultaneous ringing of phones

Associated PSTN Usage

Go to System
Navigation: Voice Routing -> Dial Plan

Default Dial plan used for this topology
Call Park Range Configuration

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

Click New.

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

Set Number Range = 1000 to 1500 is used in the test.

Set FQDN of destination server= select the desired server – fe01.sfbsp.local is used in the test.
Global Media Bypass Configuration

**Navigation:** Network Configuration -> Global

Edit Global Setting –

- Uncheck Enable media bypass

Commit the configuration.
User Configuration

Login to the Skype for Business Active Directory

Navigation: Active Directory Users and Computers ➔ Users

Add a New User
Skype for Business – New User configuration (continued)

Follow the screenshots below to add a new user:

![New Object - User dialog box](image)

- First name: text1
- Last name: 
- Full name: text1
- User logon name: text1
- @SFBSP.LOCAL
- User logon name (pre-Windows 2000): SFBSP\text1
Skype for Business – New User configuration (continued)

New Object - User

Create in: SFBSP.LOCAL/Users

Password: **********
Confirm password: **********

- User must change password at next logon
- User cannot change password
- Password never expires
- Account is disabled

< Back   Next >   Cancel
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.
Set Assign users to a pool: `fe01.sfbsp.local` from drop down menu

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

Set Telephony = Enterprise Voice

Set Line URI: `tel: +2000` is used for the test. This is the DN for the user.

Set Dial plan policy = Automatic (as configured earlier)

Set Voice policy = Cisco (as configured earlier)

Click Enable.
Skype for Business – New User configuration (continued)

- Display name: 
- SIP address: 
- Registrar pool: 
- Telephone: 
  - Enterprise voice: 
- Line SIP: 
- Dial plan policy: 
- Voice policy: 
- Conferencing policy:
Client Configuration

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

Navigation: Settings → Tools → Options → Personal → MyAccount

Set Sign-in-address= enter the sip uri of the user configured in username@domain format. test1@sfbsp.local is used for example.

Click Advanced. Select Manual Configuration.

Set Internal Server Name= Enter the FQDN of the domain (sfbsp.local.local is used for example)
Configuring the Cisco Unified Communications Manager

Cisco Unified Communications Manager Software Version

SIP Trunk Security Profile for Trunk to Skype for Business

**Navigation:** System → Security → SIP trunk security profile
Set Name* = SFB_SECURITYPROFILE. This is used for the test.

Set Device Security mode = Encrypted

Set Incoming Transport Type = TLS

Set Outgoing Transport Type = TLS

Set X.509 Subject Name = fe01.sbsp.local

Set Incoming Port = 5061

Check Accept Presence Subscription

Check Accept out of dialog refer

Check Accept unsolicited notification

Check Accept Replaces header

Check Transmit security status

All other values are default.
SIP Trunk Security Profile for Trunk to Unity Connection

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

Set **Name** = TLS_CUC. This is used for the test.

Set **Device Security mode** = Encrypted

Set **Incoming Transport Type** = TLS

Set **Outgoing Transport Type** = TLS

Set **X.509 Subject Name** = clus21unity.lab.tekvizion.com

Set **Incoming Port** = 5061

Uncheck **Accept Presence Subscription**

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

Check **Accept unsolicited notification**

Check **Accept Replaces header**

Check **Transmit security status**

All other values are default.

Commented [RSM1]: This has been checked in the screenshot
SIP Profile

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

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

Set SIP Rel1XX Options = Send PRACK if 1xx Contains SDP

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

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

All other values are default.
Cisco Unified CM Administration

SIP Profile Configuration

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

SIP Profile Information
- Name: STE_SIP_PROFILE
- Description: STE_SIP_PROFILE
- Default HTTs: [4-7]
- Early Offer for 0.0.0.0 Calls:
- User-Agent and Server Header Information:
- Version in User Agent and Server Header:
- Dial String Interpretation:
- Confidential Access Level Headers:

