Skype for Business 2015 using SIP trunk (TCP) to Cisco Unified Communications Manager Release 11.5. (1) SU5
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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.

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 works from Cisco UCM to Skype for Business and vice versa using G711 ulaw and alaw
- Skype for Business sends ‘488 Gateway is not in connected state’ for delayed offer during Call Hold, this causes call disconnection from Cisco UCM. Forcing the MTP on Cisco trunk helps resolved this issue but there were no re-INVITE sent during Hold/Resume scenario.
- Caller Name and Number is not updated correctly for 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 in Skype for Business needs to be disabled for Call Park scenario
• Skype for Business does not support “privacy:id” in 18x and 2xx message sent to and 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 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 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

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

• 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

• Calling restriction between Skype for Business clients is not supported

**System Components**

**Hardware Requirements**

The following hardware is used:

- Cisco UCS-C240-M3S VMWare Host
- Cisco 8945,8841 ,7841, 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 11.5.1.15900-18
- Cisco Unified Communications Manager IM & Presence Service 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 Server 2015 version 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 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

- G729 voice codec
- Call completion (callback, automatic callback)
- Shared Line on Skype for Business
- Message Waiting Indicator on Skype for Business Endpoints
- Blind transfer
- Video calls
- Scenarios that required third PBXs.
- Scenarios involving Non-SIP interfaces.
- Alerting Name in Skype for Business
- Connected party restriction send and receive 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:

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<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
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<td>Skype for Business Media Bypass</td>
<td>ENABLED</td>
</tr>
<tr>
<td>Skype for Business Encryption Support</td>
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<tr>
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<td>Cisco UCM Early Offer</td>
<td>ENABLED</td>
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<tr>
<td>Transport type Cisco UCM to Skype for Business</td>
<td>TCP</td>
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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 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”

![Topology Builder Screenshot](image)

Set FQDN = `<FQDN or IP of the Cisco UCM>` → 10.80.11.2 is used in this test.

Click Next.

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

![Define New IP/PSTN Gateway](image)

Define the PSTN Gateway FQDN

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

FQDN: `*`  

10.80.11.2

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

Click Next.
Set **Trunk Name** = FQDN of the Cisco UCM – **10.80.11.2.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 the 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 – 10.80.11.2 is used for the test.

Set Maximum early dialogs supported = 20

Set Encryption support level = Optional

Set Refer Support = None

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
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: 10.80.11.2**
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

Dial Plan Configuration

**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 –

- Check Enable media bypass
- Check Always 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)

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

Skype for Business – New User configuration (continued)

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

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.
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@sbsp.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**: Non-Secure Profile for SFB. 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.
SIP Trunk Security Profile for Trunk to Unity Connection

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

Set **Name** = `Unity_Connection_Trunk_Security_Profile`. This is used for the test.

Set **Device Security mode** = Non Secure

Set **Incoming Transport Type** = TCP+UDP

Set **Outgoing Transport Type** = UDP

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

Check **Accept unsolicited notification**

Check **Accept Replaces header**

All other values are default.
SIP Profile

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

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

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

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

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

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

All other values are default.
SIP Profile Configuration

Status: Ready

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

SIP Profile Information

- Name: SFB_SIP_PROFILE
- Description: SFB_SIP_PROFILE
- Default MTP Telephony Event Payload Type: 101
- Early Offer for G.729 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
- 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)

