Avaya Communication Manager Release 6.3 using SIP trunk to Cisco Unified Communications Manager Release 10.5.2
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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 Avaya Communication Manager Release 6.3 and Avaya Aura Session Manager Release 6.3 using SIP Early-Offer.

The following items were tested:
- Basic call between the two systems and verification of voice path, using both SIP and H323 on the Avaya side, and SIP and SCCP IP phones on the Cisco side
- CLIP/CLIR/CNIP/CNIR features: calling party Name and number delivery (allowed and restricted)
- COLP/CONP/COLR/CONR features: connected Name and number delivery (allowed and restricted)
- Call transfer: attended, early attended and blind transfer (see caveats for details)
- Alerting Name Identification
- 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
- Voice messaging and MWI activation-deactivation
- Audio Codec Preference List
- Extend and Connect
- Call Park
- Video

Listed below are the highlights of the integration issues:
- Basic calls worked from Cisco UCM to Avaya PBX and vice versa. Avaya’s Media Shuffling feature was enabled throughout this testing exercise.
- CLIR/CNIR—The Avaya SIP trunk does not support calling/connected Name and number restriction. Restriction of calling number on Avaya H323 and SIP phones is achieved by configuring the Avaya station configuration page and not the SIP trunk page. This restriction is honored by Cisco UCM.
- COLR/CONR—as with calling Name and number presentation restrictions, the Avaya PBX does not support connected Name and number restriction on SIP trunks.
Below are the key results:

- Basic call, call transfer, call forwarding, conference call, and hold and resume work successfully
- Centralized voicemail, using Unity Connection server integrated to Cisco UCM via SIP was used for testing. This voicemail solution can provide centralized voicemail services, supporting both Avaya and Cisco end-users

Network Topology

Basic Call Setup
Limitations
These are the known limitations, caveats, or integration issues:

- Avaya doesn’t support Alerting Name feature
- Avaya couldn’t block caller id when calls were local (internal)
- Although the Codec Preference List was used and the INVITE message displayed the right codec, Avaya would respond with the INVITE with their preferred Codec Preference for the call
- Avaya one-X@Communicator in H323 mode does not update the caller ID (connected Name) nor privacy call from Cisco UCM
- Avaya one-X@Communicator in SIP mode updated the caller ID as Avaya SIP trunk description for privacy call from Cisco UCM
- Avaya one-X@Communicator can do blind transfer. Avaya 9630G and 4610 phones do not support blind call transfer
- Calling/Connected Name un-available feature is not supported by Avaya
- One way video between Avaya one-X@Communicator H323 phones with Cisco 8945 skinny phone. Cisco UCM sends payload type 115 but Avaya responds with payload type 100 causing the issue.
- One way audio between Avaya video phones when Cisco UCM transfers the video Call. Cisco UCM sends re-INVITE after the redirection and updates SDP but Avaya is not updating the SDP resulting is one way audio and two way video between Avaya video end points.
**System Components**

**Hardware Requirements**

The following hardware was used:

- Cisco UCS-C240-M3S VMWare Host
- Cisco 8945, 9951, and 9971 IP phones
- G6430 Media Gateway

**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 10.5.2.10000-5
- Cisco Unified Communications Manager IM & P release 10.5.2.10000-9
- Cisco Unity Connection release 10.5.2.10000-5
- Cisco Jabber 10.5.0 Build 37889
- Avaya Communication Manager release 6.3 Service Pack 10(patch 22147)
- Avaya G430 Media Gateway firmware release 34.5.1
- Avaya Aura® Session Manager R6.3 (6.3.12.0.631208) Service Pack 12
- Avaya Aura® System Manager R6.3.12. Build No. - 6.3.0.8.5682-6.3.8.4903, Software Update Revision No: 6.3.12.9.3022
- Avaya one-X@Communicator Release 6.1.9.04-SP9-132
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 on page 6 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
- Blind call transfer (see limitation section)
- 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 (lamp ON, lamp OFF)
- Audio Codec Preference List
- Call Park/Pickup
- Extend and Connect
- Video
Features Not Supported or Not Tested

- Call completion (callback, automatic callback)
- Shared Line - Hold & Resume with MOH

Configuration

The goal of this guide is to provide an overview of the integration between Cisco Unified Communication Manager and Avaya (CM, SM) PBX’s. 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:

Avaya CM:

Configure the IP-Codec-Set, and IP-Network-Region.
Configure the IP interface for C-LAN and IP Media Processor cards.
Configure Cisco UCM as an IP node-Name.
Configure the signaling group for the SIP trunk to Cisco UCM.
Configure the trunk group for the SIP trunk to Cisco UCM.
Configure the SIP and digital station phone extension.
Configure the uniform dialing plan to the Cisco UCM extensions.
Configure the route pattern to the Cisco UCM extensions.

Cisco Unified Communications Manager:

1. SIP trunk security profile
2. Device setting SIP profile
3. Media resource group and media resource group list
4. Assign media resource group list (MRGL) in the default device pool
5. SIP trunk to Avaya PBX
6. SIP Trunk Normalization Script
7. SIP Trunk to Cisco Unity
8. Assign User in Cisco Unity
9. SIP and SCCP phones device configuration
10. Route pattern to the Avaya PBX
11. Call Manager Service Parameter “Duplex Streaming Enabled” set to “True”
12. Audio Codec Preference, Region and device pool Configuration
13. Extend and Connect Feature configuration
# Configuring the Avaya PBX

Avaya Software Version and Hardware Configuration List

```
list configuration all

SYSTEM CONFIGURATION

<table>
<thead>
<tr>
<th>Board Number</th>
<th>Board Type</th>
<th>Code</th>
<th>Vintage</th>
</tr>
</thead>
<tbody>
<tr>
<td>001V2 DS1 MM</td>
<td>MM710BP HW16 FW050</td>
<td>01 02 03 04 05 06 07 08</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>09 10 11 12 13 14 15 16</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>17 18 19 20 21 22 23 24</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>u u u u u u u u u</td>
<td></td>
</tr>
<tr>
<td>001V3 ANA MM</td>
<td>MM714AP HW10 FW088</td>
<td>01 02 u u u u u</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>03 04 u u u u u</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>05 06 u u u u u</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>07 08 u u u u u</td>
<td></td>
</tr>
<tr>
<td>001V5 DS1 MM</td>
<td>MM710BP HW11 FW044</td>
<td>01 02 u u u u u</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>03 04 u u u u u</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>05 06 u u u u u</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>07 08 u u u u u</td>
<td></td>
</tr>
<tr>
<td>001V9 MG-ANNOUNCEMENT</td>
<td>VMM-ANN</td>
<td>01 02 03 04 05 06 07 08</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>09 10 11 12 13 14 15 16</td>
<td></td>
</tr>
</tbody>
</table>
```

