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

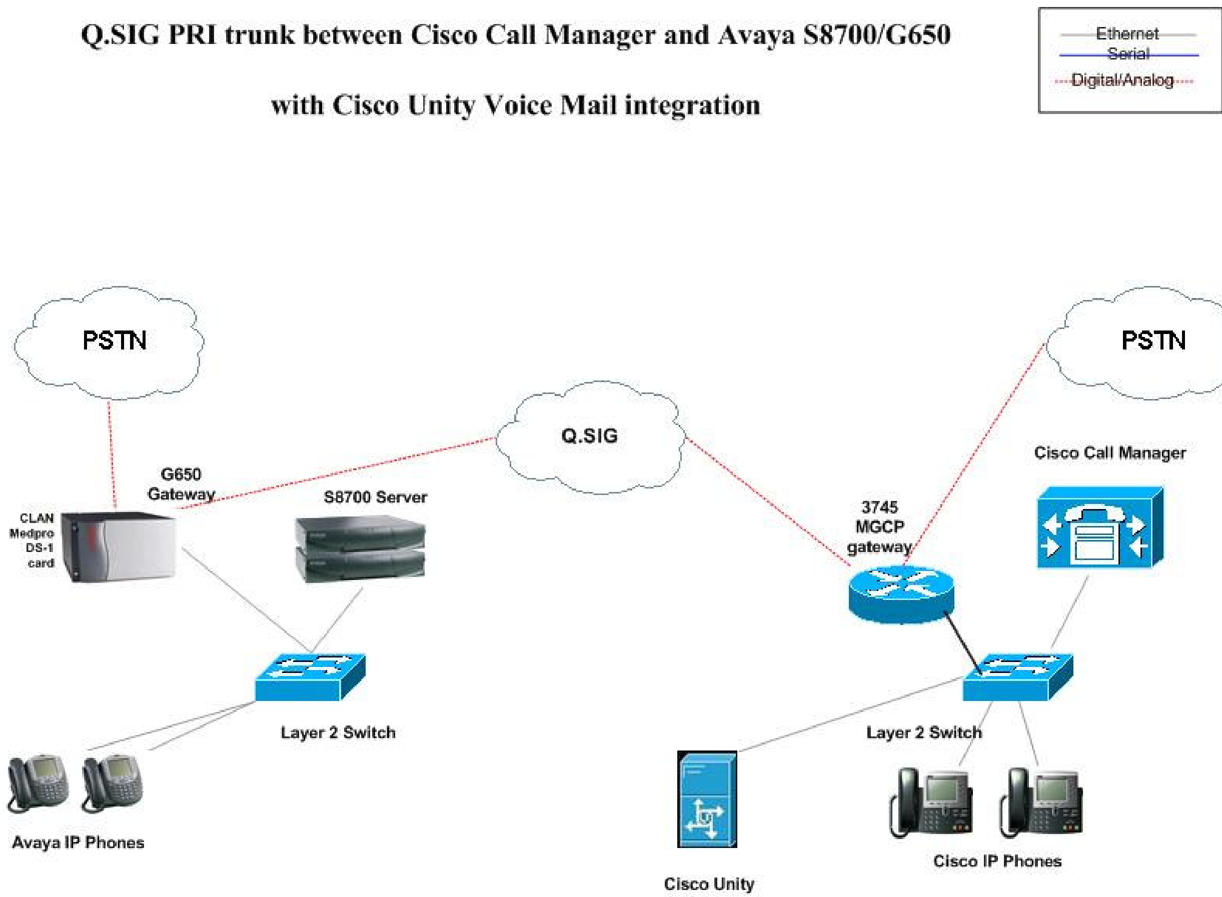
Introduction

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

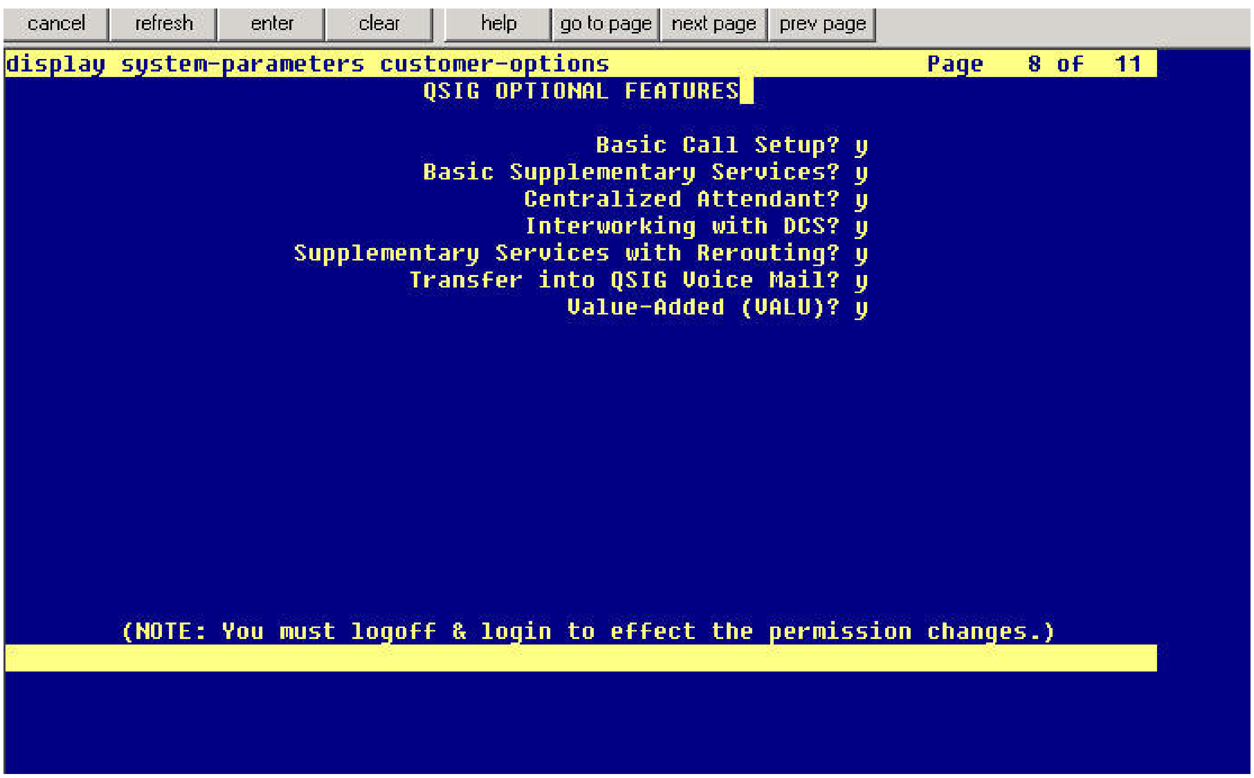


Interoperability between Cisco and Avaya IP-PBX Systems

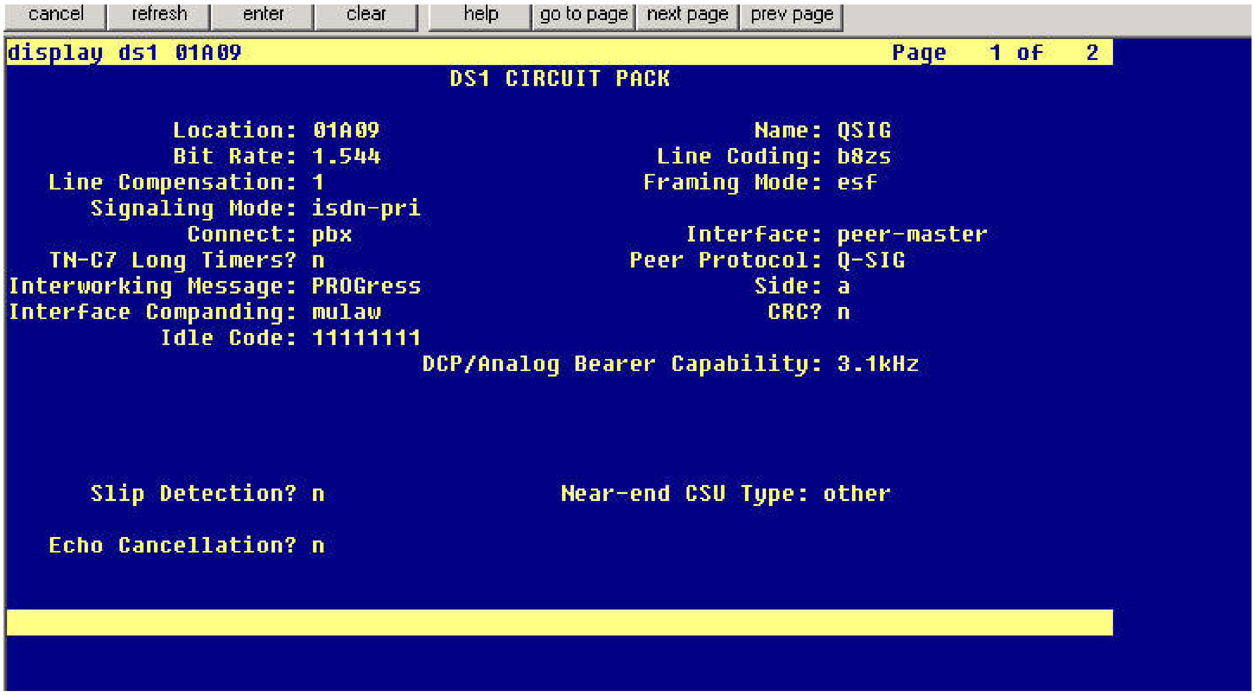
The next sections provide procedures and screen captures of how to configure the Q.SIG trunk between an Avaya S8700/G650 running Avaya Communication Manager 2.0 and a Cisco Call Manager platform running Call Manager version 4.1(2) with the Cisco 3745 MGCP device providing the physical ISDN PRI connection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

