



Configuring H.323 Signaling and IP Trunks between Avaya Communication Manager 4.0 and Cisco Unified CallManager 5.1.3 - Issue 1.1

Abstract

These Application Notes present a sample configuration using an H.323 Signaling Group and an IP Trunk Group between Avaya Communication Manager 4.0 and Cisco Unified CallManager 5.1.3. IP-IP Direct Audio calling (shuffling) between Avaya IP telephones and Cisco IP telephones is verified. The sample configuration made use of an Avaya S8710 Server but should be applicable to other Avaya Servers and Media Gateways.

1. Introduction

These Application Notes present a sample configuration for a network comprised of an Avaya S8710 Server IP Connect configuration and a Cisco Unified CallManager. The focus is on the configuration of the H.323 Signaling Group and IP Trunk Group on the Avaya S8710 Server running Avaya Communication Manager 4.0 and the corresponding configuration of the H.323 Gateways on the Cisco Unified CallManager 5.1.3. Since Cisco Unified CallManager 5.1.3 supports the equivalent of IP-IP Direct Audio functionality (shuffling), shuffling between Avaya and Cisco IP telephones is also verified. Using the configuration described herein, Cisco IP telephones controlled by the Cisco Unified CallManager 5.1.3 can call (and be called) by Avaya IP telephones and other Avaya telephones associated with the Avaya S8710 Server.

These Application Notes are an update to the previously published Application Notes entitled “Configuring H.323 Signaling and IP Trunks between Avaya Communication Manager and Cisco Call Manager 4.0 - Issue 1.0”, 4/8/2005.

Figure 1 shows the network setup used for the configuration.

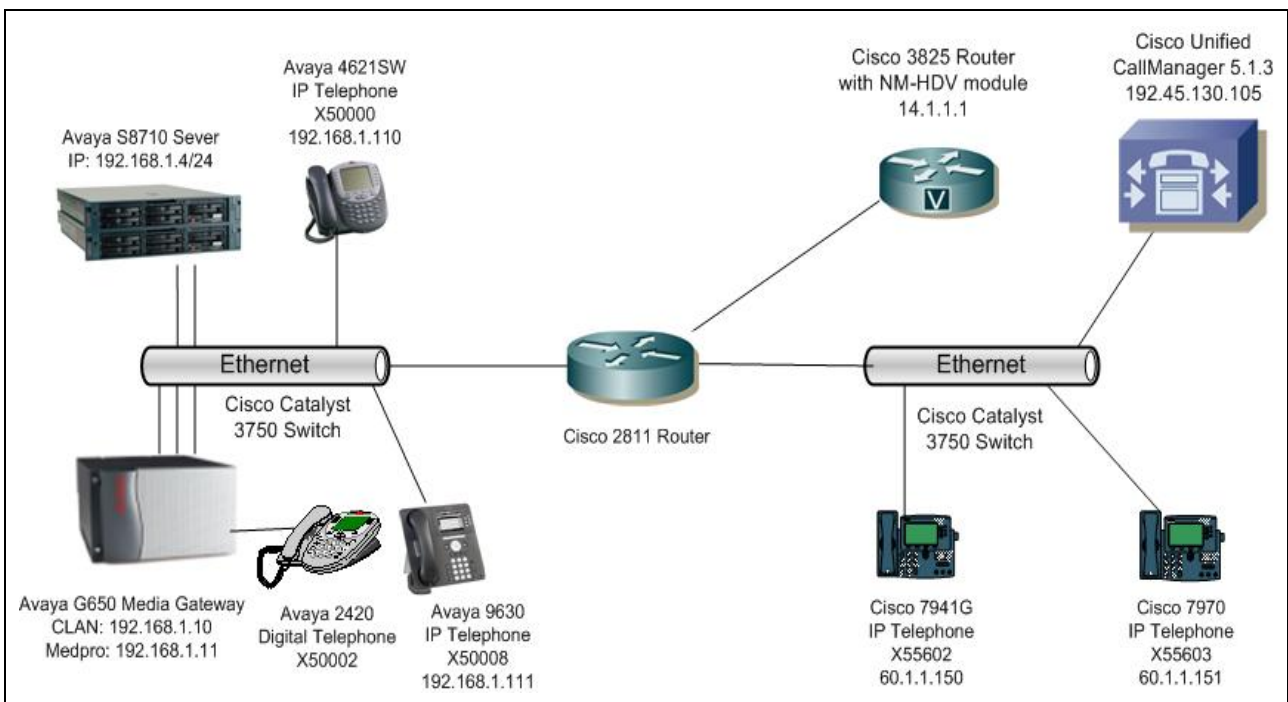


Figure 1: Avaya-Cisco H.323 Interoperability Configuration

2. Hardware and Software Used for Verification

Table 1 lists the equipment and software used for verification.

Equipment	Software
Avaya S8710 Server	R014x.00.1.731.2 with Service Pack 2 (patch 14576)
Avaya G650 Media Gateway with <ul style="list-style-type: none">• C-LAN• MEDPRO	HW01 FW024 HW20 FW095
Avaya 9630 IP Telephone	R2.1 (H.323)
Avaya 4621SW IP Telephone	R2.8 (H.323)
Cisco 3825 Router	IOS 12.4(15)T1
Cisco 2811 Router	IOS 12.4(15)T1
Cisco 3750 Catalyst Switch	IOS12.2(25)SEA
Cisco Unified CallManager	Release 5.1.3.1000-12
Cisco 7970 and 7941G IP Telephones	Release 8.3-2S

Table 1: Hardware and Software Used for Verification

3. Avaya S8710 Server Software Configuration

This section presents configuration steps for the Avaya S8710 Server. It is assumed that Avaya Communication Manager has been installed and the login and password credentials are available to the reader.

In these Application Notes, the Avaya Communication Manager administration is performed using the SAT interface.

3.1. Add Node Name and Map IP Address

The following configuration displays a subset of the **change node-names ip** screen that maps logical names to IP address. These node names are presented because they will appear in other screens, such as the screen defining the H.323 signaling group to the Cisco Unified CallManager 5.1.3.

change node-names ip				Page 1 of 1		
Name		IP Address		Name		IP Address
C-LAN	192 .168 .1 .10					.
CallManager5.1	192 .45 .130 .105					.
MedPro	192 .168 .1 .11					.

3.2. Configure C-LAN and MEDPRO

Use the command **add ip-interface** to add and configure the C-LAN and the MEDPRO of the Avaya G650 Media Gateway. The following two screens display the configurations of the C-LAN (01A02) and the MEDPRO (01A03). Note that the C-LAN and MEDPRO are assigned to Network Region 1.

```
display ip-interface 01a02                                     Page 1 of 1
1
IP INTERFACES

Type: C-LAN
Slot: 01A02
Code/Suffix: TN799 D
Node Name: C-LAN
IP Address: 192 .168 .1 .10
Subnet Mask: 255.255.255.0                                     Link: 1
Gateway Address: 192 .168 .1 .1
Enable Ethernet Port? y                                         Allow H.323 Endpoints? y
Network Region: 1                                              Allow H.248 Gateways? y
VLAN: n                                                         Gatekeeper Priority: 5

Target socket load and Warning level: 400
Receive Buffer TCP Window Size: 8320
ETHERNET OPTIONS
Auto? Y
```

```
display ip-interface 1a03
IP INTERFACES

Type: MEDPRO
Slot: 01A03
Code/Suffix: TN2302
Node Name: Medpro
IP Address: 192 .168 .1 .11
Subnet Mask: 255.255.255.0
Gateway Address: 192 .168 .1 .1
Enable Ethernet Port? y
Network Region: 1
VLAN: n

ETHERNET OPTIONS
Auto? y
```

