Avaya CS1000E Release 7.65 using SIP trunk to Cisco Unified Communications Manager Release 10.5

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Introduction

This document describes the steps and configurations necessary for Cisco Unified Communications Manager (Cisco UCM) release 10.5 to interoperate with the Avaya CS1000E 7.65 using SIP Early-Offer.

Key Results

- Basic call, call transfer, call forwarding, conference call, and hold and resume work successfully.
- Centralized voicemail, using Unity Connection server integrated to Cisco UCM with SCCP and SIP was tested. This voicemail solution can provide centralized voicemail services, supporting both Avaya and Cisco end-users.

The following items were tested:

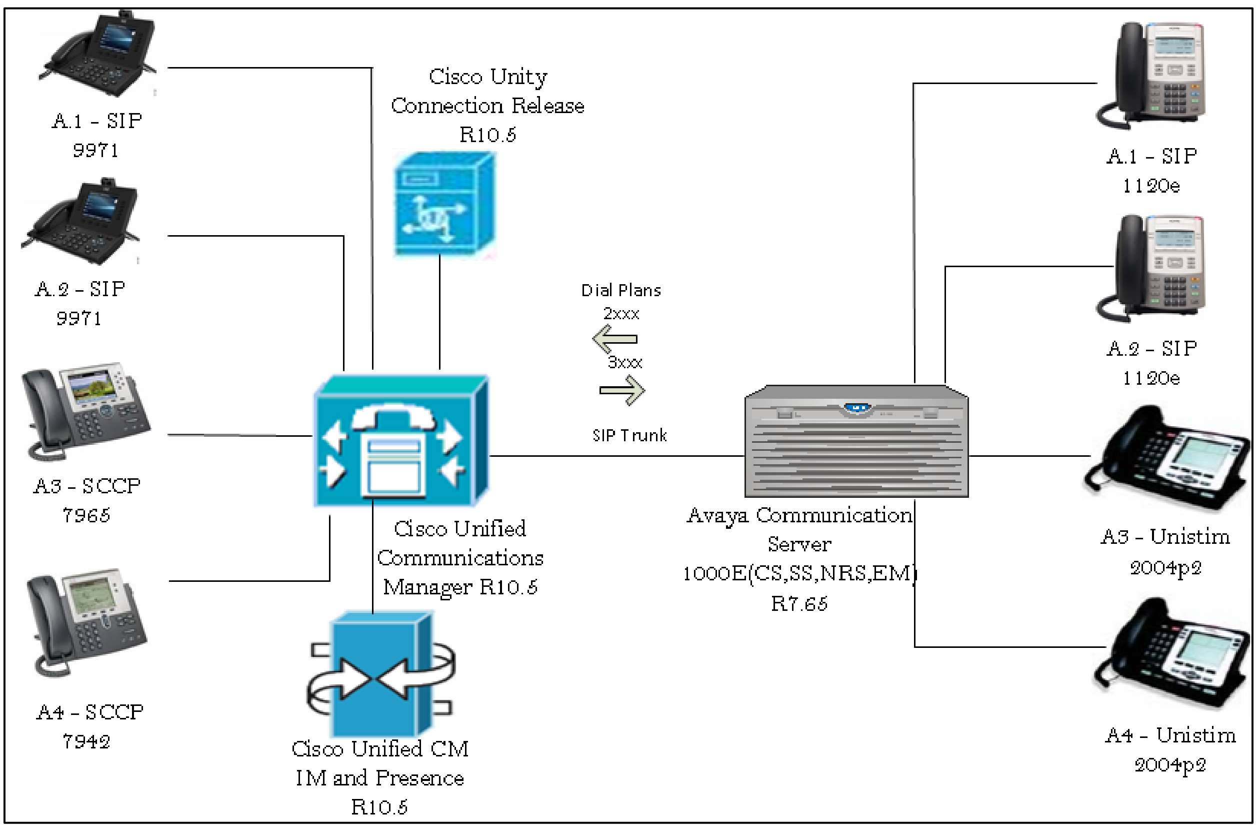
- Basic call between the two systems and verification of voice path, using both SIP and Unistim phones on the Avaya side, SIP and SCCP IP phones on the Cisco side
- CLIP/CLIR/CNIP/CNIR features: calling party name and number delivery (allowed and restricted)
- COLP/CONP/COLR/CONR features: connected name and number delivery (allowed and restricted)
- Call transfer: attended, and early attended
- Alerting Name Identification
- Call Park
- Call forwarding: call forward unconditional (CFU), call forward busy (CFB), and call forward no answer (CFNA)
- Hold and resume with music on hold
- Three-way conferencing
- Voice messaging and MWI activation-deactivation
- Audio Codec Preference List
- DTMF-relay via RFC2833 -verification of DTMF- relay by accessing each other’s VM system and responding to prompts using the keypad to send RTP Telephone Event (RFC2833) of digits pressed

Listed below are the highlights of the integration issues:

- Basic calls worked from Cisco UCM to Avaya CS1000E and vice versa. The Avaya CS1000E only supports early offer to set its media attribute to send/receive mode. Thus, for calls from Cisco UCM to Avaya CS1000E, the Cisco UCM must be set to send SIP Invite with SDP. This will ensure two-way audio once the call is connected.
- CLIR/CNIR - Restriction of calling number on Avaya CS1000E Unistim phones is achieved by configuring the Avaya station's class of service. Setting the class of service (CLS) to DDGD sets the SIP P-Asserted Identity setting to privacy = id. This restricts the calling number information. Setting the class of service to NAMD sets the SIP P-Asserted Identity setting to privacy = user. Restriction of calling name and number on the Cisco UCM can be done on the Route Pattern or SIP Trunk page. Calling name and number restrictions are honored by both sides.

Network Topology

Basic Call Setup



System Components

Hardware Requirements

The following hardware was used:

- Cisco Unified Communications Manager: Cisco UCS-C240 VMware ESXI.5.5.0
- Cisco Unity Connection Cisco: UCS-C240 VMware ESXI.5.5.0 server
- Cisco Unified CM IM and Presence: Cisco UCS-C240 VMware ESXI.5.5.0
- Cisco 7965, 7942, 9971 IP phones
- Avaya Communication Server 1000E/CPPM Linux , CPPM - Pentium M 1.4 GHz(NTDW61BAE5)
- Avaya Media Gateway Controller - NTDW60BB
- Avaya 2004P2, 1120e IP Phones

Software Requirements

The following software is required:

- Cisco UCSC-C240-M3S VMware vSphere Image Profile: ESXi-5.5.0-1331820-standard
- Cisco Unified Communications Manager release 10.5 -10.5.1.10000-7
- Cisco Unified Communications Manager IM & P release 10.5 - 10.5.1.10000-9
- Cisco Unity Connection release 10.5 -10.5.1.10000-7
- Avaya Communication Server 1000E - RELEASE 7 ISSUE 65 P - (Application configuration: CS+SS+NRS+EM_SubM)
- Avaya Media Gateway Controller - LOADWARE VERSION: PSWV 100+
- Cisco Jabber Client – Version 10.5.0 Build 37889

Features

This section lists supported and unsupported features. Deviance from the configuration presented in this guide is not supported by Cisco. Please see the Limitations section on page 7 for more information.

Features Supported

- •CLIP—calling line (number) identification presentation
- •CLIR—calling line (number) identification restriction.(Refer to Integration issues section)
- •CNIP—calling name identification presentation
- •CNIR—calling name identification restriction. (Refer to Integration issues section)
- •Alerting name.(Refer to Limitation Section)
- •Attended call transfer. (Refer to Limitation Section)
- •Early attended call transfer. (Refer to Limitation Section)
- •CFU—call forwarding unconditional. (Refer to Limitation Section)
- •CFB—call forwarding busy. (Refer to Limitation Section)
- •CFNA—call forwarding no answer. (Refer to Limitation Section)
- •COLP—connected line (number) identification presentation.(Refer to Integration issues section)
- •COLR—connected line (number) identification restriction
- •CONP—connected name identification presentation
- •CONR—connected name identification restriction. (Refer to Integration issues section)
- •Hold and resume
- •Conference call
- •MWI—Message Waiting Indicator (lamp ON, lamp OFF)
- •Audio Codec Preference List
- •Call Park (Refer to Limitation Section)
- •DTMF-relay using RFC2833

Features Not Supported or Not Tested

- •Call completion (callback, automatic callback)
- •Blind Call Transfer
- •Inter-working Test Cases with Various Calling/Connected Name and Number
- •Shared Line - Hold & Resume with MOH
- •Interworking Test Cases for Call Transfer

Limitations

These are the known limitations, caveats, or integration issues:

- Avaya CS1000E doesn't support Alerting Name feature. Although the Cisco UCM sends P-Asserted Identity (PAI) header with the alerting name(180 ringing) and connected name(200 ok), this information is not displayed by the Avaya SIP phones, However Avaya Unistim phones displayed the Alerting and connected name details.
- The Avaya PBX uses the History-Info field to send redirecting number information, while the Cisco UCM uses the Diversion header. This affects how calls are treated when redirected to a voice mail system over an SIP trunk. Since release 8.5, Cisco UCM provides the ability to translate either Diversion headers into History-Info headers or History-Info headers to Diversion headers via SIP Normalization Script. Please refer to the configuration section of this document for more details on the actual normalization script used for this testing.
- Avaya phone is configured to restrict connected name and number, it was observed that the SIP response to Cisco UCM only sets the privacy=user. However, the Cisco UCM only recognizes privacy=id to restrict presentation of both connected name and number. Cisco UCM provides the ability to covert the Privacy=user to Privacy=Id using normalization script.
- For integration where Cisco Unity is the centralized voice messaging system, a SIP normalization script is required to enable/disable MWI on Avaya phones. Please refer to the configuration section of this document for more details on the actual normalization script used for this testing.
- During a conference call hosted by the Avaya CS1000E SIP telephone, if the SIP telephone is hung up/dropped out of the conference, the conference call is dropped. The behavior is not seen with Unistim phones.
- Call Park: While retrieving Avaya CS1000E parked call from Cisco UCM, the call has been disconnected, Cisco parked calls has been retrieved successfully from Avaya CS1000E.
- Both systems support call forwarding and call transfer features. There are some call forward and transfer scenarios where the calling name and number and/or connected name and number are not updated after the call has been transferred or forwarded. This issue is found primarily when an Avaya phone is the forwarding or transferring party to a Cisco phone via the SIP trunk.

Configuration

The goal of this guide is to provide an overview of the integration between Cisco Unified Communication Manager and Avaya CS1000E PBX's. The deployment will interconnect the UC systems using SIP. No PSTN connectivity has been tested with this integration. The following sections provide the required configurations for a successful integration.

Configuring Sequence and Tasks:

Avaya Communication Server 1000E:
Configure the IP D-channel (signaling channel) between the call server and the signaling server —LD 17.

1. Zone Configuration —LD 117.
2. Configure the SIP route — LD 16.
3. Configure the SIP virtual trunks to the signal — LD 14.
4. Configure for the virtual lines for the Avaya IP phone — LD 20.
5. Configure the route list block for the virtual trunk route — LD 86.
6. Configure CDP steering codes — LD 87.
7. List trunk member — LD 21.
8. Avaya SIP Line Configuration.

Signaling Server Setup via the Avaya CS1000E Node Summary:

1. Configure a new IP telephony node summary.
2. Configure the VGW and IP phone codec profile section.
3. Configure the SIP GW setting section.
4. Configure the quality of service (QoS) section.

Network Routing Server:

1. Configure the system-wide settings.
2. Configure the NRS server settings.
3. Configure a service domain.
4. Configure an L1 domain (UDP).
5. Configure an L0 domain (CDP).
6. Configure a gateway endpoint gateway.
7. Configure the routing entries.

Cisco Unified Communications Manager:

1. Device setting SIP profile.
2. Media resource group and media resource group list.
3. Assign media resource group list (MRGL) in the default device pool.
4. SIP trunk to Avaya CS1000E PBX.
5. SIP Trunk Normalization Script.
6. SIP and SCCP phones device configuration.
7. Route pattern to the Avaya CS1000E PBX.
8. Call Manager Service Parameter "Duplex Streaming Enabled" set to "True".
9. Audio Codec Preference Configuration.
10. Region Configuration.
11. Cisco UCM Unity Integration.

Avaya CS1000E Software Version – Issue and Release

```
REQ  iss
VERSION 4121
System type is - Communication Server 1000E/CPPM Linux
CPPM - Pentium M 1.4 GHz
IPMGs Registered:            1
IPMGs Unregistered:          0
IPMGs Configured/unregistered:  0
RELEASE 7
ISSUE 65 P
IDLE_SET_DISPLAY tekvizion
IPMG  TYPE CSP/SW MSP  APP  FPGA BOOT   DBL1     DBL2
0 1 MGC    DC04 AB02 BA18 AA22 BA18 DSP1AB07      N/A

REQ  issp
02/12/14 02:03:33
TID: XXXXXXXXX
VERSION 4121
System type is - Communication Server 1000E/CPPM Linux
CPPM - Pentium M 1.4 GHz
IPMGs Registered:            1
IPMGs Unregistered:          0
IPMGs Configured/unregistered:  0
RELEASE 7
ISSUE 65 P
IDLE_SET_DISPLAY tekvizion
SYSTEM HAS NO IN-SERVICE DEPLISTS
MDP>LAST SUCCESSFUL MDP REFRESH :2014-11-03 12:46:57(Local Time)
MDP>USING DEPLIST ZIP FILE DOWNLOADED :2014-11-03 10:39:29(est)
SYSTEM HAS NO USER SELECTED PEPS IN-SERVICE

LOADWARE VERSION: PSWV 100+
INSTALLED LOADWARE PEPS : 7
PAT#  CR #         PATCH REF #    NAME          DATE       FILENAME
00    Q01981776     ISS1:1OF1     udtcab25      03/11/2014  udtcab25.lw
01    wi01171831    ISS1:1OF1     MGCCDC04      03/11/2014  MGCCDC04.LW
02    wi01057886    ISS1:1OF1     DSP1AB07      03/11/2014  DSP1AB07.LW
03    wi01057886    ISS1:1OF1     DSP2AB07      03/11/2014  DSP2AB07.LW
04    wi01057886    ISS1:1OF1     DSP3AB07      03/11/2014  DSP3AB07.LW
05    wi01057886    ISS1:1OF1     DSP4AB07      03/11/2014  DSP4AB07.LW
```

06 wi01057886 ISS1:1OF1 DSP5AB07 03/11/2014 DSP5AB07.LW

ENABLED PLUGINS : 1

PLUGIN STATUS PRS/CR_NUM MPLR_NUM DESCRIPTION

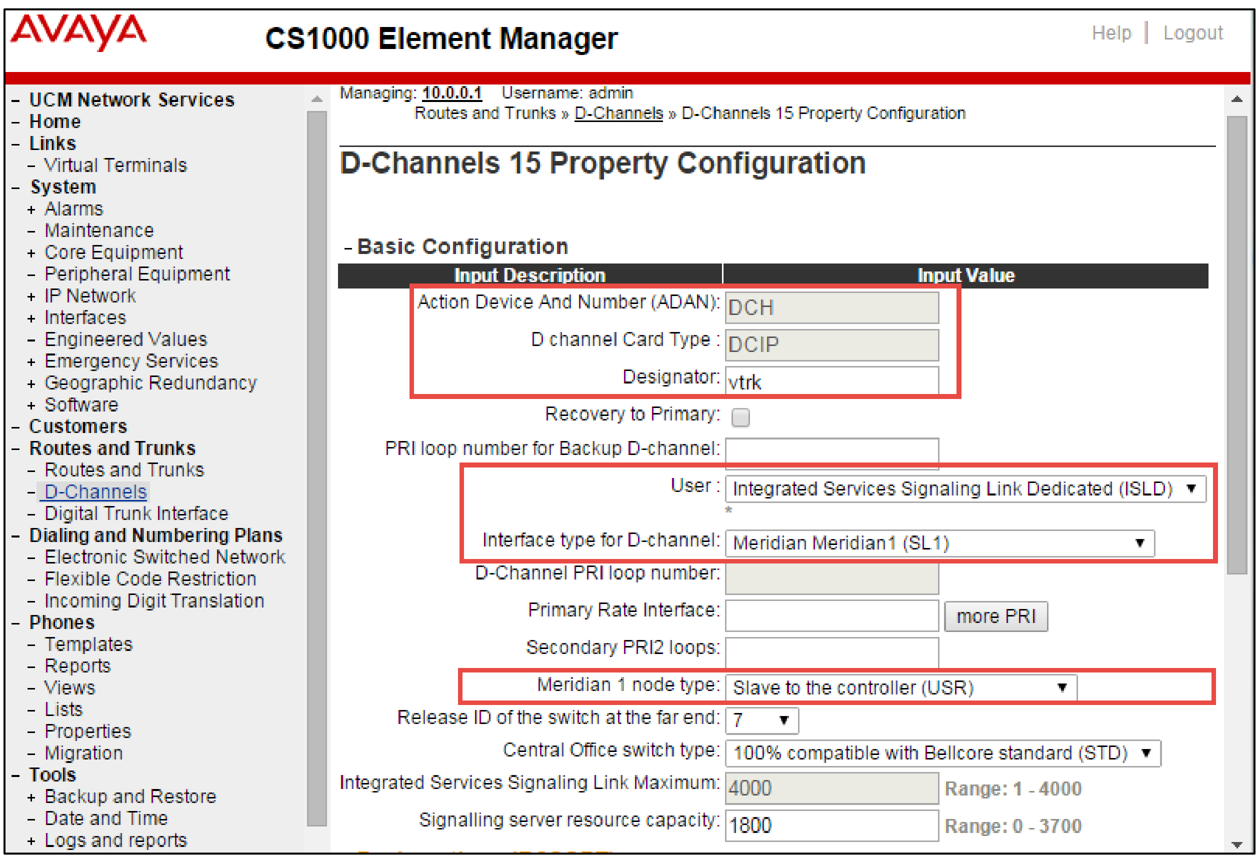
----

501 ENABLED Q02138637 MPLR30070 Enables blind transfer to a SIP endpoint even if SIP UPDATE is not supported by the far end

IP D-Channel Configuration

Configure the IP D-channel (signaling channel) between the Call Server and the Signaling Server – LD 17.The SIP Gateway application requires a D-channel over IP to communicate with the CS 1000E system. The SIP routes are associated with the D-channels and the SIP Gateway application running on a Linux server. The SIP Gateway route is used to communicate with the Call Server.

D channel Card Type (CYTP) list, select D-Channel is over IP (DCIP).
Set User = Integrated Services Signaling Link Dedicated (ISLD).
Set Interface type for D-channel (IFC) list = Meridian Meridian1 (SL1).
Set Meridian 1 Node Type = Slave to the Controller (USR).



Zone Configuration

Navigation Path: CS1000 Element Manager→ System → IP Network → Zone → BandwidthZones

Zones are used to group related information for either bandwidth or dial plan numbering purposes. Zone 6(MO) and Zone 7(VTRK) were used for Best Bandwidth (G729).Zone 3(MO) or 0(MO) and Zone 1(VTRK) were used for Best Quality (G711).

Bandwidth Zones

Add... Edit... Import... Export Maintenance... Delete Refresh

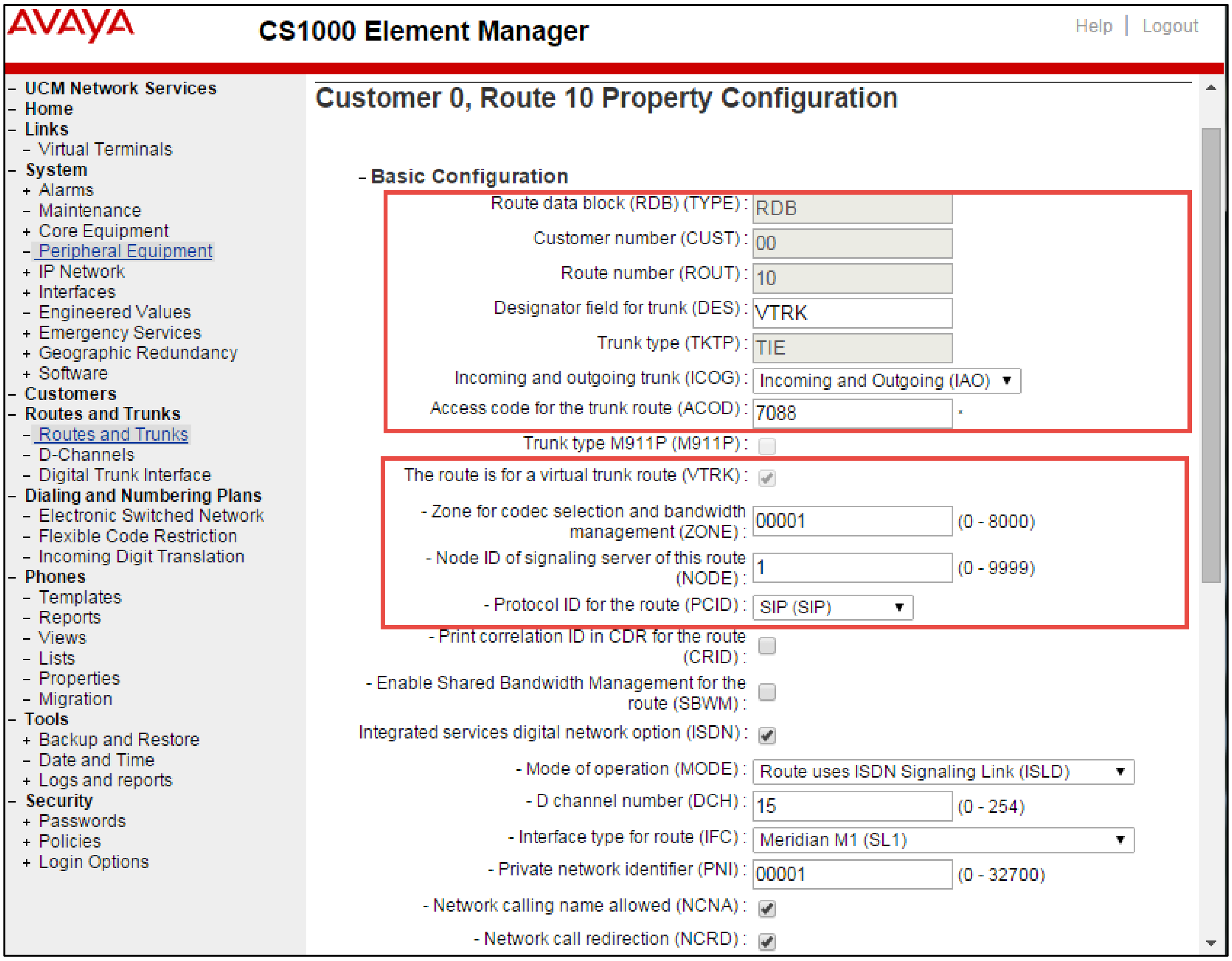
	Zone	Intrazone Bandwidth	Intrazone Strategy	Interzone Bandwidth	Interzone Strategy	Resource Type	Zone Intent	Description	Location Name	Reserved BW Block Size
1	1	100000	BQ	100000	BQ	SHARED	VTRK	ZONE1		0
2	2	100000	BQ	100000	BQ	SHARED	VTRK			0
3	3	1000000	BQ	1000000	BQ	SHARED	MO			0
4	4	1000000	BQ	1000000	BB	SHARED	MO			0
5	5	1000000	BQ	1000000	BQ	SHARED	MO			0
6	6	1000000	BB	1000000	BB	SHARED	MO			0
7	7	1000000	BB	1000000	BB	SHARED	VTRK			0
8	100	1000000	BQ	1000000	BB	SHARED	VTRK			0

SIP Route Configuration

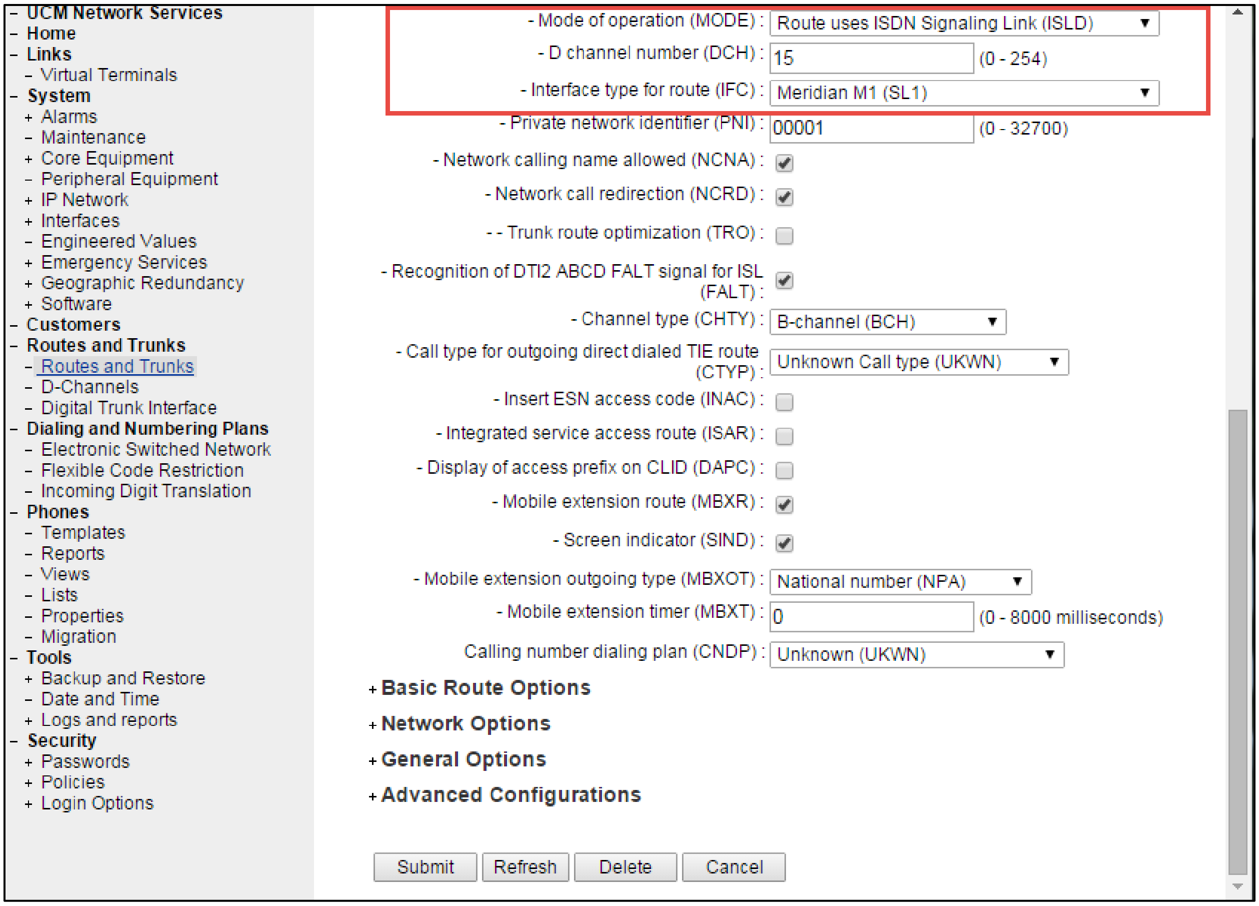
Navigation Path: CS1000 Element Manager→ Routes and Trunk

Set Route Data Block (RDB) = RDB.
Set Customer Number (CUST) = 0. This is used for this testing.
Set Route Number (ROUT) = 10. This is used for this testing.
Set Trunk type (TKTP) = TIE trunk data block (TIE).
Set Incoming and outgoing trunk (ICOG) = Incoming and Outgoing (IAO).
Set Access code for the trunk route (ACOD) = 7088. This is used for this testing.
Set Node ID of signaling server of this route (NODE) = 1. This is used for this testing.
Set Protocol ID for the route (PCID) = SIP.
Set Mode of Operation (MODE) = ISDN Signaling Link (ISLD).
Set D channel number (DCH) = 15. This is used for this testing.
Set Interface type for route (IFC) = Meridian M1 (SL1).
Check Network calling name allowed (NCNA).
Check Network call redirection (NCRD).

Avaya CS1000E SIP Trunk to Cisco UCM Configuration (Continued)

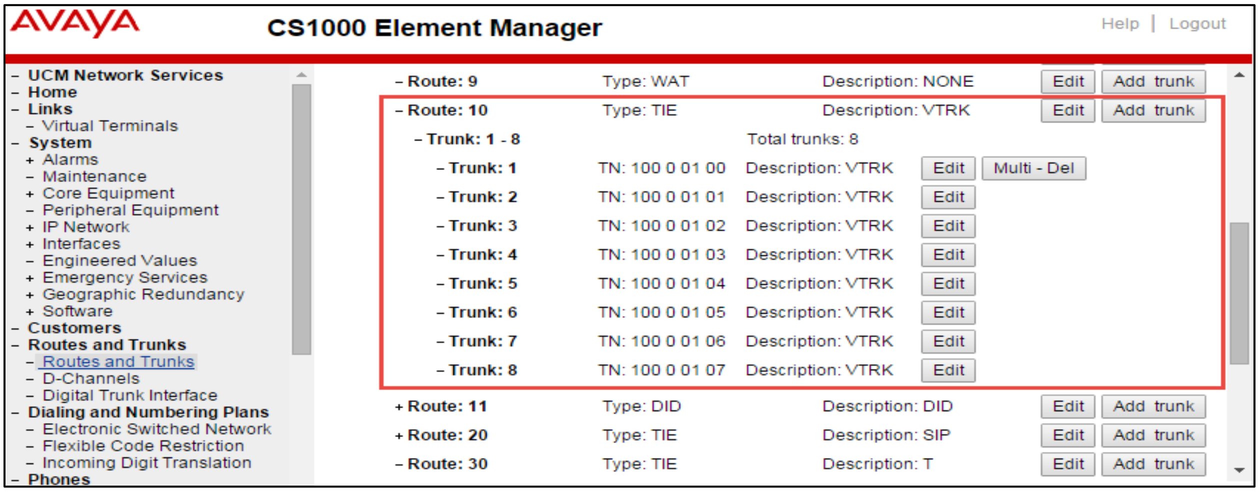


Avaya CS1000E SIP Trunk to Cisco UCM Configuration (Continued)

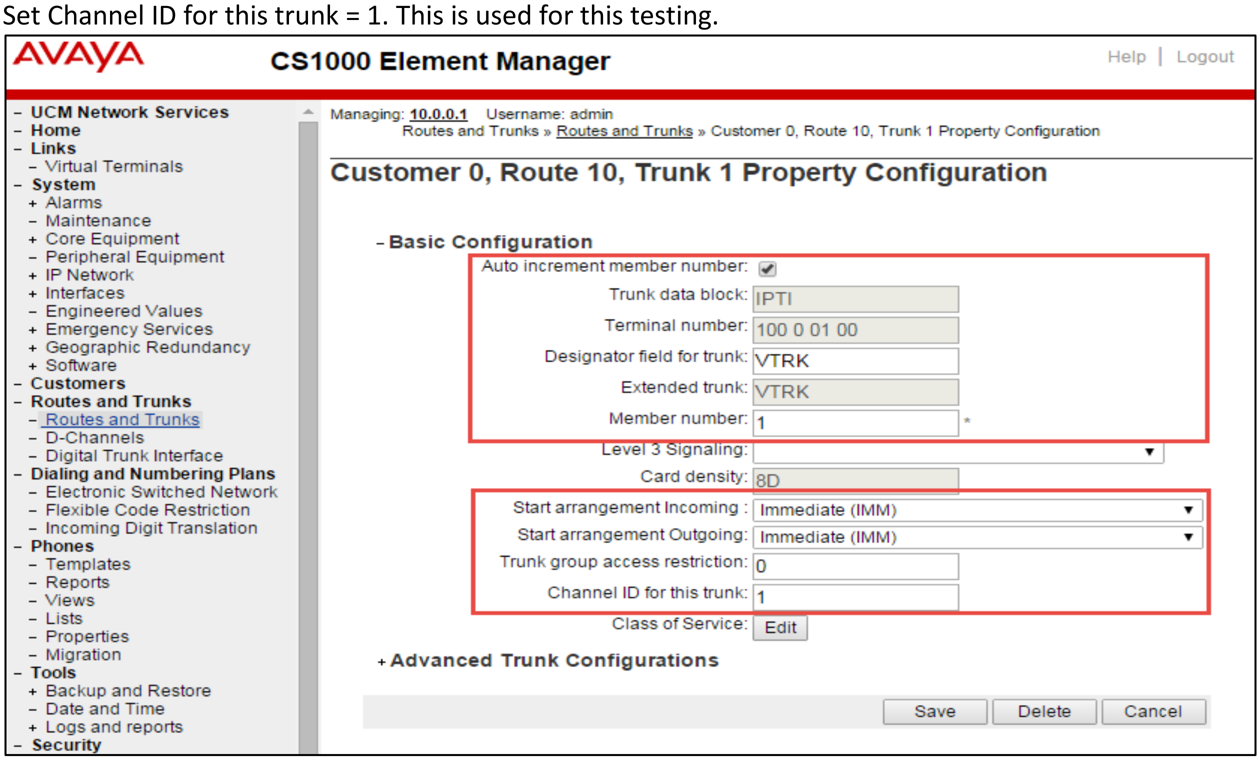


SIP Virtual trunk Configuration

Navigation Path: CS1000 Element Manager→ Routes and Trunk



Set Route Data Block (RDB) = RDB.
Set Trunk data block = IP Trunk (IPTI).
Set Terminal Number = 100 0 01 00.This is used for this testing.
Set Designator field for trunk = VTRK.
Set Member number = 1. This is used for this testing.
Set Start arrangement Incoming = Immediate (IMM).
Set arrangement Outgoing = Immediate (IMM).
Set Trunk Group Access Restriction = 1. This is used for this testing.
Set Channel ID for this trunk = 1. This is used for this testing.



