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.
2. Hardware and Software Used for Verification

Table 1 lists the equipment and software used for verification.

<table>
<thead>
<tr>
<th>Equipment</th>
<th>Software</th>
</tr>
</thead>
<tbody>
<tr>
<td>Avaya S8710 Server</td>
<td>R014x.00.1.731.2 with Service Pack 2 (patch 14576)</td>
</tr>
<tr>
<td>Avaya G650 Media Gateway with</td>
<td></td>
</tr>
<tr>
<td>• C-LAN</td>
<td>HW01 FW024</td>
</tr>
<tr>
<td>• MEDPRO</td>
<td>HW20 FW095</td>
</tr>
<tr>
<td>Avaya 9630 IP Telephone</td>
<td>R2.1 (H.323)</td>
</tr>
<tr>
<td>Avaya 4621SW IP Telephone</td>
<td>R2.8 (H.323)</td>
</tr>
<tr>
<td>Cisco 3825 Router</td>
<td>IOS 12.4(15)T1</td>
</tr>
<tr>
<td>Cisco 2811 Router</td>
<td>IOS 12.4(15)T1</td>
</tr>
<tr>
<td>Cisco 3750 Catalyst Switch</td>
<td>IOS 12.2(25)SEA</td>
</tr>
<tr>
<td>Cisco Unified CallManager</td>
<td>Release 5.1.3.1000-12</td>
</tr>
<tr>
<td>Cisco 7970 and 7941G IP Telephones</td>
<td>Release 8.3-2S</td>
</tr>
</tbody>
</table>

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.

<table>
<thead>
<tr>
<th>change node-names ip</th>
<th>Page 1 of 1</th>
</tr>
</thead>
<tbody>
<tr>
<td>Name</td>
<td>IP Address</td>
</tr>
<tr>
<td>C-LAN 192.168.1.10</td>
<td>.</td>
</tr>
<tr>
<td>MedPro 192.168.1.11</td>
<td>.</td>
</tr>
</tbody>
</table>
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
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
   Gateway Address: 192.168.1.1
   Enable Ethernet Port? y
   Network Region: 1
   VLAN: n
   Auto? Y

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

display ip-interface 1a03
1
   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

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

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

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                 AUDIO RESOURCE RESERVATION PARAMETERS
   Call Control 802.1p Priority: 6  RSVP Enabled? n
   Audio 802.1p Priority: 6
   Video 802.1p Priority: 5
H.323 IP ENDPOINTS
   H.323 Link Bounce Recovery? y
   Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
   Keep-Alive Count: 5
```

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

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

Region: 3
Location: 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

802.1P/Q PARAMETERS
Call Control 802.1p Priority: 6
Audio 802.1p Priority: 6
Video 802.1p Priority: 5

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

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