3.3. Configure IP Codec Sets

In these Application Notes, a total of two IP network regions are used. IP network region 1 is used for the Avaya location and IP network region 3 is used for the Cisco Unified CallManager location. The G.711ulaw codec is used within each region and the G.729B codec is used between these two IP network regions. The following screens display the configuration for IP codec set 1 and 3.

change ip-codec-set 1

Page 1 of 2

IP Codec Set

Codec Set: 1

Audio	Silence	Frames	Packet
Codec	Suppression	Per Pkt	Size(ms)
1: G.711MU	n	20	
2:			

Media Encryption

1: none

change ip-codec-set 3

Page1 of 2

IP Codec Set

Codec Set: 3

Audio	Silence	Frames	Packet
Codec	Suppression	Per Pkt	Size(ms)
1: G.729B	n	2	20
2: G.729AB	n	2	20

Media Encryption

1: none

3.4. Configure IP Network Regions

The following illustrates the configuration for network region 1. The intent of illustrating the network region is to show that Codec Set 1 is used in this region and that the **Intra-region IP-IP Direct Audio** is set to **yes**. The **Inter-region IP-IP Direct Audio** field is also set to **yes** to make sure the media path goes directly between phones without involving the Medpro.

```
change ip-network-region 1                                     Page 1 of 19
19

                                IP NETWORK REGION

Region: 1
Location: 1      Authoritative Domain:
      Name: Avaya
MEDIA PARAMETERS                                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1                                Inter-region IP-IP Direct Audio: yes
      UDP Port Min: 16384                        IP Audio Hairpinning? n
      UDP Port Max: 32767
DIFFSERV/TOS PARAMETERS                        RTCP Reporting Enabled? y
      Call Control PHB Value: 46                  RTCP MONITOR SERVER PARAMETERS
      Audio PHB Value: 46                        Use Default Server Parameters? y
      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
```

Note that on page 3, codec set 1 is used in IP network region 1 and codec set 3 is used between IP network region 1 and IP network region 3.

```
change ip-network-region 1                                     Page 3 of 19
19

                                Inter Network Region Connection Management

src dst codec direct  WAN-BW-limits  Video                               Dyn
rgn rgn  set   WAN  Units    Total Norm  Prio Shr Intervening-regions  CAC IGAR
1   1   1
1   2
1   3   3      y   NoLimit                                     n
```

The following screen shows the configuration for network region 3. Similar to the region 1 configuration, Codec Set 3 is configured and the **Intra-region IP-IP Direct Audio** field is set to **yes**. The **Inter-region IP-IP Direct Audio** field is also set to **yes** to make sure the media path goes directly between phones without involving the Medpro.

```

change ip-network-region 3                                     Page 1 of 19

                                IP NETWORK REGION

Region: 3
Location:                Authoritative Domain:
Name: CallManager
MEDIA PARAMETERS
  Codec Set: 3
  UDP Port Min: 16384
  UDP Port Max: 32767
  Intra-region IP-IP Direct Audio: yes
  Inter-region IP-IP Direct Audio: yes
  IP Audio Hairpinning? y
DIFFSERV/TOS PARAMETERS
  Call Control PHB Value: 46
  Audio PHB Value: 46
  Video PHB Value: 26
  RTCP Reporting Enabled? y
  RTCP MONITOR SERVER PARAMETERS
  Use Default Server Parameters? y
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
  H.323 Link Bounce Recovery? y
  Idle Traffic Interval (sec): 20
  Keep-Alive Interval (sec): 5
  Keep-Alive Count: 5
  RSVP Enabled? n

```

Also, on page 3, codec set 3 is selected for calls between region 1 and region 3.

```

change ip-network-region 3                                     Page 3 of
19

                                Inter Network Region Connection Management

src dst codec direct  WAN-BW-limits  Video
rgn rgn set  WAN Units  Total Norm  Prio Shr Intervening-regions  CAC IGA  Dyn
3  1  3  y  NoLimit
3  2
3  3  3
3  4

```

3.5. Configure IP Network Map

Use the **change ip-network-map** command to put all devices that are on 192.168.1.0 network (Avaya site) into region 1.

```
change ip-network-map                                     Page 1 of
32
```

IP ADDRESS MAPPING						
From IP Address	(To IP Address	Subnet	Region	VLAN	Emergency	Location
	or Mask)				Extension	
192 .168 .1 .1	192 .168 .1 .254		1	n		

3.6. Configure H.323 Signaling Group

This section focuses on the parameter settings recommended for the H.323 signaling group and IP trunk group used to connect with the Cisco Unified CallManager.

Signaling group 3 will be created to establish an H.323 signaling link between the C-LAN in the Avaya G650 Media Gateway and the Cisco Unified CallManager. The signaling group number is not relevant; use any available signaling group number. Use the **add signaling-group 3** command to add the signaling group.

This signaling group uses the C-LAN whose node-name is **C-LAN** as the near end, and the Cisco Unified CallManager node-name **CallManager5.1** as the far end. Retain the default near-end listen port (1720) and enter 1720 as the far-end listen port. The **Calls Share IP Signaling Connection** field should remain set to the default **n** setting. The **Direct IP-IP Audio Connections** field can be set to **yes** to allow the final media path for a call to be **direct** from the Avaya IP telephones to Cisco IP telephones.

The far-end network region field can optionally be populated with a network region number to associate with the Cisco Unified CallManager. For the signaling group shown here, the far-end network region is set to 3 so that the calls between region 1 and region 3 will use codec set 3 as configured.

```
add signaling-group 3                                     Page 1 of 5
                                     SIGNALING GROUP
Group Number: 3          Group Type: h.323
Remote Office? n        Max number of NCA TSC: 0
SBS? n                  Max number of CA TSC: 0
IP Video? n             Trunk Group for NCA TSC:
Trunk Group for Channel Selection:
TSC Supplementary Service Protocol: a      Network Call Transfer? n
T303 Timer(sec): 10

Near-end Node Name: C-LAN          Far-end Node Name: CallManager5.1
Near-end Listen Port: 1720        Far-end Listen Port: 1720
Far-end Network Region: 3
LRQ Required? n          Calls Share IP Signaling Connection? n
RRQ Required? n          H245 Control Addr On FACility? n
Media Encryption? n      Bypass If IP Threshold Exceeded? n
                          H.235 Annex H Required? n
DTMF over IP: out-of-band Direct IP-IP Audio Connections? y
Link Loss Delay Timer(sec): 90      IP Audio Hairpinning? n
Enable Layer 3 Test? n            Interworking Message: PROGRESS

DCP/Analog Bearer Capability: 3.1kHz
```

3.7. Configure IP Trunk Group

Use the **add trunk-group 3** command to create an H.323 IP trunk group on the Avaya S8710 Server. Most fields can be left at their defaults. Data has been entered in the fields shown in **bold**. Note that the trunk **Carrier Medium** is H.323 and Service type is set to **tie**.

```
add trunk-group 3                                     Page 1 of 21
                                     TRUNK GROUP
Group Number: 3                      Group Type: isdn          CDR Reports: y
  Group Name: OUTSIDE CALL              COR: 1          TN: 1          TAC: 110
  Direction: two-way          Outgoing Display? n          Carrier Medium: H.323
Dial Access? n          Busy Threshold: 255  Night Service:
Queue Length: 0
Service Type: tie                      Auth Code? n
                                     Member Assignment Method: manual
```