- Redirect by Application
- Disable Early Media on 180
- Outgoing T.38 INVITE include audio inline
- Offer valid IP and send/receive mode only for T.38 Fax Relay
- Use Fully Qualified Domain Name in SIP Requests
- Assured Services SIP conformance
- Enable External QoS**
Cisco Unified Communications Manager SIP Profile (Continued)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>SDP Information</td>
<td></td>
</tr>
<tr>
<td>SDP Session-level Bandwidth Modifier for Early Offer and Re-invites</td>
<td>TLS and AS</td>
</tr>
<tr>
<td>SDP Transparency Profile</td>
<td></td>
</tr>
<tr>
<td>Accept Audio Codec Preferences in Received Offer</td>
<td>Pass all unknown SDP attributes</td>
</tr>
<tr>
<td>Require SDP Inactive Exchange for Mid-Call Media Change</td>
<td>Default</td>
</tr>
<tr>
<td>Allow RR/AS bandwidth modifier (RFC 3055)</td>
<td></td>
</tr>
<tr>
<td>Parameters used in Phone</td>
<td></td>
</tr>
<tr>
<td>Timer Invite Expires (seconds)</td>
<td>180</td>
</tr>
<tr>
<td>Timer Register Delta (seconds)</td>
<td>5</td>
</tr>
<tr>
<td>Timer Register Expires (seconds)</td>
<td>3400</td>
</tr>
<tr>
<td>Timer T2 (max)</td>
<td>500</td>
</tr>
<tr>
<td>Timer T3 (min)</td>
<td>4000</td>
</tr>
<tr>
<td>Retry INVITE</td>
<td>6</td>
</tr>
<tr>
<td>Retry Non-INVITE</td>
<td>10</td>
</tr>
<tr>
<td>Media Port Ranges</td>
<td></td>
</tr>
<tr>
<td>Common Port Range for Audio and Video</td>
<td></td>
</tr>
<tr>
<td>Separate Port Ranges for Audio and Video</td>
<td></td>
</tr>
<tr>
<td>Start Media Port</td>
<td>16384</td>
</tr>
<tr>
<td>Stop Media Port</td>
<td>22799</td>
</tr>
<tr>
<td>DSCP for Audio Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for Video Calls</td>
<td>Use System Default</td>
</tr>
</tbody>
</table>
Cisco Unified Communications Manager SIP Profile (Continued)

<table>
<thead>
<tr>
<th>SIP Profile Configuration</th>
<th>Related Links: Back To Find/List ▼ Go</th>
</tr>
</thead>
<tbody>
<tr>
<td>DSCP for Audio Portion of Video Calls</td>
<td>Use System Default ▼</td>
</tr>
<tr>
<td>DSCP for TelePresence Calls</td>
<td>Use System Default ▼</td>
</tr>
<tr>
<td>DSCP for Audio Portion of TelePresence Calls</td>
<td>Use System Default ▼</td>
</tr>
<tr>
<td>Call Pickup URI*</td>
<td>n-cisco-serviceur-pickup</td>
</tr>
<tr>
<td>Call Pickup Group Other URI*</td>
<td>n-cisco-serviceur-pickup</td>
</tr>
<tr>
<td>Call Pickup Group URI*</td>
<td>n-cisco-serviceur-pickup</td>
</tr>
<tr>
<td>Meet Me Service URI*</td>
<td>n-cisco-serviceur-meetme</td>
</tr>
<tr>
<td>User Info*</td>
<td>None ▼</td>
</tr>
<tr>
<td>DTMF DB Level*</td>
<td>Nominal ▼</td>
</tr>
<tr>
<td>Call Hold Ring Back*</td>
<td>OFF ▼</td>
</tr>
<tr>
<td>Anonymous Call Block*</td>
<td>OFF ▼</td>
</tr>
<tr>
<td>Caller ID Blocking*</td>
<td>OFF ▼</td>
</tr>
<tr>
<td>Do Not Disturb Control*</td>
<td>User ▼</td>
</tr>
<tr>
<td>Telnet Level for 7940 and 7960*</td>
<td>Disabled ▼</td>
</tr>
<tr>
<td>Resource Priority Namespace</td>
<td>▼</td>
</tr>
<tr>
<td>Timer Keep Alive Expires (seconds)*</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Expires (seconds)*</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Delta (seconds)*</td>
<td>▼</td>
</tr>
<tr>
<td>Maximum Redirects*</td>
<td>70</td>
</tr>
<tr>
<td>OT Hook To First Digit Timer (milliseconds)*</td>
<td>15000</td>
</tr>
<tr>
<td>Call Forward URI*</td>
<td>n-cisco-serviceur-cfwdall</td>
</tr>
</tbody>
</table>
Cisco Unified Communications Manager SIP Profile (Continued)

**Speed Dial (Abbreviated Dial) URL**
- Cisco-service espionage

**Normalization Script**
- Enable Trace

**Incoming Requests FROM URI Settings**
- Caller ID DN
- Caller Name

**Trunk Specific Configuration**
- Route Incoming Request to new Trunk based on
- Resource Priority Namespace List
- SIP Re1xx Options
  - Send PRRACK for all 1xx Messages
Cisco Unified Communications Manager SIP Profile (Continued)

Media Resource Group

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

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

Set all resources in the Selected Media Resources* Box.

