



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.

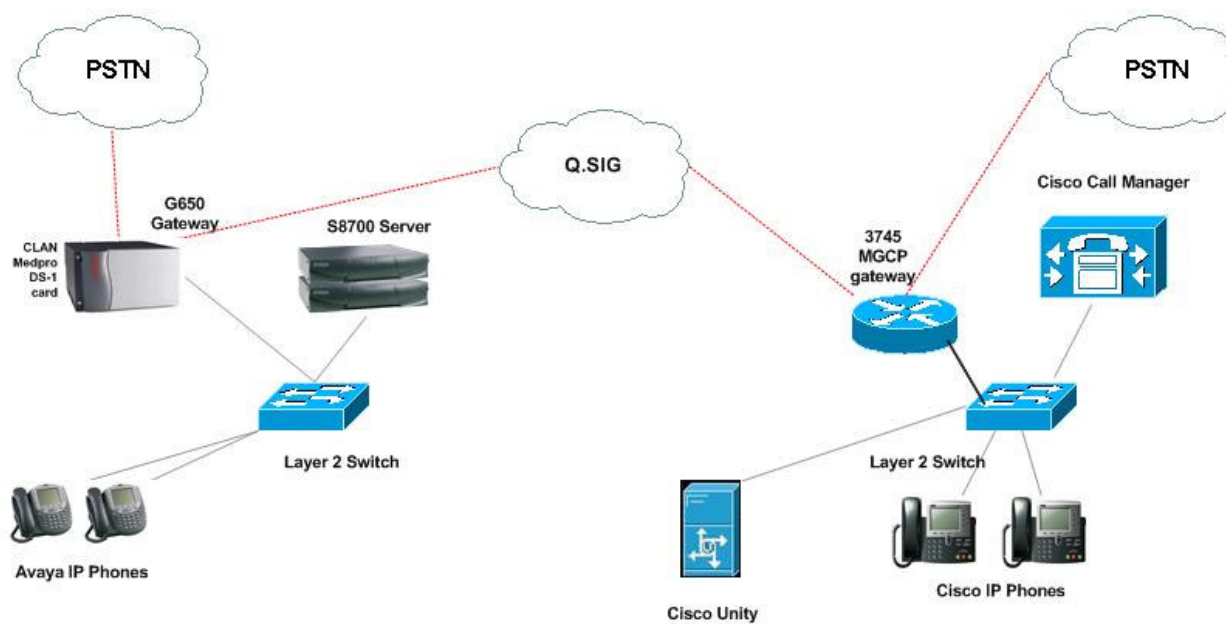
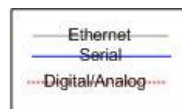
Test Setup

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**





Interoperability between Cisco and Avaya IP-PBX Systems

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.

Procedure on Avaya S8700/G650

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.

```
cancel  refresh  enter  clear  help  go to page  next page  prev page
display system-parameters customer-options Page 8 of 11
      QSIG OPTIONAL FEATURES
                Basic Call Setup? y
      Basic Supplementary Services? y
                Centralized Attendant? y
                Interworking with DCS? 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



```
cancel | refresh | enter | clear | help | go to page | next page | prev page
display ds1 01A09 Page 1 of 2
DS1 CIRCUIT PACK
Location: 01A09 Name: QSIG
Bit Rate: 1.544 Line Coding: b8zs
Line Compensation: 1 Framing Mode: esf
Signaling Mode: isdn-pri
Connect: pbx Interface: peer-master
TN-C7 Long Timers? n Peer Protocol: Q-SIG
Interworking Message: PROGRESS Side: a
Interface Companding: mulaw CRC? n
Idle Code: 11111111
DCP/Analog Bearer Capability: 3.1kHz

Slip Detection? n Near-end CSU Type: other
Echo Cancellation? n
```

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.



