Avaya Solution & Interoperability Test Lab

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.

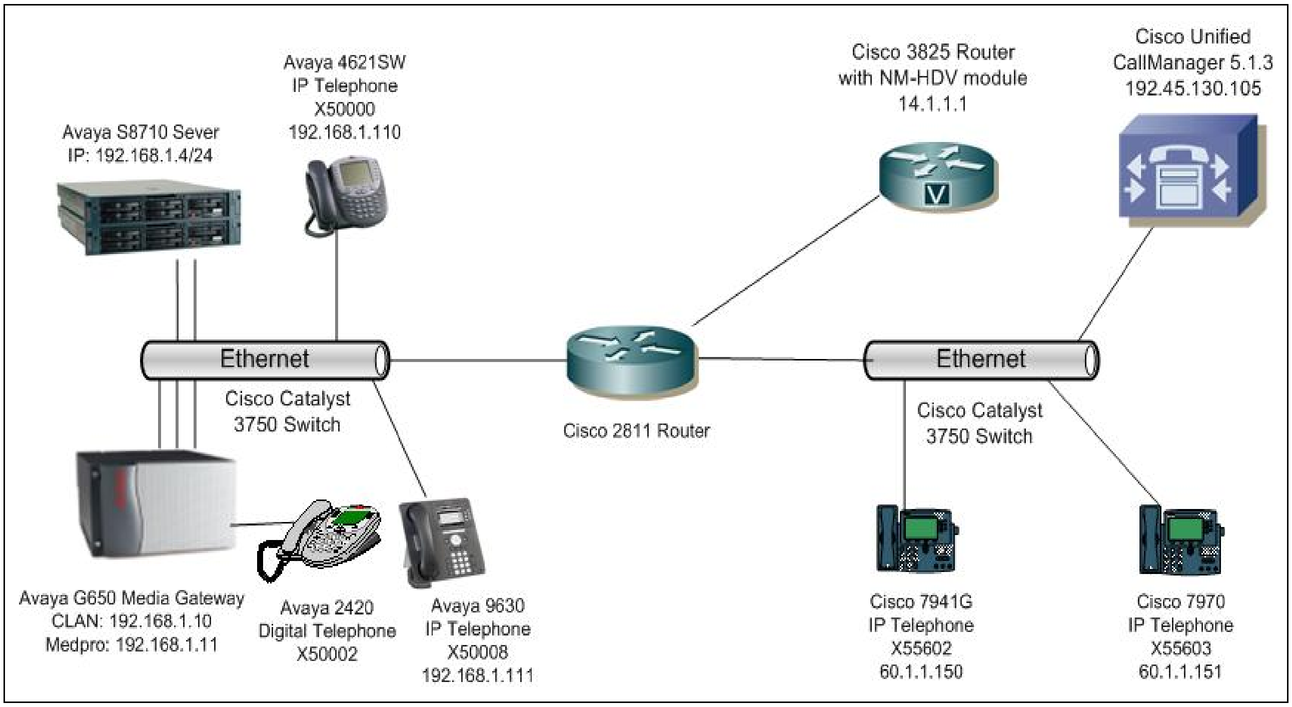


Figure 1: Avaya-Cisco H.323 Interoperability Configuration

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 • 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
                                  IP NODE NAMES
    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
                                  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                                             Page   1 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
                               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

                 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                Intra-region IP-IP Direct Audio: yes
      Codec Set: 3              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 16384                     IP Audio Hairpinning? y
   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
```

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                         Dyn
 rgn rgn  set   WAN  Units    Total Norm  Prio Shr Intervening-regions  CAC IGA
 3   1    3     y    NoLimit                                                n
 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

                                                              Emergency
                                     Subnet                   Location
 From IP Address  (To IP Address   or Mask)  Region    VLAN   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
                                             UUI 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 UUI 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
GROUP MEMBER ASSIGNMENTS                 Total Administered Members:   5

       Port            Name        Night           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
                                                     Total Administered: 5
 5  50              3                   5         Maximum Entries: 9999
```

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

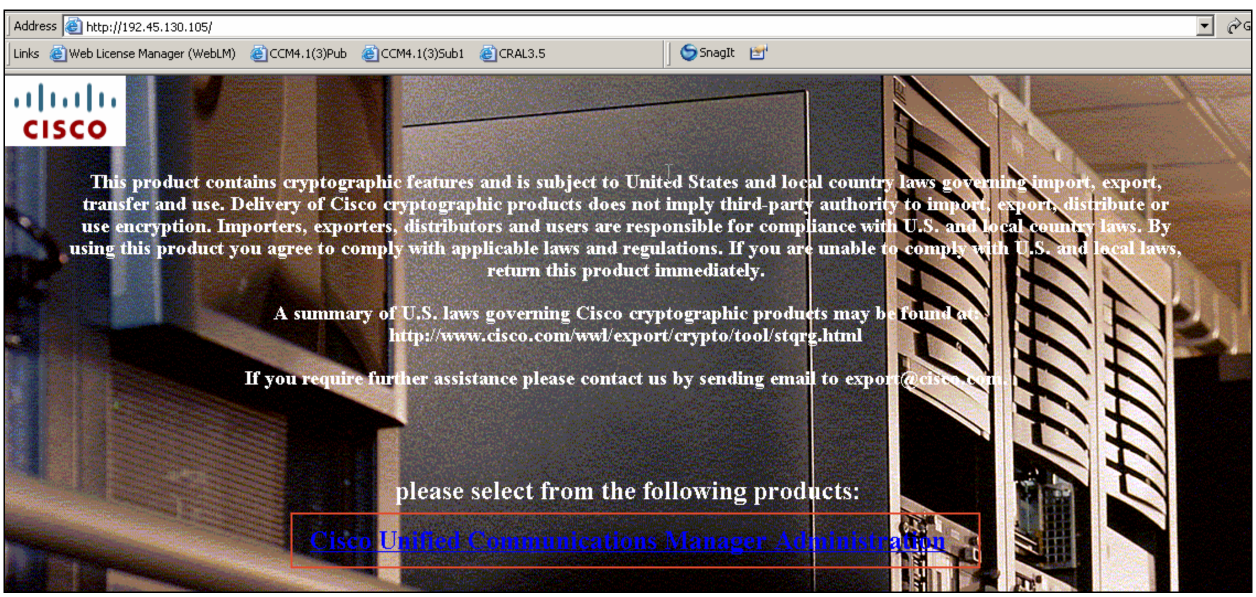
4. Cisco Unified CallManager 5.1.3 Configuration

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.

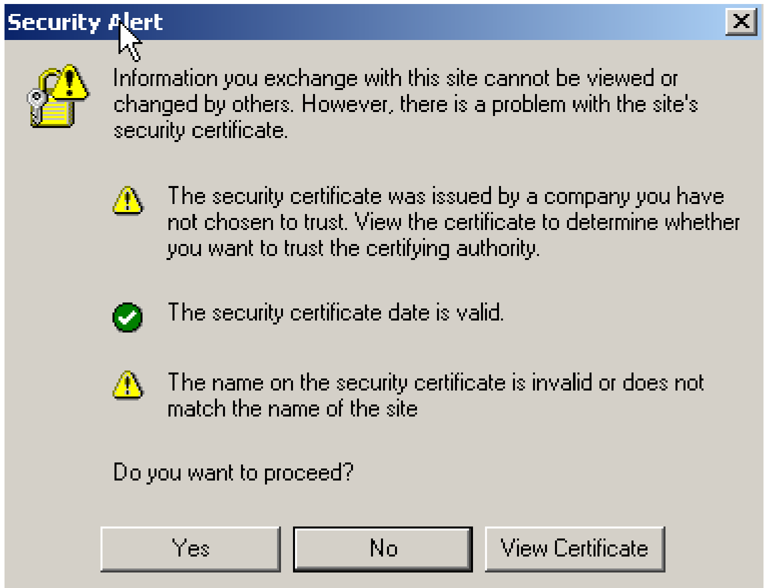
4.1. Add Regions

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.

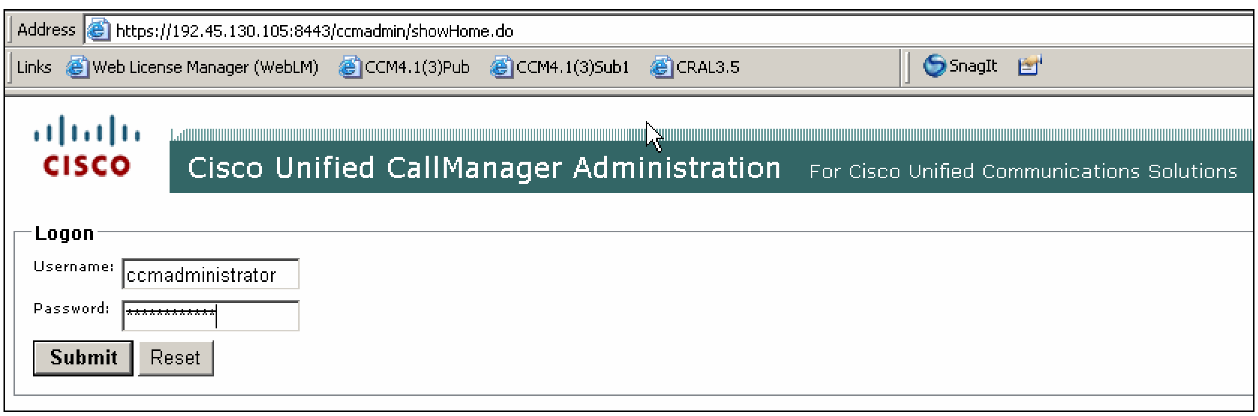
- Click the link **Cisco Unified Communication Manager Administration**



- Click **Yes** at the Security Altert

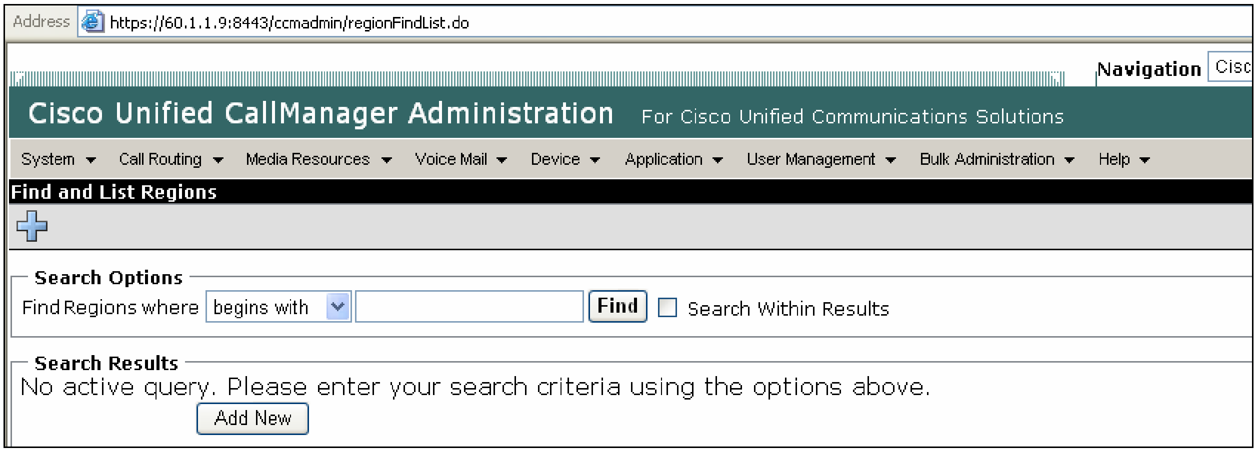


- Enter **ccmadministrator** and password into the related fields and click **Submit** as shown below.

