

Avaya Solution & Interoperability Test Lab

Configuring H.323 Signaling and IP Trunks between Avaya Communication Manager and Cisco CallManager 4.0 - Issue 1.0

Abstract

These Application Notes present a sample configuration for a network comprised of an Avaya S8700 Media Server IP Connect configuration and a Cisco CallManager. The focus is on the Avaya Communication Manager configuration for the H.323 Signaling Group and IP Trunk Group, and the corresponding configuration of the H.323 Gateway on the Cisco CallManager 4.0. The configuration described should be applicable to other Avaya Media Servers and Media Gateways.

1. Introduction

These Application Notes present a sample configuration for a network comprised of an Avaya S8700 Media Server IP Connect configuration and a Cisco CallManager. The focus is on the configuration of the H.323 Signaling Group and IP Trunk Group on the Avaya S8700 Media Servers running Avaya Communication Manager and the corresponding configuration of the H.323 Gateways on the Cisco CallManager 4.0. Using the configuration described herein, Cisco IP Telephones controlled by the Cisco CallManager can call (and be called) by Avaya IP Telephones and other Avaya telephones associated with the Avaya media servers.

These Application Notes update the previously published Application Notes entitled "Avaya S8300 Media Server and Avaya S8700 Media Server Networked with Cisco CallManager using H.323 Signaling and IP Trunk Groups - Issue 1.0". The configuration in the prior Application Notes was based on earlier versions of Avaya Communication Manager and Cisco CallManager.

Figure 1 shows the network configuration.



Figure 1: Avaya-Cisco H.323 Interoperability Configuration

Table 1 shows the device interfaces and IP network assignment.

Network Component	IP Address	Gateway	Network Mask
Avaya S8700 Media Server1	20.1.1.61	20.1.1.1	255.255.255.0
Avaya S8700 Media Server2	20.1.1.62	20.1.1.1	
G650 Gateway			
 IPSI 	20.1.1.9	20.1.1.1	
 C-LAN 	20.1.1.10	20.1.1.1	
 MEDPRO 	20.1.1.11	20.1.1.1	
Avaya C360 Stackable Switch Server			
 VLAN 20 	20.1.1.1		255.255.255.0
 VLAN 135 	135.8.31.2		
Avaya 4620 IP Telephone	20.1.1.120	20.1.1.1	255.255.255.0
Catalyst 6509 Switch			
 VLAN 135 	135.8.31.1		255.255.255.0
Cisco CallManager	135.8.31.20	135.8.31.1	255.255.255.0
Cisco 7960 IP Telephone x30001	135.8.31.100	135.8.31.1	255.255.255.0
Cisco 7960 IP Telephone x30002	135.8.31.101	135.8.31.1	255.255.255.0

Table 1: Device Interfaces and Network IP Address Assignment

2. Hardware and Software Used for Verification

Table 2 lists the equipment and software used for the verification.

Equipment	Software
Avaya S8700 Media Server	R012x.01.1.414.1 (2.1.1)
	With Patch 7689
Avaya G650 Media Gateway with	
• IPSI-2	HW36 FW052
• C-LAN	HW01 FW012
MEDPRO	HW03 FW093
Avaya 4620 IP Telephone	R2.1
Avaya C360 Stackable Switch	R4.3.10
Cisco Catalyst 6509 Switch	OS 7.5
Cisco CallManager	V.4.0
Cisco 7960 IP Telephone	V.5.0(1.1)

Table 2: Hardware and Software Used for Verification

3. Avaya S8700 Media Server Software Configuration

This section presents configuration steps for the Avaya S8700 Media Server IP Connect Configuration. It is assumed that Avaya Communication Manager has been installed and the login and password credentials are available to the reader.

The Avaya S8700 Media Server has multiple IP interfaces. Ethernet 1 is used for the control network to communicate with the IPSI circuit pack of the Avaya G650 Media Gateway. Ethernet 2 is dedicated to the services port. The services port uses the pre-configured IP address 192.11.13.6 with mask 255.255.255.252. Configure the computer's IP address as 192.11.13.5 with mask 255.255.252.252. Connect the computer's Ethernet interface to the services port with a crossover Ethernet cable. The Avaya Communication Manager SAT screens can be accessed using "telnet 192.11.13.6 5023" from a computer connected to the server's services port.