```
cancel refresh enter clear help go to page next page prev page
display trunk-group 1 Page 1 of 22
TRUNK GROUP
Group Number: 1 Group Type: isdn CDR Reports: n
Group Name: QSIG TRUNKING COR: 90 TN: 1 TAC: *01
Direction: two-way Outgoing Display? y Carrier Medium: PRI/BRI
Dial Access? y Busy Threshold: 99 Night Service:
Queue Length: 0
Service Type: tie Auth Code? n TestCall ITC: rest
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): enbloc/enbloc
Trunk Hunt: ascend QSIG Value-Added? y
Digital Loss Group: 13
Calling Number - Delete: Insert: Numbering Format: pub-unk
Bit Rate: 1200 Synchronization: async Duplex: full
Disconnect Supervision - In? y Out? y
Answer Supervision Timeout: 0
```

```
display trunk-group 1 Page 2 of 22
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
Used for DCS? n Hop Dgt? y
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 UCID? y
Send Codeset 6/7 LAI IE? y Ds1 Echo Cancellation? n
Path Replacement with Retention? y
SBS? n Network (Japan) Needs Connect Before Disconnect? y
```



```
display trunk-group 1                                     Page 6 of 22
TRUNK GROUP
Administered Members (min/max): 1/23
Total Administered Members: 23
GROUP MEMBER ASSIGNMENTS
Port      Code Sfx Name      Night      Sig Grp
1: 01A0901 TN464 G          Night      1
2: 01A0902 TN464 G          Night      1
3: 01A0903 TN464 G          Night      1
4: 01A0904 TN464 G          Night      1
5: 01A0905 TN464 G          Night      1
6: 01A0906 TN464 G          Night      1
7: 01A0907 TN464 G          Night      1
8: 01A0908 TN464 G          Night      1
9: 01A0909 TN464 G          Night      1
10: 01A0910 TN464 G          Night      1
11: 01A0911 TN464 G          Night      1
12: 01A0912 TN464 G          Night      1
13: 01A0913 TN464 G          Night      1
14: 01A0914 TN464 G          Night      1
15: 01A0915 TN464 G          Night      1
```

4. Add signaling group and point to the trunk group created above.

```
display signaling-group 1
SIGNALING GROUP
Group Number: 1
Group Type: isdn-pri
Associated Signaling? y
Primary D-Channel: 01A0924
Max number of NCA TSC: 10
Max number of CA TSC: 10
Trunk Group for NCA TSC: 1
Trunk Group for Channel Selection: 1
X-Mobility/Wireless Type: NONE
Supplementary Service Protocol: b
Network Call Transfer? n
Command: 
```




- 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.

```
cancel refresh enter clear help go to page next page prev page
display route-pattern 4 Page 1 of 3
Pattern Number: 4 Pattern Name: isdn test
Secure SIP? n
Grp FRL NPA Pfx Hop Toll No. Inserted DCS/ IXC
No Mrk Lmt List Del Digits QSIG
Dgts Intw
1: 1 0 408 4 n user
2: n user
3: n user
4: n user
5: n user
6: n user

BCC VALUE TSC CA-TSC ITC BCIE Service/Feature BAND No. Numbering LAR
0 1 2 3 4 W Request Dgts Format Subaddress
1: y y y y y n y as-needed rest pub-unk none
2: y y y y y n n rest none
3: y y y y y n n rest none
4: y y y y y n n rest none
5: y y y y y n n rest none
6: y y y y y n n rest none
```

- 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.



```
display aar analysis 4 Page 1 of 2
```

AAR DIGIT ANALYSIS TABLE

Percent Full: 2

Dialed String	Total		Route Pattern	Call Type	Node Num	ANI Reqd
	Min	Max				
4	4	4	20	aar		y
4	7	7	999	aar		n
4001	4	4	4	aar		y
4008	4	4	4	aar		y
4015	4	4	4	aar		n
44	4	4	4	aar		y
5	4	4	10	aar		n
5	7	7	999	aar		n
5001	4	4	25	aar		n
5050	4	4	10	aar		n
555	7	7	4	aar		n
7	7	7	999	aar		n
70007950	8	8	45	aar		n
8	7	7	999	aar		n
88001	5	5	65	aar		n

7. Last step is to ensure caller id is enabled on each IP phone to send calling party name.