```
change ip-network-region 3

Inter Network Region Connection Management

<table>
<thead>
<tr>
<th>src</th>
<th>dst</th>
<th>codec</th>
<th>direct</th>
<th>WAN-BW-limits</th>
<th>Video</th>
<th>Dyn</th>
</tr>
</thead>
<tbody>
<tr>
<td>rgn</td>
<td>rgn</td>
<td>set</td>
<td>WAN</td>
<td>Units</td>
<td>Total Norm</td>
<td>Prio Shr Intervening-regions</td>
</tr>
<tr>
<td>3</td>
<td>1</td>
<td>3</td>
<td>y</td>
<td>NoLimit</td>
<td>n</td>
<td></td>
</tr>
<tr>
<td>3</td>
<td>2</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>3</td>
<td>3</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>3</td>
<td>4</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
```
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.

<table>
<thead>
<tr>
<th>From IP Address</th>
<th>(To IP Address or Mask)</th>
<th>Subnet</th>
<th>Region</th>
<th>VLAN</th>
<th>Location</th>
<th>Extension</th>
</tr>
</thead>
<tbody>
<tr>
<td>192.168.1.1</td>
<td>192.168.1.254</td>
<td>1</td>
<td>1</td>
<td>n</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
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

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
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
DTMF over IP: out-of-band    H.235 Annex H Required? n
Link Loss Delay Timer(sec): 90 Direct IP-IP Audio Connections? y
Enable Layer 3 Test? n       IP Audio Hairpinning? 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.

<table>
<thead>
<tr>
<th>add trunk-group 3</th>
<th>Page 1 of 21</th>
</tr>
</thead>
<tbody>
<tr>
<td>Group Number: 3</td>
<td>Group Type: isdn</td>
</tr>
<tr>
<td>Group Name: OUTSIDE CALL</td>
<td>CDR Reports: y</td>
</tr>
<tr>
<td>Direction: two-way</td>
<td>COR: 1</td>
</tr>
<tr>
<td>Dial Access? n</td>
<td>TN: 1</td>
</tr>
<tr>
<td>Service Type: tie</td>
<td>TAC: 110</td>
</tr>
<tr>
<td>Carrier Medium: H.323</td>
<td>Busy Threshold: 255</td>
</tr>
<tr>
<td>Queue Length: 0</td>
<td>Night Service:</td>
</tr>
<tr>
<td>Service Type: tie</td>
<td>Auth Code? n</td>
</tr>
</tbody>
</table>

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.

<table>
<thead>
<tr>
<th>add trunk-group 3</th>
<th>Page 2 of 21</th>
</tr>
</thead>
<tbody>
<tr>
<td>Group Type: isdn</td>
<td>Codeset to Send Display: 0</td>
</tr>
<tr>
<td>TRUNK PARAMETERS</td>
<td>Codeset to Send National IEs: 6</td>
</tr>
<tr>
<td></td>
<td>Charge Advice: none</td>
</tr>
<tr>
<td></td>
<td>Supplementary Service Protocol: a</td>
</tr>
<tr>
<td></td>
<td>Digit Handling (in/out): enbloc/enbloc</td>
</tr>
<tr>
<td></td>
<td>Digital Loss Group: 18</td>
</tr>
<tr>
<td></td>
<td>Format:</td>
</tr>
<tr>
<td></td>
<td>Disconnect Supervision - In? y Out? n</td>
</tr>
<tr>
<td></td>
<td>Answer Supervision Timeout: 0</td>
</tr>
<tr>
<td></td>
<td>Display Incoming Digits? n</td>
</tr>
</tbody>
</table>
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.

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

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.

<table>
<thead>
<tr>
<th>add trunk-group 3</th>
<th>Page 5 of 21</th>
</tr>
</thead>
<tbody>
<tr>
<td>TRUNK GROUP</td>
<td>--------------</td>
</tr>
<tr>
<td>Administered Members (min/max):</td>
<td>1/5</td>
</tr>
<tr>
<td>Total Administered Members:</td>
<td>5</td>
</tr>
<tr>
<td>GROUP MEMBER ASSIGNMENTS</td>
<td></td>
</tr>
<tr>
<td>Port</td>
<td>Name</td>
</tr>
<tr>
<td>1: ip</td>
<td>3</td>
</tr>
<tr>
<td>2: ip</td>
<td>3</td>
</tr>
<tr>
<td>3: ip</td>
<td>3</td>
</tr>
<tr>
<td>4: ip</td>
<td>3</td>
</tr>
<tr>
<td>5: ip</td>
<td>3</td>
</tr>
<tr>
<td>6:</td>
<td></td>
</tr>
</tbody>
</table>
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.

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

### 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.

<table>
<thead>
<tr>
<th>change route-pattern 9</th>
</tr>
</thead>
<tbody>
<tr>
<td>Pattern Number: 10</td>
</tr>
<tr>
<td>Pattern Name: To CallManager</td>
</tr>
<tr>
<td>SCCAN? n</td>
</tr>
<tr>
<td>Secure SIP? n</td>
</tr>
<tr>
<td>Grp FRL NPA Pfx Hop Toll No. Inserted DCS/ IXC</td>
</tr>
<tr>
<td>No Mrk Lmt List Del Digits QSIG Intw</td>
</tr>
<tr>
<td>1: 3 0 n user</td>
</tr>
<tr>
<td>2: n user</td>
</tr>
<tr>
<td>BCC VALUE TSC CA-TSC</td>
</tr>
<tr>
<td>ITC BCIE Service/Feature PARM No. Numbering LAR</td>
</tr>