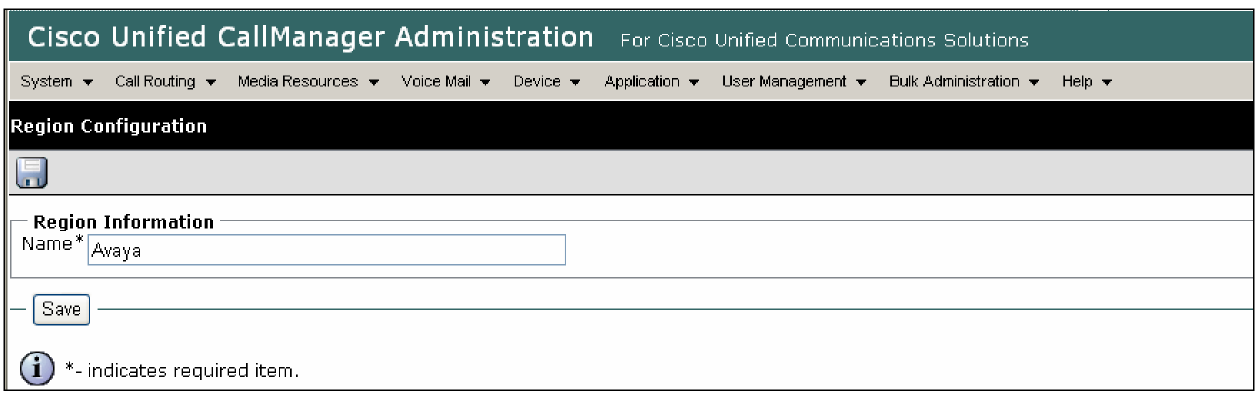


From the Cisco Unified CallManager Administration menu,

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

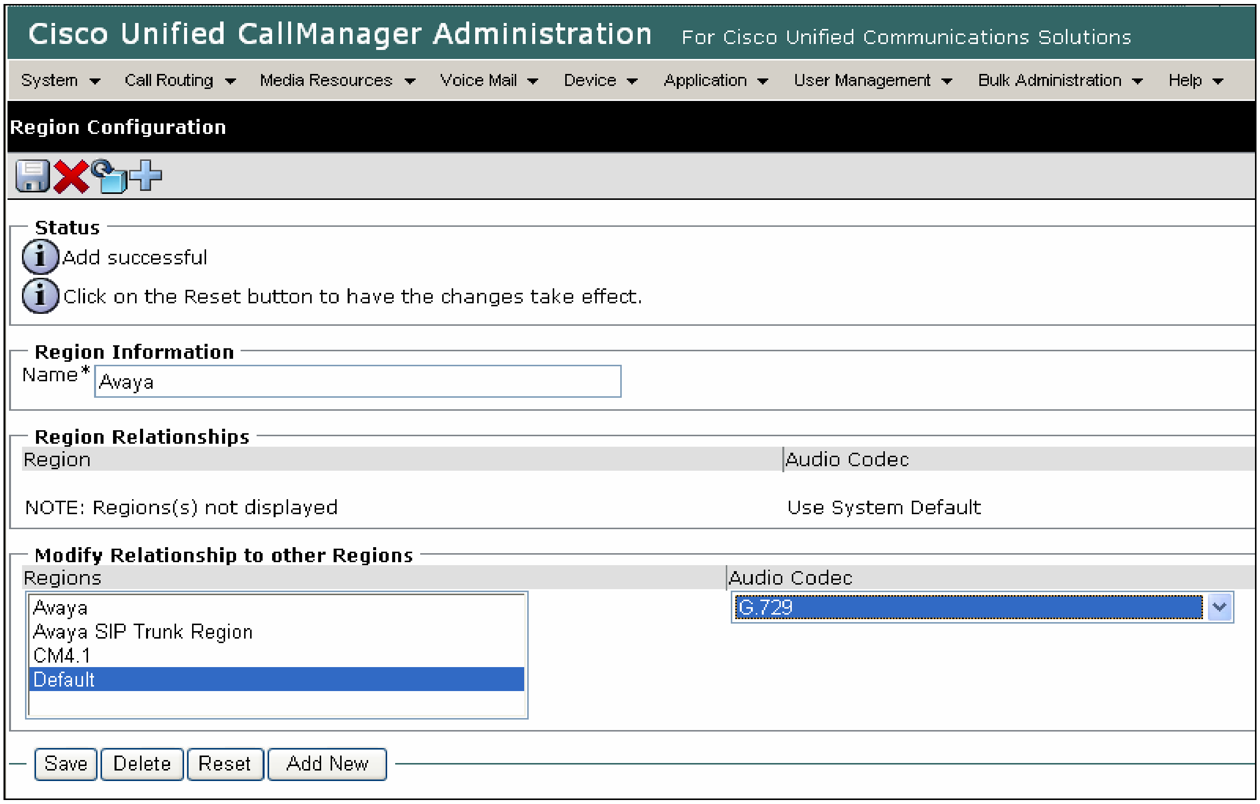


- Type **Avaya** as the region **Name***
- Click **Save**



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**.

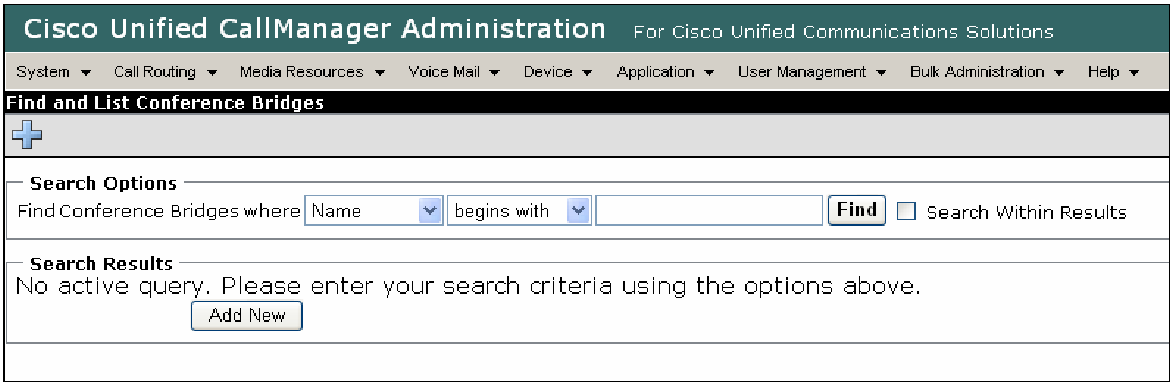


4.2. Add Conference Bridge

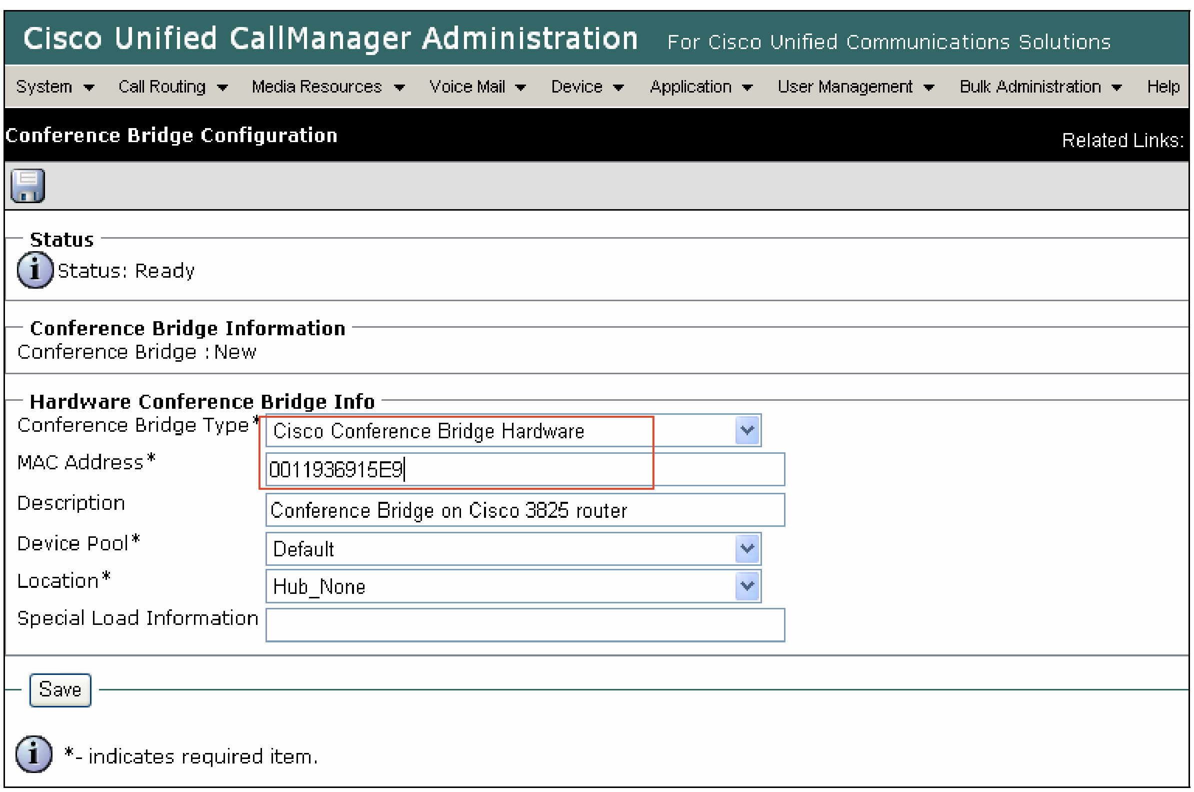
A Conference Bridge is a device used by Cisco Unified CallManager to hold Ad Hoc or Meet me conferences. It supports conferences among calling parties using different codecs. Note that Cisco Unified CallManager only supports the G.711 codec for conference calls. In these Application Notes, the calls 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**



