Tech Note: Configuring Q.SIG PRI trunk between Cisco Call Manager and Avaya S8700/G650 with Cisco Unity Voice Mail integration

Introduction

The objective of this document is to provide Cisco’s customers and business partners with exact steps to configure Q.SIG PRI trunks between the Cisco Call Manager and the Avaya S8700/G650. Also, it details steps on how to add Cisco Unity on the Cisco Call Manager platform to provide voice mail support for both Cisco and Avaya IP phones. This is particularly important for situations where IP-PBX interoperability and voice mail integration are required. The Avaya configuration screen captures have been done using the standard Emulation tool. As an alternative, the user can also use the Avaya Site Administration (ASA) tool for configuration tasks on the Avaya S8700/G650. The output display is the same in both cases. This IP-PBX interoperability and voice mail integration document is intended for external use.

Test Setup

The Avaya IP-PBX system used was the Avaya S8700/G650 running Avaya Communication Manager 2.0. The Q.SIG feature set comes standard with this software version. The AVAYA IP Phones used were the 4610SW and 4620 running Phone Firmware Version 2.01. On the Cisco side, Cisco Call Manager 4.1.2 was used to control the 3745 MGCP gateway with the NM-HDV module, running IOS version 12.2.15ZJ3. Tests were also repeated with IOS version 12.3.8.T5. Cisco Unity running version 4.0(4) SR1 was used for the voice mail integration testing.
Test Topology

Q.SIG PRI trunk between Cisco Call Manager and Avaya S8700/G650

with Cisco Unity Voice Mail integration
The next sections provide procedures and screen captures of how to configure the Q.SIG trunk between an Avaya S8700/G650 running Avaya Communication Manager 2.0 and a Cisco Call Manager platform running Call Manager version 4.1(2) with the Cisco 3745 MGCP device providing the physical ISDN PRI connection to the Avaya S8700/G650.

**Interoperability between Cisco and Avaya IP-PBX Systems**

1. Login to the S8700 server. Make sure that all the necessary Q.SIG features are enabled on the S8700 server by running the “display system-parameters customer” feature.

```
display system-parameters customer-options
```

```
QSIG OPTIONAL FEATURES
Basic Call Setup? y
Basic Supplementary Services? y
Centralized Attendant? y
Interworking with DGS? y
Supplementary Services with Rerouting? y
Transfer into QSIG Voice Mail? y
Value-Added (VALU)? y

(NOTE: You must logoff & login to effect the permission changes.)
```

2. Configure the DS-1 card for Q.SIG PRI
3. The next step is to configure a trunk group. Type "**add trunk-group #**" where # is the desired trunk group. The next 3 screen captures relate to the trunk configuration. Once the trunk group is created, add the 23 DS0 channels to the group. The following is an example of the port assignment: 01A0901 would mean: Gateway# 1, Cabinet A, Slot# 9, DS0 channel# 1.
Trunk Group

Group Number: 1  Group Type: isdn  CDR Reports: n
Group Name: QSIG TRUNKING  COR: 90  TH: 1  TAC: *01
Direction: two-way  Outgoing Display? y  Carrier Medium: PRI/BRI
Dial Access? y  Busy Threshold: 99  Night Service:
Queue Length: 0  Auth Code? n  TestCall ITC: rest
Service Type: tie  Far End Test Line No:

TestCall BCC: 4

Trunk Parameters

Codeset to Send Display: 0  Codeset to Send National IEs: 6
Max Message Size to Send: 260
Supplementary Service Protocol: b  Digit Handling (in/out): embloc/embloc
Trunk Hunt: ascend  QSIG Value-Added? y
Digital Loss Group: 13
Calling Number - Delete: Insert:
Bit Rate: 1200  Synchronization: async  Duplex: Full
Disconnect Supervision - In? y  Out? y
Answer Supervision Timeout: 0