Command successfully completed

Command: 
```
Verify system capacities and licensing:

Make sure system have enough license for SIP trunk and Video. Also make sure on page 10, the following features are enabled:

ARS?                                                Verify “y” is displayed.
ARS/AAR Partitioning?                              Verify “y” is displayed
ARS/AAR Dialing without FAC?                         Verify “y” is displayed
OPTIONAL FEATURES

IP PORT CAPACITIES

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Maximum Administered H.323 Trunks</td>
<td>12000</td>
</tr>
<tr>
<td>Maximum Concurrently Registered IP Stations</td>
<td>18000</td>
</tr>
<tr>
<td>Maximum Concurrently Registered Remote Office Trunks</td>
<td>12000</td>
</tr>
<tr>
<td>Maximum Concurrently Registered Remote Office Stations</td>
<td>18000</td>
</tr>
<tr>
<td>Maximum Concurrently Registered IP eCons</td>
<td>28</td>
</tr>
<tr>
<td>Max Concur Registered Unauthenticated H.323 Stations</td>
<td>100</td>
</tr>
<tr>
<td>Maximum Video Capable Stations</td>
<td>41000</td>
</tr>
<tr>
<td>Maximum Video Capable IP Softphones</td>
<td>20</td>
</tr>
<tr>
<td>Maximum Administered SIP Trunks</td>
<td>24000</td>
</tr>
<tr>
<td>Maximum Administered Ad-hoc Video Conferencing Ports</td>
<td>24000</td>
</tr>
<tr>
<td>Maximum Number of DS1 Boards with Echo Cancellation</td>
<td>522</td>
</tr>
<tr>
<td>Maximum TN2501 VAL Boards</td>
<td>128</td>
</tr>
<tr>
<td>Maximum Media Gateway VAL Sources</td>
<td>250</td>
</tr>
<tr>
<td>Maximum TN2602 Boards with 80 VoIP Channels</td>
<td>128</td>
</tr>
<tr>
<td>Maximum TN2602 Boards with 320 VoIP Channels</td>
<td>128</td>
</tr>
<tr>
<td>Maximum Number of Expanded Meet-me Conference Ports</td>
<td>0</td>
</tr>
</tbody>
</table>

(Note: You must logoff & login to effect the permission changes.)

----------

OPTIONAL FEATURES

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Abbreviated Dialing Enhanced List?</td>
<td>y</td>
</tr>
<tr>
<td>Access Security Gateway (ASG)?</td>
<td>n</td>
</tr>
<tr>
<td>Analog Trunk Incoming Call ID?</td>
<td>y</td>
</tr>
<tr>
<td>A/D Crp/Sys List Dialing Start at 0?</td>
<td>y</td>
</tr>
<tr>
<td>Answer Supervision by Call Classifier?</td>
<td>y</td>
</tr>
<tr>
<td>ARS/AAR Partitioning?</td>
<td>y</td>
</tr>
<tr>
<td>ARS/AAR Dialing without FAC?</td>
<td>y</td>
</tr>
<tr>
<td>ASAI Link Core Capabilities?</td>
<td>n</td>
</tr>
<tr>
<td>ASAI Link Plus Capabilities?</td>
<td>n</td>
</tr>
<tr>
<td>Async. Transfer Mode (ATM) PNC?</td>
<td>n</td>
</tr>
<tr>
<td>Async. Transfer Mode (ATM) Trunking?</td>
<td>n</td>
</tr>
<tr>
<td>ATM WAN Spare Processor?</td>
<td>n</td>
</tr>
<tr>
<td>ATMS?</td>
<td>y</td>
</tr>
<tr>
<td>Attendant Vectoring?</td>
<td>y</td>
</tr>
<tr>
<td>Audible Message Waiting?</td>
<td>y</td>
</tr>
<tr>
<td>Authorization Codes?</td>
<td>y</td>
</tr>
<tr>
<td>CAS Branch?</td>
<td>n</td>
</tr>
<tr>
<td>CAS Main?</td>
<td>n</td>
</tr>
<tr>
<td>Change COR by FAC?</td>
<td>n</td>
</tr>
<tr>
<td>Computer Telephony Adjunct Links?</td>
<td>y</td>
</tr>
<tr>
<td>Cfg Of Calls Redirected Off-net?</td>
<td>y</td>
</tr>
<tr>
<td>DCS (Basic)?</td>
<td>y</td>
</tr>
<tr>
<td>DCS Call Coverage?</td>
<td>y</td>
</tr>
<tr>
<td>DCS with Rerouting?</td>
<td>n</td>
</tr>
<tr>
<td>Digital Loss Plan Modification?</td>
<td>y</td>
</tr>
<tr>
<td>DS1 MSP?</td>
<td>y</td>
</tr>
<tr>
<td>DS1 Echo Cancellation?</td>
<td>y</td>
</tr>
</tbody>
</table>

(Note: You must logoff & login to effect the permission changes.)
Configure System Feature:

On page 1,

Set **Trunk-to-Trunk Transfer** to All

Set **CPN/ANI/ICLID Replacement for Restricted/Unavailable calls** to anonymous
<table>
<thead>
<tr>
<th>Feature-Related System Parameters</th>
</tr>
</thead>
<tbody>
<tr>
<td>CPN/ANI/ICCID Parameters</td>
</tr>
<tr>
<td>CPN/ANI/ICCID Replacement for Restricted Calls: anonymous</td>
</tr>
<tr>
<td>CPN/ANI/ICCID Replacement for Unavailable Calls: anonymous</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>DISPLAY TEXT</th>
</tr>
</thead>
<tbody>
<tr>
<td>Identity When Bridging: principal</td>
</tr>
<tr>
<td>User Guidance Display? n</td>
</tr>
<tr>
<td>Extension only label for Team button on 96xx H.323 terminals? n</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>INTERNATIONAL CALL ROUTING PARAMETERS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Local Country Code:</td>
</tr>
<tr>
<td>International Access Code:</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>SCCAN PARAMETERS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enable Enbloc Dialing without ARS FAC? n</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>CALLER ID ON CALL WAITING PARAMETERS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Caller ID on Call Waiting Delay Timer (msec): 200</td>
</tr>
</tbody>
</table>

---

Page 14 of 135
**Config IP Codec Set and IP Network Region:**

Codec set 1 is configured for this test.

Audio Codec G711MU and G.729 are select codec

Media Encryption is set to none

Allow Direct-IP Multimedia set to ‘y”

Set Maximum Call Rate for Direct-IP Multimedia: 4096:Kbits

Set Maximum Call Rate for Priority Direct-IP Multimedia: 4096:Kbits
Allow Direct-IP Multimedia? y
Maximum Call Rate for Direct-IP Multimedia: 4096:Kbit/s
Maximum Call Rate for Priority Direct-IP Multimedia: 4096:Kbit/s

<table>
<thead>
<tr>
<th>Mode</th>
<th>Redundancy</th>
<th>Size(ms)</th>
</tr>
</thead>
<tbody>
<tr>
<td>FAX</td>
<td>pass-through</td>
<td>1</td>
</tr>
<tr>
<td>Modem</td>
<td>off</td>
<td>1</td>
</tr>
<tr>
<td>TDD/TTY</td>
<td>US</td>
<td>3</td>
</tr>
<tr>
<td>H.323 Clear-channel</td>
<td>n</td>
<td>0</td>
</tr>
<tr>
<td>SIP 64K Data</td>
<td>n</td>
<td>20</td>
</tr>
</tbody>
</table>
Configure IP-Network-region 1:

Location: 1

Authoritative Domain: lab.tekvizion.com used for this testing

Name: tekvizion Lab

Codec Set: 1 which programmed in previous step

Inter/Intra-region IP-IP Direct Audio: YES

H.323 SECURITY PROFILES: any-auth

![PuTTY Terminal Screenshot]

**Display ip-network-region 1**

<table>
<thead>
<tr>
<th>Region: 1</th>
<th>Location:</th>
<th>Authoritative Domain:</th>
<th>Lab Network Region:</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>lab.tekvizion.com</td>
<td>n</td>
</tr>
</tbody>
</table>

**MEDIA PARAMETERS**

- Intra-region IP-IP Direct Audio: yes
- Inter-region IP-IP Direct Audio: yes
- IP Audio Hairpinning? y
- UDP Port Min: 2048
- UDP Port Max: 3329

**DIFFSERV/TOS PARAMETERS**

- Call Control PHB Value: 46
- Audio PHB Value: 46
- Video PHB Value: 26

**802.1p/Q PARAMETERS**

- Call Control 802.1p Priority: 6
- Audio 802.1p Priority: 6
- Video 802.1p Priority: 5

**H.323 IP ENDPOINTS**

- H.323 Link Bounce Recovery? y
- Idle Traffic Interval (sec): 20
- Keep-Alive Interval (sec): 5