Virtual Lines for Avaya CS1000E IP Phones

Avaya Unistim Phone

```
>ld 20
REQ: prt
TYPE: tn
TYPE TNB
TN   108 1 5 0
DATE
PAGE
DES
DES  2004P2
TN   108 1 05 00  VIRTUAL
TYPE 2004P2
CDEN 8D
CTYP XDLC
CUST 0
NUID
NHTN
CFG_ZONE 00000
CUR_ZONE 00000
MRT
ERL  0
ECL  0
FDN  2004
TGAR 1
LDN  NO
NCOS 0
SGRP 0
RNPG 0
SCI  0
SSU
XLST
SCPW
CLS  UNR FBA WTA LPR MTD FNA HTA TDD HFA CRPD
   MWA LMPN RMMD AAD IMD XHD IRD NID OLD VCE DRG1
   POD SLKD CCSD SWD LND CNDA
   CFTA SFD MRD DDV CNID CDCA MSID DAPA BFED RCBD
   ICDD CDMD LLCN MCTD CLBD AUTU
   GPUD DPUD DNDD CFXD ARHD CLTD ASCD
   CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
   AHD DDGA NAMA
   DRDD EXR0
   USMD USRD ULAD RTDD RBDD RBHD PGND OCBD FLXD FTTC MCBN
   VOLA VOUD CDMR PRED RECD MCDD T87D SBMD
```

KEM2 MSNV FRA PKCH MUTA MWTD DVLD CROD ELCD VMSA

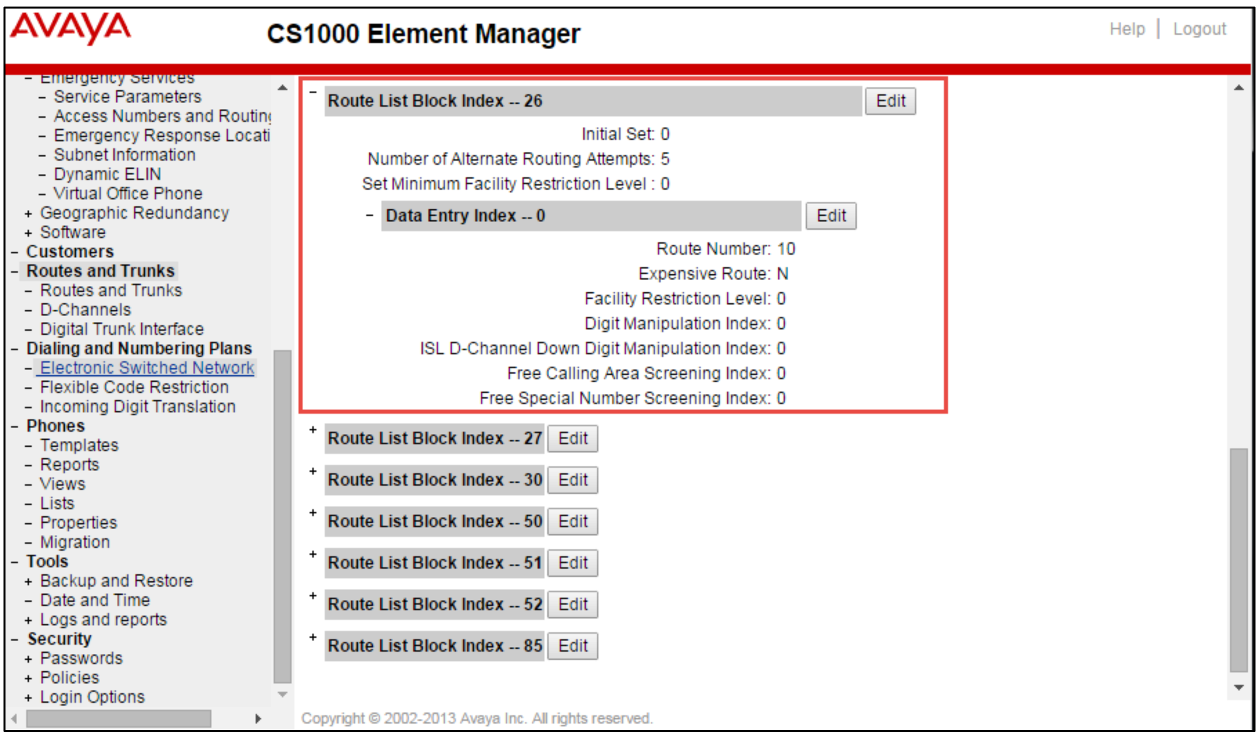
Avaya CS1000E Unistim Phone Configuration (Continued)

```
CPND_LANG ENG
RCO  0
EFD  2004
HUNT 2004
EHT  2004
LHK  0
PLEV 02
PUID
UPWD
DANI NO
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY  00 SCR 3005 0     MARP
        CPND
          NAME cs1kunis1
          XPLN 9
          DISPLAY_FMT FIRST,LAST
     01
     02
     03
     04
     05
     06
     07
     08
     09
     10
     11
     12
     13
     14
     15
     16
     17 TRN
     18 AO6
     19 CFW 16  2004
```

20 RGA
21 PRK
22 RNP
23
24 PRS
25 CHG
26 CPN
27
28
29
30
31
DATE 25 NOV 2014

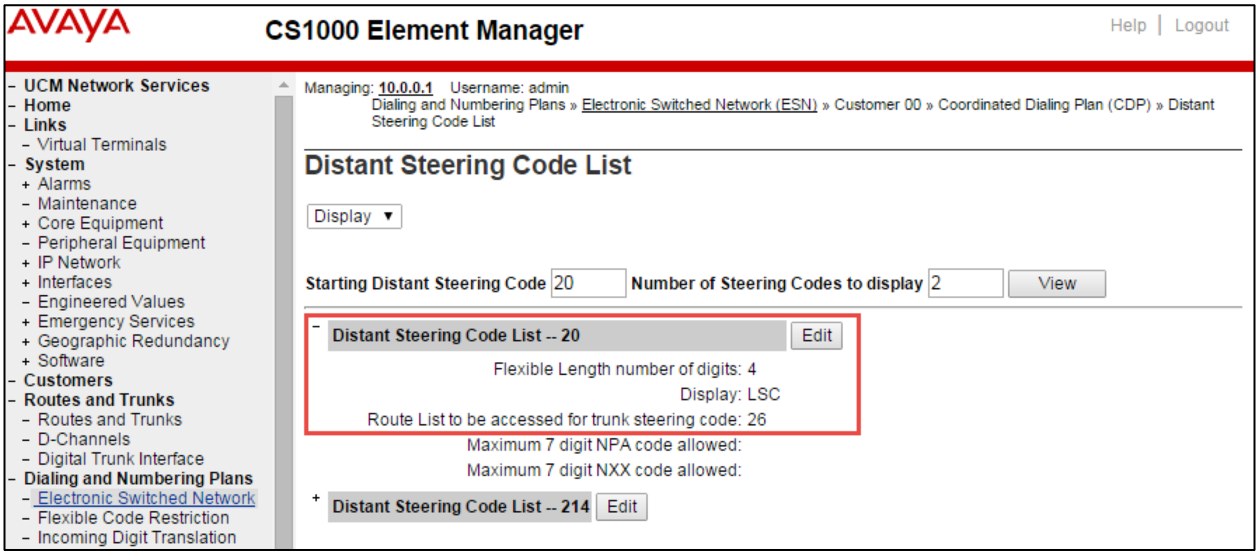
Route List Block Configuration

Navigation Path: CS1000 Element Manager→ Digital and Numbering Plans → Route List Block(RLB)



CDP Steering code configuration

Navigation Path: CS1000 Element Manager→ Digital and Numbering Plans → Coordinated Dialing Plan(CDP) list→ Distance Steering code(DSC)



Avaya SIP Line Configuration

Navigation Path: CS1000 Element Manager→ System → Customers → SIP Line Service

In the User agent DN prefix field, enter a DN prefix to build the HOT U key.information for SIP Line phones. The DN cannot conflict with the current system. The User Agent DN prefix field is the same as the UAPR prompt in LD 15 and is used on the Phones.

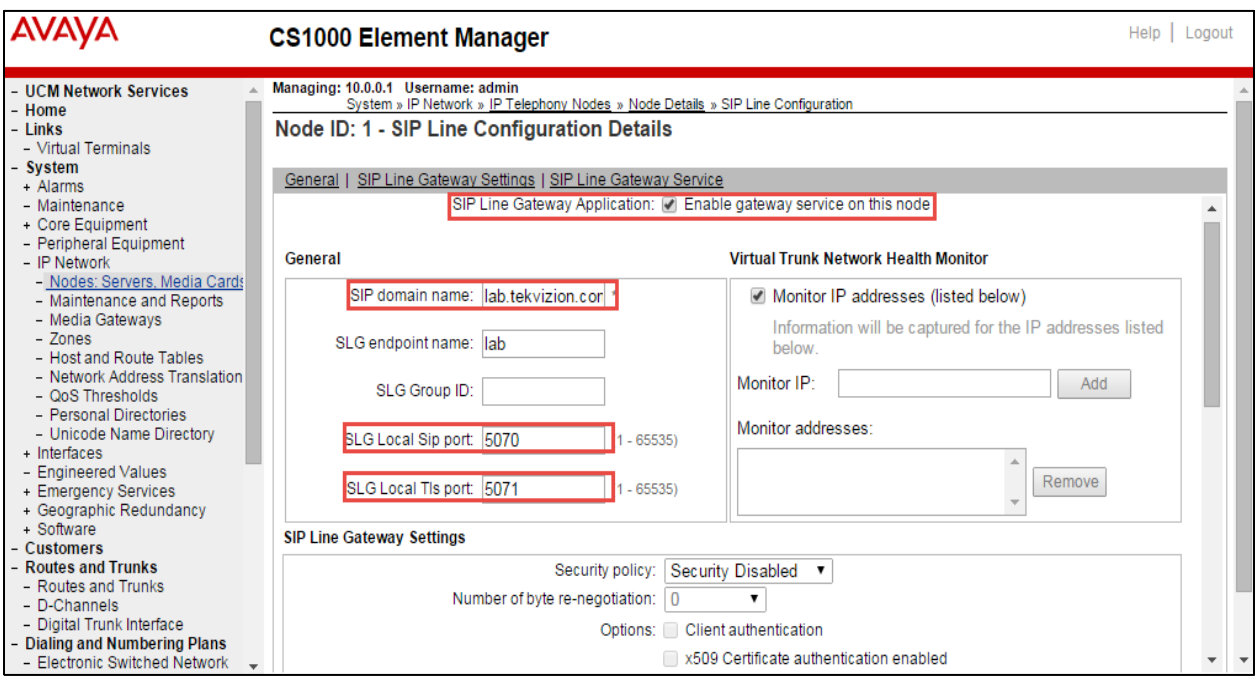
A HOT U key label is available when UXTY is SIPL. The HOT U key is also known as the User Agent Directory Number (UADN) key. The UADN key is used to make and receive calls between the SIP Line Gateway and the Universal Extensions. However, this key is used only by the SIP Line Gateway (SLG) application. (The UADN is not dialed by end users. It is only used internally between the Call Server and the SIP Line Gateway application.)

Avaya CS1000E SIP Line Configuration (Continued)

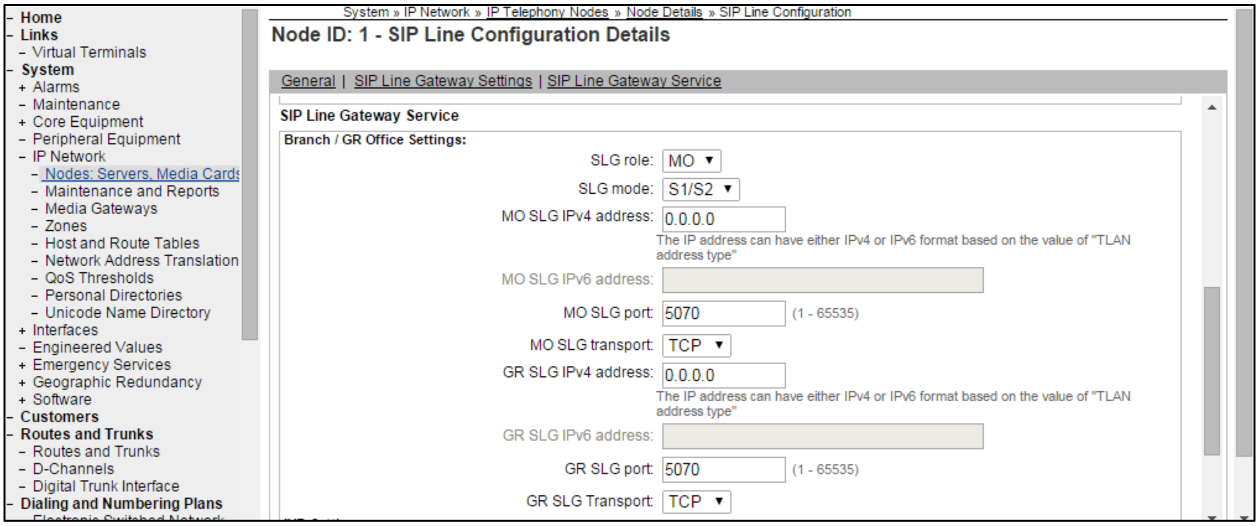
SIP Line Configuration Details

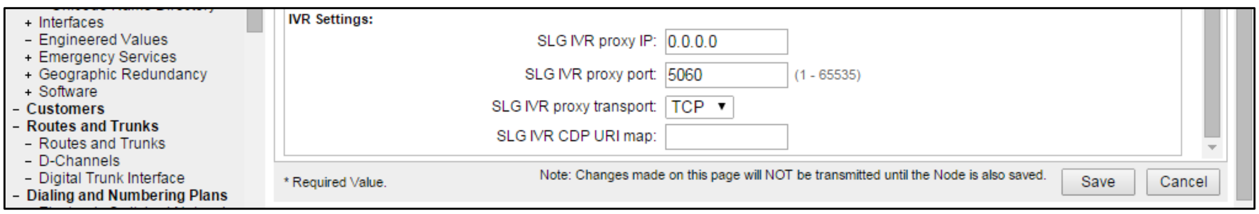
Navigation Path: CS1000 Element Manager→ System → Node ID → SIP Line under Applications

Check Enable gateway service on this node
Set domain Name * = lab.tekvizion.com. This is used for this example
Set SLG Local SIP port = 5070
Set SLG Local TLS port = 5071



Avaya CS1000E SIP Line Configuration (Continued)





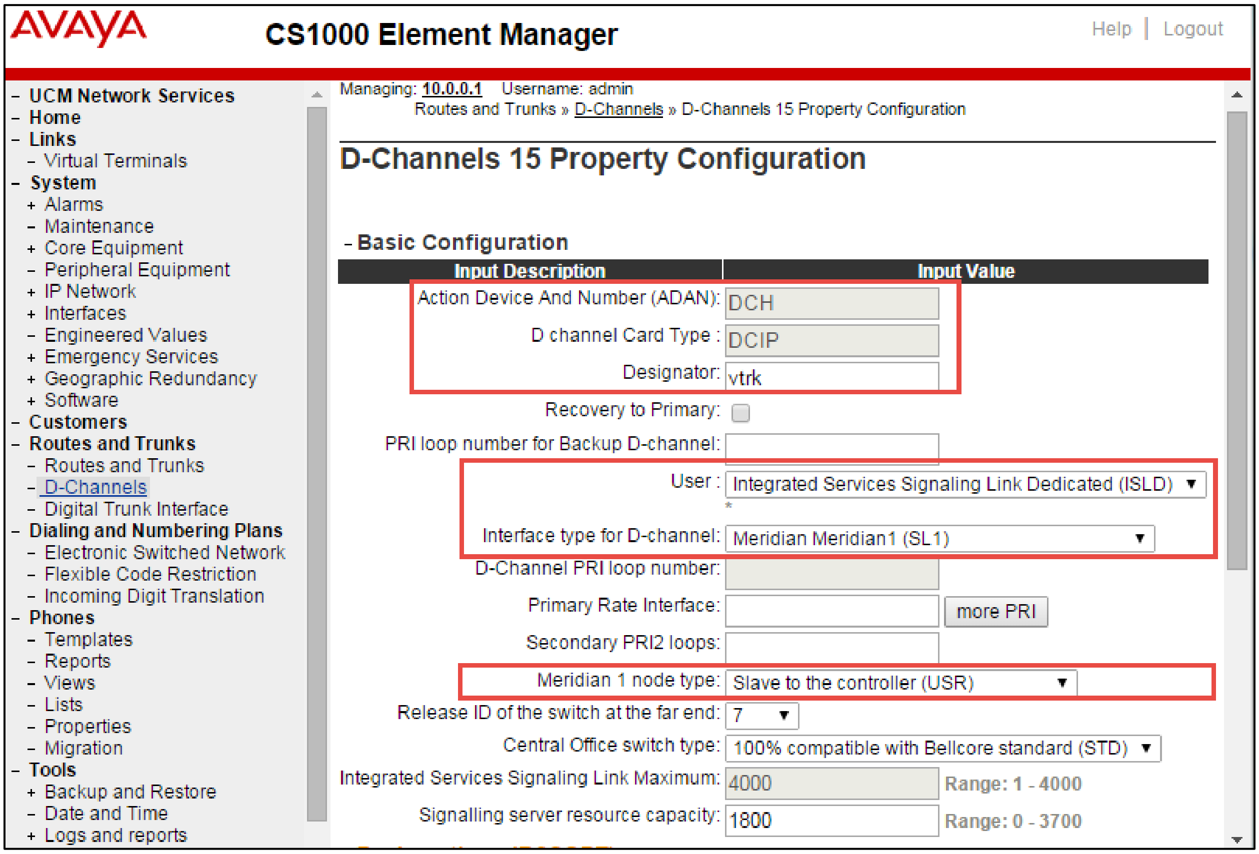
D – Channel Configuration

Navigation Path: CS1000 Element Manager→ Routes and Trunks → D-Channels

The SIP Line Gateway (SLG) application requires a D-channel over IP to communicate with the Avaya CS1000E system. The SIP Line routes are associated with the D-channels and the SLG application running on a Linux server. The SIP Line route is used to communicate with the Call Server.

D channel Card Type (CYTP) list, select D-Channel is over IP (DCIP)
Set User = Integrated Services Signaling Link Dedicated (ISLD)
Set Interface type for D-channel (IFC) list = Meridian Meridian1 (SL1)
Set Meridian 1 Node Type = Slave to the Controller (USR)

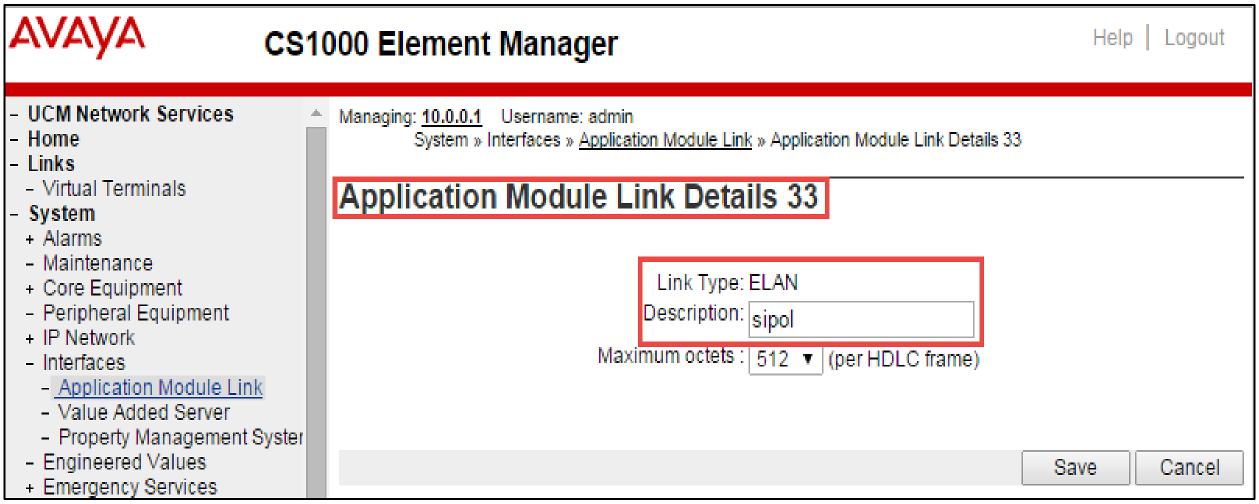
Avaya CS1000E D channel Configuration (Continued)



Application Module Link Details

Navigation Path: CS1000 Element Manager→ System → Interfaces → Application Module Link

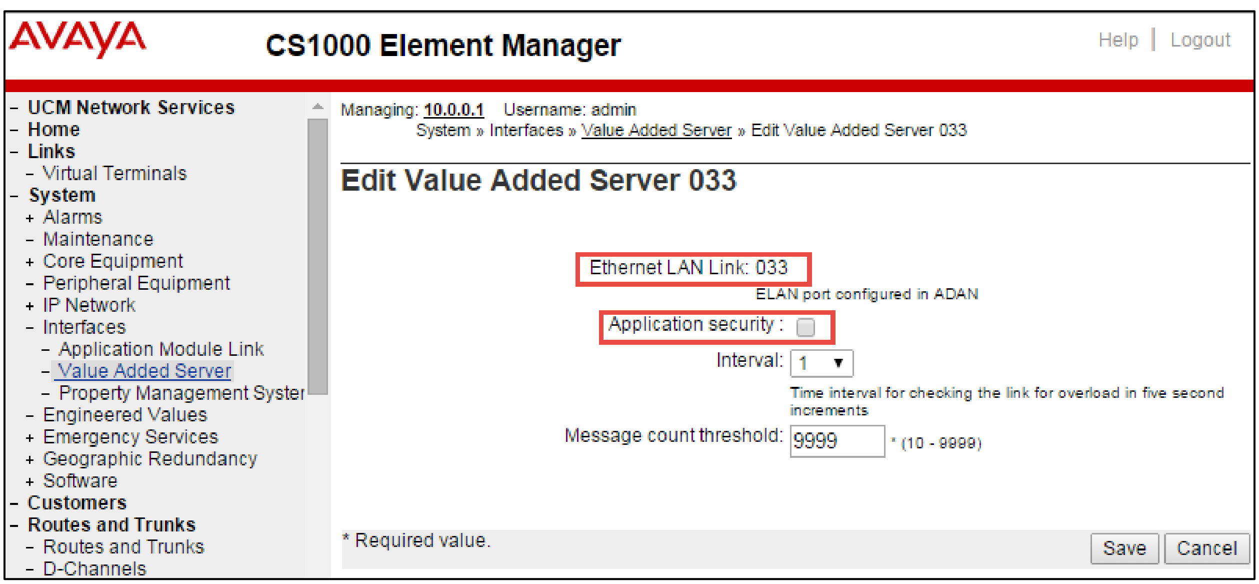
The SLG application uses the AML over ELAN link to establish a pbxlink (AML over ELAN) connection with the CS 1000 system. The SLG application can control the SIPL UEXT using AML messages with the pbxlink established. Application Module Link page, in the Port number field, enter the port number. The SIP Line service uses ports 32 to 127.
Description = enter a description for the AML.



VAS ID association with AML over ELAN link

Navigation Path: CS1000 Element Manager→ System → Interfaces → Value Added server

Every AML over ELAN link configured on the Avaya CS1000E system requires a Value Added Server (VAS) ID for the AML messages to be sent. Use the following procedure to associate a Value Added Server (VAS) with AML over ELAN.
Set Ethernet LAN Link = 33.This is used for this example.
Set Application Security check box is cleared.

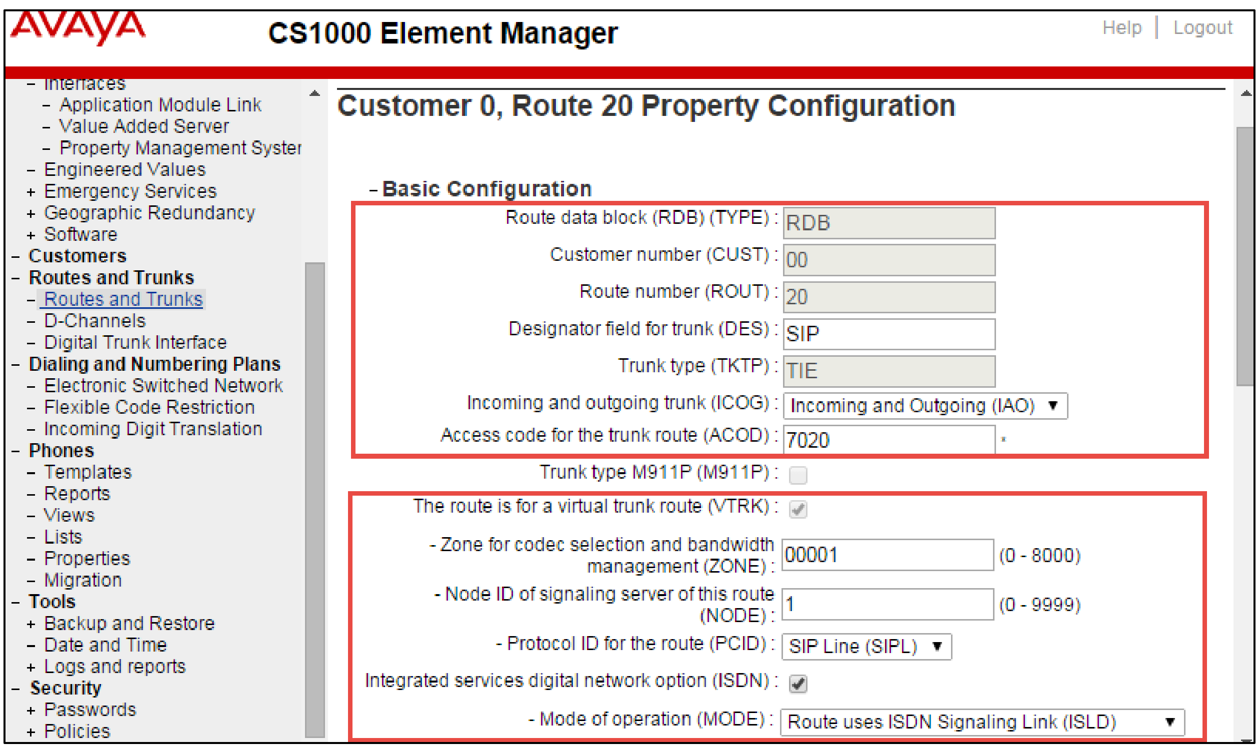


SIP Line Route Configuration

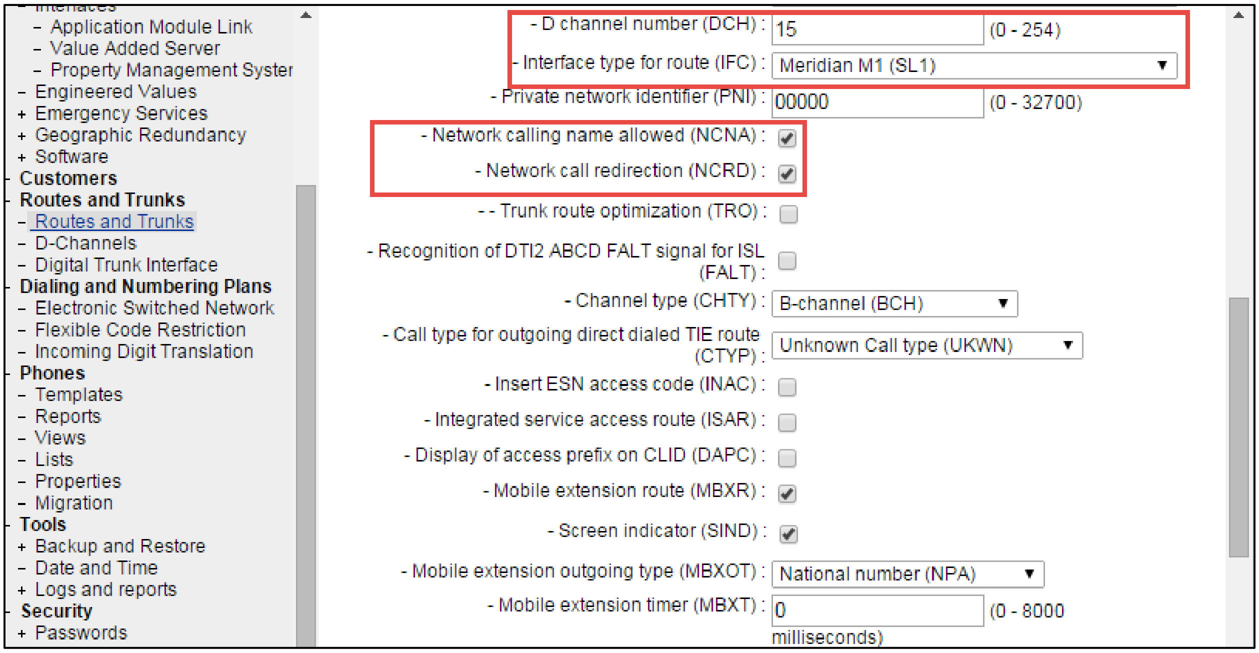
Navigation Path: CS1000 Element Manager→ Routes and Trunk

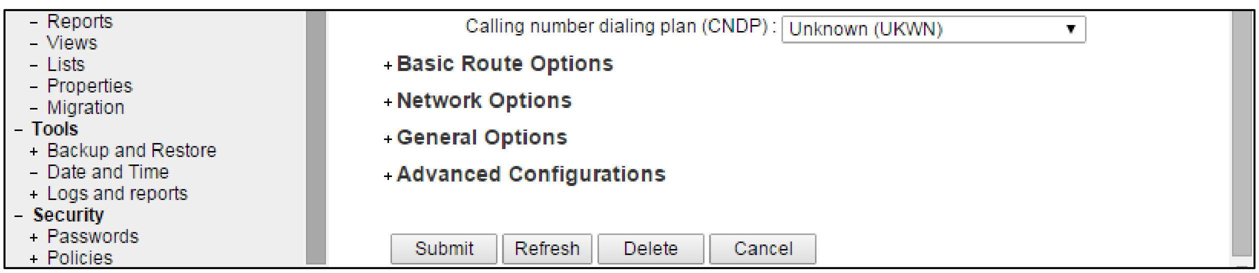
Configure a SIP Line route similar to the way to configure a virtual trunk route, such as SIP.A virtual trunk zone is required for the SIP Line route to work. Ensure to configure a virtual trunk zone.

Set Route Data Block (RDB) = RDB.
Set Customer Number (CUST) = 0. This is used for this testing.
Set Route Number (ROUT) = 20. This is used for this testing.
Set Trunk type (TKTP) = TIE trunk data block (TIE).
Set Incoming and outgoing trunk (ICOG) = Incoming and Outgoing (IAO).
Set Access code for the trunk route (ACOD) = 7020. This is used for this testing.
Set Node ID of signaling server of this route (NODE) = 1. This is used for this testing.
Set Protocol ID for the route (PCID) = SIP Line (SIPL).
Set Mode of Operation (MODE) = ISDN Signaling Link (ISLD).
Set D channel number (DCH) = 15. This is used for this testing.
Set Interface type for route (IFC) = Meridian M1 (SL1).
Check Network calling name allowed (NCNA).
Check Network call redirection (NCRD).