In Page 2 of the configuration, the **Codeset to Send Display** field is set to **0** as shown. If this field is left at the default value of 6, the Cisco CallManager will not display the calling party name or connected party name sent in the Q.931 SETUP and CONNECT messages, respectively. When set to 0, the Cisco Unified CallManager will display the calling party name on incoming calls from Avaya to Cisco telephones. Similarly, the Cisco Unified CallManager will display the connected party name on Cisco telephones when calls from Cisco telephones to Avaya telephones are answered.

```
add trunk-group 3                                     Page 2 of 21
  Group Type: isdn
TRUNK PARAMETERS
  Codeset to Send Display: 0          Codeset to Send National IEs: 6
          Charge Advice: none
  Supplementary Service Protocol: a          Digit Handling (in/out):
enbloc/enbloc
                                     Digital Loss Group: 18
Incoming Calling Number - Delete:          Insert:          Format:
Disconnect Supervision - In? y  Out? n
Answer Supervision Timeout: 0          Display Incoming Digits? n
```

In Page 3 of the configuration, set the fields **Send Name** and **Send Calling Number** to **y** as shown below. Note that the **Send Connected Number** field should remain set to **n** so that the Avaya S8710 Server will not include a Connected Number Information Element in the Q.931 CONNECT message. The Cisco Unified Call Manager software tested will not display the connected number, if present in the Q.931 CONNECT message.

add trunk-group 3		Page 3 of 21
TRUNK FEATURES		
ACA Assignment? n	Measured: none	
	Internal Alert? n	Maintenance Tests? y
	Data Restriction? n	NCA-TSC Trunk Member:
	Send Name: y	Send Calling Number: y
Used for DCS? n		Send EMU Visitor CPN? n
Suppress # Outpulsing? n	Format: private	
	UII IE Treatment: service-provider	
	Replace Restricted Numbers? n	
	Replace Unavailable Numbers? n	
	Send Connected Number: n	
Network Call Redirection: none	Hold/Unhold Notifications? n	
Send UII IE? y	Modify Tandem Calling Number? n	
Send UCID? n		
Send Codeset 6/7 LAI IE? y		

In Page 5 of the configuration, add the trunk members, as shown below. The keyword **ip** is entered in the **Port** field, and the signaling group number **3** is added in the **Sig Grp** field. The number of rows or trunk members added here will determine the number of simultaneous calls allowed on the IP trunk group.

add trunk-group 3		Page 5 of 21
TRUNK GROUP		
Administered Members (min/max):		1/5
Total Administered Members:		5
GROUP MEMBER ASSIGNMENTS		
Port	Name	Sig Grp
1: ip		3
2: ip		3
3: ip		3
4: ip		3
5: ip		3
6:		

After the trunk-group is added, use the **change signaling-group 3** command to enter the trunk group number **3** in the **Trunk Group for Channel Selection** field.

change signaling-group 3		Page	1 of	5
SIGNALING GROUP				
Group Number: 3	Group Type: h.323			
	Remote Office? n	Max number of NCA TSC: 0		
	SBS? n	Max number of CA TSC: 0		
IP Video? n	Trunk Group for NCA TSC:			
Trunk Group for Channel Selection: 3				
TSC Supplementary Service Protocol: a	Network Call Transfer? n			
T303 Timer(sec): 10				
Near-end Node Name: C-LAN		Far-end Node Name: CallManager5.1		
Near-end Listen Port: 1720		Far-end Listen Port: 1720		

3.8. Configure Route Pattern

Route pattern 9 is created on Avaya Communication Manager to route calls to Cisco Unified CallManager. With the configuration displayed below, Avaya Communication Manager will route calls with destination 55xxx using trunk group 3 configured in the previous sections.

change route-pattern 9			Page	1 of	3
Pattern Number: 10 Pattern Name: To CallManager					
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: 3 0			n user		
2:			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 n n	rest		none		
2: y y y y y n n	rest		none		

Use command **change aar analysis 55** to configure the AAR table to use route pattern 9 for dialed strings starting with 55.

change aar analysis 55			Page	1 of	2
AAR DIGIT ANALYSIS TABLE					
Percent Full: 2					
Dialed	Total	Route	Call	Node	ANI
String	Min Max	Pattern	Type	Num	Reqd
55	5 5	9	aar		n

Use **change public-unknown-numbering 5** command to configure Avaya Communication Manager to pass extensions 50xxx on trunk group 3 to the Cisco CallManager.

change public-unknown-numbering 5					Page 1 of 2
NUMBERING - PUBLIC/UNKNOWN FORMAT					
				Total	
Ext	Ext	Trk	CPN	CPN	
Len	Code	Grp(s)	Prefix	Len	
5	50	3		5	Total Administered: 5
					Maximum Entries: 9999

Use the **save translation** command to save the configuration changes.

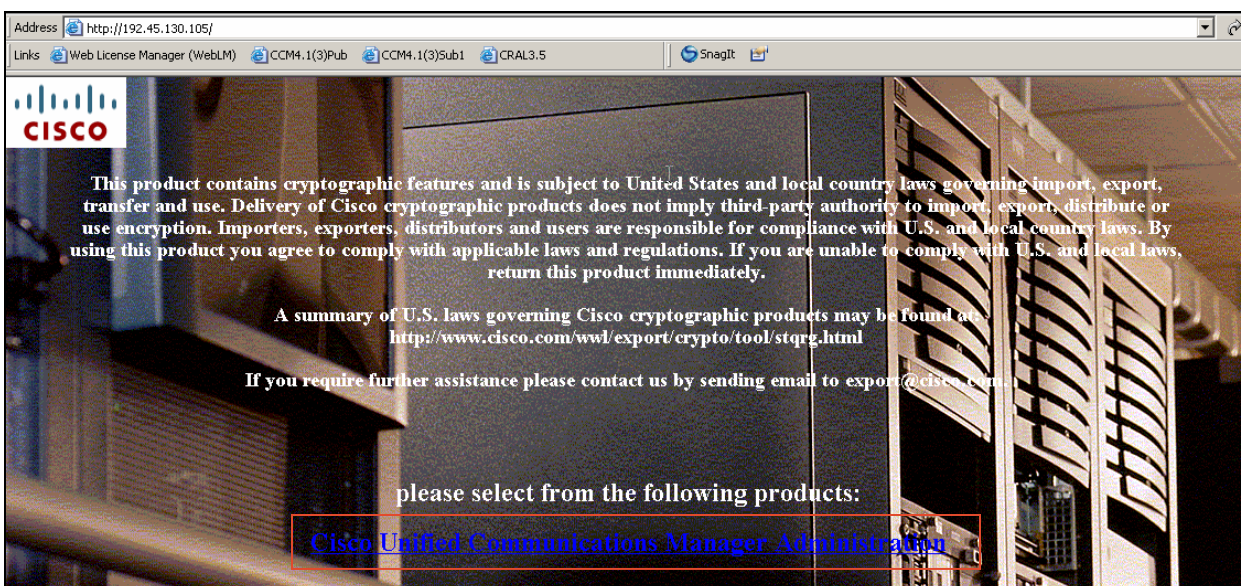
4. Cisco Unified CallManager 5.1.3 Configuration

This section illustrates the relevant Cisco Unified CallManager 5.1.3 configuration. An H.323 gateway will be configured in the Cisco Unified CallManager to connect to the IP address of the C-LAN in the Avaya G650 Media Gateway.

