

# **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.

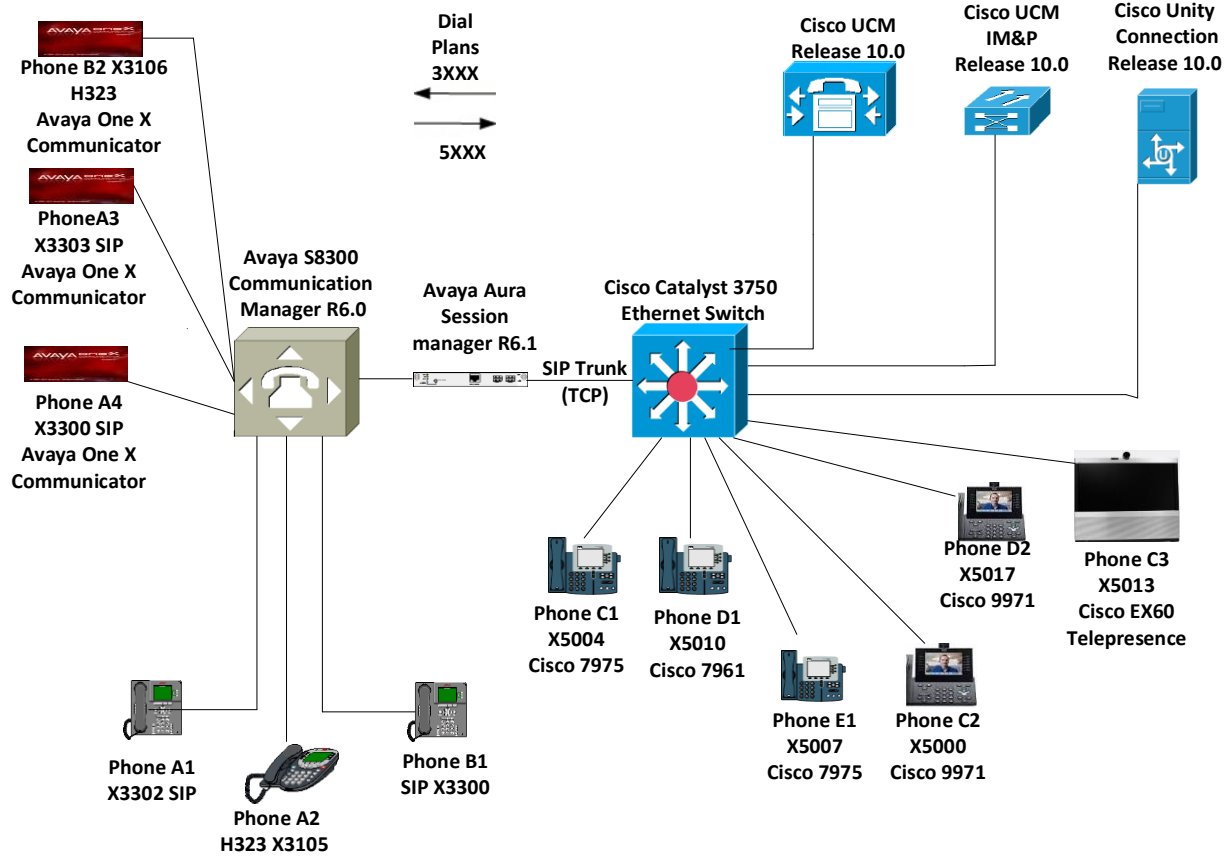
- COLR/CONR—As with calling name and number presentation restrictions, the Avaya PBX does not support connected name and number restriction on SIP trunks. Cisco UCM, on the other hand, restricts the connected name and number information the same way as for calling name and number restriction—by setting 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




## 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.

### Timer Information

Wait\*  seconds before ringing this phone when my business line is dialed.\*

Prevent this call from going straight to this phone's voicemail by using a time delay of\*  conds to detect when calls go straight to voicemail.\*

Stop ringing this phone after\*  seconds to avoid connecting to this phone's voicemail.\*

- 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

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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

```
10.70.2.14 - PuTTY
list configuration all

                          SYSTEM CONFIGURATION

Board
Number   Board Type          Code      Vintage    Assigned Ports
          u=unassigned t=tti p=psa

001V1    ICC MM                 S8300D   HW01 FW001
001V2    DCP MM                 MM712AP  HW05 FW009 01 02 u  u  u  u  u  u
001V9    MG-ANNOUNCEMENT       VMM-ANN  01 02 03 04 05 06 07 08
                                                09 10 11 12 13 14 15 16

Command successfully completed
Command:
F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```

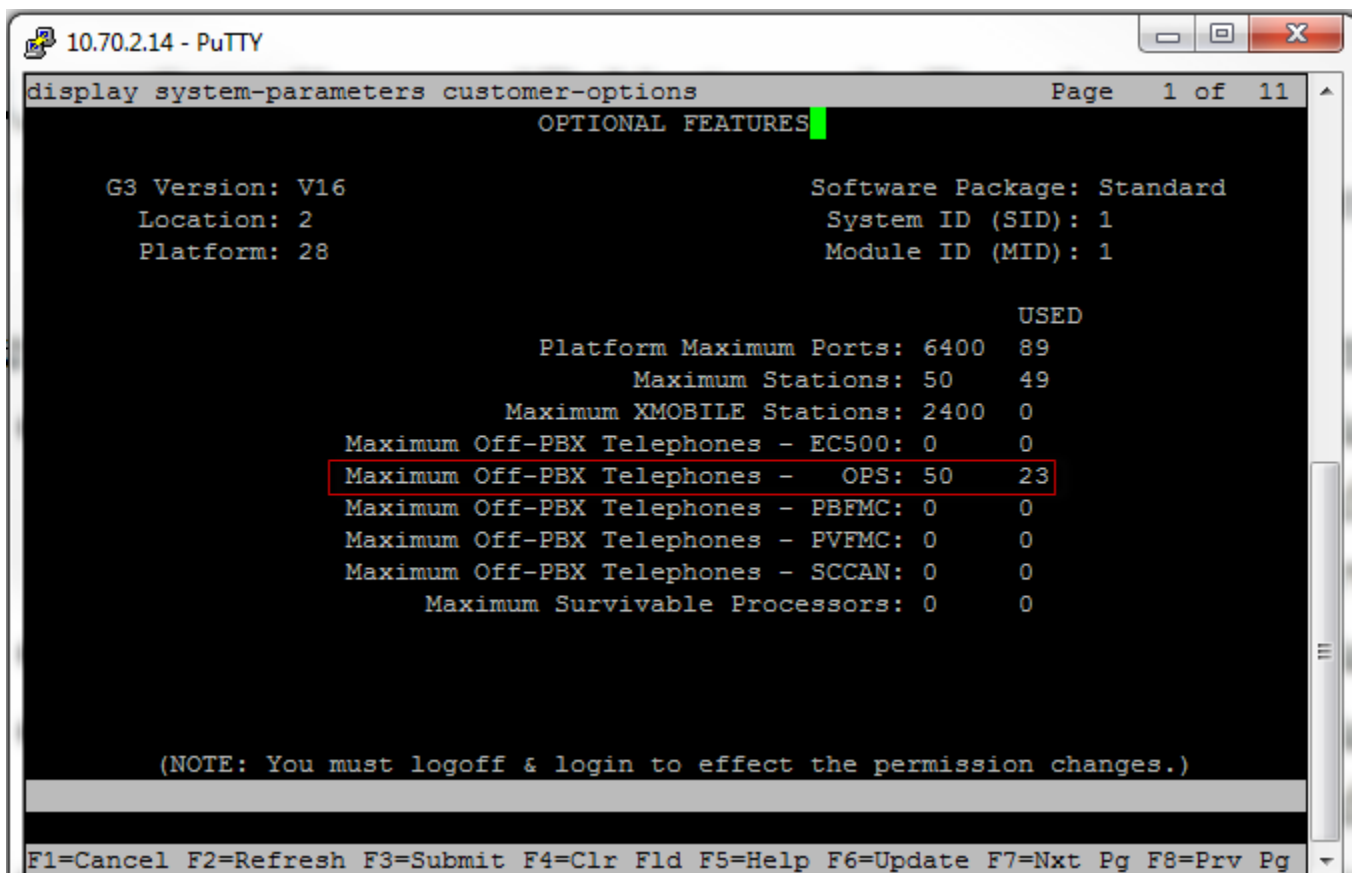
Verify system capacities and licensing:

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



The screenshot shows a PuTTY terminal window titled "10.70.2.14 - PuTTY". The terminal output displays system parameters and optional features. The "OPTIONAL FEATURES" section is highlighted with a green cursor. The output includes:

```
display system-parameters customer-options Page 1 of 11
OPTIONAL FEATURES

G3 Version: V16                               Software Package: Standard
Location: 2                                   System ID (SID): 1
Platform: 28                                  Module ID (MID): 1

                                         USED
Platform Maximum Ports: 6400 89
Maximum Stations: 50 49
Maximum XMOBILE Stations: 2400 0
Maximum Off-PBX Telephones - EC500: 0 0
Maximum Off-PBX Telephones - OPS: 50 23
Maximum Off-PBX Telephones - PBFMC: 0 0
Maximum Off-PBX Telephones - PVFMC: 0 0
Maximum Off-PBX Telephones - SCCAN: 0 0
Maximum Survivable Processors: 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
```

```
10.70.2.14 - PuTTY
display system-parameters customer-options Page 2 of 11
OPTIONAL FEATURES

IP PORT CAPACITIES                               USED
Maximum Administered H.323 Trunks: 4000 0
Maximum Concurrently Registered IP Stations: 2400 1
Maximum Administered Remote Office Trunks: 4000 0
Maximum Concurrently Registered Remote Office Stations: 2400 0
Maximum Concurrently Registered IP eCons: 50 0
Max Concur Registered Unauthenticated H.323 Stations: 100 0
Maximum Video Capable Stations: 2400 3
Maximum Video Capable IP Softphones: 4 3
Maximum Administered SIP Trunks: 4000 40
Maximum Administered Ad-hoc Video Conferencing Ports: 4000 0
Maximum Number of DS1 Boards with Echo Cancellation: 80 0
Maximum TN2501 VAL Boards: 10 0
Maximum Media Gateway VAL Sources: 50 1
Maximum TN2602 Boards with 80 VoIP Channels: 128 0
Maximum TN2602 Boards with 320 VoIP Channels: 128 0
Maximum 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
```

```
10.70.2.14 - PuTTY
display system-parameters customer-options Page 3 of 11
OPTIONAL FEATURES

Abbreviated Dialing Enhanced List? y Audible Message Waiting? y
Access Security Gateway (ASG)? n Authorization Codes? y
Analog Trunk Incoming Call ID? y CAS Branch? n
A/D Grp/Sys List Dialing Start at 01? y CAS Main? n
Answer Supervision by Call Classifier? y Change COR by FAC? n
ARS? y Computer Telephony Adjunct Links? y
ARS/AAR Partitioning? y Cvg Of Calls Redirected Off-net? y
ARS/AAR Dialing without FAC? y DCS (Basic)? y
ASAI Link Core Capabilities? n DCS Call Coverage? y
ASAI Link Plus Capabilities? n DCS with Rerouting? y
Async. Transfer Mode (ATM) PNC? n Digital Loss Plan Modification? y
Async. Transfer Mode (ATM) Trunking? n DS1 MSP? y
ATM WAN Spare Processor? n DS1 Echo Cancellation? y
ATMS? y
Attendant Vectoring? y

(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
```

## Configure System Feature:

On page 1,

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

Set **CPN/ANI/ICLID Replacement for Restricted/Unavailable calls** to anonymous

```
10.70.2.14 - PuTTY
change system-parameters features Page 1 of 19
FEATURE-RELATED SYSTEM PARAMETERS
Self Station Display Enabled? y
Trunk-to-Trunk Transfer: all
Automatic Callback with Called Party Queuing? y
Automatic Callback - No Answer Timeout Interval (rings): 3
Call Park Timeout Interval (minutes): 10
Off-Premises Tone Detect Timeout Interval (seconds): 20
AAR/ARS Dial Tone Required? y

Music (or Silence) on Transferred Trunk Calls? all
DID/Tie/ISDN/SIP Intercept Treatment: attd
Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
Automatic Circuit Assurance (ACA) Enabled? n

Abbreviated Dial Programming by Assigned Lists? n
Auto Abbreviated/Delayed Transition Interval (rings): 2
Protocol for Caller ID Analog Terminals: Bellcore
Display Calling Number for Room to Room Caller ID Calls? n

F1=Cancel F2=Refresh F3=Submit F4=Clr F1d F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```



```
10.70.2.14 - PuTTY
change system-parameters features Page 9 of 19
FEATURE-RELATED SYSTEM PARAMETERS
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 Fld 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 select codec

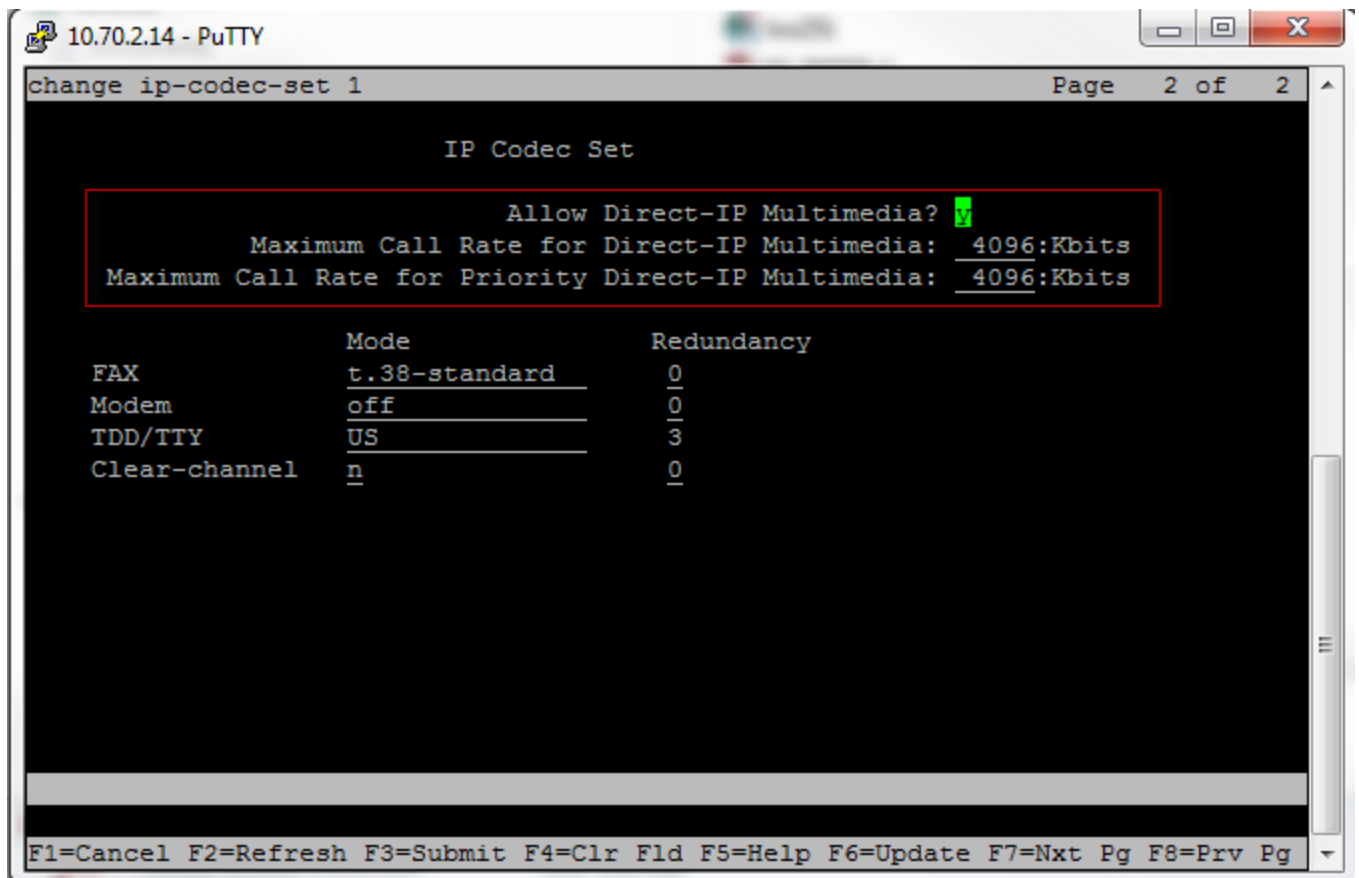
Media Encryption is set to none

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