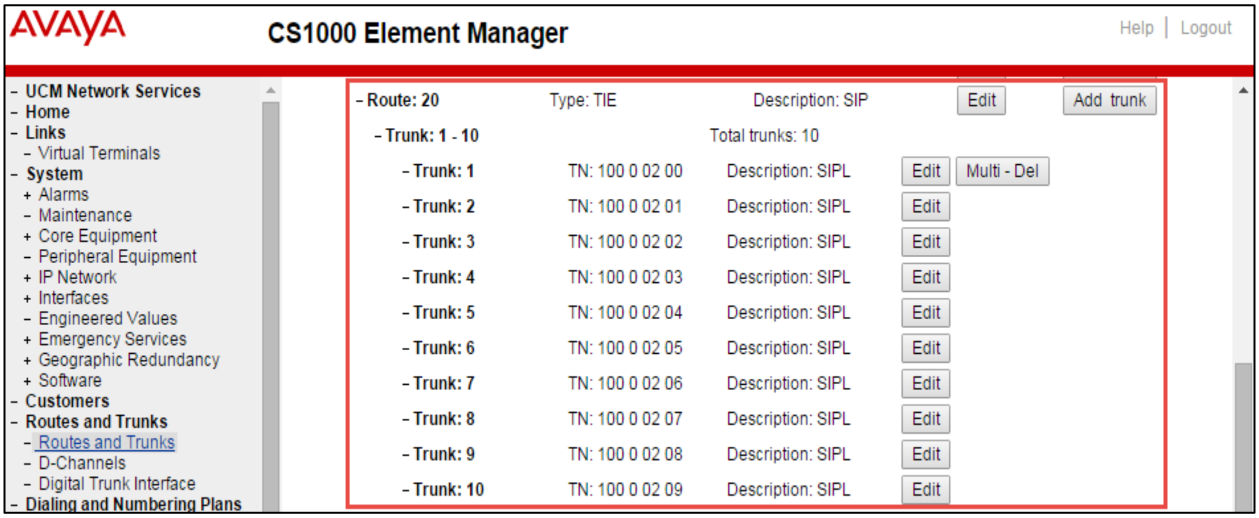
Avaya CS1000E SIP Line Configuration (Continued)





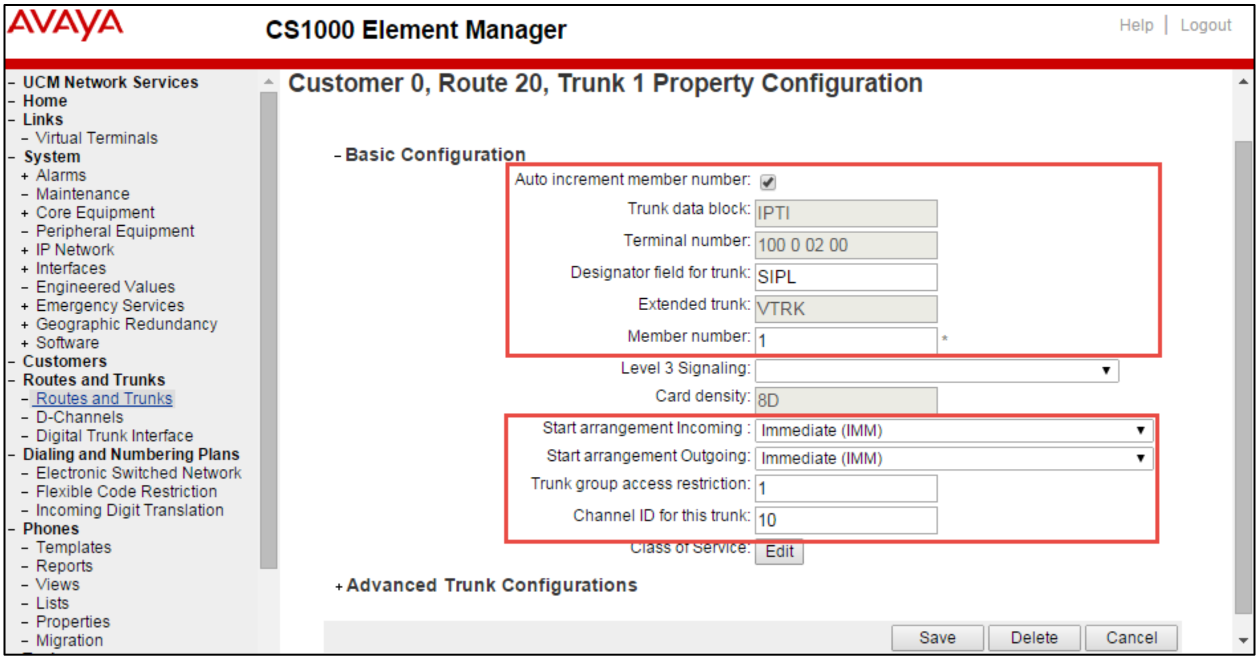
SIP Line Trunk Configuration

Navigation Path: CS1000 Element Manager→ Routes and Trunk



Set Route Data Block (RDB) = RDB.
Set Trunk data block = IP Trunk (IPTI).
Set Terminal Number = 100 0 02 00.This is used for this testing.
Set Designator field for trunk = SIPL
Set Member number = 1. This is used for this testing.
Set Start arrangement Incoming = Immediate (IMM)
Set arrangement Outgoing = Immediate (IMM)
Set Trunk Group Access Restriction = 1. This is used for this testing.
Set Channel ID for this trunk = 10. This is used for this testing.

Avaya CS1000E SIP Line Configuration (Continued)



Avaya SIP Line Phone Configuration

Ld 20
REQ: prt
TYPE: tn
TYPE TNB
TN   108 1 5 29
DATE
PAGE
DES
DES SIP1
TN   108 1 05 29 VIRTUAL
TYPE UEXT
CDEN 8D
CTYP XDLC
CUST 0
UXTY SIPL
MCCL YES
SIPN 1
SIP3 0
FMCL 0
TLSV 0
SIPU 3003
NDID 3003
SUPR NO
SUBR DFLT MWI RGA CWI MSB
UXID 3003
NUID
NHTN
CFG_ZONE 00003
CUR_ZONE 00003
MRT
ERL 0
ECL 0
FDN 2003
TGAR 1
LDN NO
NCOS 0
SGRP 0
RNPG 0
SCI 0
SSU
XLST
SCPW 3003
CLS CTD FBA WTA LPR MTD FNA HTA TDD HFD CRPD
    MWA LMPN RMMD AAD IMD XHD IRD NID OLD VCE DRG1
    POD SLKD CCSD SWD LND CNDA
    CFTA SFD MRD DDV CNID CDCA MSID DAPA BFED RCBD

Avaya CS1000E SIPL Phone Configuration (Continued)

```
    ICDD CDMD LLCN MCTD CLBD AUTU
    GPUD DPUD DNDD CFXA ARHD CLTD ASCD
    CPFA CPTA ABDD CFHA FICD NAID BUZZ AGRD MOAD
    AHA DDGA NAMA
    DRDD EXR0
    USMD USRD ULAD RTDD RBDD RBHD PGND OCBD FLXD FTTC MCBN
    VOLA VOUD CDMR PRED RECD MCDD T87D SBMD ELMD
    MSNV FRA  PKCH MWTD DVLD CROD ELCD VMSA
CPND_LANG ENG
RCO  0
EFD  2003
HUNT 2003
EHT  2003
LHK  0
PLEV 02
PUID
UPWD
DANI NO
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY  00 SCR 3003 0     MARP
      CPND
        NAME cs1ksip1
        XPLN 8
        DISPLAY_FMT FIRST,LAST
    01 HOT U 233003 MARP 0
    02
    03
    04
    05
    06
    07
    08
    09
    10
    11
    12
    13
```

Avaya CS1000E SIPL Phone Configuration (Continued)

    15
    16
    17 TRN
    18 AO6
    19 CFW 16
    20 RGA
    21 PRK
    22 RNP
    23
    24 PRS
    25 CHG
    26 CPN
    27
    28
    29
    30
    31
DATE 25 NOV 2014
NACT

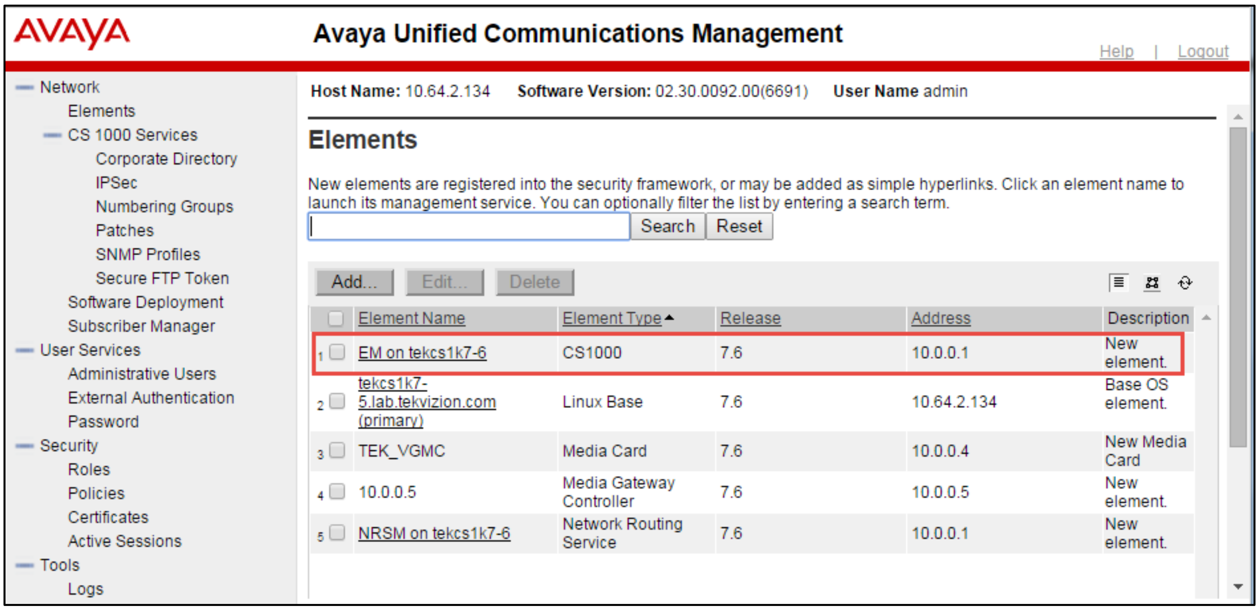
Avaya CS1000E CLI Configuration

Attached is the LD prints for Avaya CS1000E Configuration for reference

CS1k Configuration.txt

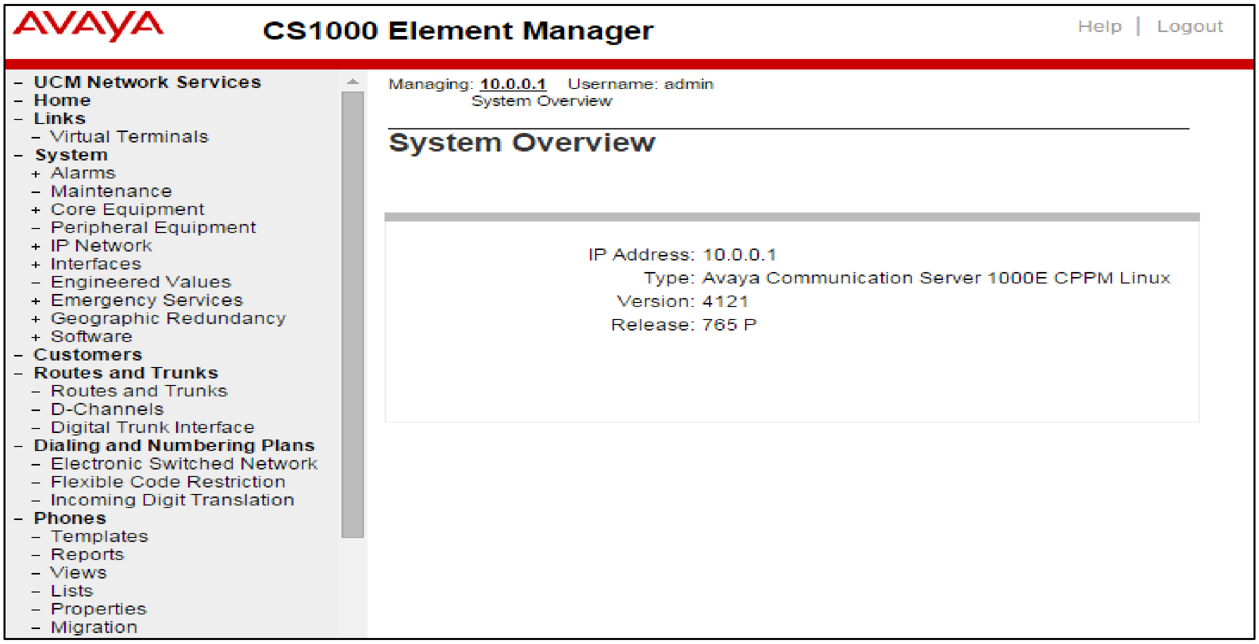
Avaya Unified Communication Management

The UCM security domain provides central authentication, authorization, auditing, certificate management, and secure navigation functionality between managed elements. All elements within the same security domain appear in a single navigation tree.



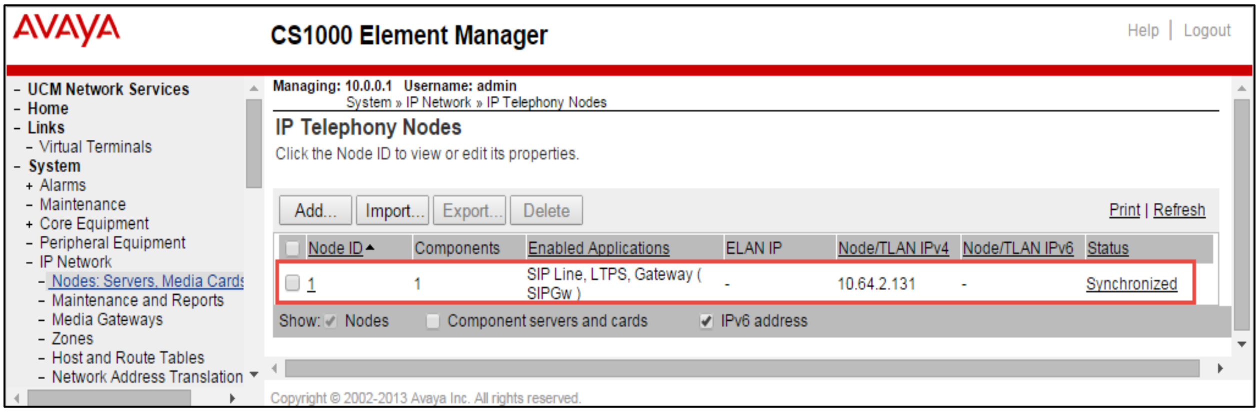
Avaya CS1000E Element Manager

Element Manager is a Web-based user interface used to configure and maintain Avaya CS1000E components.Element Manager is deployed with the Avaya Unified Communications Management solution on a Linux based operating system. Avaya UCM provides logon and security features for Element Manager.

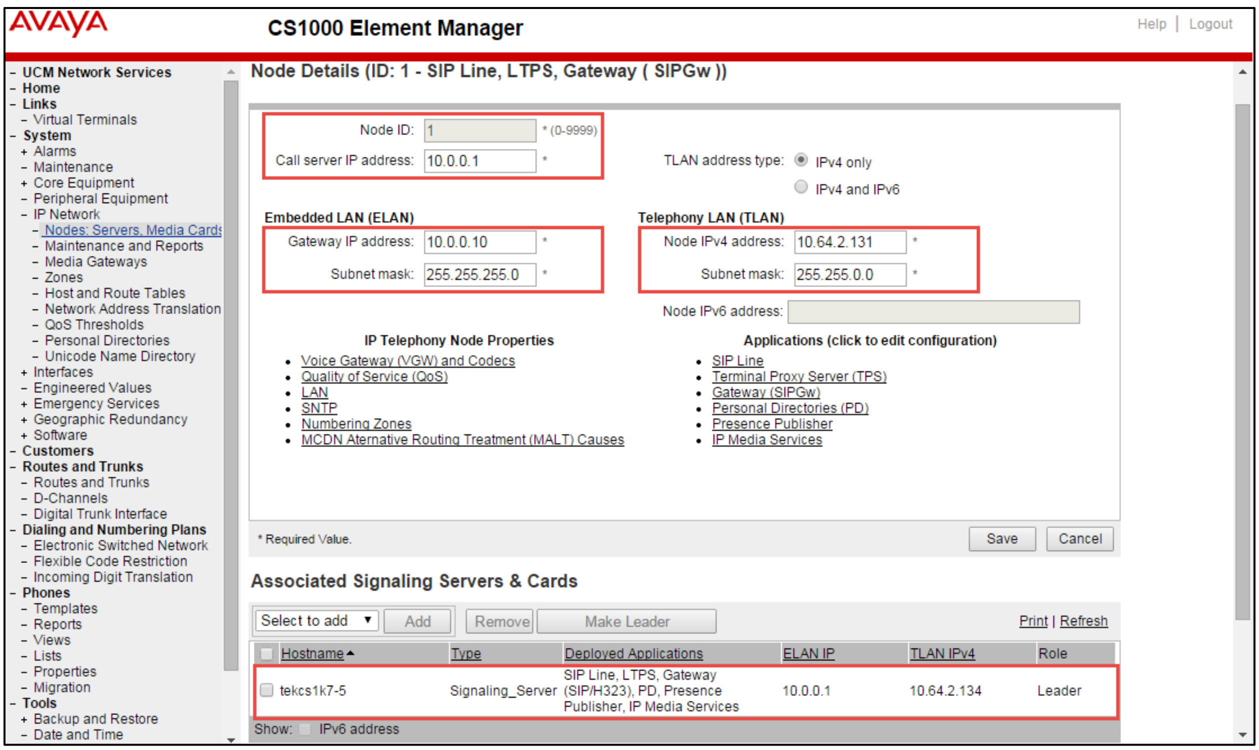

AVAYA
CS1000 Element Manager
Help | Logout
– UCM Network Services
– Home
– Links
– Virtual Terminals
– System
+ Alarms
– Maintenance
+ Core Equipment
– Peripheral Equipment
+ IP Network
+ Interfaces
– Engineered Values
+ Emergency Services
+ Geographic Redundancy
+ Software
– Customers
– Routes and Trunks
– Routes and Trunks
– D-Channels
– Digital Trunk Interface
– Dialing and Numbering Plans
– Electronic Switched Network
– Flexible Code Restriction
– Incoming Digit Translation
– Phones
– Templates
– Reports
– Views
– Lists
– Properties
– Migration
Managing: 10.0.0.1 Username: admin
System Overview
System Overview
IP Address: 10.0.0.1
Type: Avaya Communication Server 1000E CPPM Linux
Version: 4121
Release: 765 P

IP Telephony Nodes

Navigation Path: CS1000 Element Manager→ IP Network → Nodes:Servers,Media Cards



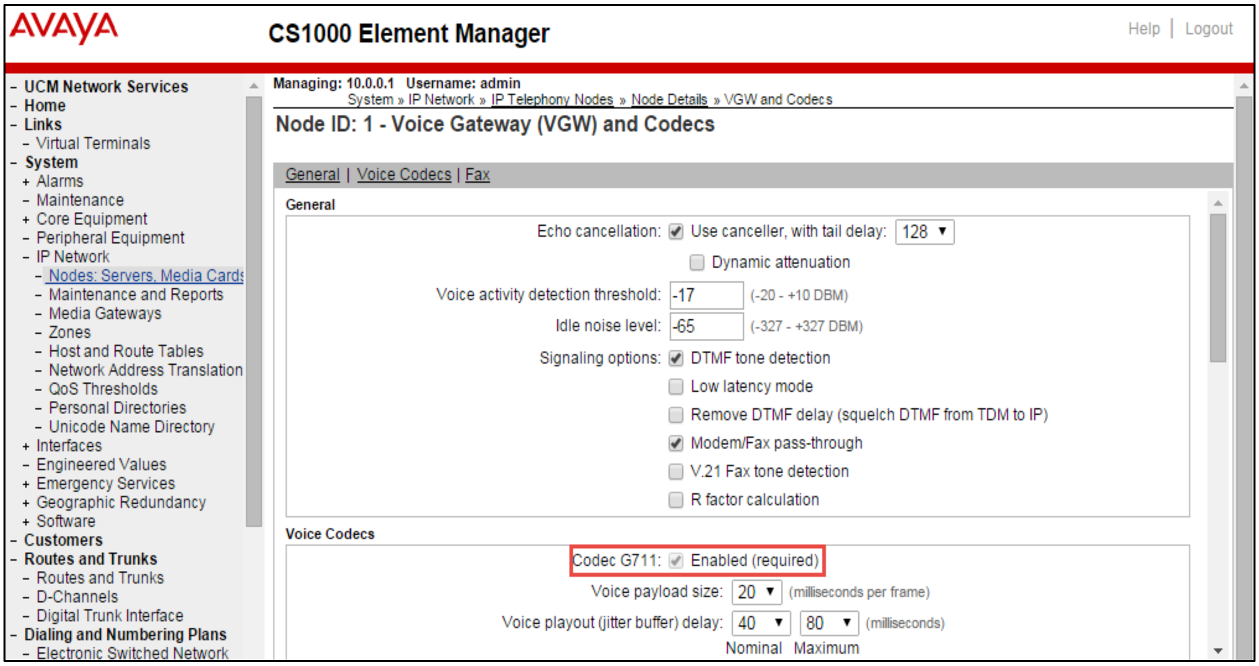
Set Node ID = 1. This is used for this testing.
Set Call server IP address = 10.0.0.1. This is used for this testing.
Set Gateway IP address= 10.0.0.10.This is used for this testing.
Set Node IPV4 address = 10.64.2.131. This is used for this testing.

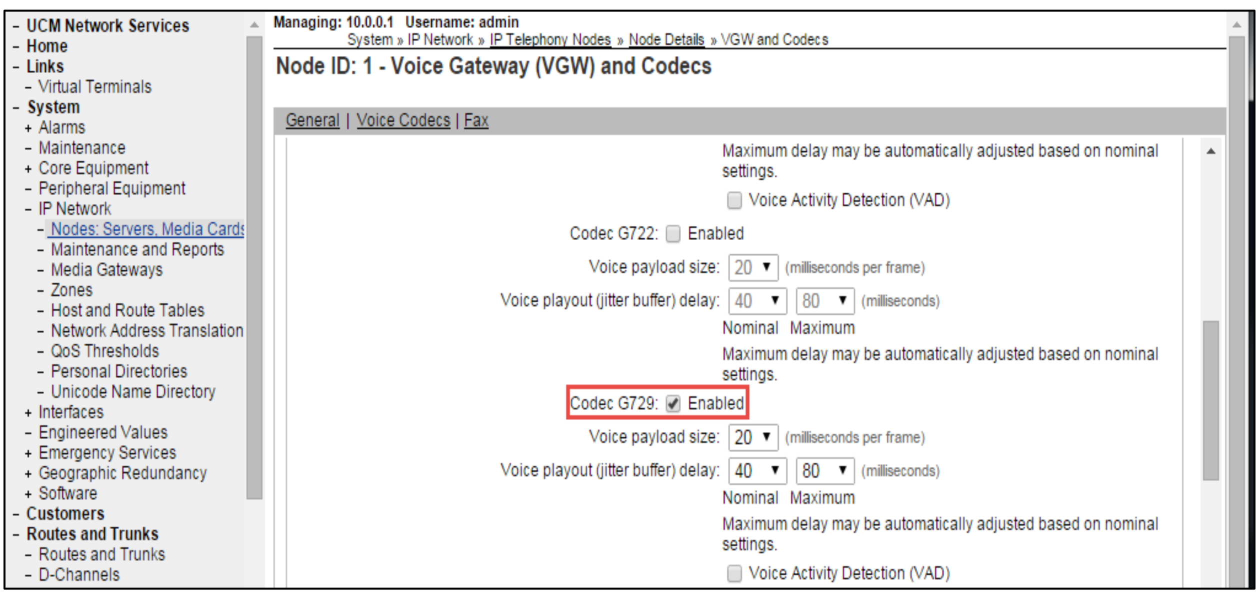


Voice Gateway (VGW) and Codecs

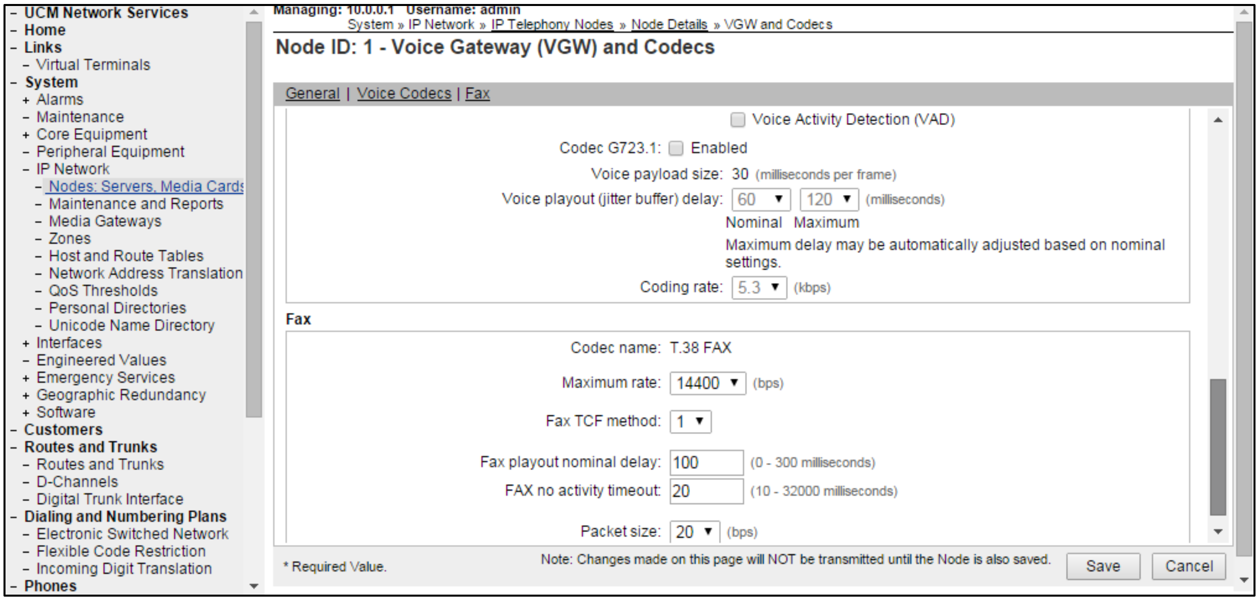
Navigation Path: CS1000 Element Manager→ IP Network → Nodes:Servers,Media Cards

Check Codec G711 Enabled required.
Check Codec G729 Enabled required.





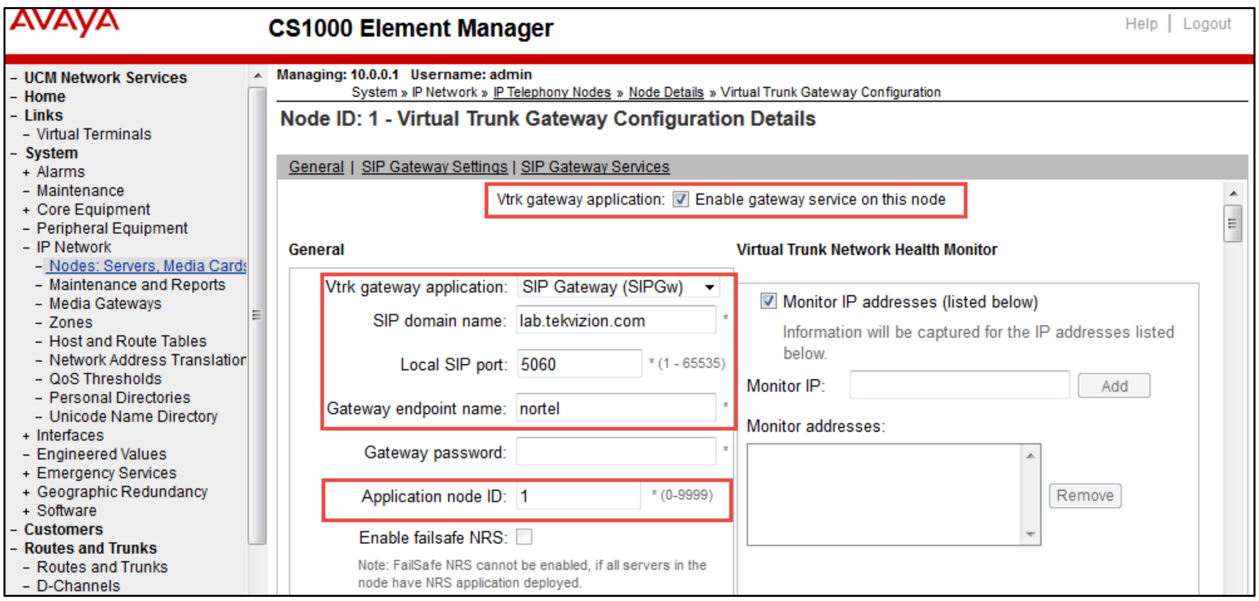
Avaya CS1000E VGW and Codecs (Continued)



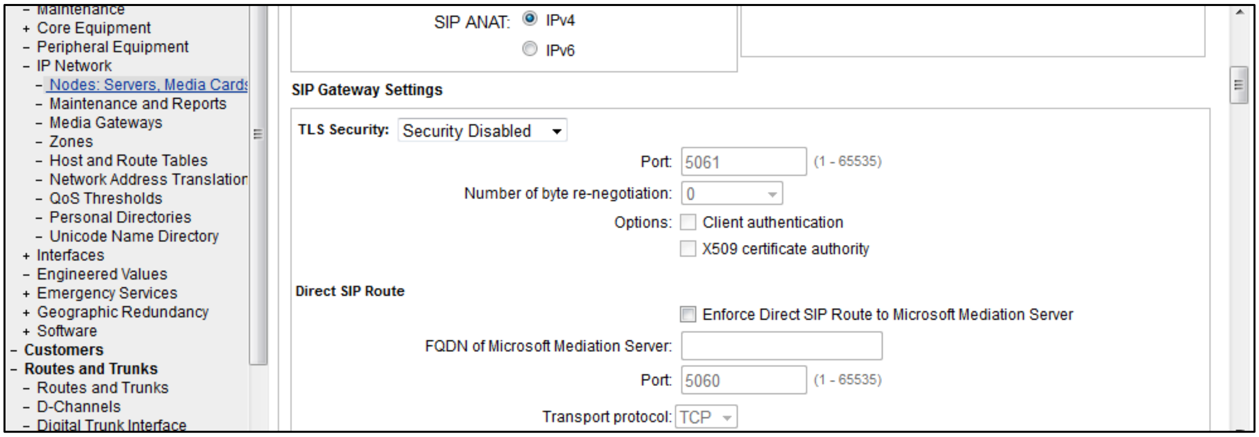
Avaya CS1000E SIP Gateway

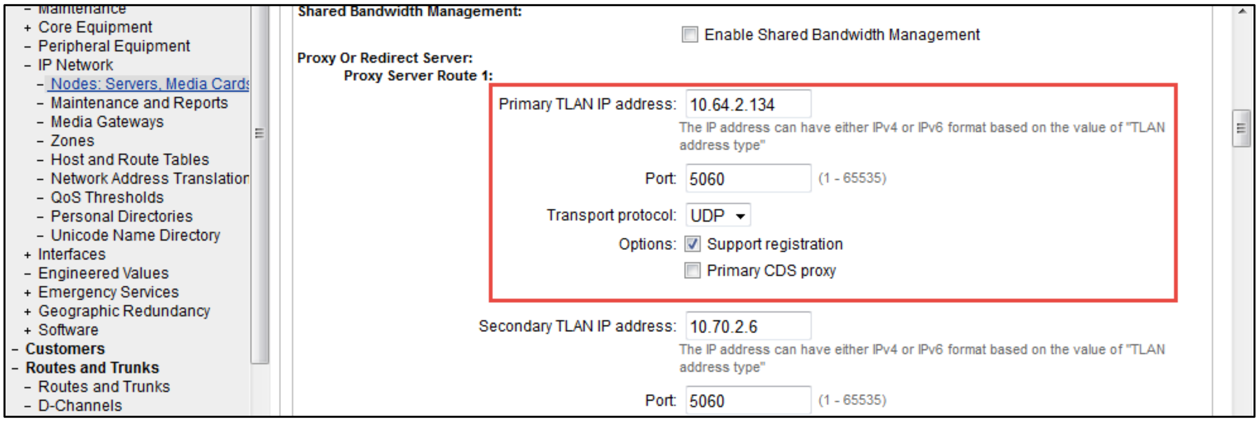
Navigation Path: CS1000 Element Manager→ IP Network → Nodes:Servers,Media Cards→ Select Node → Gateway(SIPGw)

Check VTRK gateway application Enable gateway application on this node.
Set Vtrk gateway application = SIP Gateway (SIPGw).
Set SIP domain name = lab.tekvizion.com. This is used for this testing.
Set Local SIP port = 5060.
Set Gateway endpoint name = nortel. This is used for this testing.
Application node ID = 1. This is used for this testing.
In the Proxy or Redirect Server set the Primary TLAN IP address = 10.64.2.134. This is used for this testing.
Set Port = 5060.
Set Transport Protocol= TCP. This is the transport protocol used for SIP message exchange between the Gateway and Redirect/Proxy Server. The two options are TCP and UDP. TCP is the default option.

