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

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

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

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

The following items were tested:

- Basic call between the two systems and verification of voice path, using both SIP and SCCP phones on the Cisco side, and SIP client on the Skype for Business side (Refer to limitation section for more info)
- CLIP/CLIR/CNIP/CNIR features: calling party Name and number delivery (allowed and restricted) (Refer to limitation section for more info)
- COLP/CONP/COLR/CONR features: connected Name and number delivery (allowed and restricted) (Refer to limitation section for more info)
- Call transfer: attended and early attended (Refer to limitation section for more info)
- Alerting Name Identification (Refer to limitation section for more info)
- Call forwarding: call forward unconditional(CFU), call forward busy (CFB), and call forward no answer (CFNA)
- Hold and resume with music on hold
- Three-way conferencing (Refer to limitation section for more info)
- Voice messaging and MWI activation-deactivation (Refer to limitation section for more info)
- Extend and Connect (Refer to limitation section for more info)
- Call Park (Refer to limitation section for more info)
Listed below are the highlights of the integration issues:

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

Below are the key results:

- Basic call, call transfer, call forwarding, conference call, and hold and resume tested successfully with a few caveats and limitations.
- Centralized voicemail, using Unity Connection server integrated with Cisco UCM via SIP was used for testing. This voicemail solution can provide centralized voicemail services, supporting both Skye For Business and Cisco end-users.

Network Topology
Limitations
These are the known limitations, caveats, or integration issues:

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

System Components

Hardware Requirements
The following hardware was used

- Cisco UCS-C240-M3S VMWare Host
Cisco 7960, 7965, 7975, 9951, and 9971 IP phones

Software Requirements

The following software is required:

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

Features

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

Features Supported

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

Features Not Supported or Not Tested
• Call completion (callback, automatic callback)
• Shared Line on Skype for Business
• Message Waiting Indicator on Skype for Business Endpoints
• Blind transfer
• Video calls
• Scenarios that required 3 PBXs.
• Scenarios involving Non SIP interfaces.

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

Configuring Sequence and Tasks:
Skype for Business:

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

- SIP trunk security profile
- SIP profile
- Media resource group and media resource group list
- Assign media resource group list (MRGL) in the default device pool
- Region configuration
- Normalization script
- SIP trunk to Skype for Business
- SIP Trunk to Cisco Unity Connection
- Assign User in Cisco Unity Connection
- SIP and SCCP phones device configuration
- Route Group, Route List and SIP Route Pattern
- Voice Mail
- Route pattern to Skype for Business, Unity Connection and Skype for Business call park range
- Extend and Connect Feature and User configuration

Cisco Unity Connection:

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

**Configuring the Skype for Business**

**Add Cisco UCM to Skype for Business Topology**

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

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

Right click and select “New IP/PSTN Gateway”
Set FQDN = <FQDN of the Cisco UCM>—clus20pub.skypelabsj.local is used in this test. Click Next.

Skype for Business – Add PSTN Gateway (Continued)

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

Skype for Business – Add PSTN Gateway (Continued)
Set Trunk Name = FQDN of the Cisco UCM – clus20pub.skypelabsj.local is used for this test
Set Listening port for IP/PSTN gateway = The Listening port should match the Incoming Port setting in the CISCO UCM’s SIP Trunk Security Profile – 5060 is used for this test
Set SIP Transport Protocol = TCP
Set Associate Mediation Server: Assign this PSTN gateway to the Front End co-located mediation server – fe01.skypelabsj.local is used for this test.
Click Finish.
Publish the topology so these new configurations take effect.
Skype for Business – Add PSTN Gateway (Continued)

Open the Skype for Business 2015 Control Panel.

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

Select New \(\rightarrow\) Pool Trunk
Set Service = Trunk to Cisco UCM that was created earlier as a PSTN gateway in the topology builder – clus20pub.skypelabsj.local is used for the test.

