

Configure Q.SIG PRI Trunks between Call Manager and Avaya S8700/G650 with Unity Voice Mail Integration

Document ID: 63790

Contents

Introduction

Prerequisites

- Requirements
- Components Used
- Conventions

Test Setup

Test Topology

Interoperability between Cisco and Avaya IP–PBX Systems

- Procedure on Avaya S8700/G650 IP–PBX system
- Procedure on Cisco Call Manager

Cisco 3745 Configuration

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

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

- Add Cisco Unity to Cisco Call Manager

Cisco Unity Voice Mail Features Tested

Related Information

Introduction

The objective of this document is to provide Cisco customers and business partners with the steps to configure Q.SIG PRI trunks between the Cisco Call Manager and the Avaya S8700/G650. Also, this document details steps for how to add Cisco Unity on the Cisco Call Manager platform in order provide voice mail support for both Cisco and Avaya IP phones. This is particularly important in situations where IP–PBX interoperability and voice mail integration are required. The Avaya configuration screen captures were created with the standard Emulation tool. As an alternative, you 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.

Prerequisites

Requirements

There are no specific requirements for this document.

Components Used

The information in this document is based on these software and hardware versions:

- The Avaya IP–PBX system used is 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 in this document are the 4610SW and 4620 running Phone Firmware Version 2.01.

- Cisco Call Manager 4.1.(2) was used in order to control the 3745 Media Gateway Control Protocol (MGCP) gateway with the NM–HDV module, running Cisco IOS® version 12.2.15ZJ3. Tests were also repeated with Cisco IOS® version 12.3.8.T5.
- Cisco Unity running version 4.0(4) SR1 was used for the voice mail integration testing.

The information in this document was created from the devices in a specific lab environment. All of the devices used in this document started with a cleared (default) configuration. If your network is live, make sure that you understand the potential impact of any command.

