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

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Contents

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 AvayaIP 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
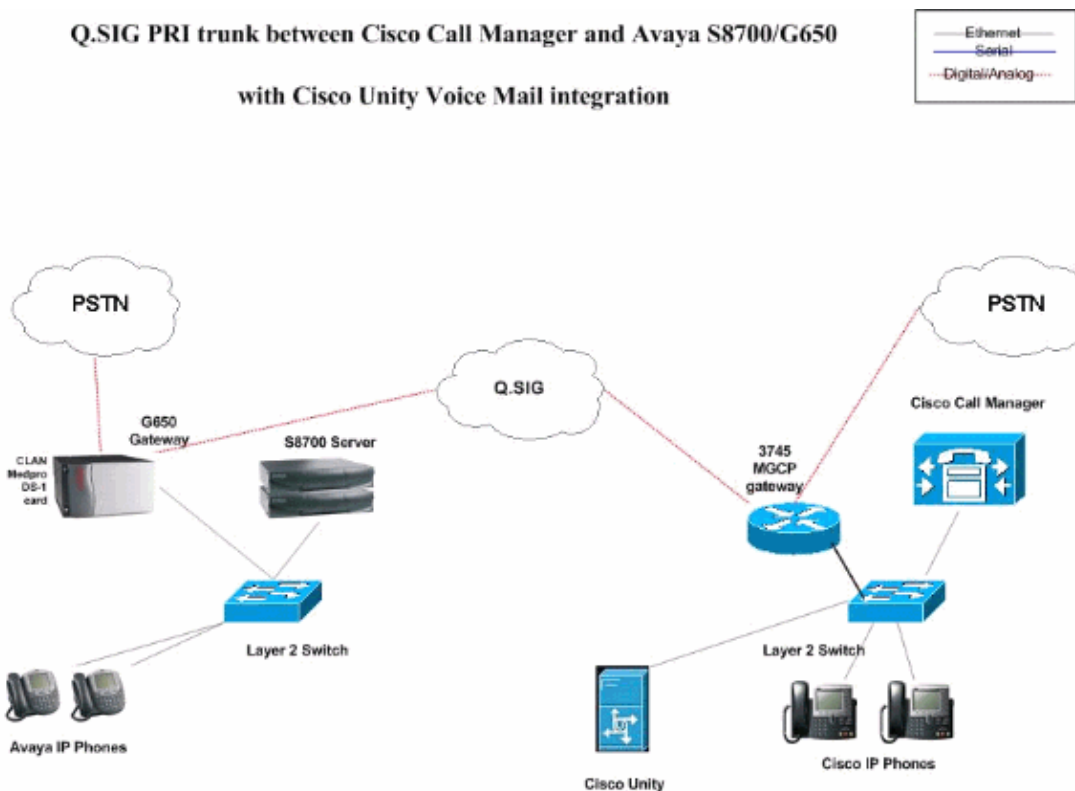
Interoperability between Cisco and Avaya IP–PBX Systems

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
GROUP MEMBER ASSIGNMENTS            Total Administered Members:  23

         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       Max number of NCA TSC: 10
                      Primary D-Channel: 01A0924   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:
```

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                            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
```

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

            Dialed        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
      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
    Emergency Location Ext: 7007                  IP Audio Hairpinning? y
```

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*	True	False
Path Replacement on Tromboned Calls*	True	True
Start Path Replacement Minimum Delay Time (sec)*	5	0
Start Path Replacement Maximum Delay Time (sec)*	10	0
Path Replacement T1 Timer (sec) *	30	30
Path Replacement T2 Timer (sec) *	15	15

Start Path Replacement Minimum Delay Time (sec)*	5	0
Start Path Replacement Maximum Delay Time (sec)*	10	0
Path Replacement T1 Timer (sec)*	30	30
Path Replacement T2 Timer (sec)*	15	15
Path Replacement PINX Id	4444	
Path Replacement Calling Search Space	< 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)	☐

Multilevel Precendence 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

☑ Inhibit restarts at PRI initialization

☐ 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*	National ▾
Calling party IE number type unknown*	National ▾
Called Numbering Plan*	ISDN ▾
Calling Numbering Plan*	ISDN ▾
Number of digits to strip*	0 ▾
Caller ID DN	
SMDI Base Port*	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**	0
Connected Line ID Presentation (QSIG Inbound Call)*	Allowed ▾

Connected Line ID Presentation (QSIG Inbound Call)*	Allowed

UUIE Configuration

☐ Passing Precedence Level Through UUIE

Security Access Level	2

Product Specific Configuration ℹ️

Line Coding*	B8ZS
Framing*	ESF
Clock*	External
Input Gain (-6..14 db)*	0
Output Attenuation (-6..14 db)*	0
Echo Cancellation Enable*	Enable
Echo Cancellation Coverage (ms)*	Default

* 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 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        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
```