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<td>SDP Transparency Profile</td>
<td>Pass all unknown SDP attributes ▼</td>
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<td>Accept Audio Codec Preferences in Received Offer *</td>
<td>Default ▼</td>
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<td>Timer Register Delta (seconds) *</td>
<td>5</td>
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<td>Timer Register Expires (seconds) *</td>
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<td>32766</td>
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<tr>
<td>DSCP for Audio Calls</td>
<td>Use System Default ▼</td>
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<tr>
<td>DSCP for Video Calls</td>
<td>Use System Default ▼</td>
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<th>SIP Profile Configuration</th>
<th>Related Links: Back To Find/List</th>
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<td>DSCP for Audio Portion of Video Calls</td>
<td>Use System Default</td>
<td></td>
</tr>
<tr>
<td>DSCP for TelePresence Calls</td>
<td>Use System Default</td>
<td></td>
</tr>
<tr>
<td>DSCP for Audio Portion of TelePresence Calls</td>
<td>Use System Default</td>
<td></td>
</tr>
<tr>
<td>Call Pickup URI</td>
<td>x-cisco-serviceuri-pickup</td>
<td></td>
</tr>
<tr>
<td>Call Pickup Group Other URI</td>
<td>x-cisco-serviceuri-opickup</td>
<td></td>
</tr>
<tr>
<td>Call Pickup Group URI</td>
<td>x-cisco-serviceuri-gpickup</td>
<td></td>
</tr>
<tr>
<td>Meet Me Service URI</td>
<td>x-cisco-serviceuri-meetme</td>
<td></td>
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<tr>
<td>User Info</td>
<td>None</td>
<td></td>
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<tr>
<td>DTMF DB Level</td>
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<td></td>
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<tr>
<td>Call Hold Ring Back</td>
<td>Off</td>
<td></td>
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<tr>
<td>Anonymous Call Block</td>
<td>Off</td>
<td></td>
</tr>
<tr>
<td>Caller ID Blocking</td>
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<td></td>
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<tr>
<td>Do Not Disturb Control</td>
<td>User</td>
<td></td>
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<tr>
<td>Telnet Level for 7940 and 7960</td>
<td>Disabled</td>
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<tr>
<td>Resource Priority Namespace</td>
<td>&lt; None &gt;</td>
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<td>120</td>
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<tr>
<td>Timer Subscribe Expires (seconds)</td>
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<td>Timer Subscribe Delta (seconds)</td>
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<tr>
<td>Maximum Redirects</td>
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<td></td>
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<tr>
<td>Off Hook To First Digit Timer (milliseconds)</td>
<td>15000</td>
<td></td>
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<tr>
<td>Call Forward URI</td>
<td>x-cisco-serviceuri-cfwdail</td>
<td></td>
</tr>
</tbody>
</table>
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 **Description = With SW_MTP** 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 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_noMTP**. This is used for the test

Set **Available Media Resources** = **MRG_SW_MTP**

Set **Selected Media Resource Groups** = **MRG_SW_NoMTP**
Cisco Unified Communications Manager Media Resource Group List Configuration

- Status
  - Status: Ready

- Media Resource Group List Status
  - Media Resource Group List: MRGL_SW_NoMTP (used by 3 devices)

- Media Resource Group List Information
  - Name: MRGL_SW_NoMTP

- Media Resources for this List
  - Available Media Resource Groups: MRG_SW_MTP
  - Selected Media Resource Groups: MRG_SW_NoMTP

- Status
  - 5 records found

Find and List Media Resource Group Lists

<table>
<thead>
<tr>
<th>Name</th>
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<tbody>
<tr>
<td>MRGL SW MTP</td>
<td></td>
</tr>
<tr>
<td>MRGL SW_NoMTP</td>
<td></td>
</tr>
</tbody>
</table>
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_SW_MTP. This is used in this example.

All other values are default.

![Cisco Unified Communications Manager Device Pool Configuration](image)

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

### Wireless LAN Profile Group
- **Standard Local Route Group**
  - Choose a local route group.

### Device Mobility Related Information
- **Device Mobility Calling Search Space**
  - Select a search space.
- **AAR Calling Search Space**
  - Select a search space.
- **AAR Group**
  - Select a group.
- **Calling Party Transformation CSS**
  - Select a CSS.
- **Called Party Transformation CSS**
  - Select a CSS.

### Geolocation Configuration
- **Geolocation**
  - Select a geolocation.
- **Geolocation Filter**
  - Select a filter.

### Call Routing Information
#### Incoming Calling Party Settings
- A table showing prefixes, strip digits, and calling search spaces.
- Options to clear or set default prefixes.

---

Page 42 of 99
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->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 does not support History-Info.
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 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 bandwith negotiation value is A=64 is not supported

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

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

Script Parameters:

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

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All rights reserved.

M = {}
M.allowHeaders = {"History-Info"}

trace.enable()

local function getDisplayName (i_header)
    local position_of_uri=string.find(i_header, ">")
    if position_of_uri <= 2
        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
    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")
end
 functions

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)
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 rseqPresent = 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 rseqPresent 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.lynclabsj.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
msg:addHeader("Diversion", diversion)
end

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

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

local function modifyUserFrom(msg)
    -- get a data from "From" header and replace
    local removeUser= ""
    local value = msg:getHeader("From")
    if value then
        value = value:gsub(";user=phone", removeUser)
        if value then
            msg:modifyHeader("From", value)
        end
    end
end
local function process_inbound_request(msg)
    modifyUserFrom(msg)
    replaceHistoryHeader(msg)
    replaceReferredByHeader(msg)
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