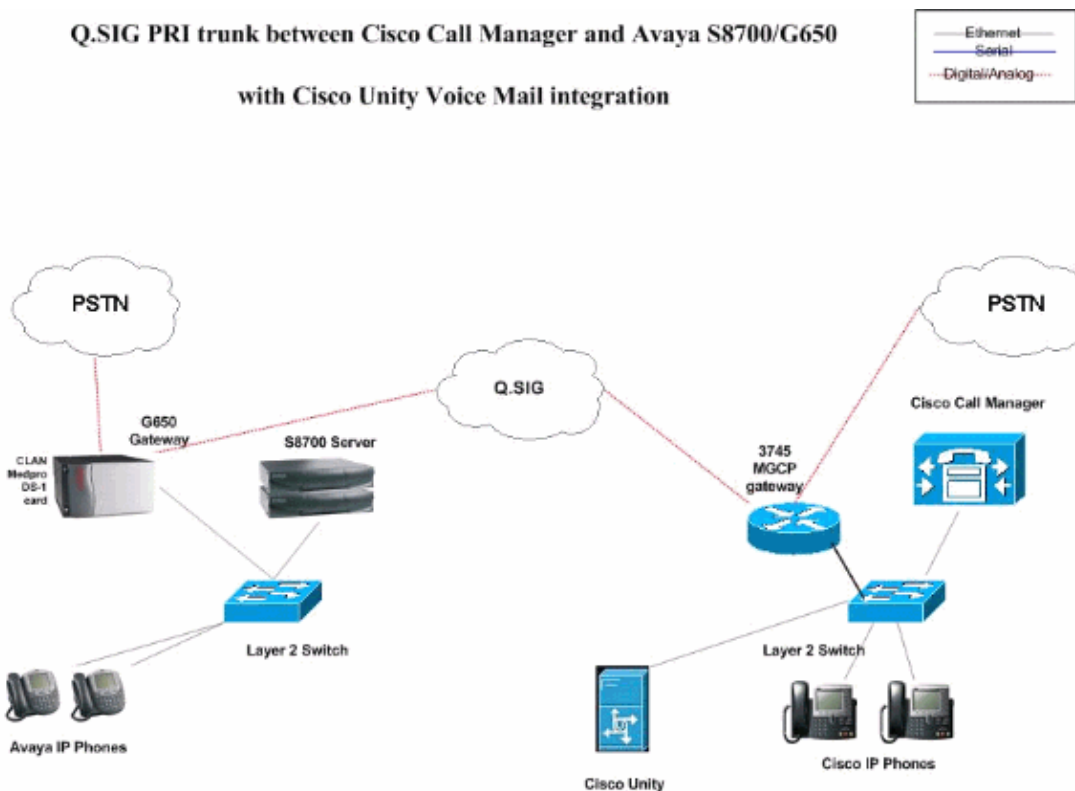
Conventions

Refer to Cisco Technical Tips Conventions for more information on document conventions.

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 Cisco IOS® version 12.2.15ZJ3. Tests were also repeated with Cisco IOS® version 12.3.8.T5. Cisco Unity running version 4.0(4) SR1 was used for the voice mail integration testing.

Test Topology



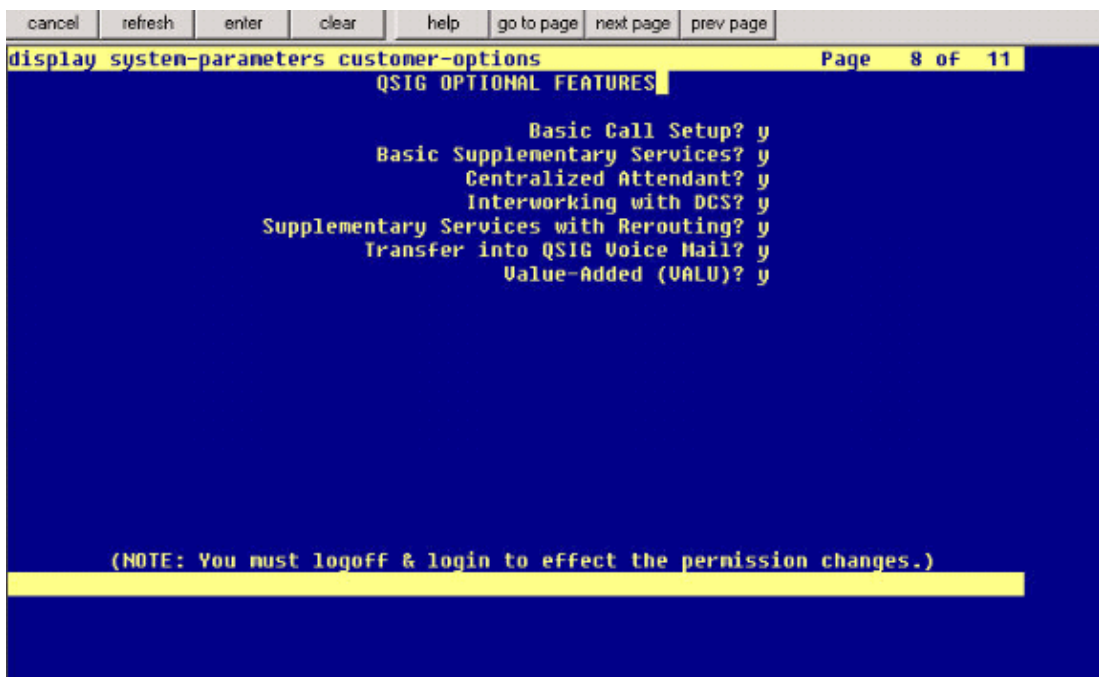
Interoperability between Cisco and Avaya IP-PBX Systems

The next sections provide procedures and screen captures to help you 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 that provides the physical ISDN PRI connection to the Avaya S8700/G650.