All other values are default.
Resource Group for MRG_NoMTP

Set Name*= MRG_NoMTP. This is used for the test.

Set Available Media Resources = MTP_2, MTP_3 and MTP_4

Set other resources in the Selected Media Resources*

All other values are default.
Cisco Unified Communications Manager Media Resource Group Configuration

Commented [RSM3]: The following screenshot is repetitive may be its correct
Media Resource Group List
Navigation Path: Media Resources → Media Resource Group List

Add New

Set Name*= MRGL_SW_MTP. This is used for this test.
Set Available Media Resources = MRG_NO-MTP
Set Selected Media Resource Groups= MRGL_SW_MTP

Add new

Set Name*= MRGL_SW_NoMTP. This is used for the test
Set Available Media Resources MRG_SW_MTP
Set Selected Media Resource Groups= MRGL_SW_NoMTP
Cisco Unified Communications Manager Media Resource Group List Configuration

Find and List Media Resource Group Lists

Find Media Resource Group List where Name begins with: Find Clear Filter

Available Media Resource Groups

- MRGL_SW_MTP

Selected Media Resource Groups

- MRGL_SW_NeHTTP

Status

- Status: Ready

Media Resource Group List Status

Media Resource Group List: MRGL_SW_NeHTTP (used by 3 devices)

Media Resource Group Information

Name: MRGL_SW_NeHTTP

Media Resource Groups for this List

Available Media Resource Groups

- MRGL_SW_MTP

Selected Media Resource Groups

- MRGL_SW_NeHTTP

Save | Delete | Copy | Add New

* Indicates required item.
Device Pool Configuration

Device Pool - G711_Pool is created in this test.

**Navigation Path:** System → Device Pool

Add New.

Set **Device Pool Name** = G711_pool. This is used in the test.

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

Set **Date/Time Group** = CMLocal

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

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

All other values are default.
Cisco Unified Communications Manager Device Pool Configuration (Continued)

<table>
<thead>
<tr>
<th>Device Pool Configuration</th>
<th>Related Links: Back To Find/List ▼ Go</th>
</tr>
</thead>
<tbody>
<tr>
<td>Wireless LAN Profile Group: None ▼</td>
<td>View Details</td>
</tr>
</tbody>
</table>

### Local Route Group Settings
- Standard Local Route Group: None ▼

### Device Mobility Related Information ****
- Device Mobility Calling Search Space: None ▼
- AAR Calling Search Space: None ▼
- AAR Group: None ▼
- Calling Party Transformation CSS: None ▼
- Called Party Transformation CSS: None ▼

### Geolocation Configuration
- Geolocation: None ▼
- Geolocation Filter: None ▼

### Call Routing Information
- **Incoming Calling Party Settings**
  - If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.
  - Clear Prefix Settings ▼ Default Prefix Settings

<table>
<thead>
<tr>
<th>Number Type</th>
<th>Prefix</th>
<th>Strip Digits</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>National Number</td>
<td>Default</td>
<td></td>
<td>None ▼</td>
</tr>
</tbody>
</table>
Cisco Unified Communications Manager Device Pool Configuration (Continued)

<table>
<thead>
<tr>
<th>Device Pool Configuration</th>
<th>Related Links:</th>
<th>Go</th>
</tr>
</thead>
<tbody>
<tr>
<td>International Number</td>
<td>Default</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Unknown Number</td>
<td>Default</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Subscriber Number</td>
<td>Default</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

**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.

<table>
<thead>
<tr>
<th>Clear Prefix Settings</th>
<th>Default Prefix Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number Type</td>
<td>Prefix</td>
</tr>
<tr>
<td>National Number</td>
<td>Default</td>
</tr>
<tr>
<td>International Number</td>
<td>Default</td>
</tr>
<tr>
<td>Unknown Number</td>
<td>Default</td>
</tr>
<tr>
<td>Subscriber Number</td>
<td>Default</td>
</tr>
</tbody>
</table>

**Phone Settings**

| Caller ID For Calls From This Phone | Calling Party Transformation CSS | < None > |

**Connected Party Settings**

| Connected Party Transformation CSS | < None > |

**Redirecting Party Settings**

| Redirecting Party Transformation CSS | < None > |

---

1. * indicates required item.
2. **Number of devices that have to be reset when this device pool is updated. To see a detailed list of these devices and other dependencies, click on Dependency Records.
3. *** Leave the field blank or enter -1 to use the configuration from the enterprise parameters.
4. **** These five parameters will override device level settings when device is roaming and in the same device mobility group.
Region Configuration

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

**Add New**

G711 Region is created in this test.