3.1. Add data-module for C-LAN

Use the command "add data-module" to enable the C-LAN. Set the field "Type" to "Ethernet" and the field "Port" to the C-LAN circuit pack (from list configuration all) location with port 17. The following snapshot displays the C-LAN configuration.

```
display data-module 20000
DATA MODULE
Data Extension: 20000
Type: ethernet
Port: 01A0317
Link: 1
```

3.2. Add Node Name and Map IP Address

The following 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 CallManager.

change node-names	ip IP	NODE NAMES	Page	1 of	1
Name	IP Address	Name	IP	Address	
C-LAN	20 .1 .1 .10		•		
Call-Manager	135.8 .31 .20				
MedPro	20 .1 .1 .11				

3.3. Configure C-LAN and MEDPRO

Uses 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 (01A03) and the MEDPRO (01A04). Note that the C-LAN and MEDPRO are assigned to Network Region 1.

```
add ip-interface 01A03

IP INTERFACES

Type: C-LAN
Slot: 01A03
Code/Suffix: TN799 D
Node Name: C-LAN
IP Address: 20.1.1.10
Subnet Mask: 255.255.0
Gateway Address: 20.1.1.1
Enable Ethernet Port? y
Network Region: 1
VLAN:
```

add ip-interface 01A04		
	IP INTERFACES	
Type:	MEDPRO	ETHERNET OPTIONS
Slot:	01A04	Auto? y
Code/Suffix:	TN2302	
Node Name:	MedPro	
IP Address:	20.1.1.11	
Subnet Mask:	255.255.255.0	
Gateway Address:	20.1.1.1	
Enable Ethernet Port?	У	
Network Region:	1	
VLAN:		

3.4. Configure the Network Region

The following illustrates the configuration for network region 1. The intent of illustrating the network region is to show that Codec Set 1 is configured and the "Direct IP-IP Audio Connections" field for both Intra-region and Inter-region may be set to "y" to allow "IP-Direct" media paths.

```
change ip-network-region 1
                                                     Page 1 of 19
                              IP NETWORK REGION
 Region: 1
Location:
                          Home Domain:
   Name:
                               Intra-region IP-IP Direct Audio: yes
                               Inter-region IP-IP Direct Audio: yes
AUDIO PARAMETERS
  Codec Set: 1
                                          IP Audio Hairpinning? y
UDP Port Min: 2048
UDP Port Max: 65535
                                        RTCP Reporting Enabled? y
                              RTCP MONITOR SERVER PARAMETERS
DIFFSERV/TOS PARAMETERS
                                Use Default Server Parameters? y
Call Control PHB Value: 34
       Audio PHB Value: 46
802.1P/Q PARAMETERS
Call Control 802.1p Priority: 7
       Audio 802.1p Priority: 6
                                 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
```

3.5. 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 CallManager.

Signaling group 2 will be created to establish an H.323 signaling link between a C-LAN in the G650 Media Gateway and the Cisco CallManager. The signaling group number is not relevant; use any available signaling group number. Use the command "add signaling-group 2" 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 CallManager node-name "Call-Manager" 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 "y" to allow the final media path for a call to be "direct" from an Avaya IP Telephone to a Cisco IP Telephone.

The far-end network region field can optionally be populated with a network region number to associate with the Cisco CallManager. For the signaling group shown here, the far-end network region is left blank. The software will treat calls using this signaling group as if they were internal to the region of the C-LAN in the G650 (i.e., region 1 in this case, using codec set 1).

add signaling-group 2			Page	1 of	5
	SIGNALII	IG GROUP			
Group Number: 2	Group Type	h.323			
	Remote Office? n	1	Max number of	NCA TSC	: 0
	SBS? n		Max number of	CA TSC:	0
			Trunk Group f	or NCA T	SC:
Trunk Group for	Channel Selection	1:	-		
Supplementary Se	ervice Protocol: a	à	Network Call '	Transfer	? n
	[303 Timer(sec):	0			
	,				
Near-end Node M	Jame: C-LAN	Far-end	d Node Name: C	all-Mana	ger
Near-end Listen H	Port: 1720	Far-end	d Listen Port:	1720	-
		Far-end No	etwork Region:		
LRO Regui	red? n Call	s Share I	P Signaling Co	nnection	? n
RRO Reguin	red?n	H245 Co	ntrol Addr On i	FACility	? n
Media Encrypt	ion? n	Bypagg If	IP Threshold i	Exceeded	? n
neara hierype		Dypabb II	II IIICOIICIA .	Inceeded	• •
DTMF over 1	P: out-of-band	Direct T	P-TP Audio Com	nections	2 🗸
	li · ouc or build	DIICCC II	TP Audio Hai	rninning	• 1 • ? n
		Thtorwo	rking Moggogo'	DBOGrog	
		THCET WO.	INTING MESSAGE.	FROGLES	Ð