Set **Maximum early dialogs supported** = 20

Set **Encryption support level** = Optional

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

Check **Enable media bypass**

Check **Centralized media processing**

Uncheck **Enable RTP latching**

Check **Enable forward call history**

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

Check **Enable outbound routing failover timer**
Skype for Business –Trunk Configuration (Continued)
Create voice routing test case information

Edit Trunk Configuration - PstnGateway:clus20pub.skypelabs.local

OK Cancel

Scope: Toll
Name: * PstnGateway:clus20pub.skypelabs.local
Description:

Maximum early dialogs supported:
20

Encryption support level:
Optional

Refer support:
Enable sending refer to the gateway

- Enable media bypass
- Centralized media processing
- Enable RTP latching
- Enable forward call history
- Enable forward P-Asserted-Identity data
- Enable outbound routing failover timer
Add a Translation rule under Called number translation rules – CUCMExtn was created in this test. This is used to remove the “+” that is added by SFB during a transfer to a Cisco UCM extension. If SFB attempts a transfer to a Cisco UCM extension without this rule, the transfer fails because the extension is not recognized by Cisco UCM.
Skype for Business – Trunk Configuration (Continued)

Skype for Business Route Configuration

**Navigation:** Voice Routing -> Route

Click New

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

Add Associated trunks = select the trunk configured earlier – PstnGateway:clus20pub.skypelabsj.local
Skype for Business Voice Policy and PSTN Usage Configuration

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

Click New

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

Set Calling Features:

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

Set Associated PSTN usages:

- Click New
- Set Name: enter a name to identify this PSTN Usage record – SFB_PSTN is used in the test.
- Set Associated Routes = select the route created earlier= SFB-Cisco
Skype for Business Dial Plan Configuration

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

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

User dial plan:

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

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

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

Edit Dial Plan - ccm

Scope: User
Name: ccm

Simple name: ccm

Description: ccmtest

Dialed conferencing region:

External access prefix:

Associated Normalization Rules

<table>
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<th>Normalization rule</th>
<th>State</th>
<th>Pattern to match</th>
<th>Translation pattern</th>
</tr>
</thead>
<tbody>
<tr>
<td>CUCM 4digit</td>
<td>Committed</td>
<td>^\d4$</td>
<td>$1</td>
</tr>
</tbody>
</table>
Skype for Business – User Dial Plan-Normalization Rule

Pool dial plan:

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

Set Simple Name= enter text to identify this pool dial plan. PstnGateway_clus20pub.skypelabsj.local is used in this test
Associated Normalization Rules ➔ New

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

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

**Skype for Business — Pool Dial Plan-Normalization Rule 1**
Add another normalization rule as below:

This is used by the client to dial out to internal extensions and to the external PBX i.e. Cisco UCM
Skype for Business – Pool Dial Plan-Normalization Rule 2

Edit Dial Plan ➤ Edit Normalization Rule - Keep All

Name: * 
Keep All

Description:

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

Starting digits:

Length:
At least 1

Digits to remove:
0

Digits to add:
+

Pattern to match: *
^\(d+\)$

Translation rule: *
→$1

[Edit, Reset, Help]
Skype for Business Call Park Range Configuration

Navigation: Voice Features -> Call Park

Click New.

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

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

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

---

Skype for Business Global Media Bypass Configuration

Navigation: Network Configuration -> Global

Edit Global Setting –

- Check Enable media bypass
- Check Always bypass

Commit the configuration.
Skype for Business User Configuration
Login to the Skype for Business Active Directory

Navigation: Active Directory Users and Computers → Users

Add a New User
Follow the screenshots below to add a new user:
Skype for Business – New User configuration (continued)
Skype for Business – New User configuration (continued)
Once the user is created, login to the Skype for Business 2015 Control Panel

Navigation: Users → Enable users

Click on the Add button and find the new user created earlier.
Skype for Business – New User configuration (continued)

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

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

Set Telephony= Enterprise Voice

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

Set Dial plan policy = cucm (as configured earlier)

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

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

**Display name:**
SFBUser1

**Enabled for Skype for Business Server**

**SIP address:**
sip:SFBUser1

**Registrar pool:**
FE01.skypelabsj.local

**Telephony:**
Enterprise Voice

**Line URI:**
tel:+8003

**Dial plan policy:**
cucm

**Voice policy:**
SFB-Cisco

**Conferencing policy:**
Automatic

**Client version policy:**
Automatic

**PIN policy:**
Automatic

**External access policy:**
Automatic

---

**Skype for Business Client Configuration**

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

Navigation: Settings ➔ Tools ➔ Options ➔ Personal ➔ MyAccount
Set Sign-in-address= enter the sip uri of the user configured in username@domain format. user2@skypelabsj.local is used for example.

Click Advanced. Select Manual Configuration.