- 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**

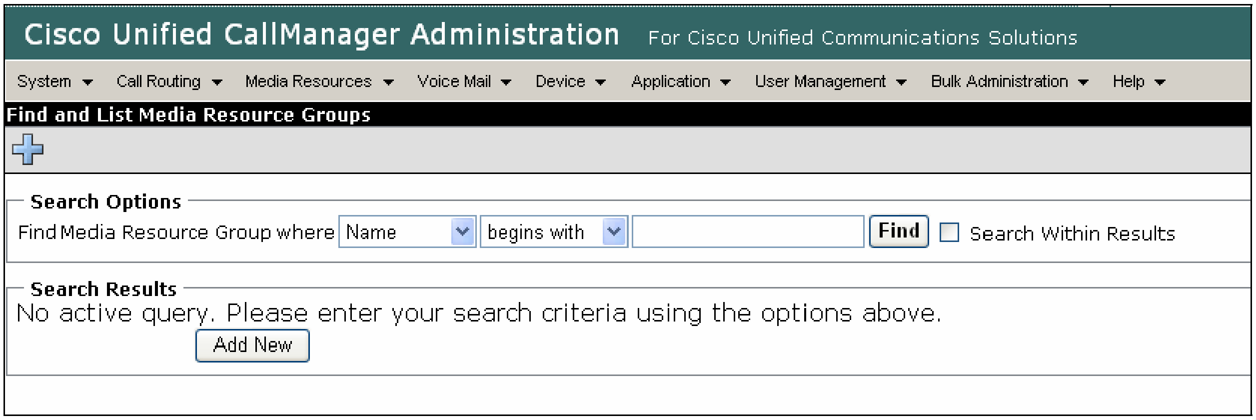


Section 5 describes the detailed Conference Bridge configuration on the Cisco 3825 router.

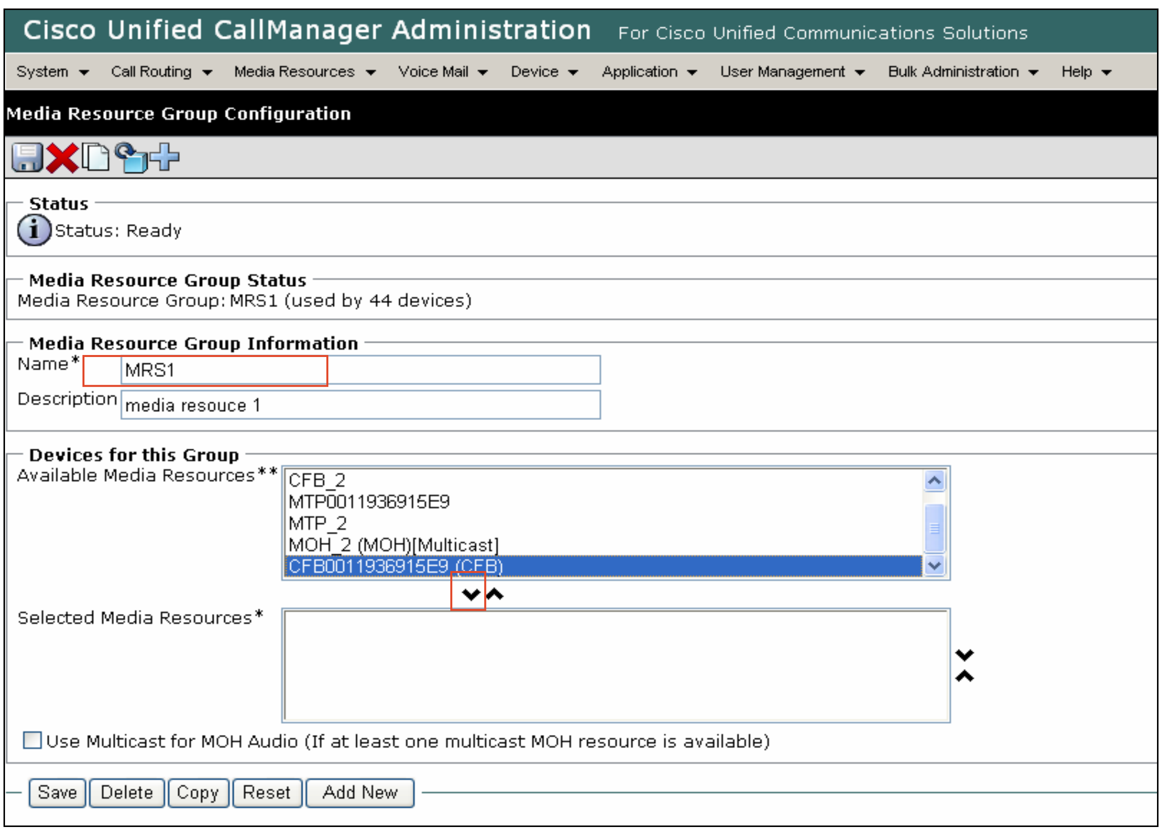
4.3. Add Media Resource Group and List

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.

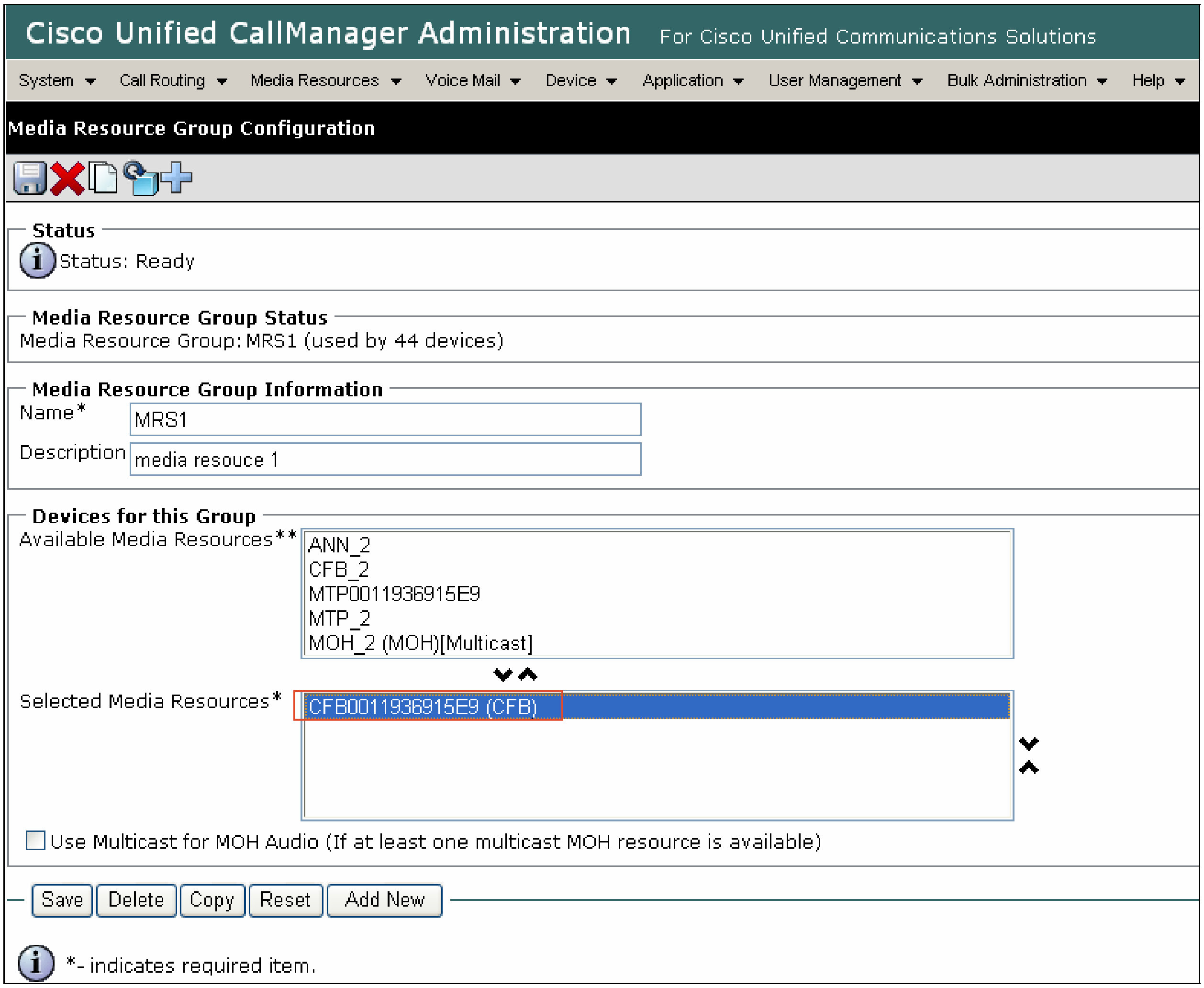
- Open **Media Resources → Media Resource Group**
- Click **Add New**