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

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

**Set Audio Codec Preference List** = G711_Preferred Codec List

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

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

**Set Audio Codec Preference List** = G711_Preferred Codec List. This is used in this example

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

All other values are default
Normalization Script

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

*Add New*

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

Set Content = add script content.

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

Handle Below Scenarios

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

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

3. There is Remote ringback hear issue
   There is issue with PRACK enabled on CUCM and media bypass enabled on Lync. Enabling media bypass on Lync allows the rtp from lync endpoint to flow through CUCM directly instead of flowing through Mediation server. The problem with PRACK enabled is that Lync endpoint is now not able to answer the incoming call. Looking into the traces, it appears that even though Lync sent updated connection information in 183 w/sdp, the call manager is still sending rtp to the mediation server which seems to be incorrect. So In this scenario CUCM expects 180 Ringing not 183 Session progress. So added the Script to convert 183 Session Progress to 180 Ringing.

4. There is incoming Invite from Lync and in From Header there is "user=phone" which cause CUCM to send malformed data in to different layers which cause call failure. So this is work around for that Scenario.

5. Script modify the AS header which from outgoing messages because call forward fails due to bandwidth negotiation value is A=64 is not supported

6. Script convert the History info to diversion Header since call forward to unity Is not supported.

7. Transfer Scenario: Referred-By in Incoming Invite is converted to Diversion Header.

Script Parameters:

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

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)
                end
            b_CT_line = b_CT_line:gsub("64", "1000")
            sdp = sdp:modifyLine("b=CT", "64", b_CT_line)
            msg:setSdp(sdp)
        end
    end
end

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

local function process_outbound_request(msg)
    local method, ruri, ver = msg:getRequestLine()
    if string.find(ruri, "@")
        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
        msg:removeHeader("Require")
    end
    local rseq = msg:getHeader("Rseq")

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

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

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

if rseqPresnt
    msg:addHeader("RSeq",seqVal[1])
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.lynclabsj.local;user=phone>;index=1;ms-retarget-reason=forwarding

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

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
end

local function replaceHistoryHeader(msg)
    local hist = msg:getHeader("History-Info")
    if hist
        then
            convertHIToDiversion(msg)
            local di = msg:getHeader("Diversion")
            if di
                then
                    msg:removeHeader("History-Info")
                end
        end
end

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

local function modifyUserFrom(msg)
    -- get a data from "From" header and replace
    local removeUser = ""
    local value = msg:getHeader("From")
    if value
        then
            value = value:sub(";user=phone", removeUser)
            if value
                then
                    msg:modifyHeader("From", value)
            end
        end
    end
local function process_inbound_request(msg)
    modifyUserFrom(msg)
    replaceHistoryHeader(msg)
    replaceReferredByHeader(msg)
    removecryptoline(msg)
    local sdp = msg:getSdp()
    if sdp
        then
            local tcap = sdp:getLine("a=tcap:", "RTP/SAVP")
            if tcap
                then
                    local a_m_line = sdp:getLine("m=audio", "RTP/AVP")
                    a_m_line = a_m_line:sub("AVP", "SAVP")
                    sdp = sdp:modifyLine("m=audio", "RTP/AVP", a_m_line)
                    sdp=sdp:removeLine("a=crypto:", "2^31")
                    msg:setSdp(sdp)
            end
        end
end

function process_inbound_any_response(msg)
    msg:addHeader("SUPPORTED","X-cisco-srtp-fallback")
    local sdp = msg:getSdp()
if sdp
then
  trace.format("Inbound SDP")
  local tcap = sdp:getLine("a=tcap:", "RTP/SAVP")
  if tcap
    then
      local a_m_line = sdp:getLine("m=audio", "RTP/AVP")
      a_m_line = a_m_line:gsub("AVP", "SAVP")
      sdp = sdp:modifyLine("m=audio", "RTP/AVP", a_m_line)
    end
  trace.format("before removing crypto")
  sdp=sdp:removeLine("a=crypto:","\[2^n\]")
  trace.format("after removing crypto")
  msg:setSdp(sdp)
end
end

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

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

M.outbound_INVITE      = process_outbound_request
M.outbound_ACK         = process_outbound_message
M.outbound_200_INVITE  = process_outbound_message
M.outbound_18X_INVITE  = process_outbound_message
M.inbound_183_INVITE   = process_inbound_progress
M.inbound_INVITE       = process_inbound_request
M.inbound_ANY_ANY      = process_inbound_any_response
M.inbound_ANY          = process_inbound_any_request

return M
SIP Trunk to Skype for Business Configuration

**Navigation:** Device → Trunk

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

Set **Device Pool** = G711_pool. This is used for the test

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

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

Uncheck Media Termination Point Required

Check Run On All Active Unified CM Nodes

Check Redirecting Diversion Header Delivery – Inbound

Set **Destination Address** = 172.16.29.62 [FQDN of Skype for Business Mediation Server] this is used in the test