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

**Skype for Business – Client configuration (continued)**
Configuring the Cisco Unified Communications Manager

Cisco Unified Communications Manager Software Version

Cisco Unified CM Administration

Navigation: System → Security → SIP trunk security profile

Set Name*= SFB-Non-secure. This is used for the test.
Set Device Security mode = Non Secure
Set Incoming Transport Type = TCP+UDP
Set Outgoing Transport Type = TCP
Check Accept Presence Subscription
Check Accept out of dialog refer
Check Accept unsolicited notification
Check Accept Replaces header
All other values are default.
Cisco Unified Communications Manager SIP Trunk Security Profile for Trunk to Unity Connection

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

Set Name* = UnityConnectionTrunkSecurityProfile. This is used for the test.
Set Device Security mode = Non Secure
Set Incoming Transport Type = TCP+UDP
Set Outgoing Transport Type = TCP
Check Accept Presence Subscription
Check Accept unsolicited notification
Check Accept Replaces header
All other values are default.

![SIP Trunk Security Profile Configuration](image-url)
Cisco Unified Communications Manager SIP Profile

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

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

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

Set SIP Rel1XX Options = Disabled

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

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

All other values are default.
### Cisco Unified Communications Manager SIP Profile (Continued)

#### SDP Information
- **SDP Session-level Bandwidth Modifier for Early Offer and Re-invites**
  - TIAS and AG
- **SDP Transparency Profile**
  - Pass all unknown SDP attributes
- **Accept Audio Codec Preferences in Received Offer**
  - Default
- **Require SDP Inactive Exchange for Mid-Call Media Change**
- **Allow RR/RS bandwidth modifier (RFC 3556)**

#### Parameters used in Phone

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
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</thead>
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<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>3600</td>
</tr>
<tr>
<td>Timer T1 (msec)</td>
<td>500</td>
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<tr>
<td>Timer T2 (msec)</td>
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<tr>
<td>Retry INVITE</td>
<td>6</td>
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<tr>
<td>Retry Non-INVITE</td>
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<tr>
<td>Start Media Port</td>
<td>16384</td>
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<tr>
<td>Stop Media Port</td>
<td>32766</td>
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<tr>
<td>Call Pickup URI</td>
<td><code>x-cisco-serviceur-pickup</code></td>
</tr>
<tr>
<td>Call Pickup Group Other URI</td>
<td><code>x-cisco-serviceur-opickup</code></td>
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<tr>
<td>Call Pickup Group URI</td>
<td><code>x-cisco-serviceur-pickup</code></td>
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</table>
Cisco Unified Communications Manager SIP Profile (Continued)

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<th>Related Links:</th>
<th>Go</th>
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<td>Call Pickup Group URL</td>
<td></td>
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<tr>
<td>Meet Me Service URL</td>
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<td></td>
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<tr>
<td>User Info</td>
<td>None</td>
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<tr>
<td>DTMF DB Level</td>
<td>Nominal</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>
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<tr>
<td>Caller ID Blocking</td>
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<td>Do Not Disturb Control</td>
<td>User</td>
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<td>Telnet Level for 7940 and 7960</td>
<td>Disabled</td>
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<td>Resource Priority Namespace</td>
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<tr>
<td>Timer Keep Alive Expires (seconds)</td>
<td>120</td>
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<tr>
<td>Timer Subscribe Expires (seconds)</td>
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<tr>
<td>Timer Subscribe Delta (seconds)</td>
<td>5</td>
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<tr>
<td>Maximum Redirects</td>
<td>70</td>
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<tr>
<td>Off Hook To First Digit Timer (milliseconds)</td>
<td>15000</td>
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<tr>
<td>Call Forward URI</td>
<td>x-cisco-serviceuri-cfwdall</td>
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<tr>
<td>Speed Dial (Abbreviated Dial) URI</td>
<td>x-cisco-serviceuri-abbrevdial</td>
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<tr>
<td>Conference Join Enabled</td>
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<tr>
<td>RFC 2343 Hold</td>
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<td></td>
</tr>
<tr>
<td>Semi Attended Transfer</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
Cisco Unified Communications Manager SIP Profile (Continued)
Cisco Unified Communications Manager Media Resource Group

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

**Media Resource Group MRG**

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

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

Set all resources in the Selected Media Resources* Box.