<tr>
<td>0 1 2 M 4 W Request Dgts Format Subaddress</td>
</tr>
<tr>
<td>1: y y y y y n n rest none</td>
</tr>
<tr>
<td>2: y y y y y n n rest none</td>
</tr>
</tbody>
</table>

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

<table>
<thead>
<tr>
<th>change aar analysis 55</th>
</tr>
</thead>
<tbody>
<tr>
<td>AAR DIGIT ANALYSIS TABLE</td>
</tr>
<tr>
<td>Percent Full: 2</td>
</tr>
<tr>
<td>Dialed String</td>
</tr>
<tr>
<td>55</td>
</tr>
</tbody>
</table>
Use **change public-unknown-numbering 5** command to configure Avaya Communication Manager to pass extensions 50xxx on trunk group 3 to the Cisco CallManager.

<table>
<thead>
<tr>
<th>Ext Len</th>
<th>Ext Code</th>
<th>Trk Genp</th>
<th>CPN Prefix</th>
<th>CPN Len</th>
<th>Total CPN Len</th>
</tr>
</thead>
<tbody>
<tr>
<td>5</td>
<td>50</td>
<td>3</td>
<td>5</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

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**

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

A summary of U.S. laws governing Cisco cryptographic products may be found at:

If you require further assistance please contact us by sending email to export@cisco.com.
• Click **Yes** at the Security Alert

![Security Alert Image]

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

![Login Image]
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**

**Cisco Unified CallManager Administration**

**Find and List Conference Bridges**

- **Search Options**
  - Find Conference Bridges where Name begins with
  - Find
  - Search Within Results

- **Search Results**
  - No active query. Please enter your search criteria using the options above.
    - 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**

**Cisco Unified CallManager Administration**

**Conference Bridge Configuration**

- **Status**
  - Status: Ready

- **Conference Bridge Information**
  - Conference Bridge: New

  **Hardware Conference Bridge Info**
  - Conference Bridge Type
    - Cisco Conference Bridge Hardware
  - MAC Address
    - 001936915E9
  - Description
    - Conference Bridge on Cisco 3825 router
  - Device Pool
    - Default
  - Location
    - Hub_Name
  - Special Load Information

- **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.

![Cisco Unified CallManager Administration](image)

**Media Resource Group Configuration**

<table>
<thead>
<tr>
<th><strong>Status</strong></th>
<th>Ready</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Media Resource Group Status</strong></td>
<td>Media Resource Group: MRS1 (used by 44 devices)</td>
</tr>
<tr>
<td><strong>Media Resource Group Information</strong></td>
<td></td>
</tr>
<tr>
<td>Name</td>
<td>MRS1</td>
</tr>
<tr>
<td>Description</td>
<td>media resource 1</td>
</tr>
</tbody>
</table>

**Devices for this Group**

**Available Media Resources**
- ANN_2
- CFB_2
- MTP_011936815E9
- MTP_2
- MOH_2 (MOH)[Multicast]

**Selected Media Resources**
- CFB0011936815E9 (CFB)

- **Use Multicast for MOH Audio** *(if at least one multicast MOH resource is available)*

- Save  Delete  Copy  Reset  Add New

* 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 MSGroup1 as Name
- Highlight the MRS1 and click the ▽ to move it to the Selected Media Resource Groups area

---

### Media Resource Group List Configuration

<table>
<thead>
<tr>
<th>Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>Status: Ready</td>
</tr>
</tbody>
</table>

### Media Resource Group List Information

<table>
<thead>
<tr>
<th>Name*</th>
<th>MRGp1</th>
</tr>
</thead>
</table>

### Media Resource Groups for this List

<table>
<thead>
<tr>
<th>Available Media Resource Groups</th>
</tr>
</thead>
<tbody>
<tr>
<td>MRS1</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Selected Media Resource Groups</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
</tr>
</tbody>
</table>

---

Save
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**

<table>
<thead>
<tr>
<th>Address</th>
<th>Web License Manager (WebUI)</th>
</tr>
</thead>
<tbody>
<tr>
<td><a href="https://192.168.10.98/locadmin/devicePoolFind.stu">https://192.168.10.98/locadmin/devicePoolFind.stu</a></td>
<td>CCN1.(10)Sub</td>
</tr>
</tbody>
</table>

### Cisco Unified CallManager Administration

**Find and List Device Pools**

**Search Options**

Find device pool where [Device Pool Name] begins with