```
display station 7007 Page 2 of 4
```

STATION

FEATURE OPTIONS

LWC Reception: spe	Auto Select Any Idle Appearance? n
LWC Activation? y	Coverage Msg Retrieval? y
LWC Log External Calls? n	Auto Answer: none
CDR Privacy? n	Data Restriction? n
Redirect Notification? y	Idle Appearance Preference? n
Per Button Ring Control? n	
Bridged Call Alerting? n	Restrict Last Appearance? y
Active Station Ringing: continuous	
H.320 Conversion? y	Per Station CPN - Send Calling Number? y
Service Link Mode: as-needed	
Multimedia Mode: enhanced	Audible Message Waiting? n
MWI Served User Type: qsig-mwi	Display Client Redirection? n
	Select Last Used Appearance? n
	Coverage After Forwarding? s
	Multimedia Early Answer? n
	Direct IP-IP Audio Connections? y
	IP Audio Hairpinning? y

Emergency Location Ext: 7007



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:

Clusterwide Parameters (Feature - Path Replacement)		
Parameter Name	Parameter Value	Suggested Value
Path Replacement Enabled*	<input type="text" value="True"/>	False
Path Replacement on Tromboned Calls*	<input type="text" value="True"/>	True
Start Path Replacement Minimum Delay Time (sec)*	<input type="text" value="5"/>	0
Start Path Replacement Maximum Delay Time (sec)*	<input type="text" value="10"/>	0
Path Replacement T1 Timer (sec)*	<input type="text" value="30"/>	30
Path Replacement T2 Timer (sec)*	<input type="text" value="15"/>	15



Start Path Replacement Minimum Delay Time (sec)*	<input type="text" value="5"/>	0
Start Path Replacement Maximum Delay Time (sec)*	<input type="text" value="10"/>	0
Path Replacement T1 Timer (sec)*	<input type="text" value="30"/>	30
Path Replacement T2 Timer (sec)*	<input type="text" value="15"/>	15
Path Replacement PINX Id	<input type="text" value="4444"/>	
Path Replacement Calling Search Space	<input type="text" value="< None >"/>	

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.



status ready

Domain Name*

Description

Cisco CallManager Group*

Installed Voice Interface Cards **Endpoint Identifiers**

Mainboard Slot	<input type="text" value="< None >"/>		
Module in Slot 1	<input type="text" value="NM-HDV"/>		
	Subunit	<input type="text" value="VWIC-2MFT-T1"/>	(1/0)
Module in Slot 2	<input type="text" value="< None >"/>		
Module in Slot 3	<input type="text" value="< None >"/>		
Module in Slot 4	<input type="text" value="NM-2V"/>		
	Subunit 0	<input type="text" value="< None >"/>	
	Subunit 1	<input type="text" value="VIC-2FXS"/>	(4/1/0) (4/1/1)

Product Specific Configuration

Global ISDN Switch Type

Switchback Timing*

Switchback uptime-delay (min)

Switchback schedule (hh:mm)



Device Information

End-Point Name*	<input type="text" value="S1/DS1-0@CCME_CUE_3745"/>
Description	<input type="text" value="S1/DS1-0@CCME_CUE_3745"/>
Device Pool*	<input type="text" value="Default"/>
Call Classification*	<input type="text" value="Use System Default"/>
Network Locale	<input type="text" value="United States"/>
Media Resource Group List	<input type="text" value=" < None >"/>
Location	<input type="text" value=" < None >"/>
AAR Group	<input type="text" value=" < None >"/>
Load Information	<input type="text"/>
V150 (subset)	<input type="checkbox"/>

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain (e.g., "0000FF")	<input type="text"/>
MLPP Indication	<input type="text" value="Off"/>
MLPP Preemption	<input type="text" value="Disabled"/>



Interface Information

PRI Protocol Type*	<input type="text" value="PRI QSIG T1"/>
Protocol Side*	<input type="text" value="User"/>
Channel Selection Order*	<input type="text" value="Top Down"/>
Channel IE Type*	<input type="text" value="Use Number when 1B"/>
PCM Type*	<input type="text" value="μ-law"/>
Delay for first restart (1/8 sec ticks)	<input type="text" value="32"/>
Delay between restarts (1/8 sec ticks)	<input type="text" value="4"/>
<input checked="" type="checkbox"/> Inhibit restarts at PRI initialization	
<input type="checkbox"/> Enable status poll	

Call Routing Information

Inbound Calls

Significant Digits*	<input type="text" value="All"/>
Calling Search Space	<input type="text" value=" < None >"/>
AAR Calling Search Space	<input type="text" value=" < None >"/>
Prefix DN	<input type="text"/>

Outbound Calls

Calling Line ID Presentation*	<input type="text" value="Allowed"/>
Calling Party Selection*	<input type="text" value="Originator"/>
Called party IE number type	<input type="text" value="National"/>



Called party IE number type unknown*	<input type="text" value="National"/>
Calling party IE number type unknown*	<input type="text" value="National"/>
Called Numbering Plan*	<input type="text" value="ISDN"/>
Calling Numbering Plan*	<input type="text" value="ISDN"/>
Number of digits to strip*	<input type="text" value="0"/>
Caller ID DN	<input type="text"/>
SMDI Base Port*	<input type="text" value="0"/>

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**	<input type="text" value="0"/>
------------------------------	--------------------------------

Connected Line ID Presentation (QSIG Inbound Call)*	<input type="text" value="Allowed"/>
---	--------------------------------------




Connected Line ID Presentation (QSIG Inbound Call)*

UUIE Configuration

Passing Precedence Level Through UUIE

Security Access Level

Product Specific Configuration 

Line Coding*

Framing*

Clock*

Input Gain (-6..14 db)*

Output Attenuation (-6..14 db)*

Echo Cancellation Enable*

Echo Cancellation Coverage (ms)*

* indicates required item
** applicable to DMS-100 protocol only
*** applicable to DMS-100 protocol and DMS-250 protocol only
**** may be required to force ringback from some PBXs

[Back to MGCP Configuration](#)
[Back to Find/List Gateways](#)

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.



System Route Plan Service Feature Device User Application Help

Cisco CallManager Administration
For Cisco IP Telephony Solutions

CISCO SYSTEM

Call Pickup Configuration

[Add a New Call Pickup Number](#)
[Back to Find/List Call Pickup Number](#)
[Dependency Record](#)

Call Pickup Number: 4444
Status: Ready

Call Pickup Number*

Description

Partition

* indicates required item

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
```