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 Description = this text is used to identify this Trunk Group
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_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 = 172.16.29.62 [FQDN of Skype for Business Front End] This is used in the test
Set SIP Trunk Security Profile* = SFB_SECURITY_PROFILE
Set SIP Profile* = SFB – Standard SIP Profile
Set DTMF Signaling Method* = No Preference
Set Normalization Script = CiscoScriptForSFB

All other values are default.
SIP Trunk to Skype for Business Configuration (Continued)
SIP Trunk to Skype for Business Configuration (Continued)

[Image of Cisco Unified CM Administration interface showing Trunk Configuration with sections for Incoming Calling Party Settings, Incoming Called Party Settings, and Connected Party Settings with configuration fields for Prefix, Strip Digits, Calling Search Space, and Use Device Pool CSS.]
SIP Trunk to Skype for Business Configuration (Continued)

### Outbound Calls
- **Called Party Transformation CSS**: <None>
- **Use Device Pool Called Party Transformation CSS**: V
- **Calling Party Transformation CSS**: <None>
- **Use Device Pool Calling Party Transformation CSS**: V
- **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**: V
- **Redirecting Party Transformation CSS**: <None>
- **Use Device Pool Redirecting Party Transformation CSS**: V

### 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**: 172.16.29.62
- **Destination Address IPv6**: 
- **Destination Port**: 5060

#### MTP Preferred Originating Codec
- **711ulaw**: 

#### BLF Presence Group
- **Standard Presence group**: 

#### SIP Trunk Security Profile
- **Non-Secure Profile For CTL**: 

#### Redirecting Calling Search Space
- **<None>**: 

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

#### SUBSCRIBE Calling Search Space
- **<None>**: 

#### SIP Profile
- **SFB_SIP_PROFILE**: View Details

#### DTMF Signaling Method
- **No Preference**: 

### Normalization Script
- **Normalization Script**: CiscoScriptForSFB

#### Enable Trace
- **Parameter Name**: 
- **Parameter Value**: 

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SIP Trunk to Skype for Business Configuration (Continued)

<table>
<thead>
<tr>
<th>Parameter Name</th>
<th>Parameter Value</th>
</tr>
</thead>
</table>

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

**GeoLocation Configuration**
- Geolocation: <None>
- Geolocation Filter: <None>

* indicates required item.
** indicates 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 **Description** = Clus21unity. This text is used to identify this Trunk Group.

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*** = Unity_Connection_Trunk_Security_Profile

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

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

All other values are default
### Device Information

- **Product:**
- **Device Protocol:** SIP
- **Trunk Service Type:**
- **Device Name:** Clus2Unity
- **Description:**
- **Device Pool:** Default
- **Common Device Configuration:**
- **Call Classification:** Use System Default
- **Media Resource Group List:**
- **Location:** Hub_None
- **AAR Group:**
- **Tunneled Protocol:** None
- **QSIG Variant:**
- **ASN.1 ROSE OID Encoding:**
- **Packet Capture Mode:** None
- **Packet Capture Duration:** 0
### SIP Trunk to Cisco Unity Connection Configuration (Continued)

#### Trunk Configuration

<table>
<thead>
<tr>
<th>Option</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Packet Capture Mode</td>
<td>None</td>
</tr>
<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 QSIG APDU</td>
<td></td>
</tr>
<tr>
<td>Unattended Port</td>
<td></td>
</tr>
<tr>
<td>SRTP Allowed</td>
<td>When 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>Run On All Active Unified CM Nodes</td>
</tr>
</tbody>
</table>

#### Intercompany Media Engine (IME)

<table>
<thead>
<tr>
<th>Option</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>E.164 Transformation Profile</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

#### MLPP and Confidential Access Level Information

<table>
<thead>
<tr>
<th>Option</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>MLPP Domain</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Confidential Access Mode</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Confidential Access Level</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>
SIP Trunk to Cisco Unity Connection Configuration (Continued)
Cisco Unified Communications Manager SIP Trunk to Cisco Unity Connection Configuration (Continued)
Route Pattern Configuration to Unity 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
Cisco Unified Communications Manager Route Pattern to 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

Set **Calling Line ID Presentation** = Default

Set **Calling Name Presentation** = Default

Set **Connected Line ID Presentation** = Default

Set **Calling Name Presentation** = Default

Set **Prefix Digits outgoing calls** = +

All other values are default
Route Pattern Configuration for 2xxx (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