- Keep-Alive Count: 5

**AUDIO RESOURCE RESERVATION PARAMETERS**

- RSVP Enabled: n
INTER-GATEWAY ALTERNATE ROUTING / DIAL PLAN TRANSPARENCY
Incoming LDW Extension:
Conversion To Full Public Number - Delete: Insert:
Maximum Number of Trunks to Use for IGAR:
Dial Plan Transparency in Survivable Mode? n

BACKUP SERVERS (IN PRIORITY ORDER)

H.323 SECURITY PROFILES
1 any-auth

TCP SIGNALING LINK ESTABLISHMENT FOR AVAYA H.323 ENDPOINTS
Near End Establishes TCP Signaling Socket? y
Near End TCP Port Min: 61440
Near End TCP Port Max: 61441

Source Region: 1 Inter Network Region Connection Management I M G A C
dst codec direct WAN-BW-limits Video Intervening Dyn A G c
rgn set WAN Units Total Norm Prio Shr Regions CAC R L =
1 1 all
2 2 y NoLimit
3
4
5
6
7
8
9
10
11
12
13
14
15
Configure the Signaling group and trunk Group

Configure the Node IP for Avaya Session Manager and CM
Configure the Signaling Group 2:
Set Group Type: sip
IMS Enabled? N
Transport Method: tcp
IP Video? Y
Priority Video? Y
Peer Detection Enabled? Y
Near-end Node Name: procr
Far-end Node Name: tekaasm
Near-end Listen Port: 5060
Far-end Listen Port: 5060
Far-end Domain: lab.tekvizion.com This is used for this testing
Far-end Network Region: 1
DTMF over IP: rtp-payload
Direct IP-IP Audio Connections? Y
Initial IP-IP Direct Media? Y This is used for this testing
Configure trunk group 2:

Group number: 2
Group Type: sip
Group Name: SIP TRUNK
TAC: #102
Member Assignment Method: auto
Service Type: tie Signalling Group: 2
Number of Members: 10
Preferred Minimum Session Refresh Interval(sec): 900
Numbering Format: private
Mark Users as Phone? Y
Support Request History? Y
Telephone Event Payload Type: 101
TRUNK FEATURES

ACF Assignment? n Measured: none Maintenance Tests? y

Numbering Format: private

UUI Treatment: service-provider

Replace Restricted Numbers? y
Replace Unavailable Numbers? y

Modify Tandem Calling Number: no

Show ANSWERED BY on Display? n

MARK USERS AS PHONE? n
Prepend ‘+’ to Calling/Alerting/Diverting/Connected Number? n
Send Transferring Party Information? n
Network Call Redirection? n

Send Diversion Header? n
Support Request History? y
Telephony Event Payload Type: 101

Identity for Calling Party Display: P-Asserted-Identity

Interworking of ISDN Clearing with In-Band Tones: keep-channel-active
Configure Route pattern:

Pattern Number: 3

Pattern Name: SIP trunk

Grp No: 2

FRL: 0

ITC: unre

Numbering Format: lev0-pvt
Dialing plan:

Configure 4 digits number start with 4 as ext.

Configure 4 digit number start with 20 as udp.

Configure 4 digit number start with 1000 as udp. This is used to access Cisco Unity Voice mail system.

8 and 9 are set as 1 digit feature access code.
Configure the AAR dialplan:

Set 4 digits dial string start with 10(Unity mail), 4006(Avaya SIP phone) and 20(Cisco phone) to use Route pattern 3 with Call Type aar.
Configure Private numbering plan:

Use the `change private-numbering` command to define the calling party number to be sent out through SIP trunk. In our case, 4 digits extension with leading digits 40 via trunk group 2 will result in a 4-digit calling number.
Fill in the indicated fields as shown below and use default values for remaining fields.

**Configure Uniform dial plan:**

**Matching Pattern** Enter the number Communication Manager matches to dialed numbers. Accepts up to seven digits. 10 and 20 and 4 are used in the example

**Len** Enter the number of user-dialed digits the system collects to match to this Matching Pattern value. 10 and 20 is used in the example

**Del** Enter number of digits to delete before routing the call. 0 is selected

**Net** The server or switch network used to analyze the converted, aar is used here

![Uniform Dial Plan Table]

Save Translation

After finished above configuration, use the “save translation” command to save these changes.
Configure Avaya Aura Session Manager

Access Avaya Aura System Manager Web login screen via https://<IP Address/FQDN>, for this test, IP address used is 10.70.4.3. Use admin as User ID and associated password, and then “Log on”

Navigation: Home → Elements → Routing
Add Domains

**Navigation:** Home → Elements → Routing → Domains

Under page Domain Management:

Name: lab.tekvizion.com

Type: sip
Add Location

**Navigation:** Home ➔ Elements ➔ Routing ➔ Locations

Name: Dallas
Add Adaptations

**Navigation:** Home → Elements → Routing → Adaptations

Adaptation for Cisco CUCM

Adaptation Name: Cucm_10_5_2

Module Name: CiscoAdapter

Module Parameter: fromto=true odstd=10.80.16.2 iosrcd=lab.tekvizion.com
### Digit Conversion for Incoming Calls to SM

<table>
<thead>
<tr>
<th>Add</th>
<th>Remove</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Item</th>
<th>Matching Pattern</th>
<th>Min</th>
<th>Max</th>
<th>Phone Context</th>
<th>Delete Digits</th>
<th>Insert Digits</th>
<th>Address to modify</th>
<th>Adaptation Data</th>
<th>Notes</th>
</tr>
</thead>
</table>

**Digit Conversion for Outgoing Calls from SM**

<table>
<thead>
<tr>
<th>Add</th>
<th>Remove</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Item</th>
<th>Matching Pattern</th>
<th>Min</th>
<th>Max</th>
<th>Phone Context</th>
<th>Delete Digits</th>
<th>Insert Digits</th>
<th>Address to modify</th>
<th>Adaptation Data</th>
<th>Notes</th>
</tr>
</thead>
</table>
Adaptation for Avaya Aura CM
Adaptation Name: Avaya_CM
Module Name: DigitConversionadapter
Module Parameter: fromto=true
Add SIP Entities and Entity Link

**Navigation:** Home → Elements → Routing → SIP Entities

SIP Entity for Session Manager

Name: tekaasm

FQDN or IP Address: 10.70.4.7

Type: Session Manager

Location: Dallas

Time Zone: America/Chicago

SIP Link Monitoring: Use Session Manager Configuration
SIP Entity and entity Link for CUCM
Name: Cucm_10_5_2
FQDN or IP Address: 10.80.16.3
Type: Other
Adaptation: Cucm_10_5_2
Location: Richardson
Time Zone: America/Chicago
SIP Link Monitoring: Use Session Manager Configuration
SIP Entity and Entity Link for Avaya Aura Communication Manager

Name: tekaacm

FQDN or IP Address: 10.70.4.4

Type: CM

Adaptation: Avaya_CM
Location: Richardson

Time Zone: America/Chicago

Sip Link Monitoring: Use Session Manager Configuration
Add Entity Links

**Navigation:** Home ➔ Elements ➔ Routing ➔ Entity Links

Add entity link between Avaya Session Manager and Cisco CUCM:
Name: ASM tekaasm_tekaacm_5060_TCP

SIP Entity 1: teaaksm
Protocol: tcp
Port 5060

SIP Entity 2: Cucm_10_5_2
Port 5060

Connection Policy: trusted
Add entity link between Avaya Session Manager and Avaya Aura Communication Manager:

Name: sm_tekaacm_5060_TCP

SIP Entity 1: tekaasm
Protocol: tcp
Port: 5060

SIP Entity 2: tekaacm
Port 5060

Connection Policy: trusted
Add Routing Policies

