Avaya S8300 Release CM 6.0.1 using SIP trunk to Cisco Unified Communications Manager Release 10.0
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Introduction

This document describes the steps and configurations necessary for Cisco Unified Communications Manager (Cisco UCM) release 10.0 to interoperate with the Avaya S8300 Communication Manager Release 6.0 and Avaya Aura Session Manager Release 6.1 using SIP Early-Offer.

The following items were tested:

- Basic call between the two systems and verification of voice path, using both SIP and digital phones 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, and early attended.
- 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
- 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 unless noted.
- CLIR/CNIR—The Avaya SIP trunk does not support calling/connected name and number restriction. Restriction of calling number on Avaya digital 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. Cisco UCM, on the other hand, restricts the connected name and number information the same way as for calling name and number restriction—by setting the SIP PRIVACY to “id” in the SIP trunk configuration page. Thus, the SIP privacy setting covers all outgoing message presentation restrictions, whether for inbound or outbound calls.

• Both systems support call forwarding (CFU, CFB, and CFNA) features. There are some call forward scenarios where the calling name and number are not updated after the call has been forwarded. This issue is found primarily when an Avaya phone is either the originating or terminating end. The Avaya phones (IP or legacy) do not display the forwarding phone’s name and number information, for a local forwarded call. Cisco phones display the forwarding information only when it is a locally forwarded call.

• Video Call Transfer and Video Conference call failures, appeared to be due to Avaya not responding to a re-INVITE sent from CUCM after the redirection, thus there is no audio after the redirection and the call drops.

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

Phone B2 X3106
H323
Avaya One X
Communicator

Phone A3
X3303 SIP
Avaya One X
Communicator

Phone A4
X3300 SIP
Avaya One X
Communicator

Avaya S8300
Communication
Manager R6.0

Avaya Aura
Session
manager R6.1

Cisco Catalyst 3750
Ethernet Switch

Cisco UCM
Release 10.0

Cisco UCM
IM&P
Release 10.0

Cisco Unity
Connection
Release 10.0

Dial Plans
3XXX

5XXX

SIP Trunk
(TCP)

Phone A1
X3302 SIP

Phone B1
SIP X3300

Phone C1
X5004
Cisco 7975

Phone D1
X5010
Cisco 7961

Phone E1
X5007
Cisco 7975

Phone D2
X5017
Cisco 9971

Phone C2
X5000
Cisco 9971

Phone C3
X5013
Cisco EX60
Telepresence

Phone A
X3300 SIP

Phone B
H323 X3105

Phone A
X3302 SIP

Phone B1 SIP X3300
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 to the INVITE with their preferred Codec Preference for the call.
- Avaya experienced Music on Hold, DTMF to Voice Mail, and one way audio on conference calls where the distant end (CUCM) couldn’t hear. Avaya Media Server had to be powered off by unplugging the AC cord and back on. Once this was performed above issues were resolved. This occurred twice during testing.
- For Extend & Connect Remote Destination to receive Voicemail when they are busy or call forward unconditional, Their Remote Destination Timer Information had to be set to 0.0.

<table>
<thead>
<tr>
<th>Timer Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Wait 4.0 seconds before ringing this phone when my business line is dialed.</td>
</tr>
<tr>
<td>Prevent this call from going straight to this phone's voicemail by using a time delay of 0.0 seconds to detect when calls go straight to voicemail.</td>
</tr>
<tr>
<td>Stop ringing this phone after 19.0 seconds to avoid connecting to this phone's voicemail.</td>
</tr>
</tbody>
</table>

- Video Call Transfer and Conference call failures, appeared to be due to Avaya not responding to a re-INVITE sent from CUCM after the redirection, thus there is no audio after the redirection and the call drops.
System Components

Hardware Requirements
The following hardware was used

- Cisco UCS-C240-M3S VMWare Host
- Catalyst switch 3750 WS-C3750X-48
- Cisco 7961, 7975, and 9971 IP phones
- Cisco EX60 Telepresence
- Avaya S8300D PBX with G6430 Media Gateway
- Avaya Common Server HP DL360 G7 with Session Manager
- Avaya Common Server HP DL360 G7 with System Manager
- Avaya Common Server HP DL360 G7 with Session border Controller

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 10.0
- Cisco Unified Communications Manager IM & P release 10.0
- Catalyst 3750 Cisco IOS Software, C3750E Software (C3750E-UNIVERSALK9-M), Version 12.2(55)SE5
- Cisco Unity Connection release 10.0
- Avaya Communication Manager release 6.01 Service Pack 11(patch 20685) (System Platform 6.0.3.10.3)
- Avaya G430 Media Gateway firmware release 30.12.1
- Avaya Aura® Session Manager R6.1 (6.1.2.0.612004) Service Pack 2
- Avaya Aura® System Manager R6.1 (System Platform 6.0.3.0.3, Template 6.1.5.0) Service Pack 2
- Avaya Aura® Session Border Controller 6.1 (System Platform 6.0.3.0.3, Template E362P4)
- Avaya One-X Communicator Release 6.1
Features

This section lists supported and unsupported features. Please see the Limitations section on page 7 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 (lamp ON, lamp OFF).
- Audio Codec Preference List
- Video
Features Not Supported or Not Tested