Procedure on Avaya S8700/G650 IP-PBX system

Complete these steps:

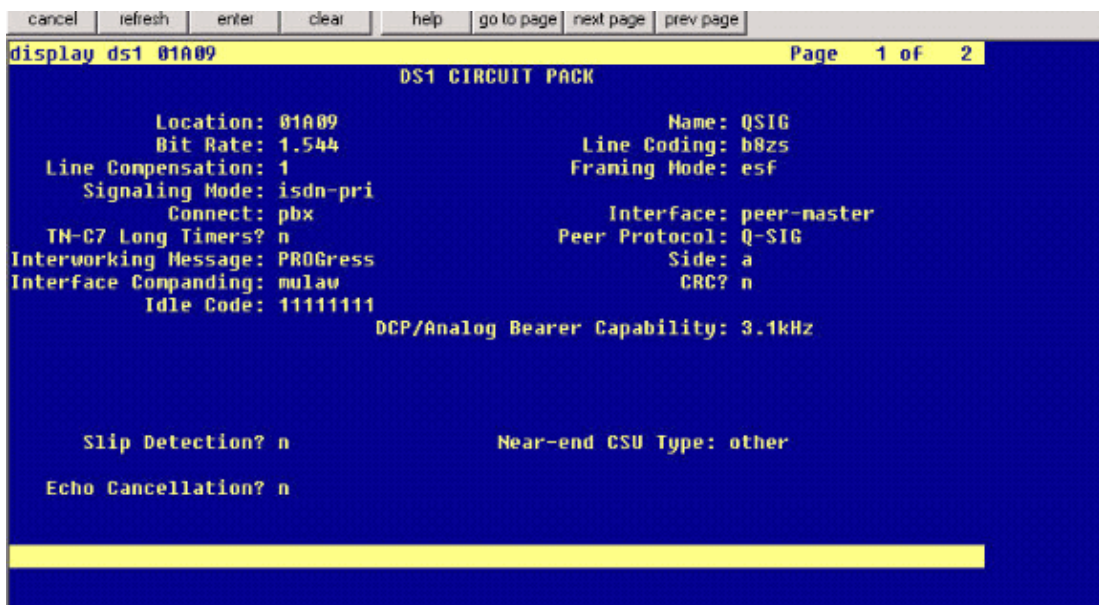
1. Login to the S8700 server. Run the **display system-parameters customer** command in order to ensure that all the necessary Q.SIG features are enabled on the S8700 server.



```
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. Configure a trunk group. Type **add trunk-group #** where # is the desired trunk.

The next three screen captures relate to the trunk configuration. Once the trunk group is created, add the 23 DS0 channels to the group. This is an example of the port assignment: 01A0901 means: Gateway# 1, Cabinet A, Slot# 9, DS0 channel# group1.

```

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		1
2:	01A0902	TN464	G		1
3:	01A0903	TN464	G		1
4:	01A0904	TN464	G		1
5:	01A0905	TN464	G		1
6:	01A0906	TN464	G		1
7:	01A0907	TN464	G		1
8:	01A0908	TN464	G		1
9:	01A0909	TN464	G		1
10:	01A0910	TN464	G		1
11:	01A0911	TN464	G		1
12:	01A0912	TN464	G		1
13:	01A0913	TN464	G		1
14:	01A0914	TN464	G		1
15:	01A0915	TN464	G		1

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

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

5. Add the 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 step 4.

```

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 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

```

6. Add an entry into the AAR table in order to use the route pattern you created to route calls. In this example, calls to Cisco IP phone extension 4XXX use 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
Dialled Total Route Call Node ANI
String Min Max Pattern Type Num Reqd
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. 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                       Restrict Last Appearance? y
    Bridged Call Alerting? n
    Active Station Ringing: continuous

    H.320 Conversion? y                             Per Station CPH - Send Calling Number? y
    Service Link Mode: as-needed
    Multimedia Mode: enhanced
    MWI Served User Type: qsig-mwi

    Audible Message Waiting? n
    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

Complete these steps:

1. Under Service parameters, make sure that the Start Path Replacement Minimum and Maximum time values are set appropriately in order to prevent any issues (such as hair pinning).

The next two screen captures relate to the Q.SIG Service Parameters settings:

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 the Q.SIG PRI.