All other values are default.
**Resource Group for MRG_NoMTP**

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

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

Set Available Media Resources = MTP_2, MTP_3 and MTP_4

Set other resources in the Selected Media Resources*

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

Navigation Path: Media Resources → Media Resource Group List

Add New

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

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

Set Available Media Resources = MRGL

Set Selected Media Resource Groups = MRG

Add new

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

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

Set Available Media Resources = MRG

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

Media Resource Group List Configuration

Status

- Status: Ready

Media Resource Group List Status

Media Resource Group List: MRG_LNoMTP (used by 26 devices)

Media Resource Group List Information

Name: MRG_LNoMTP

Media Resource Groups for this List

Available Media Resource Groups:

Selected Media Resource Groups:

MRG_LNoMTP

Find and List Media Resource Group Lists

Status

2 records found

Media Resource Group List

Name

Copy

MRG

MRG_LNoMTP

Add New Select All Clear All Delete Selected

Add New Select All Clear All Delete Selected

Page 48 of 111
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Cisco Unified Communications Manager Device Pool Configuration

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

**Navigation Path:** System → Device Pool

Add New.

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

Set Cisco Unified Communications Manager Group*= Default

Set Date/Time Group*= CMLocal

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

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

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

Device Pool Configuration

- 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.

<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>&lt;None&gt;</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>International Number</td>
<td>Default</td>
<td>&lt;None&gt;</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Unknown Number</td>
<td>Default</td>
<td>&lt;None&gt;</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Subscriber Number</td>
<td>Default</td>
<td>&lt;None&gt;</td>
<td>&lt;None&gt;</td>
</tr>
</tbody>
</table>

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

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

Add New

G711 Preferred is created in this test.

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

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

Set Audio Codec Preference List= G711 Preferred

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

Set Region= Default. This is used in this example

Set Audio Codec Preference List= G711 G729. This is used in this example

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

All other values are default
Cisco Unified Communications Manager Normalization Script

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

**Add New**

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

Set Content = add script content.

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

---

<table>
<thead>
<tr>
<th>Region</th>
<th>Audio Codec Preference List</th>
<th>Maximum Audio Bit Rate</th>
<th>Maximum Session Bit Rate for Video</th>
<th>Maximum Session Bit Rate for Immersive Video</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td>G711, G729</td>
<td>64 kbps (G.722, G.711)</td>
<td>Use System Default (304 kbps)</td>
<td>Use System Default (2000000000 kbps)</td>
</tr>
<tr>
<td>G711 Preferred</td>
<td>G711, G729</td>
<td>64 kbps (G.722, G.711)</td>
<td>Use System Default (304 kbps)</td>
<td>Use System Default (2000000000 kbps)</td>
</tr>
</tbody>
</table>