Trunk Features

ACA Assignment? n  Measured: internal  Wideband Support? n
Internal Alert? n  Maintenance Tests? y
Data Restriction? n  NCA-TSC Trunk Member: 10
Send Name: y  Send Calling Number: y
Send Dstg? y  Used for DCS? n
Suppress # Outpulsing? n  Numbering Format: public
Outgoing Channel ID Encoding: exclusive  UUI IE Treatment: service-provider

Replace Restricted Numbers? n  Replace Unavailable Numbers? n
Send Called/Busy/Connected Number: y
Send UUI IE? y  Send Dstg? y
Send Codeset 6/7 LAI? y  Dsl Echo Cancellation? n
Path Replacement with Retention? y

SBS? n  Network (Japan) Needs Connect Before Disconnect? y
4. Add signaling group and point to the trunk group created above.
5. Next, add a route pattern and point it to the signaling group. In this example, the route pattern 4 points to signaling group# 1 that was created in the previous step.

<table>
<thead>
<tr>
<th>Grp</th>
<th>FRL</th>
<th>NPA</th>
<th>Pfx</th>
<th>Hop Toll</th>
<th>No.</th>
<th>Inserted</th>
<th>Secure SIP?</th>
<th>DCS/ IXC</th>
<th>QSIG</th>
<th>Intw</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>1</td>
<td>0</td>
<td>408</td>
<td>4</td>
<td></td>
<td>D</td>
<td>n</td>
<td>n</td>
<td>user</td>
<td></td>
</tr>
<tr>
<td>2</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>n</td>
<td>n</td>
<td>user</td>
<td></td>
</tr>
<tr>
<td>3</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>n</td>
<td>n</td>
<td>user</td>
<td></td>
</tr>
<tr>
<td>4</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>n</td>
<td>n</td>
<td>user</td>
<td></td>
</tr>
<tr>
<td>5</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>n</td>
<td>n</td>
<td>user</td>
<td></td>
</tr>
<tr>
<td>6</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>n</td>
<td>n</td>
<td>user</td>
<td></td>
</tr>
</tbody>
</table>

6. Finally add an entry into the AAR table to use the route pattern above to route calls. In this example, calls to Cisco IP phone extension 4XXX are using the AAR table entry starting with 4, which in turn points to route pattern# 4.
7. Last step is to ensure caller id is enabled on each IP phone to send calling party name.
Procedure on Cisco Call Manager

1. Under Service parameters, make sure that the Start Path Replacement Minimum and Maximum time values are set appropriately to prevent any issues (such as hair pinning). The next two screen captures relate to the Q.SIG Service Parameters setting:

<table>
<thead>
<tr>
<th>Parameter Name</th>
<th>Parameter Value</th>
<th>Suggested Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Path Replacement Enabled*</td>
<td>True</td>
<td>False</td>
</tr>
<tr>
<td>Path Replacement on Tromboned Calls*</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>Start Path Replacement Minimum Delay Time (sec)*</td>
<td>5</td>
<td>0</td>
</tr>
<tr>
<td>Start Path Replacement Maximum Delay Time (sec)*</td>
<td>10</td>
<td>0</td>
</tr>
<tr>
<td>Path Replacement T1 Timer (sec)</td>
<td>30</td>
<td>30</td>
</tr>
<tr>
<td>Path Replacement T2 Timer (sec)</td>
<td>16</td>
<td>15</td>
</tr>
</tbody>
</table>
2. Add Cisco 3745 as an MGCP gateway and configure the NM-HDV T-1 module for Q.SIG PRI. The next 5 screen captures relate to this configuration.
<table>
<thead>
<tr>
<th>Installed Voice Interface Cards</th>
<th>Endpoint Identifiers</th>
</tr>
</thead>
<tbody>
<tr>
<td>Mainboard Slot</td>
<td></td>
</tr>
<tr>
<td>Module in Slot 1</td>
<td>NM-HDV</td>
</tr>
<tr>
<td>Subunit</td>
<td>VIC-2MFT-T1</td>
</tr>
<tr>
<td>Module in Slot 2</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Module in Slot 3</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Module in Slot 4</td>
<td>NW-2V</td>
</tr>
<tr>
<td>Subunit 0</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Subunit 1</td>
<td>VIC-2FXS</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Product Specific Configuration</th>
</tr>
</thead>
<tbody>
<tr>
<td>Global ISDN Switch Type</td>
</tr>
<tr>
<td>Switchback Timing</td>
</tr>
<tr>
<td>Switchback uptime-delay (min)</td>
</tr>
<tr>
<td>Switchback schedule (hh:mm)</td>
</tr>
</tbody>
</table>
## Device Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>End-Point Name</td>
<td>S1/DS1-8@CCME_CUE_3745</td>
</tr>
<tr>
<td>Description</td>
<td>S1/DS1-8@CCME_CUE_3745</td>
</tr>
<tr>
<td>Device Pool</td>
<td>Default</td>
</tr>
<tr>
<td>Call Classification*</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Network Locale</td>
<td>United States</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Location</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>AAR Group</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Load Information</td>
<td></td>
</tr>
<tr>
<td>V150 (subset)</td>
<td></td>
</tr>
</tbody>
</table>