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. These 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**	**Sugg**
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 scree 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
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 (fc

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**.
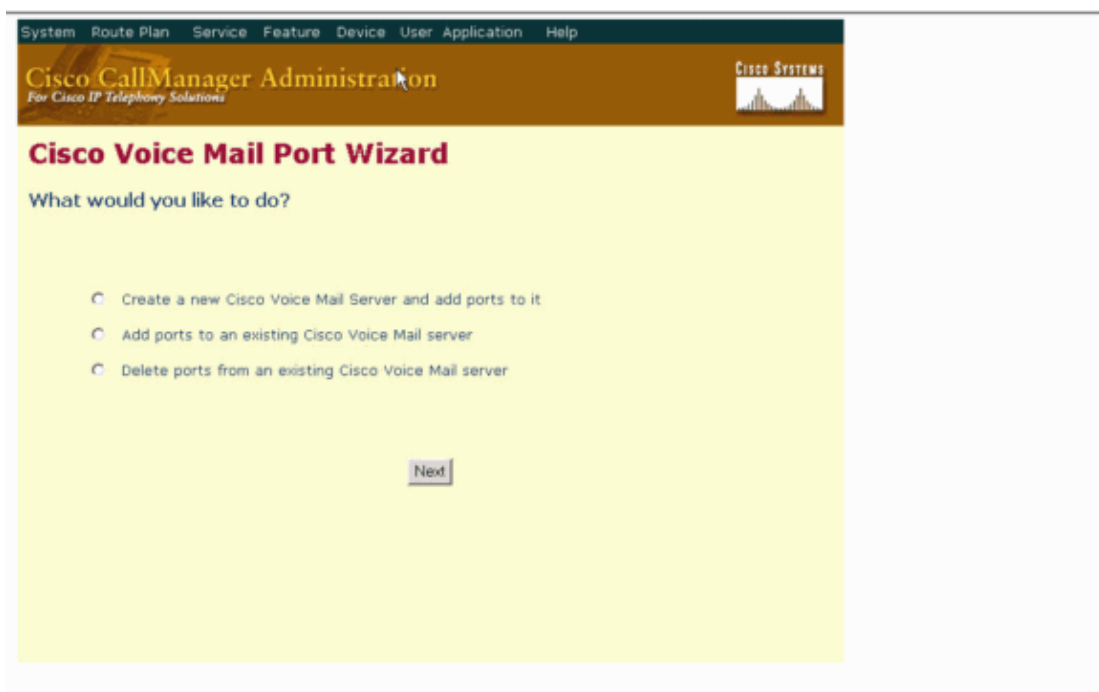
2. Enter a Cisco Voice Mail Server name, such as AvayaUM3, and click **Next**.