**Navigation:** Home → Elements → Routing → Routing Policies

Routing policy for call to go to Cisco CUCM

**Name:** to_Cucm_10_5_2

Select SIP Entity “Cucm_10_5_2” for SIP Entity as Destination

**Dial Pattern**

**Pattern:** 20 and 1000

**Min:** 4

**Max:** 4

**SIP Domain:** lab.tekvizion.com

**Original Location Name:** Richardson
Routing Policy for calls to go to Avaya Aura Communication Manager

Name: to tekaaCM

Select SIP Entity “tekaacm” for SIP Entity as Destination

Pattern: 4

Min: 4

Max: 4

SIP Domain: lab.tekvizion.com

Original Location Name: Richardson
Configuring the Cisco Unified Communications Manager
Cisco Unified Communications Manager Software Version

Cisco Unified CM Administration

System version: 10.5.2.10000-5
VMware Installation: 1 vCPU Intel(R) Xeon(R) CPU E5-2680 v2 2.70GHz, disk 1: 80G bytes, 4096Mbytes RAM, Partitions aligned
Last Successful Backup: 47 day(s) ago

User administrator last logged in to this cluster on Monday, May 4, 2015 9:20:23 AM CDT, to node 10.80.16.2, from 10.64.204.12 using HTTPS

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Cisco Unified Communications Manager SIP Trunk Security Profile

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

Set Name\* = Non Secure SIP Trunk Profile. This is used for this example.

Set Description = Non Secure SIP Trunk Profile authenticated by null String

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 Unity Connection

Set Name* = Non Secure SIP Trunk to VM Profile. This is used for this example.

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

Check Accept presence subscription

Check Accept out of dialog refer**

Check Accept unsolicited notification

Check Accept replaces header

Check Transmit security status

All other values are default.
Cisco Unified Communications Manager SIP Profile

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

Set Name*= Early Offer SIP Profile. This is used for this example.

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

Check Disable Early Media on 180

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

These values are default.

Check RFC 2543 Hold

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

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

SIP Profile Configuration

- Meet Me Service URL
- User Info
- DTMF DB Level
- Call Hold Ring Back
- Anonymous Call Block
- Caller ID Blocking
- Do Not Disturb Control
- Telnet Level for 7940 and 7960
- Resource Priority Namespace
- Timer Keep Alive Expires (seconds)
- Timer Subscribe Expires (seconds)
- Timer Subscribe Delta (seconds)
- Maximum Redirects
- Off Hook To First Digit Timer (milliseconds)
- Call Forward URL
- Speed Dial (Abbreviated Dial) URL
- Conference Join Enabled
- RFC 2543 Hold
- Semi Attended Transfer
- Enable VAD

- x-cisco-serviceuri-meetme
- None
- Nominal
- Off
- Off
- Off
- User
- Disabled
- < None >
- 120
- 120
- 5
- 70
- 15000
- x-cisco-serviceuri-cfdwell
- x-cisco-serviceuri-abbrevial

Page 49 of 135
Cisco Unified Communications Manager SIP Profile (Continued)
Cisco Unified Communications Manager SIP Profile (Continued)

Early Offer support for voice and video calls Mandatory (insert MTP if needed)
Check Enable OPTIONS Ping to monitor Destination status for Trunks with Service Type "None (Default)"
Check Send send-receive SDP in mid-call INVITE
All other values are default.
Cisco Unified Communications Manager SIP Trunk to Avaya Configuration

**Navigation:** Device → Trunk

Set Device Name* = Avaya_CM_6_3. This is used for this example.

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

Set Device Pool* = G711 Preferred. This is used for this example

Set Call Classification* = Use System Default. This is used for this example.

Set Media Resource Group List = MRGL_SW_MTP. This is used for this example.

All other values are default.
Check Redirecting Diversion Header Delivery - Inbound

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

Check Redirecting Diversion Header Delivery - Outbound

All other values are default.

![Outbound Calls Configuration](image)

In the Outbound Calls section, the following settings are highlighted:

- Redirecting Diversion Header Delivery - Outbound

Other settings include:

- Called Party Transformation CSS
- Calling Party Transformation CSS
- Calling Party Selection
- Calling Line ID Presentation
- Calling Name Presentation
- Calling and Connected Party Info Format

In the Caller Information section, the following settings are included:

- Caller ID DN
- Caller Name

[Additional settings for Caller Information are not visible in the image.]
Cisco Unified Communications Manager SIP Trunk to Avaya Configuration (Continued)

Set Destination Address = 10.70.4.7. This is used in this example.

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

Set SIP Profile* = Early Offer SIP Profile

Set DTMF Signaling Method* = No Preference

Set Normalization Script = Remove-Call-Info-Header. This example script Name was used to remove Call-Info Header to Avaya

All other values are default.

![SIP Information](image)

The table shows the configuration parameters for SIP trunking:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Destination Address</td>
<td>10.70.4.7</td>
</tr>
<tr>
<td>SIP Trunk Security Profile</td>
<td>Non Secure SIP Trunk Profile</td>
</tr>
<tr>
<td>SIP Profile</td>
<td>Early Offer SIP Profile</td>
</tr>
<tr>
<td>DTMF Signaling Method</td>
<td>No Preference</td>
</tr>
<tr>
<td>Normalization Script</td>
<td>Remove-Call-Info-Header</td>
</tr>
</tbody>
</table>

In the image, the SIP Information section is highlighted, showing the configured values for the various parameters.
Cisco Unified Communications Manager SIP Trunk to Avaya Configuration (Continued)

<table>
<thead>
<tr>
<th>Recording Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>None</td>
</tr>
<tr>
<td>This trunk connects to a recording-enabled gateway</td>
</tr>
<tr>
<td>This trunk connects to other clusters with recording-enabled gateways</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Geolocation Configuration</th>
</tr>
</thead>
<tbody>
<tr>
<td>Geolocation: &lt; None &gt;</td>
</tr>
<tr>
<td>Geolocation Filter: &lt; None &gt;</td>
</tr>
<tr>
<td>Send Geolocation Information</td>
</tr>
</tbody>
</table>

- indicates required item.

**§**: Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.
Set Name* = Remove-CallInfo-Header. This is used for this example.

Set Description = this text is used to identify this SIP Normalization Script.

Set Content* = Please see full contents on next page.

All Other values are default
**Note:** SIP Normalization script was used to remove the Call-Info Header from Cisco to Avaya.

**The full content of the SIP Normalization Script is captured below:**