3.6. Configure IP Trunk Group

Use the command "add trunk-group 2" to create an H.323 IP trunk group on the S8700 Media Server. Most fields can be left to their defaults. Data has been entered in the fields shown in bold. Note that the trunk "Carrier Medium" is IP. The "Codeset to Send Display" field is set to "0" as shown on page 1. 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 CallManager will display the calling party name on incoming calls from Avaya to Cisco telephones. Similarly, the Cisco CallManager will display the connected party name on Cisco telephones when calls from Cisco telephones to Avaya telephones are answered.

add trunk-group 2	Page 1 of 22	
	TRUNK GROUP	
Group Number: 2 Group Name: to Call-mana Direction: two-way Dial Access? y Queue Length: 0 Service Type: tie	Group Type: isdn ger COR: 1 Outgoing Display? n Busy Threshold: 255 Auth Code? n	CDR Reports: y TN: 1 TAC: 101 Carrier Medium: IP Night Service: TestCall ITC: rest
F	ar End Test Line No:	
TestCall BCC: 4 TRUNK PARAMETERS Codeset to Send Dis Max Message Size to	play: 0 Codeset to Send: 260 Charge Adv	Send National IEs: 6
Supplementary Service Prot	ocol: a Digit Handling (in/out): enbloc/enbloc
Trunk Hunt: cy	clical	igital Loss Group: 18
Incoming Calling Number - Bit Rate: 12 Disconnect Supervision - Answer Supervision Timeou	Delete: Insert: 00 Synchronization: In?y Out?n t:0	Format: async Duplex: full

In Page 2 of the configuration, set "y" in the fields of "Send Name" and "Send Calling Number" as shown below. Note that the "Send Connected Number" field should remain set to "n" so that the Avaya S8700 Media Server will not include a Connected Number Information Element in the Q.931 CONNECT message. The Cisco CallManager software tested will not display the connected number, if present in the Q.931 CONNECT message.

```
add trunk-group 2
                                                                2 of 22
                                                         Page
TRUNK FEATURES
    ACA Assignment? n
                                   Measured: none
                                                        Wideband Support? n
                            Internal Alert? n
                                                       Maintenance Tests? y
                           Data Restriction? n NCA-TSC Trunk Member:
Send Name: y Send Calling Number:
                                                  Send Calling Number: y
            Used for DCS? n
   Suppress # Outpulsing? n Format: public
Outgoing Channel ID Encoding: preferred UUI IE Treatment: service-provider
                                              Replace Restricted Numbers? n
                                             Replace Unavailable Numbers? n
                                                   Send Connected Number: n
Network Call Redirection: none
                                           Modify Tandem Calling Number? n
            Send UUI IE? y
              Send UCID? n
 Send Codeset 6/7 LAI IE? y
                SBS? n Network (Japan) Needs Connect Before Disconnect? n
```

In Page 6 of the configuration, add the trunk members, as shown below. The keyword "ip" is entered in the "Port" field, and the signaling group number "2" 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.

```
6 of 22
add trunk-group 2
                                                      Page
                                TRUNK GROUP
                                                                      1/6
                                    Administered Members (min/max):
GROUP MEMBER ASSIGNMENTS
                                        Total Administered Members:
                                                                      6
             Code Sfx Name
                                   Night
                                                   Sig Grp
      Port
 1: ip
                                                    2
 2: ip
                                                    2
                                                    2
 3: ip
                                                    2
 4: ip
 5: ip
                                                    2
  6: ip
                                                     2
```

After the trunk-group is added, use the "change signaling-group 2" command to enter the trunk group number "2" in the "Trunk Group for Channel Selection" field.

The command "save translation" must be entered to save the configuration changes.

4. Cisco CallManager Configuration

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

4.1. Add an H.323 Gateway

Start the CallManager Administration program and select "Add a New Device" from the "Device menu". Use the drop down menu to select "Gateway" as "Device type" and click "Next".



Figure 2: Add a New Device

Use the drop down menu to select "H.323 Gateway" as "Gateway type" and "H.225" as "Device Protocol". Click the "Next" button when done.