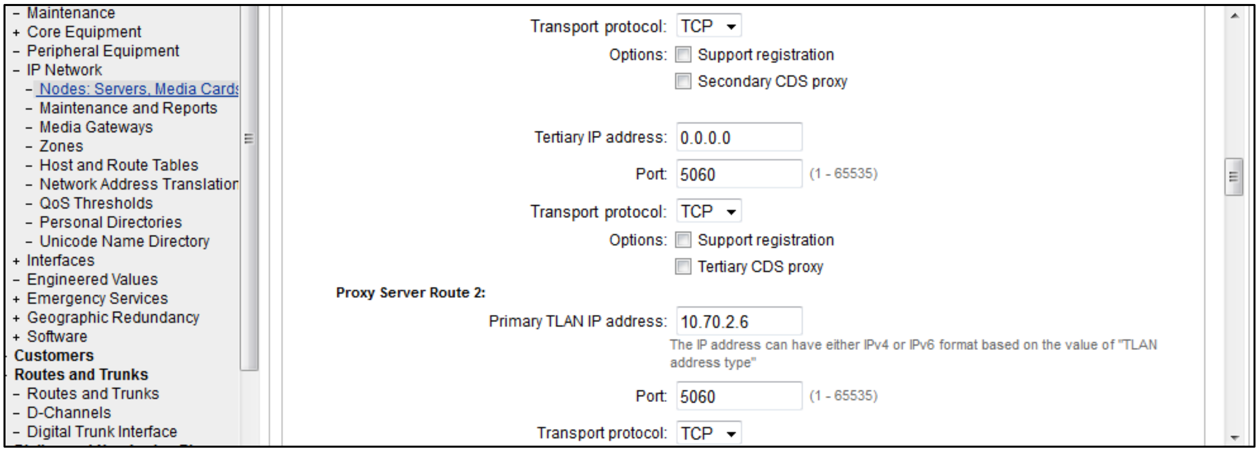


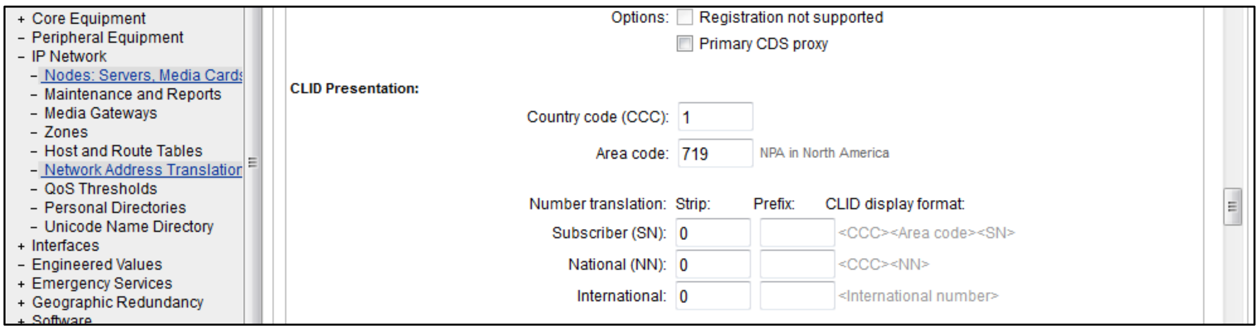
Avaya CS1000E SIP Gateway (Continued)





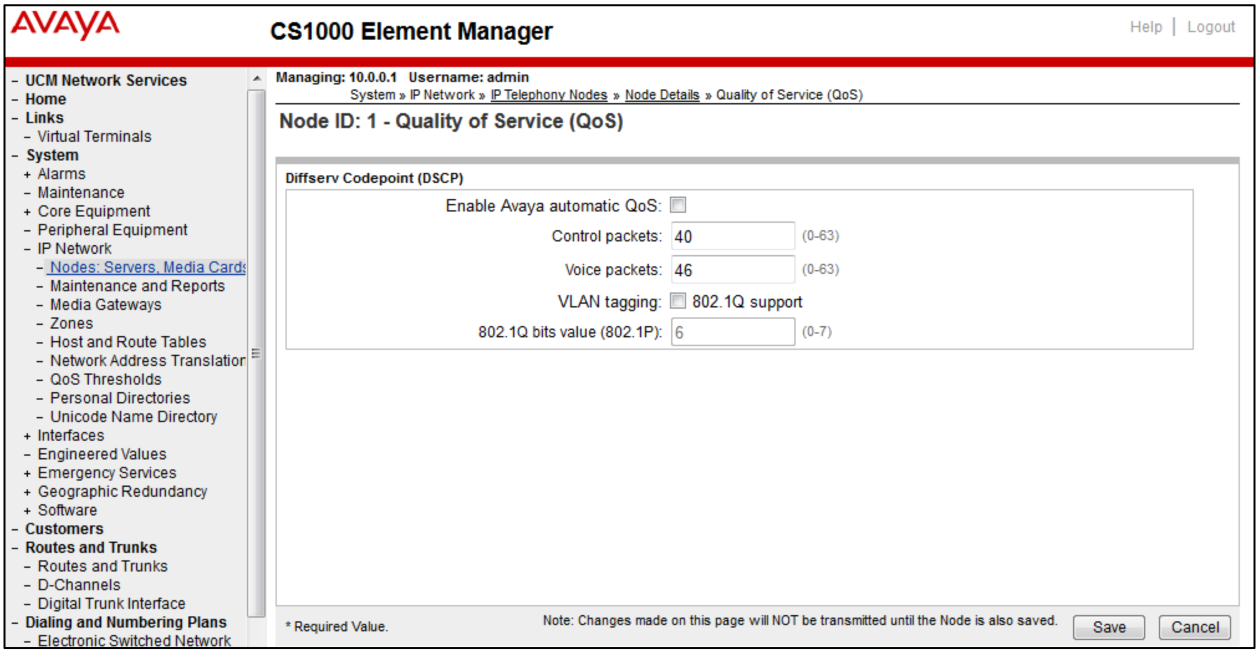
Avaya CS1000E SIP Gateway (Continued)





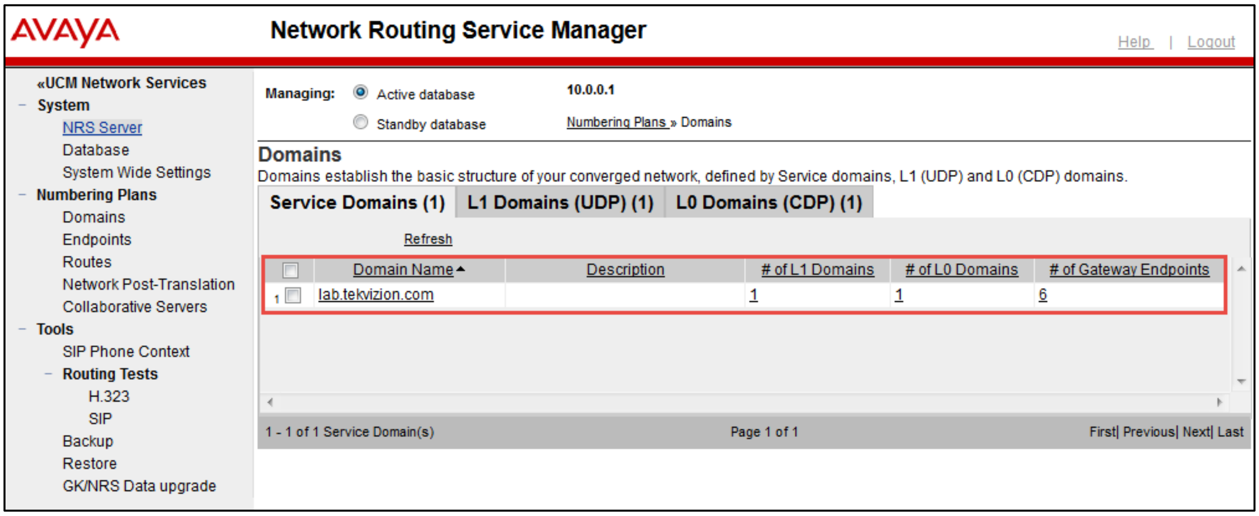
Quality of Service

Navigation Path: CS1000 Element Manager→ IP Network → Nodes:Servers,Media Cards→ Select Node → Quality of Service(Qos)

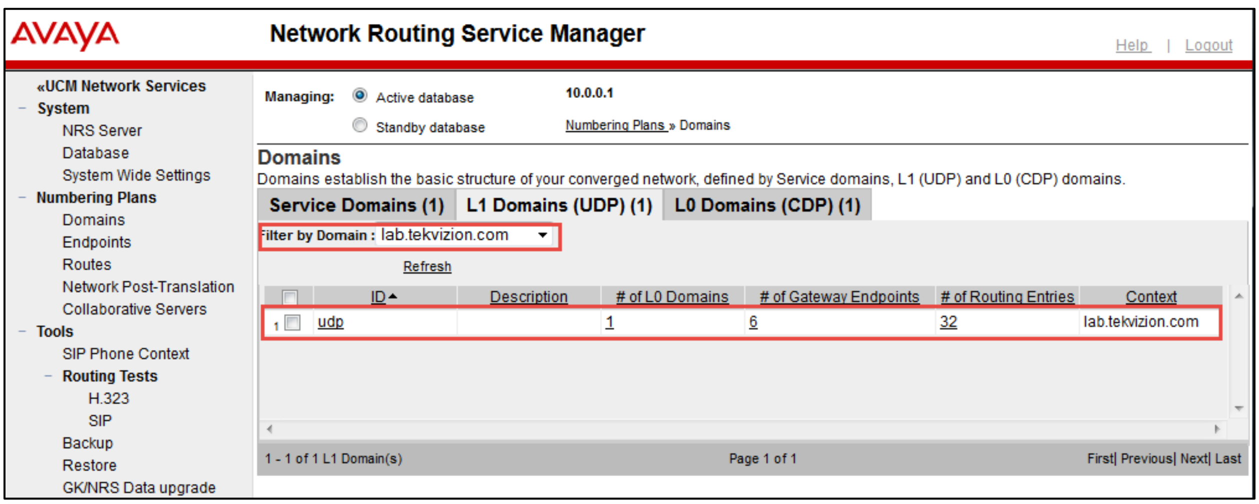


Network Routing Service Manager

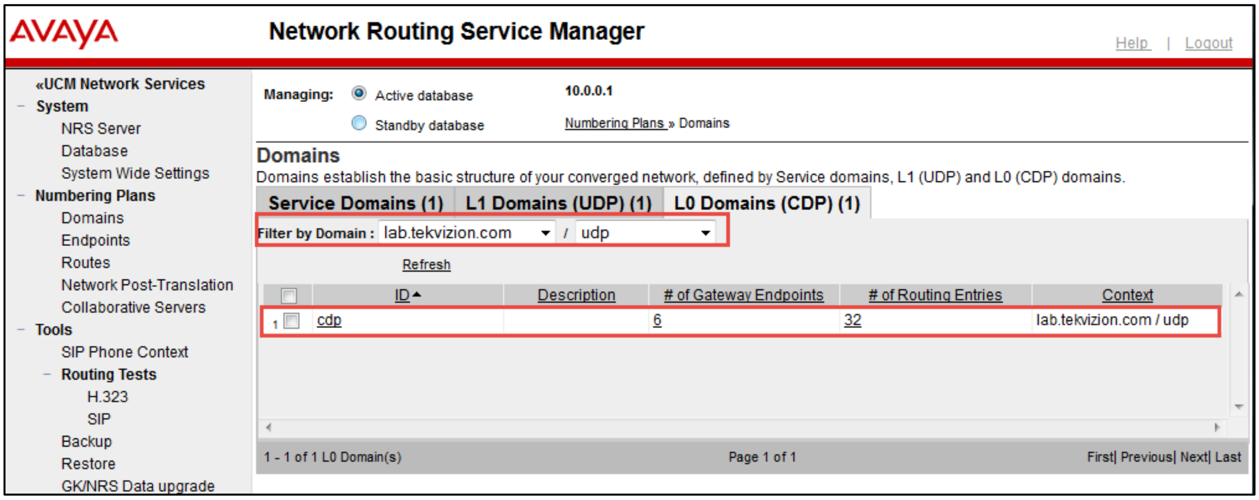
Navigation Path: CS1000 NRSM→ Numbering Plans→ Domains



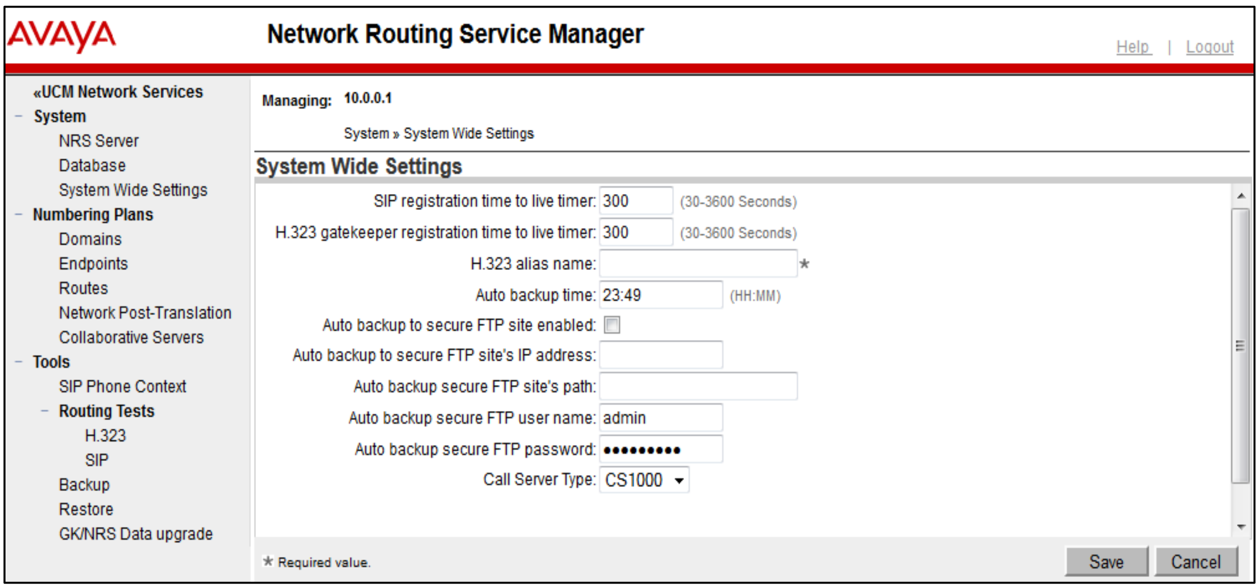
L1 Domain



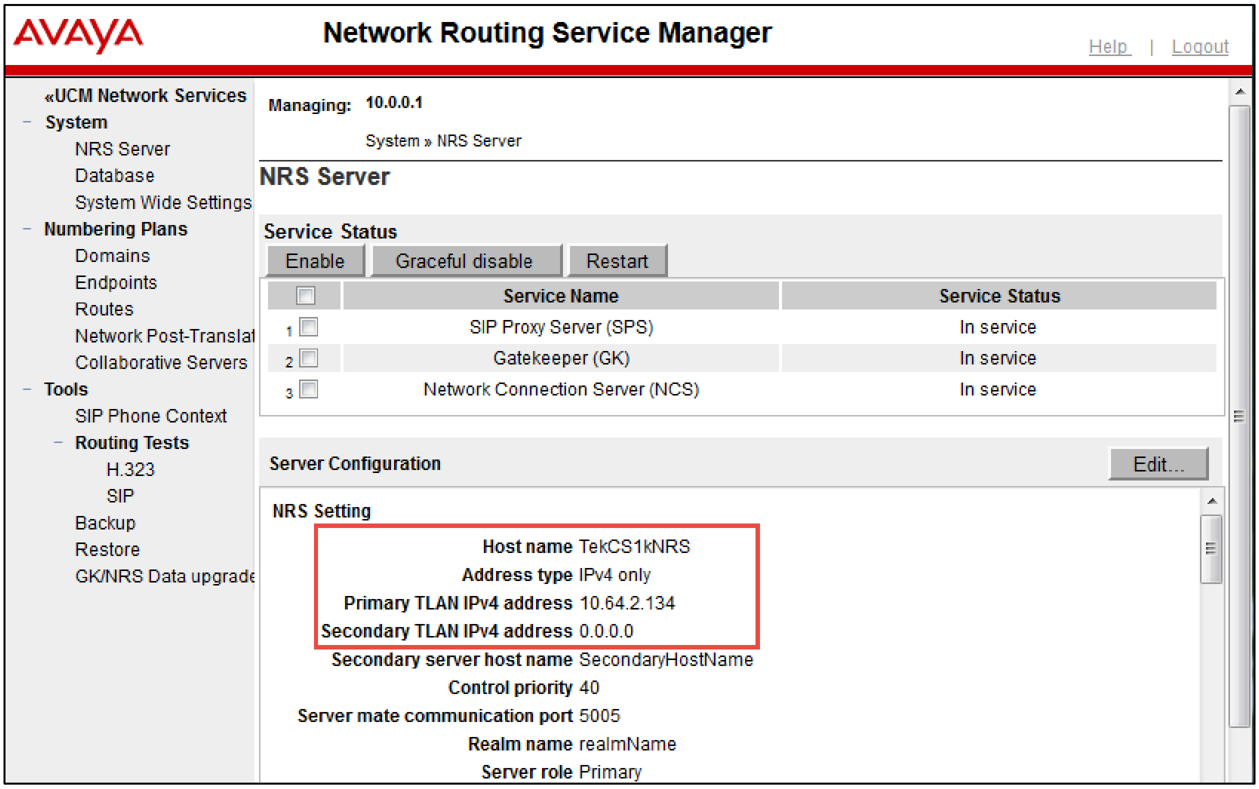
LO Domain

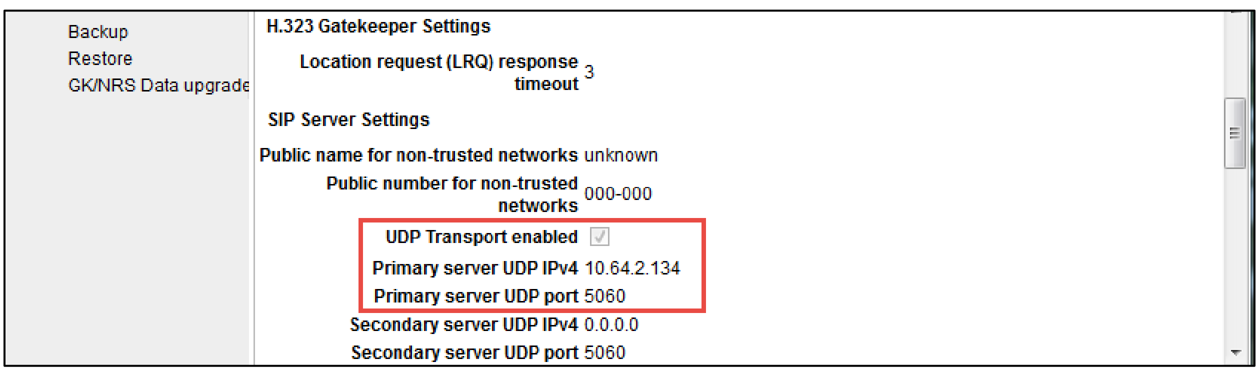


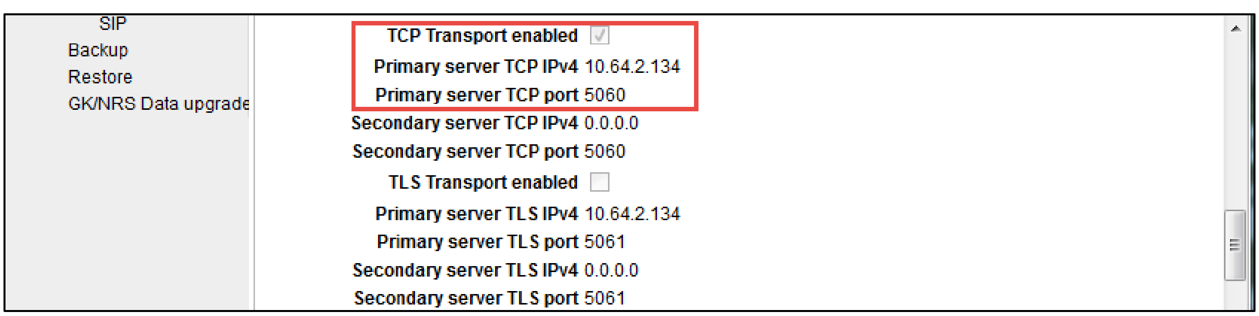
System Wide Settings



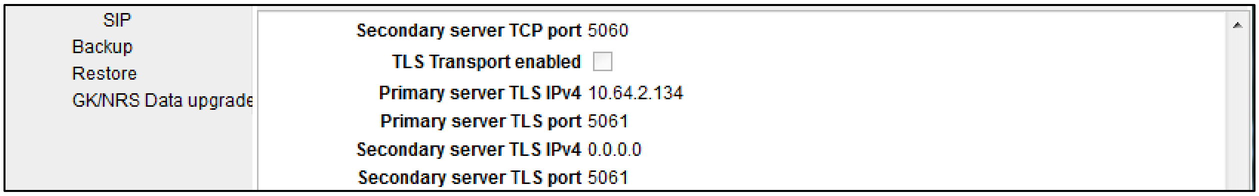
NRS Server

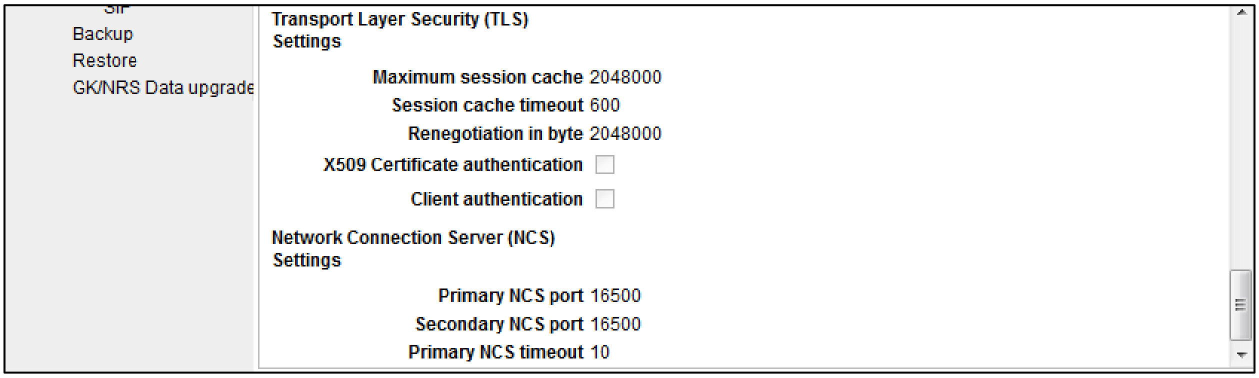

AVAYA
Network Routing Service Manager
Help | Logout
«UCM Network Services
System
NRS Server
Database
System Wide Settings
Numbering Plans
Domains
Endpoints
Routes
Network Post-Translat
Collaborative Servers
Tools
SIP Phone Context
Routing Tests
H.323
SIP
Backup
Restore
GK/NRS Data upgrade
Managing: 10.0.0.1
System » NRS Server
NRS Server
Service Status
Enable
Graceful disable
Restart
Service Name
Service Status
1 SIP Proxy Server (SPS) In service
2 Gatekeeper (GK) In service
3 Network Connection Server (NCS) In service
Server Configuration
Edit...
NRS Setting
Host name TekCS1kNRS
Address type IPv4 only
Primary TLAN IPv4 address 10.64.2.134
Secondary TLAN IPv4 address 0.0.0.0
Secondary server host name SecondaryHostName
Control priority 40
Server mate communication port 5005
Realm name realmName
Server role Primary


Backup
Restore
GK/NRS Data upgrade
H.323 Gatekeeper Settings
Location request (LRQ) response timeout 3
SIP Server Settings
Public name for non-trusted networks unknown
Public number for non-trusted networks 000-000
UDP Transport enabled
Primary server UDP IPv4 10.64.2.134
Primary server UDP port 5060
Secondary server UDP IPv4 0.0.0.0
Secondary server UDP port 5060

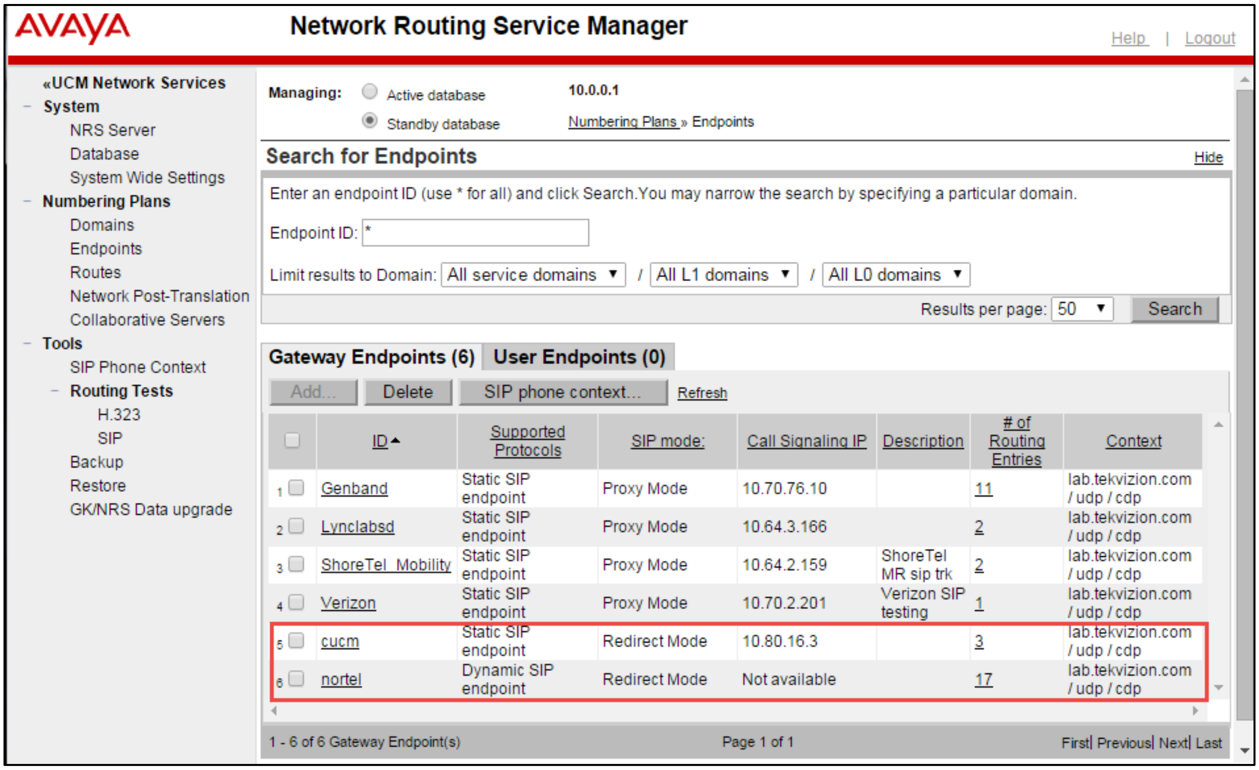

SIP
Backup
Restore
GK/NRS Data upgrade
TCP Transport enabled
Primary server TCP IPv4 10.64.2.134
Primary server TCP port 5060
Secondary server TCP IPv4 0.0.0.0
Secondary server TCP port 5060
TLS Transport enabled
Primary server TLS IPv4 10.64.2.134
Primary server TLS port 5061
Secondary server TLS IPv4 0.0.0.0
Secondary server TLS port 5061

Avaya CS1000E NRS Server Configuration (Continued)



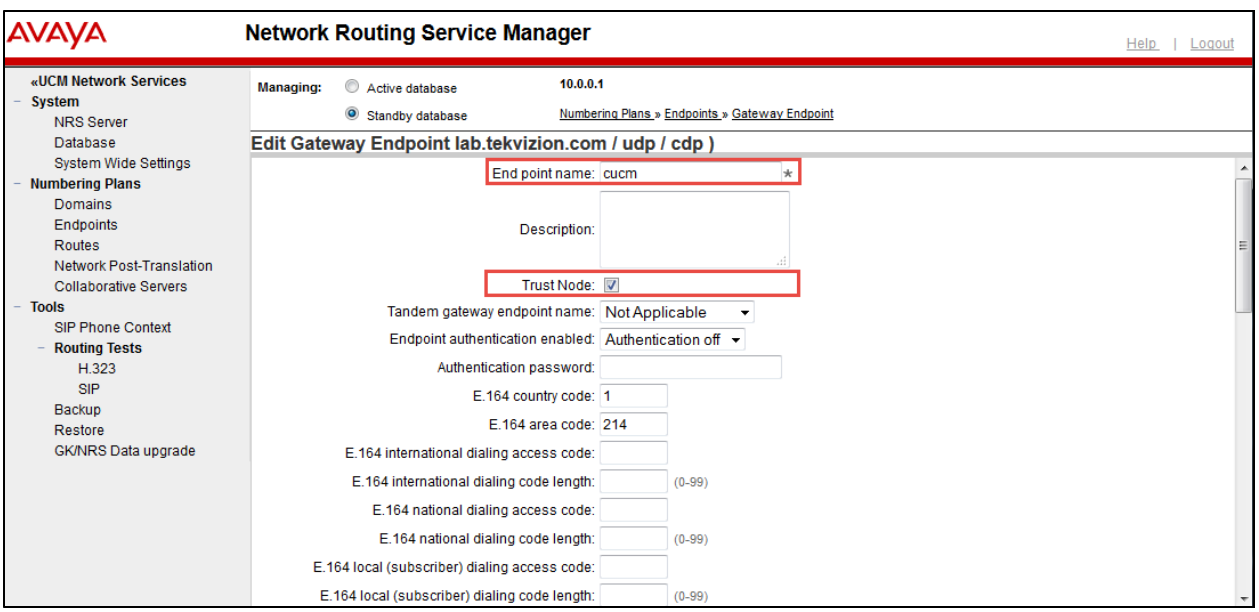


Endpoints

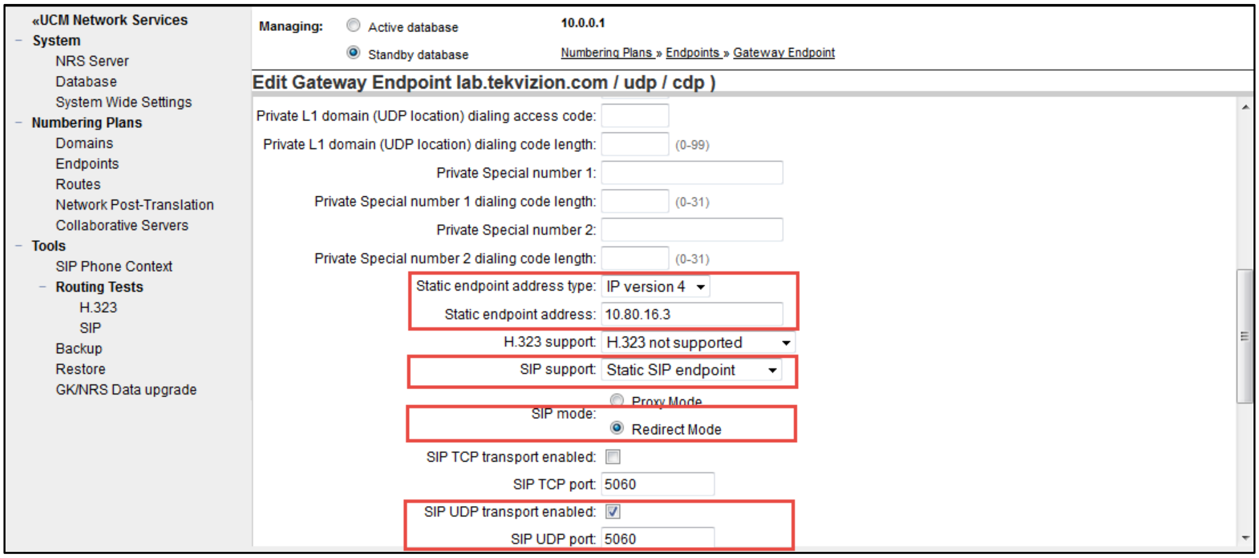


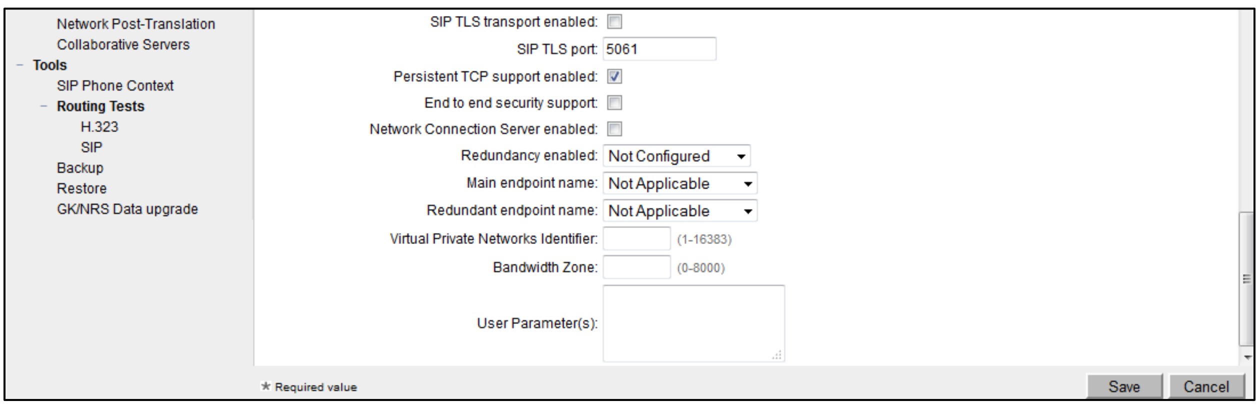
Cisco UCM Endpoint

Navigation Path: CS1000 NRSM→ Numbering Plans→ Endpoints→ cucm



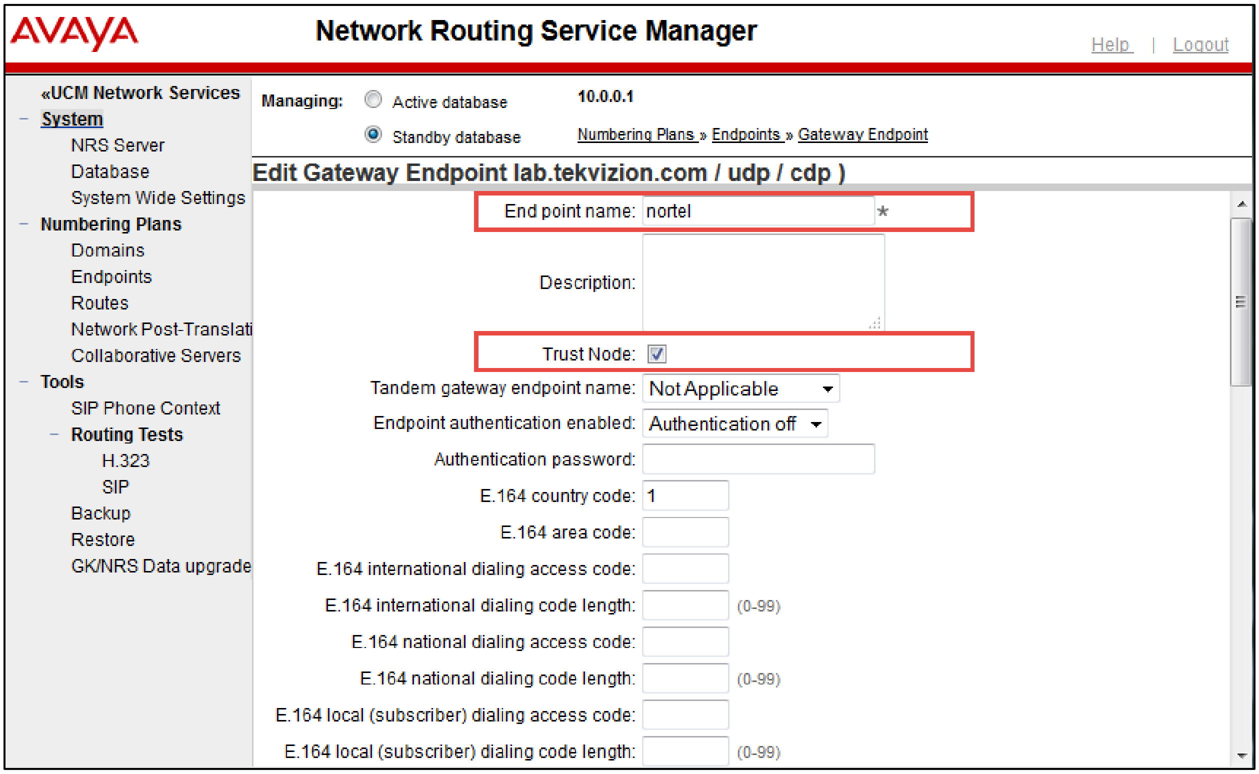
Avaya CS1000E Endpoint Configuration (Continued)