## Multilevel Precendence and Preemption (MLPP) Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>MLPP Domain (e.g., &quot;0000FF&quot;)</td>
<td></td>
</tr>
<tr>
<td>MLPP Indication</td>
<td>Off</td>
</tr>
<tr>
<td>MLPP Preemption</td>
<td>Disabled</td>
</tr>
</tbody>
</table>
### Interface Information

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>PRI Protocol Type*</td>
<td>PRI QSIG T1</td>
</tr>
<tr>
<td>Protocol Side*</td>
<td>User</td>
</tr>
<tr>
<td>Channel Selection Order*</td>
<td>Top Down</td>
</tr>
<tr>
<td>Channel IE Type*</td>
<td>Use Numberwhen1B</td>
</tr>
<tr>
<td>PCM Type*</td>
<td>μ-law</td>
</tr>
<tr>
<td>Delay for first restart (1/9 sec ticks)</td>
<td>32</td>
</tr>
<tr>
<td>Delay between restarts (1/8 sec ticks)</td>
<td>4</td>
</tr>
<tr>
<td>Inhibit restarts at PRI initialization</td>
<td>✔</td>
</tr>
<tr>
<td>Enable status poll</td>
<td>☐</td>
</tr>
</tbody>
</table>

### Call Routing Information

#### Inbound Calls

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Significant Digits*</td>
<td>All</td>
</tr>
<tr>
<td>Calling Search Space</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>AAR Calling Search Space</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Prefix DN</td>
<td></td>
</tr>
</tbody>
</table>

#### Outbound Calls

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calling Line ID Presentation*</td>
<td>Allowed</td>
</tr>
<tr>
<td>Calling Party Selection*</td>
<td>Originator</td>
</tr>
<tr>
<td>Called party IE number type</td>
<td>National</td>
</tr>
<tr>
<td>Called party IE number type unknown*</td>
<td>National</td>
</tr>
<tr>
<td>-----------------------------------------</td>
<td>---------------------------</td>
</tr>
<tr>
<td>Calling party IE number type unknown*</td>
<td>National</td>
</tr>
<tr>
<td>Called Numbering Plan*</td>
<td>ISDN</td>
</tr>
<tr>
<td>Calling Numbering Plan*</td>
<td>ISDN</td>
</tr>
<tr>
<td>Number of digits to strip*</td>
<td>0</td>
</tr>
<tr>
<td>Caller ID DN</td>
<td></td>
</tr>
<tr>
<td>SMDI Base Port*</td>
<td>0</td>
</tr>
</tbody>
</table>