🚰 Cisco CallManager 4.0 Administration	- Add a New Gateway - Microsoft Internet Explore	r
<u>File E</u> dit <u>V</u> iew F <u>a</u> vorites <u>T</u> ools <u>H</u> e	lp	
🗢 Back 👻 🤿 🗸 🚳 🖓 Searc	:h 🔝 Favorites 🛞 Media 🎯 🖏 - 🎒	
Address 🙋 http://135.8.31.20/ccmadmin/gv	vmain.asp	
System Route Plan Service	Feature Device User Application Hel	p
Cisco CallManager For Cisco IP Telephony Solutions	Administration	CISCO SYSTEMS
Add a New Gate	eway you would like to create:	
Gateway type*	H.323 Gateway	•
Device Protocol*	H.225	
* indicates required item	Next	

Figure 3: Add a New Gateway on CallManager

After clicking "Next", enter the gateway configuration information as shown in **Figure 4**. The "Device Name" corresponds to the C-LAN IP address used in the signaling group definition on the Avaya S8700 Media Server. 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. By default, the system checks this check box to specify that this device should immediately retry a video call as an audio call (if it cannot connect as a video call). 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 check box to specify that Cisco CallManager needs to receive the far-end H.245 Terminal Capability Set before it sends its H.245 Terminal Capability Set. Unchecking this check box specifies that Cisco CallManager should initiate capabilities exchange. In these Application Notes, the "Media Termination Point Required", "Retry Video Call as Audio" and "Wait for Far End H.245 Terminal Capacity Set" may all be unchecked. Click "Insert" to add this gateway.

ile <u>E</u> dit <u>V</u> iew F <u>a</u> vorites	<u>I</u> ools <u>H</u> elp	
Idress http://mcs7835/ccmad	lmin/gatewayconfig.asp?pkid={937FD57D-B369-	4274-87E6-71CA421474E5}&Status=US&Action=Update&Typ 🔽 🤗
nks @ECARE @ECisco @EM	CS7835 🝘 POST	R. 4
System Route Plan Si	ervice Feature Device User App	olication Help
Cisco CallMan	ager Administration	CISCO SYSTEMS
For Cisco IP Telephony Soluti	ons	
Catoway Co	nfiguration	Back to Find/List Gateways
Galeway Co	ingulation	Dependency Records
	Product : H.323 Gateway	
	Gateway : 20.1.1.10	
	Registration: Unknown	
	IP Address: 20.1.1.10	
	Status: Update completed.	
	Update Delete Reset G	Sateway
	Device Information	
	Device Name*	20.1.1.10
	Description	S8700-g600-1
	Device Pool*	
	Media Resource Group List	<none></none>
	Location	< None >
	AAR Group	< None >
	Signaling Port*	1720
	🗖 Media Termination Point Re	quired
	🗖 Retry Video Call as Audio	
	🔲 Wait for Far End H.245 Ter	minal Capability Set
	Multilevel Precendence and F	Preemption (MLPP) Information
	MLPP Domain (e.g., "0000FF")	
	MLPP Indication	Not available on this device
	MLPP Preemption	NUL AVAIIADIE ON THIS GEVICE
	Call Routing Information	

Figure 4: Gateway Configuration

iks 🖨 CARE 🖉 Cisco 🖉 MCS	i7835 🔮 POST		(r. 5
⊨Back • → • 🙆 🗿 🖓	🔍 Search Favorites 🎯 Media 🧭	B- 3	
	Call Routing Information		
	Inbound Calls		
	Significant Digits*		
	Calling Search Space	<none></none>	
	AAR Calling Search Space	< None >	
	Prefix DN		
	🗹 Redirecting Number IE Deliv	very - Inbound	
	Outbound Calls		
	Calling Party Selection*	Originator	
	Calling Party Presentation*	Allowed	
	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		
	🗹 Display IE Delivery		
	🔽 Redirecting Number IE Deliv	very - Outbound	
	* indicates required item		
		Back to Find/List Gateway	<u>ys</u>

Figure 4(continued): Gateway Configuration

4.2. Configure Route-pattern on CallManager

The routing pattern is configured such that calls from Cisco IP phones to extension range 5xxxx are directed to the gateway 20.1.1.10, the IP address of the C-LAN in the Avaya G650 Media Gateway. The next screen shows the configuration.

Click "Route Plan" and select "Route Pattern/Hunt Pilot" as shown below.



Figure 5: Configure Route Patten

Enter "5XXXX" in the "Route Pattern/Hunt Pilot" field as shown below.