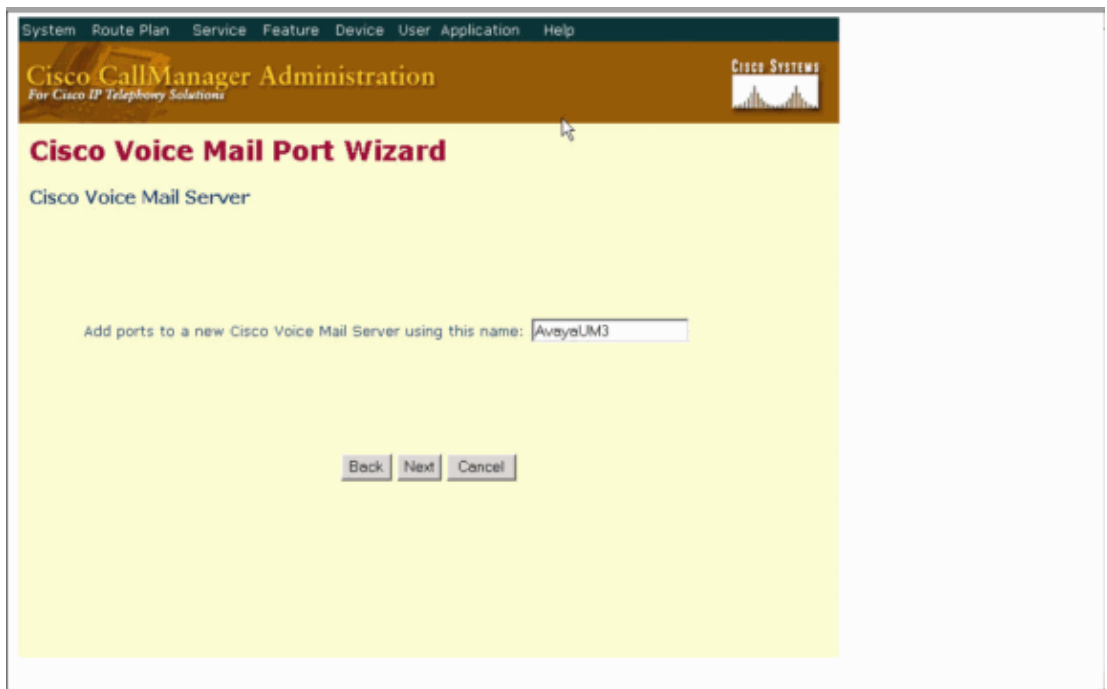
3. Select the number of Voice Mail Ports desired and click **Next**.
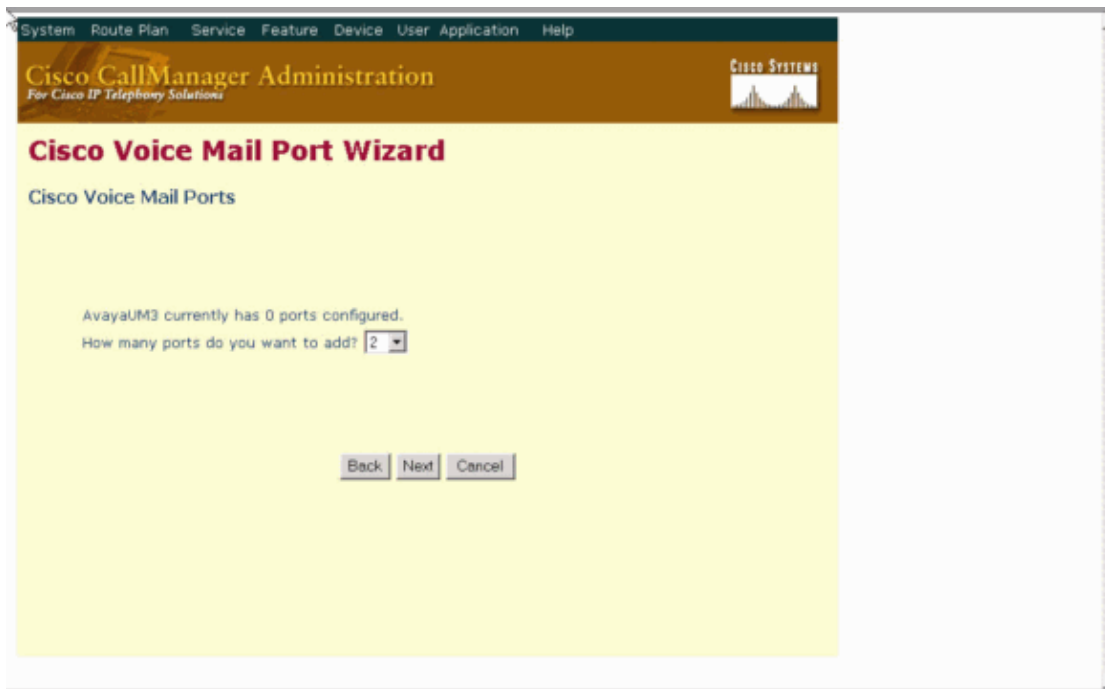
4. Enter a Description and Device Pool for the Voice Mail Ports. In the sample configuration, Avaya VMailPorts was entered as the description and Default as the device pool.

```
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
             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
```