Note: Regions not displayed use System Default.
Normalization Script

```plaintext
--[]

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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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
b_CT_line = b_CT_line:gsub("0", "1000")
sdp = sdp:modifyLine("b=CT", "0", b_CT_line)
msg:setSdp(sdp)
return
end
b_CT_line = b_CT_line:gsub("64", "1000")
sdp = sdp:modifyLine("b=CT", "64", b_CT_line)
msg:setSdp(sdp)
end
end

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

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

modify_CT_bandwidth(msg)
remove_AS_bandwidth(msg)

end

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

local function process_inbound_progress(msg)

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

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

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

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

    local sdp = msg:getSdp()
    if sdp then

msg:removeUnreliableSdp()
end

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

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

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

function convertHIToDiversion(msg)
    local historyInfos = msg:getHeaderValues("History-Info")
    for i, hi in ipairs(historyInfos) do
        hi = string.gsub(hi, "%%3B", ";")
        hi = string.gsub(hi, "%%3D", "=")
        hi = string.gsub(hi, "%%22", "\"")
        hi = string.gsub(hi, "%%20", " ")

        -- MS format: <sip:+19728522619@med02.lynlabsj.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
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
end

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

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

Cisco Unified Communications Manager SIP Trunk to Skype for Business Configuration

Navigation: Device → Trunk

Set Device Name* = SFB-FE01-CUCM. This is used for the test
Set Description = this text is used to identify this Trunk Group
Set Device Pool* = G711 Preferred. This is used for the test
Set Call Classification* = Use System Default. This is used for the test
Set Media Resource Group List = MRGL_MTP. This is used for the test
Check Media Termination Point Required
Check Run On All Active Unified CM Nodes
Check Redirecting Diversion Header Delivery – Inbound
Set Destination Address = FE01.skypelabsj.local. [FQDN of Skype for Business Front End] This is used in the test
Set SIP Trunk Security Profile* = SFB-Non-secure
Set SIP Profile* = SFB – Standard SIP Profile
Set DTMF Signaling Method* = RFC 2833
Set Normalization Script = CiscoScriptForSFB
All other values are default.
Cisco Unified Communications Manager SIP Trunk to Skype for Business Configuration (Continued)

<table>
<thead>
<tr>
<th>Trunk Configuration</th>
<th>Related Links: Back To Find/List</th>
<th>Go</th>
</tr>
</thead>
<tbody>
<tr>
<td>Packet Capture Duration</td>
<td>0</td>
<td></td>
</tr>
<tr>
<td>Media Termination Point Required</td>
<td>✔</td>
<td></td>
</tr>
<tr>
<td>Retry Video Call as Audio</td>
<td>✔</td>
<td></td>
</tr>
<tr>
<td>Path Replacement Support</td>
<td>✔</td>
<td></td>
</tr>
<tr>
<td>Transmit UTF-8 for Calling Party Name</td>
<td>✔</td>
<td></td>
</tr>
<tr>
<td>Transmit UTF-8 Names in QSIG APDU</td>
<td>✔</td>
<td></td>
</tr>
<tr>
<td>Unattended Call</td>
<td>✔</td>
<td></td>
</tr>
<tr>
<td>MRTP Allowed - When this flag is checked, encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.</td>
<td>✔</td>
<td></td>
</tr>
<tr>
<td>Consider Traffic on This Trunk Secure</td>
<td>✔</td>
<td></td>
</tr>
<tr>
<td>Route Class Signaling Enabled</td>
<td>✔</td>
<td></td>
</tr>
<tr>
<td>Use Trusted Relay Point</td>
<td>✔</td>
<td></td>
</tr>
<tr>
<td>PSTN Access</td>
<td>✔</td>
<td></td>
</tr>
<tr>
<td>Run On All Active Unified CM Nodes</td>
<td>✔</td>
<td></td>
</tr>
</tbody>
</table>

---

**Intercompany Media Engine (IME)**

| E.164 Transformation Profile | < None > |

---

**MLPP and Confidential Access Level Information**

| MLPP Domain | < None > |
| Confidential Access Mode | < None > |
| Confidential Access Level | < None > |
Cisco Unified Communications Manager SIP Trunk to Skype for Business Configuration (Continued)
### 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.

<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 Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
<td>□</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>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 Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
<td>□</td>
</tr>
</tbody>
</table>

### Connected Party Settings

Connected Party Transformation CSS: &lt; None &gt;  □  
Use Device Pool Connected Party Transformation CSS
Cisco Unified Communications Manager SIP Trunk to Skype for Business Configuration (Continued)
Cisco Unified Communications Manager SIP Trunk to Skype for Business Configuration (Continued)

Cisco Unified Communications Manager SIP Trunk to Cisco Unity Connection Configuration

Navigation: Device → Trunk
Set Device Name*= UnityConnection. This is used for the test.
Set Description = this text is used to identify this Trunk Group.