Cisco CallManager 4.0 Administration -	Route Pattern/Hunt Pilot Confi	guration - Microsoft Inte	rnet Explorer
<u>File E</u> dit <u>V</u> iew F <u>a</u> vorites <u>T</u> ools <u>H</u> elp			
🕁 Back 🔹 🤿 🚽 🙆 🛃 🛛 🐼 Search	📓 Favorites 🛛 🖓 Media	B- 3	
Address 🧃 http://135.8.31.20/ccmadmin/rout	epatternconfig.asp?pkid={C8CA691	8-84A0-440D-9CF0-18AD8A4	467E5}&status=uc
System Route Plan Service I	Feature Device User Ap	plication Help	
Cisco CallManager A For Cisco IP Telephony Solutions	Administration		CISCO SYSTEMS
Route Pattern/H Configuration	lunt Pilot	<u>Add a M</u> Back to Find/List Ro	<u>Jew Route Pattern/Hunt Pilot</u> ute Patterns and Hunt Pilots
Route Pattern/Hunt Pilot: 5X Status: Update completed Note: Any update to this Route Patter Copy Update Delete	XXX n or Hunt Pilot automatically n	esets the associated gate	way or Route/Hunt List
Pattern Definition	-		
Route Pattern/Hunt Pilot*	5xxxx		
Partition	< None >	•	
Description			
Numbering Plan*	North American Numbering	Plan 🔽	
Route Filter	< None >	•	
MLPP Precedence	Default	•	
Gateway or Route/Hunt List*	20.1.1.10	(Edit)	
Route Option	• Route this pattern		
	O Block this pattern	Not Selected —	
Provide Outside Dial Tone	Allow Overlap S	Sending 🗖	Urgent Priority
Calling Party Transformations	;		
Use Calling Party's Externa	Phone Number Mask		

Figure 6: Route Pattern/Hunt Pilot Configuration

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 Transformat	ons			
Connected Line ID Presentation	Allowed		•	
Connected Name Presentation	Allowed		•	
Called Party Transformations				
Discard Digits	PreDot		•	
Called Party Transform Mask				
Prefix Digits (Outgoing Calls)				
ISDN Network-Specific Faciliti	es Informatio	n Element		
Carrier Identification Code				
Network Service Protocol	— Not Selected	—	•	
Network Service	Sen	vice Parameter Name	Э	Service Parameter Value
- Not Selected -		ot Exist >		
* indicates required item.				

Figure 6(continued): Route Pattern/Hunt Pilot Configuration

Click "Update" to add this route pattern.

5. Verification Steps

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

• Make a phone call from the Avaya 4620 IP phone to the Cisco 7960 IP phone, and verify the voice quality is good and the IP trunk is used to carry this call. From the Avaya SAT, the command "status station 50001" displays the call signaling and audio information.

```
status station 50001
                                                    Page
                                                           3 of
                                                                  6
                               CALL CONTROL SIGNALING
                              Port: S00002
                 Switch
                                            ΙP
                                                                   ΙP
                 Port Switch-end IP Addr:Port Set-end IP Addr:Port
    IP Signaling: 01A0317 20.1.1.10 :1720
                                                 20.1.1.120:4069
           н.245:
       Node Name:
                          C-lan
  Network Region:
                          1
                                                    Page
                                                           4 of
                                                                  б
                                AUDIO CHANNEL
                                Port: S00002
                  Switch
                                             ΤP
                                                                    ΤP
                  Port Other-end IP Addr :Port
                                                   Set-end IP Addr:Port
G.711MU
           Audio:
                         135.8.31.100:32326
                                                   20.1.1.120:29798
       Node Name:
  Network Region:
                          1
                                                    1
  Audio Connection Type: ip-direct
```

• Make a phone call from the Cisco 7960 IP phone to the Avaya 6400 digital phone, and verify the call quality is good and the IP trunk is used to carry this call. The display of trunk group 2 status from the S8700 Media Server showed that voice channel 1 is in service and active.

```
status trunk 2
TRUNK GROUP STATUS
Member
                 Service State
                                   Mtce
        Port
                                           Connected Ports
                                   Busy
0002/001 T00024
                 in-service/active no
                                           S00002
0002/002 T00025 in-service/idle
                                   no
0002/003 T00032 in-service/idle
                                   no
0002/004 T00033 in-service/idle
                                   no
0002/005 T00034 in-service/idle
                                   no
0002/006 T00044 in-service/idle
                                   no
0002/007 T00045
                 in-service/idle
                                   no
0002/008 T00046 in-service/idle
                                   no
0002/009 T00047 in-service/idle
                                   no
```

- Make a phone call from the Cisco 7960 IP phone to the Avaya analog phone, and verify the voice quality is good.
- 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 the name of the connected party, when sent by the Cisco CallManager.
 - 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 of the connected party sent by the Avaya Media Server, and the dialed number (i.e., which may be different from the connected number. These Application Notes recommend that the Avaya Media Servers be provisioned to refrain from sending the Connected Number. Cisco CallManager would not display the Connected Number, if sent by Avaya).

6. Conclusion

As illustrated in these Application Notes, the Avaya S8700 Media Server and Avaya G650 Media Gateway can interoperate with Cisco CallManager using an H.323 IP trunk. Final media paths for calls can be "ip-direct" between Cisco IP telephones and Avaya IP Telephones. The calling party name and number can be displayed for calls in both directions.

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