```javascript
M = {}

function M.outbound_INVITE(msg)  
    msg:removeHeader("Call-Info")  
end

function M.outbound_18X_INVITE(msg)  
    msg:removeHeader("Call-Info")  
end

function M.outbound_200_INVITE(msg)  
    msg:removeHeader("Call-Info")  
end

function M.outbound_200_UPDATE(msg)  
    msg:removeHeader("Call-Info")  
end

return M
```
Cisco Unified Communications Manager SIP Trunk to Cisco Unity Configuration

**Navigation:** Device → Trunk

Set Device Name*= CUC. This is used for this example.

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

Set Device Pool* = Default

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

Check Run On All Active Unified CM Nodes

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

Check Redirecting Diversion Header Delivery - Inbound

All other values are default.

Check Redirecting Diversion Header Delivery – Outbound
Cisco Unified Communications Manager SIP Trunk to Cisco Unity Configuration (Continued)
Cisco Unified Communications Manager SIP Trunk to Cisco Unity Configuration (Continued)

Set Destination Address = 10.80.16.5. This is used in this example.

Set SIP Trunk Security Profile*= Non Secure SIP Trunk to VM Profile

Set SIP Profile*= Standard SIP Profile

DTMF Signaling Method *= No Preference

All other values are default.
Cisco Unity Connection User Configuration

**Navigation:** Cisco Unity connection → Users → Users

Set Alias\(^*\) = 4000. This is used for this example.
Set First Name = This text is used to identify this User.
Set Last Name\(^*\) = cisco. This is used for this example
Set Display Name = 4000. This is used in this example.
Set SMTP Address = 4000. This is used in this example.
Set Phone System= SIP. This is used in this example.
All other values are default.
Cisco Unity Connection Administration

For Cisco Unified Communications Solutions

Outgoing Fax
Number
Outgoing Fax
Server
Partition
dus26unity Partition
Search Scope
dus26unity Search Space
Phone
SIP
System
Class of
Service
Voice Mail User COS
Active Schedule
Weekdays
List in Directory
Set for Self-enrollment at Next Sign-In
Send Non-Delivery Receipts on Failed Message Delivery
Skip PIN When Calling From a Known Extension
Caution! Security risk. See Help for this page for details.
Use Short Calendar Updating Poll Interval
Recorded Name
Play/Record
Location
Address
Building
City
State
Postal Code
Country United States
Use System Default Time Zone
Cisco Unity Connection User Configuration (Continued)

All values are default.
Set System Name* = SIP. This Name used for this example
Port Group

**Navigation:** Telephony Integration → Port Group

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

Check Register with SIP server
Navigation Path: Telephony Integration ➔ Port Group ➔ Edit ➔ Servers

Port
Set Port Name = SIP-1-001. This Name used for this example

Phone System = SIP

Port Group = SIP-1

Server = clus26unity. This Name used for this example
Cisco Unified Communications Manager Service Parameter

**Navigation Path**: System → Service parameter → select server (Cluster26pub) → Select Service (Cisco CallManager (Active))

Set Duplex Streaming Enabled* = True. See Note under capture for more info.

**Note**: Cisco Unified Communications Manager Service Parameter “Duplex Streaming Enabled” should be set to “True” in order for MoH and ringback to work properly during call transfers/conferences initiated by Cisco stations to Avaya IP endpoints.
Cisco Unified Communications Manager Media Resource Group

**Navigation Path:** Media Resources → Media Resource Group

![Cisco Unified CM Administration Interface](image)

### Media Resource Group (1 - 2 of 2)

<table>
<thead>
<tr>
<th>Name</th>
<th>Description</th>
<th>Multicast</th>
<th>Copy</th>
</tr>
</thead>
<tbody>
<tr>
<td>MRG_SW_MTP</td>
<td>MRG_SW_MTP</td>
<td>false</td>
<td></td>
</tr>
<tr>
<td>MRG_SW_noMTP</td>
<td>MRG_SW_noMTP</td>
<td>false</td>
<td></td>
</tr>
</tbody>
</table>

Add New | Select All | Clear All | Delete Selected
Media Resource Group MRG_SW_MTP

Set Name*= MRG_SW_MTP This is used for this example.
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_SW_noMTP. This is used for this example.

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 List

Navigation Path: Media Resources → Media Resource Group List

Find and List Media Resource Group Lists

Status

2 records found

Media Resource Group List (1 – 2 of 2)

Find Media Resource Group List where Name begins with

<table>
<thead>
<tr>
<th>Name</th>
<th>Copy</th>
</tr>
</thead>
<tbody>
<tr>
<td>MRGL SW MTP</td>
<td></td>
</tr>
<tr>
<td>MRGL SW noMTP</td>
<td></td>
</tr>
</tbody>
</table>
Set Name* = MRGL_SW_MTP. This is used for this example.
Set Description = This text is used to identify this Media Resource Group List.
Set Available Media Resources = MRGL_SW_noMTP
Set Selected Media Resource Groups = MRG_SW_MTP
Note: noMTP Media resource group was used to test the scenarios for without using MTP test cases.
Cisco Unified Communications Manager Route Pattern to Avaya

Set Route Pattern* =4XXX. This is used to route to Avaya in this example

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

Set Gateway/Route List* = To Avaya_CM_6.3. This is used for this example

Uncheck Provide Outside Dial Tone

All other values are default.
Route Pattern Configuration for 4xxx (Continued)

Set Calling Party Transform Mask = XXXX
Set Calling Line ID Presentation= Allowed
Set Calling Name Presentation= Allowed
Set Connected Line ID Presentation* = Default
Set Calling Name Presentation* = Default

All other values are default.
Cisco Unified Communications Manager Translations Pattern to Avaya

Set Route Pattern* = 2050. This is used to route to Avaya in this example

Connected Line ID Presentation = Restricted

Connected Name Presentation = Restricted

Note: This translation pattern example shows restrict the connected name and number information for extension 200
Cisco Unified Communications Manager SIP Phone Device Level Configuration

**Navigation Path:** Device→ Phone

Set MAC Address* = 1C17D340829B. This is used in this example