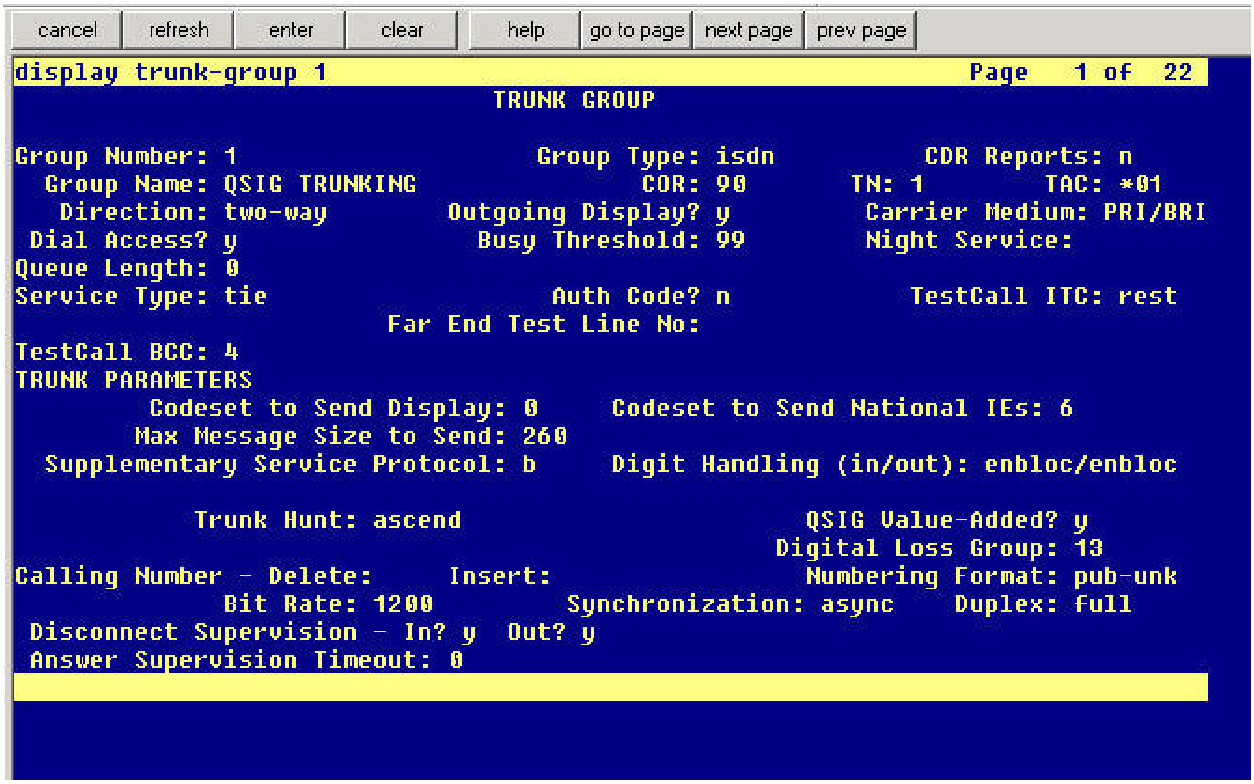


2. Configure the DS-1 card for Q.SIG PRI

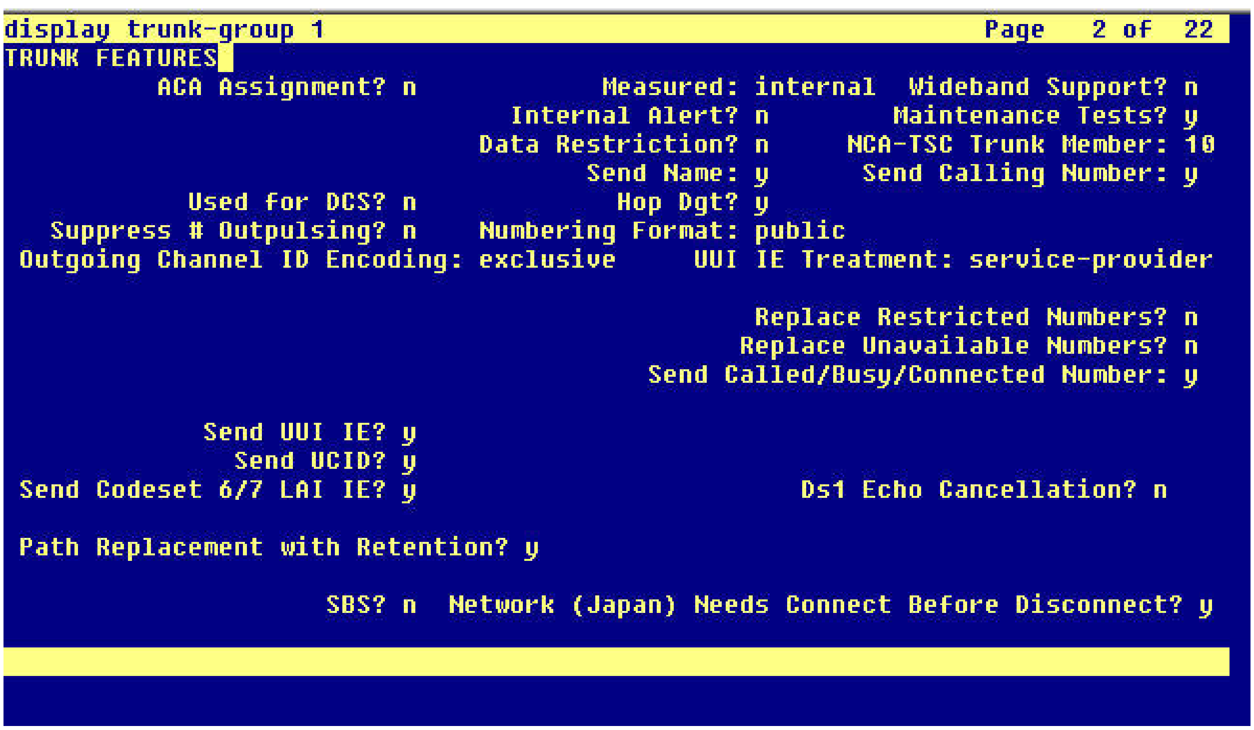


```
cancel | refresh | enter | clear | help | go to page | next page | prev page
display ds1 01A09                                              Page   1 of   2
                               DS1 CIRCUIT PACK

            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
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 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          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. 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

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

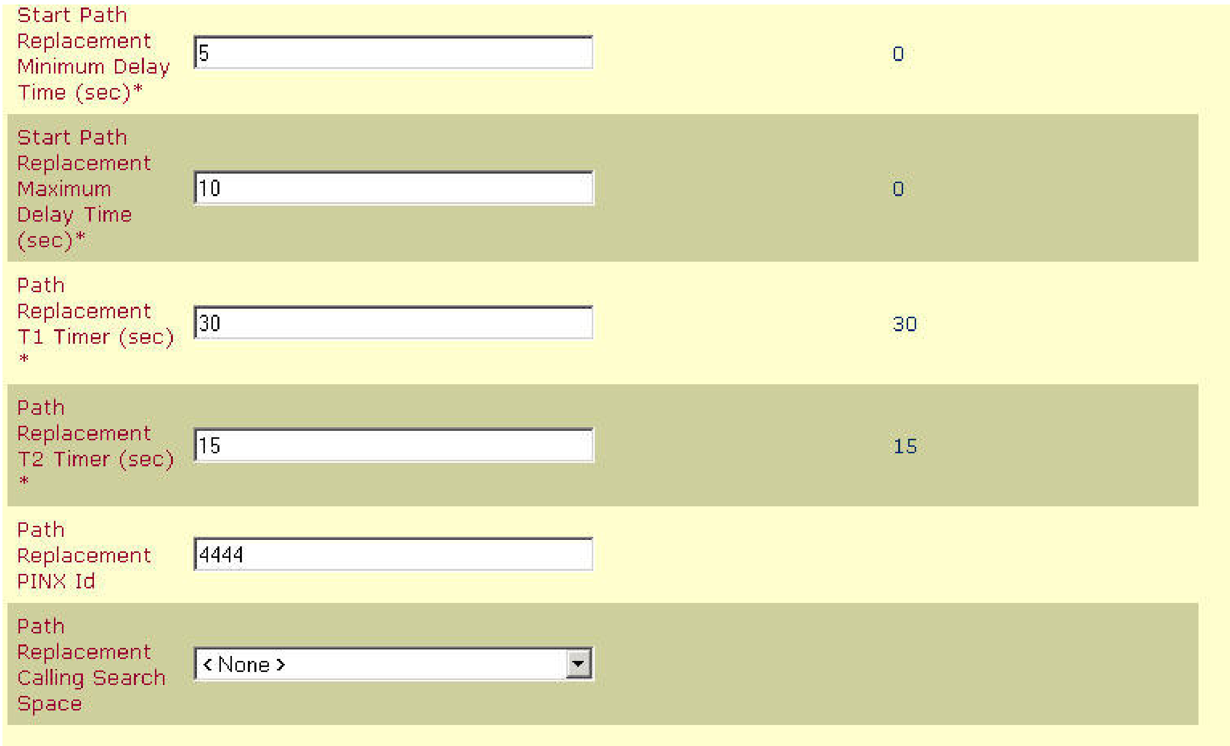
```
display station 7007                                         Page   2 of   4
                                   STATION
FEATURE OPTIONS
           LWC Reception: spe            Auto Select Any Idle Appearance? n
          LWC Activation? y                       Coverage Msg Retrieval? y
  LWC Log 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
 Emergency Location Ext: 7007                          IP Audio Hairpinning? y
```

Procedure on Cisco Call Manager

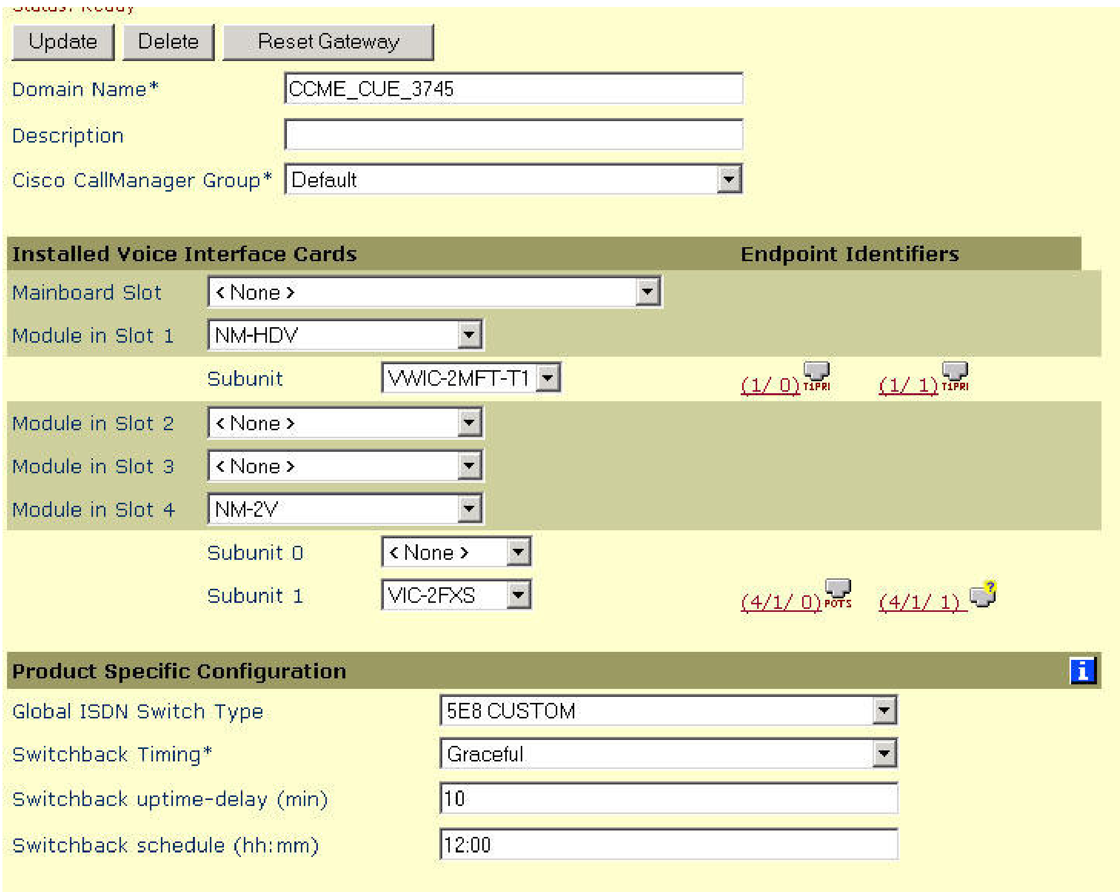
1. Under Service parameters, make sure that the Start Path Replacement Minimum and Maximum time values are set appropriately to prevent any issues (such as hair pinning). The next two screen captures relate to the Q.SIG Service Parameters setting:

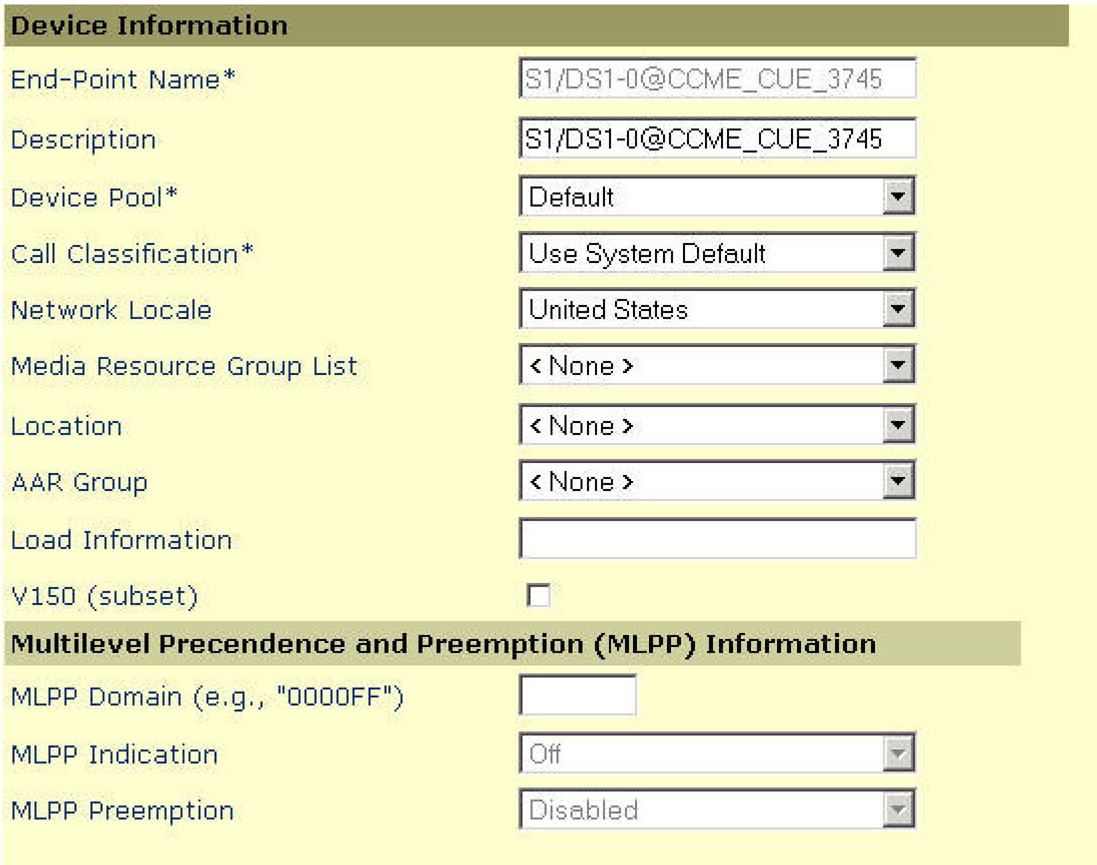
Clusterwide Parameters (Feature - Path Replacement)

Parameter Name	Parameter Value	Suggested Value
Path Replacement Enabled*	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



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.


Update
Delete
Reset Gateway
Domain Name*
CCME_CUE_3745
Description
Cisco CallManager Group*
Default
Installed Voice Interface Cards
Endpoint Identifiers
Mainboard Slot
< None >
Module in Slot 1
NM-HDV
Subunit
VWIC-2MFT-T1
(1/ 0)
(1/ 1)
Module in Slot 2
< None >
Module in Slot 3
< None >
Module in Slot 4
NM-2V
Subunit 0
< None >
Subunit 1
VIC-2FXS
(4/1/ 0)
POTS
(4/1/ 1)
Product Specific Configuration
Global ISDN Switch Type
5E8 CUSTOM
Switchback Timing*
Graceful
Switchback uptime-delay (min)
10