- Call completion (callback, automatic callback).
- Inter-working Test Cases with Various Calling/Connected Name and Number.
- Shared Line - Hold & Resume with MOH
- Call Park/Pickup
- Interworking Test Cases for Call Transfer

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 S8300 PBX:

1. Configure the IP-Codec-Set, and IP-Network-Region.
2. Configure the IP interface for C-LAN and IP Media Processor cards.
3. Configure Cisco UCM as an IP node-name.
4. Configure the signaling group for the SIP trunk to Cisco UCM.
5. Configure the trunk group for the SIP trunk to Cisco UCM.
6. Configure the SIP and digital station phone extension.
7. Configure the uniform dialing plan to the Cisco UCM extensions.
8. 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. Partitions and calling search space.
5. Assign media resource group list (MRGL) in the default device pool.
6. SIP trunk to Avaya S8300 PBX.
7. SIP Trunk Normalization Script
8. SIP Trunk to Cisco Unity
9. Assign User in Cisco Unity
10. SIP and SCCP phones device configuration.
11. Route pattern to the Avaya S8300 PBX.
12. CallManager Service Parameter “Duplex Streaming Enabled” set to “True”.
13. Audio Codec Preference Configuration
14. Region Configuration
## Configuring the Avaya S8300

### Avaya S8300D Software Version and Hardware Configuration List

<table>
<thead>
<tr>
<th>Board Number</th>
<th>Board Type</th>
<th>Code</th>
<th>Vintage</th>
<th>Assigned Ports</th>
</tr>
</thead>
<tbody>
<tr>
<td>001V1</td>
<td>ICC MM</td>
<td>S8300D</td>
<td>HW01 FW001</td>
<td>u u u u u u u</td>
</tr>
<tr>
<td>001V2</td>
<td>DCP MM</td>
<td>MM712AP</td>
<td>HW05 FW009</td>
<td>01 02 03 04 05 06 07 08</td>
</tr>
<tr>
<td>001V9</td>
<td>MG-ANNOUNCEMENT</td>
<td>VMM-ANN</td>
<td></td>
<td>09 10 11 12 13 14 15 16</td>
</tr>
</tbody>
</table>
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
display system-parameters customer-options

OPTIONAL FEATURES

IP PORT CAPACITIES

Max Administered H.323 Trunks: 4000 0
Max Concurrently Registered IP Stations: 2400 1
Max Administered Remote Office Trunks: 4000 0
Max Concurrently Registered Remote Office Stations: 2400 0
Max Concurrently Registered IP eCons: 50 0
Max Concur Registered Unauthenticated H.323 Stations: 100 0

Max Video Capable Stations: 2400 3
Max Video Capable IP Softphones: 4 3
Max Administered SIP Trunks: 4000 40

Max Administered Ad-hoc Video Conferencing Ports: 4000 0
Max Number of DS1 Boards with Echo Cancellation: 80 0
Max TN2501 VAL Boards: 10 0
Max Media Gateway VAL Sources: 50 1
Max TN2602 Boards with 80 VoIP Channels: 128 0
Max TN2602 Boards with 320 VoIP Channels: 128 0
Max Number of Expanded Meet-me Conference Ports: 0 0

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

F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
display system-parameters customer-options

OPTIONAL FEATURES

Abbreviated Dialing Enhanced List? y
Access Security Gateway (ASG)? n
Analog Trunk Incoming Call ID? y
A/D Grp/Sys List Dialing Start at 01? y
Answer Supervision by Call Classifier? y
ARS? y
ARS/AAR Partitioning? y
ARS/AAR Dialing without FAC? y
ASAI Link Core Capabilities? n
ASAI Link Plus Capabilities? n
Async. Transfer Mode (ATM) PNC? n
Async. Transfer Mode (ATM) Trunking? n
ATM WAN Spare Processor? n
ATMS? y
Attendant Vectoring? y

Audible Message Waiting? y
Authorization Codes? y
CAS Branch? n
CAS Main? n
Change COR by FAC? n
Computer Telephony Adjunct Links? y
Cvg Of Calls Redirected Off-net? y
DCS (Basic)? y
DCS Call Coverage? y
DCS with Rerouting? y
Digital Loss Plan Modification? y
DS1 MSP? y
DS1 Echo Cancellation? y

(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
**CPN/ANI/ICLID PARAMETERS**
- CPN/ANI/ICLID Replacement for Restricted Calls: **anonymous**
- CPN/ANI/ICLID Replacement for Unavailable Calls: **anonymous**

**DISPLAY TEXT**
- Identity When Bridging: **principal**
- User Guidance Display? **n**
- Extension only label for Team button on 96xx H.323 terminals? **n**

**INTERNATIONAL CALL ROUTING PARAMETERS**
- Local Country Code: **___**
- International Access Code: **___**

**ENBLOC DIALING PARAMETERS**
- Enable Enbloc Dialing without ARS FAC? **n**

**CALLER ID ON CALL WAITING PARAMETERS**
- Caller ID on Call Waiting Delay Timer (msec): **200**

[F1=Cancel F2=Refresh F3=Submit F4=Clr F5=Help F6=Update F7=Nxt Pg F8=Prv Pg]
**Config IP Codec Set and IP Network Region:**

Codec set 1 is configured for this test.