The next five screen captures relate to this configuration:

```

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

```


Device Information

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

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain (e.g., "0000FF")	
MLPP Indication	Off
MLPP Preemption	Disabled

Interface Information

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

Call Routing Information

Inbound Calls

Significant Digits*	All
Calling Search Space	< None >
AAR Calling Search Space	< None >
Prefix DN	

Outbound Calls

Calling Line ID Presentation*	Allowed
Calling Party Selection*	Originator
Called party IE number type	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


<input type="checkbox"/> Display IE Delivery	
<input type="checkbox"/> Redirecting Number IE Delivery - Outbound	
<input type="checkbox"/> Redirecting Number IE Delivery - Inbound	
<input checked="" type="checkbox"/> Send Extra Leading Character In DisplayIE***	
<input type="checkbox"/> Setup non-ISDN Progress Indicator IE Enable****	
<input type="checkbox"/> MCDN Channel Number Extension Bit Set to Zero**	
<input type="checkbox"/> Send Calling Name In Facility IE	
<input type="checkbox"/> 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](#)

- As a final step, create a Cisco Call Manager pickup group in order to provide a path proposal extension to the PBX. Make sure that the call pickup number is also entered into the Path PINX Replacement ID Service parameter (see to step# 1). Also, the Avaya system needs a route pattern in order to route to the pickup group.

```
display aar analysis 4
```

Page 1 of 2

AAR DIGIT ANALYSIS TABLE

Percent Full: 2

Dialed String	Total		Route Pattern	Call Type	Node Nun	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	

Note: Make sure that these two clusterwide parameters (**Device – PRI and MGCP Gateway**) under **Cisco CallManager Service Parameters (Advanced)** match the Q.SIG configuration in the PBX. All PBX trunks must be configured exactly as these Cisco CallManager parameters.

- ◆ **ASN.1 ROSE OID Encoding:** This parameter specifies how to encode the Invoke Object ID (OID) for the Remote Operations Service Element (ROSE). Keep this parameter set to the default value unless a Cisco support engineer instructs otherwise. This is a required field and the default is **Use Local Value**.

These are the valid values for this parameter:

- ◇ **Use Local Value**, which is supported by most telephony systems and must be used when the Q.SIG Variant service parameter is set to ISO (Protocol Profile 0x9F).
 - ◇ **Use Global Value (ISO)**, which is used only if the connected PBX does not support Use Local Value.
 - ◇ **Use Global Value (ECMA)**, which must be used if the Q.SIG Variant service parameter is set to ECMA (Protocol Profile 0x91).
- ◆ **Q.SIG Variant:** This parameter specifies the protocol profile sent in outbound Q.SIG facility information elements when the trunk is configured for Q.SIG. Keep this parameter set to the default value unless a Cisco support engineer instructs otherwise. This is a required field, and the default is **ISO (Protocol Profile 0x9F)**.

These are the available values for this parameter:

- ◇ **ECMA (Protocol Profile 0x91)**, which is typically used with ECMA PBXs and can only use Protocol Profile 0x91. If this service parameter is set to ECMA (Protocol Profile 0x91), the ASN.1 Rose OID Encoding service parameter must be set to Use Global Value (ECMA).
- ◇ **ISO (Protocol Profile 0x9F)**, which is the current ISO recommendation. If this parameter is set to ISO (Protocol Profile 0x9F), then the ASN.1 Rose OID Encoding service parameter must be set to Use Local Value.

Warning: Cisco CallManager does not support ECMA when using intercluster trunks with the Tunneled Protocol field set to Q.SIG in the Trunk Configuration window in the CallManager Administration. If you set this service parameter to ECMA (Protocol Profile 0x91), all intercluster trunks must have the Tunneled Protocol field set to None.