```
10.70.2.14 - PuTTY
change ip-codec-set 1
Page 1 of 2
IP Codec Set
Codec Set: 1
Audio Codec      Silence Suppression  Frames Per Pkt  Packet Size (ms)
1: G.711MU       n                 2               20
2: G.729         n                 2               20
3:               -                 -               -
4:               -                 -               -
5:               -                 -               -
6:               -                 -               -
7:               -                 -               -
Media Encryption
1: none
2:
3:
F1=Cancel F2=Refresh F3=Submit F4=Clr F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```



## Configure IP-Network-region 1:

Location:1

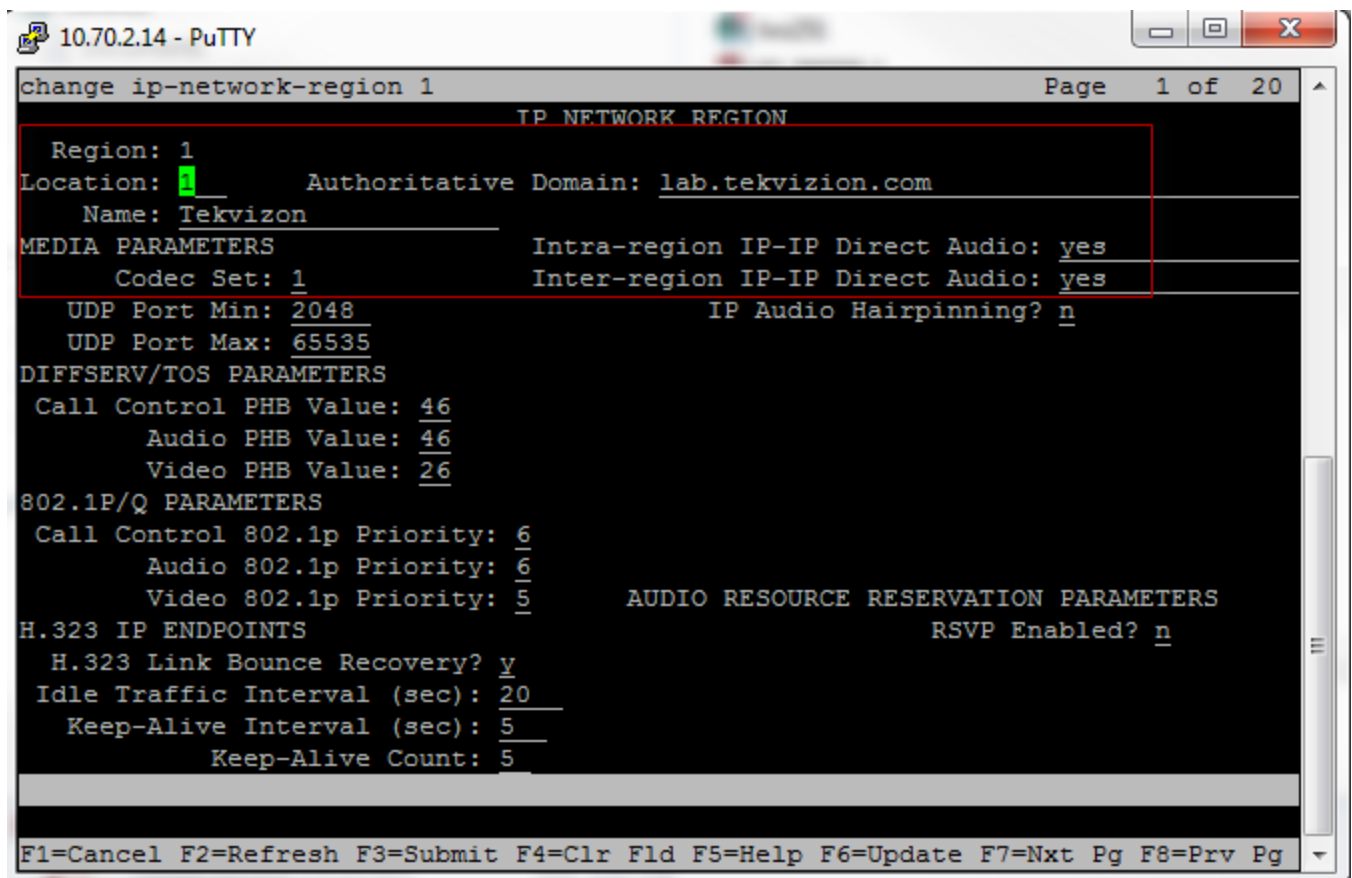
Authoritative Dimain:lab.tekvizion.com

Name:tekvizion

Codec Set: 1 which programmed in previous step

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

H.323 SECURITY PROFILES: any-auth



```
change ip-network-region 1 Page 1 of 20
IP NETWORK REGION
Region: 1
Location: 1 Authoritative Domain: lab.tekvizion.com
Name: Tekvizion
MEDIA PARAMETERS Intra-region IP-IP Direct Audio: yes
Codec Set: 1 Inter-region IP-IP Direct Audio: yes
UDP Port Min: 2048 IP Audio Hairpinning? n
UDP Port Max: 65535
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
AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS RSVP Enabled? n
H.323 Link Bounce Recovery? y
Idle Traffic Interval (sec): 20
Keep-Alive Interval (sec): 5
Keep-Alive Count: 5
F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```

```
10.70.2.14 - PuTTY
change ip-network-region 1 Page 3 of 20
IP NETWORK REGION

INTER-GATEWAY ALTERNATE ROUTING / DIAL PLAN TRANSPARENCY
Incoming LDN 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)
1 _____
2 _____
3 _____
4 _____
5 _____
6 _____

H.323 SECURITY PROFILES
1 any-auth
2 _____
3 _____
4 _____

Allow SIP URI Conversion? y

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: 61444

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

10.70.2.14 - PuTTY

change ip-network-region 1 Page 4 of 20

Source Region: 1 Inter Network Region Connection Management I M

dst codec direct WAN-BW-limits Video Intervening Dyn A G t

rgn set WAN Units Total Norm Prio Shr Regions CAC R L e

1	1											all
2	1	y	NoLimit							n		t
3	1	y	NoLimit							n		t
4	4	y	NoLimit							n		t
5												
6												
7												
8												
9												
10												
11												
12												
13												
14												
15												

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

## Configure the Signaling group and trunk Group

## Configure the Node IP for Avaya Session manager and CM

```
change node-names ip                                     Page 1 of 2
IP NODE NAMES
Name            IP Address
SM1             10.70.2.6
default         0.0.0.0
gateway         10.70.2.1
msgserver       10.70.2.14
procr           10.70.2.14
procr6          ::

```

( 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

F1=Cancel F2=Refresh F3=Submit F4=Clr F1d F5=Help F6=Update F7=Nxt Pg F8=Prv Pg

**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



```
10.70.2.14 - PuTTY
change signaling-group 4 Page 1 of 1
SIGNALING GROUP
Group Number: 4 Group Type: sip
IMS Enabled? n Transport Method: tcp
Q-SIP? n SIP Enabled LSP? n
IP Video? y Priority Video? y Enforce SIPS URI for SRTP? n
Peer Detection Enabled? y Peer Server: SM
Near-end Node Name: procr Far-end Node Name: SM1
Near-end Listen Port: 5060 Far-end Listen Port: 5060
Far-end Network Region: 1
Far-end Domain: lab.tekvizion.com
Incoming Dialog Loopbacks: eliminate Bypass If IP Threshold Exceeded? n
DTMF over IP: rtp-payload RFC 3389 Comfort Noise? n
Direct IP-IP Audio Connections? y
Session Establishment Timer (min): 3 IP Audio Hairpinning? n
Enable Layer 3 Test? y Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n Alternate Route Timer (sec): 6
F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```

**Configure trunk group 4:**

Group number: 4

Group Type:sip

Group Name:SIP to Cisco

TAC:\*104

Member Assignment Method:auto

Service Type:tieSignaling 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

```
10.70.2.14 - PuTTY
change trunk-group 4 Page 1 of 21
TRUNK GROUP
Group Number: 4 Group Type: sip CDR Reports: y
Group Name: SIP to Cisco COR: 1 TN: 1 TAC: *104
Direction: two-way Outgoing Display? n
Dial Access? n Night Service:
Queue Length: 0
Service Type: tie Auth Code? n
Member Assignment Method: auto
Signaling Group: 4
Number of Members: 10
F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```

```
10.70.2.14 - PuTTY
change trunk-group 4 Page 2 of 21
  Group Type: sip

TRUNK PARAMETERS

Unicode Name: o

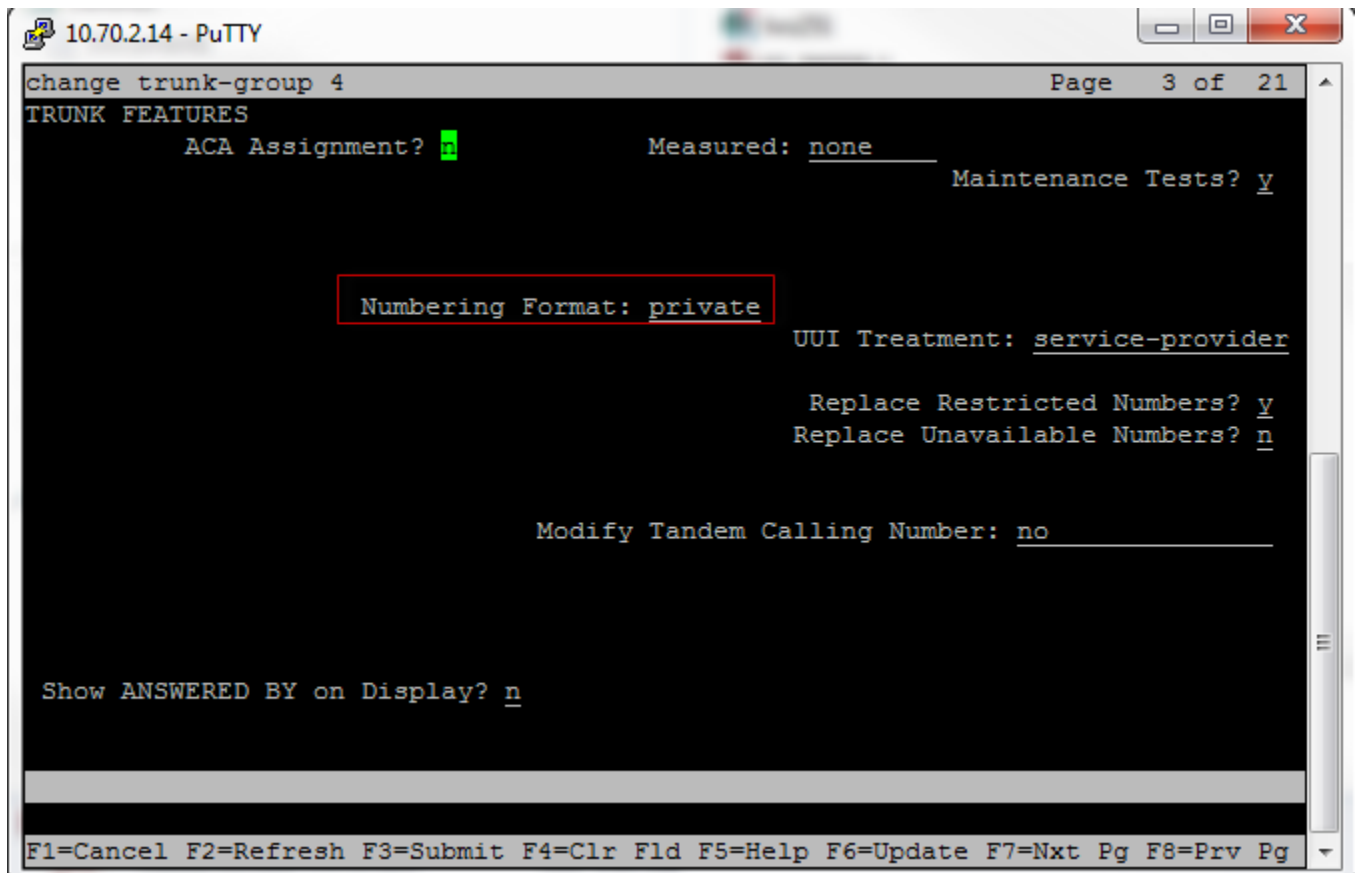
Redirect On OPTIM Failure: 5000

SCCAN? n Digital Loss Group: 18
Preferred Minimum Session Refresh Interval(sec): 900

Disconnect Supervision - In? y Out? y

XOIP Treatment: auto Delay Call Setup When Accessed Via IGAR? n

F1=Cancel F2=Refresh F3=Submit F4=Clr F1d F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```