Audio Codec G711MU and G.729 are selected 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.
IP Codec Set

Allow Direct-IP Multimedia? 
Maximum Call Rate for Direct-IP Multimedia: 4096:Kbits
Maximum Call Rate for Priority Direct-IP Multimedia: 4096:Kbits

<table>
<thead>
<tr>
<th>Mode</th>
<th>Redundancy</th>
</tr>
</thead>
<tbody>
<tr>
<td>FAX</td>
<td>t.38-standard 0</td>
</tr>
<tr>
<td>Modem</td>
<td>off        0</td>
</tr>
<tr>
<td>TDD/TTY</td>
<td>US         3</td>
</tr>
<tr>
<td>Clear-channel</td>
<td>n          0</td>
</tr>
</tbody>
</table>
Configure IP-Network-region 1:

Location: 1

Authoritative Domain: lab.tekvizion.com

Name: tekvizion

Codec Set: 1 which programmed in previous step

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

H.323 SECURITY PROFILES: any-auth
Configure the Signaling group and trunk Group

Configure the Node IP for Avaya Session manager and CM

<table>
<thead>
<tr>
<th>Name</th>
<th>IP Address</th>
</tr>
</thead>
<tbody>
<tr>
<td>M1</td>
<td>10.70.2.6</td>
</tr>
<tr>
<td>default</td>
<td>0.0.0.0</td>
</tr>
<tr>
<td>gateway</td>
<td>10.70.2.1</td>
</tr>
<tr>
<td>msgserver</td>
<td>10.70.2.14</td>
</tr>
<tr>
<td>procr</td>
<td>10.70.2.14</td>
</tr>
<tr>
<td>procr6</td>
<td>::</td>
</tr>
</tbody>
</table>

(6 of 6 administered node-names were displayed)

Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
Configure the Signaling Group 4:
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: SM1
Near-end Listen Port: 5060
Far-end Listen Port: 506
Far-end Network Region: 1
DTMF over IP: rtp-payload
Direct IP-IP Audio Connections? Y
Configure trunk group 4:

Group number: 4

Group Type: sip

Group Name: SIP to Cisco

TAC: *104

Member Assignment Method: auto

Service Type: tie

Signaling Group: 4

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
change trunk-group 4

Group Type: sip

TRUNK PARAMETERS

Unicode Name: no

Redirect On OPTIM Failure: 5000

SCCAN? n

Digital Loss Group: 18

Preferred Minimum Session Refresh Interval(sec): 900

Disconnect Supervision - In? v Out? v

XOIP Treatment: auto Delay Call Setup When Accessed Via IGAR? n
change trunk-group 4
TRUNK FEATURES

ACA Assignment? y
Measured: none
Maintenance Tests? y

Numbering Format: private
UUI Treatment: service-provider

Replace Restricted Numbers? y
Replace Unavailable Numbers? n

Modify Tandem Calling Number: no

Show ANSWERED BY on Display? n

F1=Cancel  F2=Refresh  F3=Submit  F4=Clr Fld  F5=Help  F6=Update  F7=Nxt Pg  F8=Prv Pg
PROTOCOL VARIATIONS

Mark Users as Phone? n
Prepend \'+\' to Calling Number? n
Send Transferring Party Information? n

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

Convert 180 to 183 for Early Media? n
Always Use re-INVITE for Display Updates? n
Identity for Calling Party Display: From
Enable Q-SIP? n

F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
Configure Route pattern:

Pattern Number: 4

Pattern name: Cisco

Grp No: 4

FRL: 0

ITC: unre

Numbering Format: lev0-pvt
**Dialing plan:**

Configure 4 digits number start with 31 and 33 as ext

Configure 4 digit number start with 5 as udp

8 and 9 are set as 1 digit fac code.

<table>
<thead>
<tr>
<th>Dialed String</th>
<th>Total Length</th>
<th>Call Type</th>
<th>Dialed String</th>
<th>Total Length</th>
<th>Call Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>31</td>
<td>1</td>
<td>ext</td>
<td>3234</td>
<td>4</td>
<td>ext</td>
</tr>
<tr>
<td>32</td>
<td>4</td>
<td>ext</td>
<td>3234</td>
<td>4</td>
<td>ext</td>
</tr>
<tr>
<td>33</td>
<td>4</td>
<td>ext</td>
<td>3234</td>
<td>4</td>
<td>ext</td>
</tr>
<tr>
<td>34</td>
<td>4</td>
<td>fac</td>
<td>3234</td>
<td>4</td>
<td>ext</td>
</tr>
<tr>
<td>4</td>
<td>4</td>
<td>ext</td>
<td>3234</td>
<td>4</td>
<td>ext</td>
</tr>
<tr>
<td>5</td>
<td>4</td>
<td>udp</td>
<td>3234</td>
<td>4</td>
<td>ext</td>
</tr>
<tr>
<td>6</td>
<td>1</td>
<td>udp</td>
<td>3234</td>
<td>4</td>
<td>ext</td>
</tr>
</tbody>
</table>
Configure the AAR dialplan:

Set 4 digits dial string start with 2302(Unity mail), 330(Avaya SIP phone) and 5(Cisco phone) to use Route pattern 4 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 31 and 33 via trunk group 4 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 dialplan:**

**Matching Pattern** Enter the number Communication Manager matches to dialed numbers. Accepts up to seven digits. 33 and 5 are used in the example

**Len** Enter the number of user-dialed digits the system collects to match to this Matching Pattern value. 4 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

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 10.70.2.4. Use admin as User ID and associated password, and then “Log on”