Clusterwide Parameters (Device - PRI and MGCP Gateway)

Parameter Name	Parameter Value	Sug
ASN.1 ROSE OID Encoding*	Use Local Value	Use
QSIG Variant*	ISO (Protocol Profile 0x9F)	ISO
Caller ID		
Calling Name Not Available Timeout (msec)*	2000	2000
Calling Party Number Screening Indicator*	CallManager sets the screening indicator value - Default setting	CallM scre Defa
Change B- Channel Maintenance Status 1		
Change B- Channel		

Cisco 3745 Configuration

This is the **show version** and **show running-configuration** command output 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 (FC1)
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 (FC1)

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

This section provides a list of features tested between the Cisco Call Manager 4.1(2) platform and the Avaya S8700/G650 running Communication Manager 2.0 by way of 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

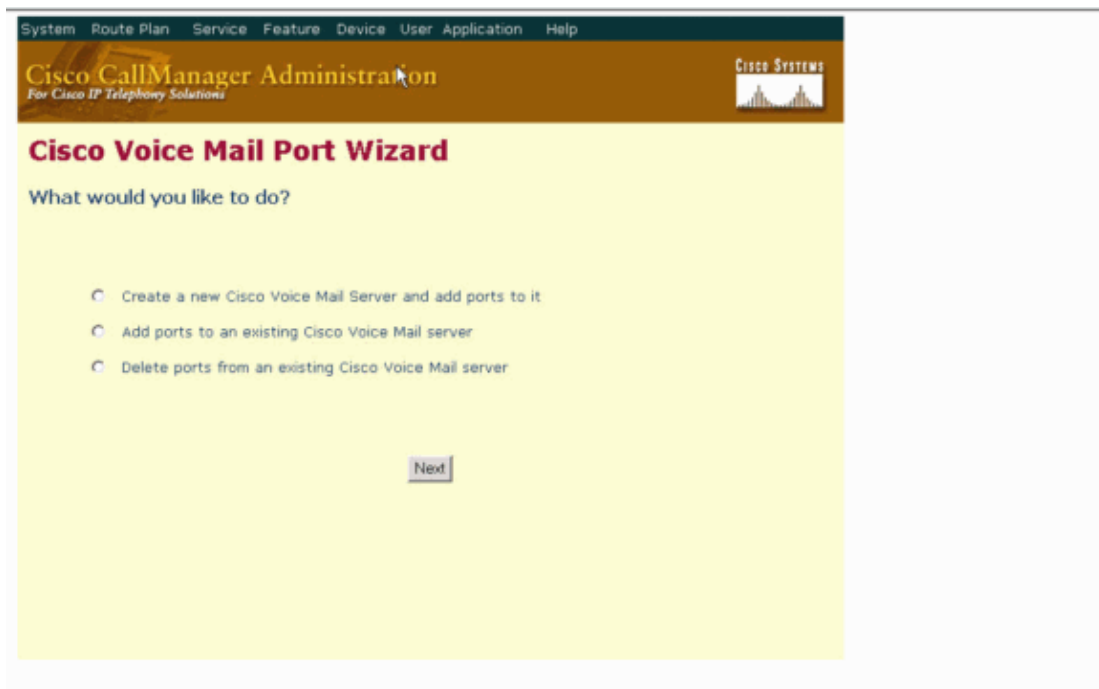
At this point, one can use the Q.SIG trunk in order to make calls between an Avaya S8700/G650 running Avaya Communication Manager 2.0 and a Cisco Call Manager platform running Call Manager version 4.1(2) with the Cisco 3745 MGCP device providing the physical ISDN PRI connection to the Avaya S8700/G650. A Cisco Unity server can be added on the Cisco Call Manager platform in order to provide voice mail support to both the Cisco and Avaya IP phones. In order to enable this, the administrator needs to configure the Cisco Unity on the Cisco Call Manager platform. This section includes the procedures with screen captures for 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.