4.1. Add Regions

Regions are used to determine which codec is selected. In this configuration, two regions are used. The default region is used for the Cisco Unified CallManager site and a new region, named **Avaya**, is created for the Avaya Communication Manager site. To save bandwidth on a WAN link, the G.729 codec is used between these two regions. Calls within each region will use the G.711 codec. The following steps show how to create a new region on Cisco Unified CallManager. Launch a web browser and use the IP address of Cisco Unified CallManager as the URL.

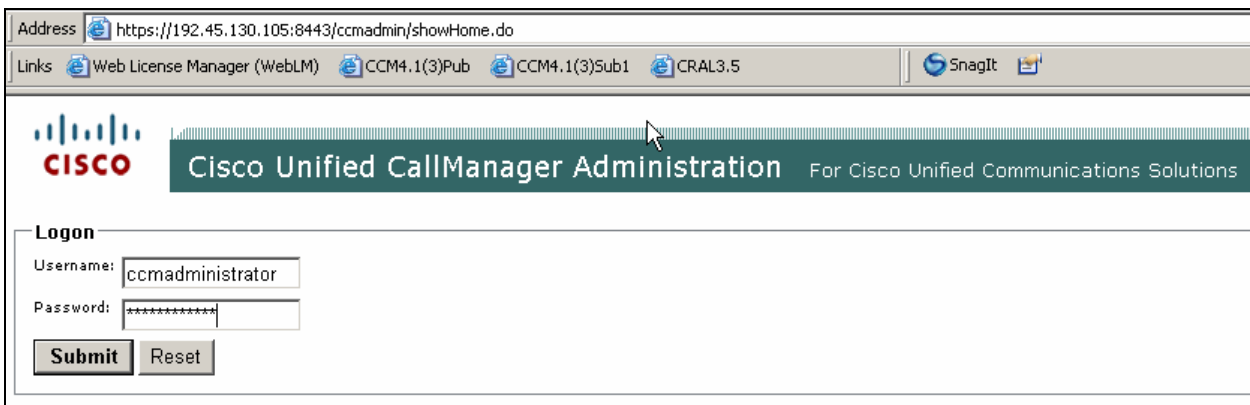
- Click the link **Cisco Unified Communication Manager Administration**



- Click **Yes** at the Security Alert



- Enter **ccadministrator** and password into the related fields and click **Submit** as shown below.



From the Cisco Unified CallManager Administration menu,

- Click **System** → **Region**
- Click **Add New** button to add a new region

The screenshot shows the Cisco Unified CallManager Administration interface. The address bar displays `https://60.1.1.9:8443/ccmadmin/regionFindList.do`. The page title is "Cisco Unified CallManager Administration For Cisco Unified Communications Solutions". A navigation menu includes System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main section is titled "Find and List Regions" and features a search form. The search form has a "Search Options" section with a "Find Regions where" dropdown set to "begins with", an empty text input field, a "Find" button, and a checkbox for "Search Within Results". Below this is a "Search Results" section with the message "No active query. Please enter your search criteria using the options above." and an "Add New" button.

- Type **Avaya** as the region **Name***
- Click **Save**

The screenshot shows the Cisco Unified CallManager Administration interface for "Region Configuration". The address bar displays `https://60.1.1.9:8443/ccmadmin/regionFindList.do`. The page title is "Cisco Unified CallManager Administration For Cisco Unified Communications Solutions". A navigation menu includes System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main section is titled "Region Configuration" and features a "Region Information" section. The "Region Information" section has a "Name*" field with the value "Avaya" entered. Below this is a "Save" button. At the bottom, there is an information icon and the text "*- indicates required item."

After clicking **Save**, the following screen shows that the **Avaya** region is added into database. .

- Under **Modify Relationship to other Regions**, highlight **Default** and use the drop-down window to select **G.729** as **Audio Codec** used between the **Avaya** and **Default** regions.
- Click **Save** and **Reset**.

The screenshot displays the Cisco Unified CallManager Administration web interface. The top navigation bar includes tabs for System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled 'Region Configuration' and contains several sections:

- Status:** Shows 'Add successful' and a message to click the Reset button for changes to take effect.
- Region Information:** A text field for 'Name*' containing 'Avaya'.
- Region Relationships:** A table with columns 'Region' and 'Audio Codec'. The 'Region' column contains the text 'NOTE: Region(s) not displayed', and the 'Audio Codec' column contains 'Use System Default'.
- Modify Relationship to other Regions:** A section with a 'Regions' list box and an 'Audio Codec' dropdown menu. The 'Regions' list box contains 'Avaya', 'Avaya SIP Trunk Region', 'CM4.1', and 'Default', with 'Default' selected. The 'Audio Codec' dropdown menu is set to 'G.729'.

At the bottom of the form are buttons for 'Save', 'Delete', 'Reset', and 'Add New'.

4.2. Add Conference Bridge

A Conference Bridge is a device used by Cisco Unified CallManager to hold Ad Hoc or Meet me conferences. It supports conferences among calling parties using different codecs. Note that Cisco Unified CallManager only supports the G.711 codec for conference calls. In these Application Notes, the calls between the two sites have been configured using G.729. For example, if a Cisco phone has an established call from an Avaya phone (G.729) and tries to conference another Cisco phone (G.711), a conference bridge is needed to provide media resources to support G.729 conference calls. Since the CallManager does not have DSP resources on its hardware, a separate hardware DSP resource is required. In this example, a Cisco 3825 router with a NM-HDV network module is used to provide DSP resources. The following steps describe the configuration of adding a Conference Bridge on a Cisco 3825 router.

From the Cisco Unified CallManager Administration menu,

- Click **Media Resources** → **Conference Bridge**
- Click **Add New**

The screenshot shows the 'Find and List Conference Bridges' page in the Cisco Unified CallManager Administration interface. The page has a header with the title 'Cisco Unified CallManager Administration' and a sub-header 'For Cisco Unified Communications Solutions'. Below the header is a navigation menu with items: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled 'Find and List Conference Bridges' and features a search section. The search section includes a 'Search Options' section with a 'Find Conference Bridges where' dropdown set to 'Name', a 'begins with' dropdown, a 'Find' button, and a 'Search Within Results' checkbox. Below the search section is a 'Search Results' section with the message 'No active query. Please enter your search criteria using the options above.' and an 'Add New' button.

- Use the **Conference Bridge Type** drop down box to select **Cisco Conference Bridge Hardware**
- Type the Cisco C3825 router's interface MAC address in the **MAC Address** field. (Note this router uses its interface FastEthernet 2/0).
- Select **Default** as **Device Pool**
- Click **Save**

The screenshot shows the 'Conference Bridge Configuration' page in the Cisco Unified CallManager Administration interface. The page has a header with the title 'Cisco Unified CallManager Administration' and a sub-header 'For Cisco Unified Communications Solutions'. Below the header is a navigation menu with items: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled 'Conference Bridge Configuration' and features a 'Related Links' section. The configuration section includes a 'Status' section with an 'i' icon and the text 'Status: Ready'. Below the status section is a 'Conference Bridge Information' section with the text 'Conference Bridge : New'. The 'Hardware Conference Bridge Info' section contains several fields: 'Conference Bridge Type*' (set to 'Cisco Conference Bridge Hardware'), 'MAC Address*' (set to '0011936915E9'), 'Description' (set to 'Conference Bridge on Cisco 3825 router'), 'Device Pool*' (set to 'Default'), 'Location*' (set to 'Hub_None'), and 'Special Load Information'. A 'Save' button is located at the bottom of the configuration section. A note at the bottom left states '*- indicates required item.'

Section 5 describes the detailed Conference Bridge configuration on the Cisco 3825 router.