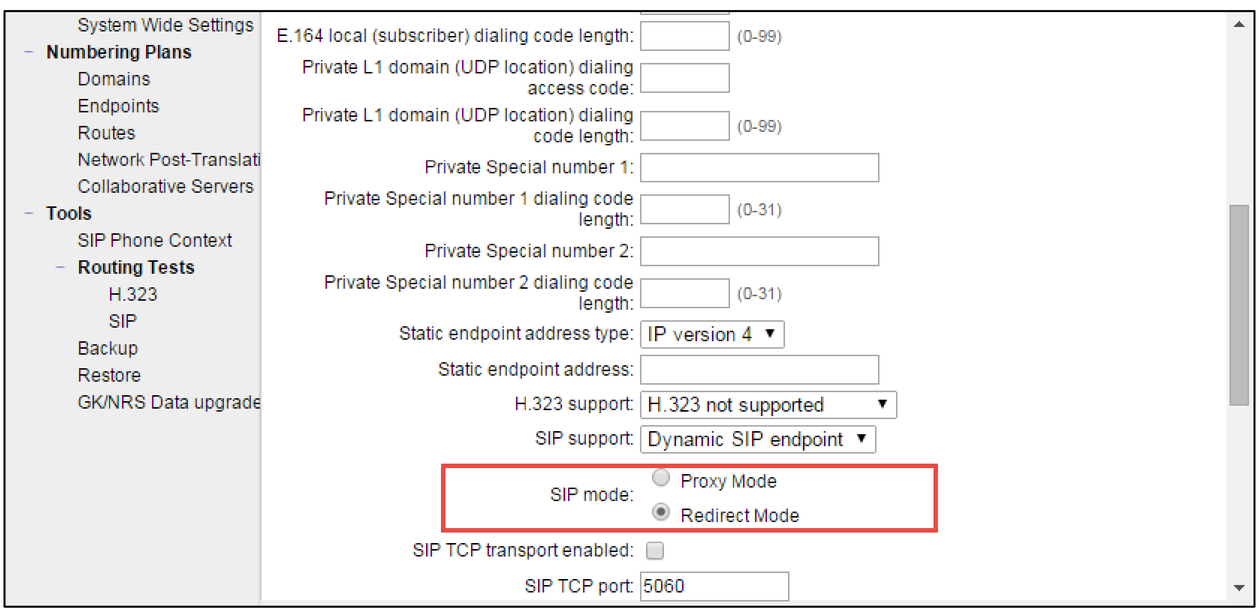




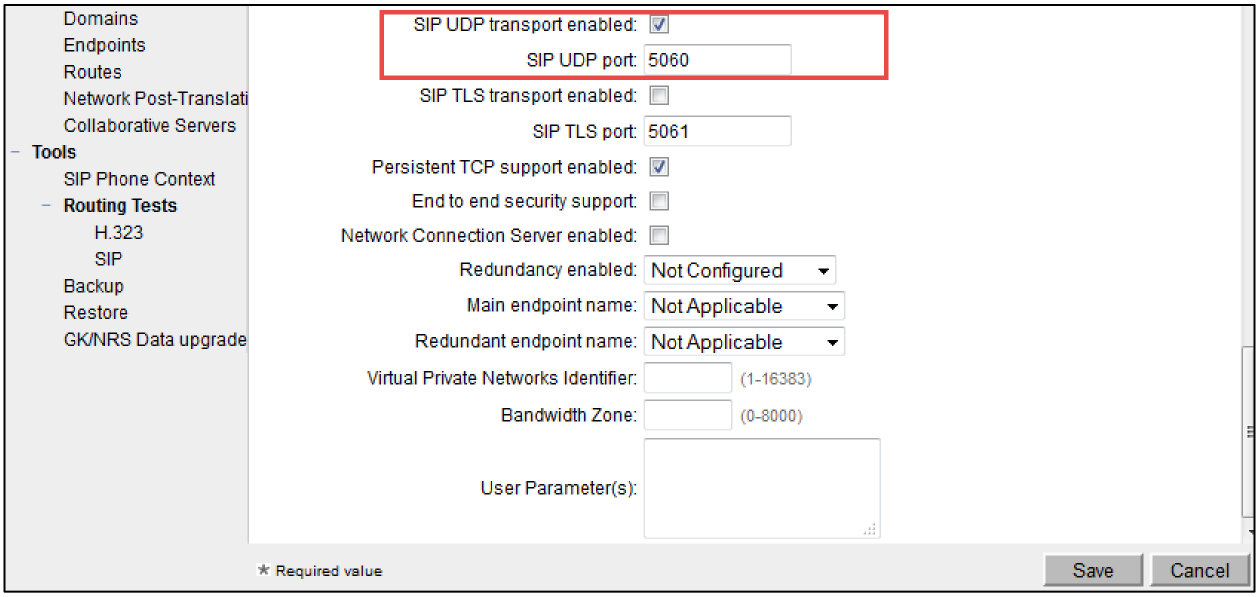
Cisco CS1000E Endpoint

Navigation Path: CS1000 NRSM→ Numbering Plans→ Endpoints→ Avaya

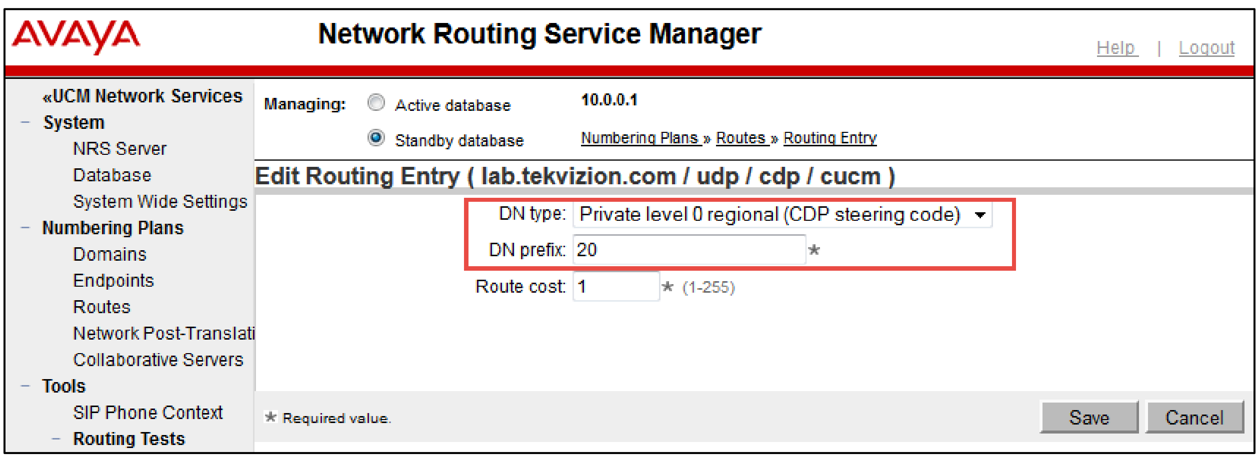




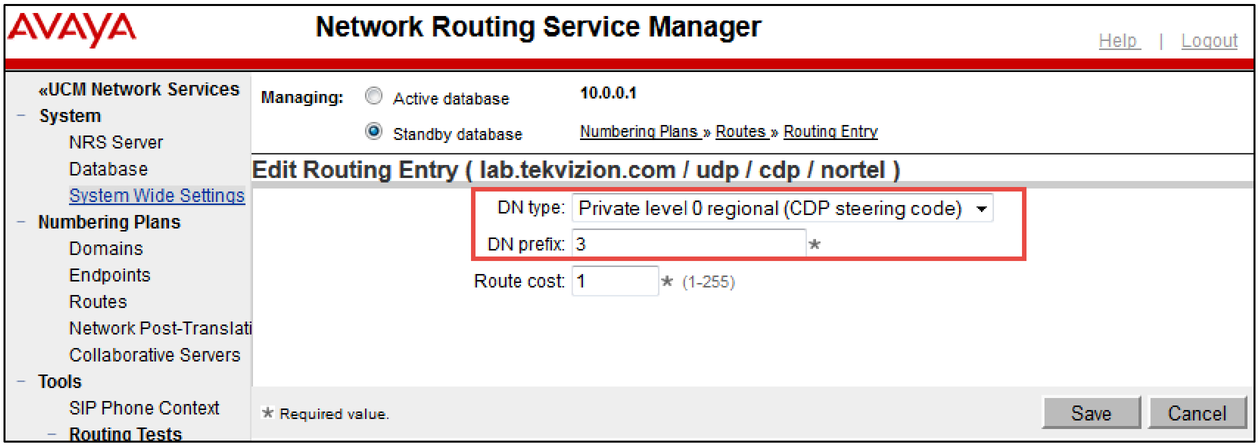
Cisco UCM Endpoint Configuration (Continued)



Cisco UCM Routing Entry



Avaya CS1000E Routing Entry

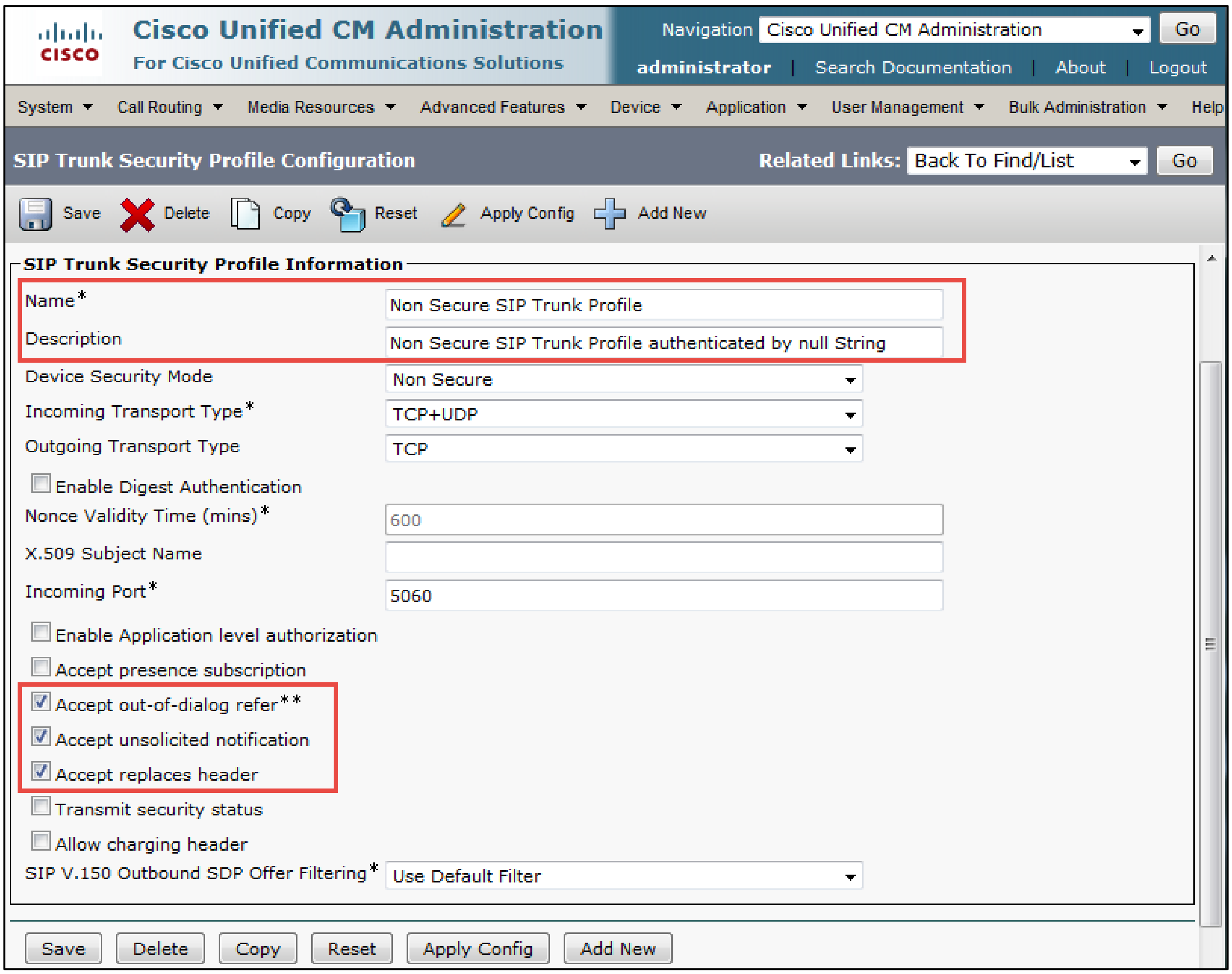


Configuring the Cisco Unified Communications Manager

Cisco Unified Communications Manager Software Version

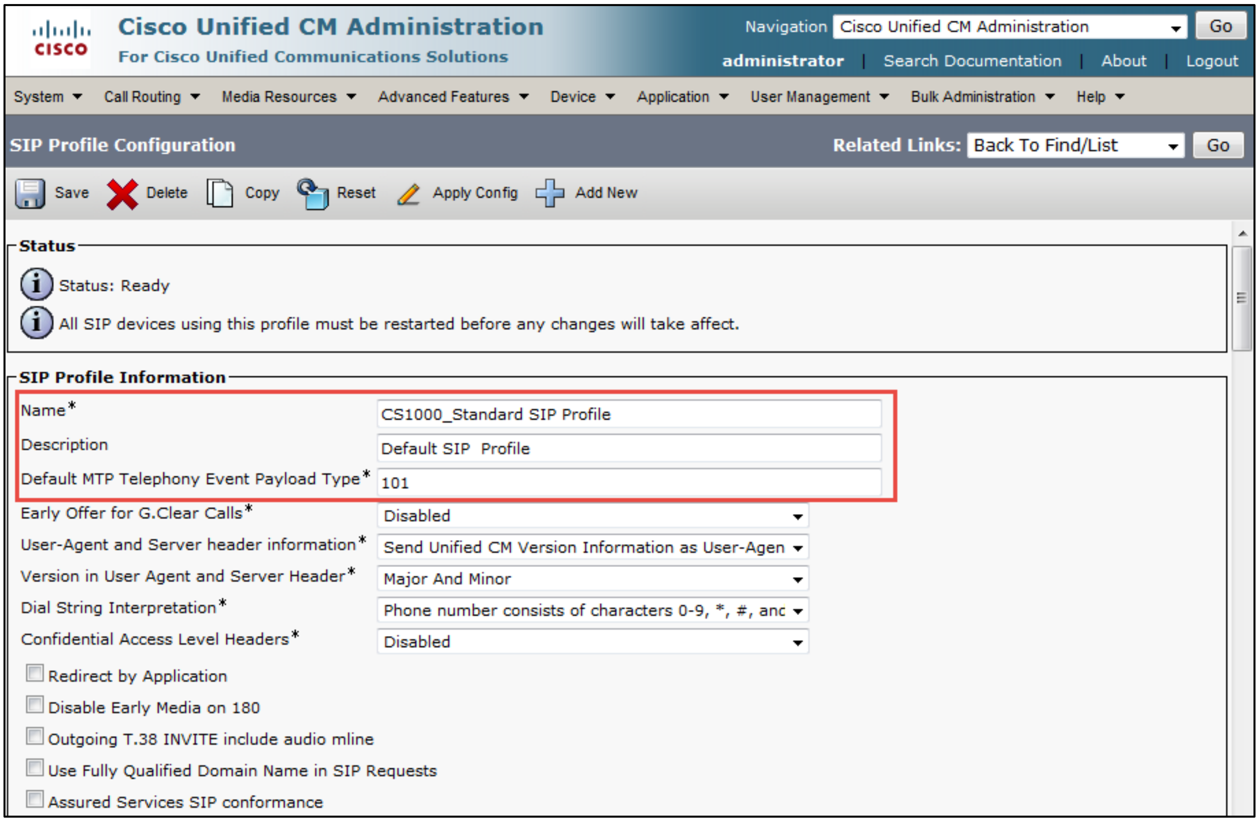
Cisco UCM SIP Trunk Security Profile

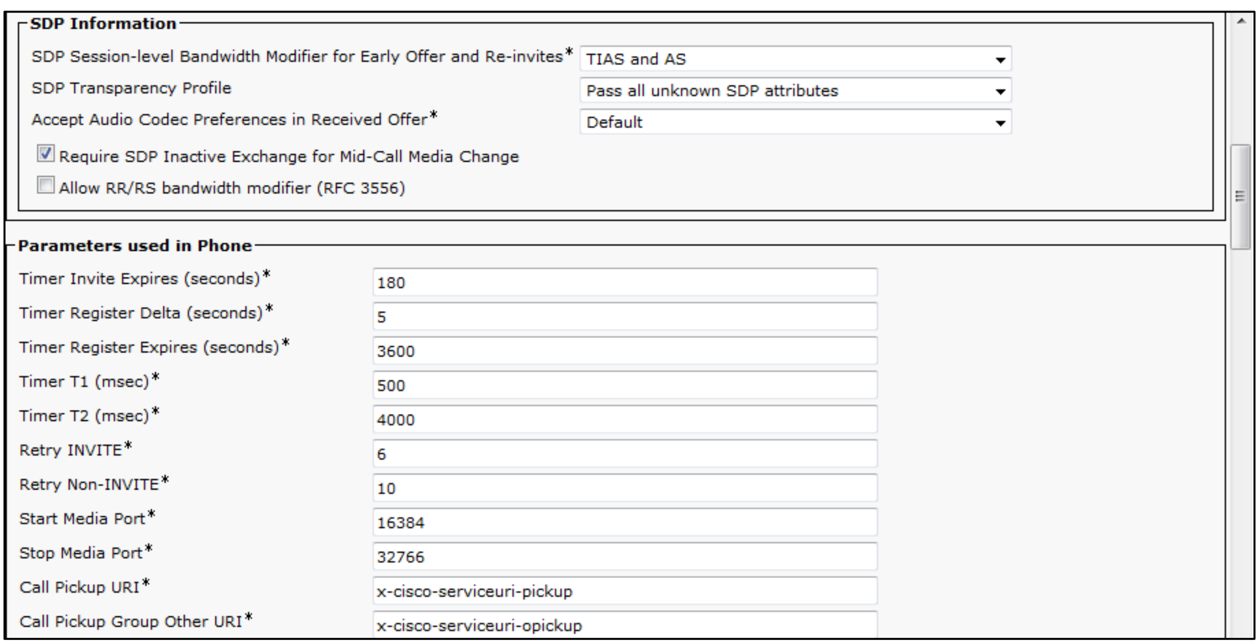
Set Name*= Non Secure SIP Trunk Profile. This is used for this example.
Set Description = This text is used to identify this SIP Trunk Security Profile.
Check Accept out of dialog refer.
Check Accept unsolicited notification.
Check Accept replaces header.
All other values are default.



Cisco UCM SIP Profile

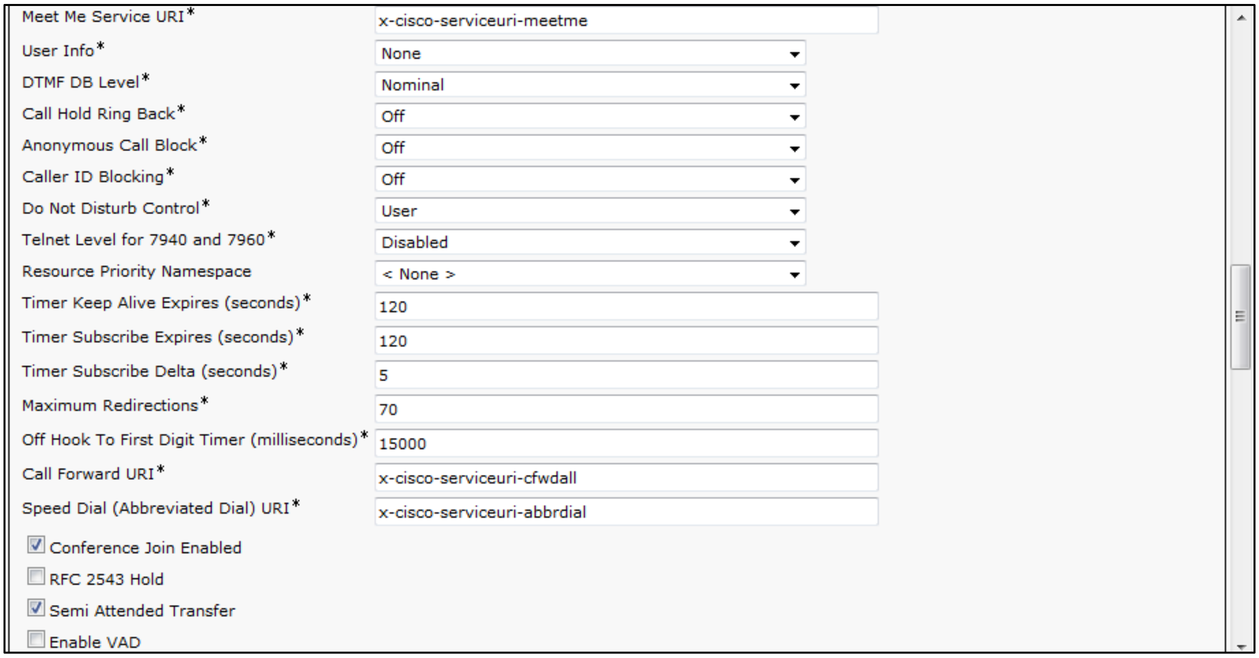
Set Name*= Early Offer SIP Profile. This is used for this example.
Set Description = This text is used to identify this SIP Profile.
All other values are default.





Cisco Unified Communications Manager SIP Profile (Continued)

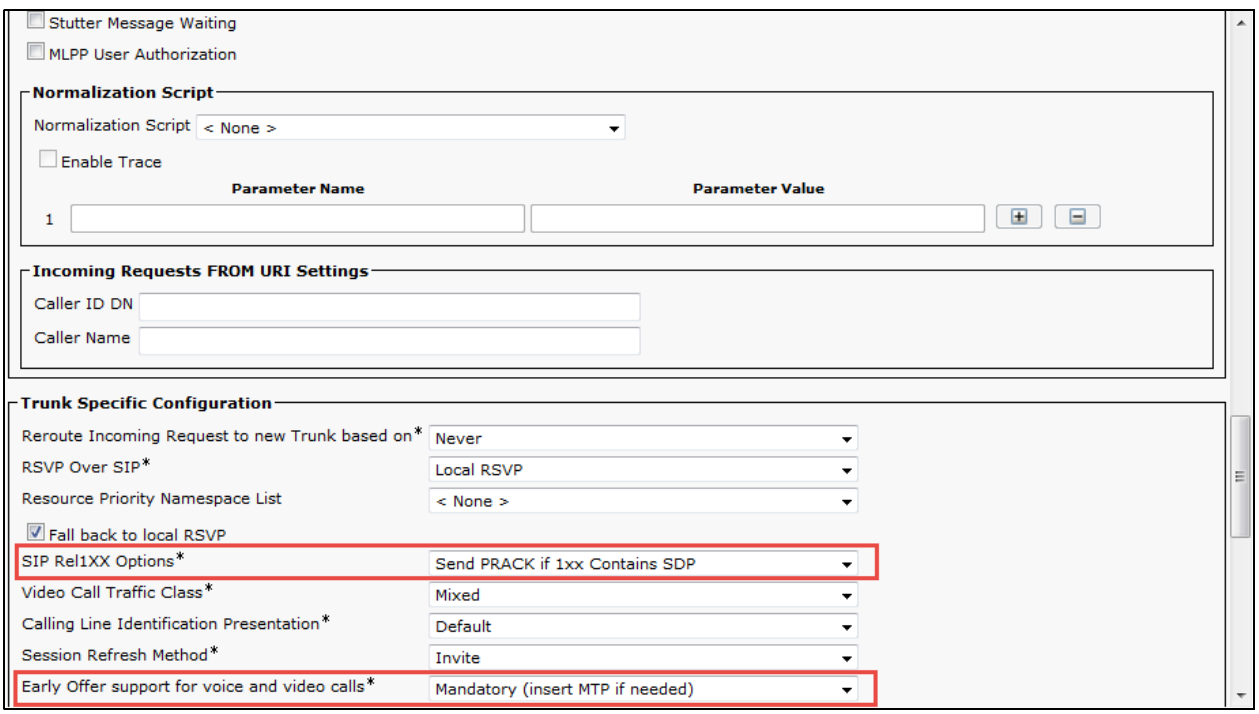
These values are default.



Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off
Caller ID Blocking*	Off
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
Resource Priority Namespace	< None >
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70
Off Hook To First Digit Timer (milliseconds)*	15000
Call Forward URI*	x-cisco-serviceuri-cfwdall
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial

- [x] Conference Join Enabled
- [ ] RFC 2543 Hold
- [x] Semi Attended Transfer
- [ ] Enable VAD

Set SIP Rel1XX Options* = Send PRACK if 1xx Contains SDP
Check Early Offer support for voice and video calls (insert MTP if needed)
All other values are default.



- [ ] Stutter Message Waiting
- [ ] MLPP User Authorization

Normalization Script

Normalization Script: < None >

- [ ] Enable Trace

	Parameter Name	Parameter Value
1		

Incoming Requests FROM URI Settings

Caller ID DN

Caller Name

Trunk Specific Configuration

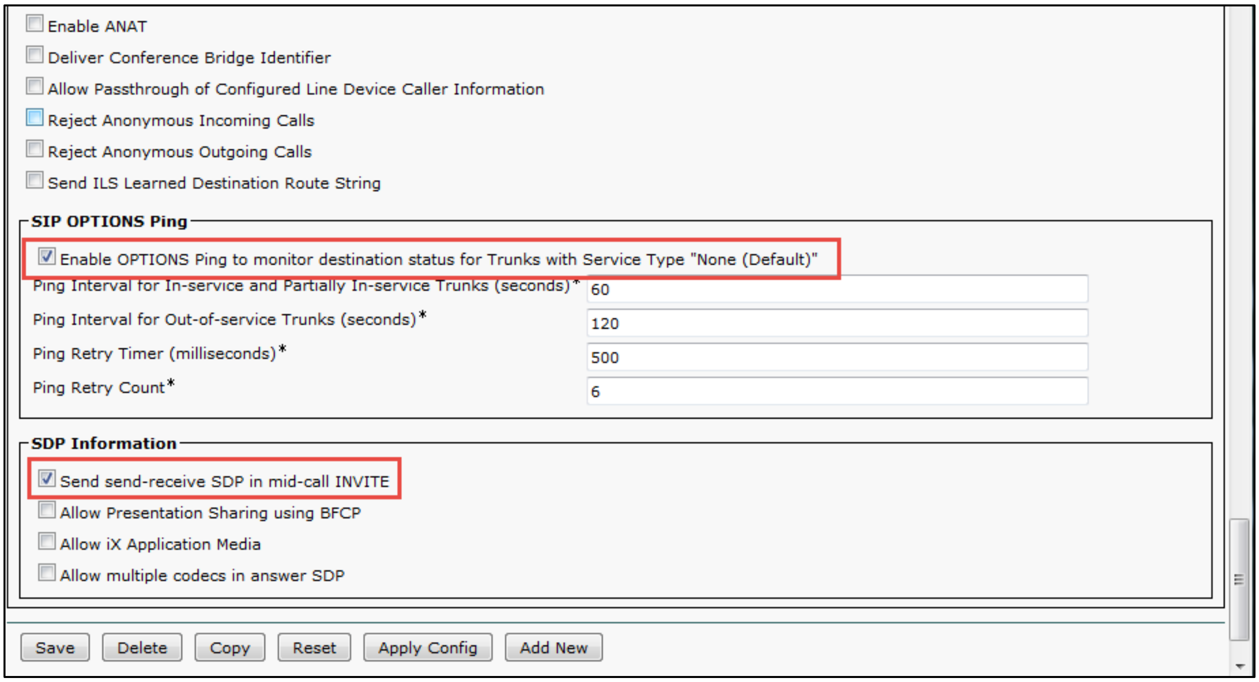
Reroute Incoming Request to new Trunk based on*	Never
RSVP Over SIP*	Local RSVP
Resource Priority Namespace List	< None >

- [x] Fall back to local RSVP

SIP Rel1XX Options*	Send PRACK if 1xx Contains SDP
Video Call Traffic Class*	Mixed
Calling Line Identification Presentation*	Default
Session Refresh Method*	Invite
Early Offer support for voice and video calls*	Mandatory (insert MTP if needed)

Cisco Unified Communications Manager SIP Profile (Continued)

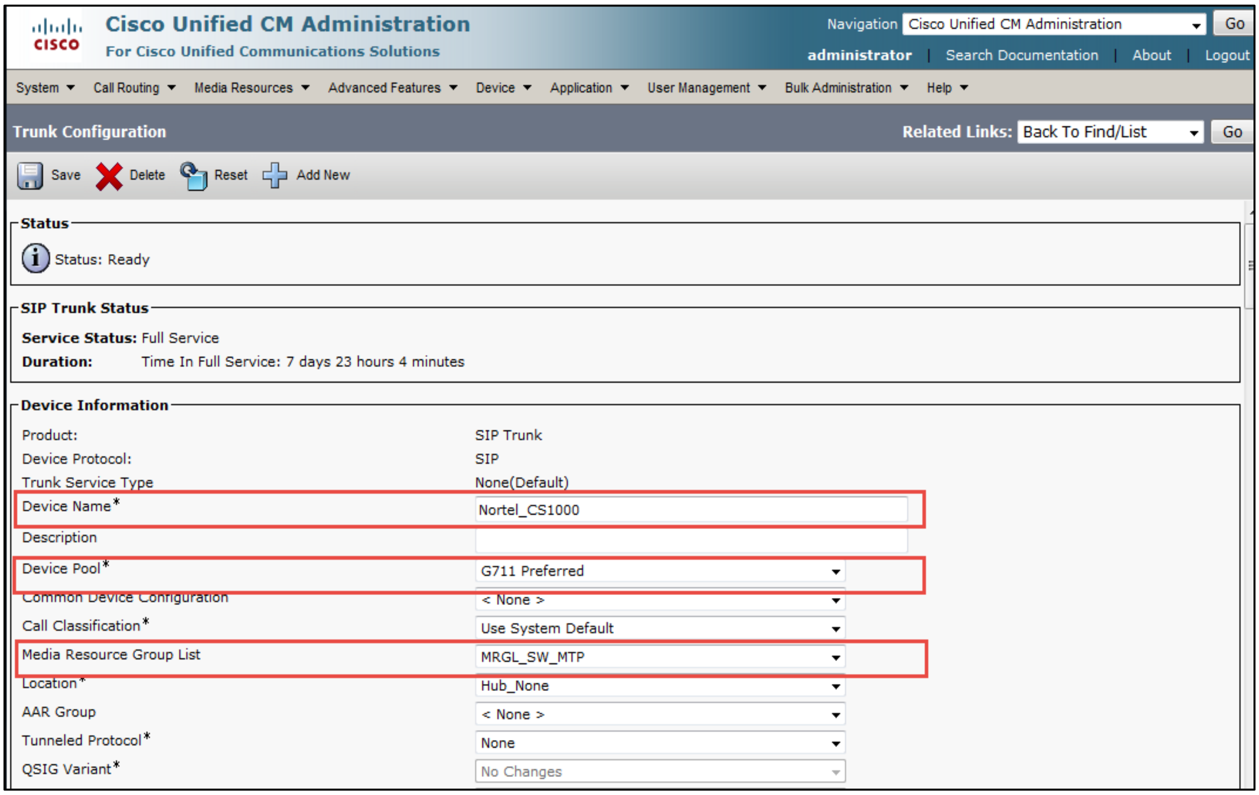
Check Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"
Check Send send-receive SDP in mid-call INVITE.
All other values are default.