Set Description = This text is used to identify this Phone

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

Set Phone Button Template* = Standard 9951 SIP. This is used in this example

Common Phone Profile *= Standard Common Phone Profile
<table>
<thead>
<tr>
<th>Line 1</th>
<th>Line 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Add new SIP</td>
<td>Add a new SD</td>
</tr>
</tbody>
</table>

### Related Links:
- Back To Find/List

### Phone Configuration

**Product Type:** Cisco 9951  
**Device Protocol:** SIP  
**Distribution:**  
**Registration:** Registered with Cisco Unified Communications Manager  
**IPv4 Address:** 172.16.31.106  
**Active Load ID:** sip9951.9-4-2-13  
**Inactive Load ID:** sip9951.9-4-1-9  
**Download Status:** None  

#### Device Information
- **MAC Address:** 1C17D3402290  
- **Description:** CISCO 9951 Audio  
- **Device Pool:** G711 Preferred  
- **Common Device Configuration:**  
- **Phone Button Template:** Standard 9951 SIP  
- **Softkey Template:** Standard User  
- **Common Phone Profile:** Standard Common Phone Profile  
- **Calling Search Space:** < None >
Cisco Unified Communications Manager SIP Phone Device Level Configuration (Continued)

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

All other values are default.
Cisco Unified Communications Manager SIP Phone Device Level Configuration (Continued)

Set Device Security Profile* = Cisco 9951- Standard SIP Non-Secure Profile. This is used in this example.

Set SIP Profile* = Early Offer SIP Profile. This is used in this example.

All other values are default.
Cisco Unified Communications Manager SIP Phone Device Level Configuration (Continued)

All other values are default.

<table>
<thead>
<tr>
<th>Certification Authority Proxy Function (CAPF) Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Certificate Operation</td>
</tr>
<tr>
<td>Authentication Mode</td>
</tr>
<tr>
<td>Authentication String</td>
</tr>
<tr>
<td>Key Size (bits)</td>
</tr>
<tr>
<td>Operation Completed By</td>
</tr>
<tr>
<td>Certificate Operation Status</td>
</tr>
<tr>
<td>Note: Security Profile Contains Additional CAPF Settings</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Expansion Module Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Module 1 Load Name</td>
</tr>
<tr>
<td>Module 2 Load Name</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>External Data Locations Information (Leave blank to use default)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Information Directory</td>
</tr>
</tbody>
</table>
Cisco Unified Communications Manager SIP Phone Device Level Configuration (Continued)

All values are default.

<table>
<thead>
<tr>
<th>Phone Configuration Details</th>
</tr>
</thead>
<tbody>
<tr>
<td>Secure Information URL</td>
</tr>
<tr>
<td>Secure Messages URL</td>
</tr>
<tr>
<td>Secure Services URL</td>
</tr>
</tbody>
</table>

**Extension Information**
- **Enable Extension Mobility**
- **Log Out Profile**: Use Current Device Settings
- **Log in Time**: < None >
- **Log out Time**: < None >

**MLPP and Confidential Access Level Information**
- **MLPP Domain**: < None >
- **MLPP Indication**: Default
- **MLPP Preemption**: Default
- **Confidential Access Mode**: < None >
- **Confidential Access Level**: < None >

**Do Not Disturb**
- **Do Not Disturb**
- **DND Option**: Use Common Phone Profile Setting
- **DND Incoming Call Alert**: < None >

**Secure Shell Information**
- **Secure Shell User**
- **Secure Shell Password**

**Product Specific Configuration Layout**
- **Override Common Settings**
- **Disable Speakerphone**
- **Disable Speakerphone and Headset**
- **PC Port**: Enabled
- **Back USB Port**: Enabled
Cisco Unified Communications Manager SIP Phone Device Level Configuration (Continued)

All other values are default.
Cisco Unified Communications Manager SIP Phone Device Level Configuration (Continued)

All other values are default.
Cisco Unified Communications Manager SIP Phone Device Level Configuration (Continued)

Set Cisco Camera* = Enabled. This is used in this example.

Set Video Capabilities* = Enabled. This is used in this example.

All values are default.
Cisco Unified Communications Manager SIP Phone Device Level Configuration (Continued)

All values are default.
Cisco Unified Communications Manager SIP Phone Device Level Configuration (Continued)
Cisco Unified Communications Manager SIP Phone Device Level Configuration (Continued)

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Discovery Protocol - Media Endpoint</td>
<td></td>
</tr>
<tr>
<td>Discovery (LLDP-MED):</td>
<td></td>
</tr>
<tr>
<td>Switch Port</td>
<td></td>
</tr>
<tr>
<td>Link Layer Discovery Protocol (LLDP):</td>
<td></td>
</tr>
<tr>
<td>LLDP Port</td>
<td></td>
</tr>
<tr>
<td>LLDP Asset ID</td>
<td></td>
</tr>
<tr>
<td>LLDP Power Priority</td>
<td></td>
</tr>
<tr>
<td>802.1X Authentication</td>
<td></td>
</tr>
<tr>
<td>FIPS Mode</td>
<td></td>
</tr>
<tr>
<td>Detect Unified CM Connection Failure</td>
<td></td>
</tr>
<tr>
<td>Switch Port Remote Configuration</td>
<td></td>
</tr>
<tr>
<td>PC Port Remote Configuration</td>
<td></td>
</tr>
<tr>
<td>Automatic Port Synchronization</td>
<td></td>
</tr>
<tr>
<td>Power Negotiation</td>
<td></td>
</tr>
<tr>
<td>ESM Access</td>
<td></td>
</tr>
</tbody>
</table>
Cisco Unified Communications Manager SIP Phone Device Level Configuration (Continued)

<table>
<thead>
<tr>
<th>Configuration Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Incoming Call Toast Timer</td>
<td>Enabled</td>
</tr>
<tr>
<td>Provide Dial Tone from Release Button</td>
<td>Disabled</td>
</tr>
<tr>
<td>Hide Video By Default</td>
<td>Disabled</td>
</tr>
<tr>
<td>Background Image</td>
<td>Disabled</td>
</tr>
<tr>
<td>Simplified New Call UI</td>
<td>Disabled</td>
</tr>
<tr>
<td>Enable VXC VPN for MAC</td>
<td>Dual Tunnel</td>
</tr>
<tr>
<td>VXC VPN Option</td>
<td>Challenge</td>
</tr>
<tr>
<td>VXC Challenge</td>
<td>Challenge</td>
</tr>
<tr>
<td>VXC-M Servers</td>
<td>Disabled</td>
</tr>
<tr>
<td>Revert to All Calls</td>
<td>Disabled</td>
</tr>
<tr>
<td>RTCP for Video</td>
<td>Enabled</td>
</tr>
<tr>