Navigation: Home→Elements→Routing
Add Domains
Under page Domain Management:

Name: lab.tekvizion.com
Type: sip
Add Location
Name: Dallas
Add Adaptations

Adaptation for Cisco CUCM

Adaptation name: Cisco_CUCM10

Module name: CiscoAdapter

Module Parameter: fromto=true odstd=10.80.10.3 iosrcd=lab.tekvizion.com
Adaptation for Avaya Aura CM

Adaptation name: Avaya_CM
Module name: DigitConversionAdapter
Module Parameter: fromto=true
Add SIP Entities and Entity Link

SIP Entity for Session Manager

Name: teksm

FQDN or IP Address: 10.70.2.6

Type: Session Manager

Location: Dallas

Time Zone: America/Chicago

SIP Link Monitoring: Use Session manager Configuration
SIP Entity and entity Link for CUCM
Name: Cisco_CUCM10
FQDN or IP Address: 10.80.10.3
Type: Other
Adaptation: Cisco_CUCM10
Location: Dallas
Time Zone: America/Chicago

SIP Link Monitoring: Use Session Manager Configuration
SIP Entity and Entity Link for Avaya Aura Communication manager

Name: tekcm
FQDN or IP Address: 10.70.2.14
Type: CM
Adaptation: Avaya_CM
Location: Dallas
Time Zone: Chicago

Sip Link Monitoring: Use Session Manager Configuration
Add Entity Links
Add entity link between Avaya Session manager and Cisco CUCM:
Name: ASM to CUCM10
SIP Entity 1: teksm
Protocol: tcp
Port 5060
SIP Entity 2: Cisco_CUCM10
Port 5060
Trusted: checked
Add entity link between Avaya Session manager and Avaya Aura Communication Manager:

Name: teksm_tekcm_5060_TCP

SIP Entity 1: teksm

Protocol: tcp

Port 5060

SIP Entity 2: tekcm

Port 5060

Trusted: checked
Add Routing Policies

Routing policy for call to go to Cisco CUCM
Name: to Cisco CUCM10

Select SIP Entity “Cisco_CUCM10” for SIP Entity as Destination

Routing Policy for calls to go to Avaya Aura Communication Manager
Name: To_tekcm

Select SIP Entity “tekcm” for SIP Entity as Destination
Add Dial Pattern

Dial pattern to Cisco CUCM

Pattern: 5
Min: 4
Max: 4
SIP Domain: lab.tekvizion.com
Original Location Name: Dallas
Routing Policy Name: to Cisco CUCM10

**Avaya Aura® System Manager 6.1**

- **Dial Pattern Details**
  - **Pattern:** 5
  - **Min:** 4
  - **Max:** 4
  - **SIP Domain:** lab.tekvizion.com
  - **Notes:** to Cisco CUCM

- **Originating Locations and Routing Policies**
  - **Routing Policy Name:** to Cisco CUCM10

- **Denied Originating Locations**

*Input Required*
Dial Pattern to Avaya Aura Communication Manager

Pattern: 310
Min: 4
Max: 4
SIP Domain: lab.tekvizion.com
Original Location Name: Dallas
Routing Policy Name: to_tekcm
Pattern: 330
Min: 4
Max: 4
SIP Domain: lab.tekvizion.com
Original Location Name: Dallas
Routing Policy Name: to_tekcm
Cisco Unified Communications Manager SIP Trunk Security Profile

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

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

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

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.
### SIP Profile Configuration

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

#### SIP Profile Information

<table>
<thead>
<tr>
<th>Name*</th>
<th>Early Offer SIP Profile</th>
</tr>
</thead>
<tbody>
<tr>
<td>Description</td>
<td>Default Early Offer SIP Profile</td>
</tr>
<tr>
<td>Default MTP Telephony Event Payload Type*</td>
<td>101</td>
</tr>
<tr>
<td>Early Offer for G.729 Calls*</td>
<td>Disabled</td>
</tr>
<tr>
<td>User-Agent and Server header information*</td>
<td>Send Unified CM Version Information as User-Agent</td>
</tr>
<tr>
<td>Version in User Agent and Server Header*</td>
<td>Major And Minor</td>
</tr>
<tr>
<td>Dial String Interpretation*</td>
<td>Phone number consists of characters 0-9, *, #, and</td>
</tr>
<tr>
<td>Confidential Access Level Headers*</td>
<td>Disabled</td>
</tr>
</tbody>
</table>

- **Disable Early Media on 180**
- Outgoing T.38 INVITE include audio mline
- Use Fully Qualified Domain Name in SIP Requests
- Assured Services SIP conformance

#### SDP Information

<table>
<thead>
<tr>
<th>SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*</th>
<th>TIAS and AS</th>
</tr>
</thead>
<tbody>
<tr>
<td>SDP Transparency Profile</td>
<td>Pass all unknown SDP attributes</td>
</tr>
<tr>
<td>Accept Audio Codec Preferences in Received Offer*</td>
<td>Off</td>
</tr>
</tbody>
</table>

- Require SDP Inactive Exchange for Mid-Call Media Change
Cisco Unified Communications Manager SIP Profile (Continued)