Set Device Pool* = G711 Preferred

Check Run On All Active Unified CM Nodes
Check Redirecting Diversion Header Delivery – Inbound
Check Redirecting Diversion Header Delivery – Outbound
Set Destination Address = 10.80.10.5. This is used for the test.

Set SIP Trunk Security Profile*= UnityConnectionTrunkSecurityProfile
Set SIP Profile*= SFB - Standard SIP Profile

DTMF Signaling Method *= RFC 2833

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

**Trunk Configuration**

<table>
<thead>
<tr>
<th>Packet Capture Duration</th>
<th></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 QSIG APDU</td>
<td></td>
</tr>
<tr>
<td>Unattended Port</td>
<td></td>
</tr>
<tr>
<td>SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.</td>
<td></td>
</tr>
<tr>
<td>Consider Traffic on This Trunk Secure*</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></td>
</tr>
<tr>
<td>Run On All Active Unified CM Nodes</td>
<td></td>
</tr>
</tbody>
</table>

**Intercompany Media Engine (IME)**

| E-164 Transformation Profile | < None > |

**MLPP and Confidential Access Level Information**

| MLPP Domain | < None > |
| Confidential Access Mode | < None > |
| Confidential Access Level | < None > |
Cisco Unified Communications Manager SIP Trunk to Cisco Unity Connection Configuration (Continued)
Cisco Unified Communications Manager SIP Trunk to Cisco Unity Connection Configuration (Continued)

### Trunk Configuration

#### 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 Number</td>
<td>Default</td>
<td>0</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: (✓)
- Calling Party Transformation CSS: < None >
- Use Device Pool Calling Party Transformation CSS: (✓)
- Calling Party Selection*: Originator
- Calling Line ID Presentation*: Default
- Calling Name Presentation*: Default
- Calling and Connected Party Info Format*: Deliver DN only in connected party
- Redirecting Diversion Header Delivery: Outbound
- Redirecting Party Transformation CSS: < None >
- Use Device Pool Redirecting Party Transformation CSS: (✓)

### Caller Information

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

### SIP Information

#### Destination

- Destination Address
- Destination Address IP: 10.80.10.5
- Destination Port: 9060

- MTP Preferred Originating Codec*: J112law
- BLF Presence Group*: Standard Presence group
- SIP Trunk Security Profile*: UnityConnectionTrunkSecurityProfile
- Redirecting Calling Search Space: < None >
- Out-Of-Dialog Refer Calling Search Space: < None >
- SUBSCRIBE Calling Search Space: < None >
- SIP Profile*: SIP - Standard SIP Profile
- DTMF Signaling Method*: RFC 2833
Cisco Unified Communications Manager Route Group

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

Add New

SFB-CUCM was configured in this test

Set Route Group Name = SFB-CUCM

Set Distribution Algorithm = Circular

Select SFB-FE-01-CUCM from Available Devices and click the Add to Route Group
Cisco Unified Communications Manager Route List

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

Add New

SFB-CUCM_Route List was created for this test.

Set Name: SFB-CUCM_Route List

Set Cisco Unified Communications Manager Group = Default

Click on Add Route Group

Set Route Group* = SFB-CUCM-[NON-QSIG]
Cisco Unified Communications Manager SIP Route Pattern

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

Add New

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

Set SIP Trunk/Route List* = SFB_CUCM_Route List
In a similar way, add SIP Route Patterns for all the servers that comprise the Skype for Business environment.

In the test, the following SIP Route Patterns were configured:
Cisco Unified Communications Manager Voice Mail Configuration

Configure Voice Mail Pilot:

Navigation: Advanced Features → Voice Mail → Voice Mail Pilot

Add new

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

Set Description = Unity Connection VM. This text is used to identify this SIP Profile

---

Cisco Unified Communications Manager Route Pattern to Skype for Business Extensions

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

Add New

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

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

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

Uncheck Provide Outside Dial Tone

Set Calling Line ID Presentation= Default

Set Calling Name Presentation= Default

Set Connected Line ID Presentation*= Default

Set Calling Name Presentation* = Default
All other values are default.
Cisco Unified Communications Manager Route Pattern to invoke Jabber client with Remote Destination configured as Skype for Business Extensions

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

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

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

Uncheck Provide Outside Dial Tone
Set Calling Line ID Presentation= Default
Set Calling Name Presentation= Default
Set Connected Line ID Presentation*= Default
Set Calling Name Presentation*= Default
Discard Digits = PreDot
All other values are default
Cisco Unified Communications Manager Route Pattern to Skype for Business Call Park range