```
10.70.2.14 - PuTTY
change trunk-group 4 Page 4 of 21
PROTOCOL VARIATIONS
  Mark Users as Phone? y
    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

```
change route-pattern 4 Page 1 of 3
Pattern Number: 4 Pattern Name: Cisco
SCCAN? n Secure SIP? n
Grp FRL NPA Pfx Hop Toll No. Inserted DCS/ IXC
No Mrk Lmt List Del Digits QSIG
Dgts Intw
1: 4 0 - - - - - n user
2: - - - - - - - n user
3: - - - - - - - n user
4: - - - - - - - n user
5: - - - - - - - n user
6: - - - - - - - n user

BCC VALUE TSC CA-TSC ITC BCIE Service/Feature PARM No. Numbering LAR
0 1 2 M 4 W Request Dgts Format Subaddress
1: Y Y Y Y Y Y n unre - lev0-pvt none
2: Y Y Y Y Y n n rest - - none
3: Y Y Y Y Y n n rest - - none
4: Y Y Y Y Y n n rest - - none
5: Y Y Y Y Y n n rest - - none
6: Y Y Y Y Y n n rest - - none

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

**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.

change dialplan analysis Page 1 of 12

DIAL PLAN ANALYSIS TABLE  
Location: all Percent Full: 3

Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type
0	1	attd	8	1	fac			
2	5	ext	9	1	fac			
21	4	ext	*	3	fac			
2302	4	ext	*	4	dac			
26	4	ext	#	3	fac			
2624	4	udp						
2633	4	udp						
28	4	ext						
30000	5	ext						
31	4	ext						
33	4	ext						
34	4	fac						
4	4	ext						
5	4	udp						
6	4	udp						

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

**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.

The screenshot shows a terminal window titled '10.70.2.14 - PuTTY' displaying the command 'change aar analysis 2'. The output is an 'AAR DIGIT ANALYSIS TABLE' for 'Location: all' and 'Percent Full: 3'. The table lists various dialed strings and their corresponding route patterns and call types. The following table represents the data shown in the screenshot:

Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Reqd
14242	10	10	4	aar		n
22	5	5	20	aar		n
2300	5	5	20	aar		n
2302	4	4	4	aar		n
240	5	5	20	aar		n
26	4	4	4	aar		n
30000	5	5	1	aar		n
330	4	4	4	aar		n
4	7	7	254	aar		n
45	4	4	4	aar		n
5	4	4	4	aar		n
6	4	4	4	aar		n
6	7	7	254	aar		n
7	7	7	254	aar		n
7193	10	10	4	aar		n

At the bottom of the terminal window, the following function key definitions are visible: F1=Cancel F2=Refresh F3=Submit F4=Clr F1d F5=Help F6=Update F7=Nxt Pg F8=Prv Pg





Fill in the indicated fields as shown below and use default values for remaining fields.

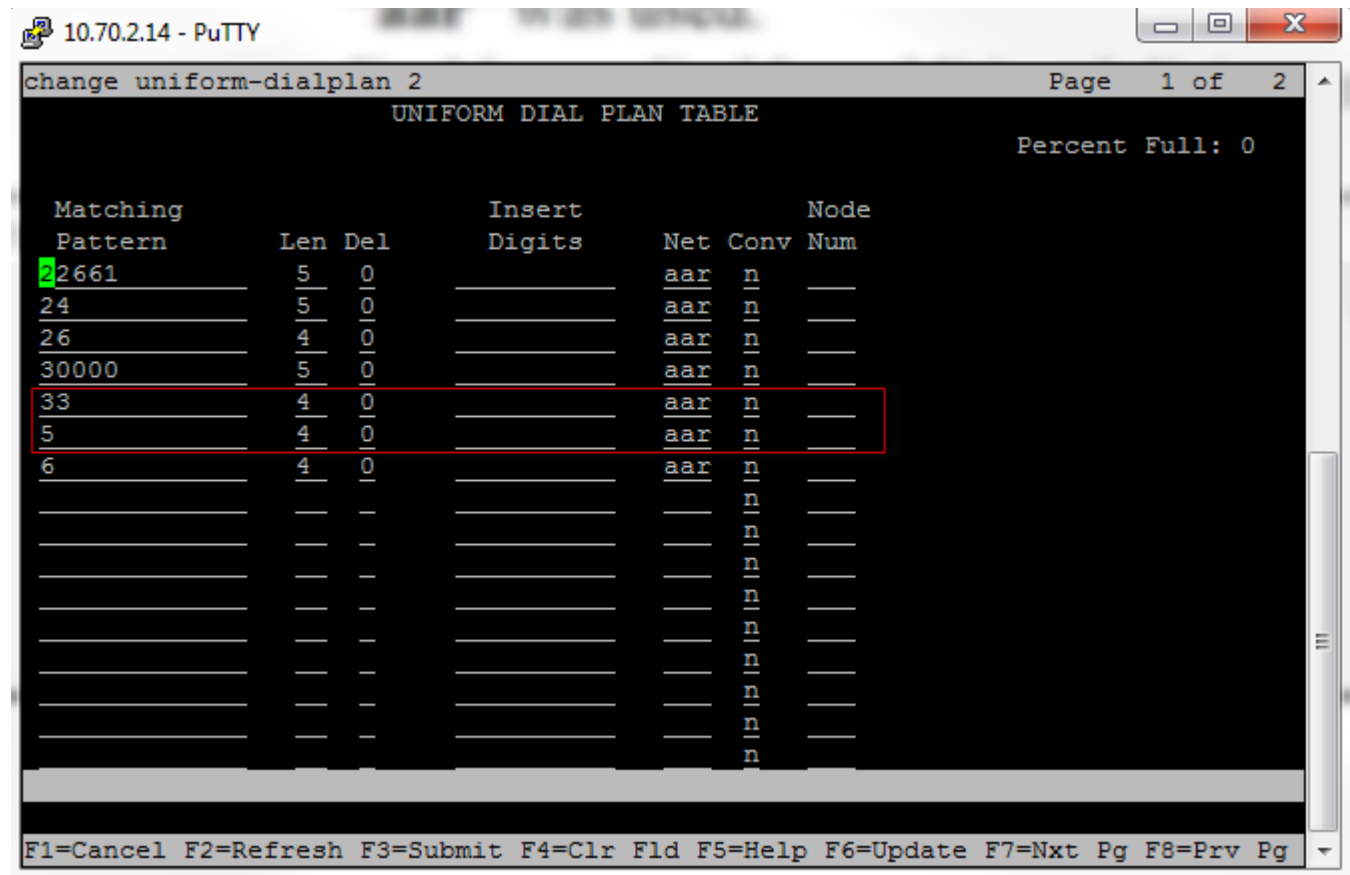
**Configure Uniform dialplan:**

**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

**AVAYA** Avaya Aura® System Manager 6.1 [Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

[Routing](#) \* [Home](#)

- Users**
  - Administrators**  
Manage Administrative Users
  - Groups & Roles**  
Manage groups, roles and assign roles to users
  - Synchronize and Import**  
Synchronize users with the enterprise directory, import users from file
  - User Management**  
Manage users, shared user resources and provision users
- Elements**
  - Application Management**  
Manage applications and application certificates
  - Communication Manager**  
Manage Communication Manager objects
  - Conferencing**  
Conferencing
  - Inventory**  
Manage, discover, and navigate to elements, update element software
  - Messaging**  
Manage Messaging System objects
  - Presence**  
Presence
  - Routing**   
Network Routing Policy
  - Session Manager**  
Session Manager Element Manager
  - SIP AS 8.1**  
SIP AS 8.1
- Services**
  - Backup and Restore**  
Backup and restore System Manager database
  - Configurations**  
Manage system wide configurations
  - Events**  
Manage alarms, view and harvest logs
  - Licenses**  
View and configure licenses
  - Replication**  
Track data replication nodes, repair replication nodes
  - Scheduler**  
Schedule, track, cancel, update and delete jobs
  - Security**  
Manage Security Certificates
  - Templates**  
Manage Templates for Communication Manager and Messaging System objects

Add Domains

Under page Domain Management:

Name: lab.tekvizion.com

Type:sip

**AVAYA** Avaya Aura® System Manager 6.1 [Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

[Routing](#) ✕ [Home](#)

Home / Elements / Routing / Domains - Domain Management [Help ?](#)

**Domain Management** [Commit](#) [Cancel](#)

1 Item | Refresh Filter: Enable

Name	Type	Default	Notes
* lab.tekvizion.com	sip	<input type="checkbox"/>	

**\* Input Required** [Commit](#) [Cancel](#)

Add Location  
Name: Dallas



Home / Elements / Routing / Locations - Location Details [Help ?](#)

**Location Details** [Commit](#) [Cancel](#)

Call Admission Control has been set to ignore SDP. All calls will be counted using the Default Audio Bandwidth.  
See Session Manager -> Session Manager Administration -> Global Setting

**General**

\* **Name:**   
**Notes:**

**Overall Managed Bandwidth**

**Managed Bandwidth Units:**    
**Total Bandwidth:**

**Per-Call Bandwidth Parameters**

\* **Default Audio Bandwidth:**

**Location Pattern**

0 Items | [Refresh](#) Filter: [Enable](#)

<input type="checkbox"/>	IP Address Pattern	Notes
--------------------------	--------------------	-------

\* **Input Required** [Commit](#) [Cancel](#)

## 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

The screenshot shows the Avaya Aura System Manager 6.1 web interface. The browser address bar is `https://10.70.24/SMGR/`. The page title is "Avaya Aura® System Manager 6.1". The navigation menu on the left includes "Routing", "Domains", "Locations", "Adaptations", "SIP Entities", "Entity Links", "Time Ranges", "Routing Policies", "Dial Patterns", "Regular Expressions", and "Defaults". The "Routing" menu is expanded, and the "Adaptations" sub-menu is selected. The main content area is titled "Adaptation Details" and contains a "General" section with the following fields:

- \* Adaptation name:** Cisco\_CUCM10
- Module name:** CiscoAdapter
- Module parameter:** fromto=true odstd=10.80.10.3 iosrcd=lab.tekvizion.com
- Egress URI Parameters:** (empty)
- Notes:** to Cisco CUCM10 test


Below the "General" section, there are two sections for "Digit Conversion for Incoming Calls to SM" and "Digit Conversion for Outgoing Calls from SM". Each section has "Add" and "Remove" buttons and a table with columns: Matching Pattern, Min, Max, Phone Context, Delete Digits, Insert Digits, Address to modify, and Notes. Both tables currently show "0 Items" and "Filter: Enable".

## Adaptation for Avaya Aura CM

Adaptation name:Avaya\_CM

Module name: DigitConversionAdapter

Module Parameter: fromto=true

Avaya Aura® System Manager 6.1Help | About | Change Password | Log off admin

Routing Home

- Routing
- Domains
- Locations
- Adaptations
- SIP Entities
- Entity Links
- Time Ranges
- Routing Policies
- Dial Patterns
- Regular Expressions
- Defaults

Home / Elements / Routing / Adaptations - Adaptation Details

Adaptation Details

Help ?

Commit Cancel

General

\* Adaptation name:

Module name:

Module parameter:

Egress URI Parameters:

Notes:

Digit Conversion for Incoming Calls to SM

Add Remove

0 Items Refresh Filter: Enable

<input type="checkbox"/>	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Notes
--------------------------	------------------	-----	-----	---------------	---------------	---------------	-------------------	-------

Digit Conversion for Outgoing Calls from SM

Add Remove

0 Items Refresh Filter: Enable

<input type="checkbox"/>	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Notes
--------------------------	------------------	-----	-----	---------------	---------------	---------------	-------------------	-------

\* Input Required

Commit Cancel

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

The screenshot displays the Avaya Aura System Manager 6.1 web interface. The top navigation bar includes the Avaya logo, the product name 'Avaya Aura® System Manager 6.1', and user options like 'Help | About | Change Password | Log off admin'. A breadcrumb trail shows 'Home / Elements / Routing / SIP Entities - SIP Entity Details'. The left sidebar lists various configuration categories, with 'SIP Entities' selected. The main content area is titled 'SIP Entity Details' and contains several sections: 'General' with fields for Name (teksm), FQDN or IP Address (10.70.2.6), Type (Session Manager), Location (Dallas), Time Zone (America/Chicago), and SIP Link Monitoring (Use Session Manager Configuration); 'Entity Links' with a warning that links can only be modified after the entity is committed; and a 'Port' table with two entries for port 5060 using TCP and UDP protocols on the domain lab.tekvizion.com. The interface includes 'Commit' and 'Cancel' buttons and a 'Filter: Enable' option for the table.

**AVAYA** Avaya Aura® System Manager 6.1 Help | About | Change Password | Log off admin

Routing x Home

Home / Elements / Routing / SIP Entities - SIP Entity Details Help ?

SIP Entity Details Commit Cancel

**General**

\* Name:

\* FQDN or IP Address:

Type:

Notes:

Location:

Outbound Proxy:

Time Zone:

Credential name:

**SIP Link Monitoring**

SIP Link Monitoring:

**Entity Links**

Entity Links can be modified after SIP Entity is committed.

**Port**

2 Items  Filter: Enable

<input type="checkbox"/>	Port	Protocol	Default Domain	Notes
<input type="checkbox"/>	<input type="text" value="5060"/>	<input type="text" value="TCP"/>	<input type="text" value="lab.tekvizion.com"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text" value="5060"/>	<input type="text" value="UDP"/>	<input type="text" value="lab.tekvizion.com"/>	<input type="text"/>



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



Routing x Home

Home / Elements / Routing / SIP Entities - SIP Entity Details

SIP Entity Details

Commit Cancel Help ?

**General**

\* Name: Cisco\_CUCM10

\* FQDN or IP Address: 10.80.10.3

Type: Other

Notes:

Adaptation: Cisco\_CUCM10

Location: Dallas

Time Zone: America/Chicago

Override Port & Transport with DNS SRV:

\* SIP Timer B/F (in seconds): 4

Credential name:

Call Detail Recording: none

**SIP Link Monitoring**

SIP Link Monitoring: Use Session Manager Configuration

**Entity Links**

Add Remove

0 Items | Refresh Filter: Enable

	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted
--	--------------	----------	------	--------------	------	---------

## 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

The screenshot shows the Avaya Aura System Manager 6.1 interface. The top navigation bar includes the Avaya logo, the product name "Avaya Aura® System Manager 6.1", and links for "Help | About | Change Password | Log off admin". The main content area is titled "SIP Entity Details" and is divided into several sections:

- General:** Contains fields for Name (tekcm), FQDN or IP Address (10.70.2.14), Type (CM), and Notes.
- Adaptation:** Contains dropdown menus for Adaptation (Avaya\_CM), Location (Dallas), and Time Zone (America/Chicago).
- Override Port & Transport with DNS SRV:** A checkbox that is currently unchecked.
- SIP Timer B/F (in seconds):** A text input field containing the value 4.
- Credential name:** An empty text input field.
- Call Detail Recording:** A dropdown menu set to "none".
- SIP Link Monitoring:** A dropdown menu set to "Use Session Manager Configuration".
- Entity Links:** Includes "Add" and "Remove" buttons and a table with 0 items.

The table under "Entity Links" has the following structure:

	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted
0 Items   Refresh						

At the bottom of the page, there is a "\* Input Required" message and "Commit" and "Cancel" buttons.

## 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



Routing \* Home

Home / Elements / Routing / Entity Links - Entity Links

Entity Links [Help ?](#)

1 Item Refresh Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
* ASM to CUCM10	* teksm	TCP	* 5060	* Cisco_CUCM10	* 5060	<input checked="" type="checkbox"/>	to Cisco CUCM10

\* Input Required

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



Routing \* Home

Home / Elements / Routing / Entity Links - Entity Links

Entity Links [Help ?](#)

1 Item Refresh Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
* teksm_tekcm_5060_TCP	* teksm	TCP	* 5060	* tekcm	* 5060	<input checked="" type="checkbox"/>	

\* Input Required

## 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



The screenshot shows the 'Routing Policy Details' page in the Avaya Aura System Manager 6.1. The left sidebar contains a navigation menu with 'Routing Policies' selected. The main content area is titled 'Routing Policy Details' and includes a 'General' section with the following fields: '\* Name: to Cisco CUCM10', 'Disabled: ', and 'Notes: to Cisco CUCM10'. Below this is the 'SIP Entity as Destination' section with a 'Select' button. At the bottom, a table lists the destination SIP entity:

Name	FQDN or IP Address	Type	Notes
Cisco_CUCM10	10.80.10.3	Other	

Routing Policy for calls to go to Avaya Aura Communication Manager

Name: To\_tekcm

Select SIP Entity "tekcm" for SIP Entity as Destination



The screenshot shows the 'Routing Policy Details' page in the Avaya Aura System Manager 6.1. The left sidebar contains a navigation menu with 'Routing Policies' selected. The main content area is titled 'Routing Policy Details' and includes a 'General' section with the following fields: '\* Name: to\_tekcm', 'Disabled: ', and 'Notes:'. Below this is the 'SIP Entity as Destination' section with a 'Select' button. At the bottom, a table lists the destination SIP entity:

Name	FQDN or IP Address	Type	Notes
tekcm	10.70.2.14	CM	

## 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 6.1Help | About | Change Password | Log off admin

Routing Home

- Routing
- Domains
- Locations
- Adaptations
- SIP Entities
- Entity Links
- Time Ranges
- Routing Policies
- Dial Patterns
- Regular Expressions
- Defaults

Home / Elements / Routing / Dial Patterns - Dial Pattern Details

### Dial Pattern Details

Commit Cancel

#### General

\* Pattern:

\* Min:

\* Max:

Emergency Call:

SIP Domain:

Notes:

#### Originating Locations and Routing Policies

Add Remove

1 Item Refresh Filter: Enable

<input type="checkbox"/>	Originating Location Name 1 ▲	Originating Location Notes	Routing Policy Name	Rank 2 ▲	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Dallas	enterprise	to Cisco CUCM10	0	<input type="checkbox"/>	Cisco_CUCM10	to Cisco CUCM10

Select : All, None

#### Denied Originating Locations

Add Remove

0 Items Refresh Filter: Enable

<input type="checkbox"/>	Originating Location	Notes
--------------------------	----------------------	-------