All other values are default
Route Pattern to Skype for Business Call Park range

The Skype for Business Call Park range configured is 1000-1999. The following route pattern “1XXX” is therefore configured to enable a parked call to be retrieved from Cisco UCM.
Cisco Unified Communications Manager Call Park Range Configuration (Continued)

Cisco Unified Communications Manager 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: Back To Find/List ▼</th>
<th>Go</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Calling Party Transformations</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Use Calling Party's External Phone Number Mask</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Calling Party Transform Mask</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Calling Line ID Presentation</td>
<td>Default</td>
<td></td>
</tr>
<tr>
<td>Calling Name Presentation</td>
<td>Default</td>
<td></td>
</tr>
<tr>
<td>Calling Party Number Type</td>
<td>Cisco CallManager</td>
<td></td>
</tr>
<tr>
<td>Calling Party Numbering Plan</td>
<td>Cisco CallManager</td>
<td></td>
</tr>
<tr>
<td><strong>Connected Party Transformations</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Connected Line ID Presentation</td>
<td>Default</td>
<td></td>
</tr>
<tr>
<td>Connected Name Presentation</td>
<td>Default</td>
<td></td>
</tr>
<tr>
<td><strong>Called Party Transformations</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Discard Digits</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td>Called Party Transform Mask</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Prefix Digits (Outgoing Calls)</td>
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<tr>
<td>Called Party Number Type</td>
<td>Cisco CallManager</td>
<td></td>
</tr>
<tr>
<td>Called Party Numbering Plan</td>
<td>Cisco CallManager</td>
<td></td>
</tr>
<tr>
<td><strong>ISDN Network Specific Facilities Information Element</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Network Service Protocol</td>
<td>-- Not Selected --</td>
<td></td>
</tr>
<tr>
<td>Carrier Identification Code</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

<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>&lt; Not Exist &gt;</td>
<td></td>
</tr>
</tbody>
</table>

* indicates required item.
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** = **10.80.11.2**; 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.
Cisco UCM service Profile Configuration

**Navigation:** User Management ➔ User setting ➔ Service Profile
## Service Profile Configuration

### Status
Status: Ready

### Service Profile Information
- **Name**: Jabber SVC Profile
- **Description**: Jabber SVC Profile

### Voicemail Profile
- **Primary**: Not set
- **Secondary**: Not set
- **Tertiary**: Not set

### MailStore Profile
- **Primary**: Not set
- **Secondary**: Not set
- **Tertiary**: Not set
- **Inbox Folder**: INBOX
- **Trash Folder**: Deleted Items
- **Polling Interval (in seconds)**: 60
- **Allow duel folder mode**:
Cisco UCM service profile Configuration (Continued)

### Service Profile Configuration

#### Conferencing Profile
- **Primary**: <None>
- **Secondary**: <None>
- **Tertiary**: <None>
- **Server Certificate Verification**: Any
- **Credentials source for web conference service**: Not set

#### Directory Profile
- **Primary**: <None>
- **Secondary**: <None>
- **Tertiary**: <None>
- **Use UDS for Contact Resolution**
- **Use Logged On User Credentials**

- **Username**
- **Password**
- **Search Base 1**
- **Search Base 2**
- **Search Base 3**
- **Recursive Search on All Search Bases**

- **Search Timeout (seconds)**: 5
- **Base Filter (Only used for Advanced Directory)**
- **Predictive Search Filter (Only used for Advanced Directory)**

- **Search Timeout (seconds)**: 5

#### IM and Presence Profile
- **Primary**: IMP IR
- **Secondary**: <None>
- **Tertiary**: <None>

#### CTI Profile
- **Primary**: CTI_SRV
- **Secondary**: <None>
- **Tertiary**: <None>

#### Video Conference Scheduling Portal Profile
- **Primary**: <None>
- **Secondary**: <None>
- **Tertiary**: <None>

---

* indicates required item.
Cisco Unified CM IM Presence – CCMCIP Profile Configuration

**Navigation Path:** Application → CCMCIP Profile

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

Set Primary CCMCIP Host *: 10.80.11.2. 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.
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 **Device Pool** = Default. This is used for the test.