Set **SIP Trunk Security Profile** = SFB_SECURITY_PROFILE

Set **SIP Profile** = SFB_SIP_PROFILE – Standard SIP Profile

Set **DTMF Signaling Method** = No Preference

Set **Normalization Script** = lync_interop_updated

All other values are default.
### Device Information

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Product</td>
<td>SIP</td>
</tr>
<tr>
<td>Device Protocol</td>
<td>SIP</td>
</tr>
<tr>
<td>Trunk Service Type</td>
<td>Name (Default)</td>
</tr>
<tr>
<td>Description</td>
<td></td>
</tr>
<tr>
<td>Device Port</td>
<td>Default</td>
</tr>
<tr>
<td>Common Device Configuration</td>
<td></td>
</tr>
<tr>
<td>Call Classifier</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>Media_SIP_HotTP</td>
</tr>
<tr>
<td>Location</td>
<td>Hub_None</td>
</tr>
<tr>
<td>AAI Group</td>
<td></td>
</tr>
<tr>
<td>Tunneld Protocol</td>
<td>Name</td>
</tr>
<tr>
<td>QOS Variant</td>
<td>No Changes</td>
</tr>
<tr>
<td>ATM L ROSE PID Encoding</td>
<td>No Changes</td>
</tr>
<tr>
<td>Packet Capture Mode</td>
<td>Name</td>
</tr>
<tr>
<td>Packet Capture Duration</td>
<td></td>
</tr>
</tbody>
</table>
Cisco Unified Communications Manager SIP Trunk to Skype for Business Configuration (Continued)
Cisco Unified Communications Manager SIP Trunk to Skype for Business Configuration (Continued)

<table>
<thead>
<tr>
<th>Parameter Name</th>
<th>Parameter Value</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Recording Information</strong></td>
<td></td>
</tr>
<tr>
<td>- None</td>
<td></td>
</tr>
<tr>
<td>- This trunk connects to a recording-enabled gateway</td>
<td></td>
</tr>
<tr>
<td>- This trunk connects to other clusters with recording-enabled gateways</td>
<td></td>
</tr>
<tr>
<td><strong>Geolocation Configuration</strong></td>
<td></td>
</tr>
<tr>
<td>Geolocation: &lt; Name &gt;</td>
<td></td>
</tr>
<tr>
<td>Geolocation Filter: &lt; name &gt;</td>
<td></td>
</tr>
<tr>
<td>Send Geolocation Information</td>
<td></td>
</tr>
</tbody>
</table>

- * indicates required item.
- **- Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.
SIP Trunk to Cisco Unity Connection Configuration

**Navigation**: Device → Trunk

Set **Device Name** = Clus21unity. This is used for the test.

Set **Device Pool** = Default

Check Run On All Active Unified CM Nodes

Check Redirecting Diversion Header Delivery – Inbound

Check Redirecting Diversion Header Delivery – Outbound

Set **Destination Address** = 10.80.11.6. This is used for the test.

Set **SIP Trunk Security Profile** = TLS_CUC

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

Set **DTMF Signaling Method** = No Preference

All other values are default
Cisco Unified Communications Manager SIP Trunk to Cisco Unity Connection Configuration (Continued)
Cisco Unified Communications Manager SIP Trunk to Cisco Unity Connection Configuration (Continued)
Voice Mail Configuration

Configure Voice Mail Pilot:

Navigation: Advanced Features → Voice Mail → Voice Mail Pilot

Add new

Set Voice Mail Pilot Number = 4900. This is used for the test

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

Route Pattern to Skype for Business Extensions

Navigation: Call Routing → Route/Hunt → Route Pattern

Add New

Set Route Pattern* = 2XXX. This is used to route to the Skype for Business in this test

Set Gateway/Route List* = SFB. 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

Commented [RSM4]: Checked in the screenshot
Set Calling Name Presentation* = Default

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

All other values are default.
Route Pattern Configuration for 8xx (Continued)

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

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

- **Set Route Pattern** = \+2XXX. This is used to route to the Skype for Business when using the Extend and Connect functionality in this test
- **Set Gateway/Route List** = SFB. 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
Set Discard Digits = PreDot

All other values are default

Commented [RSM5]: Not used in the screenshot
Cisco Unified Communications Manager Route pattern Configuration (Continued)