\* Input Required

Commit Cancel

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

The screenshot shows the Avaya Aura System Manager 6.1 web interface. The browser address bar indicates the URL is https://10.70.24/SMGR/. The page title is "Avaya Aura® System Manager 6.1". The navigation menu on the left includes "Routing", "Domains", "Locations", "Adaptations", "SIP Entities", "Entity Links", "Time Ranges", "Routing Policies", "Dial Patterns", "Regular Expressions", and "Defaults". The "Dial Patterns" section is active, showing "Dial Pattern Details". The "General" tab is selected, and a red box highlights the following fields: "Pattern: 310", "Min: 4", "Max: 4", "Emergency Call: [checkbox]", "SIP Domain: lab.tekvizion.com", and "Notes: to Avaya CM". Below this, the "Originating Locations and Routing Policies" section shows a table with one item: "Dallas" (Originating Location Name), "enterprise" (Originating Location Notes), "to\_tekcm" (Routing Policy Name), "0" (Rank), and "tekcm" (Routing Policy Destination). The "Denied Originating Locations" section is empty. The page includes "Commit" and "Cancel" buttons at the bottom right.

Pattern: 330

Min: 4

Max: 4

SIP Domain: lab.tekvizion.com

Original Location Name: Dallas

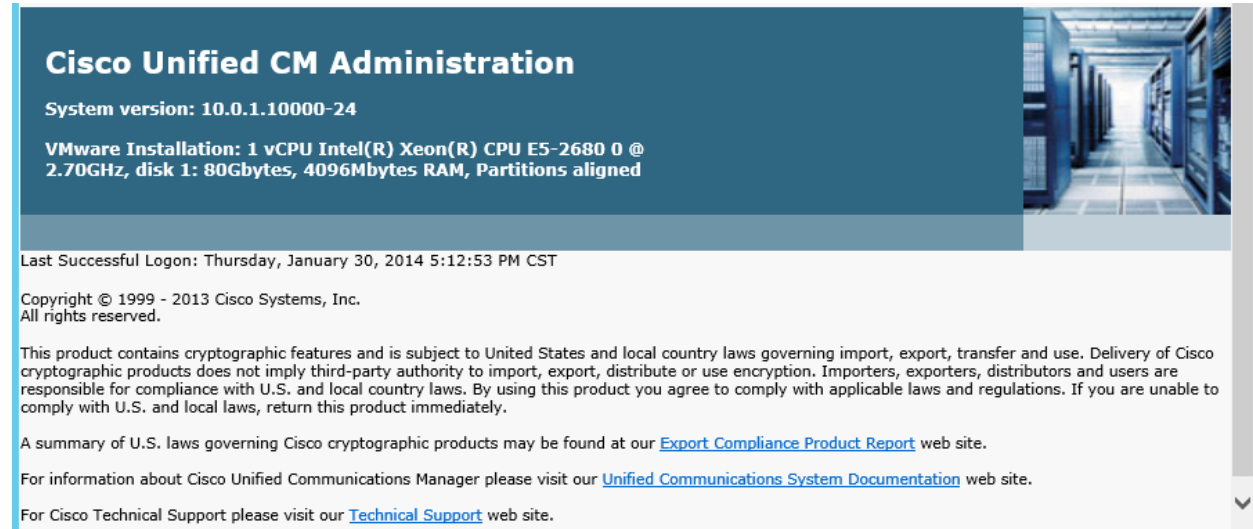
Routing Policy Name: to\_tekcm

The screenshot shows the Avaya Aura System Manager 6.1 interface. The top navigation bar includes the Avaya logo, the title "Avaya Aura® System Manager 6.1", and links for "Help | About | Change Password | Log off admin". The breadcrumb trail is "Home / Elements / Routing / Dial Patterns - Dial Pattern Details". The left sidebar contains a menu with "Routing" selected, and sub-items: Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns (highlighted), Regular Expressions, and Defaults. The main content area is titled "Dial Pattern Details" and has a "General" section. A red box highlights the following fields: "\* Pattern: 330", "\* Min: 4", "\* Max: 4", "Emergency Call: ", "SIP Domain: lab.tekvizion.com", and "Notes: ". Below this is the "Originating Locations and Routing Policies" section, which includes "Add" and "Remove" buttons, "1 Item | Refresh", and a table with columns: "Originating Location Name", "Originating Location Notes", "Routing Policy Name", "Rank", "Routing Policy Disabled", "Routing Policy Destination", and "Routing Policy Notes". The table contains one row: Dallas, enterprise, to\_tekcm, 0, , tekcm. Below the table is "Denied Originating Locations" section with "Add" and "Remove" buttons, "0 Items | Refresh", and a table with columns: "Originating Location" and "Notes". At the bottom, there is a "\* Input Required" message and "Commit" and "Cancel" buttons.



# Configuring the Cisco Unified Communications Manager

Cisco Unified Communications Manager Software Version



**Cisco Unified CM Administration**

System version: 10.0.1.10000-24

VMware Installation: 1 vCPU Intel(R) Xeon(R) CPU E5-2680 0 @ 2.70GHz, disk 1: 80Gbytes, 4096Mbytes RAM, Partitions aligned

Last Successful Logon: Thursday, January 30, 2014 5:12:53 PM CST

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This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco cryptographic products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with U.S. and local country laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at our [Export Compliance Product Report](#) web site.

For information about Cisco Unified Communications Manager please visit our [Unified Communications System Documentation](#) web site.

For Cisco Technical Support please visit our [Technical Support](#) web site.

## 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 CM Administration**  
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration  
administrator | Search Documentation

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾  
Help ▾

**SIP Trunk Security Profile Configuration** Related Links: Back To Find/List ▾ Go

Save Delete Copy Reset Apply Config Add New

**Status**  
Status: Ready

**SIP Trunk Security Profile Information**

Name*	Non Secure SIP Trunk Profile
Description	Non Secure SIP Trunk Profile authenticated by null St
Device Security Mode	Non Secure ▾
Incoming Transport Type*	TCP+UDP ▾
Outgoing Transport Type	TCP ▾
<input type="checkbox"/> Enable Digest Authentication	
Nonce Validity Time (mins)*	600
X.509 Subject Name	
Incoming Port*	5060
<input type="checkbox"/> Enable Application level authorization	
<input type="checkbox"/> Accept presence subscription	
<input checked="" type="checkbox"/> Accept out-of-dialog refer**	
<input checked="" type="checkbox"/> Accept unsolicited notification	
<input checked="" type="checkbox"/> Accept replaces header	
<input type="checkbox"/> Transmit security status	
<input type="checkbox"/> Allow charging header	
SIP V.150 Outbound SDP Offer Filtering*	Use Default Filter ▾

Save Delete Copy Reset Apply Config Add New



## 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.

**SIP Trunk Security Profile Information**

Name\*

Description

Device Security Mode

Incoming Transport Type\*

Outgoing Transport Type

Enable Digest Authentication

Nonce Validity Time (mins)\*

X.509 Subject Name

Incoming Port\*

Enable Application level authorization

- Accept presence subscription
- Accept out-of-dialog refer\*\*
- Accept unsolicited notification
- Accept replaces header
- Transmit security status

Allow charging header

SIP V.150 Outbound SDP Offer Filtering\*

## 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

Related Links: Back To Find/List ▾ Go

Save Delete Copy Reset Apply Config Add New

**Status**

Status: Ready

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

**SIP Profile Information**

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

Redirect by Application

Disable Early Media on 180

Outgoing T.38 INVITE include audio mline

Use Fully Qualified Domain Name in SIP Requests

Assured Services SIP conformance

**SDP Information**

SDP Session-level Bandwidth Modifier for Early Offer and Re-invites\* TIAS and AS ▾

SDP Transparency Profile Pass all unknown SDP attributes ▾

Accept Audio Codec Preferences in Received Offer\* Off ▾

Require SDP Inactive Exchange for Mid-Call Media Change

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

These values are default.



### SIP Profile Configuration

Related Links: [Back To Find/List](#) ▾ [Go](#)

Save Delete Copy Reset Apply Config Add New

#### Parameters used in Phone

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

### 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.

**SIP Profile Configuration** Related Links: [Back To Find/List](#)

Save  Copy

Conference Join Enabled  
 **RFC 2543 Hold**  
 Semi Attended Transfer  
 Enable VAD  
 Stutter Message Waiting  
 MLPP User Authorization

**Normalization Script**

Normalization Script   
 Enable Trace

	Parameter Name	Parameter Value	
1	<input type="text"/>	<input type="text"/>	<input type="button" value="+"/>

**Incoming Requests FROM URI Settings**

Caller ID DN   
Caller Name

**Trunk Specific Configuration**

Reroute Incoming Request to new Trunk based on\*   
RSVP Over SIP\*   
Resource Priority Namespace List   
 Fall back to local RSVP  
**SIP Rel1XX Options\***   
Video Call Traffic Class\*   
Calling Line Identification Presentation\*   
Session Refresh Method\*   
 Enable ANAT  
 Deliver Conference Bridge Identifier  
 **Early Offer support for voice and video calls (insert MTP if needed)**  
 Allow Passthrough of Configured Line Device Caller Information  
 Reject Anonymous Incoming Calls  
 Reject Anonymous Outgoing Calls

### 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.

Send ILS Learned Destination Route String

**SIP OPTIONS Ping**

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

Ping Interval for In-service and Partially In-service Trunks (seconds)*	60
Ping Interval for Out-of-service Trunks (seconds)*	120
Ping Retry Timer (milliseconds)*	500
Ping Retry Count*	6

**SDP Information**


Send send-receive SDP in mid-call INVITE

Allow Presentation Sharing using BFCP

Allow iX Application Media

Allow multiple codecs in answer SDP

Save Delete Copy Reset Apply Config Add New

 \*- indicates required item.

## 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.

The screenshot displays the Cisco Unified CM Administration interface for configuring a SIP Trunk. The page title is "Trunk Configuration" and the user is logged in as "administrator". The configuration is for a SIP Trunk named "Trunk\_to\_Avaya\_SM".

Device Information	
Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	Trunk_to_Avaya_SM
Description	SIP Trunk to Avaya
Device Pool*	G711 Pool
Common Device Configuration	< None >
Call Classification*	OnNet
Media Resource Group List	MRGL_G711
Location*	Hub_None
AAR Group	< None >
Tunneled Protocol*	None
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes
Packet Capture Mode*	None
Packet Capture Duration	0
<input type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Path Replacement Support	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Transmit UTF-8 Names in QSIG APDU	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.	
Consider Traffic on This Trunk Secure*	When using both sRTP and TLS
Route Class Signaling Enabled*	Default
Use Trusted Relay Point*	Default
<input checked="" type="checkbox"/> PSTN Access	
<input type="checkbox"/> Run On All Active Unified CM Nodes	

## **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 CM Administration**  
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration administrator | Search Documentation | About | Log

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

**Trunk Configuration** Related Links: Back To Find/List ▾ Go

Save Delete Reset Add New

**Intercompany Media Engine (IME)**

E.164 Transformation Profile < None > ▾

**MLPP and Confidential Access Level Information**

MLPP Domain < None > ▾  
Confidential Access Mode < None > ▾  
Confidential Access Level < None > ▾

**Call Routing Information**

Remote-Party-Id  
 Asserted-Identity  
Asserted-Type\* Default ▾  
SIP Privacy\* Default ▾

**Inbound Calls**

Significant Digits\* All ▾  
Connected Line ID Presentation\* Allowed ▾  
Connected Name Presentation\* Allowed ▾  
Calling Search Space < None > ▾  
AAR Calling Search Space < None > ▾  
Prefix DN   
 Redirecting Diversion Header Delivery - Inbound

**Incoming Calling Party Settings**

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

Clear Prefix Settings Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None > ▾	<input checked="" type="checkbox"/>

## 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.

The screenshot displays the Cisco Unified CM Administration web interface. At the top, the navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and the user role "administrator". Below this is a breadcrumb trail: System > Call Routing > Media Resources > Advanced Features > Device > Application > User Management > Bulk Administration > Help. The main heading is "Trunk Configuration" with a "Related Links" section containing "Back To Find/List".

Below the heading are action buttons: Save, Delete, Reset, and Add New. The configuration is organized into several sections:

- Incoming Called Party Settings:** Includes a text box for "Prefix" (set to "Default"), "Strip Digits" (set to "0"), "Calling Search Space" (set to "< None >"), and a checked "Use Device Pool CSS" checkbox. Buttons for "Clear Prefix Settings" and "Default Prefix Settings" are also present.
- Connected Party Settings:** Includes a "Connected Party Transformation CSS" dropdown (set to "< None >") and a checked "Use Device Pool Connected Party Transformation CSS" checkbox.
- Outbound Calls:** This section contains several settings, with a red box highlighting the following:
  - Called Party Transformation CSS: < None >
  - Use Device Pool Called Party Transformation CSS: checked
  - Calling Party Transformation CSS: < None >
  - Use Device Pool Calling Party Transformation CSS: checked
  - Calling Party Selection\*: Originator
  - Calling Line ID Presentation\*: Allowed
  - Calling Name Presentation\*: Allowed
  - Calling and Connected Party Info Format\*: Deliver URI and DN in connected party, if available
  - Redirecting Diversion Header Delivery - Outbound: checked
  - Redirecting Party Transformation CSS: < None >
  - Use Device Pool Redirecting Party Transformation CSS: checked
- Caller Information:** Includes text boxes for "Caller ID DN" and "Caller Name", and a checkbox for "Maintain Original Caller ID DN and Caller Name in Identity Headers" (unchecked).

## **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 CM Administration**  
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go  
administrator | Search Documentation | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾  
Help ▾

**Trunk Configuration** Related Links: Back To Find/List Go

Save Delete Reset Add New

**SIP Information**

Destination Address is an SRV

1*	Destination Address	Destination Address IPv6	Destination Port
	10.70.2.6		5060

MTP Preferred Originating Codec\* 711ulaw ▾  
 BLF Presence Group\* Standard Presence group ▾  
 SIP Trunk Security Profile\* Non Secure SIP Trunk Profile ▾  
 Rerouting Calling Search Space < None > ▾  
 Out-Of-Dialog Refer Calling Search Space < None > ▾  
 SUBSCRIBE Calling Search Space < None > ▾  
 SIP Profile\* Early Offer SIP Profile ▾ [View Details](#)  
 DTMF Signaling Method\* RFC 2833 ▾

**Normalization Script**

Normalization Script Remove-CallInfo-Header ▾

Enable Trace

1	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

Save Delete Reset Add New

## 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





### SIP Normalization Script Configuration

Related Links: [Back To Find/List](#) ▾ [Go](#)

#### SIP Normalization Script Info

NOTE: Scripts may only be updated on the publisher. Please connect to the publisher and try updating the script there.

Name *	<input type="text" value="Remove-CallInfo-Header"/>
Description	<input type="text" value="Remove-CallInfo-Header"/>
Content *	<pre>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)</pre>
Script Execution Error Recovery Action *	<input type="text" value="Message Rollback Only"/>
System Resource Error Recovery Action *	<input type="text" value="Disable Script"/>
Memory Threshold *	<input type="text" value="50"/> kilobytes
Lua Instruction Threshold *	<input type="text" value="1000"/> instructions

**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:**