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

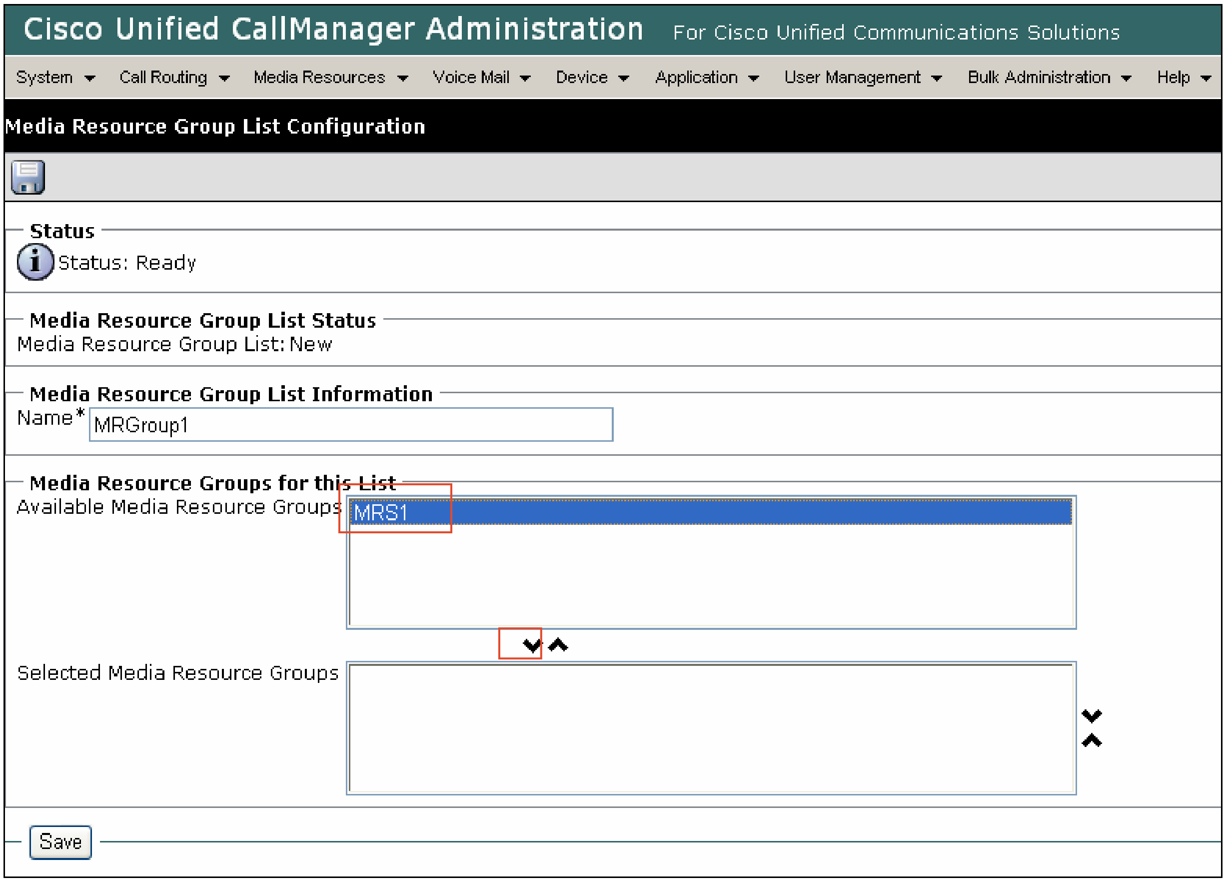


- Click the to move it to the **Selected Media Resources** area as shown below
- Click **Save** and **Reset**.

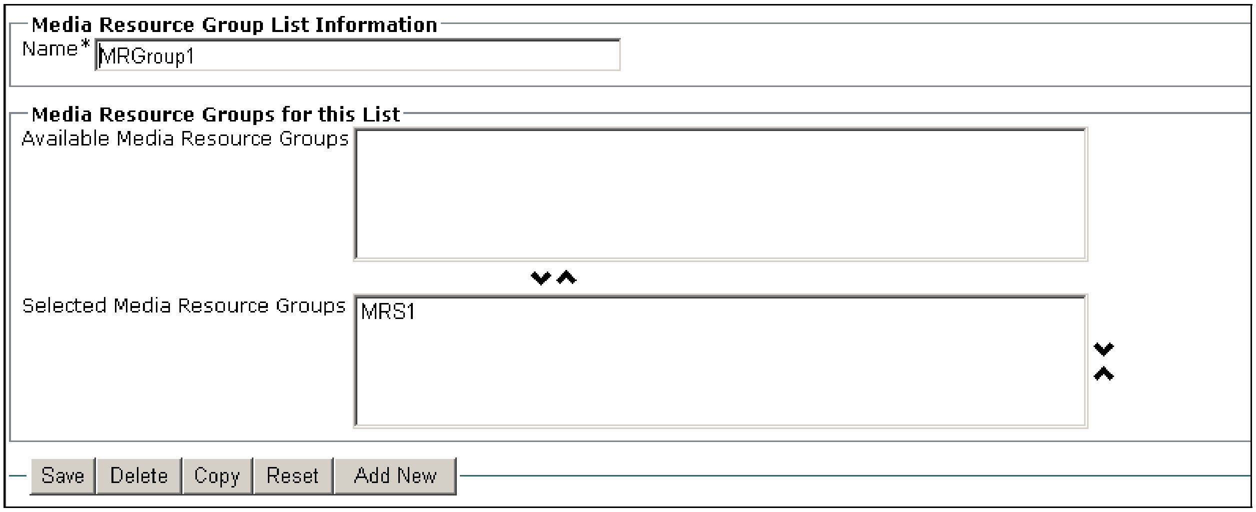


Follow the configuration steps below to add a media resource list.
From the configuration menu,

- Open **Media Resources → Media Resource List**
- Click **Add New**
- Type **MSGroup1** as **Name**
- Highlight the **MRS1** and click the ⌄ to move it to the **Selected Media Resource Groups** area