4.3. Add Media Resource Group and List

To use the Conference Bridge, the Cisco CallManager needs a Media Resource Group and a Media Resource List to include the conference bridge created in the previous section. Follow the steps below to add a media resource group and list.

- Open **Media Resources** → **Media Resource Group**
- Click **Add New**

The screenshot shows the 'Find and List Media Resource Groups' page in the Cisco Unified CallManager Administration interface. The page has a header with the title 'Cisco Unified CallManager Administration' and a sub-header 'For Cisco Unified Communications Solutions'. Below the header is a navigation bar with tabs: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled 'Find and List Media Resource Groups' and features a search section with a plus icon, search options (Find Media Resource Group where Name begins with), a Find button, and a checkbox for 'Search Within Results'. Below the search section is a 'Search Results' section with the message 'No active query. Please enter your search criteria using the options above.' and an 'Add New' button.

- Type **MRS1** in the **Name** field.
- Highlight the conference bridge **CFB001936915E9(CFB)** in the **Available Media Resources** block.

The screenshot shows the 'Media Resource Group Configuration' page in the Cisco Unified CallManager Administration interface. The page has a header with the title 'Cisco Unified CallManager Administration' and a sub-header 'For Cisco Unified Communications Solutions'. Below the header is a navigation bar with tabs: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled 'Media Resource Group Configuration' and features a status section (Status: Ready), a media resource group status section (Media Resource Group: MRS1 (used by 44 devices)), a media resource group information section (Name: MRS1, Description: media resource 1), and a devices for this group section. The devices for this group section includes a list of available media resources (CFB_2, MTP0011936915E9, MTP_2, MOH_2 (MOH)[Multicast], CFB0011936915E9 (CFB)) and a selected media resources section. The 'CFB0011936915E9 (CFB)' resource is highlighted in the available list. Below the selected media resources section is a checkbox for 'Use Multicast for MOH Audio (If at least one multicast MOH resource is available)'. At the bottom of the page are buttons for Save, Delete, Copy, Reset, and Add New.

- Click the ▼ to move it to the **Selected Media Resources** area as shown below
- Click **Save** and **Reset**.

Cisco Unified CallManager Administration
For Cisco Unified Communications Solutions

System ▼
Call Routing ▼
Media Resources ▼
Voice Mail ▼
Device ▼
Application ▼
User Management ▼
Bulk Administration ▼
Help ▼

Media Resource Group Configuration

Status
 Status: Ready

Media Resource Group Status
Media Resource Group: MRS1 (used by 44 devices)

Media Resource Group Information
Name*
Description

Devices for this Group

Available Media Resources**

ANN_2
CFB_2
MTP0011936915E9
MTP_2
MOH_2 (MOH)[Multicast]

Selected Media Resources*

CFB0011936915E9 (CFB)

☐ Use Multicast for MOH Audio (If at least one multicast MOH resource is available)

*- indicates required item.

Follow the configuration steps below to add a media resource list.

From the configuration menu,

- Open **Media Resources** → **Media Resource List**
- Click **Add New**
- Type **MSGGroup1** as **Name**
- Highlight the **MRS1** and click the ▼ to move it to the **Selected Media Resource Groups** area

Cisco Unified CallManager Administration For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Media Resource Group List Configuration

Status
 Status: Ready

Media Resource Group List Status
Media Resource Group List: New

Media Resource Group List Information
Name*

Media Resource Groups for this List
Available Media Resource Groups:

MRS1

Selected Media Resource Groups:

- Click **Save** and **Reset**

Media Resource Group List Information	
Name*	MRGroup1
Media Resource Groups for this List	
Available Media Resource Groups	
	▼ ▲
Selected Media Resource Groups	MRS1
	▼ ▲
<input type="button" value="Save"/> <input type="button" value="Delete"/> <input type="button" value="Copy"/> <input type="button" value="Reset"/> <input type="button" value="Add New"/>	

4.4. Add Device Pool

There is a default device pool pre-defined on the Call Manager. This configuration will use this default device pool for all Cisco IP telephones on the CallManager. A new device pool, named Avaya CM, will be created for the Avaya Communication Manager site. The purpose of creating a new device pool is to use different regions to select different codecs. The following configuration shows how to add a new device pool to the Cisco Unified CallManager database

- Click **System → Device Pool**
- Click **Add New**

Cisco Unified CallManager Administration For Cisco Unified Communications Solutions	
System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾	
Find and List Device Pools	
+	
Search Options	
Find device pool where	Device Pool Name ▾ begins with ▾ <input type="text"/> <input type="button" value="Find"/> <input type="checkbox"/> Search Within Results
Search Results	
No active query. Please enter your search criteria using the options above.	
<input type="button" value="Add New"/>	

- Enter **Avaya CM** as **Device Pool Name**
- Select **Avaya** in the **Region*** field
- Select **Standard User** in the **Softkey Template*** field
- Select **MRGroup1** in the **Media Resource Group List** field
- Leave other fields as default as shown below
- Click **Save** and **Reset**

Cisco Unified CallManager Administration
For Cisco Unified Communications Solutions

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Voice Mail
Device
Application
User Management
Bulk Administration

Device Pool Configuration

Status
 Status: Ready

Device Pool: Avaya CM (1 members**)

Device Pool Settings

Device Pool Name*	Avaya CM
Cisco Unified CallManager Group*	Default
Date/Time Group*	CMLocal
Region*	Avaya
Softkey Template*	Standard User
SRST Reference*	Disable
Calling Search Space for Auto-registration	< None >
Media Resource Group List	MRGroup1
Network Hold MOH Audio Source	< None >
User Hold MOH Audio Source	< None >
Network Locale	< None >
User Locale	< None >
Connection Monitor Duration	

Multilevel Precedence and Preemption (MLPP) Information

MLPP Indication*	Default
MLPP Preemption*	Default
MLPP Domain	< None >

Save
Delete
Copy
Reset
Add New

Note: The **Default** device pool is created automatically during the Cisco Unified CallManager installation. Follow the steps below to edit the **Default** device pool properties.

- Click **System** → **Device Pool**
- Click **find**

Address: <https://192.45.130.105/ccmadmin/devicePoolFindList.do>

Links: Web License Manager (WebLM) CCM4.1(3)Pub CCM4.1(3)Sub1 CRAL3.5 SnagIt

Cisco Unified CallManager Administration For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Find and List Device Pools

Search Options

Find device pool where Device Pool Name ▾ begins with ▾ **Find** ☐ Search Within Results

Search Results

No active query. Please enter your search criteria using the options above.

Add New

- Click **Default** under **Search Results**

Cisco Unified CallManager Administration For Cisco Unified Communications Solutions Logged in as

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Find and List Device Pools

Status

4 records found

Search Options

Find device pool where Device Pool Name ▾ begins with ▾ **Find** ☐ Search Within Results
(devicePool.name begins with any)

Search Results






Name	Unified CallManager Group	Region	Date/Time Group
<input type="checkbox"/> Avaya CM	Default	Avaya	CMLocal
<input type="checkbox"/> Avaya SIP Trunk Region	Default	Avaya SIP Trunk Region	CMLocal
<input type="checkbox"/> Default	Default	Default	CMLocal


- Use the drop-down window to select **MRGroup1** as the **Media Resource Group List**
- Leave other fields as default
- Click **Save** and **Reset**

Cisco Unified CallManager Administration For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Device Pool Configuration

Status
 Status: Ready

Device Pool: Default (43 members**)

Device Pool Settings

Device Pool Name*	Default
Cisco Unified CallManager Group*	Default ▾
Date/Time Group*	CMLocal ▾
Region*	Default ▾
Softkey Template*	Standard User ▾
SRST Reference*	Disable ▾
Calling Search Space for Auto-registration	< None > ▾
Media Resource Group List	MRGroup1 ▾
Network Hold MOH Audio Source	< None > ▾
User Hold MOH Audio Source	< None > ▾
Network Locale	United States ▾
User Locale	English, United States ▾
Connection Monitor Duration	

Multilevel Precedence and Preemption (MLPP) Information

MLPP Indication*	Default ▾
MLPP Preemption*	Default ▾
MLPP Domain	< None > ▾

Save Delete Copy Reset Add New

In order for Cisco IP telephones to use the Conference Bridge, **MRGroup1** must be set as the **Media Resource Group List** in telephone administration. The following illustrates the configuration for Extension 55602. Repeat this configuration for all other IP telephones.