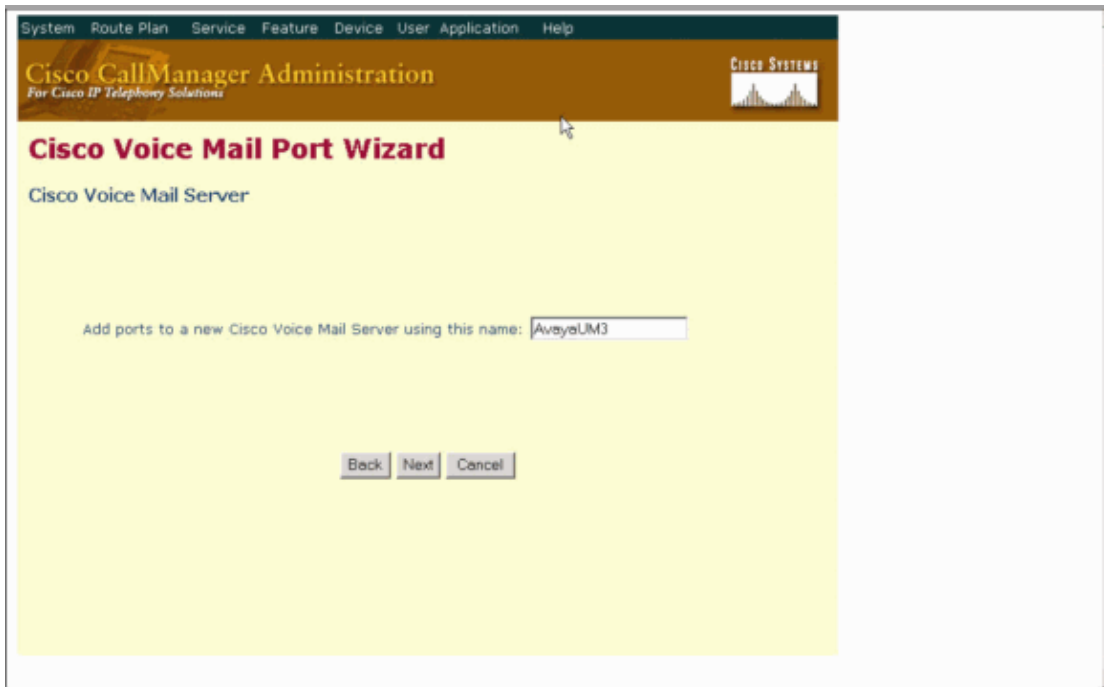
Add Cisco Unity to Cisco Call Manager

Complete these steps:

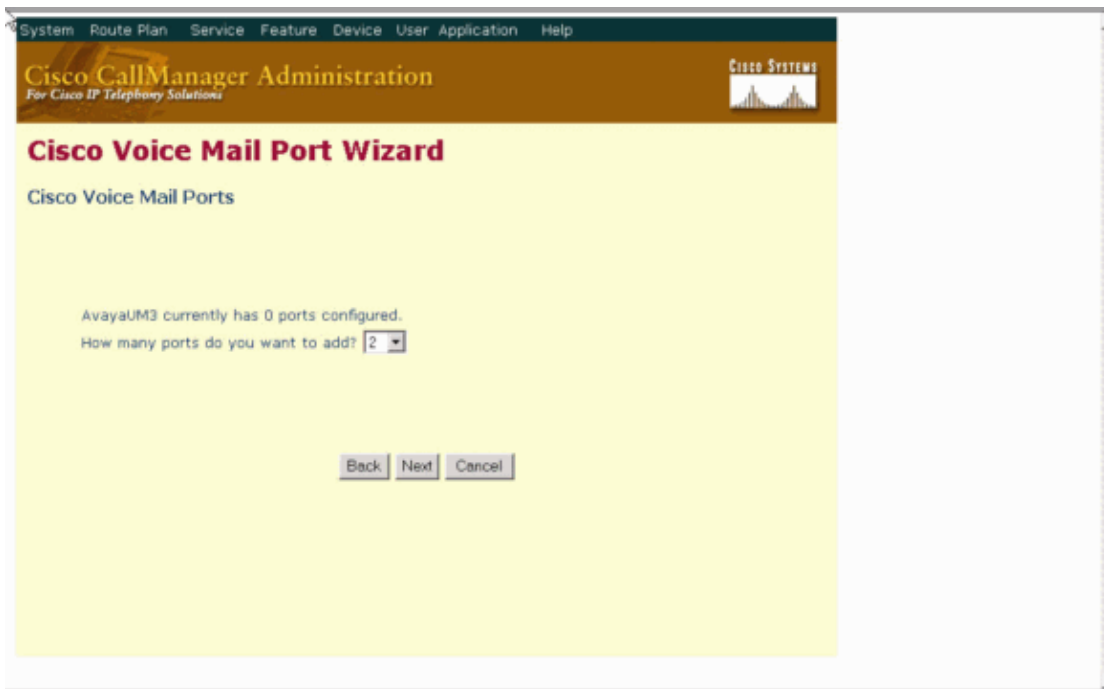
1. Under Feature, select **Voice Mail > Voice Mail Port Wizard**. Select **Create a new voice mail server** and add ports to it and click **Next**.



