



Tech Note: Configuring H.323 IP trunk between Cisco Call Manager and Avaya S8700/G600

A decorative graphic on the left side of the page, consisting of a blue square with white curved lines forming a globe-like pattern, and a vertical black line extending downwards from the bottom of the square.

Introduction

The objective of this document is to provide the Cisco field engineers or Cisco partners with exact steps to configure H.323 IP trunks between the Cisco Call Manager and the Avaya S8700/G600. This is particularly important for situations where interoperability is required. It is recommended to use the Avaya Site Administration (ASA) tool for configuration tasks on the Avaya S8700/G600. This interoperability document is intended for external use as well. There is no Cisco Confidential information within this document.

Procedure on Avaya S8700/G600

1. The first step is to check whether the Avaya system is capable of supporting IP trunks. Unlike the Cisco Call Manager, this is a licensed feature. This can be obtained by running the “*display system-parameters customer*” command from the ASA tool. The screenshot on the next page shows the output captured from our lab setup.



```
display system-parameters custo | send (return) | help (f5) | cancel (esc) | enter (f3) | schedule (f9) | next (f7) | previous (f8)
1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 | 9 |
OPTIONAL FEATURES
G3 Version: U11                      Maximum Ports: 125  35
Location: 1                          Maximum XMOBILE Stations: 10  0
Platform: 8
IP PORT CAPACITIES
Maximum Administered IP Trunks: 25  17
Maximum Concurrently Registered IP Stations: 100  5
Maximum Administered Remote Office Trunks: 0  0
Maximum Concurrently Registered Remote Office Stations: 0  0
Maximum Concurrently Registered IP eCons: 0  0
Maximum Number of DS1 Boards with Echo Cancellation: 400  0
Maximum TN2501 UAL Boards: 1  0
Maximum G700 UAL Sources: 10  0
(NOTE: You must logoff & login to effect the permission changes.)
```

2. The next step is to configure a trunk group. Type “*add trunk-group #*” where # is the desired trunk group.

```
change trunk-group 1 | send (return) | help (f5) | cancel (esc) | enter (f3) | schedule (f9) | next (f7) | previous (f8)
1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 | 9 | 10 | 11 | 12 | 13 | 14 | 15 | 16 | 17 | 18 | 19 | 20 | 21 | 22 |
TRUNK GROUP
Group Number: 1                      Group Type: isdn      CDR Reports: y
Group Name: IP TRUNKS TO CALLMANAGER  COR: 1              TN: 1              TAC: 81
Direction: two-way                   Outgoing Display? y  Carrier Medium: IP
Dial Access? y                       Busy Threshold: 255  Night Service:
Queue Length: 0
Service Type: tie                     Auth Code? n        TestCall ITC: rest
Far End Test Line No:
TestCall BCC: 0
TRUNK PARAMETERS
Codeset to Send Display: 6           Codeset to Send National IEs: 6
Max Message Size to Send: 260       Charge Advice: none
Supplementary Service Protocol: a    Digit Handling (in/out): enbloc/enbloc
Trunk Hunt: cyclical
Calling Number - Delete:             Insert:              Digital Loss Group: 18
Bit Rate: 1200                       Synchronization: async Duplex: full
Disconnect Supervision - In? y Out? n
Answer Supervision Timeout: 0
```



- Then add the IP address of the remote end (which in this case is the Call Manager) by typing the “*change node-names ip*” command. Create a signaling group, specify the remote end and then associate it with the trunk group previously created.

```
change signaling-group 10 | send (return) | help (f5) | cancel (esc) | enter (f3) | schedule (f9) | next (f7) | previous (f8) |
1 | 2 | 3 | 4 | 5 |
SIGNALING GROUP
Group Number: 10      Group Type: h.323
Remote Office? [n]    Max number of NCA TSC: 0
SBS? [n]              Max number of CA TSC: 0
Trunk Group for Channel Selection: 1  Trunk Group for NCA TSC:
Supplementary Service Protocol: a

Near-end Node Name: ClanIP  Far-end Node Name: CallManager
Near-end Listen Port: 1720  Far-end Listen Port: 1720
Far-end Network Region: 1

LRQ Required? [n]      Calls Share IP Signaling Connection? [n]
RRQ Required? [n]
Media Encryption? [n]  Bypass If IP Threshold Exceeded? [n]

DTMF over IP: out-of-band  Direct IP-IP Audio Connections? [y]
IP Audio Hairpinning? [y]
Interworking Message: PROGRESS
```

- Now that the trunk and signaling groups have been successfully created, the next step is to take care of the routing patterns. In this lab setup, the Avaya IP phones are in the extension range 2XXX while the Call Manager IP phones are in the extension range 4XXX. So we need to create a dialplan to route to 4XXX from the Avaya system. For the purposes of our lab setup, we are inserting digits 443 while dialing the 4 digit extension and then stripping them out (Note: This is not necessary. You can also have a simple setup which does not require any digit manipulation). Check the uniform dial plan table by typing “*change uniform #*” where # is the matching pattern.



- Next, add the CLAN addresses to a route group, and associate the route group to a route list

Route/Hunt List Configuration

[Add a new Route/Hunt List](#)
[Back to Find/List Route/Hunt Lists](#)
[Dependency Records](#)

Route/Hunt List Details

Avaya S8700 Xeta

Route/Hunt List: S8700 Avaya Xeta

Status: Ready

Copy Update Delete Reset

Route/Hunt List Information

Route/Hunt List Name* S8700 Avaya Xeta

Description

Cisco CallManager Group* Default

Enable this Route/Hunt List (change effective on Update; no reset required)

Route/Hunt List Member Information

Add Route Group Add Line Group

Selected Groups* (ordered by highest priority)

Avaya S8700 Xeta[non-QSIG]

- Next step is to point the route pattern 2XXX to the route list created

Pattern Definition

Route Pattern/Hunt Pilot* 2XXX

Partition < None >

Description

Numbering Plan* North American Numbering Plan

Route Filter < None >

MLPP Precedence Default

Gateway or Route/Hunt List* S8700 Avaya Xeta (Edit)

Route Option

Route this pattern

Block this pattern — Not Selected —

Provide Outside Dial Tone Allow Overlap Sending Urgent

- For supplementary services features such as conference calls, enable Media Termination Point (MTP) resources.



Media Resource Group List	< None >
Location	< None >
AAR Group	< None >
Signaling Port*	1720
<input checked="" type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input checked="" type="checkbox"/> Wait for Far End H.245 Terminal Capability Set	
Multilevel Precedence and Preemption (MLPP) Information	
MLPP Domain (e.g., "0000FF")	
MLPP Indication	Not available on this device
MLPP Preemption	Not available on this device
Call Routing Information	
Inbound Calls	
Significant Digits*	All
Calling Search Space	< None >
AAR Calling Search Space	< None >

5. To allow Caller ID and Caller Name Display, enable Display IE Delivery.

Prefix DN	
<input type="checkbox"/> Redirecting Number IE Delivery - 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	
<input checked="" type="checkbox"/> Display IE Delivery	
<input type="checkbox"/> Redirecting Number IE Delivery - Outbound	
* indicates required item	
Back to Find/List Gateways	



Features Tested

The following are the lists of features tested between the Cisco Call Manager 4.0 and Avaya S8700/G600 Communication Manager 1.3:

Name and Number Display (Bi-directional)

Call Transfer

Conference Call between the two systems

Call Park