These values are default.
### Parameters used in Phone

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Timer Invite Expires (seconds)</td>
<td>180</td>
</tr>
<tr>
<td>Timer Register Delta (seconds)</td>
<td>5</td>
</tr>
<tr>
<td>Timer Register Expires (seconds)</td>
<td>3600</td>
</tr>
<tr>
<td>Timer T1 (msec)</td>
<td>500</td>
</tr>
<tr>
<td>Timer T2 (msec)</td>
<td>4000</td>
</tr>
<tr>
<td>Retry INVITE*</td>
<td>6</td>
</tr>
<tr>
<td>Retry Non-INVITE*</td>
<td>10</td>
</tr>
<tr>
<td>Start Media Port*</td>
<td>16384</td>
</tr>
<tr>
<td>Stop Media Port*</td>
<td>32766</td>
</tr>
<tr>
<td>Call Pickup URI*</td>
<td>x-cisco-serviceuri-pickup</td>
</tr>
<tr>
<td>Call Pickup Group Other URI*</td>
<td>x-cisco-serviceuri-pickup</td>
</tr>
<tr>
<td>Call Pickup Group URI*</td>
<td>x-cisco-serviceuri-pickup</td>
</tr>
<tr>
<td>Meet Me Service URI*</td>
<td>x-cisco-serviceuri-meetme</td>
</tr>
<tr>
<td>User Info*</td>
<td>None</td>
</tr>
<tr>
<td>DTMF DB Level*</td>
<td>Nominal</td>
</tr>
<tr>
<td>Call Hold Ring Back*</td>
<td>Off</td>
</tr>
<tr>
<td>Anonymous Call Block*</td>
<td>Off</td>
</tr>
<tr>
<td>Caller ID Blocking*</td>
<td>Off</td>
</tr>
<tr>
<td>Do Not Disturb Control*</td>
<td>User</td>
</tr>
<tr>
<td>Telnet Level for 7940 and 7960*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Resource Priority Namespace</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Timer Keep Alive Expires (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Expires (seconds)*</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Delta (seconds)*</td>
<td>5</td>
</tr>
<tr>
<td>Maximum Redirections*</td>
<td>70</td>
</tr>
<tr>
<td>Off Hook To First Digit Timer (milliseconds)*</td>
<td>15000</td>
</tr>
<tr>
<td>Call Forward URI*</td>
<td>x-cisco-serviceuri-cfwdall</td>
</tr>
<tr>
<td>Speed Dial (Abbreviated Dial) URI*</td>
<td>x-cisco-serviceuri-abbrevdial</td>
</tr>
</tbody>
</table>

### Cisco Unified Communications Manager SIP Profile (Continued)

Check RFC 2543 Hold

Set SIP Rel1XX Options* = Send PRACK if 1xx Contains SDP
Check Early Offer support for voice and video calls (insert MTP if needed)

All other values are default.

Cisco Unified Communications Manager SIP Profile (Continued)

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

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

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

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

Set Call Classification* = OnNet. This is used for this example.

Set Media Resource Group List = MRGL_G711. This is used for this example.

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

Set Connected Line ID Presentation* = Allowed
Set Connected Name Presentation* = Allowed
Check Redirecting Diversion Header Delivery - Inbound
All other values are default.
Cisco Unified Communications Manager SIP Trunk to Avaya Configuration (Continued)

Set Calling Line ID Presentation*= Allowed
Set Calling Name Presentation*= Allowed
Set Calling and Connected Party Info Format* = Deliver URI and DN in connected party, if available

Check Redirecting Diversion Header Delivery - Outbound

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

Set Destination Address = 10.70.2.6. This is used in this example.

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

Set SIP Profile* = EarlyOffer SIP Profile

Set DTMF Signaling Method* = RFC 2833

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.
**Cisco Unified Communications Manager SIP Trunk Normalization Script**