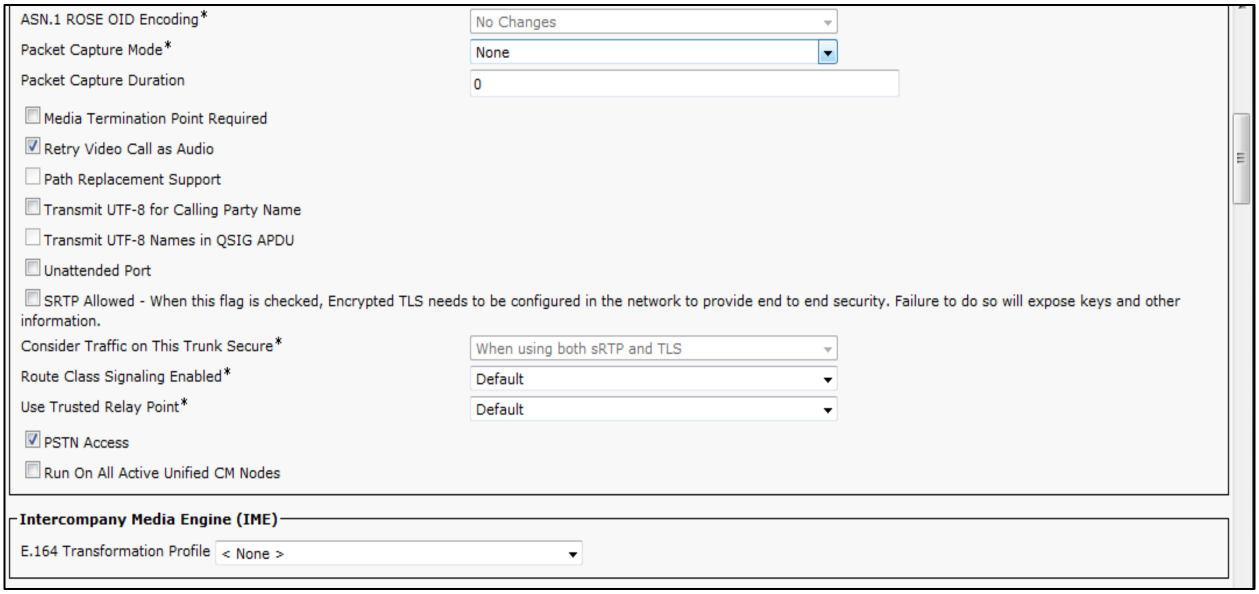


Cisco UCM SIP Trunk to Avaya Configuration

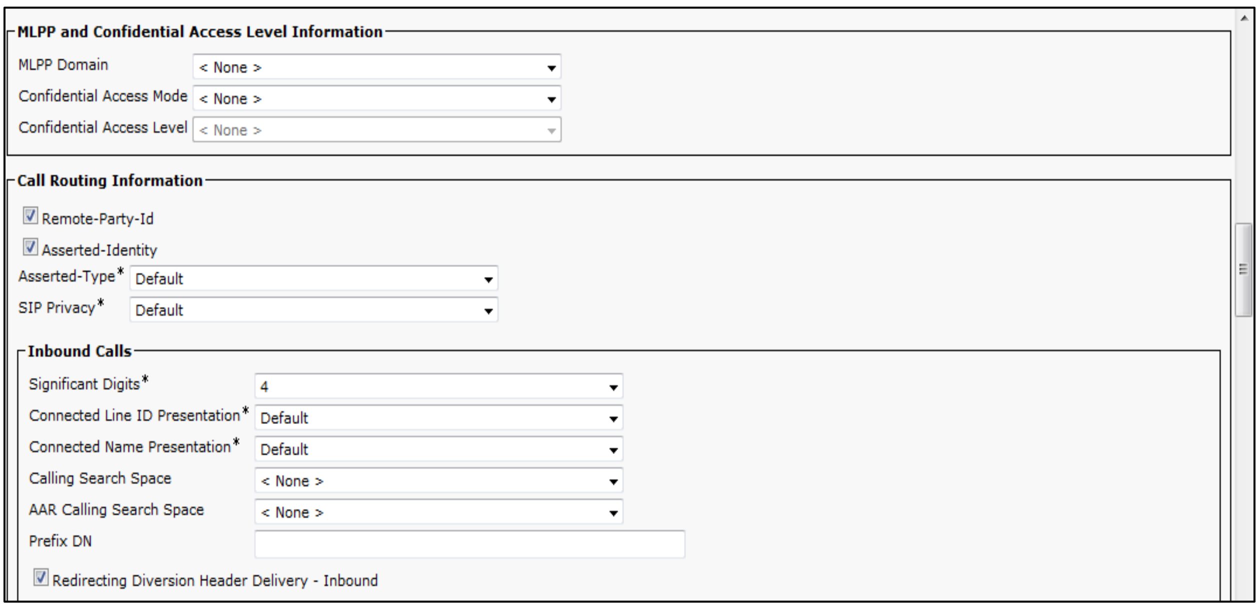
Set Device Name*= Avaya_CS1000. This is used for this example.
Set Description = This text is used to identify this Trunk Group.
Set Device Pool* = G711 Preferred. This is used for this example.
Set Media Resource Group List = MRGL_SW_MTP. This is used for this example.

All other values are default.



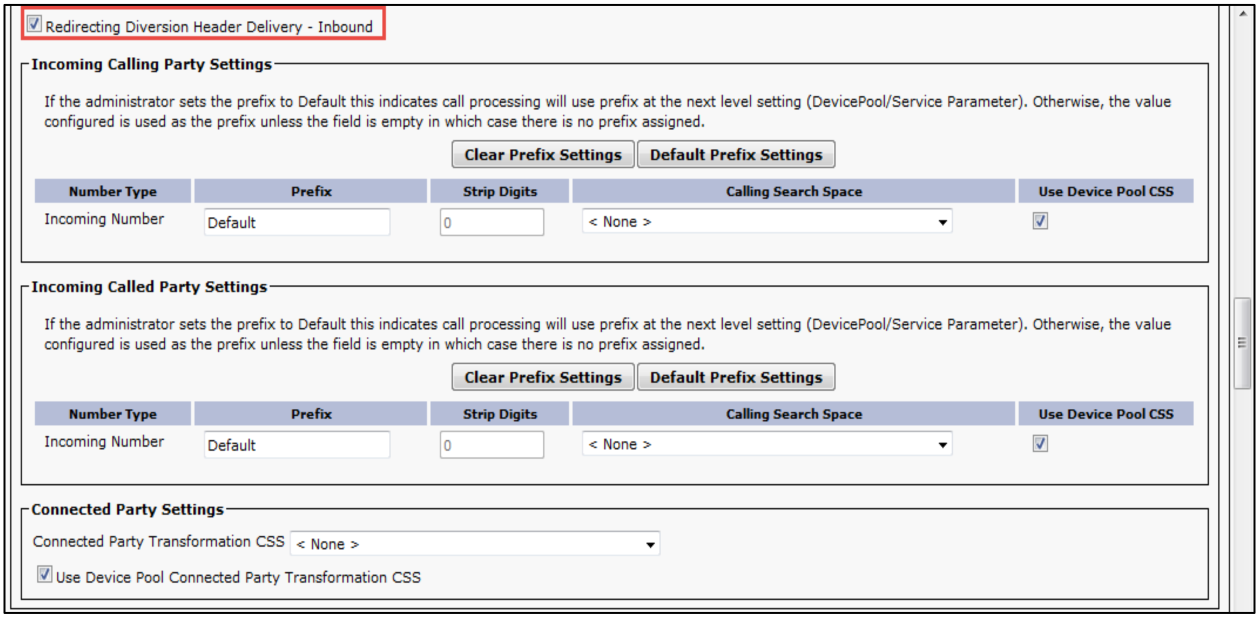


All other values are default.

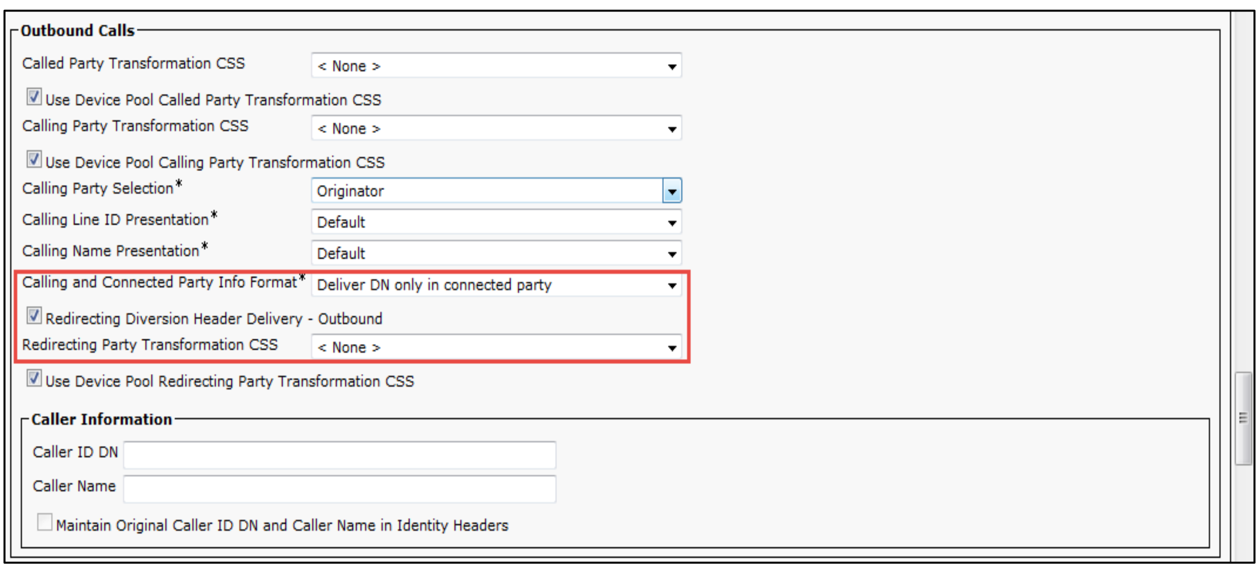


Check Redirecting Diversion Header Delivery – Inbound

Cisco Unified Communications Manager SIP Trunk to Avaya Configuration (Continued)



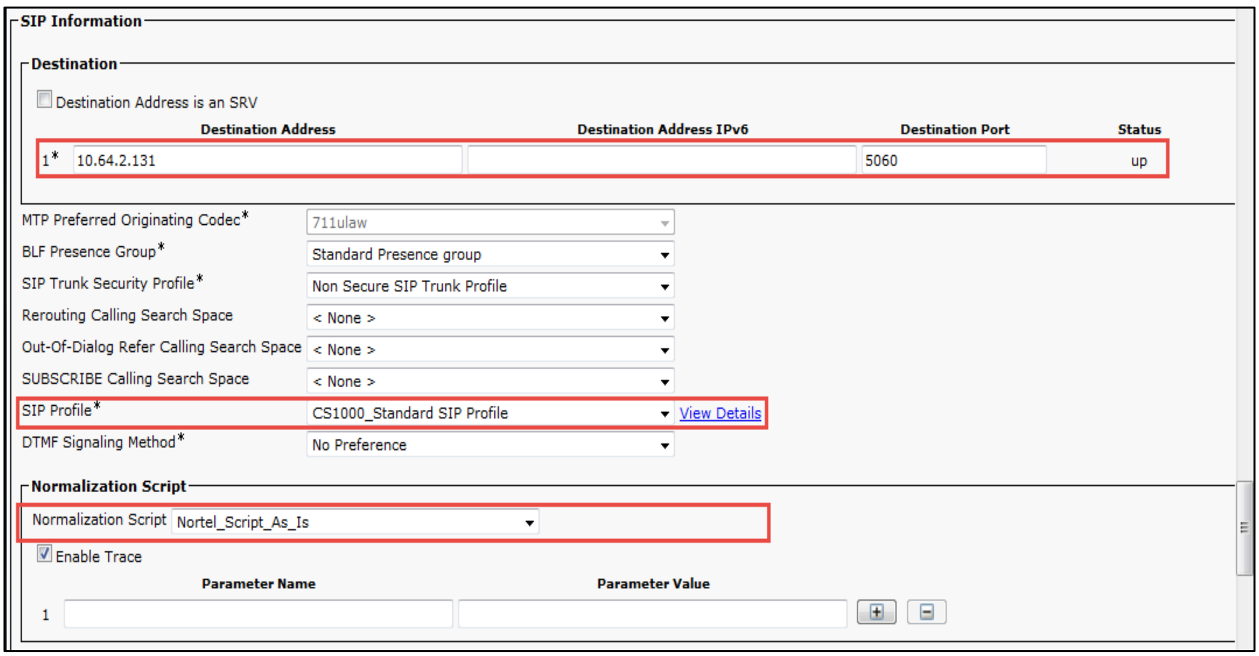
Set Calling and Connected Party Info Format* = Deliver URI and DN in connected party, if available.
Check Redirecting Diversion Header Delivery – Outbound.
All other values are default.

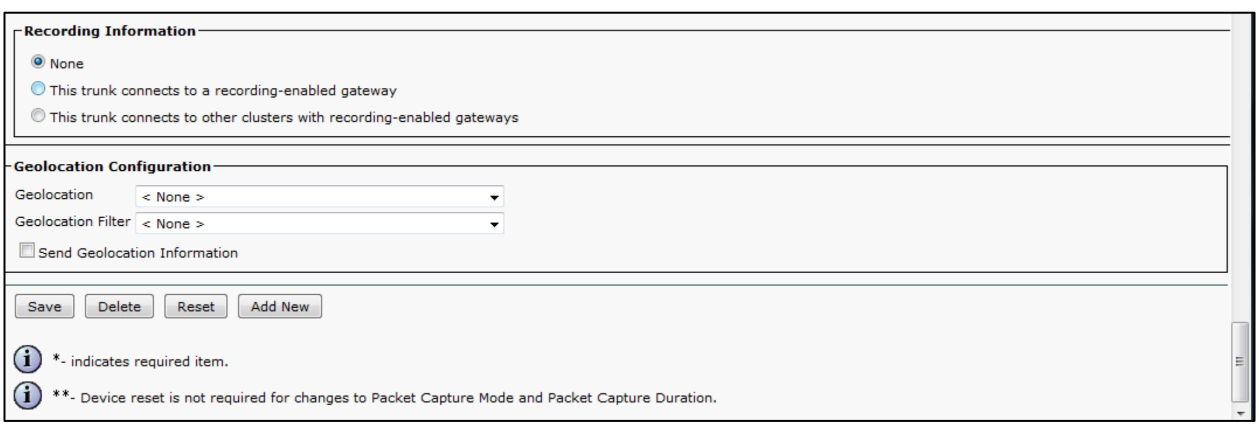


Cisco Unified Communications Manager SIP Trunk to Avaya Configuration (Continued)

Set Destination Address = 10.64.2.131. This is used in this example.
Set SIP Trunk Security Profile*= Non Secure SIP Trunk Profile.
Set SIP Profile*= Early Offer SIP Profile.
Set DTMF Signaling Method*= No Preference.
Set Normalization Script = nortel_Script_As_Is. The Cisco UCM-Software Script should be applied at SIP trunk toward Avaya PBX. This script normalizes the SIP messaging to/from the Avaya for UC Voice Mail center MWI, History-Info to Diversion Header conversion, Diversion Header to History- Info header conversion and privacy=user to privacy=id conversion.

All other values are default.

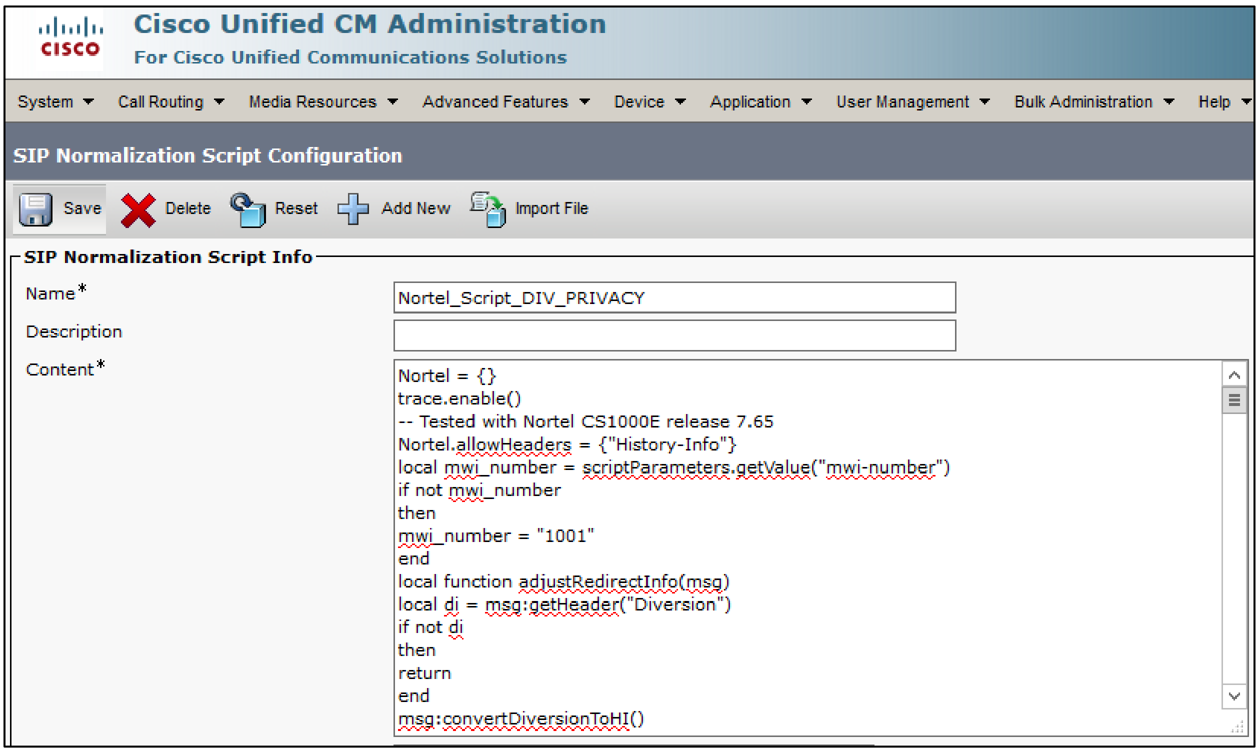


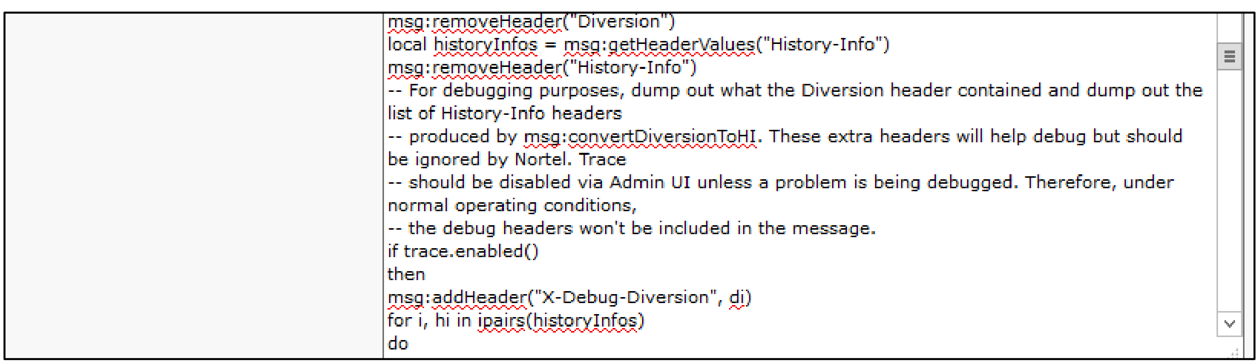


Cisco UCM SIP Trunk Normalization Script

Set Name*= Nortel_Script_DIV_PRIVACY. This is used for this example.
Set Description = This text is used to identify this SIP Normalization Script.
Set Content*= Please see full contents on next page.

All Other values are default





SIP Normalization script Configuration (Continued)

```
msg:addHeader("X-Debug-History-Info", hi)
end
end
-- Example:
-- Original Diversion header generated natively by CUCM might have been this:
-- Diversion: <sip:1000@10.10.10.100>;reason=unconditional;privacy=off;screen=yes
--
-- The call to convertDiversionToHI will produce these:
-- History-Info: <sip:1000@10.10.10.100:5060?Reason=sip;cause=302;
text="unconditional">;index=1
-- History-Info: <sip:3005@10.10.10.200:5060>;index=1.1
--
-- However, Nortel needs something that looks like this:
-- History-Info:<sip:1000@10.10.10.100?reason=sip%3Bcause%3D302%3Btext
%3D%22Moved%20Temporarily%22>;index=1
```

```
-- History-Info: <sip:3005@10.10.10.200>;index=2
--
-- This loop generates the additional History-Info header and uses the index value for the
first header generated by
-- convertDiversionToHI. Each header uses the index from the next. The last header uses the
last value plus one.
-- While processing each header, it also removes the port number from the URI and does any
necessary conversion of
-- to special characters to the escaped value for the embedded header.
for i, hi in ipairs(historyInfos)
do
local uri = string.match(hi, '<(.*)%?') or string.match(hi, "<(.*)>;index=.*") or ""
-- Strip out the port number.
uri = string.gsub(uri, "@(.*):%d+", "@%1")
-- Get the embedded header but without the ?Reason=sip part.
```

```
if embed_header
then
embed_header = string.gsub(embed_header, "unconditional", "Moved Temporarily")
embed_header = string.gsub(embed_header, ";", "%%3B")
embed_header = string.gsub(embed_header, "=", "%%3D")
embed_header = string.gsub(embed_header, "\"", "%%22")
embed_header = string.gsub(embed_header, " ", "%%20")
embed_header = string.format("?reason=sip%s", embed_header)
end
hi = string.format("<%s%s>;index=%s", uri, embed_header or "", i)
msg:addHeader("History-Info", hi)
end
end
-- Remove OPTIONS from outbound INVITE requests.
-- Convert Diversion to History-Info.
```

```
function Nortel.outbound_INVITE(msg)
msg:removeHeaderValue("Allow","OPTIONS")
adjustRedirectInfo(msg)
end
-- Remove OPTIONS from any outbound request
function Nortel.outbound_ANY(msg)
msg:removeHeaderValue("Allow","OPTIONS")
end
-- Remove OPTIONS from any ourbound response to any request
function Nortel_outbound_ANY_ANY(msg)
msg:removeHeaderValue("Allow","OPTIONS")
end
-- Modify the From header so that the userpart is numeric. CUCM will natively send
-- 'voicemail' as the userpart. Nortel does not handle that. This code changes
-- the user part to 1000 or the value of the configured script parameter: mwi-number.
```

SIP Normalization script Configuration (Continued)

```
function Nortel.outbound_NOTIFY(msg)
msg:removeHeaderValue("Allow", "OPTIONS")
local from = msg:getHeader("From")
if from
then
from = from:gsub("voicemail", mwi_number)
msg:modifyHeader("From", from)
msg:addHeaderUriParameter("From", "user", "phone")
end
end
-- Convert History-Info to Diversion for inbound invites. Also, remove the
-- phone-context userpart parameter and user=phone URI parameter if either
-- is present.
function Nortel.inbound_INVITE(msg)
msg:removeHeaderValue("Allow", "UPDATE")
```

```
local hist = msg:getHeader("History-Info")
if not hist
then
return
end
msg:convertHIToDiversion()
msg:removeHeader("History-Info")
local diversion = msg:getHeader("Diversion")
if diversion
then
-- This first regex will remove the phone-context userpart parameter if there
-- are other parameters after it but before the @.
diversion = diversion:gsub(";phone%-context=[^;]*;([^@]*)@", ";%1@")
-- This second regex will remove the phone-context userpart parameter if it
-- is immediately before the @.
```

```
diversion = diversion:gsub(";phone%-context=[^@]*@", "@")
-- Remove user=phone URI parameter.
diversion = diversion:gsub(";user=phone", "")
diversion = diversion:gsub(";reason=deflection", ";reason=no-answer")
-- Save the changes.
msg:modifyHeader("Diversion", diversion)
end
end
function Nortel.inbound_ANY_INVITE(msg)
msg:removeHeaderValue("Allow", "UPDATE")
end
function Nortel.outbound_ANY_INVITE(msg)
msg:removeHeaderValue("Allow", "OPTIONS")
end
```

```
function Nortel.inbound_200_INVITE(msg)
    -- Modify Privacy header value user to id
    local privacyValues = msg:getHeaderValues("Privacy")
    local paramName  = string.match(privacyValues[1], "user")
    if paramName == "user"
    then
      msg:modifyHeader("Privacy", "id")
    end

end
return Nortel
```

Note: The Cisco UCM-Software Script should be applied at SIP trunk toward Avaya PBX. This script normalizes the SIP messaging to/from the Avaya for UC Voice Mail center MWI, History-Info to Diversion Header conversion, Diversion Header to History-Info header conversion, Omitting Option and Update from Allow header.

Download the script "Normalization Script" at Cisco Downloads Home > Products > Unified Communications > Call Control > Cisco Unified Communications Manager (CallManager) > Cisco Unified Communications Manager Version 10.5 > SIP Normalization and Transparency Scripts:

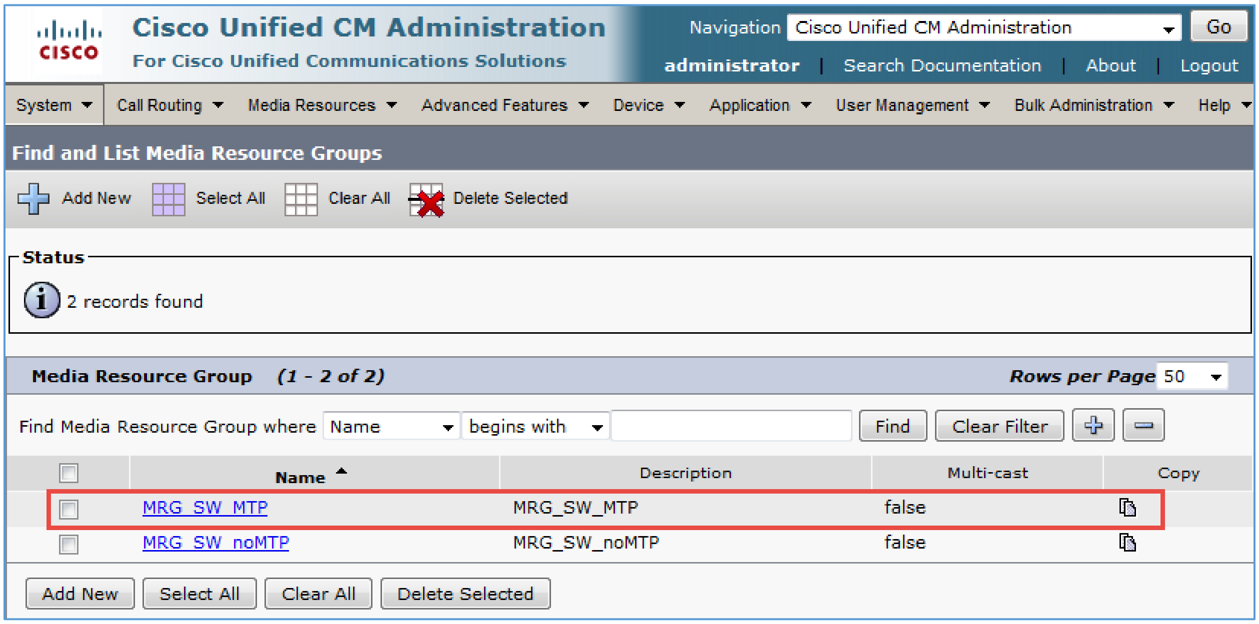
https://software.cisco.com/download/release.html?i=!y&mdfid=285963825&softwareid=284695022&release=Scripts&os=

Cisco UCM Service Parameter

Set Duplex Streaming Enabled* = True. See Note under capture for more info.

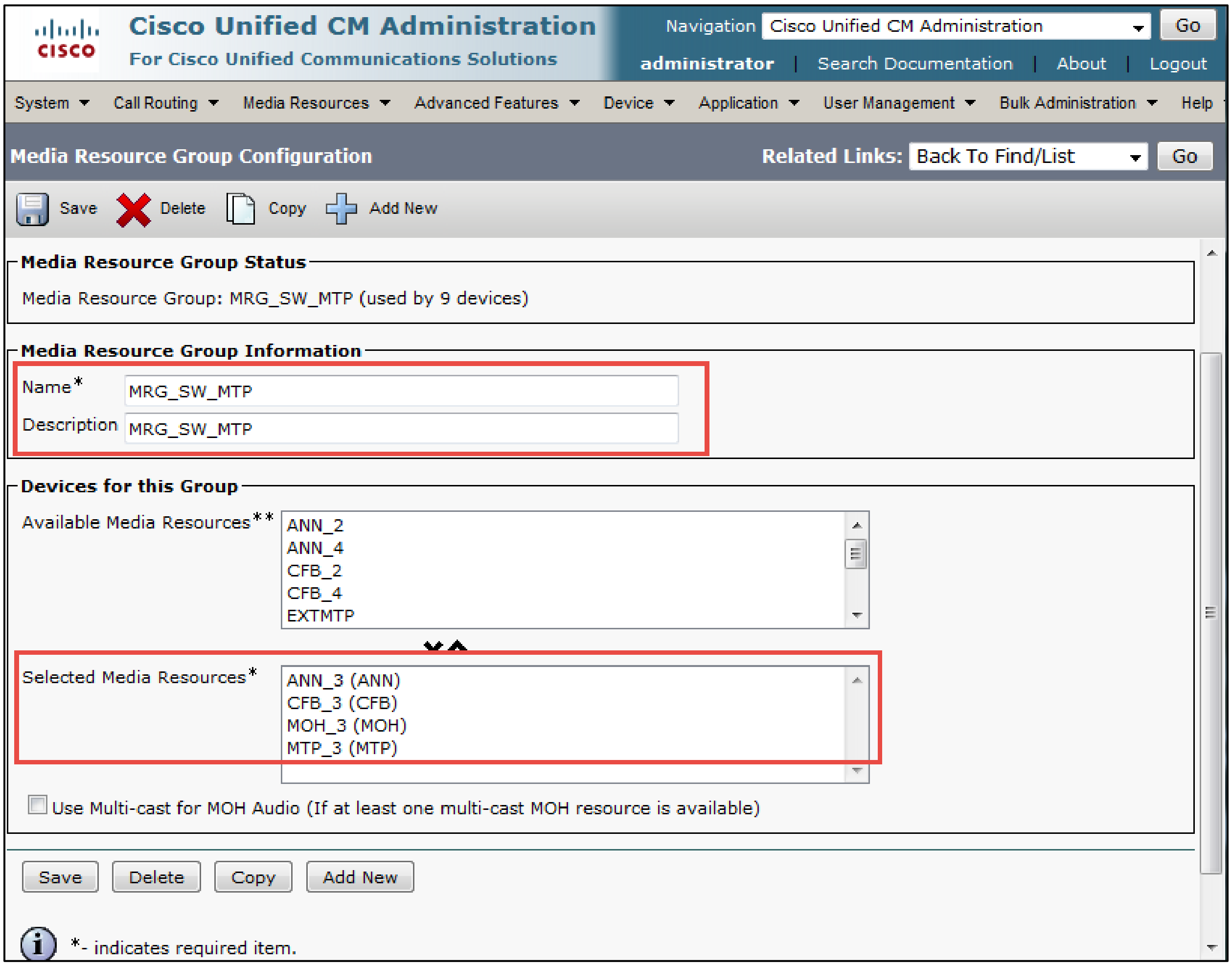
Note: Cisco Unified Communications Manager Service Parameter “Duplex Streaming Enabled” should be set to “True” in order for MoH and ring back to work properly during call transfers/conferences initiated by Cisco stations to Avaya IP endpoints.