```
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.

The screenshot displays the Cisco Unified CM Administration interface for configuring a SIP Trunk. The page title is "Trunk Configuration" and the user is logged in as "administrator". The "Device Information" section is highlighted with a red box, showing the following configuration:

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	To_Unity_Connection
Description	SIP Trunk for Cisco Unity Connection
Device Pool*	Default
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	< None >
Location*	Hub_None
AAR Group	< None >
Tunneled Protocol*	None
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes
Packet Capture Mode*	None
Packet Capture Duration	0

Below the table, several checkboxes are visible:

- Media Termination Point Required
- Retry Video Call as Audio
- Path Replacement Support
- Transmit UTF-8 for Calling Party Name
- Transmit UTF-8 Names in QSIG APDU
- Unattended Port
- SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.

The "Consider Traffic on This Trunk Secure\*" dropdown is set to "When using both sRTP and TLS".

Other fields include:

- Route Class Signaling Enabled\*: Default
- Use Trusted Relay Point\*: Default
- PSTN Access
- Run On All Active Unified CM Nodes



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

Check Redirecting Diversion Header Delivery - Inbound

All other values are default.

The screenshot shows the Cisco Unified CM Administration interface for Trunk Configuration. The page includes a navigation bar with 'Cisco Unified CM Administration' and 'administrator' roles. The main content area is titled 'Trunk Configuration' and contains several sections:

- Intercompany Media Engine (IME):** E.164 Transformation Profile is set to '< None >'.
- MLPP and Confidential Access Level Information:** MLPP Domain, Confidential Access Mode, and Confidential Access Level are all set to '< None >'.
- Call Routing Information:** Remote-Party-Id and Asserted-Identity are checked. Asserted-Type\* and SIP Privacy\* are set to 'Default'.
- Inbound Calls:** Significant Digits\* is 'All', Connected Line ID Presentation\*, Connected Name Presentation\*, Calling Search Space, and AAR Calling Search Space are all 'Default'. Prefix DN is empty.
- Redirecting Diversion Header Delivery - Inbound:** This checkbox is checked and highlighted with a red box.
- Incoming Calling Party Settings:** A text box explains that the prefix 'Default' indicates call processing will use the prefix at the next level setting. Below this are buttons for 'Clear Prefix Settings' and 'Default Prefix Settings'. A table shows settings for 'Incoming Number':

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

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

Check Redirecting Diversion Header Delivery - Outbound

All other values are default.

The screenshot displays the Cisco Unified CM Administration interface for configuring a SIP Trunk. The page title is "Trunk Configuration" and it includes a navigation menu at the top with options like "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", and "Help". The user is logged in as "administrator".

The configuration is divided into several sections:

- Incoming Called Party Settings:** This section includes a text box explaining that the prefix setting affects call processing. It has two buttons: "Clear Prefix Settings" and "Default Prefix Settings". Below is a table with columns for "Number Type", "Prefix", "Strip Digits", "Calling Search Space", and "Use Device Pool CSS". The "Incoming Number" row shows "Default" for Prefix, "0" for Strip Digits, "< None >" for Calling Search Space, and a checked "Use Device Pool CSS" checkbox.
- Connected Party Settings:** This section has a dropdown for "Connected Party Transformation CSS" set to "< None >" and a checked checkbox for "Use Device Pool Connected Party Transformation CSS".
- Outbound Calls:** This section contains several dropdown menus and checkboxes:
  - "Called Party Transformation CSS" set to "< None >"
  - Checked checkbox for "Use Device Pool Called Party Transformation CSS"
  - "Calling Party Transformation CSS" set to "< None >"
  - Checked checkbox for "Use Device Pool Calling Party Transformation CSS"
  - "Calling Party Selection\*" set to "Originator"
  - "Calling Line ID Presentation\*" set to "Default"
  - "Calling Name Presentation\*" set to "Default"
  - "Calling and Connected Party Info Format\*" set to "Deliver DN only in connected party"
  - Checked checkbox for "Redirecting Diversion Header Delivery - Outbound" (highlighted with a red box)
  - "Redirecting Party Transformation CSS" set to "< None >"
  - Checked checkbox for "Use Device Pool Redirecting Party Transformation CSS"
- Caller Information:** This section has input fields for "Caller ID DN" and "Caller Name", and an unchecked checkbox for "Maintain Original Caller ID DN and Caller Name in Identity Headers".

## 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.

The screenshot displays the Cisco Unified CM Administration interface for configuring a SIP Trunk. The page title is "Trunk Configuration" and it includes navigation menus for System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The user is logged in as "administrator".

**SIP Information**

**Destination**

Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port	Status
1*	10.80.10.5		5060	N/A

MTP Preferred Originating Codec\* 711ulaw  
BLF Presence Group\* Standard Presence group  
SIP Trunk Security Profile\* Non Secure SIP Trunk to VM Profile  
Rerouting Calling Search Space < None >  
Out-Of-Dialog Refer Calling Search Space < None >  
SUBSCRIBE Calling Search Space < None >  
SIP Profile\* Standard SIP Profile [View Details](#)  
DTMF Signaling Method\* No Preference

**Normalization Script**

Normalization Script < None >  
 Enable Trace

	Parameter Name	Parameter Value
1		

**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.

The screenshot displays the Cisco Unity Connection Administration web interface. The left sidebar shows a navigation tree with categories like Users, Class of Service, Templates, Contacts, Distribution Lists, Call Management, Message Storage, Networking, and Unified Messaging. The main content area is titled 'Name' and contains the following configuration fields:

- Alias\*: 5017
- First Name: 5017
- Last Name: cisco
- Display Name: 5017 cisco
- SMTP Address: 5017 @lab.tekvizion.com
- Initials: (empty)
- Title: (empty)
- Employee ID: (empty)

Below the name fields is the 'LDAP Integration Status' section with two radio buttons: 'Integrate with LDAP Directory' (unselected) and 'Do Not Integrate with LDAP Directory' (selected).

The 'Phone' section includes the following fields:

- Extension\*: 5017
- Cross-Server Transfer Extension: (empty)
- Outgoing Fax Number: (empty)
- Outgoing Fax Server: --- Not Selected ---
- Partition: clus20unity Partition
- Search Scope: clus20unity Search Space
- Phone System: Cluster 20
- Class of Service: Voice Mail User COS
- Active Schedule: Weekdays

At the bottom of the phone section, there are several checkboxes: 'Set for Self-enrollment at Next Sign-In' (unchecked), 'List in Directory' (checked), 'Send Non-Delivery Receipts on Failed Message Delivery' (checked), 'Skip PIN When Calling From a Known Extension' (checked), and 'Use Short Calendar Caching Poll Interval' (unchecked). A 'Caution!' note is present below the 'Skip PIN' checkbox. A 'Recorded Name' field with a 'Play/Record' button is at the very bottom.



## Cisco Unity Connection User 5017 Configuration (Continued)

All values are default.

**Cisco Unity Connection Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unity Connection Administration

administrator | Search Documentation | About | Sign Out

**Cisco Unity Connection**

- Users
  - Users
  - Import Users
  - Synch Users
- Class of Service
  - Class of Service
  - Class of Service Membership
- Templates
  - User Templates
  - Call Handler Templates
  - Contact Templates
  - Notification Templates
- Contacts
  - Contacts
- Distribution Lists
  - System Distribution Lists
- Call Management
  - System Call Handlers
  - Directory Handlers
  - Interview Handlers
  - Custom Recordings

**Location**

Address

Building

City

State

Postal Code

Country

Use System Default Time Zone

Time Zone

Language  Use System Default Language

English(United States)

Department

Manager

Billing ID

Corporate Email Address

Generate SMTP Proxy Address From Corporate Email Address

Corporate Phone Number

## Cisco Unified Communications Manager Service Parameter

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

The screenshot shows the Cisco Unified CM Administration interface. The page title is "Service Parameter Configuration". The navigation menu includes "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", and "Bulk Administration". The user is logged in as "administrator".

The "Clusterwide Parameters (External QoS)" section contains the following parameters:


External QoS Enabled *	False	False
------------------------	-------	-------

The "Clusterwide Parameters (Service)" section contains the following parameters:

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

**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





**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions


Navigation Cisco Unified CM Administration Go

administrator | [Search Documentation](#) | [About](#) | [Logout](#)



System ▾ | Call Routing ▾ | Media Resources ▾ | Advanced Features ▾ | Device ▾ | Application ▾ | User Management ▾ | Bulk Administration ▾ | Help ▾



**Find and List Media Resource Groups**

 Add New  Select All  Clear All  Delete Selected

**Status**  
 2 records found

**Media Resource Group (1 - 2 of 2)** Rows per Page 50

Find Media Resource Group where Name ▾ begins with ▾  Find Clear Filter  

<input type="checkbox"/>	Name ^	Description	Multi-cast	Copy
<input type="checkbox"/>	<a href="#">MRG_MTP</a>	MRG with MTP	false	
<input type="checkbox"/>	<a href="#">MRG_noMTP</a>	MRG without MTP	false	

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.

The screenshot displays the Cisco Unified CM Administration interface for configuring a Media Resource Group. The page title is "Media Resource Group Configuration" and the user is logged in as "administrator". The navigation menu includes System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, and Bulk Administration. The configuration page includes a "Status" section showing "Ready", a "Media Resource Group Status" section showing "MRG\_MTP (used by 13 devices)", and a "Media Resource Group Information" section with fields for "Name\*" (MRG\_MTP) and "Description" (MRG with MTP). Below this is a "Devices for this Group" section with an "Available Media Resources\*\*" list (empty) and a "Selected Media Resources\*" list containing ANN\_2 (ANN), ANN\_3 (ANN), ANN\_4 (ANN), CFB\_2 (CFB), and CFB\_3 (CFB). The "Media Resource Group Information" and "Selected Media Resources\*" sections are highlighted with red boxes.

**Media Resource Group Configuration** Related Links: [Back To Find/List](#) Go

Save Delete Copy Add New

**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\*  
ANN\_2 (ANN)  
ANN\_3 (ANN)  
ANN\_4 (ANN)  
CFB\_2 (CFB)  
CFB\_3 (CFB)

## MRG\_MTP Resource Group (Continued)

 **Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions





Navigation Cisco Unified CM Administration Go


administrator | [Search Documentation](#) | [About](#) | [Logout](#)

System ▾ | Call Routing ▾ | Media Resources ▾ | Advanced Features ▾ | Device ▾ | Application ▾ | User Management ▾ | Bulk Administration ▾

Help ▾

**Media Resource Group Configuration** Related Links: Back To Find/List Go

 Save  Delete  Copy  Add New

**Status**  
 Status: Ready

**Media Resource Group Status**  
Media Resource Group: MRG\_MTP (used by 13 devices)


**Media Resource Group Information**  
Name\*   
Description

**Devices for this Group**  
Available Media Resources\*\*  

▼ ▲

Selected Media Resources*
CFB_4 (CFB)
MOH_2 (MOH)
MOH_3 (MOH)
MOH_4 (MOH)
MTP_2 (MTP)





## MRG\_MTP Resource Group (Continued)

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go  
**administrator** | [Search Documentation](#) | [About](#) | [Logout](#)

System ▾ | Call Routing ▾ | Media Resources ▾ | Advanced Features ▾ | Device ▾ | Application ▾ | User Management ▾ | Bulk Administration ▾  
Help ▾

**Media Resource Group Configuration** Related Links: [Back To Find/List](#) Go

 Save  Delete  Copy  Add New

Name\*   
Description

**Devices for this Group**

Available Media Resources\*\*

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.


The screenshot displays the Cisco Unified CM Administration interface for configuring a Media Resource Group. The page title is "Media Resource Group Configuration" and the user is logged in as "administrator". The navigation menu includes System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, and Bulk Administration. The "Media Resource Group Information" section is highlighted with a red box and contains the following fields:

- Name\*: MRG\_noMTP
- Description: MRG without MTP

The "Devices for this Group" section is also highlighted with a red box and contains two lists:

- Available Media Resources\*\*: MTP\_2, MTP\_3, MTP\_4
- Selected Media Resources\*: ANN\_2 (ANN), ANN\_3 (ANN), ANN\_4 (ANN), CFB\_2 (CFB), CFB\_3 (CFB)

## Resource Group for MRG noMTP (Continued)

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions





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System ▾ | Call Routing ▾ | Media Resources ▾ | Advanced Features ▾ | Device ▾ | Application ▾ | User Management ▾ | Bulk Administration ▾

Help ▾

**Media Resource Group Configuration** Related Links: Back To Find/List Go

 Save  Delete  Copy  Add New

**Status**  
 Status: Ready


**Media Resource Group Status**  
Media Resource Group: MRG\_noMTP (used by 29 devices)

**Media Resource Group Information**  
Name\*   
Description

**Devices for this Group**  
Available Media Resources\*\*  
  
  
  
**Selected Media Resources\***



# Cisco Unified Communications Manager Media Resource Group List

**Cisco Unified CM Administration**  
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Help ▾



**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)** Rows per Page 50

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

<input type="checkbox"/>	Name ^	Copy
<input type="checkbox"/>	<a href="#">MRGL_Default</a>	
<input type="checkbox"/>	<a href="#">MRGL_G711</a>	
<input type="checkbox"/>	<a href="#">MRGL_G729</a>	

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

The screenshot displays the Cisco Unified CM Administration interface for configuring a Media Resource Group List. The page title is "Media Resource Group List Configuration". The navigation bar includes "Navigation" with a dropdown menu set to "Cisco Unified CM Administration" and a "Go" button. Below the navigation bar, there are several tabs: "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", and "Bulk Administration". The "Media Resources" tab is active. The main content area shows the configuration for "Media Resource Group List: MRGL\_G711 (used by 13 devices)". The "Media Resource Group List Information" section is highlighted with a red box and contains a "Name\*" field with the value "MRGL\_G711". The "Media Resource Groups for this List" section contains two fields: "Available Media Resource Groups" with the value "MRG\_noMTP" and "Selected Media Resource Groups" with the value "MRG\_MTP". The "Selected Media Resource Groups" field is also highlighted with a red box. At the bottom of the page, there are buttons for "Save", "Delete", "Copy", and "Add New".