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:

```plaintext
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

Set Device Name* = To_Unity_Connection. 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.

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

Check Redirecting Diversion Header Delivery - Outbound

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

Set Destination Address = 10.80.10.5. This is used in this example.

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

Set SIP Profile*= Standard SIP Profile

All other values are default.

<table>
<thead>
<tr>
<th>Destination Address</th>
<th>Destination Address IPv6</th>
<th>Destination Port</th>
<th>Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>10.80.10.5</td>
<td></td>
<td>5060</td>
<td>N/A</td>
</tr>
</tbody>
</table>

- MTP Preferred Originating Codec*: 711ulaw
- BLF Presence Group*: Standard Presence group
- SIP Trunk Security Profile*: Non Secure SIP Trunk to VM Profile
- Reouting Calling Search Space: < None >
- Out-Of-Dialog Refer Calling Search Space: < None >
- SUBSCRIBE Calling Search Space: < None >
- SIP Profile*: Standard SIP Profile
- DTMF Signalling Method*: No Preference

**Normalization Script**

**Enable Trace**

Parameter Name | Parameter Value
--- | ---

**Recording Information**

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

**Geolocation Configuration**

Geolocation: < None >
Geolocation Filter: < None >
Send Geolocation Information
Cisco Unity Connection User 5017 Configuration

Set Alias* = 5017. 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= 5017 cisco. This is used in this example.

Set SMTP Address =5017. This is used in this example.

Set Phone System= Cluster 20. This is used in this example.

All other values are default.
Cisco Unity Connection User 5017 Configuration (Continued)

All values are default.
Cisco Unified Communications Manager Service Parameter

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

<table>
<thead>
<tr>
<th>Clusterwide Parameters (External QoS)</th>
<th>False</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default Network Hold MOH Audio Source ID</td>
<td>1</td>
</tr>
<tr>
<td>Default User Hold MOH Audio Source ID</td>
<td>1</td>
</tr>
<tr>
<td>Duplex Streaming Enabled*</td>
<td>True</td>
</tr>
<tr>
<td>Media Exchange Interface Capability Timer</td>
<td>8</td>
</tr>
<tr>
<td>Send Multicast MOH in H.245 QLC Message*</td>
<td>True</td>
</tr>
<tr>
<td>Media Exchange Timer*</td>
<td>12</td>
</tr>
<tr>
<td>Media Exchange Stop Streaming Timer</td>
<td>8</td>
</tr>
<tr>
<td>Open Video Channel Response Timer for SIP Interop*</td>
<td>500</td>
</tr>
<tr>
<td>Port Received Timer After Call Connection*</td>
<td>500</td>
</tr>
<tr>
<td>Media Resource Allocation Timer*</td>
<td>12</td>
</tr>
<tr>
<td>MTP and Transcoder Resource Throttling Percentage</td>
<td>95</td>
</tr>
<tr>
<td>Intercluster Capabilities Mismatch Timer</td>
<td>1000</td>
</tr>
<tr>
<td>Silence Suppression</td>
<td>False</td>
</tr>
<tr>
<td>Silence Suppression for Gateways*</td>
<td>False</td>
</tr>
<tr>
<td>Strip G.729 Annex B (Silence Suppression) from Capabilities*</td>
<td>False</td>
</tr>
<tr>
<td>Enable Source IP Address Verification</td>
<td>True</td>
</tr>
</tbody>
</table>

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

<table>
<thead>
<tr>
<th>Name</th>
<th>Description</th>
<th>Multi-cast</th>
<th>Copy</th>
</tr>
</thead>
<tbody>
<tr>
<td>MRG MTP</td>
<td>MRG with MTP</td>
<td>false</td>
<td></td>
</tr>
<tr>
<td>MRG noMTP</td>
<td>MRG without MTP</td>
<td>false</td>
<td></td>
</tr>
</tbody>
</table>

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

Set Name *= MRG_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.
MRG_MTP Resource Group (Continued)

Media Resource Group Configuration

Status

Status: Ready

Media Resource Group Status

Media Resource Group: MRG_MTP (used by 13 devices)

Media Resource Group Information

Name: MRG_MTP
Description: MRG with MTP

Devices for this Group

Available Media Resources

Selected Media Resources

CFB_2 (CFB)
MOH_2 (MOH)
MOH_3 (MOH)
MOH_4 (MOH)
MTP_2 (MTP)
**MRG_MTP Resource Group (Continued)**

![Cisco Unified CM Administration interface](image)

### Media Resource Group Configuration

<table>
<thead>
<tr>
<th>Name</th>
<th>MRG_MTP</th>
</tr>
</thead>
<tbody>
<tr>
<td>Description</td>
<td>MRG with MTP</td>
</tr>
</tbody>
</table>

#### Devices for this Group

**Available Media Resources**

- MOH_3 (MOH)
- MOH_4 (MOH)
- MTP_2 (MTP)
- MTP_3 (MTP)
- MTP_4 (MTP)

**Selected Media Resources**

- MOH_3 (MOH)
- MOH_4 (MOH)
- MTP_2 (MTP)
- MTP_3 (MTP)
- MTP_4 (MTP)

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

Set Name*= MRG_noMTP  This is used for 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.
Resource Group for MRG noMTP (Continued)

### Media Resource Group Configuration

**Status**
- Status: Ready

**Media Resource Group Status**
- Media Resource Group: MRG_noMTP (used by 29 devices)

**Media Resource Group Information**

<table>
<thead>
<tr>
<th>Name</th>
<th>MRG_noMTP</th>
</tr>
</thead>
<tbody>
<tr>
<td>Description</td>
<td>MRG without MTP</td>
</tr>
</tbody>
</table>

**Devices for this Group**

**Available Media Resources**
- MTP_2
- MTP_3
- MTP_4

**Selected Media Resources**
- CFB_3 (CFB)
- CFB_4 (CFB)
- MOH_2 (MOH)
- MOH_3 (MOH)
- MOH_4 (MOH)
## Cisco Unified Communications Manager Media Resource Group List

### Find and List Media Resource Group Lists

- Add New
- Select All
- Clear All
- Delete Selected

### Status

3 records found

### Media Resource Group List (1 - 3 of 3)

<table>
<thead>
<tr>
<th>Name</th>
<th>Copy</th>
</tr>
</thead>
<tbody>
<tr>
<td>MRGL Default</td>
<td><img src="#" alt="Copy" /></td>
</tr>
<tr>
<td>MRGL 6711</td>
<td><img src="#" alt="Copy" /></td>
</tr>
<tr>
<td>MRGL 6729</td>
<td><img src="#" alt="Copy" /></td>
</tr>
</tbody>
</table>

Add New | Select All | Clear All | Delete Selected
Set Name*= MRGL_G711 This is used for this example.

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

Set Available Media Resources = MTP_2, MTP_3 and MTP_4

Set Selected Media Resource Groups= MRG_MTP

Note: This Media Resource Group List was added to provide early offer on the invite from Cisco to Avaya for SCCP phones.
Cisco Unified Communications Manager Route Pattern to Avaya
Set Route Pattern* = 3XXX 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* = Trunk_to_Avaya_SM. This is used for this example.
Uncheck Provide Outside Dial Tone
Set Calling Party Transform Mask = XXXX
Set Calling Line ID Presentation= Allowed
Set Calling Name Presentation= Allowed
All other values are default.