Switchback schedule (hh:mm)
12:00


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

- [x] Inhibit restarts at PRI initialization
- [ ] Enable status poll

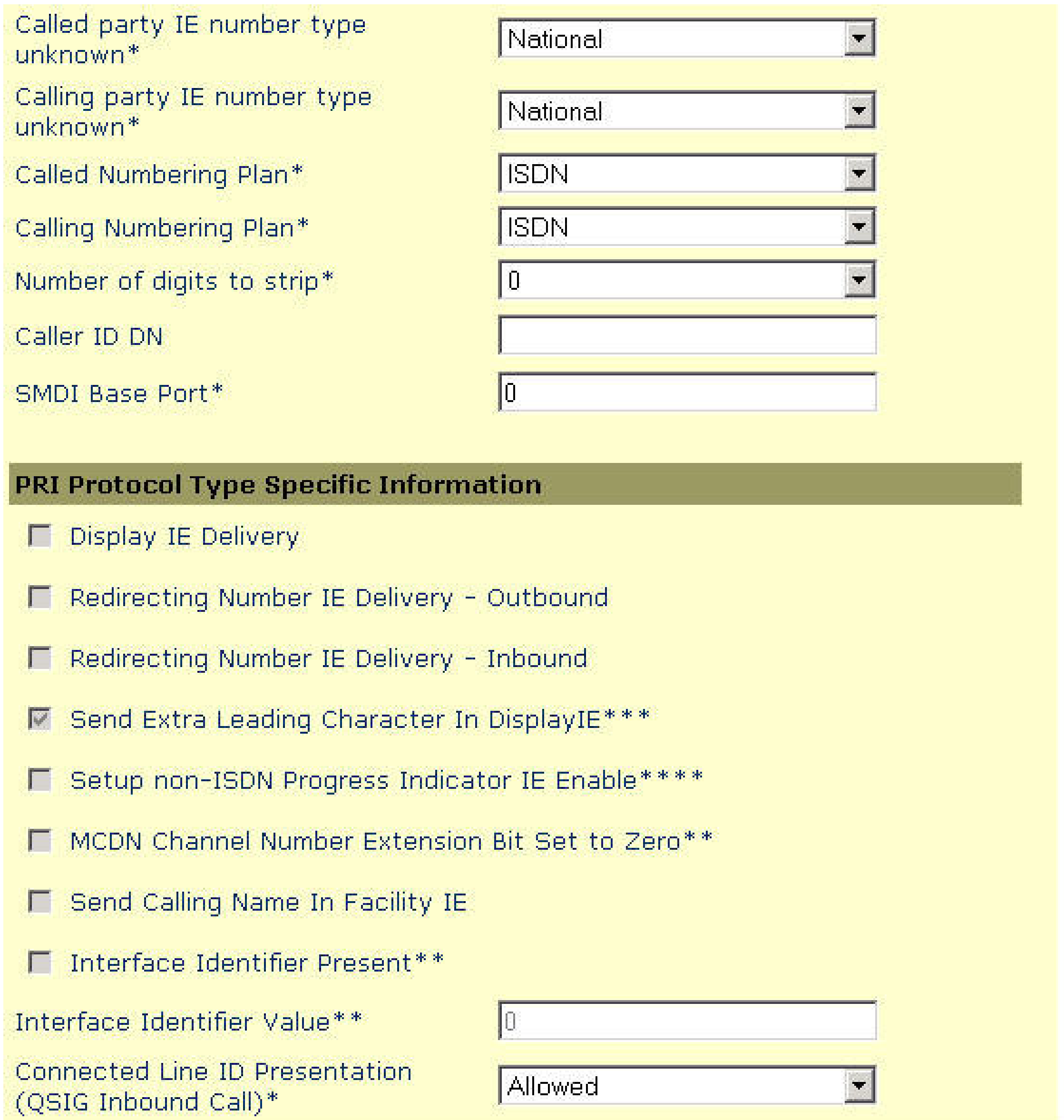
Call Routing Information

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

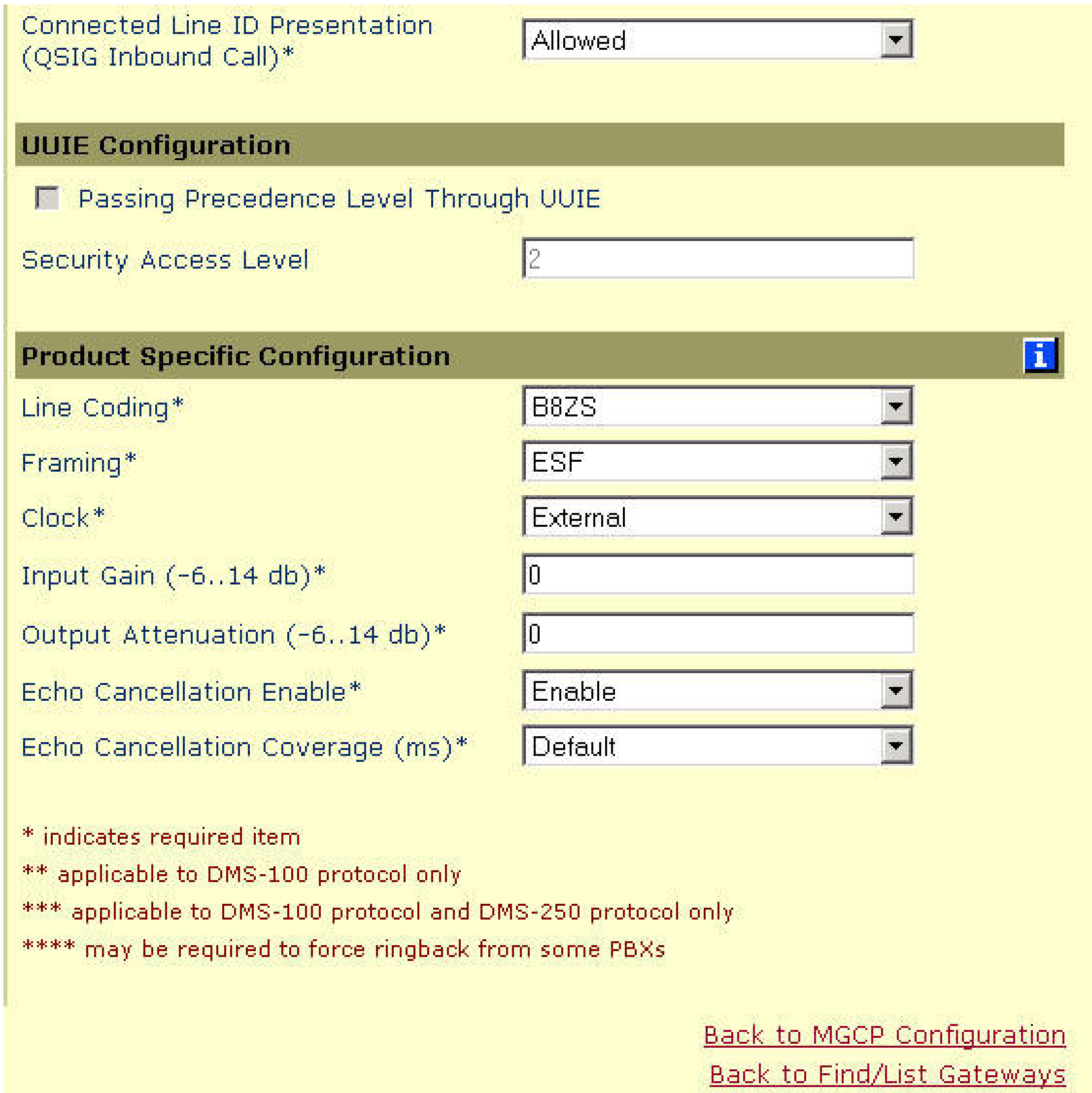


Called party IE number type unknown*	National
Calling party IE number type unknown*	National
Called Numbering Plan*	ISDN
Calling Numbering Plan*	ISDN
Number of digits to strip*	0
Caller ID DN	
SMDI Base Port*	0

PRI Protocol Type Specific Information

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

Interface Identifier Value**	0
Connected Line ID Presentation (QSIG Inbound Call)*	Allowed



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

Cisco 3745 Configuration

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

CCME_CUE_3745#sh vers

Cisco Internetwork Operating System Software

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

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

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

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

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

ROM: System Bootstrap, Version 12.2(8r)T2, RELEASE SOFTWARE (fc1)

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

CCME_CUE_3745 uptime is 39 minutes

System returned to ROM by reload

System image file is "flash:c3745-is-mz.122-15.ZJ3.bin"

cisco 3745 (R7000) processor (revision 2.0) with 246784K/15360K bytes of memory.

Processor board ID JMX0814L3E2

R7000 CPU at 350Mhz, Implementation 39, Rev 3.3, 256KB L2, 2048KB L3 Cache

Bridging software.

X.25 software, Version 3.0.0.

SuperLAT software (copyright 1990 by Meridian Technology Corp).

Primary Rate ISDN software, Version 1.1.

2 FastEthernet/IEEE 802.3 interface(s)

25 Serial network interface(s)

1 terminal line(s)

2 Channelized T1/PRI port(s)

1 ATM AIM(s)

2 Voice FXS interface(s)

2 Voice E & M interface(s)

1 cisco service engine(s)

DRAM configuration is 64 bits wide with parity disabled.

151K bytes of non-volatile configuration memory.

125184K bytes of ATA System CompactFlash (Read/Write)

```
Configuration register is 0x2102


CCME_CUE_3745#sh run
Building configuration...