- Click **Device** → **Phone**
- Click **Find**

Cisco Unified CallManager Administration For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Find and List Phones

Search Options

Find Phone where Device Name ▾ begins with ▾ **Find** ☐ Search Within Results


Select item or enter search text ▾

Search Results

No active query. Please enter your search criteria using the options above.

[Add New](#)

- Click the phone's MAC address link

	7941G-GE	SEP0019563CBF86	Auto 55602	Default	SCCP
---	----------	---------------------------------	------------	-------------------------	------

Set **Media Resource Group List** to **MRGroup1** as shown below.

Cisco Unified CallManager Administration For Cisco Unified Communications Solutions Logged in as

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

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

Status

Status: Ready

Association Information

[Modify Button Items](#)

1 [Line \[1\] - 55602 \(no partition\)](#)

2 [Line \[2\] - Add a new DN](#)

----- Unassigned Associated Items -----

3 [Add a new SD](#)

4 [Add a new SURL](#)

5 [Add a new BLF SD](#)

6 Privacy

7 None

Phone Type

Product Type: Cisco 7941G-GE

Device Protocol: SCCP

Device Information

Registration Registered with Cisco Unified CallManager cuc

IP Address [60.1.1.150](#)

MAC Address*

Description

Device Pool* [View Details](#)

Phone Button Template*

Softkey Template

Common Phone Profile*

Calling Search Space

AAR Calling Search Space

Media Resource Group List

User Hold MOH Audio Source

Network Hold MOH Audio Source

Location*

4.5. Add an H.323 Gateway

From the Cisco Unified CallManager Administration screen,

- Select **Device** → **Gateway**
- Click **Add New**

The screenshot shows the 'Find and List Gateway' page in the Cisco Unified CallManager Administration interface. The page has a dark green header with the title 'Cisco Unified CallManager Administration' and a subtitle 'For Cisco Unified Communications Solutions'. Below the header is a navigation bar with tabs: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled 'Find and List Gateway' and features a blue plus icon. Under the 'Status' section, it says '0 records found'. The 'Search Options' section includes a search bar with 'Name' selected, a 'begins with' dropdown, a 'Hide endpoints' checkbox, and a 'Find' button. Below the search bar, it says '(device.name begins with any)'. The 'Search Results' section states 'No active query. Please enter your search criteria using the options above.' and includes an 'Add New' button and a 'Rows per Page' dropdown set to '50'.

From the Gateway Type drop-down list box,

- Choose **H.323 Gateway** and click **Next**.

The screenshot shows the 'Add a new Gateway' page in the Cisco Unified CallManager Administration interface. The page has a dark green header with the title 'Cisco Unified CallManager Administration' and a subtitle 'For Cisco Unified Communications Solutions'. Below the header is a navigation bar with tabs: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled 'Add a new Gateway' and features a green right-pointing arrow. Under the 'Select the type of gateway you would like to add:' section, the 'Gateway Type*' dropdown is set to 'H.323 Gateway'. Below this is a 'Next' button. At the bottom, there is an information icon and a note: '*- indicates required item.'

After clicking **Next**, enter the gateway configuration information as shown below. The **Device Name** corresponds to the C-LAN IP address used in the signaling group definition on the Avaya S8710 Server. Select **Avaya CM** for **Device Pool** and **MRGroup1** for **Media Resource Group List**. Note that **Media Termination Point Required** is only needed if the H.323 clients and H323 devices do not support the H.245 Empty Capabilities Set message. **Retry Video Call as Audio** applies only to video endpoints. In this configuration, there is no need to check this box. **Wait for Far End H.245 Terminal Capability Set** applies only to H.323 devices. By default, the system checks this box to specify that Cisco Call Manager needs to receive the far-end H.245 Terminal Capability Set before it sends its H.245 Terminal Capability Set. Leave **Signaling Port** at the default of 1720.

Cisco Unified CallManager Administration
For Cisco Unified Communications Solutions

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Voice Mail
Device
Application
User Management
Bulk Administration
Help

Gateway Configuration

Status
i Status: Ready

Device Information

Product	H.323 Gateway
Device Protocol	H.225
Registration	Unknown
IP Address	192.168.1.10
Device Name*	192.168.1.10
Description	C-LAN
Device Pool*	Avaya CM
Call Classification*	Use System Default
Media Resource Group List	MRGroup1
Packet Capture Mode*	None
Packet Capture Duration	0
Location*	Hub_None
AAR Group	< None >
Tunneled Protocol*	None
Signaling Port*	1720

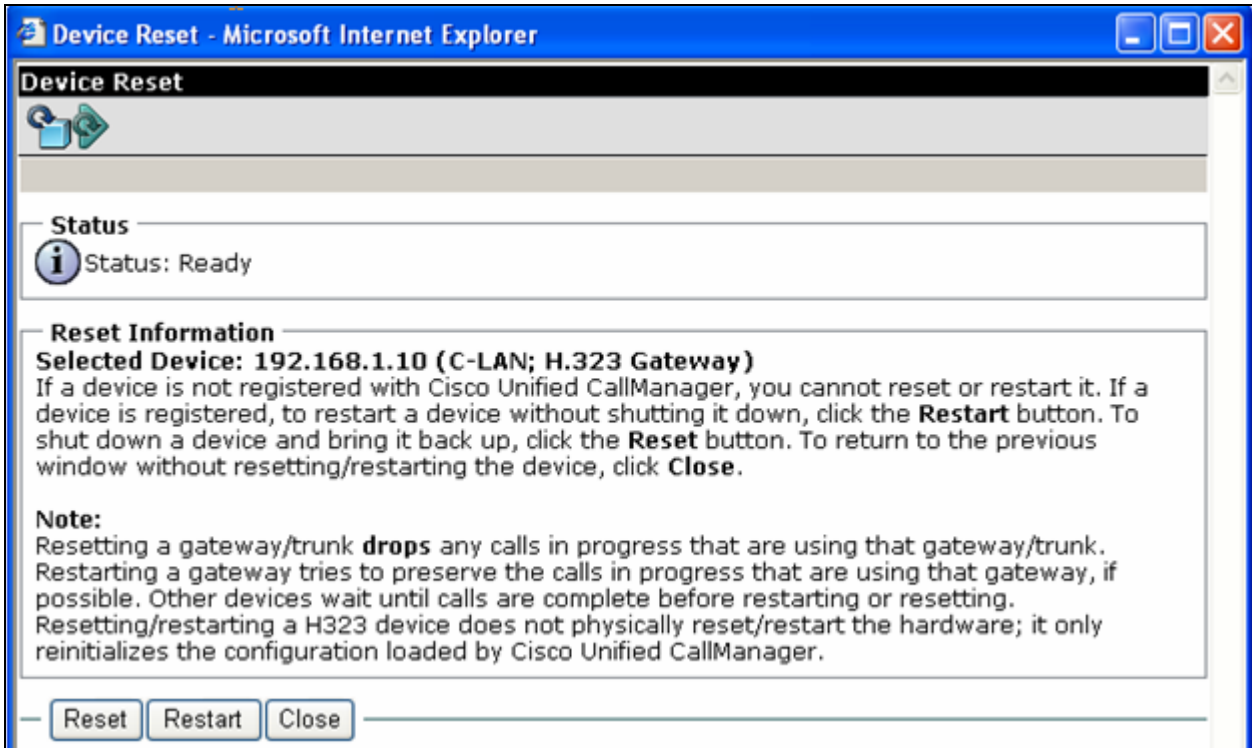
☐ Media Termination Point Required
☐ Retry Video Call As Audio
☒ Wait for Far End H.245 Terminal Capability Set
☐ Path Replacement Support
☐ Transmit UTF-8 for Calling Party Name
☐ SRTP Allowed - When this flag is checked, IPsec needs to be configured in the network to provide end to end security information.