<td>Record Call Log from Shared Line</td>
<td>Disabled</td>
</tr>
<tr>
<td>Show Remote Private Calls</td>
<td>Disabled</td>
</tr>
<tr>
<td>Record Call Log For Remote Private Calls</td>
<td>Enabled</td>
</tr>
</tbody>
</table>
Cisco Unified Communications Manager SIP Phone Device Level Configuration (Continued)
Cisco Unified Communications Manager SCCP Phone Device Level Configuration

**Navigation Path:** Device → Phone

Set MAC Address* = 64E950CB2AE6. This is used in this example.

Set Description = This text is used to identify this Phone

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

Set Phone Button Template* = Standard 8945 SCCP. 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 SCCP Phone Device Level Configuration (Continued)

All other values are default.
Set Device Security Profile* = Cisco 8945 – Standard SCCP Non-Secure Profile. This is used in this example.

All other values are default.
Cisco Unified Communications Manager SCCP Phone Device Level Configuration (Continued)

These values are default.
Cisco Unified Communications Manager SCCP Phone Device Level Configuration (Continued)

Set Video Capabilities* = Enabled

These values are default.
Cisco Unified Communications Manager SCCP Phone Device Level Configuration (Continued)

These values are default.
Cisco Unified Communications Manager SCCP Phone Device Level Configuration (Continued)

These values are default.
Cisco Unified Communications Manager Audio Codec Preference List Configuration

**Navigation Path:** System → Service parameter → select server (Cluster26pub) → Select Service (Cisco CallManager (Active))

Set Accept Audio Codec Preference in Received Offer *= Off. This needs to be set when you are wanting to use the Codec Preference List created.
Cisco Unified Communications Manager Audio Codec Preference List Configuration (Continued)

G711 Preferred and G729 Preferred Audio Codec Preference List created in this example.

All other values are default.
Cisco Unified Communications Manager Audio Codec Preference List Configuration (Continued)

Set Name*= G711 G729. This is used for this example

Set Description*= This text is used to identify this Audio Codec Preference List

Set Codec in List*= G.711 U-Law 64k. First choice in this example

Set Codec in List*= G.729 8k. Second choice in this example

All other values are default.
Cisco Unified Communications Manager Audio Codec Preference List Configuration (Continued)

Set Name*= G729 G711. This is used for this example.

Set Description* = this text is used to identify this Audio Codec Preference List.

Set Codec in List*= G.729 8k. First choice for this example.

Set Codec in List*= G.711 U-Law 64. Second choice for this example.

All other values are default.
Cisco Unified Communications Manager Region Configuration

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

G711 Preferred and G729 Preferred created in this example.

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

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 G729
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 Region Configuration (Continued)

Set Name*= G729 Preferred. This is used in this example.
Set Region= G729 Preferred. This is used in this example
Set Audio Codec Preference List= G729 G711. This is used in this example
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= G729 G711. This is used in this example
Set Maximum Audio Bit Rate= 64 Kbps (G7.22, G7.11). This is used in this example
All other values are default.
Cisco Unified Communications Manager Device Pool Configuration

**Navigation Path:** System → Device Pool

G711 Preferred and G729 Preferred created in this example.

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

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

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_SW_MTP. This is used in this example.

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

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

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

Set Device Pool Name*= G729 Preferred. This is used in this example.
Set Cisco Unified Communications Manager Group* = Default
Set Date/Time Group* = CMLocal
Set Region* = G729 Preferred. 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 (Continued)

All values are default.

<table>
<thead>
<tr>
<th>Device Pool Configuration</th>
<th>Related Links</th>
<th>Back To Find/Use</th>
<th>Go</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calling Party Transformation CSS</td>
<td>&lt; None &gt;</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Called Party Transformation CSS</td>
<td>&lt; None &gt;</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

- **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>Type</th>
<th>Prefix</th>
<th>Strip Digits</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>National Number</td>
<td>Default</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>International Number</td>
<td>Default</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Unknown Number</td>
<td>Default</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Subscriber Number</td>
<td>Default</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

- **Incoming Called Party Settings**

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

All values are default.
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 end user configuration

Add user to Cisco UCM

**Navigation Path:** User Management → End user

Set User ID*= user2. This is used for this example.
Set Last Name = Jabber2. This is used for this example.
Check Home Cluster.

![Cisco UCM End User Configuration](image-url)
Cisco UCM end user Configuration (Continued)

Set Controlled Devices = CTIRDuser2. This is used for this example.
Check Allow Control of Device from CTI
Select the Primary Extension for this user. 2007 is used for this example.
Check Enable Mobility

Add the following permissions for Standard Users:
- Standard CCM End-Users
- Standard CTI Enabled
- Standard CCMUSER Administration
Add Phone: CTI Remote Device