<table>
<thead>
<tr>
<th>Route Pattern Configuration</th>
<th>Related Links</th>
<th>Back To Find/List</th>
<th>Go</th>
</tr>
</thead>
</table>

**Calling Party Transformations**
- Use Calling Party's External Phone Number Mask
- Calling Party Transform Mask
- Prefix Digits (Outgoing Calls)
- Calling Line ID Presentation
  - Default
- Calling Name Presentation
  - Default
- Calling Party Number Type
  - Cisco CallManager
- Calling Party Numbering Plan
  - Cisco CallManager

**Connected Party Transformations**
- Connected Line ID Presentation
  - Default
- Connected Name Presentation
  - Default

**Called Party Transformations**
- Discard Digits
  - < None >
- Called Party Transform Mask
- Prefix Digits (Outgoing Calls)
- Called Party Number Type
  - Cisco CallManager
- Called Party Numbering Plan
  - Cisco CallManager

**ISDN Network-Specific Facilities Information Element**
- Network Service Protocol
  - Not Selected
- Carrier Identification Code

<table>
<thead>
<tr>
<th>Network Service</th>
<th>Service Parameter Name</th>
<th>Service Parameter Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Not Selected</td>
<td>Not Selected</td>
<td>Not Exist</td>
</tr>
</tbody>
</table>

Save  Delete  Copy  Add New

* indicates required item.
Route Pattern to Skype for Business Call Park range

The Skype for Business Call Park range configured is 1000-1999. The following route pattern “1XX” is therefore configured to enable a parked call to be retrieved from Cisco UCM.

![Route Pattern Configuration](image-url)
Cisco Unified Communications Manager Call Park Range Configuration (Continued)

<table>
<thead>
<tr>
<th>Route Pattern Configuration</th>
<th>Related Links</th>
<th>Back To Find/List</th>
<th>Go</th>
</tr>
</thead>
</table>

**Calling Party Transformations**
- Use Calling Party's External Phone Number Mask
- Calling Party Transform Mask
- Prefix Digits (Outgoing Calls)
- Calling Line ID Presentation
  - Default
- Calling Name Presentation
  - Default
- Calling Party Number Type
  - Cisco CallManager
- Calling Party Numbering Plan
  - Cisco CallManager

**Connected Party Transformations**
- Connected Line ID Presentation
  - Default
- Connected Name Presentation
  - Default

**Called Party Transformations**
- Discard Digits
  - < None >
- Called Party Transform Mask
- Prefix Digits (Outgoing Calls)
- Called Party Number Type
  - Cisco CallManager
- Called Party Numbering Plan
  - Cisco CallManager

**ISDN Network-Specific Facilities Information Element**
- Network Service Protocol
  - < Not Selected >
- Carrier Identification Code

<table>
<thead>
<tr>
<th>Network Service</th>
<th>Service Parameter Name</th>
<th>Service Parameter Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>&lt; Not Selected &gt;</td>
<td>&lt; Not Exist &gt;</td>
<td></td>
</tr>
</tbody>
</table>

* indicates required item.
Route Pattern to Unity Connection Voice Mail

A route pattern 4900 (which is the voice mail pilot), is configured to reach Unity Connection Voice Mail.
Cisco Unified Communications Manager Route Pattern Configuration (Continued)

<table>
<thead>
<tr>
<th>Route Pattern Configuration</th>
<th>Related Links</th>
<th>Back To Find/List</th>
<th>Go</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

- **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**
  - Distinct Digits
  - Distinct Digits (Outgoing Calls)
  - Caller ID Transform Mask
  - Caller ID Numbering Plan

- **ISDN Network-Specific Facilities Information Element**
  - Network Service Protocol
  - Carrier Identification Code

<table>
<thead>
<tr>
<th>Network Service</th>
<th>Service Parameter Name</th>
<th>Service Parameter Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>-- Not Selected --</td>
<td>-- Not Selected --</td>
<td>-- Not Selected --</td>
</tr>
</tbody>
</table>

* indicates required item.
Cisco UCM Extend and Connect

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

UC service Configuration

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

Add New

Select Service Type as CTI

Set **Name** = IMPTrunk

Set **Host Name/IP Address** = 10.80.11.2; this is the Cisco UCM publisher IP.