Below is the continuation of the previous screen.

- Check the boxes as shown below and leave other settings at their default.
- Click **Save** to save configuration
- Click **Reset** to reset gateway

Multilevel Precedence and Preemption (MLPP) Information	
MLPP Domain	< None >
MLPP Indication	Not available on this device
MLPP Preemption	Not available on this device
Call Routing Information - Inbound Calls	
Significant Digits*	All
Calling Search Space	< None >
AAR Calling Search Space	< None >
Prefix DN	
<input checked="" type="checkbox"/> Redirecting Number IE Delivery - Inbound	
<input type="checkbox"/> Enable Inbound FastStart	
Call Routing Information - Outbound Calls	
Calling Party Selection*	Originator
Calling Party Presentation*	Default
Called party IE number type unknown*	Cisco CallManager
Calling party IE number type unknown*	Cisco CallManager
Called Numbering Plan*	Cisco CallManager
Calling Numbering Plan*	Cisco CallManager
Caller ID DN	
<input checked="" type="checkbox"/> Display IE Delivery	
<input checked="" type="checkbox"/> Redirecting Number IE Delivery - Outbound	
<input type="checkbox"/> Enable Outbound FastStart	
Codec For Outbound FastStart	G711 u-law 64K
<input type="button" value="Save"/> <input type="button" value="Delete"/> <input type="button" value="Copy"/> <input type="button" value="Reset"/> <input type="button" value="Add New"/>	

- Click **Reset** again at the following pop-up screen

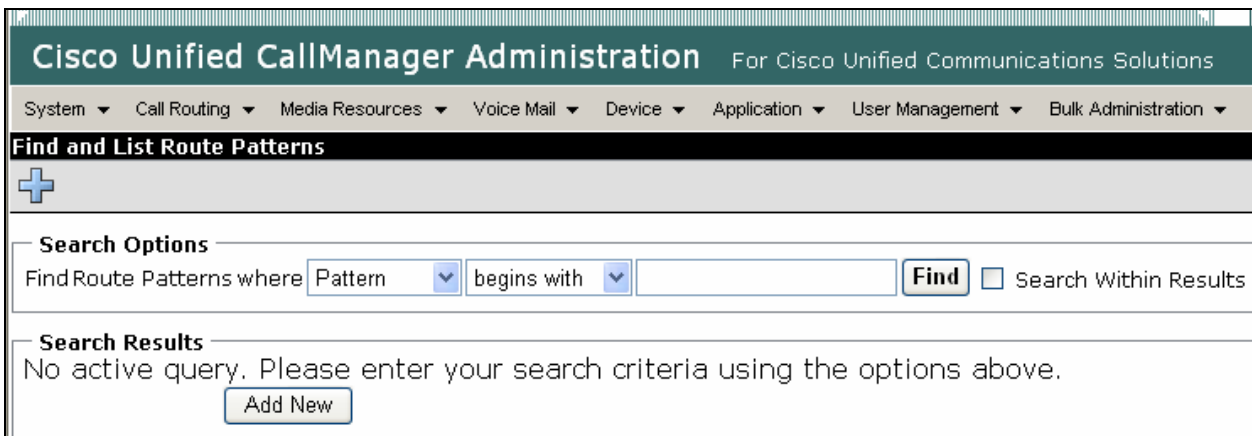


4.6. Configure Route-pattern on the Cisco Unified CallManager

The routing pattern is configured such that calls from the Cisco IP phones to extension range 50xxx are directed to the gateway 192.168.1.10, the IP address of the C-LAN in the Avaya G650 Media Gateway. The next screen shows the configuration.

From the Cisco Unified CallManager Administration screen,

- Click **Call Routing** → **Route/Hunt** → **Route Pattern** as shown below
- Click **Add New**








- Enter **50XXX** in the **Route Pattern** field as shown below
- From the **Gateway/Route List** drop down box, select gateway **192.168.1.10**
- Click **Route this pattern** from **Route Option**
- Leave other settings as shown

Cisco Unified CallManager Administration For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Route Pattern Configuration

Status
 Status: Ready

Pattern Definition

Route Pattern*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence*

Gateway/Route List* (Edit) Find

Route Option
☒ Route this pattern
☐ Block this pattern

Call Classification*


☐ Allow Device Override ☐ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority

☐ Require Forced Authorization Code

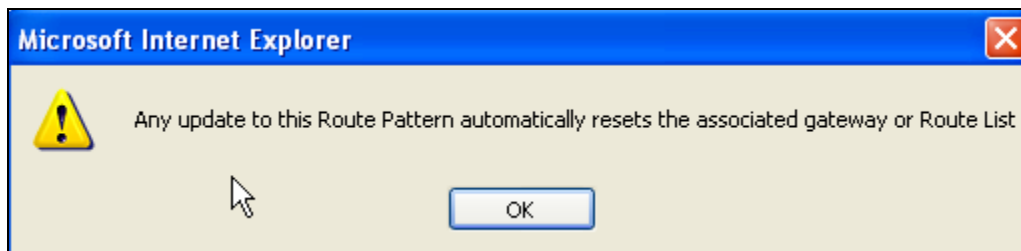
Authorization Level*

☐ Require Client Matter Code

This screen continues on next page.

Calling Party Transformations		
<input type="checkbox"/> Use Calling Party's External Phone Number Mask		
Calling Party Transform Mask	<input type="text"/>	
Prefix Digits (Outgoing Calls)	<input type="text"/>	
Calling Line ID Presentation*	Default	
Calling Name Presentation*	Default	
Connected Party Transformations		
Connected Line ID Presentation*	Default	
Connected Name Presentation*	Default	
Called Party Transformations		
Discard Digits	< None >	
Called Party Transform Mask	<input type="text"/>	
Prefix Digits (Outgoing Calls)	<input type="text"/>	
ISDN Network-Specific Facilities Information Element		
Network Service Protocol	-- Not Selected --	
Carrier Identification Code	<input type="text"/>	
Network Service	Service Parameter Name	Service Parameter Value
-- Not Selected --	< Not Exist >	<input type="text"/>
<input type="button" value="Save"/> <input type="button" value="Delete"/> <input type="button" value="Copy"/> <input type="button" value="Add New"/>		
 *- indicates required item.		