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** Related Links: [Back To Find/List](#)

---

**Pattern Definition**

Route Pattern*	3XXX
Route Partition	< None >
Description	Route to Avaya SM
Numbering Plan	-- Not Selected --
Route Filter	< None >
MLPP Precedence*	Default
<input type="checkbox"/> Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	Trunk_to_Avaya_SM <a href="#">(Edit)</a>
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern <input type="text" value="No Error"/>
Call Classification*	OnNet
External Call Control Profile	< None >
<input type="checkbox"/> Allow Device Override <input type="checkbox"/> Provide Outside Dial Tone <input type="checkbox"/> Allow Overlap Sending <input type="checkbox"/> Urgent Priority	
<input type="checkbox"/> Require Forced Authorization Code	
Authorization Level*	0
<input type="checkbox"/> Require Client Matter Code	

---

**Calling Party Transformations**

<input type="checkbox"/> Use Calling Party's External Phone Number Mask	
Calling Party Transform Mask	XXXX
Prefix Digits (Outgoing Calls)	
Calling Line ID Presentation*	Allowed
Calling Name Presentation*	Allowed
Calling Party Number Type*	Cisco CallManager
Calling Party Numbering Plan*	Cisco CallManager

## Route Patter Configuration for 3xxx (Continued)

Set Connected Line ID Presentation\* = Allowed

Set Calling Name Presentation\* = Allowed

All other values are default.

The screenshot displays the Cisco Unified CM Administration interface for configuring a route pattern. The page title is "Route Pattern Configuration" and it includes a navigation menu with options like "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", and "Bulk Administration". The user is logged in as "administrator".

The main configuration area is divided into three sections:

- Connected Party Transformations:** This section is highlighted with a red box. It contains two dropdown menus: "Connected Line ID Presentation\*" and "Connected Name Presentation\*", both of which are set to "Allowed".
- Called Party Transformations:** This section includes fields for "Discard Digits" (set to "< None >"), "Called Party Transform Mask", "Prefix Digits (Outgoing Calls)", "Called Party Number Type\*" (set to "Cisco CallManager"), and "Called Party Numbering Plan\*" (set to "Cisco CallManager").
- ISDN Network-Specific Facilities Information Element:** This section includes a "Network Service Protocol" dropdown (set to "-- Not Selected --"), a "Carrier Identification Code" field, and a table for "Network Service" parameters. The table has three columns: "Network Service", "Service Parameter Name", and "Service Parameter Value". The "Network Service" dropdown is set to "-- Not Selected --" and the "Service Parameter Name" dropdown is set to "< Not Exist >".

At the bottom of the configuration area, there are buttons for "Save", "Delete", "Copy", and "Add New".

# Cisco Unified Communications Manager SIP Phone Ext. 5000 Device Level Configuration

**Cisco Unified CM Administration**

For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go

**administrator** | [Search Documentation](#) | [About](#) | [Logout](#)

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Help ▾

**Find and List Phones** Related Links: [Actively Logged In Device Report](#) Go

+ Add New ⌘ Select All ⌘ Clear All ✖ Delete Selected ↺ Reset Selected ✍ Apply Config to Selected

**Status**

i 8 records found

**Phone (1 - 8 of 8)** Rows per Page 50 ▾

Find Phone where Device Pool ▾ contains ▾ G711 Pool Find Clear Filter ⊕ ⊖

Select item or enter search text ▾

<input type="checkbox"/>		Device Name(Line) ^	Description	Device Pool	Device Protocol	Status	IPv4 Address	Copy	Super Copy
<input type="checkbox"/>		<a href="#">SEP1C17D337D1C9</a>	5000	<a href="#">G711 Pool</a>	SIP	Registered with clus20sub1	<a href="#">10.80.10.36</a>		
<input type="checkbox"/>		<a href="#">SEP005060084CFB</a>	EX60 5013	<a href="#">G711 Pool</a>	SIP	Registered with clus20sub1	<a href="#">10.80.10.32</a>		
<input type="checkbox"/>		<a href="#">CFSUSER02</a>	Cisco Framework User2	<a href="#">G711 Pool</a>	SIP	Registered with clus20sub1	10.64.1.138		
<input type="checkbox"/>		<a href="#">SEP001A2FA6CF0A</a>	5004	<a href="#">G711 Pool</a>	SCCP	Registered with clus20sub1	<a href="#">10.80.10.23</a>		
<input type="checkbox"/>		<a href="#">SEP1C17D337D19F</a>	5017	<a href="#">G711 Pool</a>	SIP	Registered with clus20sub1	<a href="#">10.80.10.35</a>		
<input type="checkbox"/>		<a href="#">SEP001C5856D737</a>	5010	<a href="#">G711 Pool</a>	SCCP	Registered with clus20sub1	<a href="#">10.80.10.34</a>		
<input type="checkbox"/>		<a href="#">CTIRDAvatar1</a>	CTI Avatar Device 1	<a href="#">G711 Pool</a>	CTI Remote Device	Registered with clus20sub1	None		
<input type="checkbox"/>		<a href="#">CTIRDAvatar2</a>	CTI Avatar Device 2	<a href="#">G711 Pool</a>	CTI Remote Device	Registered with clus20sub1	None		

+ Add New ⌘ Select All ⌘ Clear All ✖ Delete Selected ↺ Reset Selected ✍ Apply Config to Selected

## 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.

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and the role "administrator". The main menu shows "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", and "Help". The current page is "Phone Configuration" for a device with ID 5000. The "Status" section shows the device is "Ready". The "Association" section lists various call services, with "Line [1] - 5000 (no partition)" and "Line [2] - Add a new DN" selected. The "Phone Type" section shows "Product Type: Cisco 9971" and "Device Protocol: SIP". The "Real-time Device Status" section shows registration details, including IP address 10.80.10.36. The "Device Information" section contains a list of configuration fields, with several highlighted in red boxes: MAC Address\* (1C17D337D1C9), Description (5000), Device Pool\* (G711 Pool), Phone Button Template\* (Standard 9971 SIP), Media Resource Group List (MRGL\_Default), User Hold MOH Audio Source (1-SampleAudioSource), and Network Hold MOH Audio Source (1-SampleAudioSource).

Field	Value
MAC Address*	1C17D337D1C9
Description	5000
Device Pool*	G711 Pool
Phone Button Template*	Standard 9971 SIP
Media Resource Group List	MRGL_Default
User Hold MOH Audio Source	1-SampleAudioSource
Network Hold MOH Audio Source	1-SampleAudioSource

## 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.

The screenshot displays the Cisco Unified CM Administration interface for configuring a SIP Phone. The page title is "Cisco Unified CM Administration" and the user is logged in as "administrator". The navigation menu includes "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", and "Help". The current page is "Phone Configuration" for "Ext. 5000".

The configuration is organized into several sections:

- BLF:** A list of BLF services (24-31) including Answer Oldest, Do Not Disturb, Services, Record, Alerting Calls, Queue Status, Privacy, and None.
- General Settings:** AAR Group, User Locale, Network Locale, Built In Bridge\*, Privacy\*, Device Mobility Mode\* (with a "View Current" link), and Owner (radio buttons for User and Anonymous (Public/Shared Space)).
- Owner User ID\*:** A dropdown menu currently set to "user1".
- Personalization and Services:** Phone Personalization\*, Services Provisioning\*, Phone Load Name, Use Trusted Relay Point, BLF Audible Alert Setting (Phone Idle)\*, BLF Audible Alert Setting (Phone Busy)\*, Always Use Prime Line\*, Always Use Prime Line for Voice Message\*, Geolocation, and Feature Control Policy.
- Control and Security:** Checkboxes for "Ignore Presentation Indicators (internal calls only)", "Allow Control of Device from CTI" (checked), "Logged Into Hunt Group" (unchecked), "Remote Device", "Protected Device\*\*\*\*", and "Require off-premise location".
- Number Presentation Transformation:** A section for "Caller ID For Calls From This Phone" with a "Calling Party Transformation CSS" dropdown set to "< None >".

## 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.

The screenshot displays the Cisco Unified CM Administration interface for configuring a SIP phone. The top navigation bar includes the Cisco logo, the title 'Cisco Unified CM Administration', and a navigation menu with options like 'System', 'Call Routing', 'Media Resources', 'Advanced Features', 'Device', 'Application', 'User Management', and 'Help'. Below this, the 'Phone Configuration' section is active, showing 'Related Links: Back To Find/List' and a 'Go' button. A toolbar contains icons for 'Save', 'Delete', 'Copy', 'Reset', 'Apply Config', and 'Add New'. The main configuration area is divided into three sections:

- Number Presentation Transformation:**
  - Caller ID For Calls From This Phone:** Calling Party Transformation CSS is set to '< None >'. The checkbox 'Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)' is checked.
  - Remote Number:** Calling Party Transformation CSS is set to '< None >'. The checkbox 'Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)' is checked.
- Protocol Specific Information:**
  - Packet Capture Mode\*: None
  - Packet Capture Duration: 0
  - BLF Presence Group\*: Standard Presence group
  - SIP Dial Rules: < None >
  - MTP Preferred Originating Codec\*: 711ulaw
  - Device Security Profile\*:** Cisco 9971 - Standard SIP Non-Secure Profile (highlighted with a red box)
  - Rerouting Calling Search Space: < None >
  - SUBSCRIBE Calling Search Space: < None >
  - SIP Profile\*:** Early Offer SIP Profile (highlighted with a red box, with a 'View Details' link)
  - Digest User:** user1 (highlighted with a red box)
  - Media Termination Point Required:
  - Unattended Port:
  - Require DTMF Reception:



## Cisco Unified Communications Manager SIP Phone Ext. 5000 Device Level Configuration (Continued)

All other values are default.

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Help

**Phone Configuration** | Related Links: Back To Find/List

Save | Delete | Copy | Reset | Apply Config | Add New

### Certification Authority Proxy Function (CAPF) Information

Certificate Operation\*

Authentication Mode\*

Authentication String

Key Size (Bits)\*

Operation Completes By     (YYYY:MM:DD:HH)

Certificate Operation Status: None  
Note: Security Profile Contains Addition CAPF Settings.

### Expansion Module Information

Module 1

Module 1 Load Name

Module 2

Module 2 Load Name

Module 3

Module 3 Load Name

### External Data Locations Information (Leave blank to use default)

Information

Directory

Messages

Services

Authentication Server

Proxy Server

Idle

Idle Timer (seconds)

Secure Authentication URL

Secure Directory URL

Secure Idle URL

Secure Information URL

## Cisco Unified Communications Manager SIP Phone Ext. 5000 Device Level Configuration (Continued)

All values are default.

**Cisco Unified CM Administration**  
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Navigation Cisco Unified CM Administration Go

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System ▾ | Call Routing ▾ | Media Resources ▾ | Advanced Features ▾ | Device ▾ | Application ▾ | User Management ▾ | Help ▾

**Phone Configuration**

Related Links: Back To Find/List Go

Save ✖ Delete 📄 Copy 🔄 Reset ✎ Apply Config + Add New

Secure Information URL   
 Secure Messages URL   
 Secure Services URL

**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**  

?
Param
Override Common Settings

 Disable Speakerphone  
 Disable Speakerphone and Headset  
 PC Port\* Enabled ▾  
 Back USB Port\* Enabled ▾

## 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.

The screenshot displays the Cisco Unified CM Administration interface for configuring a SIP Phone. The top navigation bar includes the Cisco logo, the title 'Cisco Unified CM Administration', and the user role 'administrator'. Below the navigation bar, there are tabs for 'System', 'Call Routing', 'Media Resources', 'Advanced Features', 'Device', 'Application', 'User Management', and 'Help'. The main content area is titled 'Phone Configuration' and includes a 'Related Links' section with a 'Back To Find/List' button. A toolbar at the top of the configuration area contains icons for 'Save', 'Delete', 'Copy', 'Reset', 'Apply Config', and 'Add New'. The configuration table lists various settings for the SIP Phone, with 'Cisco Camera\*' and 'Video Capabilities\*' highlighted in red. The 'Audio Class' and 'Handsfree' options are also highlighted in blue.

Setting	Value	Checkbox
Side USB Port*	Enabled	<input type="checkbox"/>
<b>Cisco Camera*</b>	<b>Enabled</b>	<input checked="" type="checkbox"/>
Console Access*	Disabled	<input type="checkbox"/>
<b>Video Capabilities*</b>	<b>Enabled</b>	<input checked="" type="checkbox"/>
Enable/Disable USB Classes	Mass Storage Human Interface Device <b>Audio Class</b>	<input type="checkbox"/>
SDIO *	Disabled	<input type="checkbox"/>
Bluetooth *	Enabled	<input type="checkbox"/>
Wifi *	Enabled	<input type="checkbox"/>
Bluetooth Profiles*	<b>Handsfree</b> Human Interface Device	<input type="checkbox"/>
Settings Access*	Enabled	<input type="checkbox"/>
Gratuitous ARP*	Disabled	<input type="checkbox"/>
PC Voice VLAN Access*	Enabled	<input type="checkbox"/>
Web Access*	Disabled	<input type="checkbox"/>
Show All Calls on Primary Line*	Disabled	<input type="checkbox"/>
Days Display Not Active	<b>Sunday</b> Monday Tuesday	<input type="checkbox"/>
Display On Time	07:30	<input type="checkbox"/>
Display On Duration	10:30	<input type="checkbox"/>
Display Idle Timeout	01:00	<input type="checkbox"/>
HTTPS Server*	http and https Enabled	<input type="checkbox"/>
Enable Power Save Plus	Sunday Monday Tuesday	<input type="checkbox"/>
Phone On Time	00:00	<input type="checkbox"/>
Phone Off Time	24:00	<input type="checkbox"/>
Phone Off Idle Timeout*	60	<input type="checkbox"/>
<input type="checkbox"/> Enable Audible Alert		<input type="checkbox"/>
EnergyWise Domain		<input type="checkbox"/>

## Cisco Unified Communications Manager SIP Phone Ext. 5000 Device Level Configuration (Continued)

Set RTCP\* = Enabled. This is used in this example.

All other values are default.

The screenshot displays the Cisco Unified CM Administration interface for configuring a SIP Phone. The page title is "Phone Configuration" and the user is logged in as "administrator". The "RTCP\*" setting is highlighted with a red box and is set to "Enabled".

Setting	Value	Checkbox
EnergyWise Domain		<input type="checkbox"/>
EnergyWise Endpoint Security Secret		<input type="checkbox"/>
Allow EnergyWise Overrides		<input type="checkbox"/>
Span to PC Port*	Disabled	
Logging Display*	Disabled	
Load Server		<input type="checkbox"/>
IPv6 Load Server		<input type="checkbox"/>
Recording Tone*	Disabled	
Recording Tone Local Volume*	100	
Recording Tone Remote Volume*	50	
Recording Tone Duration		
Display On When Incoming Call*	Enabled	<input type="checkbox"/>
<b>RTCP*</b>	<b>Enabled</b>	<input checked="" type="checkbox"/>
Log Server		<input type="checkbox"/>
IPv6 Log Server		<input type="checkbox"/>
Remote Log*	Disabled	<input type="checkbox"/>
Log Profile	Default	<input type="checkbox"/>
Advertise G.722 and iSAC Codecs*	Use System Default	
Wideband Headset UI Control*	Enabled	
Wideband Headset*	Enabled	
Peer Firmware Sharing*	Enabled	<input type="checkbox"/>
Cisco Discovery Protocol (CDP): Switch Port*	Enabled	<input type="checkbox"/>
Cisco Discovery Protocol (CDP): PC Port*	Enabled	<input type="checkbox"/>
Link Layer Discovery Protocol	Enabled	<input type="checkbox"/>

## Cisco Unified Communications Manager SIP Phone Ext. 5000 Device Level Configuration (Continued)

All values are default.