login

!

end

Features Tested for Interoperability between Cisco and Avaya IP-PBX Systems

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.



System Route Plan Service Feature Device User Application Help

Cisco CallManager Administration
For Cisco IP Telephony Solutions

CISCO SYSTEMS

Cisco Voice Mail Port Wizard

What would you like to do?

- Create a new Cisco Voice Mail Server and add ports to it
- Add ports to an existing Cisco Voice Mail server
- Delete ports from an existing Cisco Voice Mail server

Next

2. Enter a Cisco Voice Mail Server name, such as AvayaUM3, and click Next.



System Route Plan Service Feature Device User Application Help

Cisco CallManager Administration
For Cisco IP Telephony Solutions

CISCO SYSTEMS

Cisco Voice Mail Port Wizard

Cisco Voice Mail Server

Add ports to a new Cisco Voice Mail Server using this name:

3. Select the Voice Mail Ports you want configured and click Next.



System Route Plan Service Feature Device User Application Help

Cisco CallManager Administration
For Cisco IP Telephony Solutions

CISCO SYSTEMS

Cisco Voice Mail Port Wizard

Cisco Voice Mail Ports

AvayaUM3 currently has 0 ports configured.
How many ports do you want to add?

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.



System Route Plan Service Feature Device User Application Help

Cisco CallManager Administration
For Cisco IP Telephony Solutions

CISCO SYSTEMS

Cisco Voice Mail Port Wizard

Cisco Voice Mail Device Information

Enter the device information for ports 1 through 2 of AvayaUM3. A Device Pool selection is required. The Wizard applies these settings to all new ports.

Device Information

Description	<input type="text" value="Avaya VMailPorts"/>
Device Pool*	<input type="text" value="Default"/>
Calling Search Space	<input type="text" value="< None >"/>
AAR Calling Search Space	<input type="text" value="< None >"/>
Location	<input type="text" value="< None >"/>

* indicates required item

5. Enter the Beginning Directory Number, such as 4406, and the Display, such as Voicemail and click Next.



System Route Plan Service Feature Device User Application Help

Cisco CallManager Administration
For Cisco IP Telephony Solutions

CISCO SYSTEMS

Cisco Voice Mail Port Wizard

Cisco Voice Mail Directory Numbers

Enter the directory number settings for the new Cisco Voice Mail Server (AvayaUM3). If a Partition is selected, you must select a Calling Search Space that includes the selected Partition.

Beginning Directory Number* (each new port receives the next available directory number)

Partition

Calling Search Space

Display

AAR Group

External Number Mask

* indicates required item

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.



System Route Plan Service Feature Device User Application Help

Cisco CallManager Administration
For Cisco IP Telephony Solutions

CISCO SYSTEMS

Cisco Voice Mail Port Wizard

Do you want to add these directory numbers to a Line Group?

For using these ports, you need to add corresponding directory numbers to a line group. You can add them to an existing line group or to a new line group. If you decide to add it later, you can do so by using Line Group configuration option.