The CTI Remote Device type represents the user’s remote device(s).
Select the desired Owner User ID. User2 is used in this example.
Set the Device Name populated automatically. Modify if desired - CTIRDuser2 used this example.
Set Device Pool: Default. This is used in this example.
Set RD* = 4000. This is used for this example. 4000 is the Avaya extension.
Remote Destination Configuration

Set Destination Number* = 4000. This is used for this example. Check Enable Extend and Connect.

<table>
<thead>
<tr>
<th>CTI Remote Device</th>
<th>Line</th>
<th>Association</th>
</tr>
</thead>
<tbody>
<tr>
<td>Line [1] - 2007 (no partition)</td>
<td>✓</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Remote Destination Information</th>
<th>Line</th>
<th>Association</th>
</tr>
</thead>
<tbody>
<tr>
<td>Name</td>
<td>RD</td>
<td></td>
</tr>
<tr>
<td>Destination Number*</td>
<td>4000</td>
<td></td>
</tr>
<tr>
<td>Owner User ID*</td>
<td>user2</td>
<td></td>
</tr>
</tbody>
</table>

- Enable Unified Mobility features
- Remote Destination Profile
- Single Number Reach Voicemail Policy* | -- Not Selected -- | |
- Enable Single Number Reach
  - Ring this phone and my business phone at the same time when my business line(s) is dialed.
- Enable Move to Mobile
  - If this is a mobile phone, transfer active calls to this phone when the mobility button on your Cisco IP Phone is pressed.

- Enable Extend and Connect
  - Allow this phone to be controlled by CTI applications (e.g., jabber)
  - CTI Remote Device* | CTIMobile2 | |

Timer Information

Wait* | 4.0 | seconds before ringing this phone when my business line is dialed.*
Prohibit this call from going straight to this phone's voicemail by using a time delay of* | 1.5 | seconds to detect when calls go straight to voicemail.*
Stop ringing this phone after* | 19.0 | seconds to avoid connecting to this phone's voicemail.*
Cisco UCM UC service Configuration

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

![UCM UC service Configuration](image)

The image shows the UC Service configuration in Cisco Unified CM Administration. The navigation path is indicated as follows:

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

The screen displays a list of UC services with their details such as Name, UC Service Type, Product Type, Host/IPv Address, Port, and Protocol. There are three records found:

1. **CTI_SRV**
   - UC Service Type: CTI
   - Product Type: CTI
   - Host/IPv Address: 10.80.16.2
   - Port: 2748
   - Protocol: TCP

2. **CTI_SUB1**
   - UC Service Type: CTI
   - Product Type: CTI
   - Host/IPv Address: 10.80.16.3
   - Port: 2748
   - Protocol: TCP

3. **IMF_SRV**
   - UC Service Type: IM and Presence
   - Product Type: Unified CM (IM and Presence)
   - Host/IPv Address: 10.80.16.6
   - Port: 2748
   - Protocol: TCP

The interface also includes options for adding new services, selecting all, clearing all, and deleting selected services.
Cisco UCM service Profile Configuration

**Navigation Path:** User Management ➔ User setting ➔ Service Profile

![Cisco Unified CM Administration](image)

<table>
<thead>
<tr>
<th>Service Profile Configuration</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Status</strong></td>
</tr>
<tr>
<td>Status: Ready</td>
</tr>
<tr>
<td><strong>Service Profile Information</strong></td>
</tr>
<tr>
<td>Name: Jabber_svc_Profile</td>
</tr>
<tr>
<td>Description: Jabber Service Profile</td>
</tr>
<tr>
<td>Make this the default service profile for the system</td>
</tr>
<tr>
<td><strong>Voicemail Profile</strong></td>
</tr>
<tr>
<td>Primary: &lt;None&gt;</td>
</tr>
<tr>
<td>Secondary: &lt;None&gt;</td>
</tr>
<tr>
<td>Tertiary: &lt;None&gt;</td>
</tr>
<tr>
<td>Credentials source for voicemail service: Not set</td>
</tr>
<tr>
<td><strong>Mailstore Profile</strong></td>
</tr>
<tr>
<td>Primary: &lt;None&gt;</td>
</tr>
<tr>
<td>Secondary: &lt;None&gt;</td>
</tr>
<tr>
<td>Tertiary: &lt;None&gt;</td>
</tr>
<tr>
<td>Inbox Folder: InBOX</td>
</tr>
<tr>
<td>Trash Folder: Deleted Items</td>
</tr>
<tr>
<td>Polling Interval (in seconds): 60</td>
</tr>
</tbody>
</table>
Cisco Unified CM IM Presence – CCMCIP Profile Configuration

**Navigation Path:** Application → Legacy Clients → CCMCIP Profile

Set Name *: remotedesk, this is used in this example.
Set Primary CCMCIP Host *: 10.80.16.2. Cisco Publisher IP. This is used in this example.
Set Backup CCMCIP Host *: 10.80.16.3. Cisco Publisher IP. This is used in this example.
Add Users to Profile: user2. This is used in this example.
Cisco UCM – SIP trunk to Cisco IM&Presence Trunk Configuration

**Navigation Path:** Device → Trunk

Set Device Name*= IMPTrunk. This is used for this example.
Set Description = This text is used to identify this Trunk Group.
Set Device Pool* = Default. This is used for this example.
Set Media Resource Group List = MRGL_SW_MTP. This is used for this example.

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

All other values are default.
Cisco UCM SIP Trunk to CUP 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: < 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
Cisco UCM SIP Trunk to CUP Configuration (Continued)

Set Destination Address = 10.80.16.6. 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.
# Acronyms

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>EXT</td>
<td>Extension</td>
</tr>
<tr>
<td>UDP</td>
<td>Uniform Dial Plan</td>
</tr>
<tr>
<td>AAR</td>
<td>Automatic Alternate route</td>
</tr>
<tr>
<td>FAC</td>
<td>Feature Access Code</td>
</tr>
<tr>
<td>PBX</td>
<td>Private Branch Exchange</td>
</tr>
<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>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>Cisco UCM</td>
<td>Cisco Unified Communications Manager</td>
</tr>
<tr>
<td>DNS</td>
<td>Domain Name Server</td>
</tr>
<tr>
<td>FQDN</td>
<td>Fully Qualified Domain Name</td>
</tr>
<tr>
<td>MWI</td>
<td>Message Waiting Indicator</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>PSTN</td>
<td>Public Switched Telephone Network</td>
</tr>
<tr>
<td>RTP</td>
<td>Real Time Protocol</td>
</tr>
<tr>
<td>SIP</td>
<td>Session Initiated Protocol</td>
</tr>
</tbody>
</table>