Set **Media Resource Group List** = MRGL_SW_MTP. 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>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>None (Default)</td>
</tr>
<tr>
<td>Device Name</td>
<td>IMTrunk</td>
</tr>
<tr>
<td>Description</td>
<td>Default</td>
</tr>
<tr>
<td>Device Pool</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Common Device Configuration</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Call Classification</td>
<td>MRGL_SW_MTP</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>Hub_None</td>
</tr>
<tr>
<td>Location</td>
<td>None</td>
</tr>
<tr>
<td>AAR Group</td>
<td>None</td>
</tr>
<tr>
<td>Tunneled Protocol</td>
<td>No Changes</td>
</tr>
<tr>
<td>QSIG Variant</td>
<td>No Changes</td>
</tr>
<tr>
<td>ASN.1 ROSE OID Encoding</td>
<td>None</td>
</tr>
<tr>
<td>Packet Capture Mode</td>
<td>None</td>
</tr>
</tbody>
</table>
Cisco Unified Communications Manager SIP Trunk to CUP Configuration (Continued)
Cisco UCM SIP Trunk to CUP Configuration (Continued)

<table>
<thead>
<tr>
<th>Number Type</th>
<th>Prefix</th>
<th>Strip Digits</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>Incoming Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

- **Incoming Called Party Settings**

- **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
Cisco UCM SIP Trunk to CUP Configuration (Continued)

### Trunk Configuration

- **Save** | **Delete** | **Reset** | **Add New**

- **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**

### Caller Information

- **Caller ID DN**
- **Caller Name**
- **Maintain Original Caller ID DN and Caller Name in Identity Headers**

### SIP Information

#### Destination

<table>
<thead>
<tr>
<th>Destination Address</th>
<th>Destination Address IPv6</th>
<th>Destination</th>
</tr>
</thead>
<tbody>
<tr>
<td>10.80.11.5</td>
<td>5060</td>
<td></td>
</tr>
</tbody>
</table>

- **MTP Preferred Originating Code**: 711lowlow
- **BLF Presence Group**: Standard Presence group
- **SIP Trunk Security Profile**: Non Secure SIP Trunk Profile
- **Rerouting Calling Search Space**: <None>
- **Out-Of-Dial Ref Call Sea Search Space**: <None>
- **SUBSCRIBE Call Search Space**: <None>
- **SIP Profile**: Standard SIP Profile
- **DTMF Signaling Method**: No Preference

#### Normalization Script

- **Normalization Script**: <None>
- **Enable Trace**

<table>
<thead>
<tr>
<th>Parameter Name</th>
<th>Parameter Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td></td>
</tr>
</tbody>
</table>

#### Recording Information

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

#### Geolocation Configuration

- **Geolocation**: <None>
- **Geolocation Filter**: <None>

### Related Links

- **Back To Find/List** | **Go**

---

*- indicates required item.

**XY**: Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.
Cisco UCM 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 CT1 from User Device Association screen
Cisco UCM end user Configuration (Continued)

<table>
<thead>
<tr>
<th>Manager User ID</th>
<th>Department</th>
<th>User Locale</th>
<th>Associated PC/Site Code</th>
<th>Digest Credentials</th>
<th>Confirm Digest Credentials</th>
<th>User Profile</th>
<th>User Rank</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>Use System Default</td>
<td>1-Default User Rank</td>
</tr>
</tbody>
</table>

**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)

**Device Information**

<table>
<thead>
<tr>
<th>Controlled Devices</th>
<th>Available Profiles</th>
</tr>
</thead>
<tbody>
<tr>
<td>CTIRD1</td>
<td></td>
</tr>
</tbody>
</table>

**Controlled Devices**

- Device Association
- Line Appearance Association for Presence

**Presence Viewer for User**

- UC Service Profile: Jabber_SVC_Profile

**Related Links:** Back to Find List Users ▼ Go
Cisco UCM end user Configuration (Continued)

[Image of Cisco Unified CM Administration interface showing End User Configuration]

- Device Information
  - Controlled Devices
    - CTI1

- Available Profiles
  - CTI Controlled Device Profiles

- Extension Mobility
  - Available Profiles

Related Links: Back to Find List Users  Go
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.

![Cisco UCM end user Configuration screenshot](image-url)
Check Enable Mobility

Add the following permissions for Standard Users:
- Standard CCM End-Users
- Standard CTI Enabled
- Standard CCMUSER Administration
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
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
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** = 10.80.11.2 [This is the Cisco UCM publisher IP]

Set **Port** = 5060

Check Register with SIP server
Click Save.
Cisco UCM Unity Connection Port Group Configuration (Continued)

Cisco Unity Connection Telephony Integration – Add Ports

![Cisco Unity Connection Telephony Integration](image)

Cisco Unity Connection 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** = clus21inilitypartition

All other values are default.
Cisco Unity Connection User Configuration (Continued)
Cisco Unity Connection User Configuration (Continued)

All values are default.

Similarly, create a user that has a Cisco extension.
<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>