- Yes. Add directory numbers to a **new** Line Group
- Yes. Add directory numbers to an **existing** Line Group
- No. I will add them later.

7. Enter a Line Group Name which matches the Voice Mail Server you previously entered, such as AvayaUM3.



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Cisco CallManager Administration
For Cisco IP Telephony Solutions

CISCO SYSTEMS

Cisco Voice Mail Port Wizard

Line Group

Enter the Line Group settings for Cisco Voice Mail Server (AvayaUM3).

Line Group Name

* indicates required item

8. The next screen shows the configuration entered so far. Click Finish if there are no changes to the configuration.



Ready to Add Cisco Voice Mail Ports

The information shown below will be applied to the Cisco Voice Mail Ports being created. If this information is correct, click Finish to add the new ports. If the information shown is not correct, click the Back button to edit the information, or Cancel to quit without adding any ports.

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 VMailPorts
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

9. Click Add a New Hunt List on the Hunt List Administration web page.

The screenshot shows the Cisco CallManager Administration web interface. At the top, there is a navigation menu with links for System, Route Plan, Service, Feature, Device, User, Application, and Help. Below the navigation menu is a header banner with the text "Cisco CallManager Administration For Cisco IP Telephony Solutions" and the Cisco Systems logo. The main content area is titled "Find and List Hunt Lists" and contains a search interface. The search interface includes a dropdown menu for "Find Hunt Lists where" (set to "HuntListName"), a dropdown menu for "begins with" (set to "begins with"), a text input field, and a "Find" button. Below the search interface, there is a message "No current search" and a note "and show 20 items per page". A link "Add a New Hunt List" is visible in the top right corner of the main content area.



10. Enter a Hunt List Name and Description, such as Avaya VMailHL. Also select Default for the Cisco Call Manager Group.

The screenshot shows the Cisco CallManager Administration web interface. At the top, there is a navigation menu with links for System, Route Plan, Service, Feature, Device, User, Application, and Help. The main header area includes the Cisco CallManager Administration logo and the Cisco Systems logo. The page title is "Hunt List Configuration". On the right side, there are two links: "Add a new Hunt List" and "Back to Find/List Hunt Lists". The main content area is titled "Hunt List: New" and shows the status as "Ready". There is an "Insert" button. Below this, the "Hunt List Information" section contains three input fields: "Hunt List Name*" with the value "Avaya VMail HL", "Description" with the value "Avaya VMail HL", and "Cisco CallManager Group*" with a dropdown menu set to "Default". A note at the bottom of this section states "* indicates required item".

11. The following screen capture is the result of successfully adding the Hunt List. Click Add Line Group.



Hunt List Details

Hunt List: Avaya VMail HL

Status: Insert completed

Hunt List Information

Hunt List Name*

Description

Cisco CallManager Group*

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

Hunt List Member Information

Selected Groups*
(ordered by highest priority)

▼ ▲

Removed Groups
(to be removed from Hunt List when you click Update)

* indicates required item

12. Select the Line Group previously configured. In our case, it's AvayaUM3.



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Hunt List Detail Configuration

[Add a new Hunt List](#)
[Back to Hunt List Configuration](#)
[Back to Find/List Hunt Lists](#)

Hunt List Details

Hunt List: Avaya VMail HL
Status: Ready

Details for New Hunt List Member

Line Group*

13. The next screen capture shows the result of successfully inserting the line group.



Hunt List Configuration

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

Hunt List Details
AvayaUM3

Hunt List: Avaya VMail HL
Status: Line Group insert completed

Hunt List Information

Hunt List Name*

Description

Cisco CallManager Group*

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

Hunt List Member Information

Selected Groups*
(ordered by highest priority)

Removed Groups
(to be removed from Hunt List when you click Update)

14. Go to Route Plan, Route/Hunt, Hunt Pilot. Click Add a New Hunt Pilot from the resulting Hunt Pilot screen.



System **Route Plan** Service Feature Device User Application Help

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CISCO SYSTEMS

Find and List Hunt Pilots

[Add a New Hunt Pilot](#)

No current search

Find Hunt Pilots where

and show items per page

To list all items, click Find without entering any search text.