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.
Find and List UC Services

5 records found

<table>
<thead>
<tr>
<th>UC Service</th>
<th>(1 – 5 of 5)</th>
<th>Rows per Page 50</th>
</tr>
</thead>
<tbody>
<tr>
<td>Find UC Service where Name \begin{tabular}{l} CTI \end{tabular}</td>
<td>\begin{tabular}{l} \text{begins with} \end{tabular}</td>
<td>Find Clear Filter</td>
</tr>
<tr>
<td>CTI SEV</td>
<td>CTI</td>
<td>CTI</td>
</tr>
<tr>
<td>CTI SUB1</td>
<td>CTI</td>
<td>CTI</td>
</tr>
<tr>
<td>IM. PB</td>
<td>IM and Presence</td>
<td>Unified CM (IM and Presence)</td>
</tr>
</tbody>
</table>
service Profile Configuration

**Navigation:** User Management → User setting → Service Profile
Cisco UCM service profile Configuration (Continued)
Cisco UCM service profile Configuration (Continued)

- **Search Timeout (seconds)**: 5
- **Ring Filter (Only used for Advance Directory)**: 
- **Predicitive Search Filter (Only used for Advance Directory)**: 
- **Allow Jabber to Search and Add Security Groups**: 

**IM and Presence Profile**
- **Primary**: IM 1E
- **Secondary**: <Name> 
- **Tertiary**: <Name> 

**CTI Profile**
- **Primary**: CTI-5ky
- **Secondary**: <Name> 
- **Tertiary**: <Name> 

**Video Conference Scheduling Portal Profile**
- **Primary**: <Name> 
- **Secondary**: <Name> 
- **Tertiary**: <Name>

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

Set **Backup CCMCIP Host**: 10.80.11.3. Cisco Publisher IP. This is used in this test.

Add Users to Profile: jabber1, test2 and test3. This is used in this test.
SIP trunk to Cisco IM&Presence Trunk Configuration

Navigation Path: Device ⇒ Trunk

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

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

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

Set Destination Address = 10.80.11.5. This is used in this example.

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

Set SIP Profile* = Standard SIP Profile.

Set DTMF Signaling Method* = No Preference.

All other values are default.
<table>
<thead>
<tr>
<th>Device Information</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Product</strong></td>
<td>SIP Trunk</td>
</tr>
<tr>
<td><strong>Device Protocol</strong></td>
<td>SIP</td>
</tr>
<tr>
<td><strong>Trunk Service Type</strong></td>
<td>None (Default)</td>
</tr>
<tr>
<td><strong>Device Name</strong></td>
<td>MG-Trunk</td>
</tr>
<tr>
<td><strong>Device Pool</strong></td>
<td>Default</td>
</tr>
<tr>
<td><strong>Common Device Configuration</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>Call Classification</strong></td>
<td>Use System Default</td>
</tr>
<tr>
<td><strong>Media Resource Group List</strong></td>
<td>MGSL_Du_iSoftIP</td>
</tr>
<tr>
<td><strong>Location</strong></td>
<td>Hub_1007</td>
</tr>
<tr>
<td><strong>AAR Group</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>Tunneled Protocol</strong></td>
<td>None</td>
</tr>
<tr>
<td><strong>QSIG Variant</strong></td>
<td>No Changes</td>
</tr>
<tr>
<td><strong>ASN.1 KRO D-ID Encoding</strong></td>
<td>No Changes</td>
</tr>
<tr>
<td><strong>Packets Capture Mode</strong></td>
<td>None</td>
</tr>
<tr>
<td><strong>Packets Capture Duration</strong></td>
<td>0</td>
</tr>
</tbody>
</table>
Cisco Unified Communications Manager SIP Trunk to CUP Configuration (Continued)

### Trunk Configuration

<table>
<thead>
<tr>
<th>Option</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Media Termination Point Required</td>
<td></td>
</tr>
<tr>
<td>Retry Video Call as Audio</td>
<td></td>
</tr>
<tr>
<td>Path Replacement Support</td>
<td></td>
</tr>
<tr>
<td>Transmit UTF-8 for Calling Party Name</td>
<td></td>
</tr>
<tr>
<td>Transmit UTF-8 Names in ONS APDU</td>
<td></td>
</tr>
<tr>
<td>Unattended Port</td>
<td></td>
</tr>
<tr>
<td>SRTP Allowed</td>
<td>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 either using both SRTP and TLS.</td>
</tr>
<tr>
<td>Route Class Signaling Enabled*</td>
<td>Default</td>
</tr>
<tr>
<td>Use Trusted Relay Point*</td>
<td>Default</td>
</tr>
<tr>
<td>PSTN Access</td>
<td></td>
</tr>
<tr>
<td>Run On All Active Unified CM Nodes</td>
<td></td>
</tr>
</tbody>
</table>

### Intercompany Media Engine (IME)

<table>
<thead>
<tr>
<th>Transformation Profile</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>1.104</td>
<td>None</td>
</tr>
</tbody>
</table>