**PRI Protocol Type Specific Information**

- Display IE Delivery
- Redirecting Number IE Delivery - Outbound
- Redirecting Number IE Delivery - Inbound
- Send Extra Leading Character In DisplayIE***
- Setup non-ISDN Progress Indicator IE Enable****
- MCDN Channel Number Extension Bit Set to Zero**
- Send Calling Name In Facility IE
- Interface Identifier Present**

Interface Identifier Value** | 0 |

Connected Line ID Presentation (QSIG Inbound Call)* | Allowed |
3. As a final step, create a Call Manager pickup group to provide path proposal extension to the PBX. Make sure that the call pickup number is also entered into the Path PINX Replacement ID Service parameter (Look at Step# 1). Also, the Avaya system needs a route pattern to route to the pickup group.
Cisco 3745 Configuration

Included below is the show version and show running-configuration on the Cisco 3745 MGCP device. Controller T1 1/0 on the Cisco 3745 is connected to the Avaya S8700/G650 DS1 PRI card. Q.SIG signaling is configured on PRI link between the Cisco 3745 and the Avaya S8700/G650.

CCME_CUE_3745#sh vers

Cisco Internetwork Operating System Software

IOS (tm) 3700 Software (C3745-IS-M), Version 12.2(15)ZJ3, EARLY DEPLOYMENT RELEASE SOFTWARE (fc2)

TAC Support: http://www.cisco.com/tac

Copyright (c) 1986-2003 by cisco Systems, Inc.

Compiled Thu 25-Sep-03 22:25 by eaarmas

Image text-base: 0x60008954, data-base: 0x61C2C000

ROM: System Bootstrap, Version 12.2(8r)T2, RELEASE SOFTWARE (fc1)
ROM: 3700 Software (C3745-IS-M), Version 12.2(15)ZJ3, EARLY DEPLOYMENT RELEASE SOFTWARE (fc2)

CCME_CUE_3745 uptime is 39 minutes
System returned to ROM by reload
System image file is "flash:c3745-is-mz.122-15.ZJ3.bin"

cisco 3745 (R7000) processor (revision 2.0) with 246784K/15360K bytes of memory.
Processor board ID JMX0814L3E2
R7000 CPU at 350Mhz, Implementation 39, Rev 3.3, 256KB L2, 2048KB L3 Cache
Bridging software.
X.25 software, Version 3.0.0.
SuperLAT software (copyright 1990 by Meridian Technology Corp).
Primary Rate ISDN software, Version 1.1.
2 FastEthernet/IEEE 802.3 interface(s)
25 Serial network interface(s)
1 terminal line(s)
2 Channelized T1/PRI port(s)
1 ATM AIM(s)
2 Voice FXS interface(s)
2 Voice E & M interface(s)
1 cisco service engine(s)
DRAM configuration is 64 bits wide with parity disabled.
151K bytes of non-volatile configuration memory.
125184K bytes of ATA System CompactFlash (Read/Write)
Configuration register is 0x2102

CCME_CUE_3745#sh run
Building configuration...

Current configuration : 3291 bytes
!
version 12.2
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname CCME_CUE_3745
!
logging queue-limit 100
!
voice-card 1
dspfarm
!
voice-card 5
dspfarm
!
ip subnet-zero
!

no ip domain lookup
!