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



3. Select the number of Voice Mail Ports desired and click **Next**.



4. Enter a Description and Device Pool for the Voice Mail Ports. In the sample configuration, Avaya VMailPorts was entered as the description and Default as the device pool.

```

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
  Hop Dgt? 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 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

```

5. Enter the Beginning Directory Number, such as 4406, and the Display, such as Voice Mail, and click **Next**.

```

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                             Interface: peer-master
  Connect: pbx                                         Peer Protocol: Q-SIG
  TN-C7 Long Timers? n                                 Side: a
  Interworking Message: PROGress                       CRC? n
  Interface Companding: mulaw
  Idle Code: 11111111
  DCP/Analog Bearer Capability: 3.1kHz

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

```

6. The next screen asks, "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**.

```

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

```

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

```

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

```

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


```

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		1
2:	01A0902	TN464	G		1
3:	01A0903	TN464	G		1
4:	01A0904	TN464	G		1
5:	01A0905	TN464	G		1
6:	01A0906	TN464	G		1
7:	01A0907	TN464	G		1
8:	01A0908	TN464	G		1
9:	01A0909	TN464	G		1
10:	01A0910	TN464	G		1
11:	01A0911	TN464	G		1
12:	01A0912	TN464	G		1
13:	01A0913	TN464	G		1
14:	01A0914	TN464	G		1
15:	01A0915	TN464	G		1

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

```

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

```

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

```

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 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

```

11. This screen capture is the result of the successful addition of the Hunt List. Click **Add Line Group**.

The screenshot shows the 'Hunt List Details' page for 'Avaya VMail HL'. The status is 'Insert completed'. There are buttons for 'Copy', 'Update', 'Delete', and 'Reset'. The 'Hunt List Information' section includes fields for 'Hunt List Name*' (Avaya VMail HL), 'Description' (Avaya VMail HL), and 'Cisco CallManager Group*' (Default). A checkbox 'Enable this Hunt List' is checked. The 'Hunt List Member Information' section has an 'Add Line Group' button and two lists: 'Selected Groups*' (ordered by highest priority) and 'Removed Groups' (to be removed from Hunt List when you click Update). A note at the bottom states '* indicates required item'.

12. Select the Line Group previously configured. In this case, it is AvayaUM3.

```
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                           Restrict Last Appearance? y
  Bridged Call Alerting? n
  Active Station Ringing: continuous

  H.320 Conversion? y                                 Per Station CPN - Send Calling Number? y
  Service Link Mode: as-needed
  Multimedia Mode: enhanced
  MWI Served User Type: qsig-mwi