- Click **Save**
- Click **OK** on the subsequent pop-up



5. Configure Conference Bridge on the Cisco 3825 Router

This section only presents the Conference Bridge related configuration on the Cisco 3825 router.

```
voice-card 1
no dspfarm
dsp services dspfarm           --- enable DSP farm services for the voice card

voice service voip             --- enable voip service on router
allow-connections h323 to h323
redirect ip2ip
h323

interface FastEthernet2/0
ip address 14.1.1.1 255.255.255.0
ip pim sparse-dense-mode
duplex auto
speed auto
h323-gateway voip interface
h323-gateway voip bind srcaddr 14.1.1.1

scp local FastEthernet2/0 ! --- select the interface that SCCP applications use to register with Cisco
Unified CallManager

scp                            --- enable the Skinny Client Control Protocol (SCCP)
                                protocol and bring it up administratively

scp ccm 192.45.130.105 priority 1 --- add Cisco Unified CallManager as SCCP Server with
                                priority 1
scp codec g711ulaw mask        --- Add codecs supported by this conference bridge
scp codec g729r8 mask
scp codec g729ar8 mask
scp codec g729abr8 mask

!
dspfarm transcoder maximum sessions 24
dspfarm confbridge maximum sessions 6 --- set Max session 6 for conference bridge
dspfarm codec g729 vad disable       --- disable vad for codec g729
dspfarm
!
gateway
```


6. Verification Steps

The following steps can be used to verify the configuration described in these Application Notes.

- Make a phone call from the Avaya 9630 IP Telephone (50008) to the Cisco 7970 Telephone (55603), and verify the voice quality is good and the IP trunk is used to carry this call. From the Avaya SAT, use the command **status station 50008** to display the call signaling and audio information.

```
status station 50008                                     Page 1 of 7
                                     GENERAL STATUS
Administered Type: 4620      Service State: in-service/off-hook
Connected Type: 9640      TCP Signal Status: connected
Extension: 50008
Port: S00026      Parameter Download: complete
Call Parked? no      SAC Activated? no
Ring Cut Off Act? no
Active Coverage Option: 1

EC500 Status: N/A      Off-PBX Service State: N/A
Message Waiting:
Connected Ports: T00063

Limit Incoming Calls? no

User Cntrl Restr: none
Group Cntrl Restr: none

                                     HOSPITALITY STATUS
Awaken at:
User DND: not activated
Group DND: not activated
Room Status: non-guest room

status station 50008                                     Page 3 of 7
                                     CALL CONTROL SIGNALING
Port: S00026      Switch-End IP Signaling Loc: 01A0217 H.245 Port:
IP Address      Port Node Name      Rgn
Switch-End: 192.168. 1. 10      61441 c-lan      1
Set End: 192.168. 1.111      1720      1
H.245 Near:
H.245 Set:

status station 50008                                     Page 4 of 7
                                     AUDIO CHANNEL Port: S00026
G.729A+B      Switch-End Audio Location:
IP Address      Port Node Name      Rgn
Other-End: 60. 1. 1.151      21898      3
Set-End: 192.168. 1.111      2868      1
Audio Connection Type: ip-direct

status station 50008                                     Page 6 of 7
                                     SRC PORT TO DEST PORT TALKPATH
src port: S00026
S00026:TX:192.168.1.111:2868/g729ab/20ms
T00063:RX:60.1.1.151:21898/g729b/20ms
```

- When the call is up, use the command **status trunk 3** to verify that trunk group 3 is used to carry this call. The display below shows that trunk group 3, channel 4 is in service/active. The signaling path is between C-LAN and CallManager and the audio path is between Avaya IP Telephone (x50008) and Cisco IP telephone (x55603). The codec used is G.729B.

```

status trunk 3

                                TRUNK GROUP STATUS

Member   Port      Service State      Mtce Connected Ports
                                Busy

0003/001 T00060    in-service/idle    no
0003/002 T00061    in-service/idle    no
0003/003 T00062    in-service/idle    no
0003/004 T00063    in-service/active  no    S00026
0003/005 T00064    in-service/idle    no
0003/006 T00065    in-service/idle    no

status trunk 3/4                                     Page 1 of 2

                                TRUNK STATUS

Trunk Group/Member: 0009/004      Service State: in-service/active
Port: T00063                      Maintenance Busy? no
Signaling Group ID:              CA-TSC state: not allowed

IGAR Connection? no

Connected Ports: S00026

                                Port      Near-end IP Addr : Port      Far-end IP Addr : Port
                                Signaling: 01A0217 192.168. 1. 10 : 13874 192. 45.130.105 : 1720
                                H.245: 01A0217 192.168. 1. 10 : 13875 192. 45.130.105 : 59602
G.729B Audio: 192.168. 1.111 : 2868 60. 1. 1.151: 21898
                                Video:
Video Codec:
H.245 Tunneled in Q.931? no      Authentication Type: None
Audio Connection Type: ip-direct

```

- Make a phone call from the Cisco 7941G (55602) IP phone to the Avaya digital phone (50002), and verify the voice quality is good. Transfer the call to the Avaya 4621SW IP Telephone (50000) and verify that the transfer is successful.

- Make a phone call from the Avaya 4621SW IP telephone (50000) to the Cisco 7941G telephone (55602). While the call is up, conference the Cisco 7970 telephone (55603) from the Cisco7941G telephone (55602) and verify that all three parties are in conference. Use command **show sccp connections** to display the Cisco 3825 router and verify that all three IP telephones using the conference bridge with G.729b codec.

C3825#show sccp connections							
sess_id	conn_id	stype	mode	codec	ripaddr	rport	sport
16778332	16778182	conf	sendrecv	g729b	60.1.1.151	22708	20238
16778332	16778184	conf	sendrecv	g729b	60.1.1.150	22578	24380
16778332	16778186	conf	sendrecv	g729b	192.168.1.110	2144	24616
Total number of active session(s) 1, and connection(s) 3							

- Display verifications:
 - For calls from an Avaya telephone to a Cisco IP telephone, the Cisco IP telephone will display the name and number of the Avaya caller, provided the Avaya server is provisioned to send the calling party name and number. When the Cisco telephone is answered, the Avaya telephone will display the number and name of the Cisco telephone.
 - For calls from a Cisco telephone to an Avaya telephone, the Avaya telephone will display the calling party name and number, when sent by the Cisco CallManager. When the Avaya telephone is answered, the Cisco telephone will display the name and the dialed number of the connected party sent by Avaya Communication Manager.

7. Conclusion

As illustrated in these Application Notes, the Avaya S8710 Server and Avaya G650 Media Gateway can interoperate with the Cisco Unified CallManager 5.1.3 using an H.323 IP trunk. A Cisco 3825 router can be configured as a conference bridge device to support conference calls among the Avaya and Cisco telephones. IP-IP Direct Audio calling (shuffling) is supported between Avaya IP telephones and Cisco IP telephones and calling party name and number can be displayed for calls in both directions.

8. Additional References

The following documents are available at <http://support.avaya.com/>

[1] *Application Notes for Configuring H.323 Signaling and IP Trunks between Avaya Communication Manager 4.0 and Cisco Unified CallManager 5.1 - Issue 1.0*

[2] *Administrator Guide for Avaya Communication Manager, Document ID: 03-300509, Issue 3.1, February 2007*

The following Cisco document is available at
http://cisco.com/en/US/products/sw/voicesw/ps556/products_administration_guide_chapter09186a00808bac81.html

[3] *Cisco Unified CallManager Administration Guide, Release 5.1(3)*

9. Change History

Issue	Date	Reason
1.1		Update to incorporate newer version of Avaya Communication Manager and Cisco Unified CallManager and to incorporate IP-IP Direct Audio calling (shuffling).
1.0	4/8/2005	Initial issue.

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