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

```
!
no voice hpi capture buffer
no voice hpi capture destination
!
!
ccm-manager mgcp
ccm-manager music-on-hold
ccm-manager config server 172.28.221.18
ccm-manager config
mta receive maximum-recipients 0
!
!
controller T1 1/0
 framing esf
 linecode b8zs
 pri-group timeslots 1-24 service mgcp
!
controller T1 1/1
 framing sf
 linecode ami
!
!
!
interface FastEthernet0/0
 description CCME-CUE-3745_to_cat3550
 no ip address
 duplex auto
 speed auto
```

```
!
interface FastEthernet0/0.1
 encapsulation dot1Q 99
!
interface FastEthernet0/0.2
 description NEW_S8700_G650
 encapsulation dot1Q 300
 ip address 172.28.221.49 255.255.255.240
 ip helper-address 172.28.221.19
 h323-gateway voip bind srcaddr 172.28.221.49
!
interface FastEthernet0/0.3
 description MODULAR_MESSAGING_SOLUTION
 encapsulation dot1Q 900
 ip address 172.28.221.129 255.255.255.240
 ip helper-address 172.28.221.19
!
interface FastEthernet0/0.4
 encapsulation dot1Q 301
 ip address 10.1.3.1 255.255.255.128
 ip helper-address 172.28.221.19
!
interface FastEthernet0/0.5
 encapsulation dot1Q 302
 ip address 10.1.3.129 255.255.255.128
 ip helper-address 172.28.221.19
!
interface FastEthernet0/0.6
```

```
 encapsulation dot1Q 90
 ip address 90.1.1.254 255.255.255.0
 ip helper-address 172.28.221.19
!
interface Serial0/0
 description CCME-CUE-3745_to_3600
 ip address 25.0.0.1 255.0.0.0
 clockrate 256000
 no fair-queue
!
interface Serial1/0:23
 no ip address
 no logging event link-status
 isdn switch-type primary-qsig
 isdn incoming-voice voice
 isdn bind-l3 ccm-manager
 isdn bchan-number-order ascending
 no cdp enable
!
interface Service-Engine2/0
 no ip address
 shutdown
!
router eigrp 100
 network 10.0.0.0
 network 25.0.0.0
 network 90.0.0.0
 network 172.28.0.0
```

```
 auto-summary
!
ip http server
ip classless
!
call rsvp-sync
!
voice-port 1/0:23
!
voice-port 4/0/0
!
voice-port 4/0/1
!
voice-port 4/1/0
!
voice-port 4/1/1
!
mgcp
mgcp call-agent 172.28.221.18 2427 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode out-of-band
mgcp rtp unreachable timeout 1000 action notify
mgcp package-capability rtp-package
no mgcp package-capability res-package
mgcp package-capability sst-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax t38 inhibit
mgcp rtp payload-type g726r16 static
```

```
!
mgcp profile default
!
!
!
dial-peer cor custom
!
dial-peer voice 1 pots
 application mgcpapp
 port 1/0:23
!
dial-peer voice 999410 pots
 application mgcpapp
 port 4/1/0
!
!
line con 0
 password cisco
 login
line 65
 flush-at-activation
 no activation-character
 no exec
 transport preferred none
 transport input all
line aux 0
line vty 0 4
 password cisco
```

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

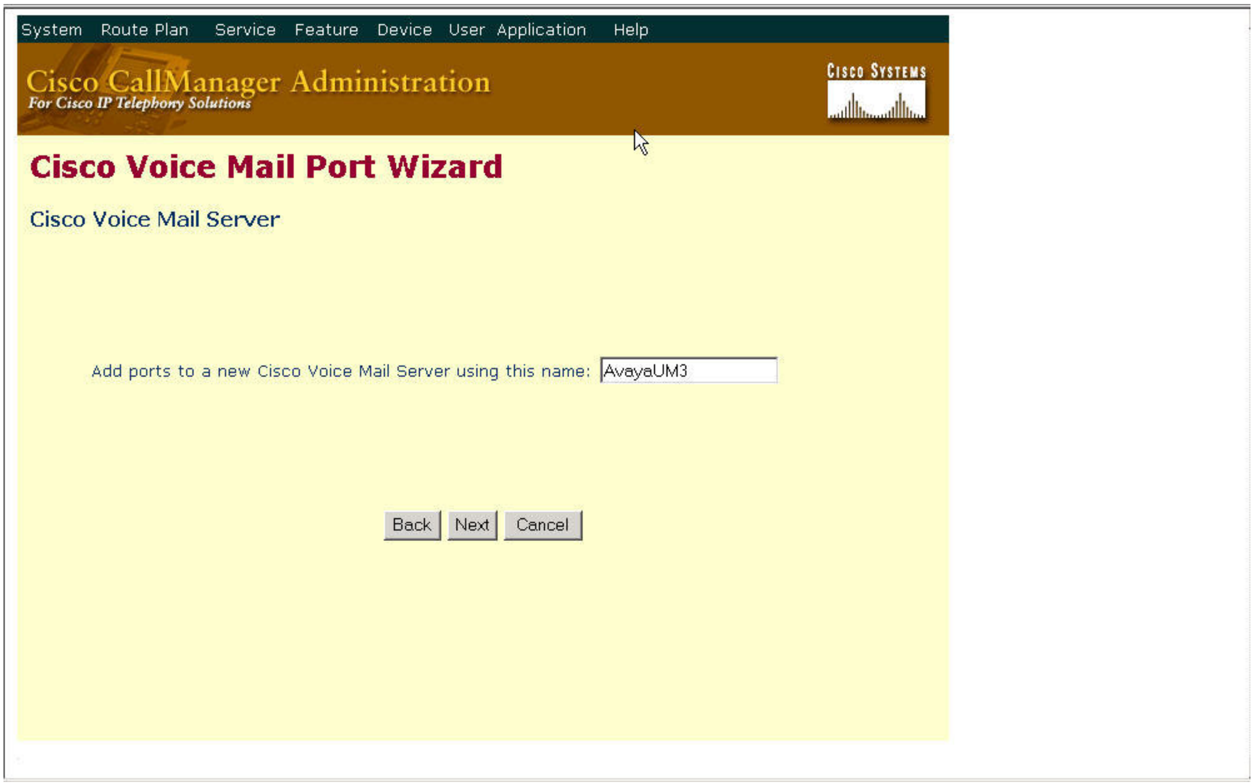
Integration of Cisco Unity Voice Mail to support Cisco and Avaya IP Phones

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.

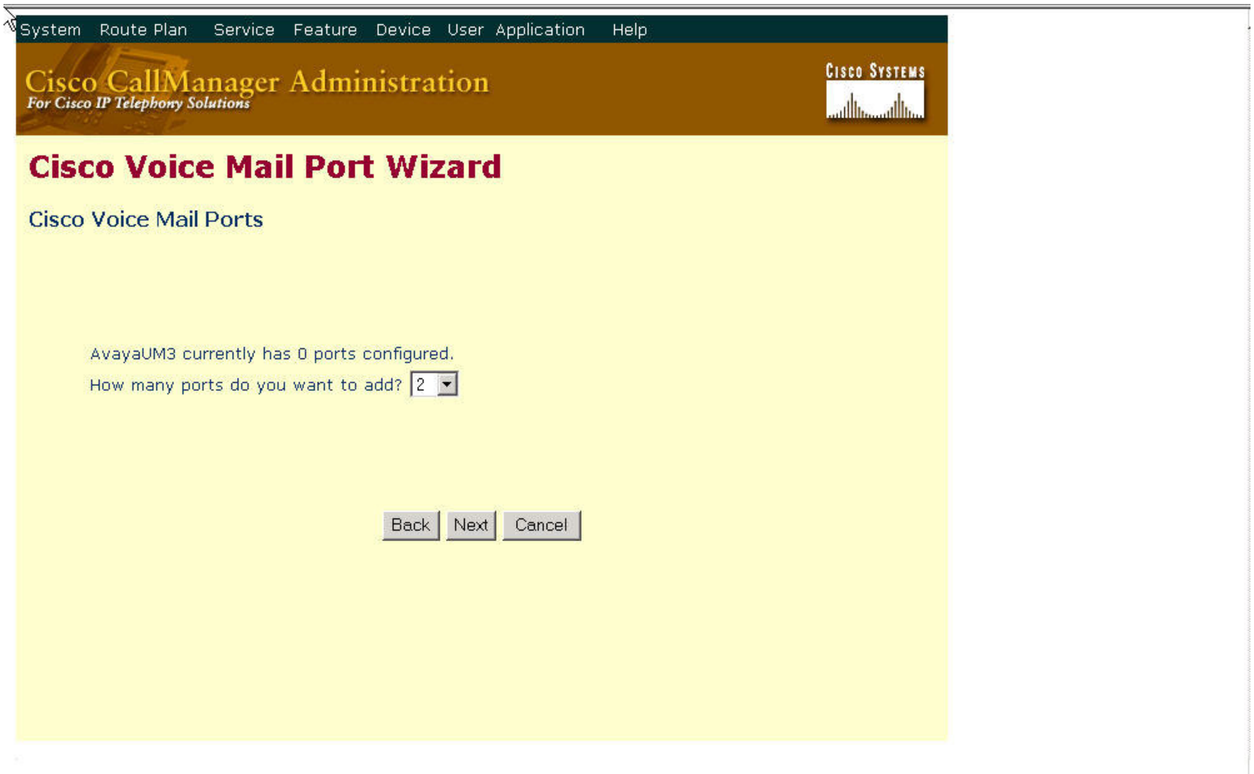
Procedure for adding Cisco Unity to Cisco Call Manager

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.

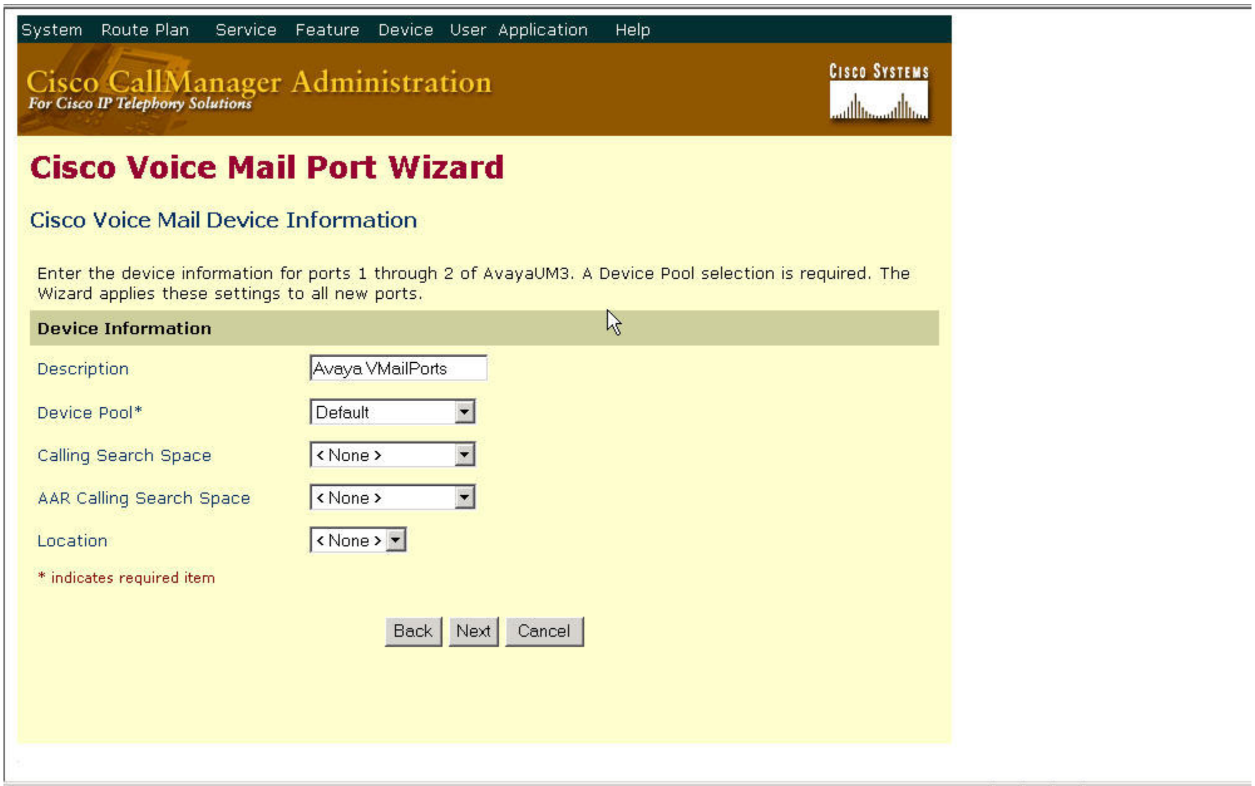
2. Enter a Cisco Voice Mail Server name, such as AvayaUM3, and click Next.