  Audible Message Waiting? n
  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
```

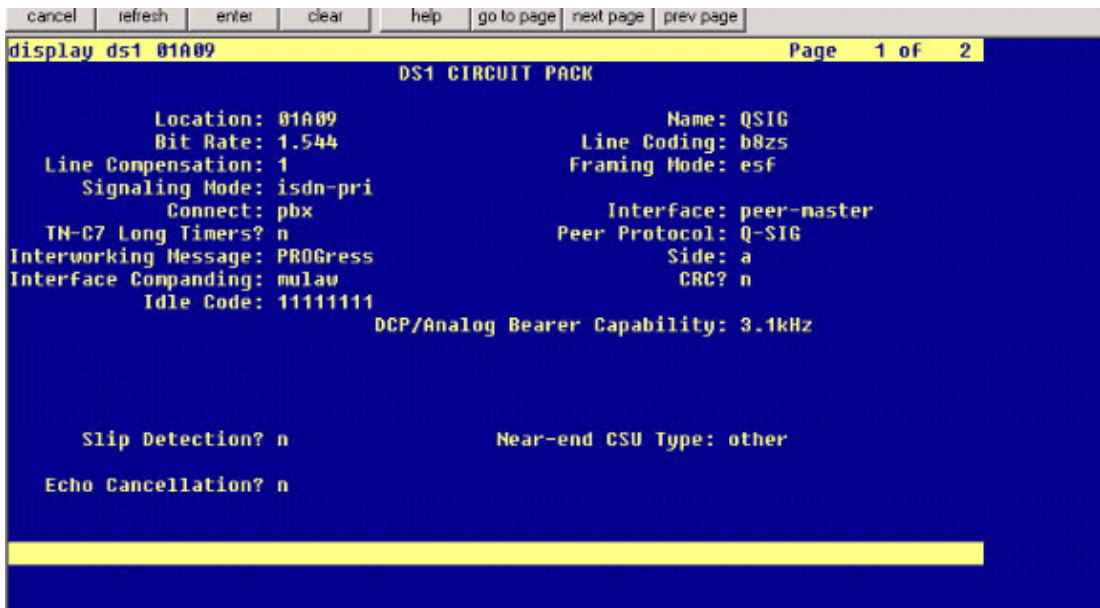
13. The next screen capture shows the result of the successful insertion of the line group.



14. Go to **Route Plan > Route/Hunt > Hunt Pilot**. Click **Add a New Hunt Pilot** from the Hunt Pilot screen that results.



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



16. Go to **Feature > Voice Mail > Voice Mail Pilot** and click **Add a New Voice Mail Pilot** on the screen that results.

```

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

```

17. Enter the Voice Mail Pilot number matching the Hunt Pilot number previously configured. In this case, both the Hunt Pilot and Voice Mail Pilot numbers are 4408.

```

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

```

18. Go to **Feature > Voice Mail > Voice Mail Profile** and click **Add a New Voice Mail Profile**.


```

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		1	
2:	01A0902	TN464	G		1	
3:	01A0903	TN464	G		1	
4:	01A0904	TN464	G		1	
5:	01A0905	TN464	G		1	
6:	01A0906	TN464	G		1	
7:	01A0907	TN464	G		1	
8:	01A0908	TN464	G		1	
9:	01A0909	TN464	G		1	
10:	01A0910	TN464	G		1	
11:	01A0911	TN464	G		1	
12:	01A0912	TN464	G		1	
13:	01A0913	TN464	G		1	
14:	01A0914	TN464	G		1	
15:	01A0915	TN464	G		1	

19. Enter the Voice Mail Profile Name and Description, such as AvayaVMailProfile, and select the Voice Mail Pilot number in step 17. In this 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 (MWD) On/Off numbers. Included here are two screen captures for Message Waiting Indicator On/Off numbers.

System Route Plan Service Feature Device User Application Help

Cisco CallManager Administration
For Cisco IP Telephony Solutions

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

Cisco CallManager Administration
For Cisco IP Telephony Solutions

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

This is a list of Cisco Unity Voice Mail features tested with the Avaya IP phones used to access Cisco Unity Voice Mail by way of 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

Related Information

- [Voice Technology Support](#)
 - [Voice and Unified Communications Product Support](#)
 - [Troubleshooting Cisco IP Telephony](#) 
 - [Technical Support & Documentation – Cisco Systems](#)
-

[Contacts & Feedback](#) | [Help](#) | [Site Map](#)

© 2014 – 2015 Cisco Systems, Inc. All rights reserved. [Terms & Conditions](#) | [Privacy Statement](#) | [Cookie Policy](#) | [Trademarks of Cisco Systems, Inc.](#)

Updated: Aug 20, 2006

Document ID: 63790