**Cisco Unified CM Administration**

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**administrator** | [Search Documentation](#) | [About](#) | [Logout](#)

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**Phone Configuration** Related Links: Back To Find/List Go

Save
 Delete
 Copy
 Reset
 Apply Config
 Add New

Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port *	<input type="text" value="Enabled"/>	<input type="checkbox"/>
Link Layer Discovery Protocol (LLDP): PC Port *	<input type="text" value="Enabled"/>	<input type="checkbox"/>
LLDP Asset ID	<input type="text"/>	
LLDP Power Priority *	<input type="text" value="Unknown"/>	
802.1x Authentication *	<input type="text" value="User Controlled"/>	<input type="checkbox"/>
FIPS Mode *	<input type="text" value="Disabled"/>	<input type="checkbox"/>
Detect Unified CM Connection Failure *	<input type="text" value="Normal"/>	<input type="checkbox"/>
Switch Port Remote Configuration *	<input type="text" value="Disabled"/>	<input type="checkbox"/>
PC Port Remote Configuration *	<input type="text" value="Disabled"/>	<input type="checkbox"/>
Automatic Port Synchronization *	<input type="text" value="Disabled"/>	<input type="checkbox"/>
Power Negotiation *	<input type="text" value="Enabled"/>	<input type="checkbox"/>
Restrict Data Rates *	<input type="text" value="Disabled"/>	
SSH Access *	<input type="text" value="Disabled"/>	<input type="checkbox"/>
Incoming Call Toast Timer *	<input type="text" value="5"/>	<input type="checkbox"/>
Provide Dial Tone from Release Button *	<input type="text" value="Disabled"/>	<input type="checkbox"/>
Hide Video By Default *	<input type="text" value="Disabled"/>	<input type="checkbox"/>
Background Image	<input type="text"/>	<input type="checkbox"/>
Simplified New Call UI *	<input type="text" value="Disabled"/>	<input type="checkbox"/>
Enable VXC VPN for MAC	<input type="text"/>	
VXC VPN Option *	<input type="text" value="Dual Tunnel"/>	<input type="checkbox"/>
VXC Challenge *	<input type="text" value="Challenge"/>	<input type="checkbox"/>

## Cisco Unified Communications Manager SIP Phone Ext. 5000 Device Level Configuration (Continued)

All values are default.

 **Cisco Unified CM Administration**  
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**Phone Configuration** Related Links: Back To Find/List

VXC-M Servers	<input type="text"/>	<input type="checkbox"/>
Revert to All Calls*	Disabled <input type="button" value="v"/>	<input type="checkbox"/>
80-bit SRTCP*	Disabled <input type="button" value="v"/>	<input type="checkbox"/>
RTCP for Video*	Enabled <input type="button" value="v"/>	<input type="checkbox"/>
Record Call Log from Shared Line*	Disabled <input type="button" value="v"/>	<input type="checkbox"/>
Show Call History for Selected Line Only.*	Disabled <input type="button" value="v"/>	<input type="checkbox"/>
Actionable Incoming Call Alert*	Disabled <input type="button" value="v"/>	<input type="checkbox"/>
DF bit*	0 <input type="button" value="v"/>	<input type="checkbox"/>
Default Line Filter	<input type="text"/>	<input type="checkbox"/>
Separate Audio and Video Mute*	Disabled <input type="button" value="v"/>	<input type="checkbox"/>
Softkey Control*	Feature Control Policy <input type="button" value="v"/>	<input type="checkbox"/>
Start Video Port	<input type="text"/>	<input type="checkbox"/>
Stop Video Port	<input type="text"/>	<input type="checkbox"/>
Lowest Alerting Line State Priority*	Disabled <input type="button" value="v"/>	<input type="checkbox"/>
TLS Resumption Timer*	3600 <input type="text"/>	<input type="checkbox"/>

## 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 CM Administration**  
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go  
administrator | Search Documentation | About | Logout

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**Phone Configuration** Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

**Association**

Modify Button Items

- 1 Line [1] - 5010 (no partition)
- 2 Line [2] - Add a new DN
- 3 Add a new SD
- 4 Add a new SD
- 5 Add a new SD
- 6 Add a new SD
- Unassigned Associated Items -----
- 7 Add a new SD
- 8 Add a new SURL
- 9 Add a new BLF SD
- 10 Add a new BLF Directed Call Park
- 11 CallBack
- 12 Call Park
- 13 Call Pickup
- 14 Conference List
- 15 Conference
- 16 Do Not Disturb
- 17 End Call
- 18 Forward All
- 19 Group Call Pickup
- 20 Hold
- 21 Hunt Group Logout
- 22 Intercom [1] - Add a new Intercom
- 23 Malicious Call Identification
- 24 Meet Me Conference
- 25 Mobility
- 26 New Call
- 27 Other Pickup

**Phone Type**

**Product Type:** Cisco 7961  
**Device Protocol:** SCCP

---

**Real-time Device Status**

**Registration:** Registered with Cisco Unified Communications Manager clus20sub1  
**IPv4 Address:** 10.80.10.34  
**Active Load ID:** SCCP41.9-3-1SR3-1S  
**Download Status:** None

---

**Device Information**

Device is Active  
 Device is trusted

MAC Address\* 001C5856D737  
Description 5010  
Device Pool\* G711 Pool [View Details](#)

Common Device Configuration < None > [View Details](#)

Phone Button Template\* Standard 7961 SCCP

Softkey Template < None >  
Common Phone Profile\* Standard Common Phone Profile [View Details](#)

Calling Search Space < None >  
AAR Calling Search Space < None >

Media Resource Group List MRGL\_G711  
User Hold MOH Audio Source 1-SampleAudioSource  
Network Hold MOH Audio Source 1-SampleAudioSource  
Location\* Hub\_None

## Cisco Unified Communications Manager SCCP Phone Ext. 5010 Device Level Configuration (Continued)

All other values are default.



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administrator | Search Documentation | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Help ▾

**Phone Configuration** Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

25	Mobility
26	New Call
27	Other Pickup
28	Quality Reporting Tool
29	Redial
30	Remove Last Participant
31	Transfer
32	Video Mode
33	Queue Status
34	Privacy
35	None

AAR Group < None >

User Locale < None >

Network Locale < None >

Built In Bridge\* Default

Privacy\* Default

Device Mobility Mode\* Default [View](#)  
[Current Device Mobility Settings](#)

Owner  User  Anonymous (Public/Shared Space)

Owner User ID

Phone Personalization\* Default

Services Provisioning\* Default

Phone Load Name

Single Button Barge Default

Join Across Lines Default

Use Trusted Relay Point\* Default

BLF Audible Alert Setting (Phone Idle)\* Default

BLF Audible Alert Setting (Phone Busy)\* Default

Always Use Prime Line\* Default

Always Use Prime Line for Voice Message\* Default

Geolocation < None >

Retry Video Call as Audio

Ignore Presentation Indicators (internal calls only)

Allow Control of Device from CTI

Logged Into Hunt Group

Remote Device

Protected Device\*\*\*\*

**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.

The screenshot displays the Cisco Unified CM Administration interface for Phone Configuration. The top navigation bar includes the Cisco logo, the title 'Cisco Unified CM Administration', and navigation options like 'Navigation', 'administrator', 'Search Documentation', 'About', and 'Logout'. Below this is a menu bar with categories like 'System', 'Call Routing', 'Media Resources', 'Advanced Features', 'Device', 'Application', 'User Management', and 'Help'. The main content area is titled 'Phone Configuration' and includes a 'Related Links' section with 'Back To Find/List'. A toolbar at the top of the configuration area contains icons for 'Save', 'Delete', 'Copy', 'Reset', 'Apply Config', and 'Add New'. The configuration is organized into several sections:

- Hot line Device\*\*\*\***: Includes checkboxes for 'Hot line Device\*\*\*\*' and 'Require off-premise location'.
- Number Presentation Transformation**:
  - Caller ID For Calls From This Phone**: Includes a dropdown for 'Calling Party Transformation CSS' (set to '< None >') and a checked checkbox for 'Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)'.
  - Remote Number**: Includes a dropdown for 'Calling Party Transformation CSS' (set to '< None >') and a checked checkbox for 'Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)'.
- Protocol Specific Information**: Includes dropdowns for 'Packet Capture Mode\*' (None), 'Packet Capture Duration' (0), 'BLF Presence Group\*' (Standard Presence group), and 'Device Security Profile\*' (Cisco 7961 - Standard SCCP Non-Secure Profile, highlighted with a red box). It also includes a dropdown for 'SUBSCRIBE Calling Search Space' (< None >) and checkboxes for 'Unattended Port', 'Require DTMF Reception', and 'RFC2833 Disabled'.
- Certification Authority Proxy Function (CAPF) Information**: Includes dropdowns for 'Certificate Operation\*' (No Pending Operation), 'Authentication Mode\*' (By Null String), and 'Authentication String'. It features a 'Generate String' button, a dropdown for 'Key Size (Bits)\*' (1024), and a field for 'Operation Completes By' (2014 2 2 12 (YYYY:MM:DD:HH)). It also shows 'Certificate Operation Status: None' and a note: 'Note: Security Profile Contains Addition CAPF Settings.'

## Cisco Unified Communications Manager SCCP Phone Ext. 5010 Device Level Configuration (Continued)

These values are default.

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

administrator | Search Documentation | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Help ▾

**Phone Configuration** Related Links: Back To Find/List

**Expansion Module Information**

Module 1	< None >
Module 1 Load Name	
Module 2	< None >
Module 2 Load Name	

**External Data Locations Information (Leave blank to use default)**

Information	
Directory	
Messages	
Services	
Authentication Server	
Proxy Server	
Idle	
Idle Timer (seconds)	
Secure Authentication URL	
Secure Directory URL	
Secure Idle URL	
Secure Information URL	
Secure Messages URL	
Secure Services URL	

**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 >

## Cisco Unified Communications Manager SCCP Phone Ext. 5010 Device Level Configuration (Continued)

These values are default.

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Help ▾

**administrator** | [Search Documentation](#) | [About](#) | [Logout](#)

**Phone Configuration**

Related Links: Back To Find/List Go

Save
 Delete
 Copy
 Reset
 Apply Config
 Add New

**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**

	Param	Override Common Settings
<input type="checkbox"/> Disable Speakerphone		
<input type="checkbox"/> Disable Speakerphone and Headset		
Forwarding Delay*	Disabled ▾	
PC Port *	Enabled ▾	
Settings Access*	Enabled ▾	<input type="checkbox"/>
Gratuitous ARP*	Disabled ▾	
PC Voice VLAN Access*	Enabled ▾	
Video Capabilities*	Disabled ▾	<input type="checkbox"/>
Auto Line Select*	Disabled ▾	
Web Access*	Disabled ▾	<input type="checkbox"/>
Enable Power Save Plus	Sunday Monday Tuesday ▾	<input type="checkbox"/>
Phone On Time	00:00	<input type="checkbox"/>
Phone Off Time	24:00	<input type="checkbox"/>
Phone Off Idle Timeout*	60	<input type="checkbox"/>
<input type="checkbox"/> Enable Audible Alert		<input type="checkbox"/>
EnergyWise Domain		<input type="checkbox"/>
EnergyWise Endpoint Security Secret		<input type="checkbox"/>
<input type="checkbox"/> Allow EnergyWise Overrides		<input type="checkbox"/>


## Cisco Unified Communications Manager SCCP Phone Ext. 5010 Device Level Configuration (Continued)

These values are default.

Cisco Unified CM Administration		Navigation	Cisco Unified CM Administration	Go
For Cisco Unified Communications Solutions		administrator	Search Documentation	About   Logout
System	Call Routing	Media Resources	Advanced Features	Device
Application	User Management	Help		
Phone Configuration		Related Links: Back To Find/List		
Save            Delete            Copy            Reset            Apply Config            Add New				
Span to PC Port*	Disabled			
Logging Display*	PC Controlled			
Load Server	<input type="text"/>			
Recording Tone*	Disabled			
Recording Tone Local Volume*	100			
Recording Tone Remote Volume*	50			
Recording Tone Duration	<input type="text"/>			
RTCP*	Disabled			
"more" Soft Key Timer	5			
Auto Call Select*	Enabled			
Log Server	<input type="text"/>			
Advertise G.722 Codec*	Use System Default			
Wideband Headset UI Control*	Enabled			
Wideband Handset UI Control*	Enabled			
Wideband Headset*	Enabled			
Wideband Handset*	Use Phone Default			
Peer Firmware Sharing*	Enabled			
Cisco Discovery Protocol (CDP): Switch Port*	Enabled			
Cisco Discovery Protocol (CDP): PC Port*	Enabled			
Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port*	Enabled			
Link Layer Discovery Protocol (LLDP): PC Port*	Enabled			
LLDP Asset ID	<input type="text"/>			
LLDP Power Priority*	Unknown			
Display Refresh Rate*	Normal			

## Cisco Unified Communications Manager SCCP Phone Ext. 5010 Device Level Configuration (Continued)

These values are default.







 **Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration  Go

administrator | Search Documentation | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Help ▾

**Phone Configuration** Related Links: Back To Find/List  Go

 Save  Delete  Copy  Reset  Apply Config  Add New

IPv6 Load Server	<input type="text"/>	<input type="checkbox"/>
IPv6 Log Server	<input type="text"/>	<input type="checkbox"/>
802.1x Authentication*	User Controlled <input type="text"/>	<input type="checkbox"/>
Detect Unified CM Connection Failure*	Normal <input type="text"/>	<input type="checkbox"/>
Minimum Ring Volume*	0-Silent <input type="text"/>	<input type="checkbox"/>
Headset Sidetone Level*	Use Phone Default <input type="text"/>	<input type="checkbox"/>
HTTPS Server*	http and https Enabled <input type="text"/>	<input type="checkbox"/>
Enbloc Dialing*	Enabled <input type="text"/>	<input type="checkbox"/>
Switch Port Remote Configuration*	Disabled <input type="text"/>	<input type="checkbox"/>
PC Port Remote Configuration*	Disabled <input type="text"/>	<input type="checkbox"/>
Automatic Port Synchronization*	Disabled <input type="text"/>	<input type="checkbox"/>
SSH Access*	Disabled <input type="text"/>	<input type="checkbox"/>
LOGIN Access*	Enabled <input type="text"/>	<input type="checkbox"/>
FIPS Mode*	Disabled <input type="text"/>	<input type="checkbox"/>
80-bit SRTP*	Disabled <input type="text"/>	<input type="checkbox"/>

## 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.

The screenshot displays the Cisco Unified CM Administration interface for configuring service parameters. The page title is "Service Parameter Configuration" and it shows a list of parameters under the heading "Clusterwide Parameters (System - Location and Region)". The parameters are listed in a table with columns for the parameter name, its current value, and its default value. The parameter "Accept Audio Codec Preferences in Received Offer" is highlighted with a red box and is currently set to "Off".