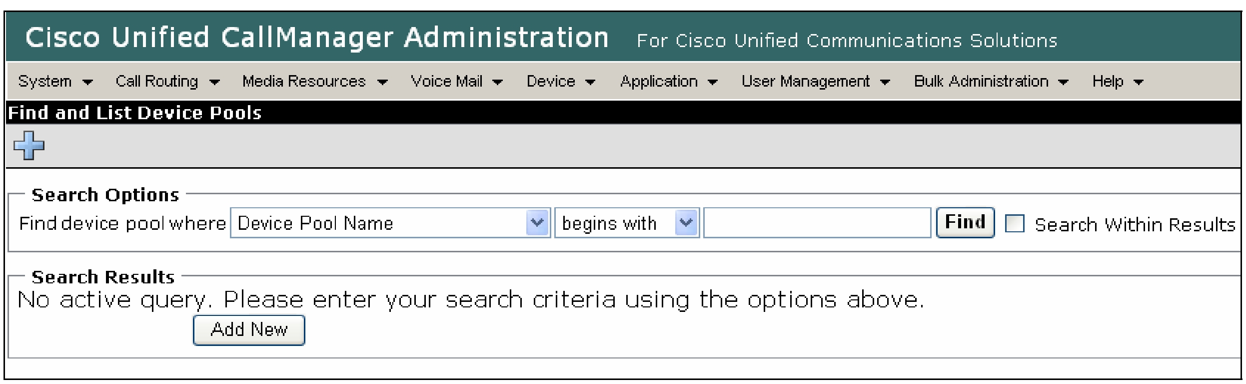
- Click **Save** and **Reset**



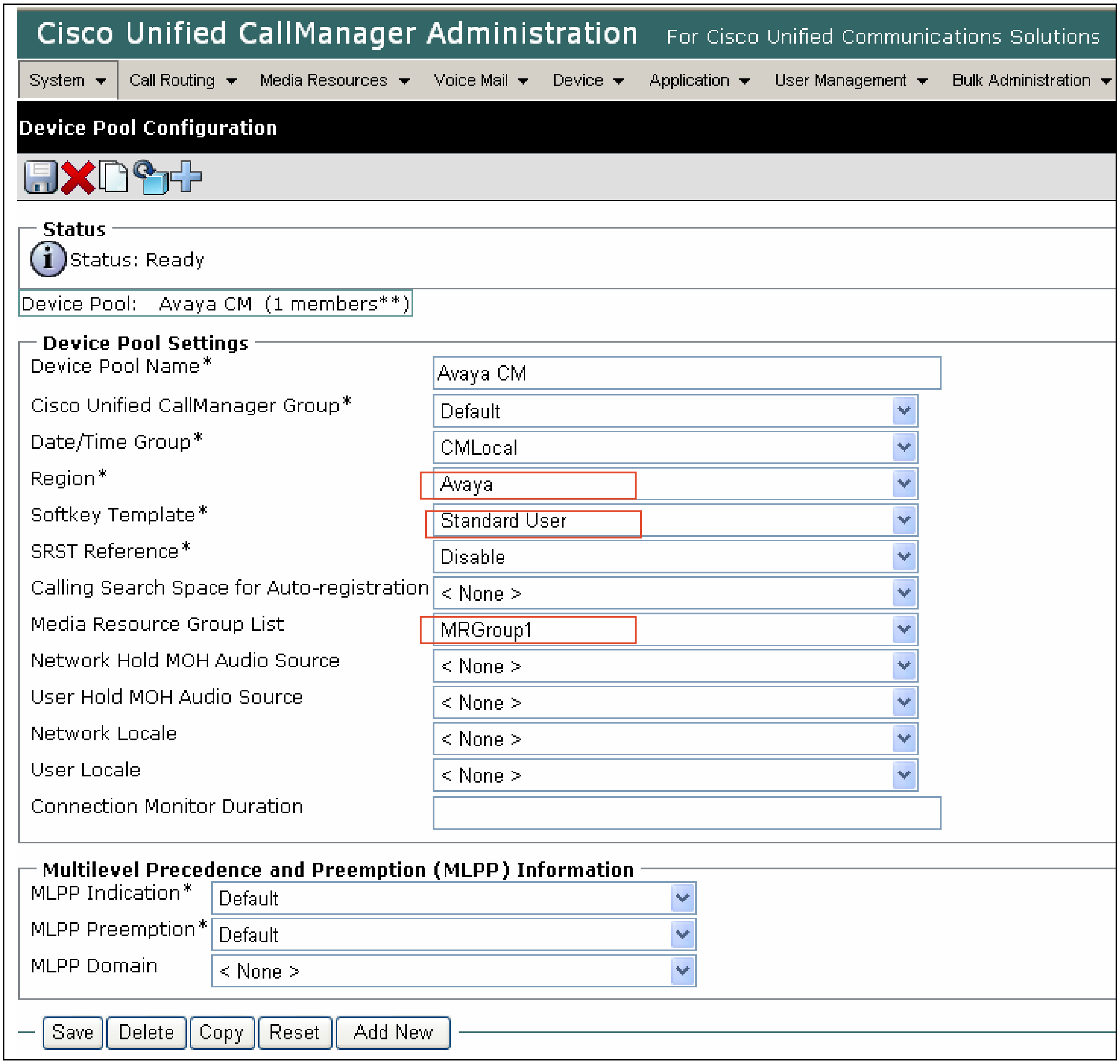
4.4. Add Device Pool

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**

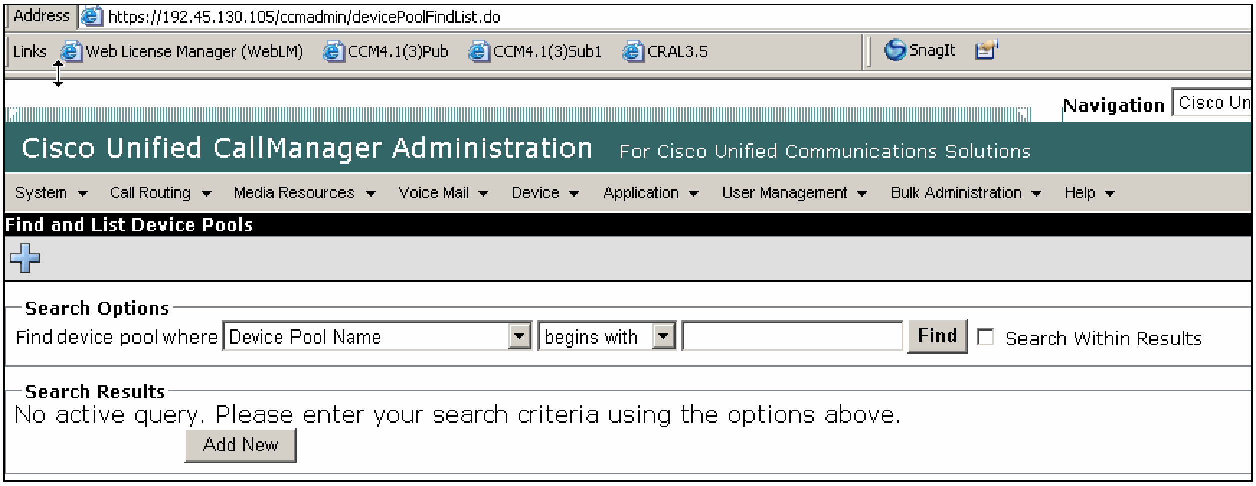


- Enter **Avaya CM** as **Device Pool Name**
- Select **Avaya** in the **Region*** field
- Select **Standard User** in the **Softkey Template*** field
- Select **MRGroup1** in the **Media Resource Group List** field
- Leave other fields as default as shown below
- Click **Save** and **Reset**

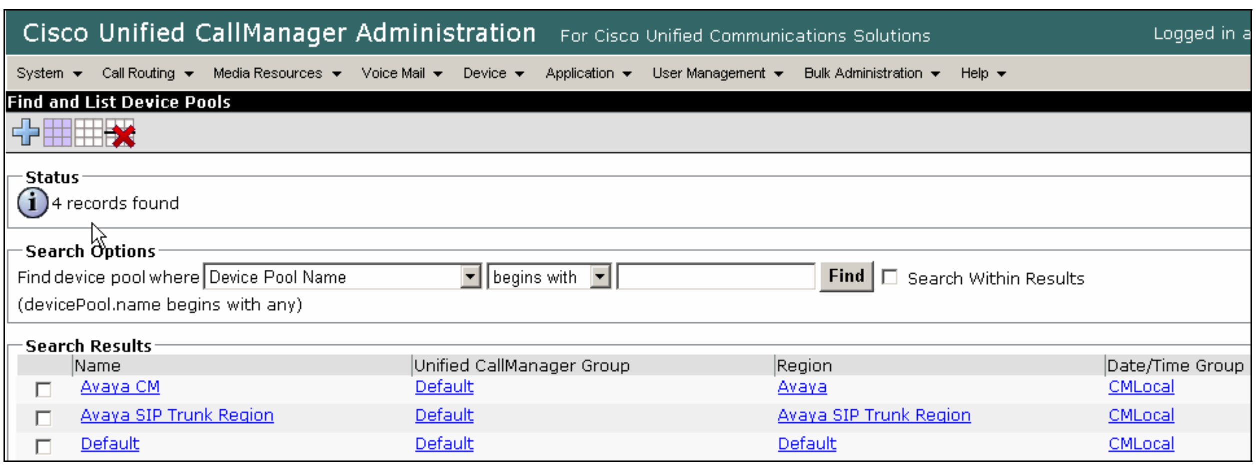


Note: The **Default** device pool is created automatically during the Cisco Unified CallManager installation. Follow the steps below to edit the **Default** device pool properties.

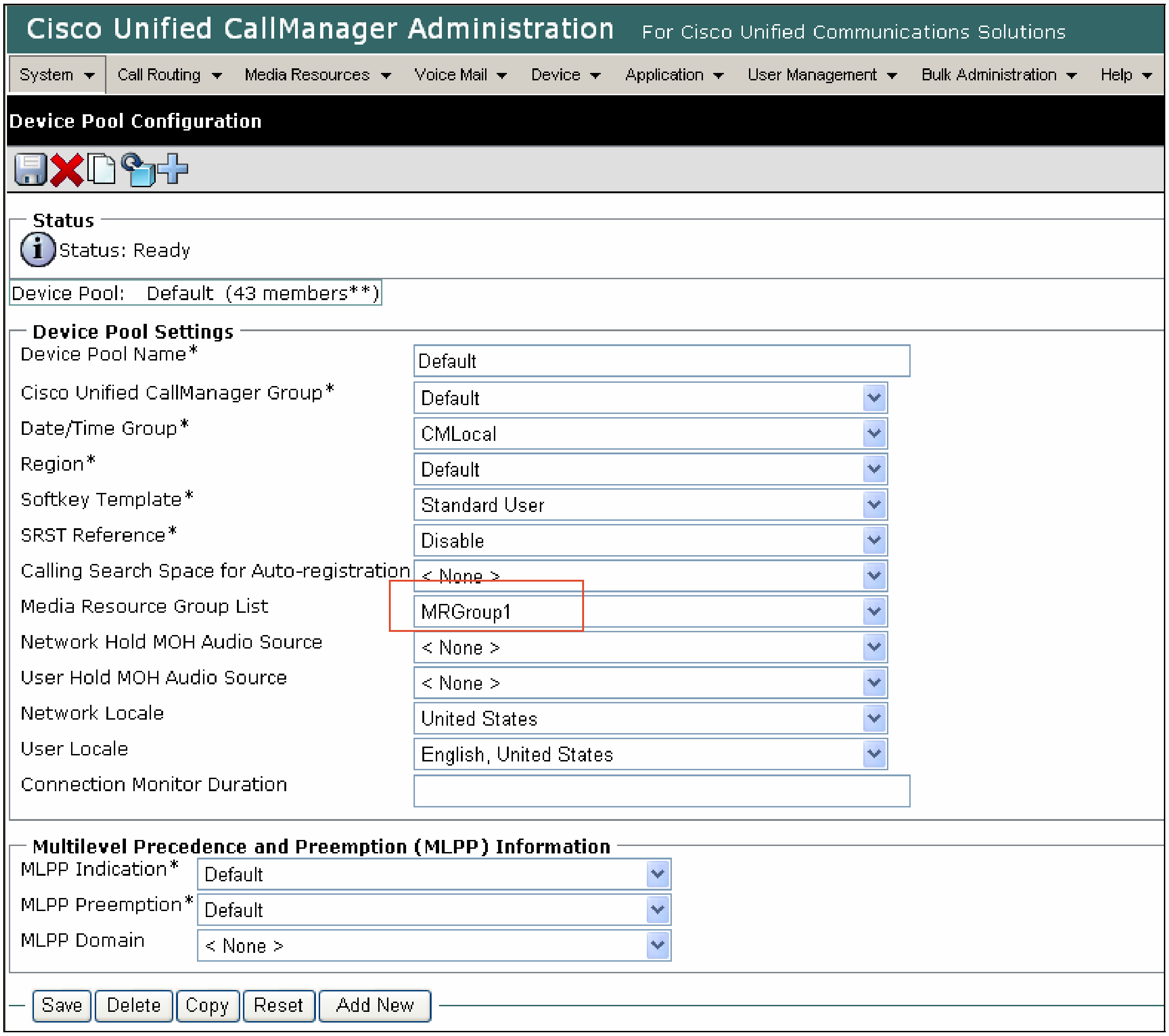
- Click **System → Device Pool**
- Click **find**



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

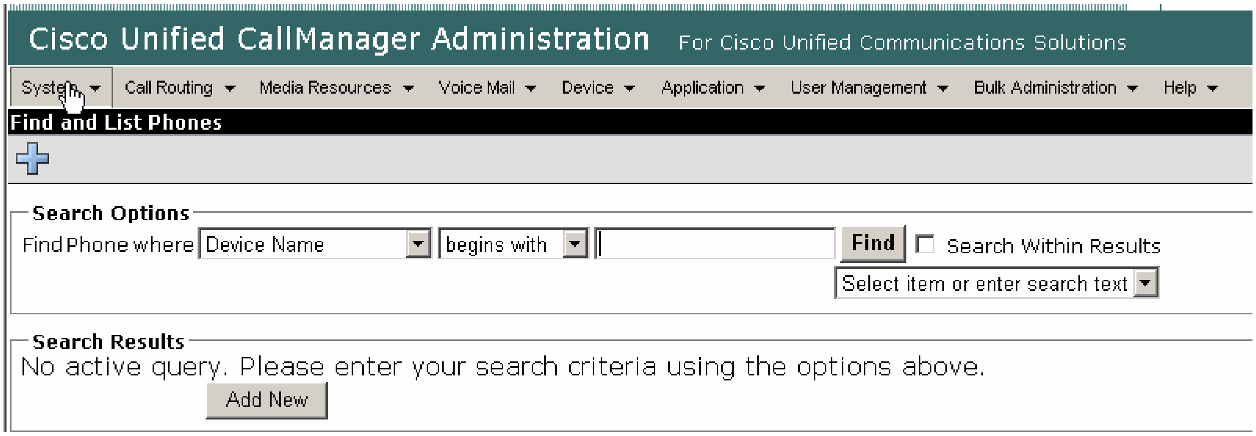


- Use the drop-down window to select **MRGroup1** as the **Media Resource Group List**
- Leave other fields as default
- Click **Save** and **Reset**