The Skype for Business Call Park range configured is 100-199. The following route pattern “1XX” is therefore configured to enable a parked call to be retrieved from Cisco UCM.
Cisco Unified Communications Manager Route Pattern to Unity Connection Voice Mail

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

<table>
<thead>
<tr>
<th>Pattern Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Pattern <strong>Route Pattern</strong>: 7000</td>
</tr>
<tr>
<td>Route Partition <strong>Route Partition</strong>: &lt; None &gt;</td>
</tr>
<tr>
<td>Description <strong>Description</strong>: Voice mail to unity Connection</td>
</tr>
<tr>
<td>Numbering Plan <strong>Numbering Plan</strong>: -- Not Selected --</td>
</tr>
<tr>
<td>Route Filter <strong>Route Filter</strong>: &lt; None &gt;</td>
</tr>
<tr>
<td>MLPP Precedence <strong>MLPP Precedence</strong>: Default</td>
</tr>
<tr>
<td>Apply Call Blocking Percentage <strong>Apply Call Blocking Percentage</strong>:</td>
</tr>
<tr>
<td>Resource Priority Namespace Network Domain <strong>Resource Priority Namespace Network Domain</strong>: &lt; None &gt;</td>
</tr>
<tr>
<td>Route Class <strong>Route Class</strong>: Default</td>
</tr>
<tr>
<td>Gateway/Route List <strong>Gateway/Route List</strong>: UnityConnection</td>
</tr>
<tr>
<td>Route Option <strong>Route Option</strong>: Route this pattern</td>
</tr>
<tr>
<td>Call Classification <strong>Call Classification</strong>: OffSet</td>
</tr>
<tr>
<td>External Call Control Profile <strong>External Call Control Profile</strong>: &lt; None &gt;</td>
</tr>
<tr>
<td>Allow Device Override <strong>Allow Device Override</strong>:</td>
</tr>
<tr>
<td>Provide Outside Dial Tone <strong>Provide Outside Dial Tone</strong>:</td>
</tr>
<tr>
<td>Allow Overlap Sending <strong>Allow Overlap Sending</strong>:</td>
</tr>
<tr>
<td>Urgent Priority <strong>Urgent Priority</strong>:</td>
</tr>
<tr>
<td>Authorization Level <strong>Authorization Level</strong>:</td>
</tr>
<tr>
<td>Require Client Matter Code <strong>Require Client Matter Code</strong>:</td>
</tr>
</tbody>
</table>

### Calling Party Transformations

- **Use Calling Party’s External Phone Number Mask**:  
- **Calling Party Transform Mask**:  
- **Prefix Digits (Outgoing Calls)**:  

---
Route Pattern Configuration for 7000 (Continued)

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

Cisco UCM UC service Configuration
Navigation Path: User Management → User setting → UC Service

Add New

Select Service Type as CTI

Set Name = CTI_SRV
Set Host Name/IP Address* = 10.80.10.2; this is the Cisco UCM publisher IP.

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
Cisco UCM service profile Configuration (Continued)
Cisco UCM service profile Configuration (Continued)
Cisco Unified CM IM Presence – CCMCIP Profile Configuration

**Navigation Path:** Application → CCMCIP Profile

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

Set Primary CCMCIP Host *: 10.80.10.2. Cisco Publisher IP. This is used in this test.

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

Add Users to Profile: user1, user 2 and user3. This is used in this test.
Cisco UCM – SIP trunk to Cisco IM&Presence Trunk Configuration

**Navigation Path:** Device → Trunk

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

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

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

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

Set **Destination Address** = 10.80.10.6. This is used in this example.

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

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

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

All other values are default.
Cisco Unified Communications Manager SIP Trunk to CUP Configuration (Continued)
Cisco UCM SIP Trunk to CUP Configuration (Continued)

![Cisco UCM SIP Trunk to CUP Configuration](image)

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

<table>
<thead>
<tr>
<th>SIP Information</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Destination</strong></td>
</tr>
<tr>
<td>Destination Address</td>
</tr>
<tr>
<td>10.80.10.0</td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td>MTP Preferred Originating Codec</td>
</tr>
<tr>
<td>DLF Presence Group</td>
</tr>
<tr>
<td>SIP Trunk Security Profile</td>
</tr>
<tr>
<td>Relocating Calling Search Space</td>
</tr>
<tr>
<td>Out-Of-Dialing Refer Calling Search Space</td>
</tr>
<tr>
<td>SUBSCRIBE Calling Search Space</td>
</tr>
<tr>
<td>SIP Profile</td>
</tr>
<tr>
<td>DTMF Signaling Method</td>
</tr>
</tbody>
</table>
Cisco UCM end user configuration
Add user to Cisco UCM

**Navigation:** User Management → End user

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

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

Check Home Cluster.

Click the Device Association

Select CTI1 from User Device Association screen
### User Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>User Status</td>
<td>Enabled Local User</td>
</tr>
<tr>
<td>User ID</td>
<td>user1</td>
</tr>
<tr>
<td>Password</td>
<td>******************************</td>
</tr>
<tr>
<td>Confirm Password</td>
<td>******************************</td>
</tr>
<tr>
<td>Self-Service User ID</td>