isdn switch-type primary-qsig
! no voice hpi capture buffer
no voice hpi capture destination
!
!
ccm-manager mgcp
ccm-manager music-on-hold
ccm-manager config server 172.28.221.18
ccm-manager config
mta receive maximum-recipients 0
!
!
controller T1 1/0
framing esf
linecode b8zs
pri-group timeslots 1-24 service mgcp
!
controller T1 1/1
framing sf
linecode ami
!
!
!
interface FastEthernet0/0
description CCME-CUE-3745_to_cat3550
no ip address
duplex auto
speed auto
!  
interface FastEthernet0/0.1  
  encapsulation dot1Q 99  
!  
interface FastEthernet0/0.2  
  description NEW_S8700_G650  
  encapsulation dot1Q 300  
  ip address 172.28.221.49 255.255.255.240  
  ip helper-address 172.28.221.19  
  h323-gateway voip bind srcaddr 172.28.221.49  
!  
interface FastEthernet0/0.3  
  description MODULAR_MESSAGING_SOLUTION  
  encapsulation dot1Q 900  
  ip address 172.28.221.129 255.255.255.240  
  ip helper-address 172.28.221.19  
!  
interface FastEthernet0/0.4  
  encapsulation dot1Q 301  
  ip address 10.1.3.1 255.255.255.128  
  ip helper-address 172.28.221.19  
!  
interface FastEthernet0/0.5  
  encapsulation dot1Q 302  
  ip address 10.1.3.129 255.255.255.128  
  ip helper-address 172.28.221.19  
!  
interface FastEthernet0/0.6
encapsulation dot1Q 90
ip address 90.1.1.254 255.255.255.0
ip helper-address 172.28.221.19
!
interface Serial0/0
description CCME-CUE-3745_to_3600
ip address 25.0.0.1 255.0.0.0
clockrate 256000
no fair-queue
!
interface Serial1/0:23
no ip address
no logging event link-status
isdn switch-type primary-qsig
isdn incoming-voice voice
isdn bind-l3 ccm-manager
isdn bchan-number-order ascending
no cdp enable
!
interface Service-Engine2/0
no ip address
shutdown
!
router eigrp 100
network 10.0.0.0
network 25.0.0.0
network 90.0.0.0
network 172.28.0.0
auto-summary
!
ip http server
ip classless
!
call rsvp-sync
!
voice-port 1/0:23
!
voice-port 4/0/0
!
voice-port 4/0/1
!
voice-port 4/1/0
!
voice-port 4/1/1
!
mgcp
mgcp call-agent 172.28.221.18 2427 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode out-of-band
mgcp rtp unreachable timeout 1000 action notify
mgcp package-capability rtp-package
no mgcp package-capability res-package
mgcp package-capability sst-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax t38 inhibit
mgcp rtp payload-type g726r16 static
mgcp profile default

dial-peer cor custom

dial-peer voice 1 pots
  application mgcpapp
  port 1/0:23


dial-peer voice 999410 pots
  application mgcpapp
  port 4/1/0


line con 0
  password cisco
  login

line 65
  flush-at-activation
  no activation-character
  no exec
  transport preferred none
  transport input all

line aux 0
line vty 0 4
  password cisco
The following are the lists of features tested between the Cisco Call Manager 4.1(2) platform and the Avaya S8700/G650 running Communication Manager 2.0 via the Q.SIG PRI trunk:

Name and Number Display (Bi-directional)
Call Transfer
Conference Call between the two systems

Integration of Cisco Unity Voice Mail to support Cisco and Avaya IP Phones

At this point, one can make calls via the Q.SIG trunk between an Avaya S8700/G650 running Avaya Communication Manager 2.0 and a Cisco Call Manager platform running Call Manager version 4.1(2) with the Cisco 3745 MGCP device providing the physical ISDN PRI connection to the Avaya S8700/G650. A Cisco Unity server can be added on the Cisco Call Manager platform to provide voice mail support to both the Cisco and Avaya IP phones. To do this, the administrator only needs to configure the Cisco Unity on the Cisco Call Manager platform. Included below are the procedures with screen captures of how to configure Cisco Unity on the Cisco Call Manager Administration management page. Note most of the configuration is performed on the Cisco Voice Mail Port Wizard.