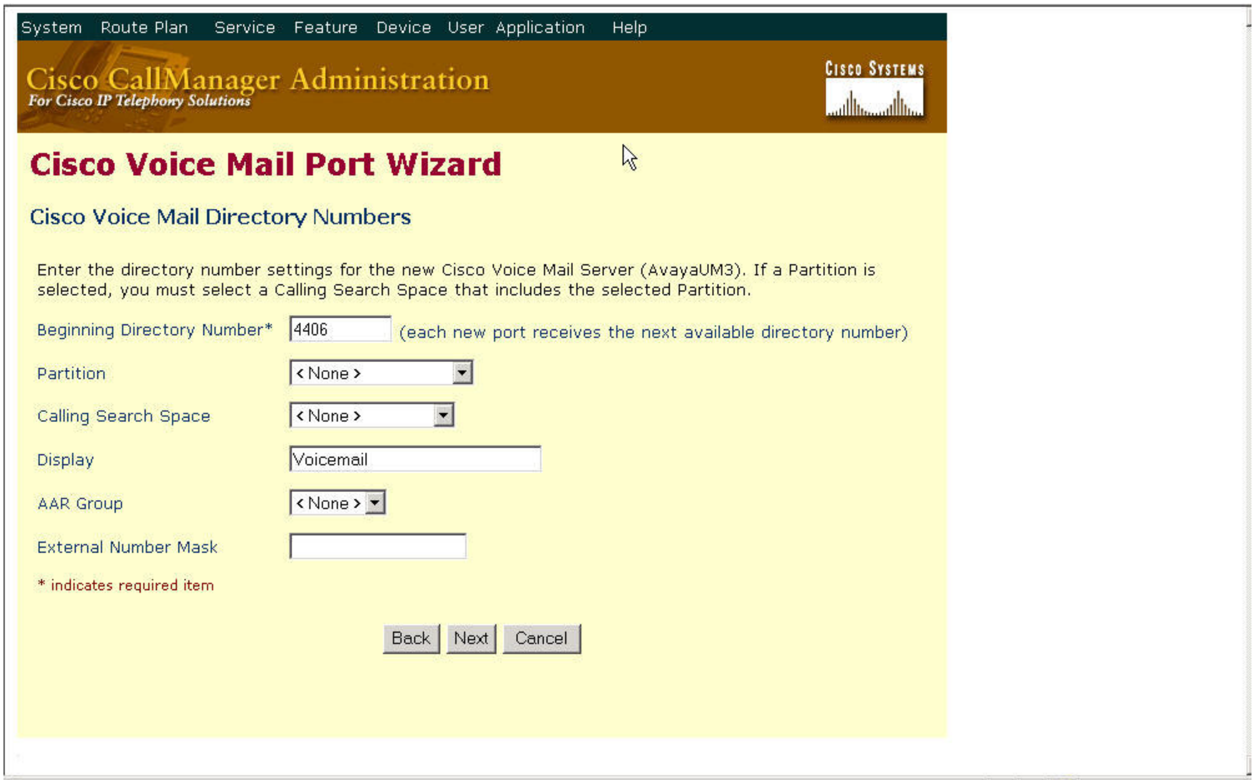
3. Select the Voice Mail Ports you want configured and click Next.



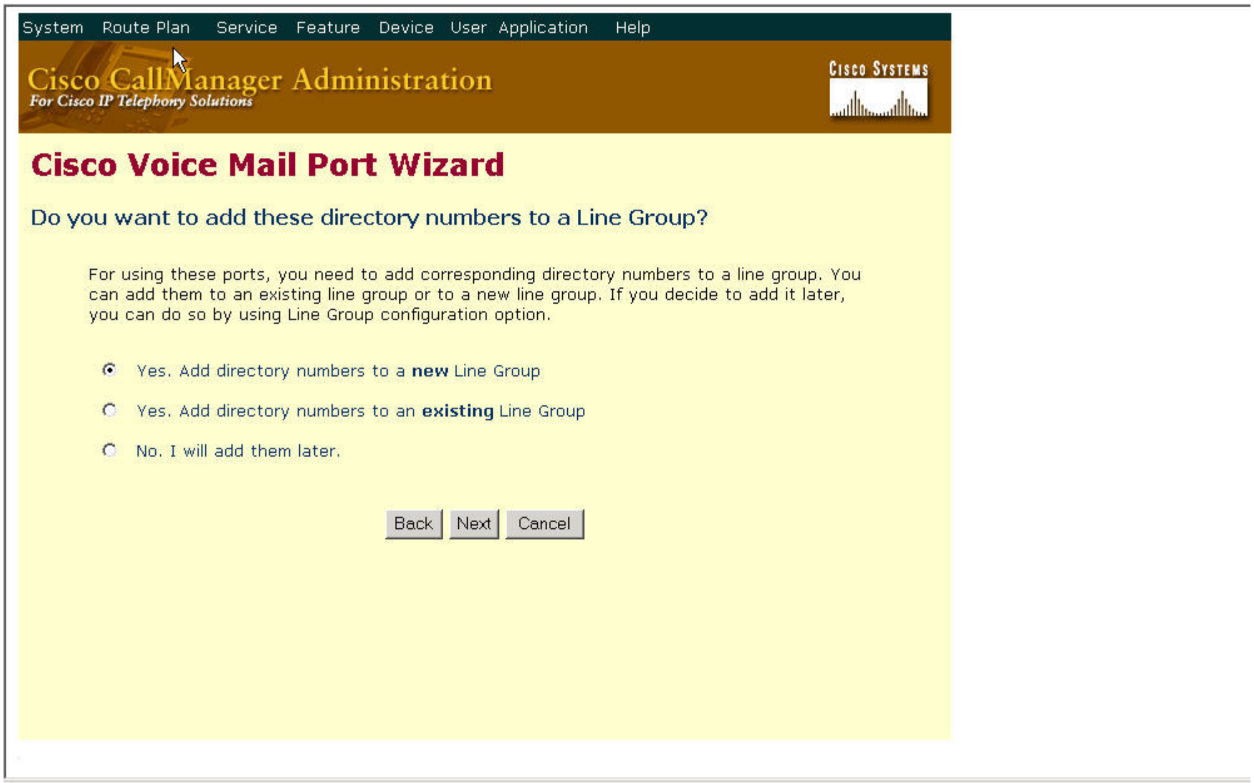
4. Enter a Description and Device Pool for the Voice Mail Ports. In our configuration we entered Avaya VMailPorts as the description and Default as the device pool.



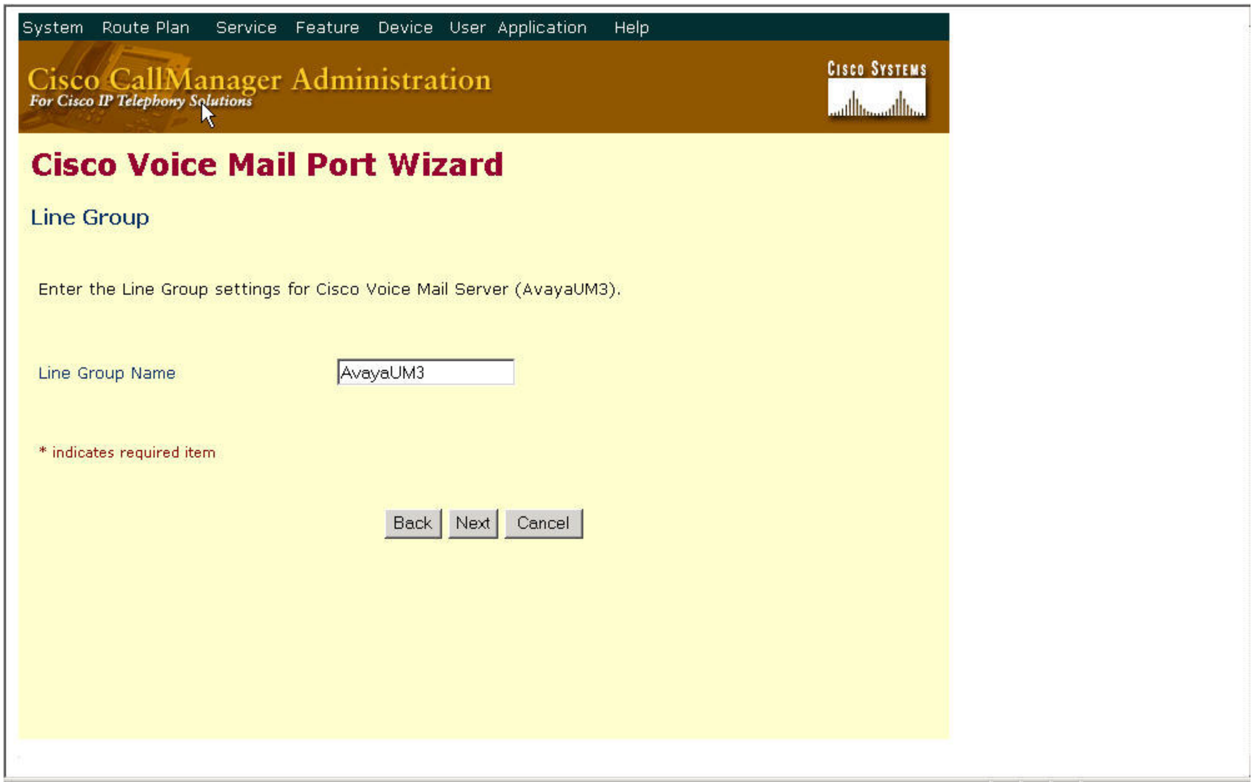
5. Enter the Beginning Directory Number, such as 4406, and the Display, such as Voicemail and click Next.



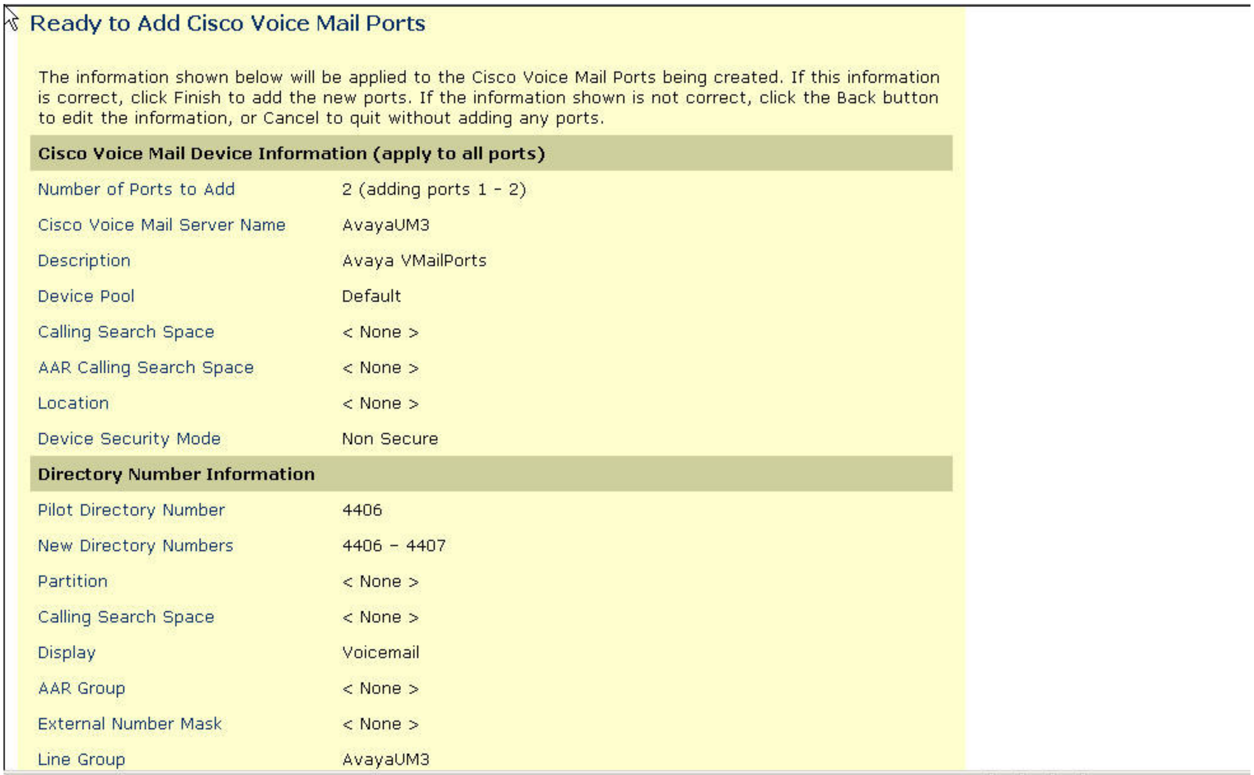
6. The next screen will ask Do you want to add these directory numbers to a Line Group? Select Yes. Add directory numbers to a new Line Group and click Next.