Cisco UCM Media Resource Group

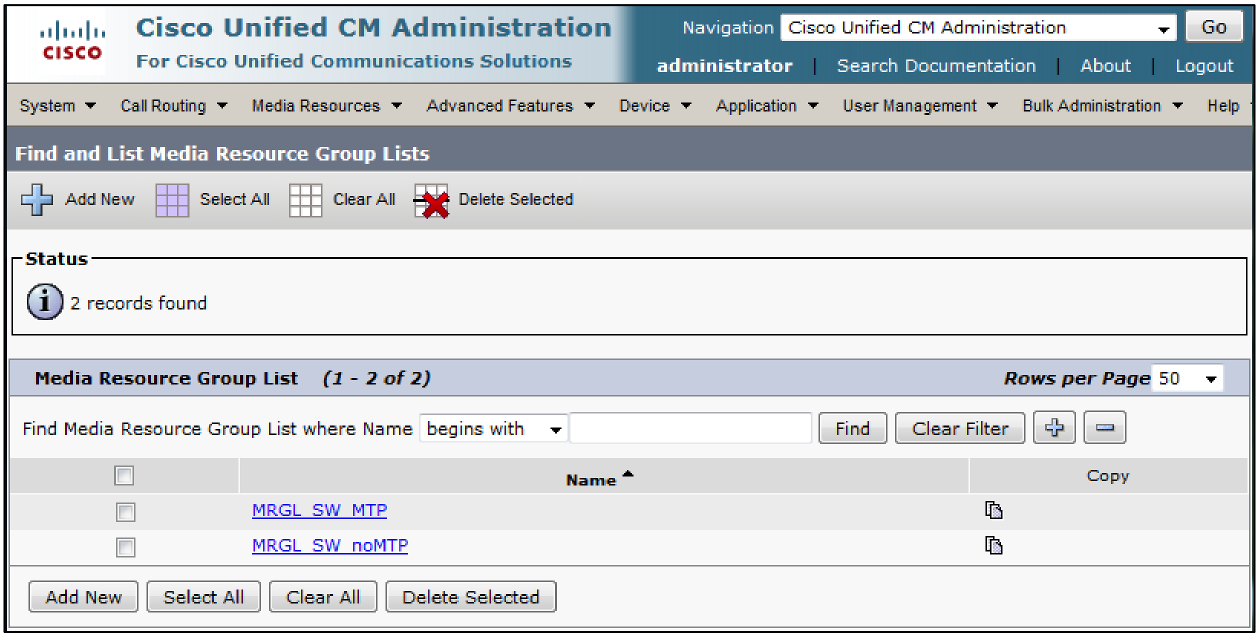


Media Resource Group MRG_MTP

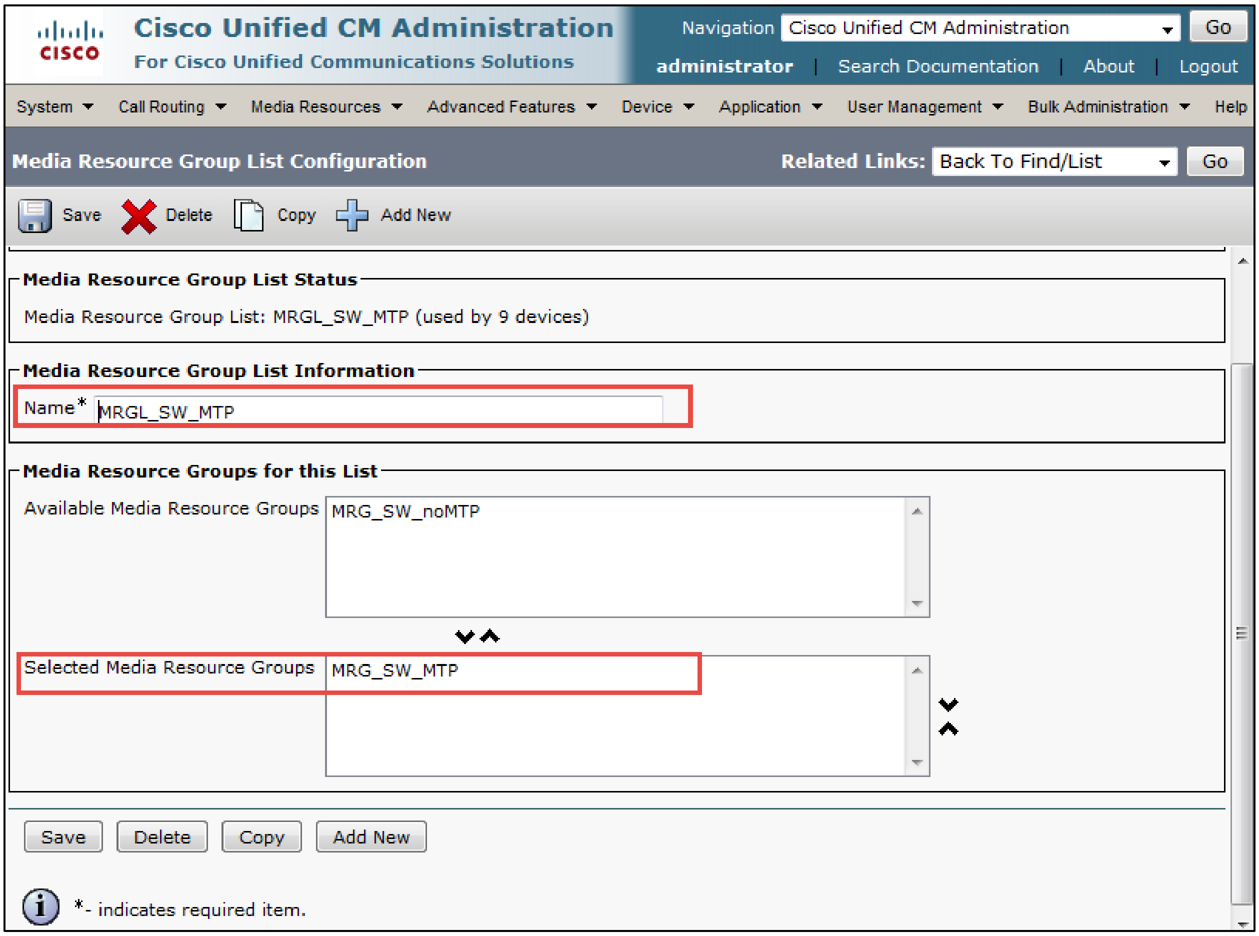
Set Name*= MRG_SW_MTP. This is used for this example.
Set Description = This text is used to identify this Media Resource Group.
Set all resources in the Selected Media Resources* Box.
All other values are default.



Cisco UCM Media Resource Group List



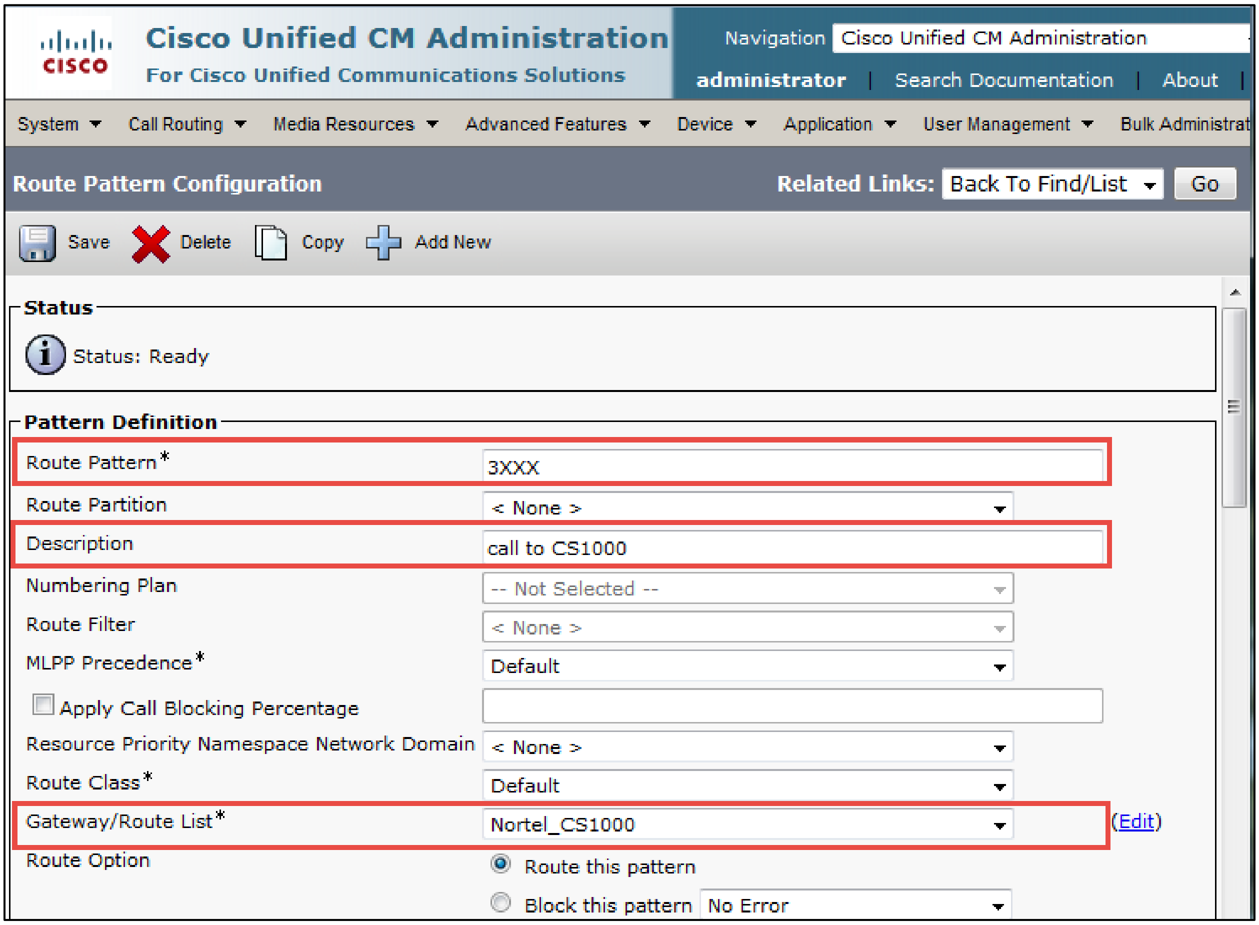
Set Name*= MRGL_SW_MTP This is used for this example.
Set Selected Media Resource Groups= MRG_SW_MTP



Note: This Media Resource Group List was added to provide early offer on the invite from Cisco to Avaya for SCCP phones.

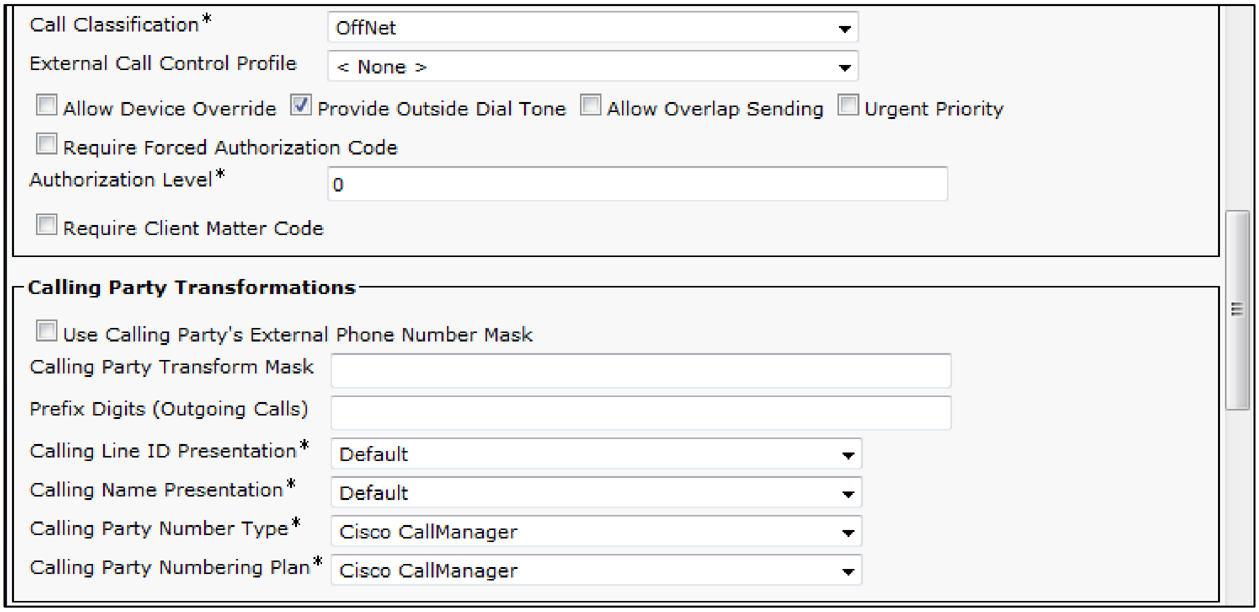
Cisco UCM Route Pattern to Avaya

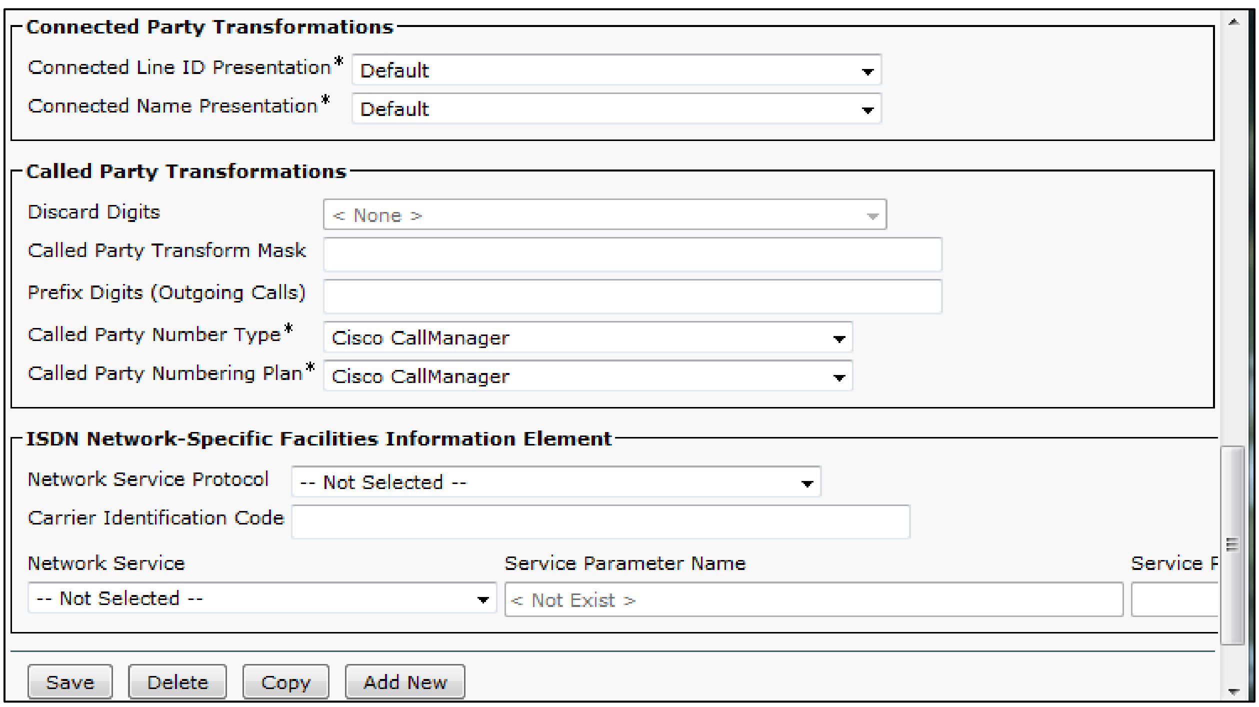
Set Route Pattern* =3XXX. This is used to route Avaya in this example.
Set Description = this text is used to identify this Route Pattern.
Set Gateway/Route List* = Avaya_CS1000. This is used for this example.
All other values are default.

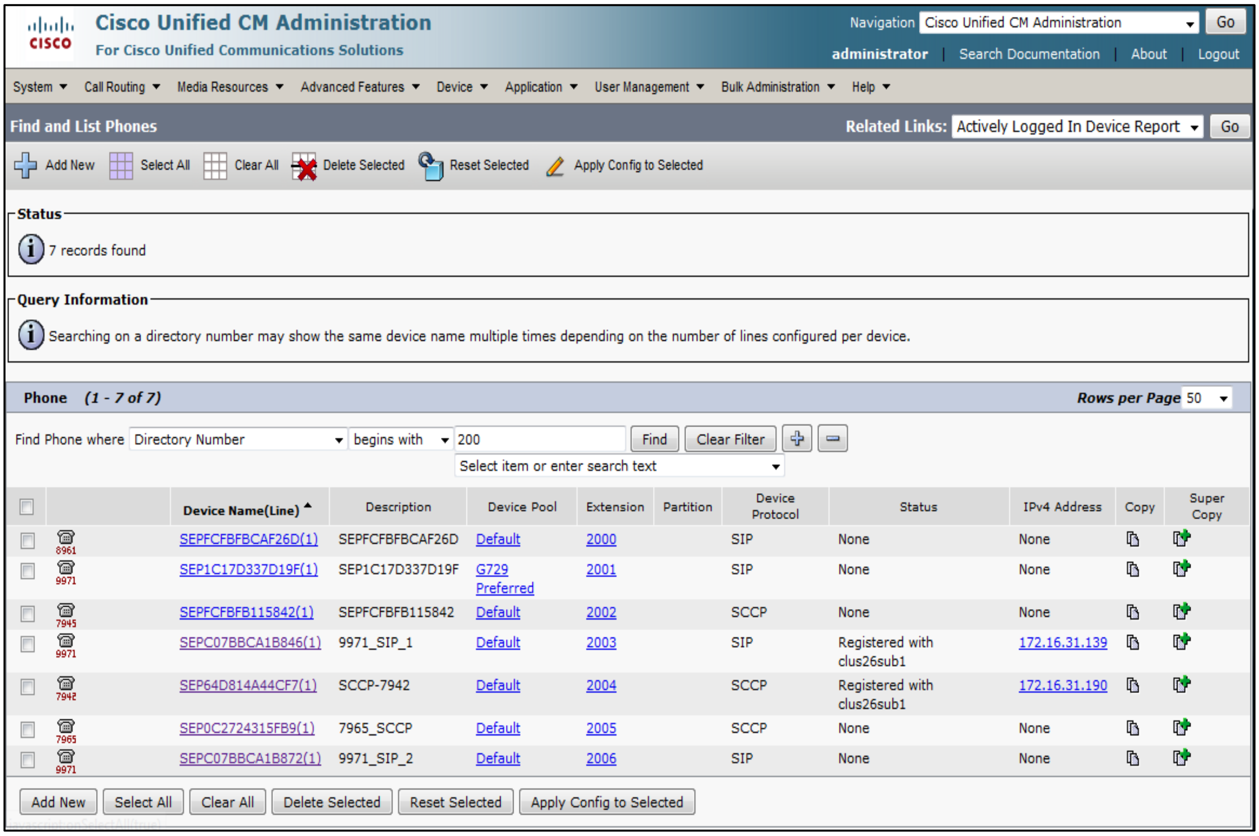


Route Patter Configuration for 3xxx (Continued)

All other values are default.

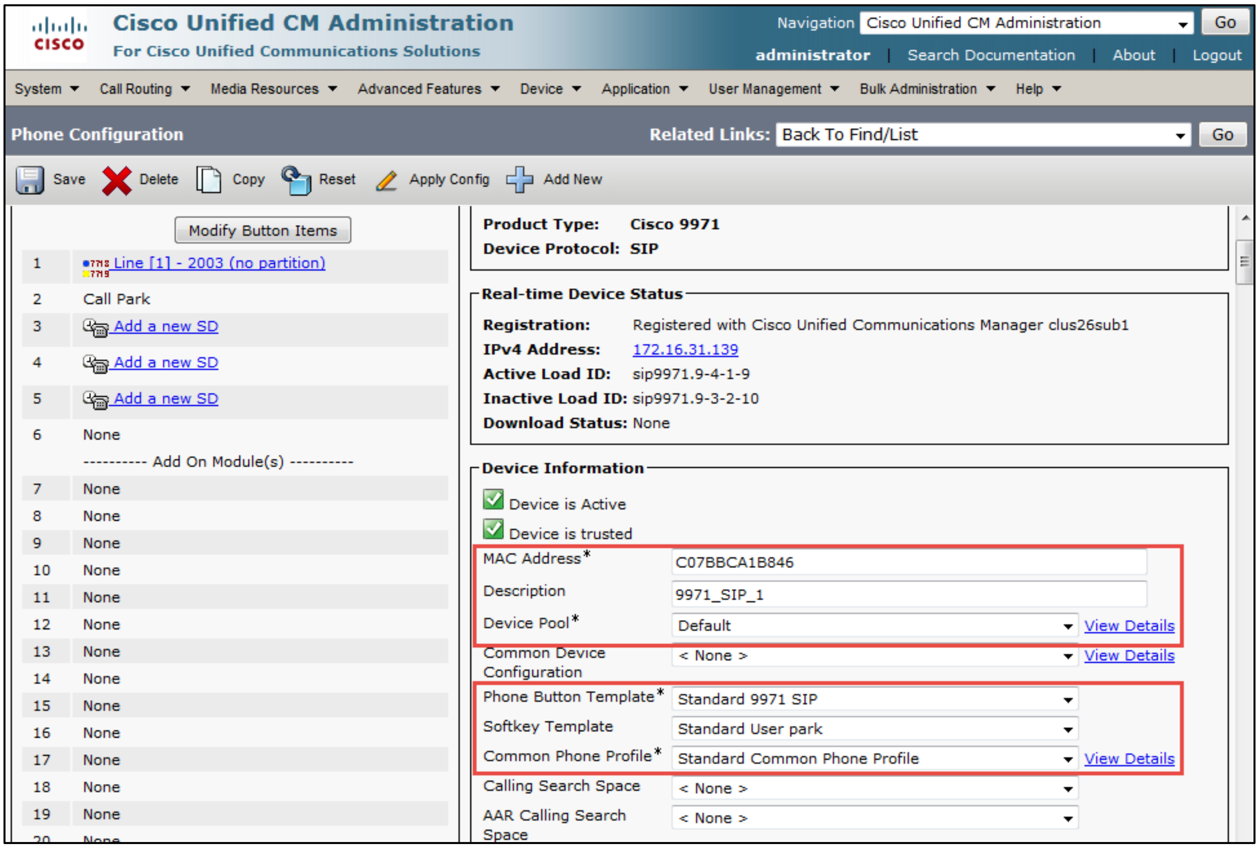






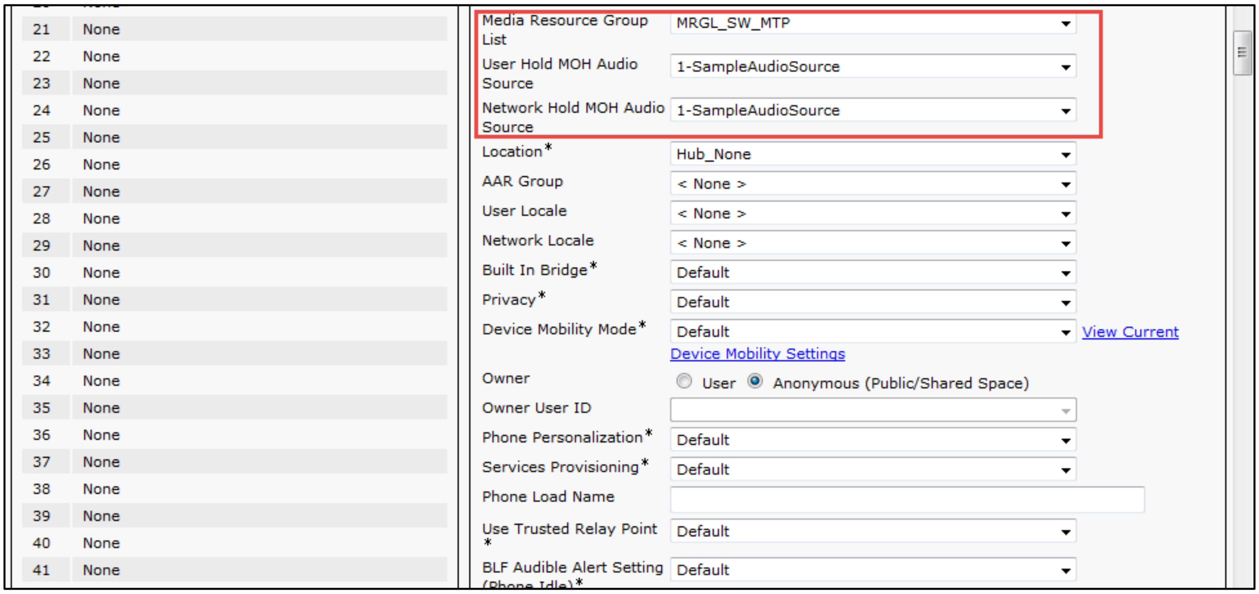
Set MAC Address* = C07BBCA1B846. This is used in this example.
Set Description = 9971_SIP_1.This text is used to identify this Phone.
Set Device Pool*= Default. This is used in this example.
Set Phone Button Template*= Standard 9971 SIP. This is used in this example.
All other values are default.

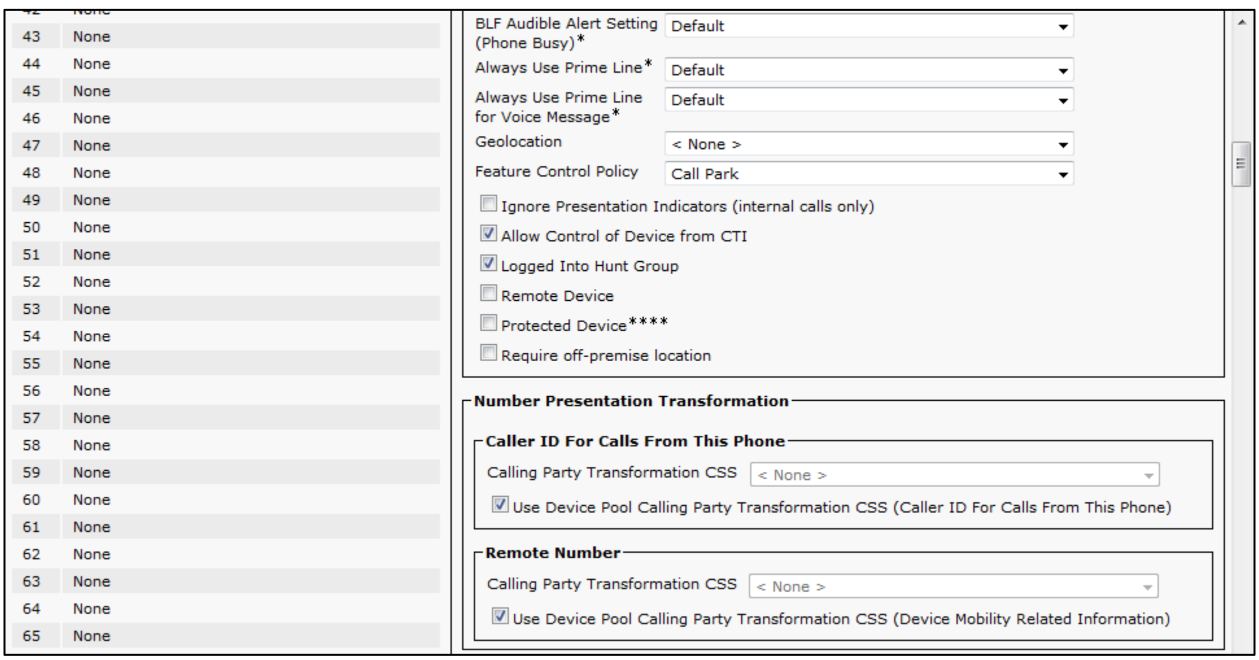
Cisco Unified Communications Manager SIP Phone Ext. 5000 Device Level Configuration (Continued)



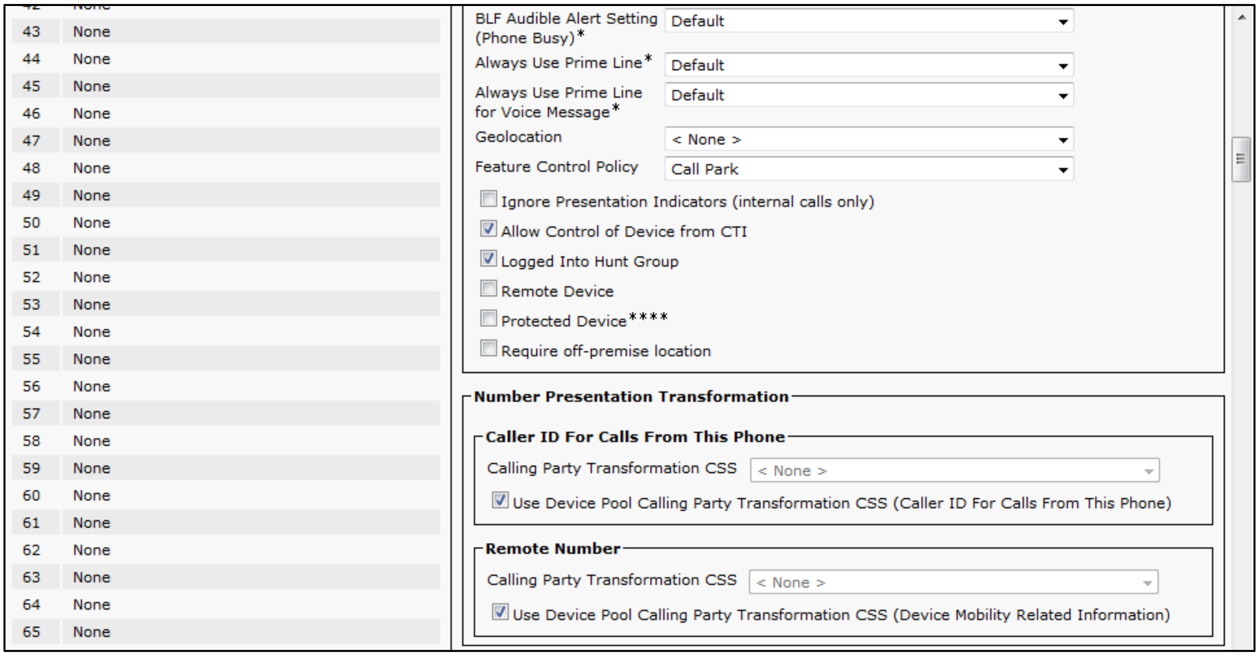
Cisco Unified Communications Manager SIP Phone Ext. 2003 Device Level Configuration (Continued)

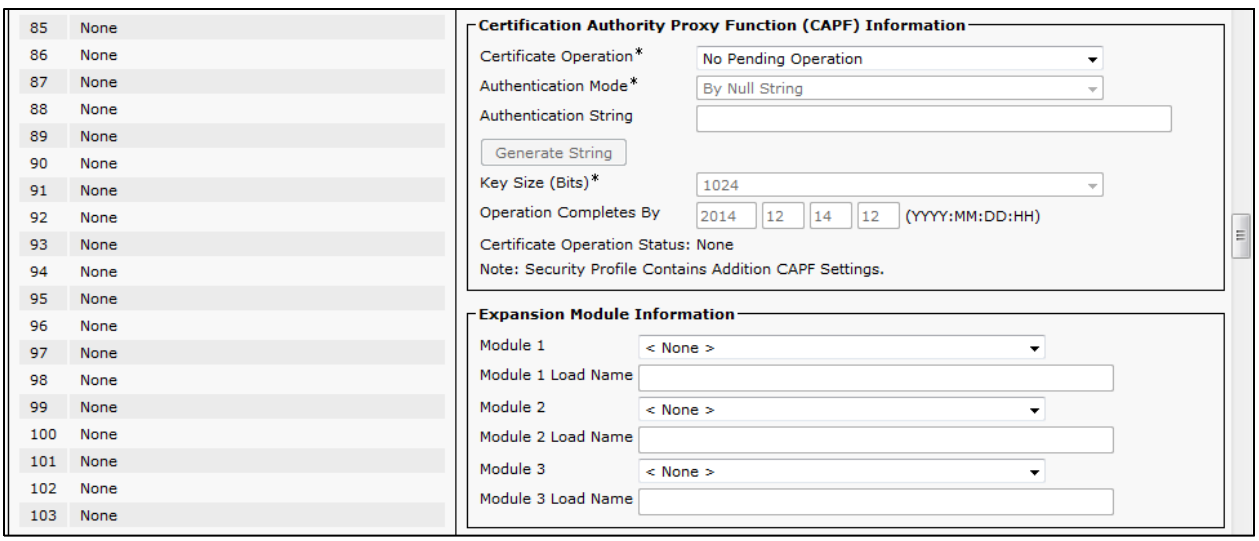
All other values are default.
Set Media Resource Group List = MRGL_SW_MTP. This is used in this example.
Set User Hold MOH Audio Source = 1-SampleAudioSource.
Set Network Hold MOH Audio Source = 1-SampleAudioSource.