**Search Results**

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

Add New

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

<table>
<thead>
<tr>
<th>Device Pool Name</th>
<th>Unified CallManager Group</th>
<th>Region</th>
<th>Date/Time Group</th>
</tr>
</thead>
<tbody>
<tr>
<td>Avaya CM</td>
<td>Default</td>
<td>Avaya</td>
<td>DMSLocal</td>
</tr>
<tr>
<td>Avaya SIP Trunk Region</td>
<td>Default</td>
<td>Avaya SIP Trunk Region</td>
<td>DMSLocal</td>
</tr>
<tr>
<td>Default</td>
<td>Default</td>
<td>Default</td>
<td>DMSLocal</td>
</tr>
</tbody>
</table>
• 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 Device → Phone

Click Find

Search Options
Find Phone where

Find Phone

Device Name

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

SEF0019563CBF86
Auto 55602
Default
SCCP

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.

---

**Cisco Unified CallManager Administration**

**Gateway Configuration**

<table>
<thead>
<tr>
<th>Status</th>
<th>Ready</th>
</tr>
</thead>
</table>

**Device Information**

<table>
<thead>
<tr>
<th>Product</th>
<th>H.323 Gateway</th>
</tr>
</thead>
<tbody>
<tr>
<td>Registration</td>
<td>H.3225</td>
</tr>
<tr>
<td>IP Address</td>
<td>192.168.1.10</td>
</tr>
<tr>
<td>Device Name</td>
<td>CLAN</td>
</tr>
<tr>
<td>Description</td>
<td>192.168.1.10</td>
</tr>
</tbody>
</table>

**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

### Call Routing Information - Inbound Calls
- **Significant Digits**
- **Calling Search Space**
- **AAA Calling Search Space**
- **Prefix DN**

- **Redirecting Number IE Delivery - Inbound**
- **Enable Inbound FastStart**

### Call Routing Information - Outbound Calls
- **Calling Party Selection**
- **Calling Party Presentation**
- **Called party IE number type unknown**
- **Calling party IE number type unknown**
- **Called Numbering Plan**
- **Calling Numbering Plan**
- **Caller ID DN**

- **Display IE Delivery**
- **Redirecting Number IE Delivery - Outbound**
- **Enable Outbound FastStart**
- **Codec For Outbound FastStart**

### Multilevel Precedence and Preemption (MLPP) Information
- **MLPP Domain**
- **MLPP Indication**
- **MLPP Preemption**
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

**Route Pattern Configuration**

<table>
<thead>
<tr>
<th>Status</th>
<th>Status: Ready</th>
</tr>
</thead>
</table>

**Pattern Definition**

- **Route Pattern**: 50XXX
- **Route Partition**: < None >
- **Description**: Route Pattern to Avaya CM
- **Numbering Plan**: < Not Selected >
- **Route Filter**: < None >
- **MLPP Precedence**: Default
- **Gateway/Route List**: 192.168.1.10
- **Route Option**: Route this pattern
- **Call Classification**: 
- **Allow Device Override**: [ ]
- **Provide Outside Dial Tone**: [ ]
- **Allow Overlap Sending**: [ ]
- **Urgent Priority**: [ ]
- **Require Forced Authorization Code**: [ ]
- **Authorization Level**: 0
- **Require Client Matter Code**: [ ]

This screen continues on next page.
- **Calling Party Transformations**
  - Use Calling Party’s External Phone Number Mask
  - Calling Party Transform Mask
  - Prefix Digits (Outgoing Calls)
  - 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
  - Prefix Digits (Outgoing Calls)

- **ISDN Network-Specific Facilities Information Element**
  - Network Service Protocol: -- Not Selected --
  - Carrier Identification Code

<table>
<thead>
<tr>
<th>Network Service</th>
<th>Service Parameter Name</th>
<th>Service Parameter Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>-- Not Selected --</td>
<td>&lt; Not Exist &gt;</td>
<td></td>
</tr>
</tbody>
</table>

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

---

Any update to this Route Pattern automatically resets the associated gateway or Route List.

![Microsoft Internet Explorer](image)
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

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

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

sc cp ccm 192.45.130.105 priority 1  --- add Cisco Unified CallManager as SCCP Server with priority 1

sc cp codec g711ulaw mask
sc cp codec g729r8 mask
sc cp codec g729ar8 mask
sc cp 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
!
```
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