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



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

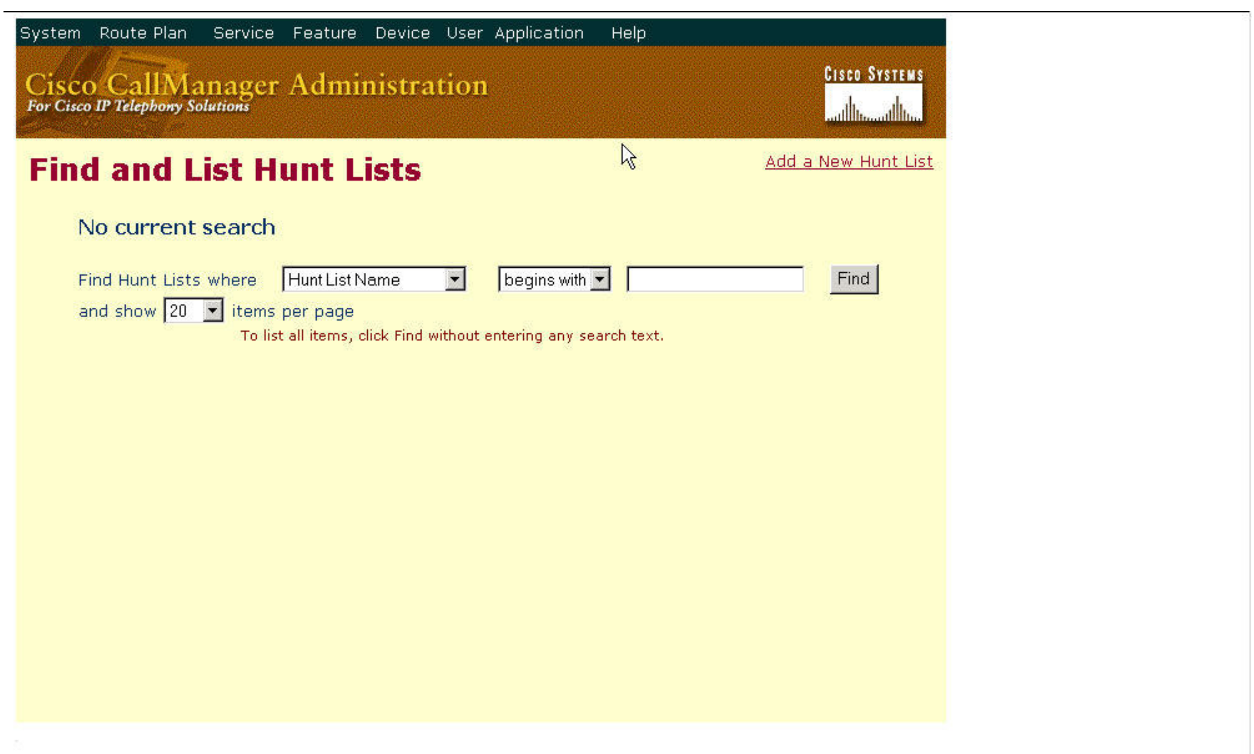


Ready to Add Cisco Voice Mail Ports

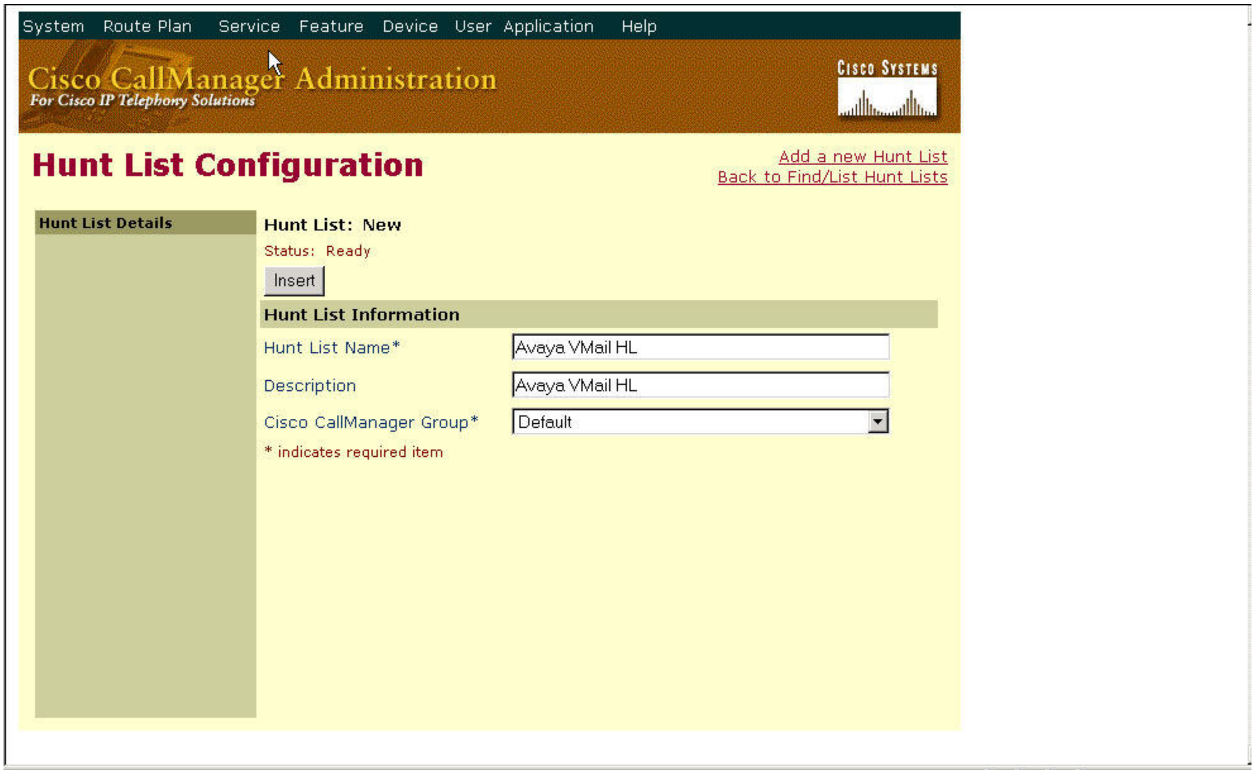
The information shown below will be applied to the Cisco Voice Mail Ports being created. If this information is correct, click Finish to add the new ports. If the information shown is not correct, click the Back button to edit the information, or Cancel to quit without adding any ports.

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

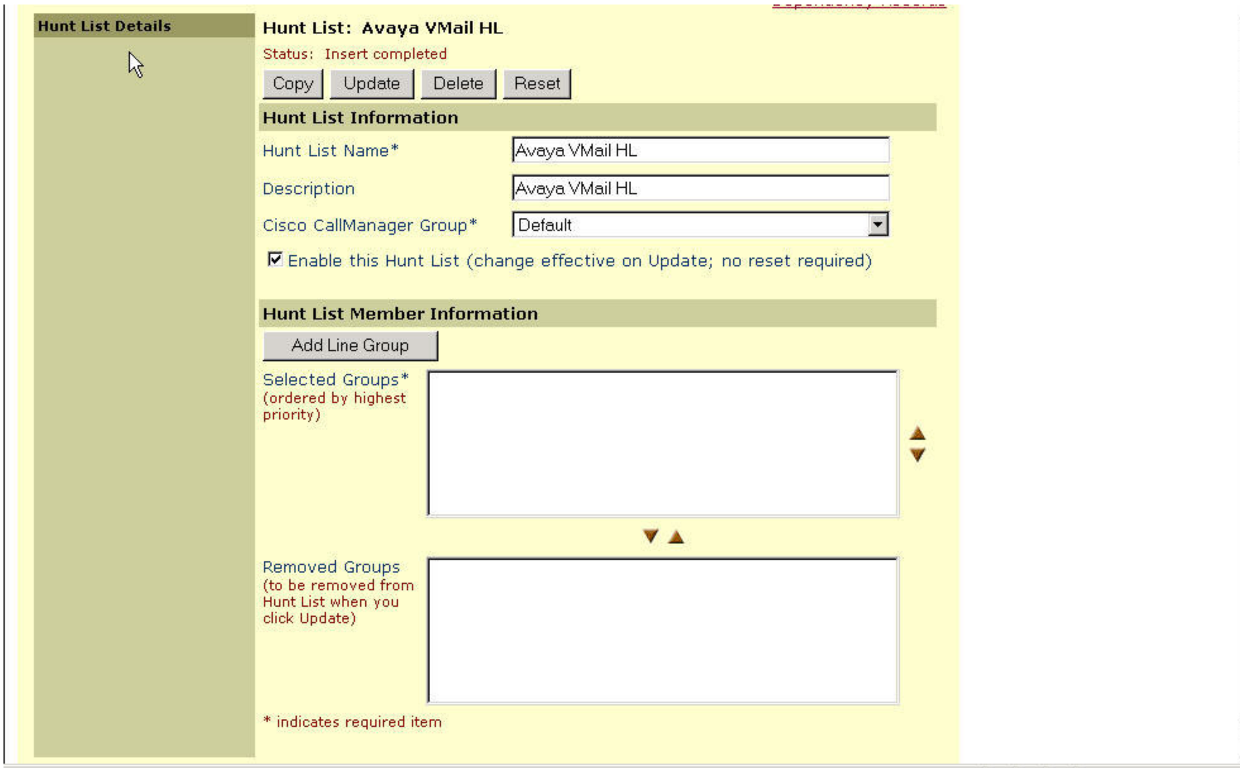
9. Click Add a New Hunt List on the Hunt List Administration web page.