### MLPF and Confidential Access Level Information

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>MLPF Domain</td>
<td>None</td>
</tr>
<tr>
<td>Confidential Access Mode</td>
<td>None</td>
</tr>
<tr>
<td>Confidential Access Level</td>
<td>None</td>
</tr>
</tbody>
</table>

### Call Routing Information

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Remote Party Id</td>
<td></td>
</tr>
</tbody>
</table>
Cisco UCM SIP Trunk to CUP Configuration (Continued)

### 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.

<table>
<thead>
<tr>
<th>Number Type</th>
<th>Prefix</th>
<th>Strip Digits</th>
<th>Calling Search Space</th>
<th>Use Device Pool CSS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Incoming</td>
<td>Default</td>
<td></td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
</tbody>
</table>

### 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
- Use Device Pool Calling Party Transformation CSS
- Calling Party Selection
- Calling Line ID Presentation Default
- Calling Name Presentation Default
end user configuration

Add user to Cisco UCM

**Navigation:** User Management → End user

Set **User ID** = *jabber1*. This is used for the test.

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

Check Home Cluster.

Click the Device Association

Select C711 from User Device Association screen
Cisco UCM end user Configuration (Continued)

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Manager User ID</td>
<td></td>
</tr>
<tr>
<td>Department</td>
<td></td>
</tr>
<tr>
<td>User Locale</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Associated PC/Site Code</td>
<td></td>
</tr>
<tr>
<td>Digest Credentials</td>
<td></td>
</tr>
<tr>
<td>Confirm Digest Credentials</td>
<td></td>
</tr>
<tr>
<td>User Profile</td>
<td>Use System Default (Standard, Factory Default)</td>
</tr>
<tr>
<td>User Rank</td>
<td>1-Default User Rank</td>
</tr>
<tr>
<td>Service Settings</td>
<td></td>
</tr>
<tr>
<td>Home Cluster</td>
<td></td>
</tr>
<tr>
<td>Enable User for Unified IM and Presence</td>
<td>Configure IM and Presence in the associated UC Service Profile</td>
</tr>
<tr>
<td>UC Service Profile</td>
<td>Jabber_SVC_Profile</td>
</tr>
<tr>
<td>Device Information</td>
<td></td>
</tr>
<tr>
<td>Controlled Devices</td>
<td>CTI01</td>
</tr>
<tr>
<td>Available Profiles</td>
<td></td>
</tr>
</tbody>
</table>
Cisco UCM end user Configuration (Continued)

Check Allow Control of Device from CTI
Select the Primary Extension for this user. 4007 is used for this example.
Check Enable Mobility
Add the following permissions for Standard Users:

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

Commented [RSM6]: Allow control of all devices in the screenshot
Remote Destination Configuration

**Navigation:** Device ➔ Remote Destination

Add New

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

Set Destination Number* = +2001. This is used for the test. [8004 is a Skype for Business extension]

Check Enable Extend and Connect.

Set CTI Remote Device = CTIRD1

The CTI Remote Device configuration is updated with the remote destination:
Two Remote Destinations were configured for this test:

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**. Jabber1 is used in this test.

Set **Device Pool**: Default

Save.
Add a DN to this device.

DN +2001 was configured for this test.
Cisco Unity Connection
Telephony Integration – Add Phone System

Navigation: Telephony Integrations → Phone system

Add New

Set Phone System Name* = SFB_TLS. This Name used for this test

---

Check Use Same Port for Enabling and Disabling MWIs

---

Field marked with an asterisk (*) are required.
Telephony Integration – Add Port Group

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

Go

Set **Phone System** = SFB_TLS

Set **Create From** = Port group

Set **Type** = SIP

Set Display Name* = SFB_TLS-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

Set **Security Profile** = 5061/TLS

Set **Security Mode** = Encrypted

Check Secure RTP

Check Enable Message Waiting Indicators
Click Save.
Telephony Integration – Add Ports

Set **Number of Ports** = 10

Set **Phone System** = SFB_TLS

Set **Port Group** = SFB_TLS-1

Set **Server** = clus21unity.lab.tekvizion.com (which is the FQDN of Cisco Unity Server)

---

User Configuration

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

Set **Alias** = +2002 (This is used for the test)

Set **First Name** = +2002 (This is used to identify the User)

Set **Extension** = +2002 (This is user’s extension number)

Set **Partition** = clus21UnityPartition

Set **Phone System** = SFB_TLS
Cisco Unity Connection User Configuration (Continued)

All values are default.

Similarly, create a user that has a Cisco extension.
# Acronyms

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