GENERAL STATUS
Administered Type: 4620           Service State: in-service/off-hook
Connected Type: 9640           TCP Signal Status: connected
Extension: 50008
          Port: S00026
Call Parked? 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

CALL CONTROL SIGNALING
Port: S00026
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

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

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

<table>
<thead>
<tr>
<th>Member</th>
<th>Port</th>
<th>Service State</th>
<th>Mtce Connected Ports</th>
</tr>
</thead>
<tbody>
<tr>
<td>0003/001</td>
<td>T00060</td>
<td>in-service/idle</td>
<td>no</td>
</tr>
<tr>
<td>0003/002</td>
<td>T00061</td>
<td>in-service/idle</td>
<td>no</td>
</tr>
<tr>
<td>0003/003</td>
<td>T00062</td>
<td>in-service/idle</td>
<td>no</td>
</tr>
<tr>
<td>0003/004</td>
<td>T00063</td>
<td>in-service/active</td>
<td>no</td>
</tr>
<tr>
<td>0003/005</td>
<td>T00064</td>
<td>in-service/idle</td>
<td>no</td>
</tr>
<tr>
<td>0003/006</td>
<td>T00065</td>
<td>in-service/idle</td>
<td>no</td>
</tr>
</tbody>
</table>

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:
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 Cisco 7941G 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.

<table>
<thead>
<tr>
<th>sess_id</th>
<th>conn_id</th>
<th>stype</th>
<th>mode</th>
<th>codec</th>
<th>ripaddr</th>
<th>rport</th>
<th>sport</th>
</tr>
</thead>
<tbody>
<tr>
<td>16778332</td>
<td>16778182</td>
<td>conf</td>
<td>sendrecv</td>
<td>g729b</td>
<td>60.1.1.151</td>
<td>22708</td>
<td>20238</td>
</tr>
<tr>
<td>16778332</td>
<td>16778184</td>
<td>conf</td>
<td>sendrecv</td>
<td>g729b</td>
<td>60.1.1.150</td>
<td>22578</td>
<td>24380</td>
</tr>
<tr>
<td>16778332</td>
<td>16778186</td>
<td>conf</td>
<td>sendrecv</td>
<td>g729b</td>
<td>192.168.1.110</td>
<td>2144</td>
<td>24616</td>
</tr>
</tbody>
</table>

Total number of active session(s) 1, and connection(s) 3

• Display verifications:
  o 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.
  o 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


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


9. Change History

<table>
<thead>
<tr>
<th>Issue</th>
<th>Date</th>
<th>Reason</th>
</tr>
</thead>
<tbody>
<tr>
<td>1.1</td>
<td></td>
<td>Update to incorporate newer version of Avaya Communication Manager and Cisco Unified CallManager and to incorporate IP-IP Direct Audio calling (shuffling).</td>
</tr>
</tbody>
</table>