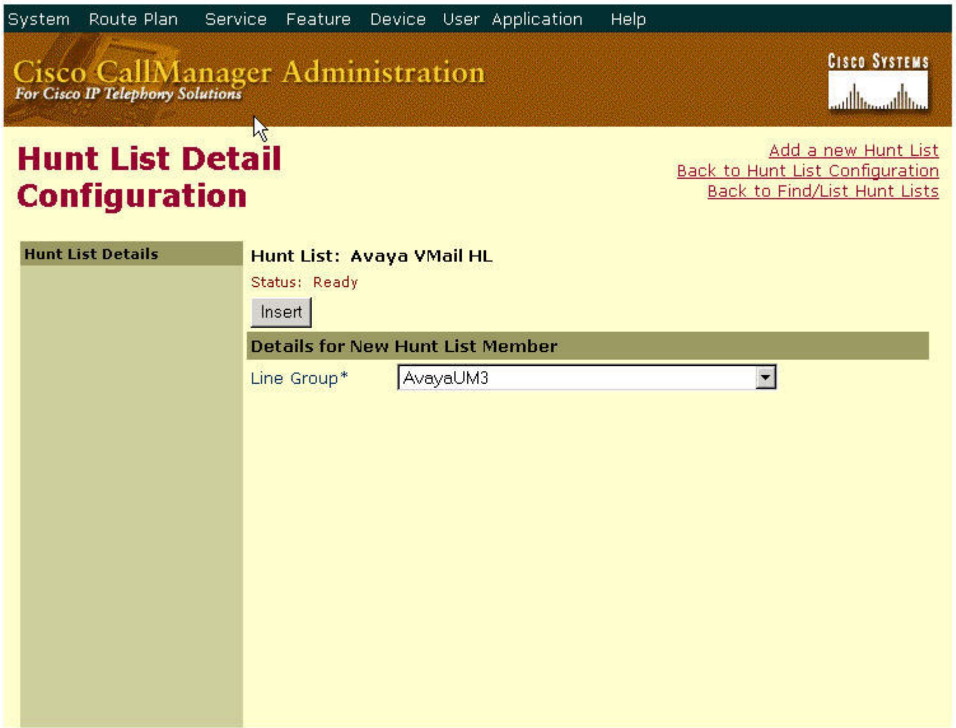
10. Enter a Hunt List Name and Description, such as Avaya VMailHL. Also select Default for the Cisco Call Manager Group.



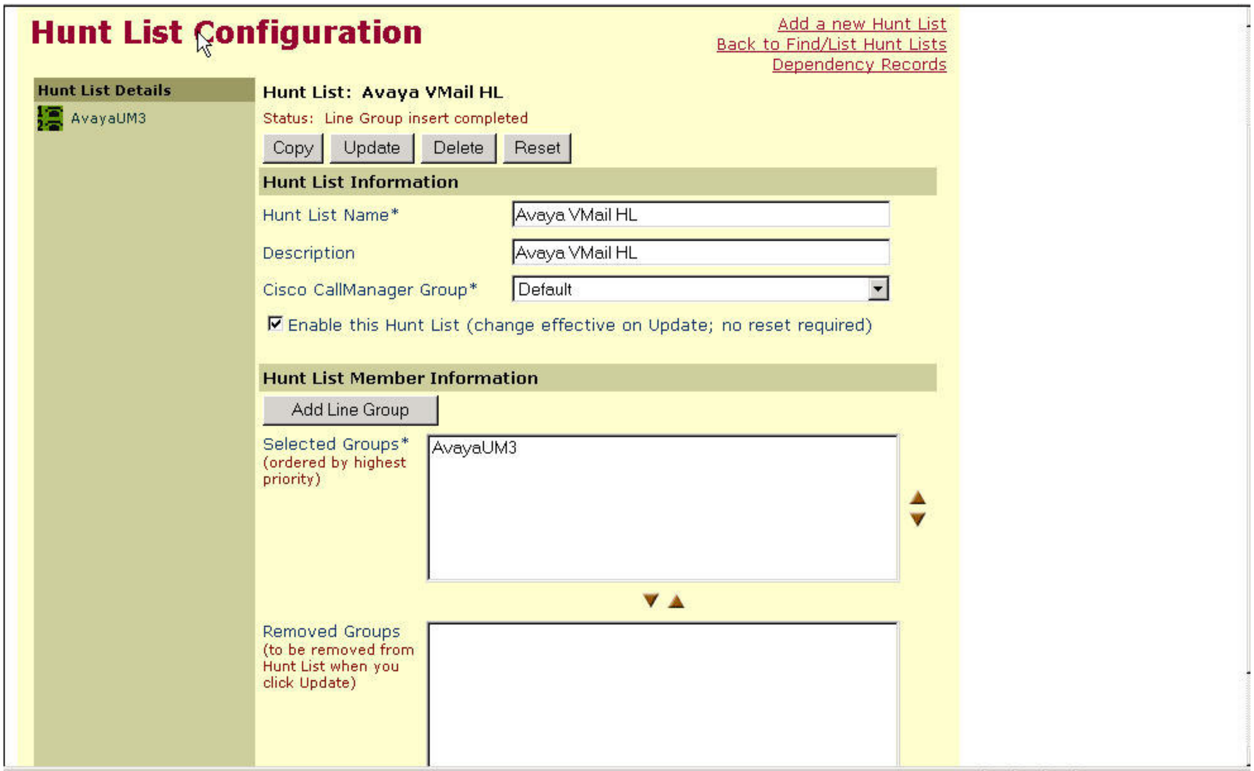
11. The following screen capture is the result of successfully adding the Hunt List. Click Add Line Group.



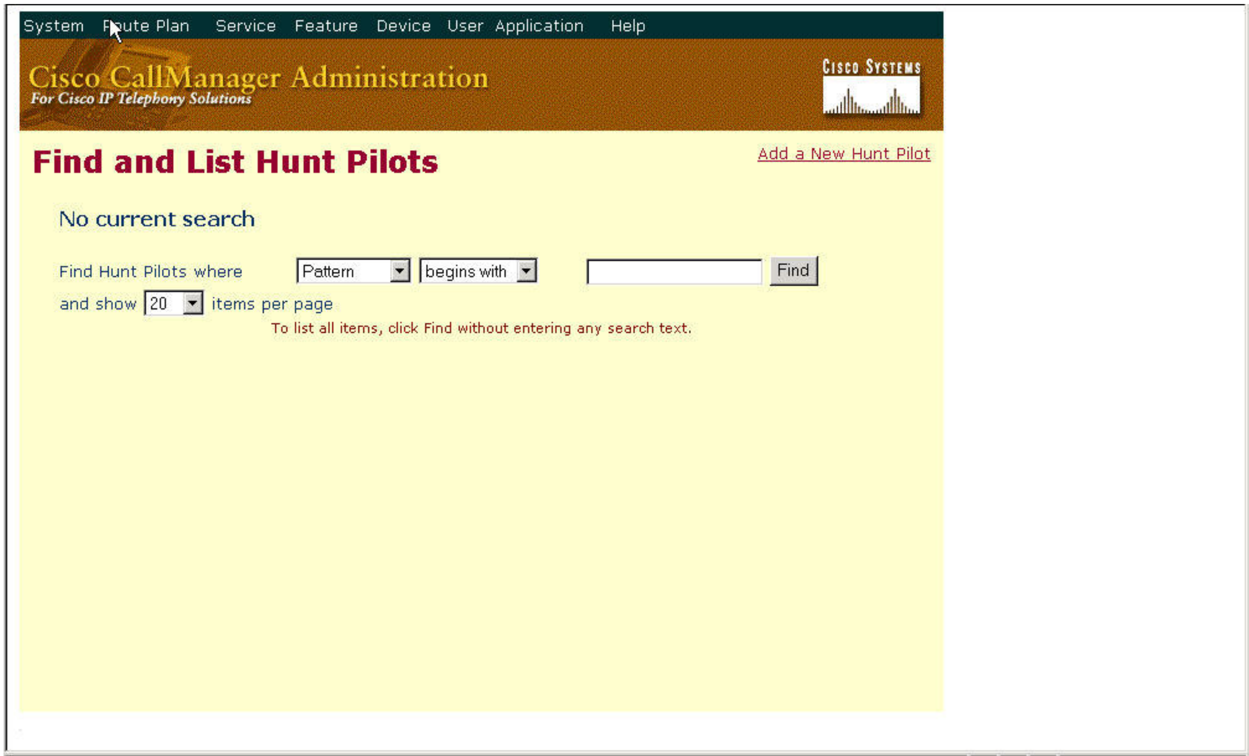
12. Select the Line Group previously configured. In our case, it's AvayaUM3.



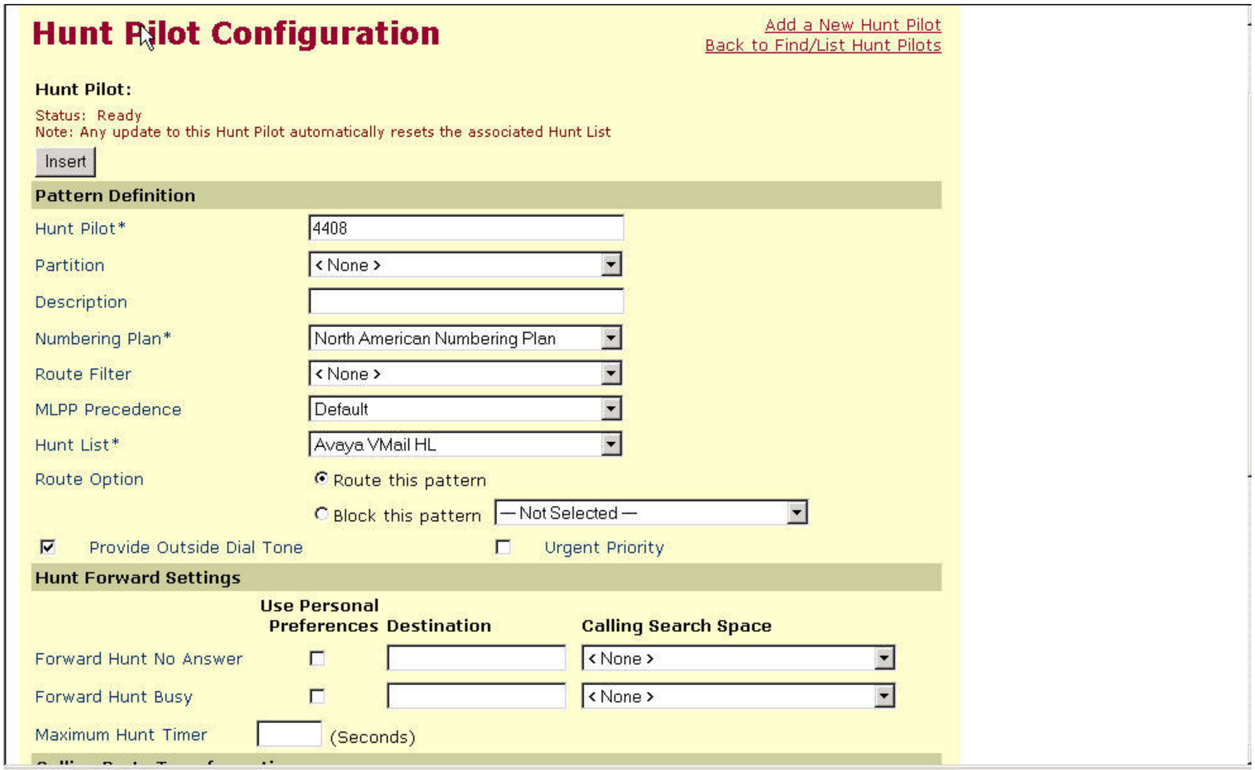
13. The next screen capture shows the result of successfully inserting the line group.