Route Pattern Configuration for 3xxx (Continued)

Set Connected Line ID Presentation* = Allowed

Set Calling Name Presentation* = Allowed

All other values are default.
### Cisco Unified Communications Manager SIP Phone Ext. 5000 Device Level Configuration

#### Phone Table

<table>
<thead>
<tr>
<th>Device Name(Time)</th>
<th>Description</th>
<th>Device Pool</th>
<th>Device Protocol</th>
<th>Status</th>
<th>IP Addr Address</th>
<th>Copy</th>
<th>Super Copy</th>
</tr>
</thead>
<tbody>
<tr>
<td>5000</td>
<td>G711 Pool</td>
<td>SIP</td>
<td>Registered with class2web1 16.80.10.36</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>5013</td>
<td>G711 Pool</td>
<td>SIP</td>
<td>Registered with class2web1 16.80.10.32</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Cisco Framework User2</td>
<td>G711 Pool</td>
<td>SIP</td>
<td>Registered with class2web1 16.64.1.128</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>5004</td>
<td>G711 Pool</td>
<td>SCCP</td>
<td>Registered with class2web1 16.80.10.23</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>5017</td>
<td>G711 Pool</td>
<td>SIP</td>
<td>Registered with class2web1 16.80.10.35</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>5010</td>
<td>G711 Pool</td>
<td>SCCP</td>
<td>Registered with class2web1 16.80.10.24</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>CTI Avatar 1</td>
<td>CTI Avatar Device 1</td>
<td>G711 Pool</td>
<td>CTI Remote Device</td>
<td>Registered with class2web1 None</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>CTI Avatar 2</td>
<td>CTI Avatar Device 2</td>
<td>G711 Pool</td>
<td>CTI Remote Device</td>
<td>Registered with class2web1 None</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
Cisco Unified Communications Manager SIP Phone Ext. 5000 Device Level Configuration (Continued)

Set MAC Address* = 1C17D337D1C9. This is used in this example.

Set Description = This text is used to identify this Phone

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

Set Phone Button Template* = Standard 9971 SIP. This is used in this example.

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

Set User Hold MOH Audio Source = 1-SampleAudioSource.

Set Network Hold MOH Audio Source = 1-SampleAudioSource.

All other values are default.
Cisco Unified Communications Manager SIP Phone Ext. 5000 Device Level Configuration (Continued)

Set Owner User ID*= User1. Leave Blank if Phone is not provisioned for Jabber Avatar

Uncheck Logged Into Hunt Group

All other values are default.
Cisco Unified Communications Manager SIP Phone Ext. 5000 Device Level Configuration (Continued)

Set Device Security Profile* = Cisco 9971-Standard SIP Non-Secure Profile. This is used in this example.

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

Set Digest User = user1. If this is not a Jabber Avatar Phone. Leave as none.

All other values are default.
Cisco Unified Communications Manager SIP Phone Ext. 5000 Device Level Configuration (Continued)

All other values are default.
Cisco Unified Communications Manager SIP Phone Ext. 5000 Device Level Configuration (Continued)

All values are default.
Cisco Unified Communications Manager SIP Phone Ext. 5000 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 other values are default.
Set RTCP* = Enabled. This is used in this example.

All other values are default.
Cisco Unified Communications Manager SIP Phone Ext. 5000 Device Level Configuration (Continued)

All values are default.

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port</td>
<td>Enabled</td>
</tr>
<tr>
<td>Link Layer Discovery Protocol (LLDP): PC Port</td>
<td>Enabled</td>
</tr>
<tr>
<td>LLDP Asset ID</td>
<td></td>
</tr>
<tr>
<td>LLDP Power Priority</td>
<td>Unknown</td>
</tr>
<tr>
<td>802.1x Authentication*</td>
<td>User Controlled</td>
</tr>
<tr>
<td>FIPS Mode*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Detect Unified CM Connection Failure *</td>
<td>Normal</td>
</tr>
<tr>
<td>Switch Port Remote Configuration*</td>
<td>Disabled</td>
</tr>
<tr>
<td>PC Port Remote Configuration*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Automatic Port Synchronization</td>
<td>Disabled</td>
</tr>
<tr>
<td>Power Negotiation</td>
<td>Enabled</td>
</tr>
<tr>
<td>Restrict Data Rates*</td>
<td>Disabled</td>
</tr>
<tr>
<td>SSH Access*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Incoming Call Toll Charge Timer</td>
<td>5</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></td>
</tr>
<tr>
<td>Simplified New Call UI*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Enable VXCM VPN for MAC</td>
<td>Disabled</td>
</tr>
<tr>
<td>VXCM VPN Option*</td>
<td>Dual Tunnel</td>
</tr>
<tr>
<td>VXCM Challenge*</td>
<td>Challenge</td>
</tr>
</tbody>
</table>
Cisco Unified Communications Manager SIP Phone Ext. 5000 Device Level Configuration (Continued)

All values are default.
Cisco Unified Communications Manager SCCP Phone Ext. 5010 Device Level Configuration

Set MAC Address* = 001C5856D737. This is used in this example.

Set Description = This text is used to identify this Phone

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

Set Phone Button Template* = Standard 7961 SCCP. This is used in this example.

Set Media Resource Group List = MRGL_G711. This is used in this example.

Set User Hold MOH Audio Source = 1-SampleAudioSource.

Set Network Hold MOH Audio Source = 1-SampleAudioSource.

All other values are default.
Cisco Unified Communications Manager SCCP Phone Ext. 5010 Device Level Configuration (Continued)

All other values are default.
Cisco Unified Communications Manager SCCP Phone Ext. 5010 Device Level Configuration (Continued)
Set Device Security Profile* = Cisco 7961 – Standard SCCP Non-Secure Profile. This is used in this example.

All other values are default.
Cisco Unified Communications Manager SCCP Phone Ext. 5010 Device Level Configuration (Continued)

These values are default.
Cisco Unified Communications Manager SCCP Phone Ext. 5010 Device Level Configuration (Continued)

These values are default.
These values are default.
Cisco Unified Communications Manager SCCP Phone Ext. 5010 Device Level Configuration (Continued)

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

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.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Default</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Accept Audio Codec Preferences in Received Offer</td>
<td>Off</td>
<td>Enables or disables the use of the codec preference list in received offers.</td>
</tr>
</tbody>
</table>
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 Preferred. 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 Preferred. 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.729a 8k. Second choice for this example.