<td>5007</td>
</tr>
<tr>
<td>PIN</td>
<td>******************************</td>
</tr>
<tr>
<td>Confirm PIN</td>
<td>******************************</td>
</tr>
<tr>
<td>Last name</td>
<td>user1</td>
</tr>
<tr>
<td>Middle name</td>
<td></td>
</tr>
<tr>
<td>First name</td>
<td>Jabber</td>
</tr>
<tr>
<td>Title</td>
<td></td>
</tr>
<tr>
<td>Directory URI</td>
<td><a href="mailto:user1@skypelabg.local">user1@skypelabg.local</a></td>
</tr>
<tr>
<td>Telephone Number</td>
<td></td>
</tr>
<tr>
<td>Home Number</td>
<td></td>
</tr>
<tr>
<td>Mobile Number</td>
<td></td>
</tr>
<tr>
<td>Pager Number</td>
<td></td>
</tr>
<tr>
<td>Mail ID</td>
<td></td>
</tr>
<tr>
<td>Manager User ID</td>
<td></td>
</tr>
<tr>
<td>Department</td>
<td></td>
</tr>
<tr>
<td>User Locals</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>
Cisco UCM end user Configuration (Continued)
Check Allow Control of Device from CTI
Select the Primary Extension for this user. 5007 is used for this example.
Add the following permissions for Standard Users:
Remote Destination Configuration

**Navigation:** Device ➔ Remote Destination

Add New

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

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

Check Enable Extend and Connect.

Set CTI Remote Device = CTI1
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. user1 is used in this test.

Set Device Pool: G711 Preferred

Save.
Add a DN to this device.

DN 7009 was configured for this test.
Cisco UCM CTI Remote Device DN Configuration

[Diagram of Directory Number Configuration with fields for Directory Number, Route Partition, Description, Alerting Name, ASCII Alerting Name, External Call Control Profile, Allow Control of Device from CTI, Associated Devices, and other settings such as Voice Mail Profile, Calling Search Space, DLF Presence Group, User Hold MOH Audio Source, Network Hold MOH Audio Source, and Reject Anonymous Calls.]
### Line 1 on Device CT11

- Display (Caller ID):
  - Jabber_SFB8004
- ASCII Display:
  - Jabber_SFB8004
- Line Text Label:
- External Phone Number Mask:
- Recording Option:
  - Call Recording Disabled
- Recording Profile:
  - < None >
- Recording Media Source:
  - Gateway Preferred
- Monitoring Calling Search Space:
  - < None >

### Multiple Call/Call Waiting Settings on Device CT11

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

- Maximum Number of Calls:
  - 4
- Busy Trigger:
  - 1
  - (Less than or equal to Max. Calls)

### Forwarded Call Information Display on Device CT11

- Caller Name
- Caller Number
- Redirected Number
- Dialed Number
Cisco Unity Connection

Cisco Unity Connection Telephony Integration – Add Phone System

**Navigation:** Telephony Integrations → Phone system

Add New

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

---

Cisco Unity Connection Telephony Integration – Add Port Group

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

Go

Set Phone System = SFB_CUCM

Set Create From – Port group Type = SIP

Set Display Name* = SFB_CUCM-1. This Name used for this example.

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

---

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Check Register with SIP server

Click Save.

Cisco UCM Unity Connection Port Group Configuration (Continued)
Cisco UCM Unity Connection Port Group Configuration (Continued)
Cisco Unity Connection Telephony Integration – Add Ports

**Cisco Unity Connection User Configuration**

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

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

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

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

Save.

Set Phone System= SFB_CUCM. This is used in this example.

All other values are default.

**Cisco Unity Connection User Configuration (Continued)**
Cisco Unity Connection User Configuration (Continued)
<table>
<thead>
<tr>
<th>Name</th>
<th>Alias</th>
<th>First Name</th>
<th>Last Name</th>
<th>Display Name</th>
<th>SMTP Address</th>
<th>Title</th>
<th>Initials</th>
<th>Employee ID</th>
<th>Extension</th>
<th>Cross-Server Transfer Extension or URI</th>
<th>Outgoing Fax Number</th>
<th>Outgoing Fax Server</th>
<th>Search Scope</th>
<th>Phone System</th>
<th>Class of Service</th>
<th>Active Schedule</th>
<th>Self-enrollment at Next Sign-In</th>
<th>List in Directory</th>
<th>Send Non-Delivery Receipts on Failed Message Delivery</th>
<th>Skip PIN When Calling From a Known Extension</th>
</tr>
</thead>
<tbody>
<tr>
<td>User1</td>
<td>U1004</td>
<td>USER1</td>
<td></td>
<td>SFB, User1</td>
<td>@du120unity.lab.tokvision.com</td>
<td></td>
<td></td>
<td></td>
<td>1004</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>SFB_CUCM</td>
<td>Voice Mail User COS</td>
<td>Weekdays</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
</tr>
</tbody>
</table>

**LDAP Integration Status**
- [ ] Integrate with LDAP Directory
- [x] Do Not Integrate with LDAP Directory
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>Acronym</td>
<td>Definition</td>
</tr>
<tr>
<td>---------</td>
<td>------------</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>

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