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
             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
```

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.

```
                              TRUNK GROUP
                             Administered Members (min/max):   1/23
GROUP MEMBER ASSIGNMENTS         Total Administered Members:  23

        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           Max number of NCA TSC: 10
                        Primary D-Channel: 01A0924        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:
```

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

```
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
```

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

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

                                    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
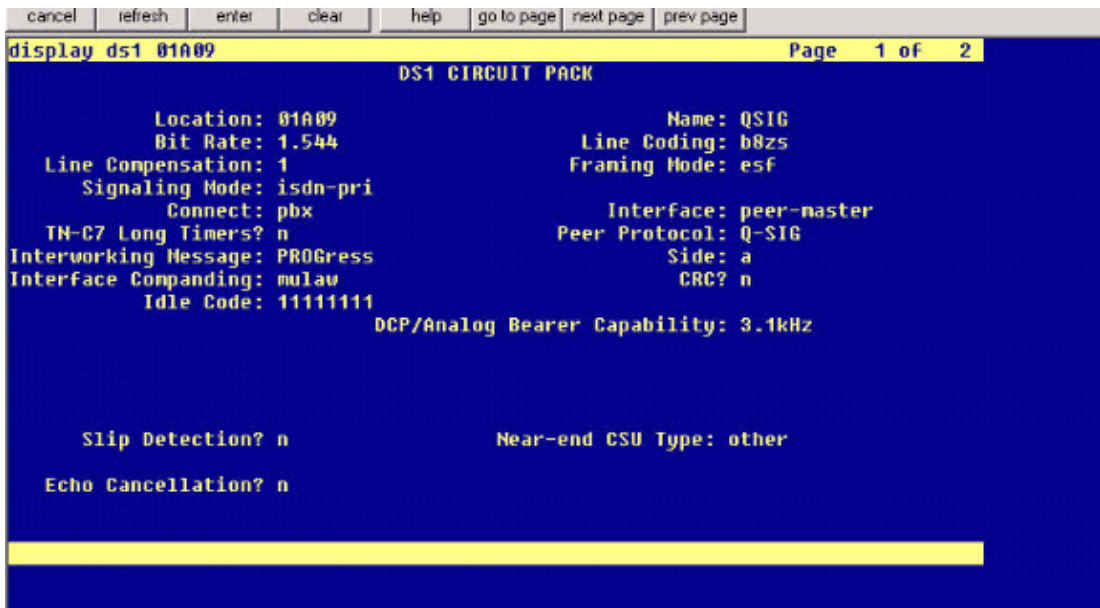   Emergency Location Ext: 7007   IP Audio Hairpinning? y

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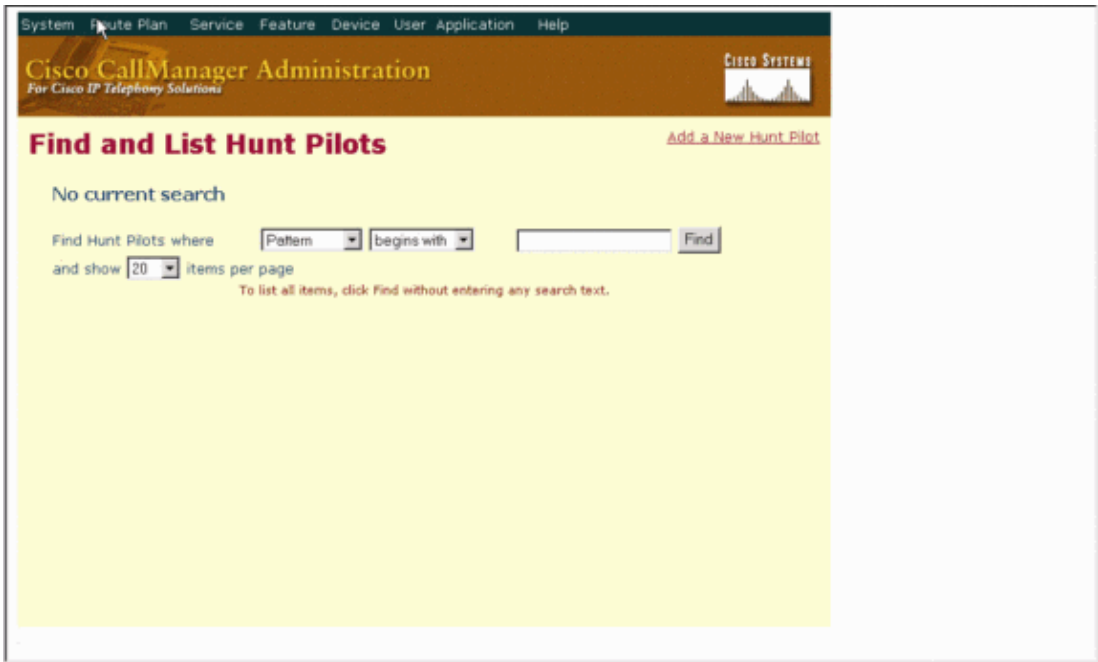
15. Enter in the Hunt Pilot, such as 4408, and select a Hunt List, such as Avaya VMail HL and click **Insert**.

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**.

```
                                 TRUNK GROUP
                          Administered Members (min/max):    1/23
GROUP MEMBER ASSIGNMENTS            Total Administered Members:  23

        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 (MWI) On/Off numbers. Included here are two screen captures for Message Waiting Indicator On/Off numbers.

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**

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