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

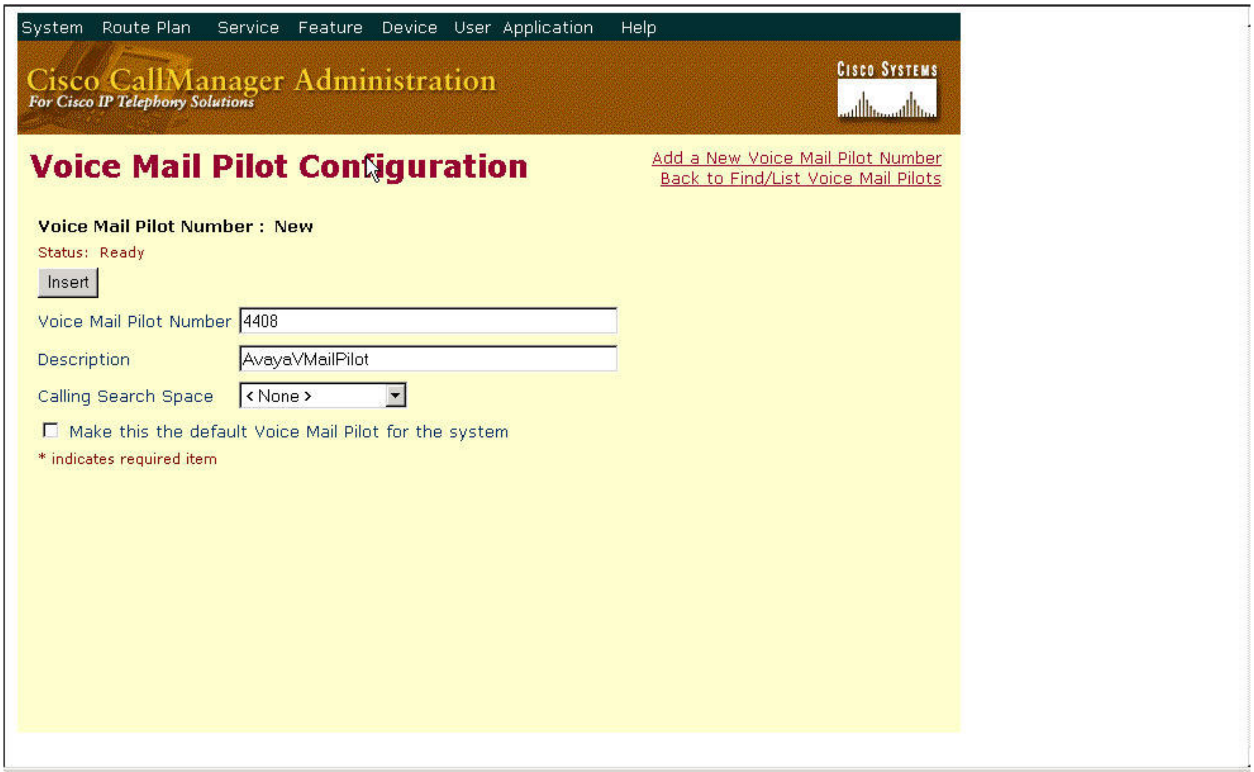


15. Enter in the Hunt Pilot, such as 4408, and select a Hunt List, such as Avaya VMail HL and click Insert.



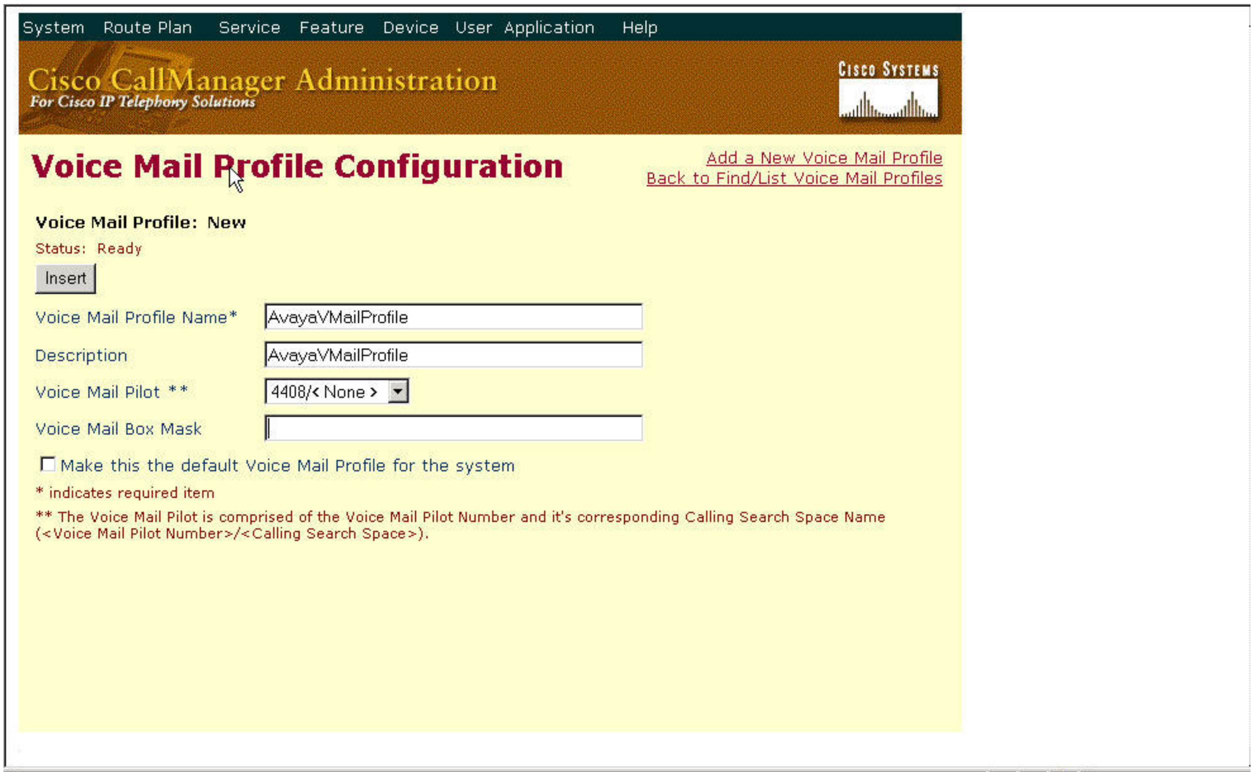
16. Go to Feature, Voice Mail, Voice Mail Pilot and click Add a New Voice Mail Pilot on the resulting screen.

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

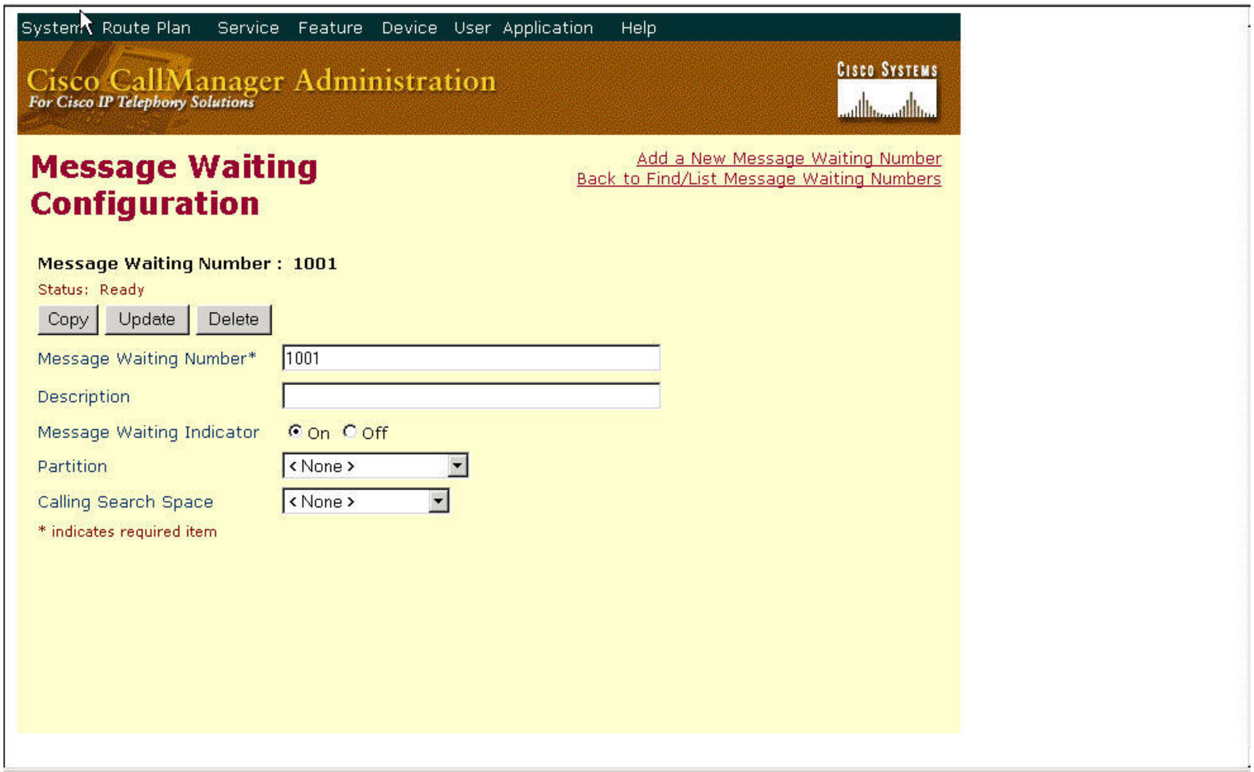


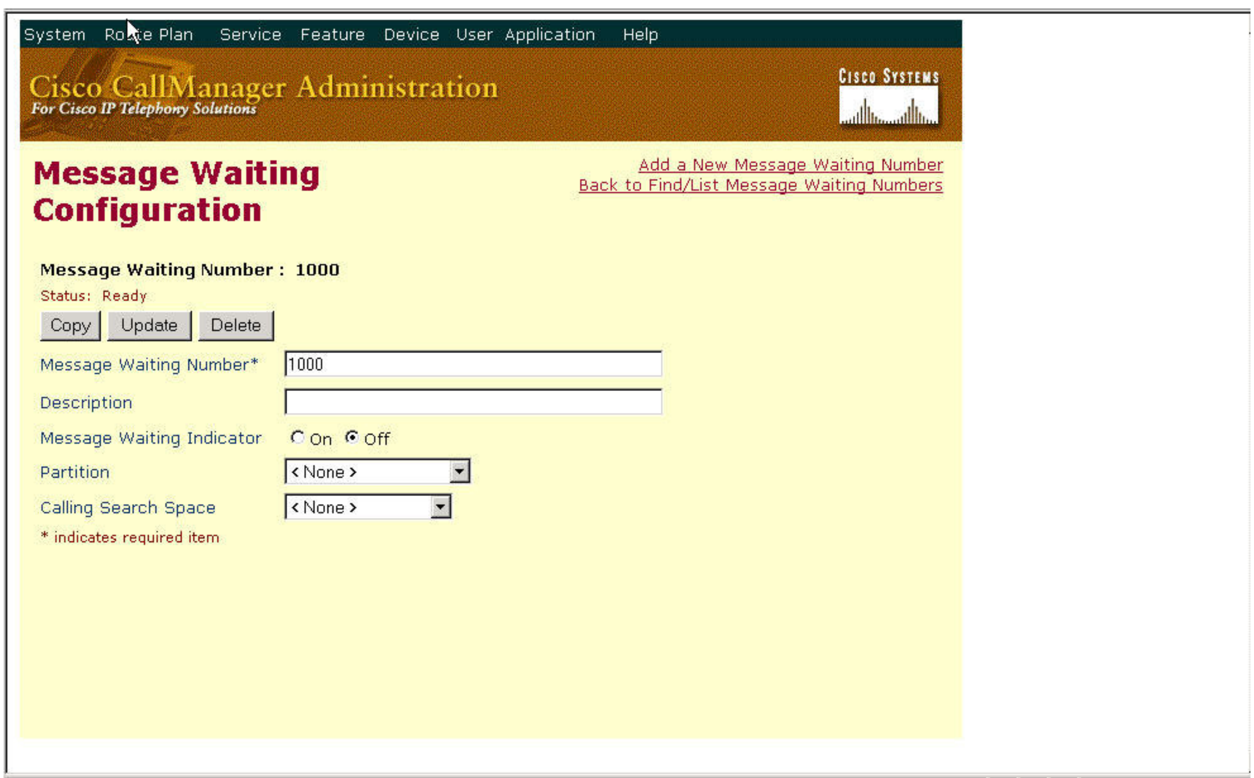
18. Next, go to Feature, Voice Mail, Voice Mail Profile and click Add a New Voice Mail Profile.

19. Enter the Voice Mail Profile Name and Description, such as AvayaVMailProfile and select the Voice Mail Pilot number in step 17. In our case the Voice Mail Pilot number is 4408.



20. Click Features, Voice Mail, Message Waiting Indicator, Add a New Message Waiting Number to add the Message Waiting Indicator On/Off numbers. Included below are two screen captures for Message Waiting Indicator On/Off numbers.


System Route Plan Service Feature Device User Application Help
Cisco CallManager Administration
For Cisco IP Telephony Solutions
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
Cisco CallManager Administration
For Cisco IP Telephony Solutions
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.