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

G711 Region and G729 Region created in this example.

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

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

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

Set **Audio Codec Preference List**: G711 Preferred.

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

Set **Region**: G729 Region. This is used in this example.

Set **Audio Codec Preference List**: G729 Preferred. This is used in this example.

Set **Maximum Audio Bit Rate**: 8 Kbps (G7.29). This is used in this example.

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

Set Name*= G729 Region. This is used in this example.

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

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

Set Maximum Audio Bit Rate= 8 Kbps (G.729). This is used in this example.

Set Region=G729 Region. This is used in this example.

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

Set Maximum Audio Bit Rate= 64 Kbps (G.7.22, G.7.11). This is used in this example.

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

G711 Pool and G729 Pool created in this example.

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

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

Set Cisco Unified Communications Manager Group* = Default

Set Date/Time Group* = CMLocal

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

Set Media Resource Group List = MRGL_G711. This is used in this example.

All other values are default.
#### Device Pool Settings

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Pool Name</td>
<td>G711 Pool</td>
</tr>
<tr>
<td>Cisco Unified Communications Manager Group*</td>
<td>Default</td>
</tr>
<tr>
<td>Calling Search Space for Auto-registration</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Adjunct CSS</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Reverted Call Focus Priority</td>
<td>Default</td>
</tr>
<tr>
<td>Intercompany Media Services Enrolled Group</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

#### Local Route Group Settings

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Standard Local Route Group</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

#### Roaming Sensitive Settings

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Date/Time Group</td>
<td>CMLocal</td>
</tr>
<tr>
<td>Region*</td>
<td>G711 Region</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>MRGL_G711</td>
</tr>
<tr>
<td>Location</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Network Locale</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>SRST Reference*</td>
<td>Disable</td>
</tr>
<tr>
<td>Connection Monitor Duration***</td>
<td></td>
</tr>
<tr>
<td>Single Button Barge*</td>
<td>Default</td>
</tr>
<tr>
<td>Join Across Lines*</td>
<td>Default</td>
</tr>
<tr>
<td>Physical Location</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Device Mobility Group</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Wireless LAN Profile Group</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

---

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

All values are default.
- **Device Mobility Related Information**

<table>
<thead>
<tr>
<th>Device Mobility Calling Search Space</th>
<th>&lt; None &gt;</th>
</tr>
</thead>
<tbody>
<tr>
<td>AAR Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>AAR Group</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Calling Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Called Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

- **Geolocation Configuration**

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

- **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 (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty 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</td>
<td>Default</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>International</td>
<td>Default</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Unknown</td>
<td>Default</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Subscriber</td>
<td>Default</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>
Cisco Unified Communications Manager Device Pool Configuration (Continued)

All values are default.

### Incoming Called Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level (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>0</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>International Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Unknown Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Subscriber Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

### Phone Settings

- **Caller ID For Calls From This Phone**

  Calling Party Transformation CSS: < None >

### Connected Party Settings

Connected Party Transformation CSS: < None >

### Redirecting Party Settings

Redirecting Party Transformation CSS: < None >
Cisco Unified Communications Manager Device Pool Configuration (Continued)

Set Device Pool Name*= G729 Pool. This is used in this example.

Set Cisco Unified Communications Manager Group*= Default

Set Date/Time Group*= CMLocal

Set Region*= G729 Region. This is used in this example

Set Media Resource Group List=MRGL_G729. This is used in this example.

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

All values are default.

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 of (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></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)

All values are default.

Incoming Called Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level (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>0</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>International</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Unknown Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Subscriber Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

Phone Settings

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

Connected Party Settings
Connected Party Transformation CSS < None >

Redirecting Party Settings
Redirecting Party Transformation CSS < None >
### Acronyms

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>CCBS</td>
<td>Call Completion to Busy Subscriber</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>CFNR</td>
<td>Call Forwarding No Reply</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>SIP</td>
<td>Session Initiated Protocol</td>
</tr>
</tbody>
</table>
Important Information

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