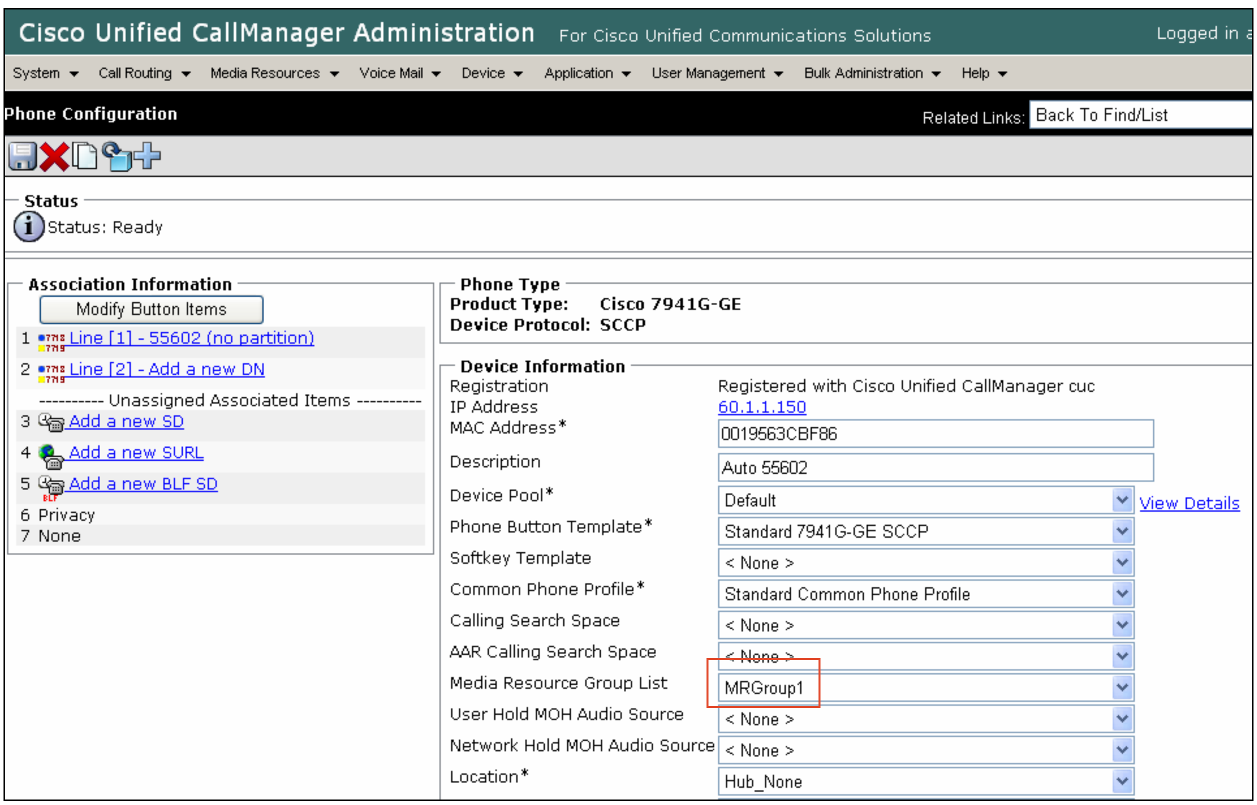
In order for Cisco IP telephones to use the Conference Bridge, **MRGroup1** must be set as the **Media Resource Group List** in telephone administration. The following illustrates the configuration for Extension 55602. Repeat this configuration for all other IP telephones.

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



- Click the phone's MAC address link

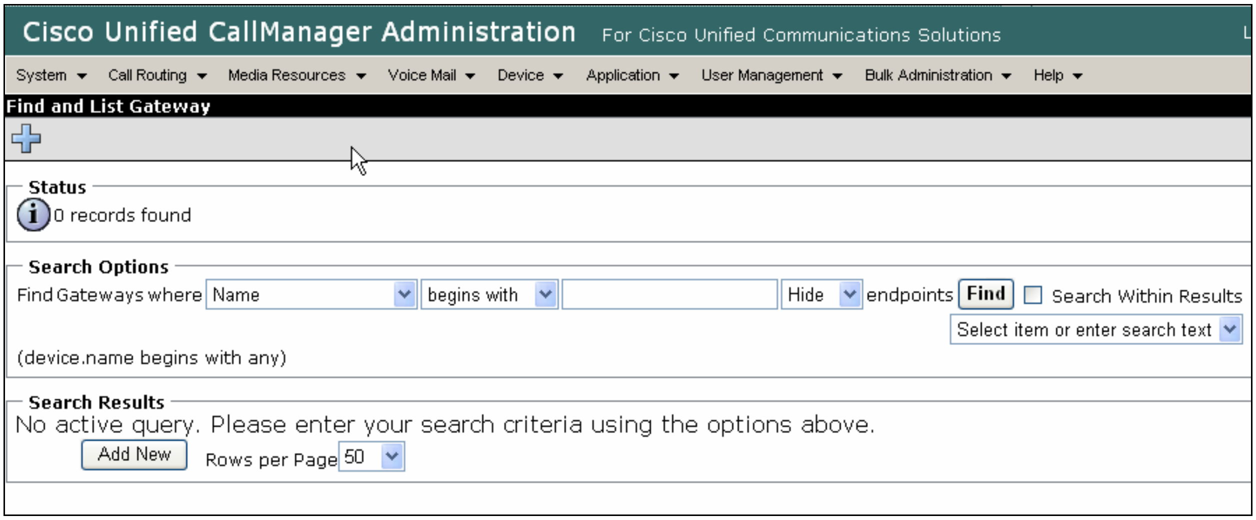
Set **Media Resource Group List** to **MRGroup1** as shown below.



4.5. Add an H.323 Gateway

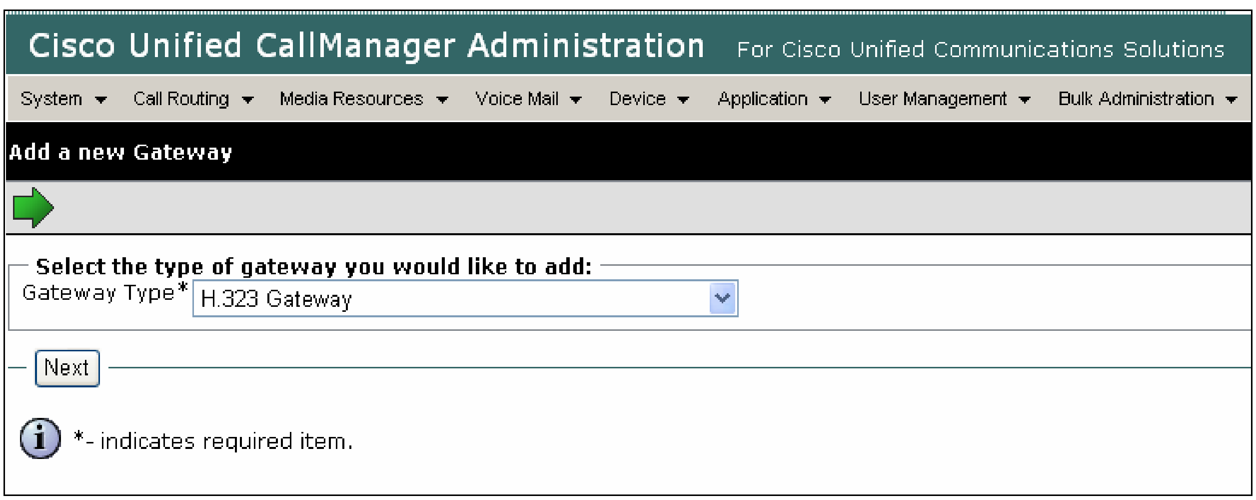
From the Cisco Unified CallManager Administration screen,