Parameter Name	Current Value	Default Value
Enforce Millisecond Packet Size *	True	True
Locations Trace Details Enabled *	False	False
Preferred G.711 Millisecond Packet Size *	20	20
Preferred G.722 Millisecond Packet Size *	20	20
Preferred G.723.1 Millisecond Packet Size *	30	30
Preferred G.729 Millisecond Packet Size *	20	20
Always Use Preferred G.729 Packet Size For SIP Trunk Answers *	False	False
Preferred GSM EFR Bytes Packet Size *	31	31
G.711 A-law Codec Enabled *	Enabled for All Devices	Enabled for All Devices
G.711 mu-law Codec Enabled *	Enabled for All Devices	Enabled for All Devices
G.722 Codec Enabled *	Disabled	Enabled for All Devices
iLBC Codec Enabled *	Disabled	Enabled for All Devices
iSAC Codec Enabled *	Enabled for All Devices	Enabled for All Devices
Default Intra-region Max Audio Bit Rate *	64 kbps (G.722, G.711)	64 kbps (G.722, G.711)
Default Inter-region Max Audio Bit Rate *	8 kbps (G.729)	8 kbps (G.729)
Default Intra-region Max Video Call Bit Rate (Includes Audio) *	384	384
Default Inter-region Max Video Call Bit Rate (Includes Audio) *	384	384
Default Intra-region Max Immersive Video Call Bit Rate (Includes Audio) *	2000000000	2000000000
Default Inter-region Max Immersive Video Call Bit Rate (Includes Audio) *	2000000000	2000000000
Use Video BandwidthPool for Immersive Video Calls *	True	True
Default Intra-region and Inter-region Link Loss Type *	Low Loss	Low Loss
Default Audio Codec List between Regions *	Factory Default low loss	Factory Default low loss
Default Audio Codec List within Region *	Factory Default low loss	Factory Default low loss
Accept Audio Codec Preferences in Received Offer *	Off	Off
G.Clear Bandwidth Override	False	False

## 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.

The screenshot displays the Cisco Unified CM Administration web interface. At the top, the navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration For Cisco Unified Communications Solutions", and a navigation dropdown menu set to "Cisco Unified CM Administration" with a "Go" button. Below this is a secondary navigation bar with links for "administrator", "Search Documentation", "About", and "Logout". A main navigation menu contains dropdowns for "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", and "Help".

The main content area is titled "Find and List Audio Codec Preference Lists". It features a toolbar with buttons for "Add New", "Select All", "Clear All", and "Delete Selected". Below the toolbar is a "Status" box containing two information icons: "1 records deleted" and "5 records found".

The "Audio Codec Preference Lists (1 - 5 of 5)" section includes a search filter with a dropdown for "Name" and a "begins with" input field, along with "Find", "Clear Filter", and pagination buttons. Below the search is a table with the following data:

<input type="checkbox"/>	Name ^	Description	Copy
<input type="checkbox"/>	<a href="#">Codec Pref.List - Conf Bridges and G711 Endpoints</a>	Codec Preference for Conf Bridges and G711 Endpoints	
	<a href="#">Factory Default lossy</a>	Lossy Codec List	
	<a href="#">Factory Default low loss</a>	Low Loss Codec List	
<input type="checkbox"/>	<a href="#">G711 Preferred</a>	G711 Preferred	
<input type="checkbox"/>	<a href="#">G729 Preferred</a>	G729 Preferred	

At the bottom of the table area are buttons for "Add New", "Select All", "Clear All", and "Delete Selected". The rows for "G711 Preferred" and "G729 Preferred" are highlighted with a red rectangular box.



## 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.

The screenshot shows the Cisco Unified CM Administration interface for configuring an Audio Codec Preference List. The page title is "Audio Codec Preference List Configuration". The navigation menu includes System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, and Help. The breadcrumb trail is "Navigation > Cisco Unified CM Administration > administrator | Search Documentation". The "Related Links" section contains "Back To Find/List" and "Go". The action bar includes "Save", "Delete", "Copy", and "Add New". The configuration form has the following fields:

Name*	G711 Preferred
Description*	G711 Preferred
Codecs in List*	G.711 U-Law 64k G.729 8k AMR-WB (7k-24k) AMR (5k-13k) MP4A-LATM 128k AAC-LD (MP4A Generic) MP4A-LATM 64k MP4A-LATM 56k L16 256k MP4A-LATM 48k ISAC 32k MP4A-LATM 32k G.722 64k G.722.1 32k G.722 56k G.722.1 24k G.722 48k MP4A-LATM 24k G.711 A-Law 64k G.711 U-Law 56k G.711 A-Law 56k ILBC 16k G.728 16k

## 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.

The screenshot displays the Cisco Unified CM Administration interface for configuring an Audio Codec Preference List. The page title is "Audio Codec Preference List Configuration". The navigation menu includes System, Call Routing, Media Resources, Advanced Features, Device, Application, and User Management. The user is logged in as "administrator".

Below the navigation menu, there are action buttons: Save, Delete, Copy, and Add New. The "Audio Codec Preference List Information" section contains the following fields:

Name*	G729 Preferred
Description*	G729 Preferred
Codecs in List*	G.729 8k G.729a 8k AMR-WB (7k-24k) AMR (5k-13k) MP4A-LATM 128k AAC-LD (MP4A Generic) MP4A-LATM 64k MP4A-LATM 56k L16 256k MP4A-LATM 48k ISAC 32k MP4A-LATM 32k G.722 64k G.722.1 32k G.722 56k G.722.1 24k G.722 48k MP4A-LATM 24k G.711 A-Law 64k G.711 U-Law 64k G.711 U-Law 56k G.711 A-Law 56k ILBC 16k G.728 16k

## Cisco Unified Communications Manager Region Configuration

G711 Region and G729 Region created in this example.

All other values are default.

The screenshot displays the Cisco Unified CM Administration web interface. At the top, the navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration For Cisco Unified Communications Solutions", and the user role "administrator". A secondary navigation bar lists menu items: System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, and Help. Below this is a "Find and List Regions" section with action buttons: Add New, Select All, Clear All, and Delete Selected. A "Status" box indicates "3 records found". The main content area shows a table of regions with a search filter set to "begins with". The table lists three regions: "Default", "G711 Region", and "G729 Region". The "G711 Region" and "G729 Region" rows are highlighted with a red border. At the bottom of the table are buttons for "Add New", "Select All", "Clear All", and "Delete Selected".

	Name ^
<input type="checkbox"/>	Default
<input type="checkbox"/>	G711 Region
<input type="checkbox"/>	G729 Region

## 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.

The screenshot shows the Cisco Unified CM Administration interface for Region Configuration. The 'Name\*' field is set to 'G711 Region'. The 'Region Relationships' table is as follows:

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
Default	Use System Default (Factory Default low loss)	64 kbps (G.722, G.711)	384 kbps	2147483647 kbps
G711 Region	G711 Preferred	64 kbps (G.722, G.711)	384 kbps	2147483647 kbps
G729 Region	G729 Preferred	8 kbps (G.729)	384 kbps	2147483647 kbps
NOTE: Regions not displayed	Use System Default	Use System Default	Use System Default	Use System 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 (G7.22, G7.11). This is used in this example

All other values are default.

The screenshot displays the Cisco Unified CM Administration interface for Region Configuration. The 'Name' field is set to 'G729 Region'. Below it, the 'Region Relationships' table is shown with the following data:

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
Default	Use System Default (Factory Default low loss)	64 kbps (G.722, G.711)	384 kbps	2147483647 kbps
G711 Region	G729 Preferred	8 kbps (G.729)	384 kbps	2147483647 kbps
G729 Region	G729 Preferred	64 kbps (G.722, G.711)	384 kbps	2147483647 kbps

NOTE: Regions not displayed Use System Default Use System Default Use System Default Use System Default

## Cisco Unified Communications Manager Device Pool Configuration

G711 Pool and G729 Pool created in this example.

All other values are default.

The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration For Cisco Unified Communications Solutions", and the user role "administrator". Below the navigation bar is a menu with options like System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, and Help. The main content area is titled "Find and List Device Pools" and includes buttons for "Add New", "Select All", "Clear All", and "Delete Selected". A status bar indicates "3 records found". Below this is a table of device pools. The table has columns for Name, Cisco Unified CM Group, Region, Date/Time Group, and Copy. The rows are: Default (Default, Default, CMLocal), G711 Pool (Default, G711 Region, CMLocal), and G729 Pool (Default, G729 Region, CMLocal). The G711 Pool and G729 Pool rows are highlighted with a red box.

<input type="checkbox"/>	Name ^	Cisco Unified CM Group	Region	Date/Time Group	Copy
<input type="checkbox"/>	<a href="#">Default</a>	<a href="#">Default</a>	<a href="#">Default</a>	<a href="#">CMLocal</a>	
<input type="checkbox"/>	<a href="#">G711 Pool</a>	<a href="#">Default</a>	<a href="#">G711 Region</a>	<a href="#">CMLocal</a>	
<input type="checkbox"/>	<a href="#">G729 Pool</a>	<a href="#">Default</a>	<a href="#">G729 Region</a>	<a href="#">CMLocal</a>	

## **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 Configuration

Related Links: [Back To Find/List](#) ▾ [Go](#)

Save Delete Copy Reset Apply Config Add New

#### Device Pool Settings

Device Pool Name*	G711 Pool
Cisco Unified Communications Manager Group*	Default ▾
Calling Search Space for Auto-registration	< None > ▾
Adjunct CSS	< None > ▾
Reverted Call Focus Priority	Default ▾
Intercompany Media Services Enrolled Group	< None > ▾

#### Local Route Group Settings

Standard Local Route Group	< None > ▾
----------------------------	------------

#### Roaming Sensitive Settings

Date/Time Group*	CMLocal ▾
Region*	G711 Region ▾
Media Resource Group List	MRGL_G711 ▾
Location	< None > ▾
Network Locale	< None > ▾
SRST Reference*	Disable ▾
Connection Monitor Duration***	
Single Button Barge*	Default ▾
Join Across Lines*	Default ▾
Physical Location	< None > ▾
Device Mobility Group	< None > ▾
Wireless LAN Profile Group	< None > ▾ <a href="#">View Details</a>

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

All values are default.





**Device Pool Configuration**

Related Links: Back To Find/List ▾ Go

Save 
 Delete 
 Copy 
 Reset 
 Apply Config 
 Add New

**- Device Mobility Related Information \*\*\*\***

Device Mobility Calling Search Space

AAR Calling Search Space

AAR Group

Calling Party Transformation CSS

Called Party Transformation CSS

**- Geolocation Configuration**

Geolocation

Geolocation Filter

**- 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 in which case there is no prefix assigned.

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	<input type="text" value="Default"/>	<input type="checkbox"/>	<input style="width: 100px;" type="text" value=" &lt; None &gt; "/>
International Number	<input type="text" value="Default"/>	<input type="checkbox"/>	<input style="width: 100px;" type="text" value=" &lt; None &gt; "/>
Unknown Number	<input type="text" value="Default"/>	<input type="checkbox"/>	<input style="width: 100px;" type="text" value=" &lt; None &gt; "/>
Subscriber Number	<input type="text" value="Default"/>	<input type="checkbox"/>	<input style="width: 100px;" type="text" value=" &lt; None &gt; "/>

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

All values are default.



**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administr

**administrator** | [Search Documentation](#)

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management

Help ▾

**Device Pool Configuration** Related Links: Back To Find/List Go

Save
 Delete
 Copy
 Reset
 Apply Config
 Add New

**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 er which case there is no prefix assigned.

Clear Prefix Settings
Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value=" &lt; None &gt;"/>
International Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value=" &lt; None &gt;"/>
Unknown Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value=" &lt; None &gt;"/>
Subscriber Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value=" &lt; None &gt;"/>

**Phone Settings**

**Caller ID For Calls From This Phone**

Calling Party Transformation CSS

**Connected Party Settings**

Connected Party Transformation CSS

**Redirecting Party Settings**

Redirecting Party Transformation CSS

## 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


The screenshot displays the Cisco Unified CM Administration interface for Device Pool Configuration. The page is titled "Device Pool Configuration" and includes a navigation bar with "Navigation Cisco Unified CM Administration" and "administrator" user information. The main content area is divided into three sections:

- Device Pool Settings:** This section contains several configuration fields. A red box highlights the following fields:
  - Device Pool Name\*: G729 Pool
  - Cisco Unified Communications Manager Group\*: Default
- Local Route Group Settings:** This section contains a single field: Standard Local Route Group: < None >
- Roaming Sensitive Settings:** This section contains several configuration fields. A red box highlights the following fields:
  - Date/Time Group\*: CMLocal
  - Region\*: G729 Region
  - Media Resource Group List: MRGL\_G729

Other fields in the Device Pool Settings section include: Calling Search Space for Auto-registration: < None >, Adjunct CSS: < None >, Reverted Call Focus Priority: Default, and Intercompany Media Services Enrolled Group: < None >. Other fields in the Roaming Sensitive Settings section include: Location: < None >, Network Locale: < None >, SRST Reference\*: Disable, Connection Monitor Duration\*\*\*: (empty), Single Button Barge\*: Default, Join Across Lines\*: Default, Physical Location: < None >, Device Mobility Group: < None >, and Wireless LAN Profile Group: < None >.

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

All values are default.



**Cisco Unified CM Administration**  
 For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administr  
**administrator** | [Search Documentation](#)

---

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management
Help ▾

**Device Pool Configuration** Related Links: Back To Find/List Go

Save  Delete  Copy  Reset  Apply Config  Add New

**- 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 (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings
Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	<span style="border: 1px solid #ccc; padding: 2px;">Default</span>	<input type="checkbox"/>	<span style="border: 1px solid #ccc; padding: 2px;">&lt; None &gt;</span>
International Number	<span style="border: 1px solid #ccc; padding: 2px;">Default</span>	<input type="checkbox"/>	<span style="border: 1px solid #ccc; padding: 2px;">&lt; None &gt;</span>
Unknown Number	<span style="border: 1px solid #ccc; padding: 2px;">Default</span>	<input type="checkbox"/>	<span style="border: 1px solid #ccc; padding: 2px;">&lt; None &gt;</span>
Subscriber Number	<span style="border: 1px solid #ccc; padding: 2px;">Default</span>	<input type="checkbox"/>	<span style="border: 1px solid #ccc; padding: 2px;">&lt; None &gt;</span>

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

All values are default.

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**Device Pool Configuration**
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**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 er which case there is no prefix assigned.

Clear Prefix Settings
Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value=" &lt; None &gt;"/>
International Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value=" &lt; None &gt;"/>
Unknown Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value=" &lt; None &gt;"/>
Subscriber Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value=" &lt; None &gt;"/>

**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

<b>Acronym</b>	<b>Definition</b>
CCBS	Call Completion to Busy Subscriber
CCNR	Call Completion on No Reply
CFB	Call Forwarding on Busy
CFNR	Call Forwarding No Reply
CFU	Call Forwarding Unconditional
CLIP	Calling Line (Number) Identification Presentation
CLIR	Calling Line (Number) Identification Restriction
CNIP	Calling Name Identification Presentation
CNIR	Calling Name Identification Restriction
COLP	Connected Line (Number) Identification Presentation
COLR	Connected Line (Number) Identification Restriction
CONP	Connected Name Identification Presentation
CONR	Connected Name Identification Restriction
CT	Call Transfer
Cisco UCM	Cisco Unified Communications Manager
DNS	Domain Name Server
FQDN	Fully Qualified Domain Name
MWI	Message Waiting Indicator
MRGL	Media Resource Group List
MTP	Media Termination Point
PSTN	Public Switched Telephone Network
SIP	Session Initiated Protocol

## **Important Information**

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