Procedure for adding Cisco Unity to Cisco Call Manager

1. Under Feature, select Voice Mail, Voice Mail Port Wizard. Select Create a new voice mail server and add ports to it and click Next.
2. Enter a Cisco Voice Mail Server name, such as AvayaUM3, and click Next.
3. Select the Voice Mail Ports you want configured and click Next.
4. Enter a Description and Device Pool for the Voice Mail Ports. In our configuration we entered Avaya VMailPorts as the description and Default as the device pool.
5. Enter the Beginning Directory Number, such as 4406, and the Display, such as Voicemail and click Next.
6. The next screen will ask Do you want to add these directory numbers to a Line Group? Select Yes. Add directory numbers to a new Line Group and click Next.
7. Enter a Line Group Name which matches the Voice Mail Server you previously entered, such as AvayaUM3.
8. The next screen shows the configuration entered so far. Click Finish if there are no changes to the configuration.
9. Click Add a New Hunt List on the Hunt List Administration web page.

|Cisco Voice Mail Device Information (apply to all ports) |
|---|---|
|Number of Ports to Add| 2 (adding ports 1 - 2)|
|Cisco Voice Mail Server Name| AvayaUM3|
|Description| Avaya VMMailPorts|
|Device Pool| Default|
|Calling Search Space| < None >|
|AAR Calling Search Space| < None >|
|Location| < None >|
|Device Security Mode| Non Secure|

|Directory Number Information |
|---|---|
|Pilot Directory Number| 4406|
|New Directory Numbers| 4406 - 4407|
|Partition| < None >|
|Calling Search Space| < None >|
|Display| Voicemail|
|AAR Group| < None >|
|External Number Mask| < None >|
|Line Group| AvayaUM3|
10. Enter a Hunt List Name and Description, such as Avaya VMailHL. Also select Default for the Cisco Call Manager Group.

11. The following screen capture is the result of successfully adding the Hunt List. Click Add Line Group.
12. Select the Line Group previously configured. In our case, it’s AvayaUM3.
13. The next screen capture shows the result of successfully inserting the line group.
15. Enter in the Hunt Pilot, such as 4408, and select a Hunt List, such as Avaya VMail HL and click Insert.
17. Enter the Voice Mail Pilot number matching the Hunt Pilot number previously configured. In our case, both the Hunt Pilot and Voice Mail Pilot numbers are 4408.
18. Next, go to Feature, Voice Mail, Voice Mail Profile and click Add a New Voice Mail Profile.
19. Enter the Voice Mail Profile Name and Description, such as AvayaVMailProfile and select the Voice Mail Pilot number in step 17. In our case the Voice Mail Pilot number is 4408.
20. Click Features, Voice Mail, Message Waiting Indicator, Add a New Message Waiting Number to add the Message Waiting Indicator On/Off numbers. Included below are two screen captures for Message Waiting Indicator On/Off numbers.
### Message Waiting Configuration

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Message Waiting Number</td>
<td>1001</td>
</tr>
<tr>
<td>Status: Status:</td>
<td>Ready</td>
</tr>
<tr>
<td>Copy</td>
<td></td>
</tr>
<tr>
<td>Update</td>
<td></td>
</tr>
<tr>
<td>Delete</td>
<td></td>
</tr>
<tr>
<td>Description</td>
<td></td>
</tr>
<tr>
<td>Message Waiting Indicator</td>
<td>On</td>
</tr>
<tr>
<td>Partition</td>
<td>None</td>
</tr>
<tr>
<td>Calling Search Space</td>
<td>None</td>
</tr>
</tbody>
</table>

*Indicates required item*
**Cisco Unity Voice Mail Features Tested**

The following are the lists of Cisco Unity Voice Mail features tested using the Avaya IP phones to access Cisco Unity Voice Mail via the Q.SIG PRI trunk between the Cisco Call Manager 4.1(2) platform and the Avaya S8700/G650 running Communication Manager 2.0:

- Internal greeting
- Busy greeting
- MWI
- Easy message access

**Conclusion**

This document has been created to provide Cisco’s customers or business partners with exact steps to configure Q.SIG PRI trunks between the Cisco Call Manager and the Avaya S8700/G650. Also, it details steps on how to add Cisco Unity on the Cisco Call Manager platform to provide voice mail support for both Cisco and Avaya IP phones.