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

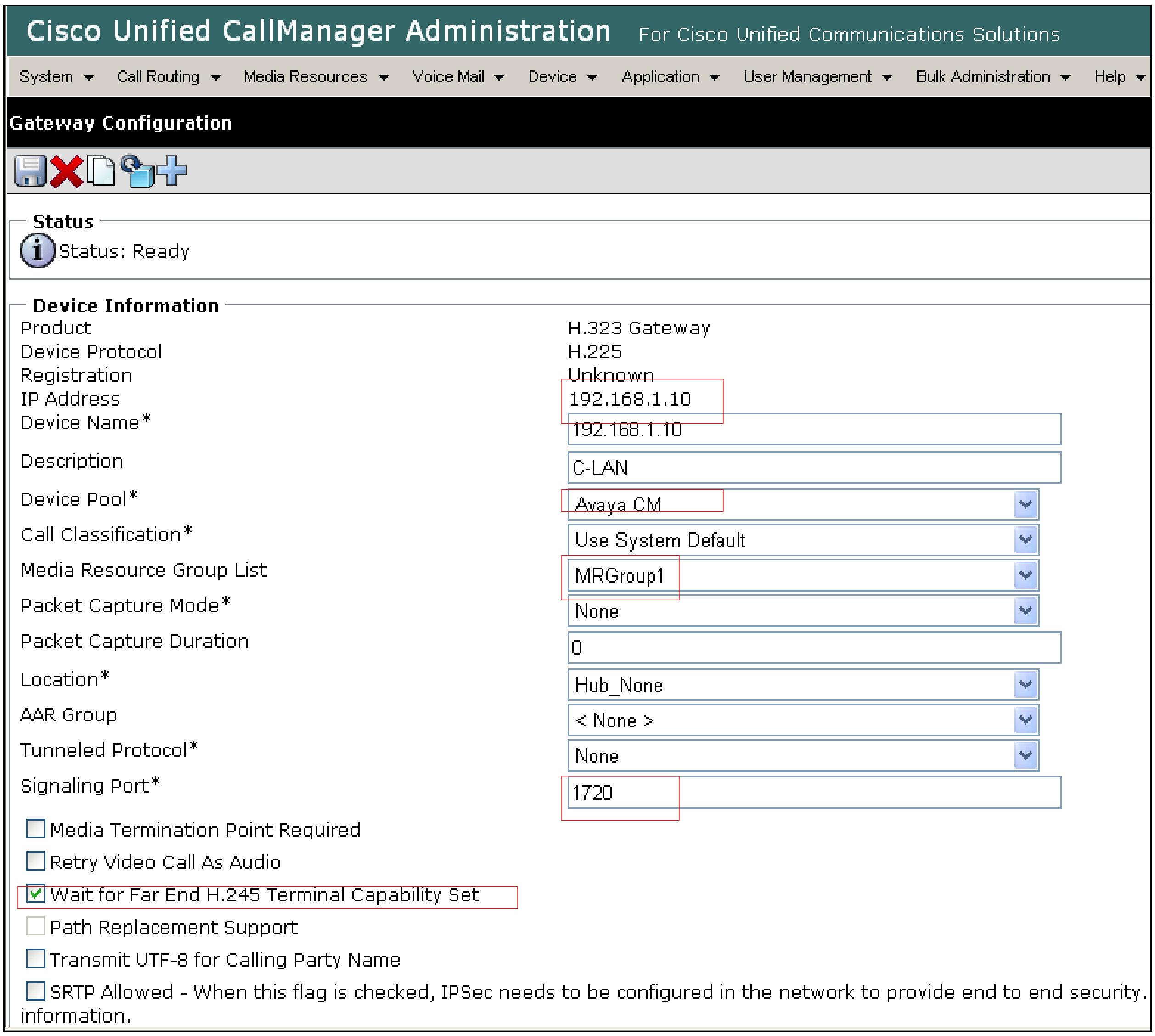


From the Gateway Type drop-down list box,

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

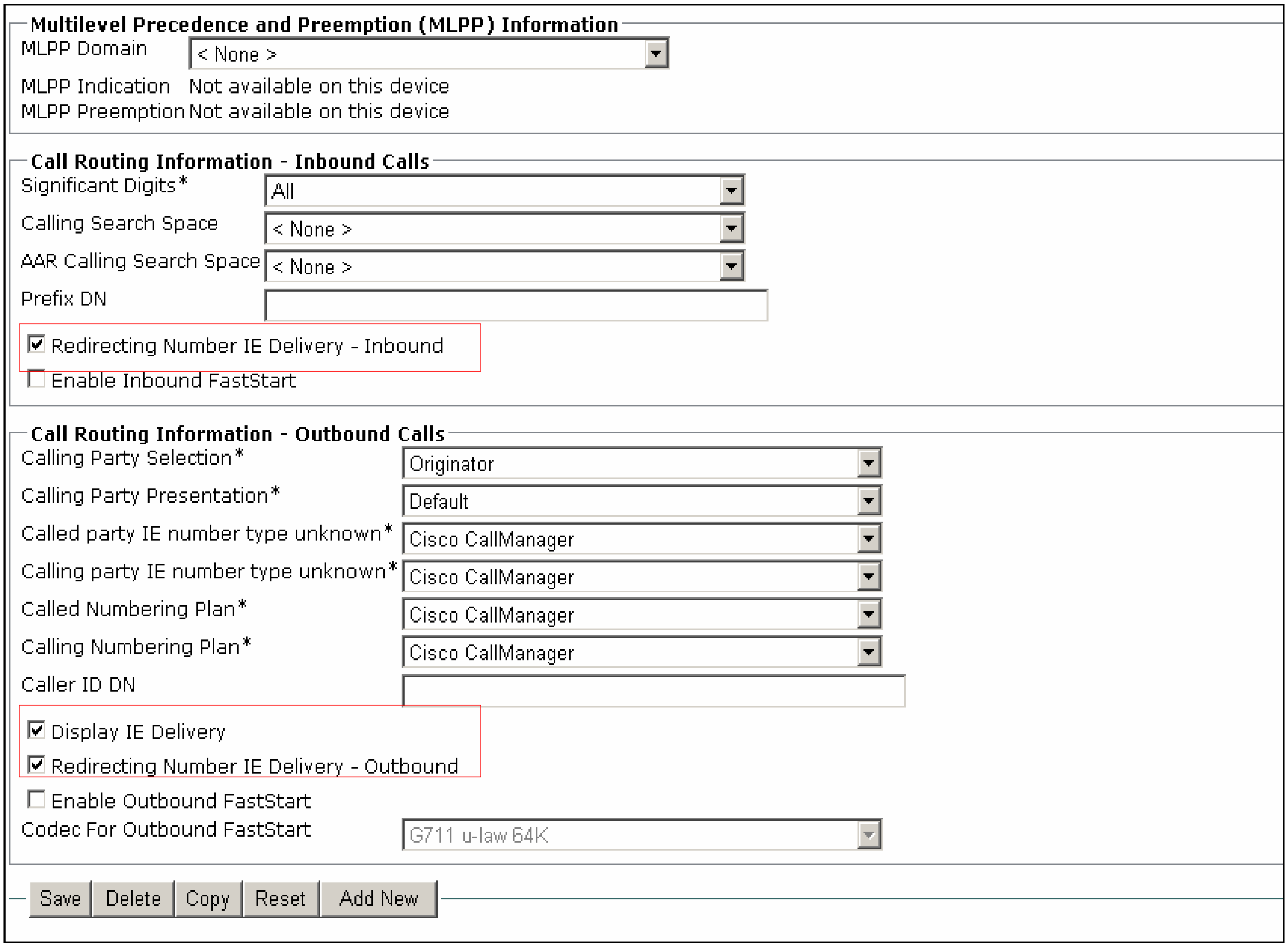


After clicking **Next**, enter the gateway configuration information as shown below. The **Device Name** corresponds to the C-LAN IP address used in the signaling group definition on the Avaya S8710 Server. Select **Avaya CM** for **Device Pool** and **MRGroup1** for **Media Resource Group List**. Note 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.

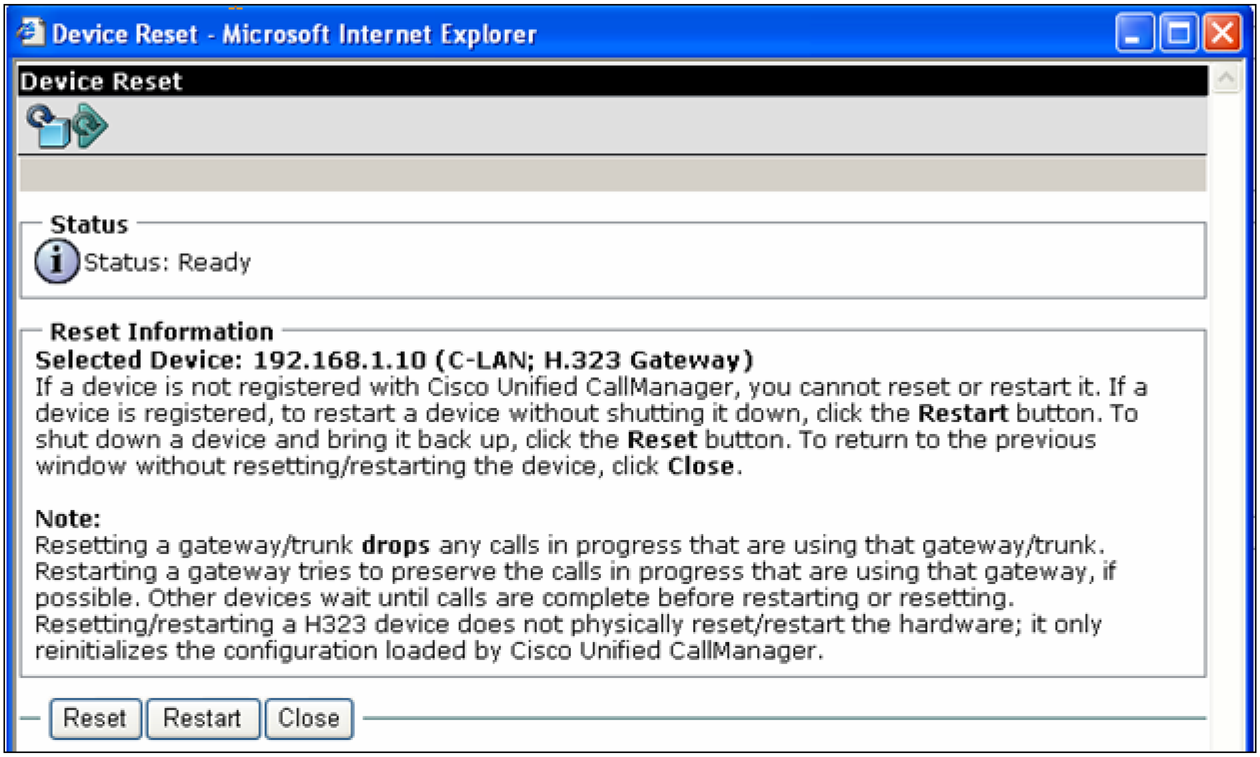


Below is the continuation of the previous screen.