15. Enter in the Hunt Pilot, such as 4408, and select a Hunt List, such as Avaya VMail HL and click Insert.



Hunt Pilot Configuration

[Add a New Hunt Pilot](#)
[Back to Find/List Hunt Pilots](#)

Hunt Pilot:
Status: Ready
Note: Any update to this Hunt Pilot automatically resets the associated Hunt List

Pattern Definition

Hunt Pilot*	<input type="text" value="4408"/>
Partition	<input style="border: none;" type="text" value=" < None > "/>
Description	<input type="text"/>
Numbering Plan*	<input style="border: none;" type="text" value="North American Numbering Plan"/>
Route Filter	<input style="border: none;" type="text" value=" < None > "/>
MLPP Precedence	<input style="border: none;" type="text" value="Default"/>
Hunt List*	<input style="border: none;" type="text" value="Avaya VMail HL"/>
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern <input style="border: none;" type="text" value=" - Not Selected - "/>

Provide Outside Dial Tone Urgent Priority

Hunt Forward Settings

	Use Personal Preferences	Destination	Calling Search Space
Forward Hunt No Answer	<input type="checkbox"/>	<input type="text"/>	<input style="border: none;" type="text" value=" < None > "/>
Forward Hunt Busy	<input type="checkbox"/>	<input type="text"/>	<input style="border: none;" type="text" value=" < None > "/>
Maximum Hunt Timer	<input type="text"/>	(Seconds)	

16. Go to Feature, Voice Mail, Voice Mail Pilot and click Add a New Voice Mail Pilot on the resulting screen.



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CISCO SYSTEMS

Find and List Voice Mail Pilots

[Add a New Voice Mail Pilot](#)

2 matching record(s) for Voice Mail Pilot Number begins with ""

Find voice mail pilots where begins with

and show items per page

To list all items, click Find without entering any search text.

Matching record(s) 1 to 2 of 2

<input type="checkbox"/>	Pilot Number	Description	Calling Search Space
<input type="checkbox"/>		No Voice Mail	
<input type="checkbox"/>	4405	Default	

Page of 1

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.



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Voice Mail Pilot Configuration

[Add a New Voice Mail Pilot Number](#)
[Back to Find/List Voice Mail Pilots](#)

Voice Mail Pilot Number : New
Status: Ready

Voice Mail Pilot Number

Description

Calling Search Space

Make this the default Voice Mail Pilot for the system

* indicates required item

18. Next, go to Feature, Voice Mail, Voice Mail Profile and click Add a New Voice Mail Profile.



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Cisco CallManager Administration
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CISCO SYSTEMS

Find and List Voice Mail Profiles

[Add a New Voice Mail Profile](#)

2 matching record(s) for Voice Mail Profile Name begins with ""

Find Voice Mail Profiles where Voice Mail Profile Name
and show items per page.
To list all items, click Find without entering any search text.

Matching record(s) 1 to 2 of 2

<input type="checkbox"/>	VM Profile Name	Description	Pilot/Calling Search Space	Copy
<input type="checkbox"/>	Default	Default voice messaging profile	4405/< None >	
	NoVoiceMail**	No Voice Mail	< None >/< None >	

* Voice Mail Profile using the default Voice Mail Pilot for the system (0/< None >)
** This is the special No Voice Mail Profile for the system (cannot be deleted)

First Previous Next Last Page of 1

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.



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Voice Mail Profile Configuration

[Add a New Voice Mail Profile](#)
[Back to Find/List Voice Mail Profiles](#)

Voice Mail Profile: New
Status: Ready

Voice Mail Profile Name*

Description

Voice Mail Pilot **

Voice Mail Box Mask

Make this the default Voice Mail Profile for the system

* indicates required item

** The Voice Mail Pilot is comprised of the Voice Mail Pilot Number and it's corresponding Calling Search Space Name (<Voice Mail Pilot Number>/<Calling Search Space>).

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.



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CISCO SYSTEMS

Message Waiting Configuration

[Add a New Message Waiting Number](#)
[Back to Find/List Message Waiting Numbers](#)

Message Waiting Number : 1001
Status: Ready

Copy Update Delete

Message Waiting Number* 1001

Description

Message Waiting Indicator On Off

Partition < None >

Calling Search Space < None >

* indicates required item

System **Route Plan** Service Feature Device User Application Help

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CISCO SYSTEMS

Message Waiting Configuration

[Add a New Message Waiting Number](#)
[Back to Find/List Message Waiting Numbers](#)

Message Waiting Number : 1000
Status: Ready

Copy Update Delete

Message Waiting Number* 1000

Description

Message Waiting Indicator On Off

Partition < None >

Calling Search Space < None >

* 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.