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



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

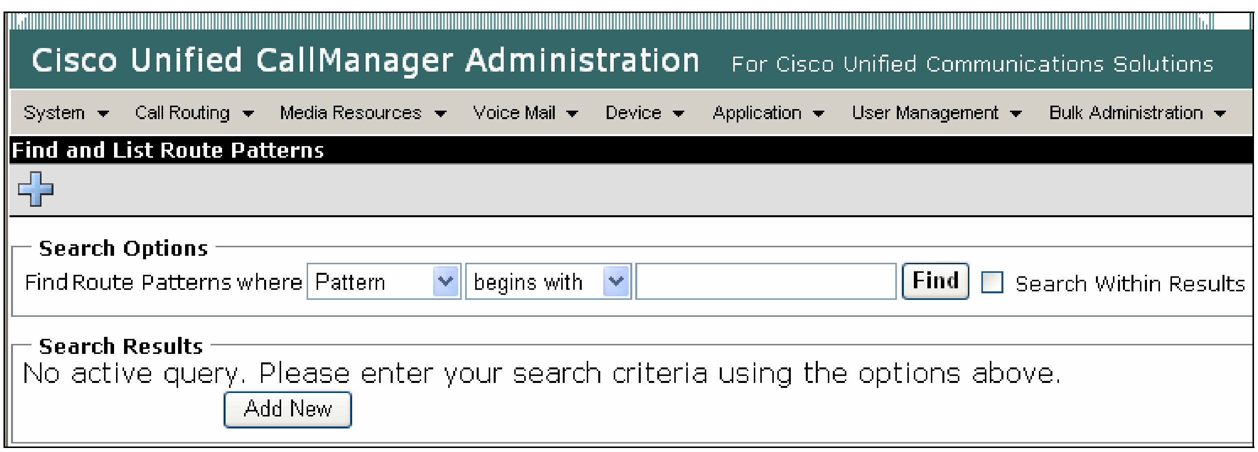


4.6. Configure Route-pattern on the Cisco Unified CallManager

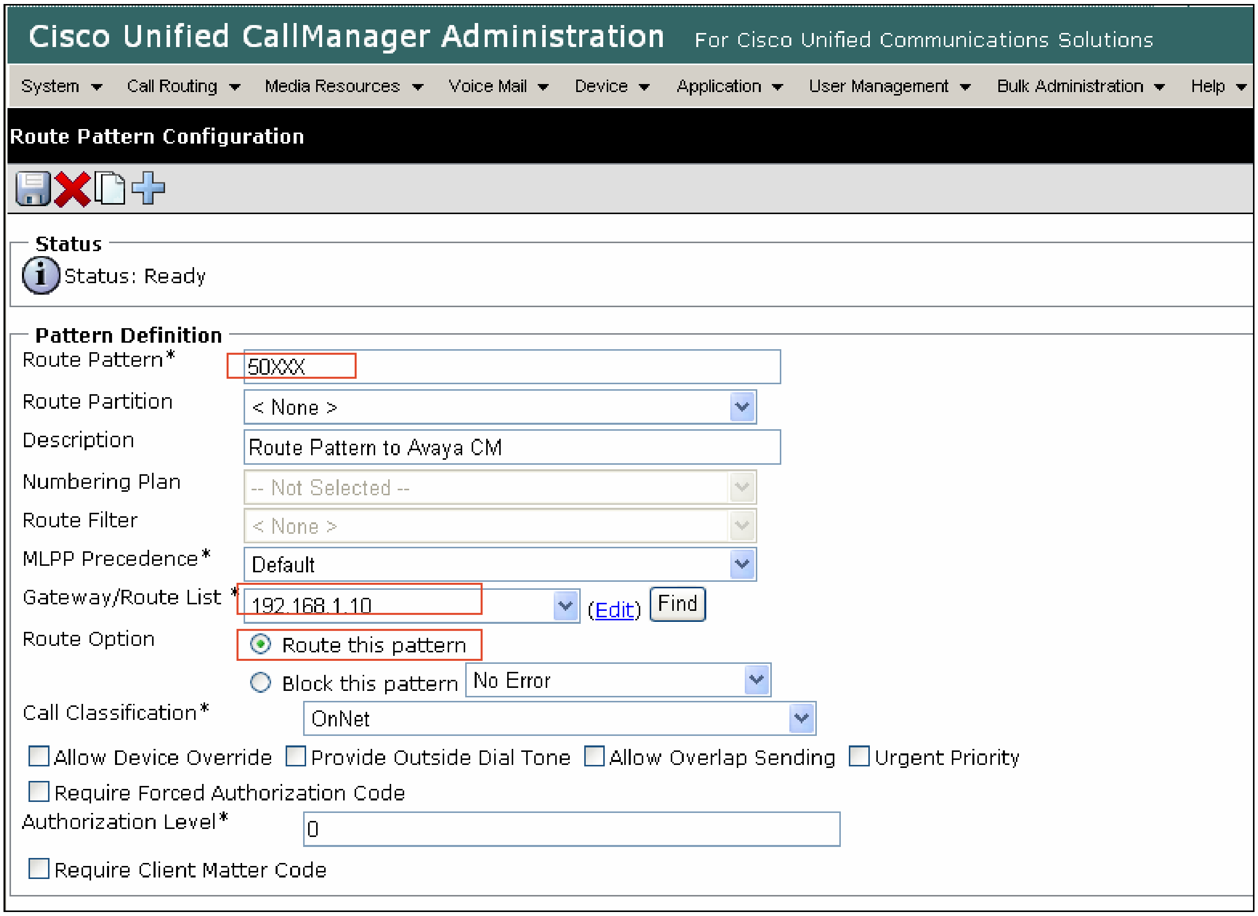
The routing pattern is configured such that calls from the Cisco IP phones to extension range 50xxx are directed to the gateway 192.168.1.10, the IP address of the C-LAN in the Avaya G650 Media Gateway. The next screen shows the configuration.

From the Cisco Unified CallManager Administration screen,

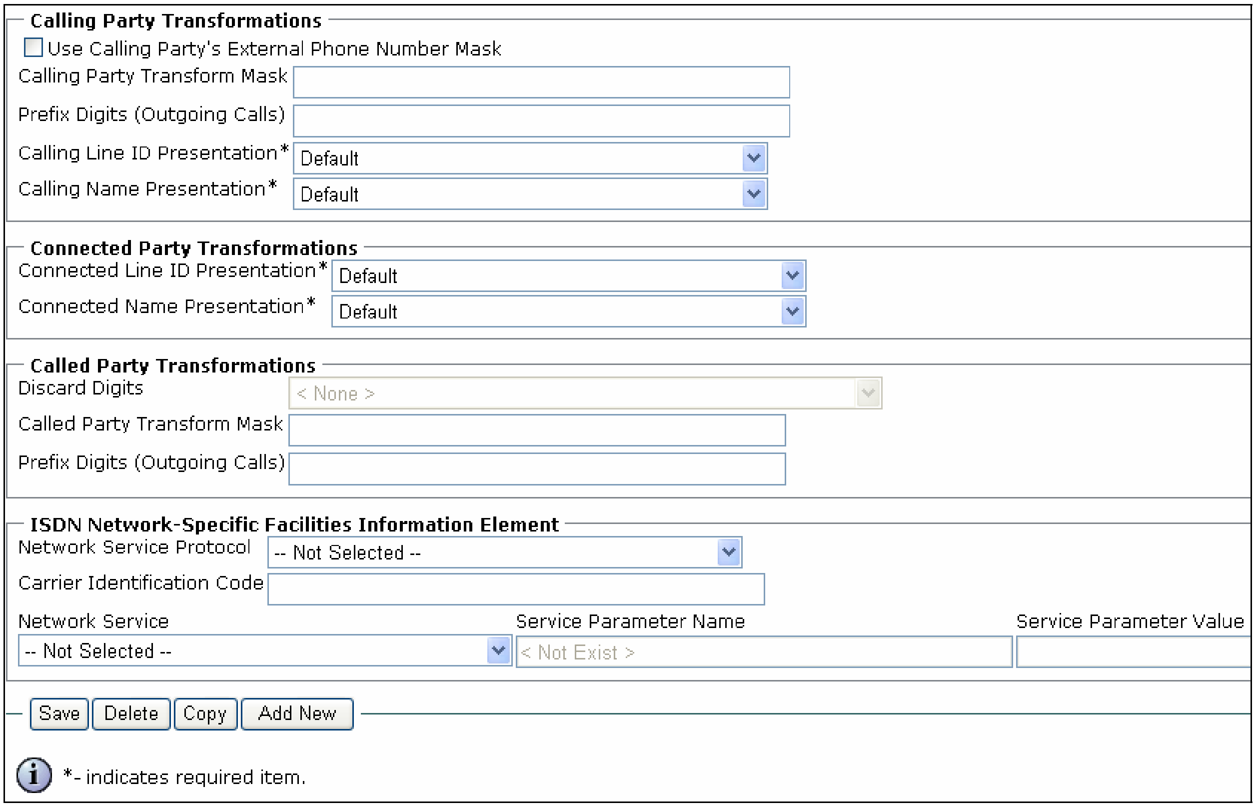
- Click **Call Routing → Route/Hunt → Route Pattern** as shown below
- Click **Add New**



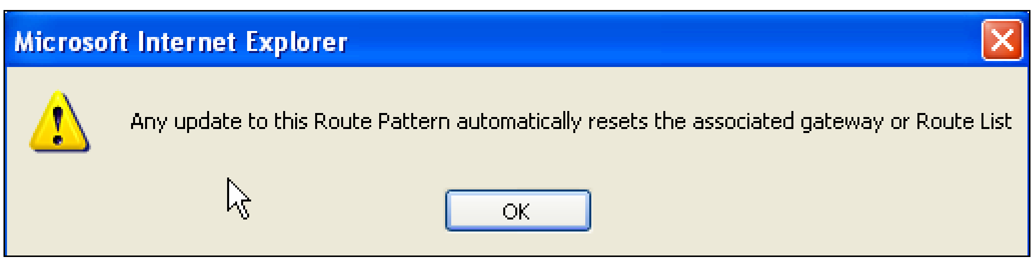
- Enter **50XXX** in the **Route Pattern** field as shown below
- From the **Gateway/Route List** drop down box, select gateway **192.168.1.10**
- Click **Route this pattern** from **Route Option**
- Leave other settings as shown



This screen continues on next page.



- 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

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

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

sccp ccm 192.45.130.105 priority 1   --- add Cisco Unified CallManager as SCCP Server with
                                         priority 1
sccp codec g711ulaw mask             --- Add codecs supported by this conference bridge
sccp codec g729r8 mask
sccp codec g729ar8 mask
sccp 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                      HOSPITALITY STATUS
Group Cntrl Restr: none                   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.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya Solution & Interoperability Test Lab at interoplabnotes@list.avaya.com