Cisco Unified Communications Manager SIP Phone Ext. 2003 Device Level Configuration (Continued)





Cisco Unified Communications Manager SIP Phone Ext. 2003 Device Level Configuration (Continued)

104	None
105	None
106	None
107	None
108	None
109	None
110	None
111	None
112	None
113	None
114	None
	---------- Unassigned Associated Items ----------
115	Add a new SD
116	Line [2] - Add a new DN
117	All Calls
118	Add a new BLF Directed Call Park
119	Call Pickup
120	CallBack

External Data Locations Information (Leave blank to use default)

Information	
Directory	
Messages	
Services	
Authentication Server	
Proxy Server	
Idle	
Idle Timer (seconds)	
Secure Authentication URL	
Secure Directory URL	
Secure Idle URL	
Secure Information URL	
Secure Messages URL	
Secure Services URL	

121	Group Call Pickup
122	Hunt Group Logout
123	Intercom [1] - Add a new Intercom
124	Malicious Call Identification
125	Meet Me Conference
126	Mobility
127	Other Pickup
128	Quality Reporting Tool
129	Redial
130	Add a new SURL
131	Add a new BLF SD
132	Answer Oldest
133	Do Not Disturb
134	Services
135	Record
136	Alerting Calls
137	Queue Status
138	Privacy
139	None

Extension Information

☐ Enable Extension Mobility

Log Out Profile	-- Use Current Device Settings --
Log in Time	< None >
Log out Time	< None >

MLPP and Confidential Access Level Information

MLPP Domain	< None >
MLPP Indication*	Default
MLPP Preemption*	Default
Confidential Access Mode	< None >
Confidential Access Level	< None >

Do Not Disturb

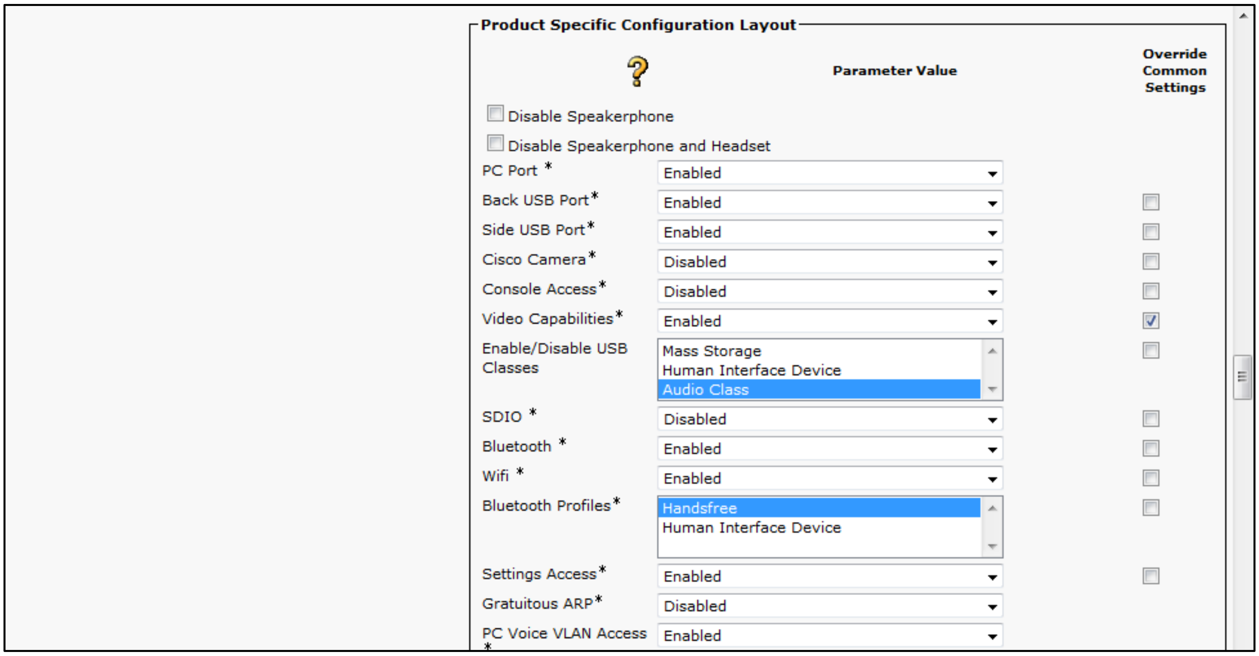
☐ Do Not Disturb

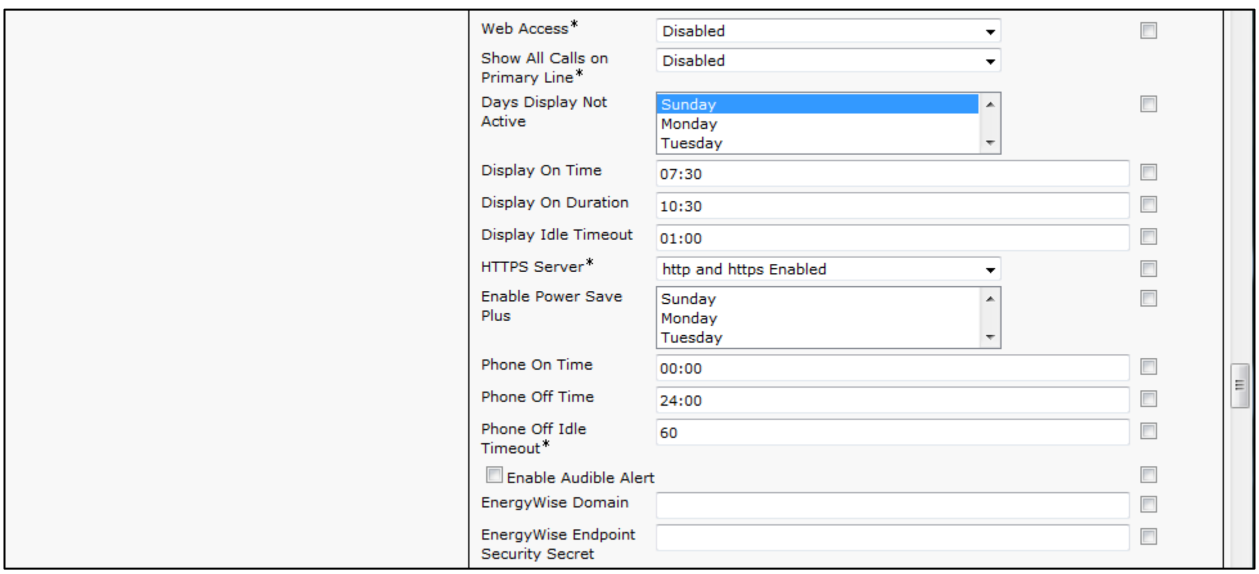
DND Option*	Use Common Phone Profile Setting
DND Incoming Call Alert	< None >

Secure Shell Information

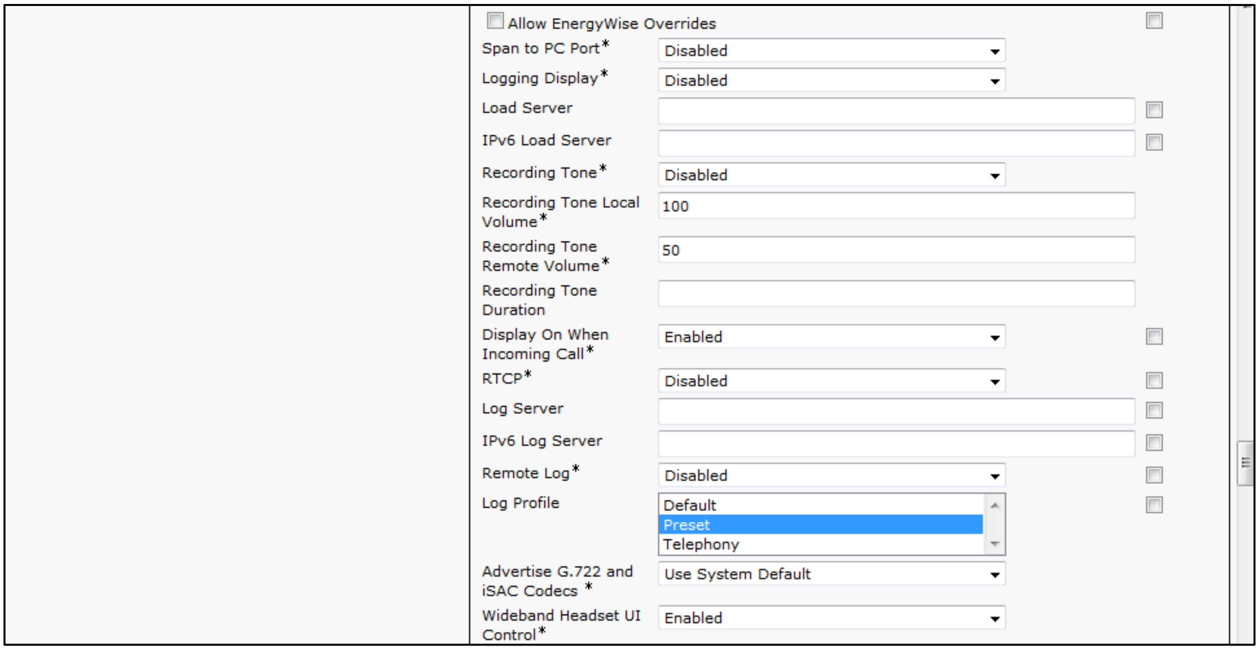
Secure Shell User	
Secure Shell Password	

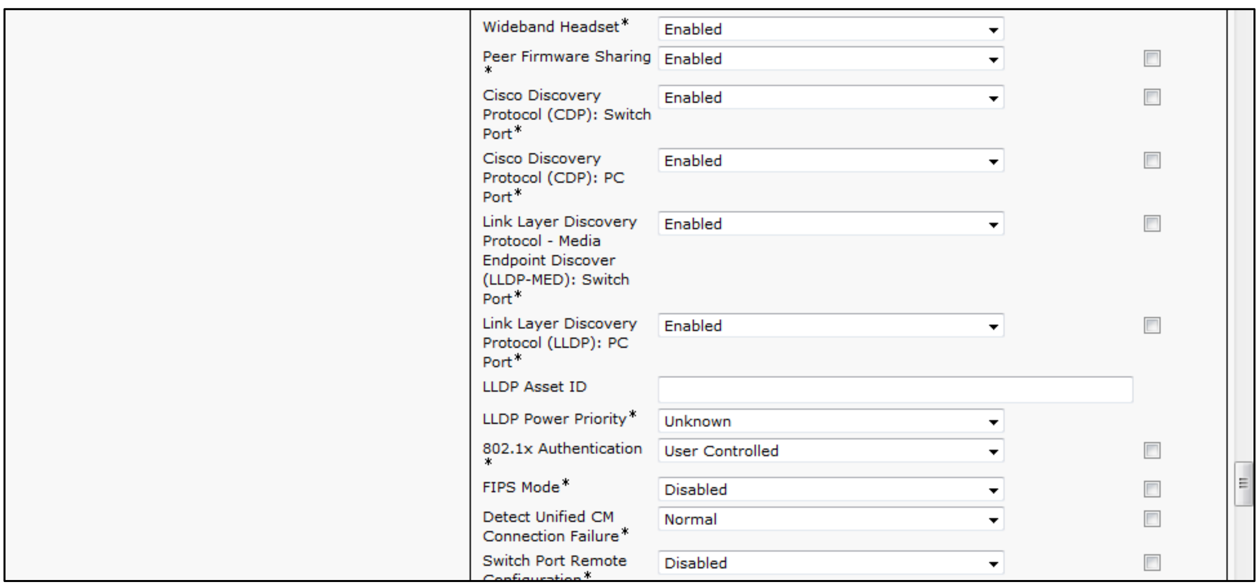
Cisco Unified Communications Manager SIP Phone Ext. 2003 Device Level Configuration (Continued)



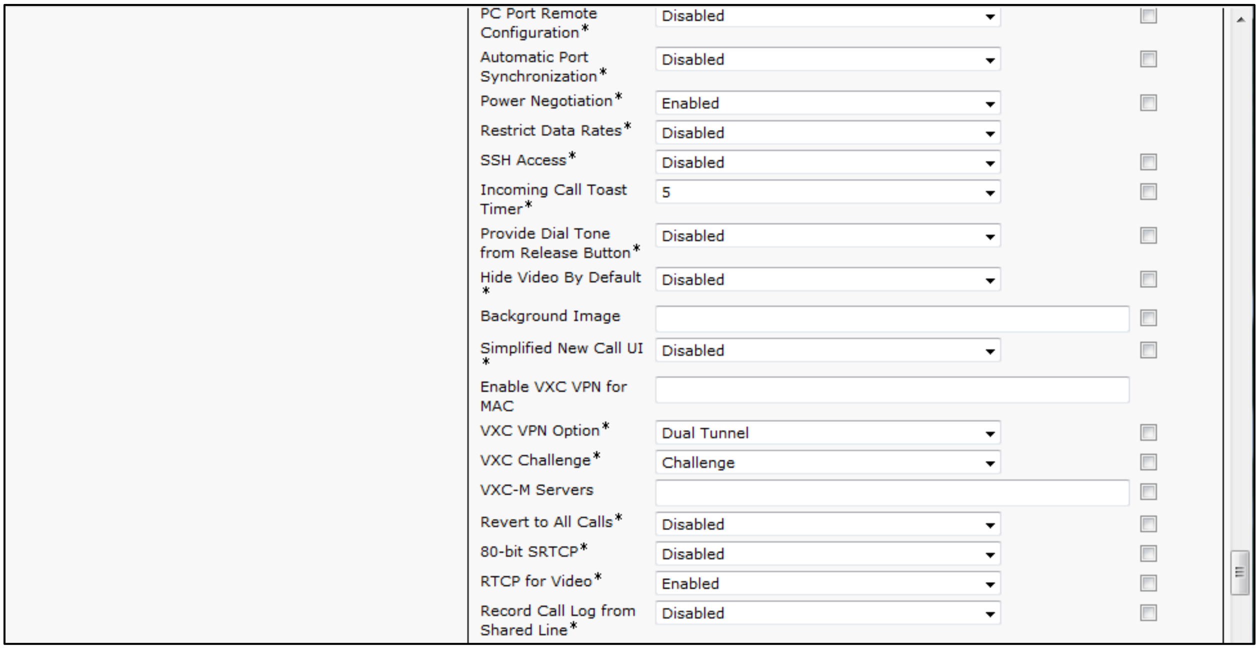


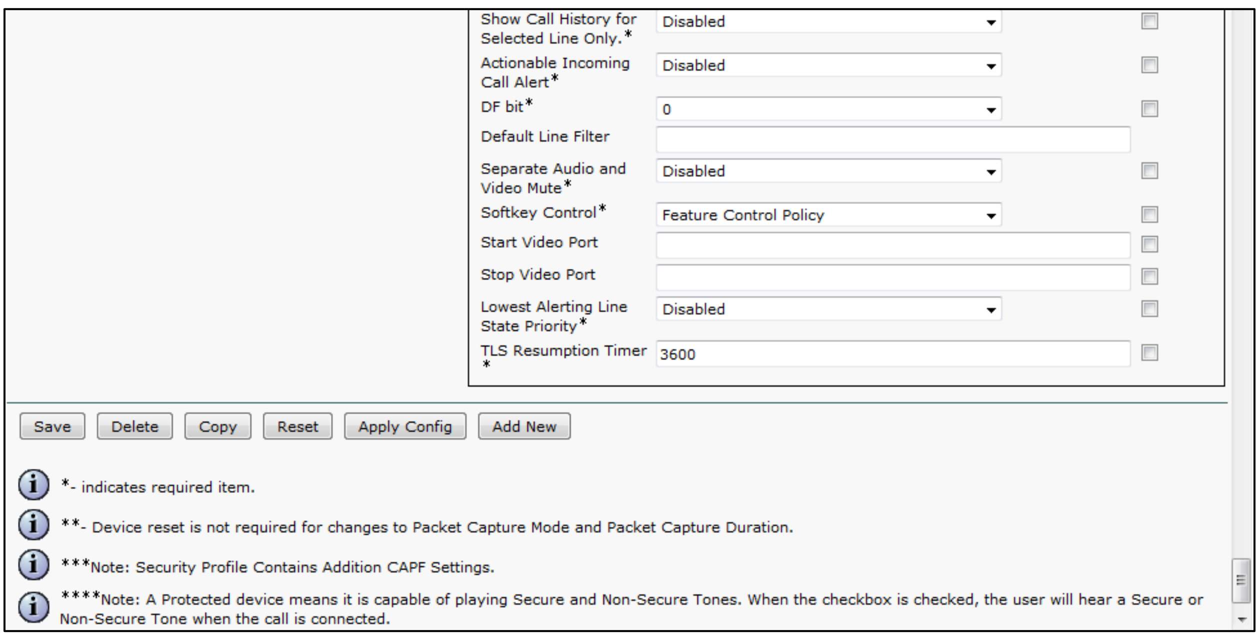
Cisco Unified Communications Manager SIP Phone Ext. 2003 Device Level Configuration (Continued)





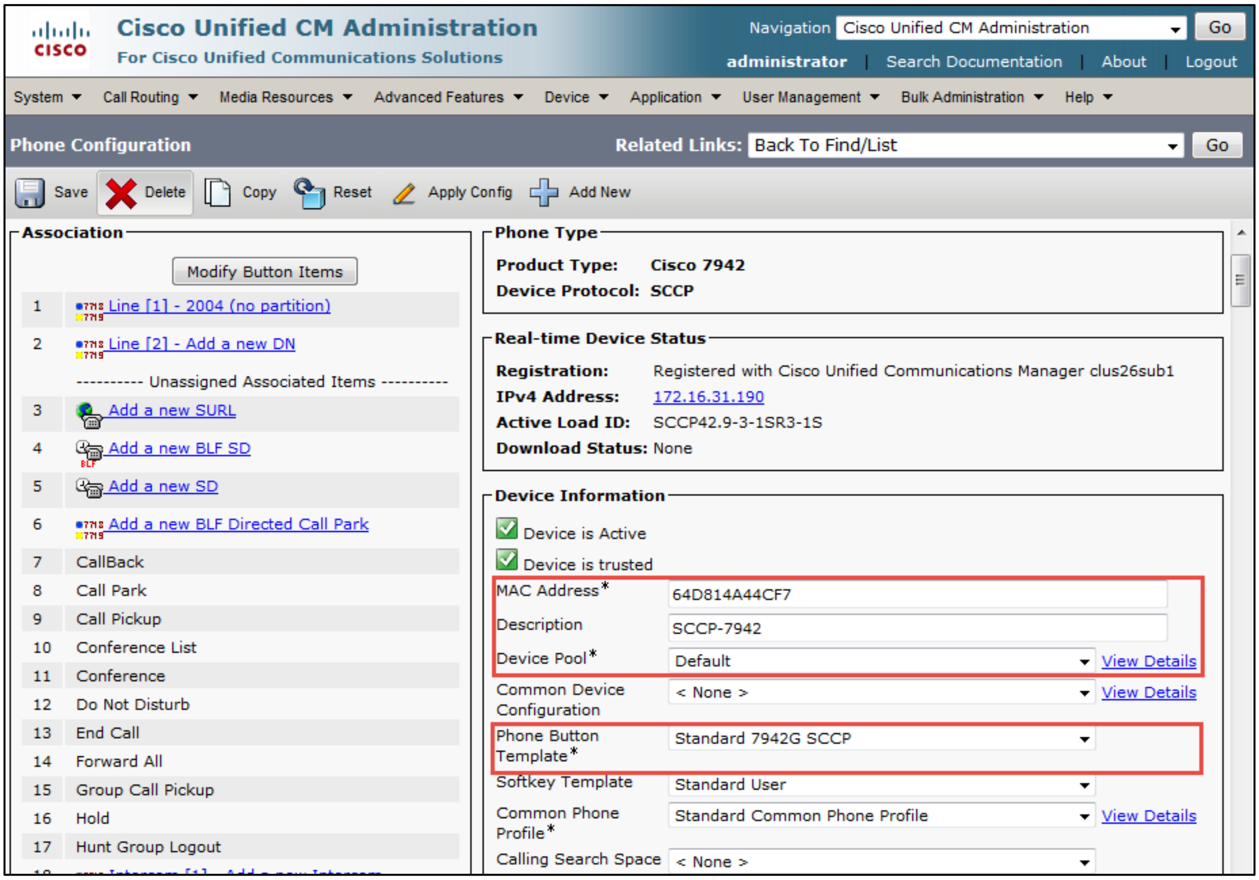
Cisco Unified Communications Manager SIP Phone Ext. 2003 Device Level Configuration (Continued)



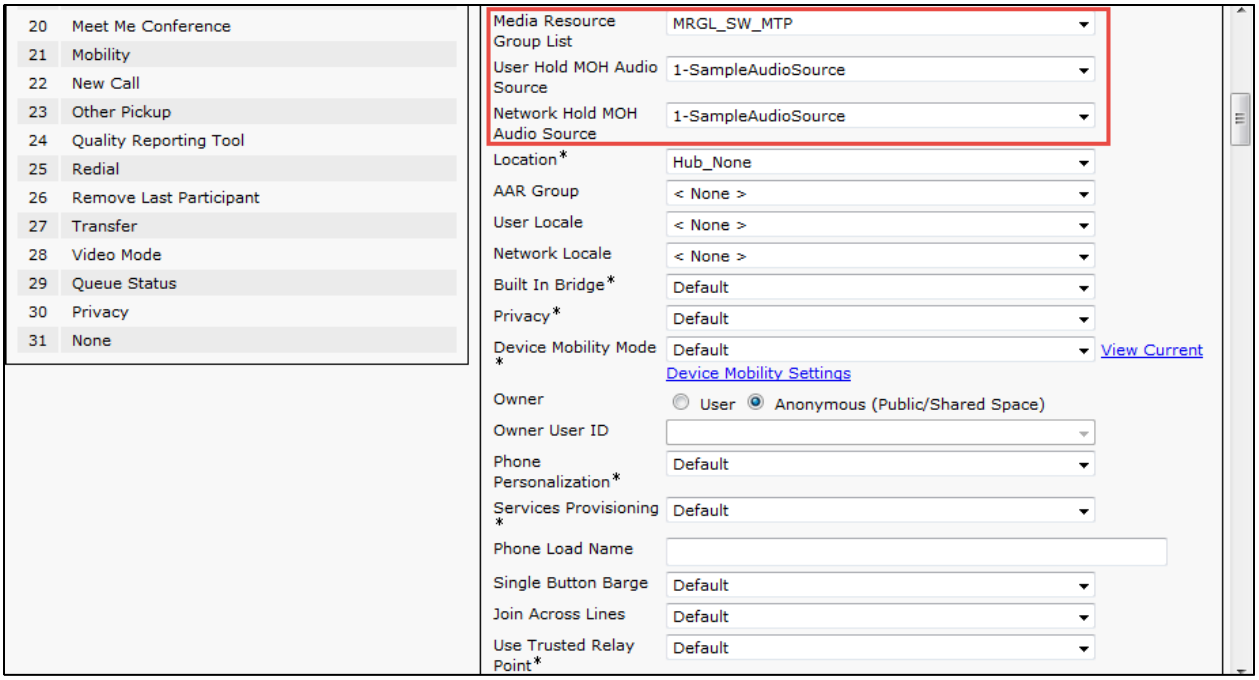


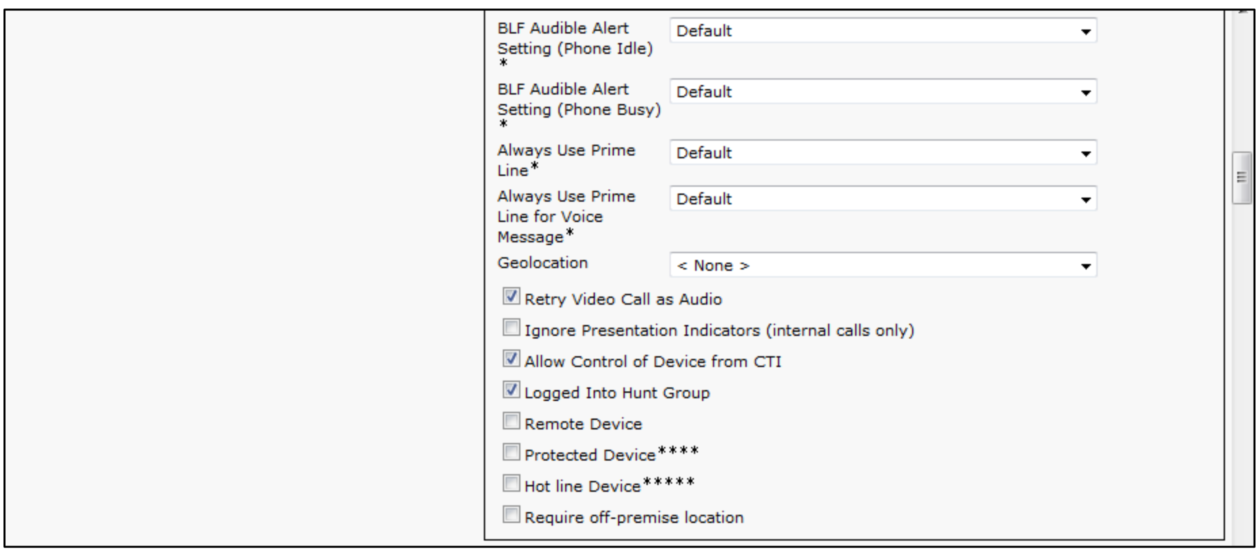
Cisco UCM SCCP Phone Ext. 2004 Device Level Configuration

Set MAC Address* = 64D814A44CF7. This is used in this example.
Set Description = SCCP-7942.This text is used to identify this Phone
Set Device Pool*= Default. This is used in this example.
Set Phone Button Template*= Standard 7942G SCCP. This is used in this example
Set Media Resource Group List = MRGL_SW_MTP. This is used in this example.
Set User Hold MOH Audio Source = 1-SampleAudioSource.
Set Network Hold MOH Audio Source = 1-SampleAudioSource.
All other values are default.

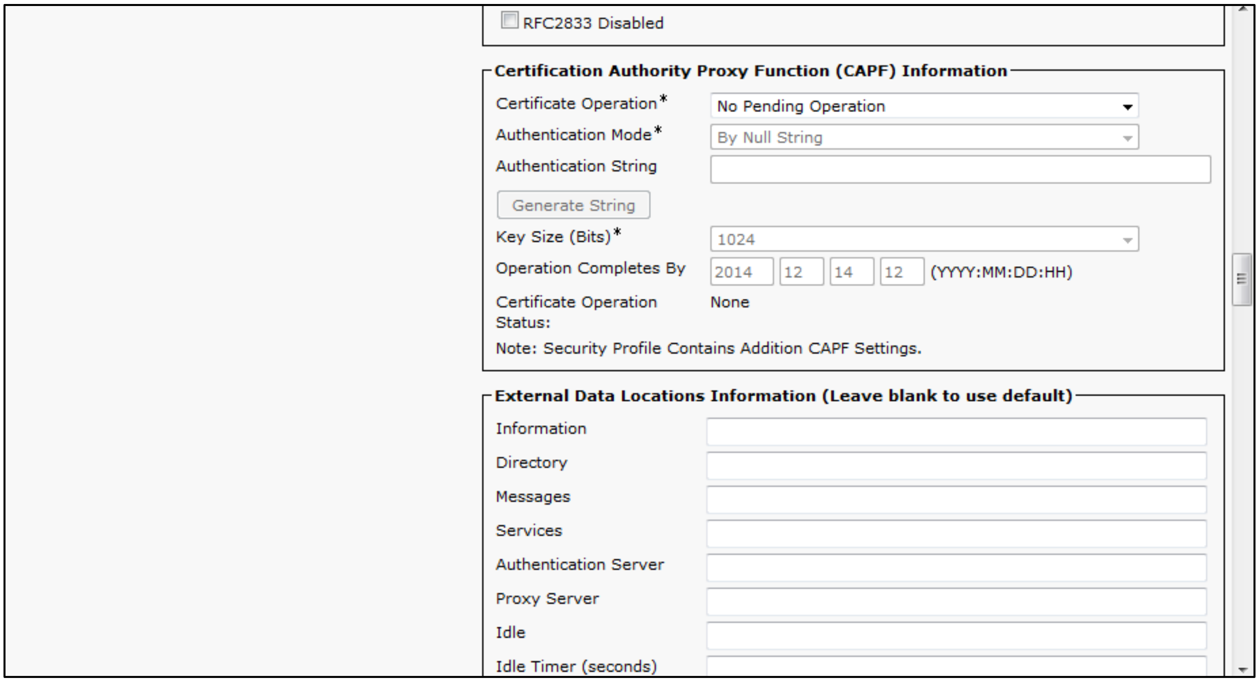


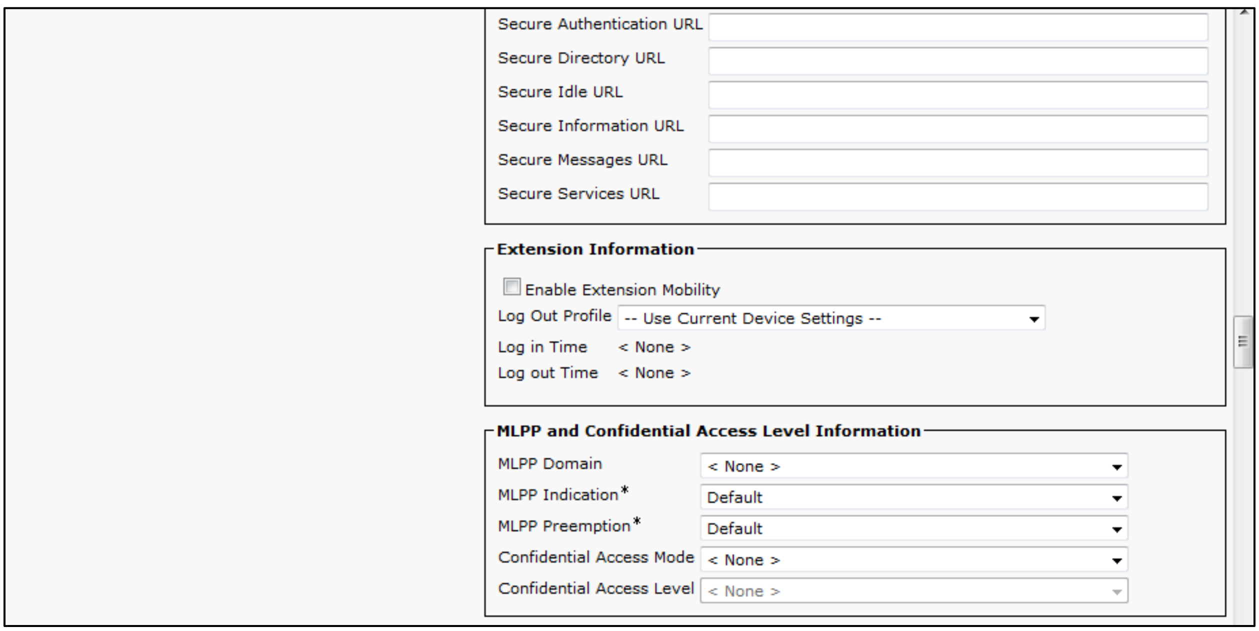
Cisco Unified Communications Manager SCCP Phone Ext. 2004 Device Level Configuration (Continued)



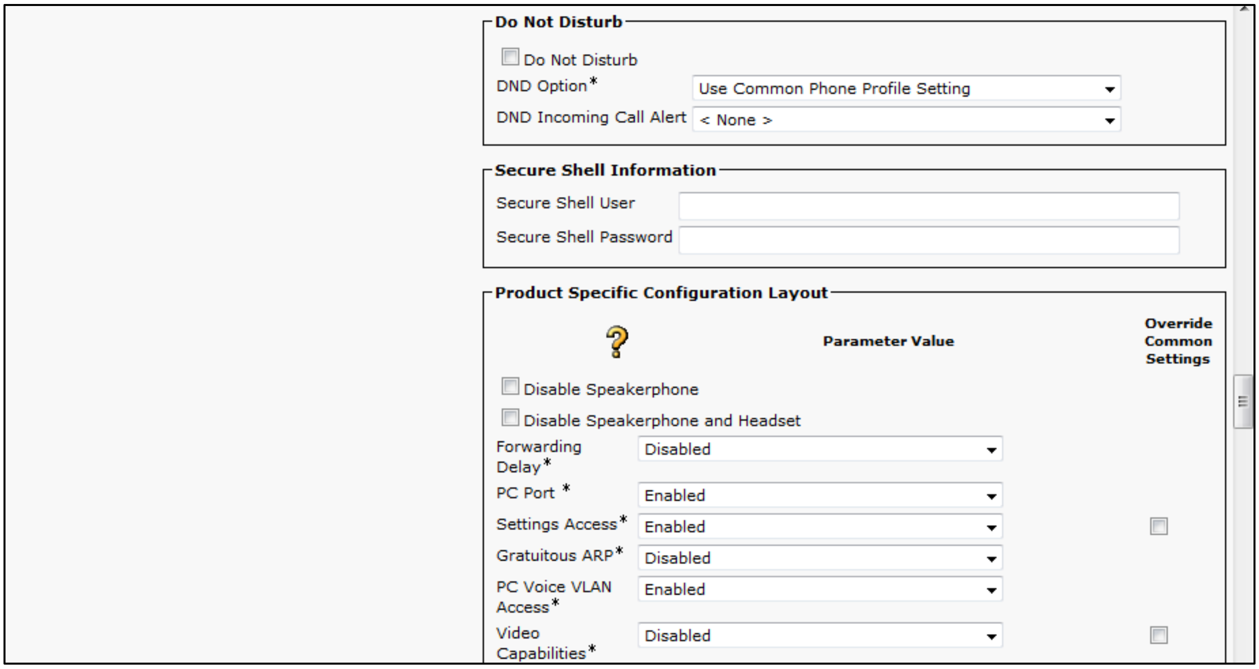


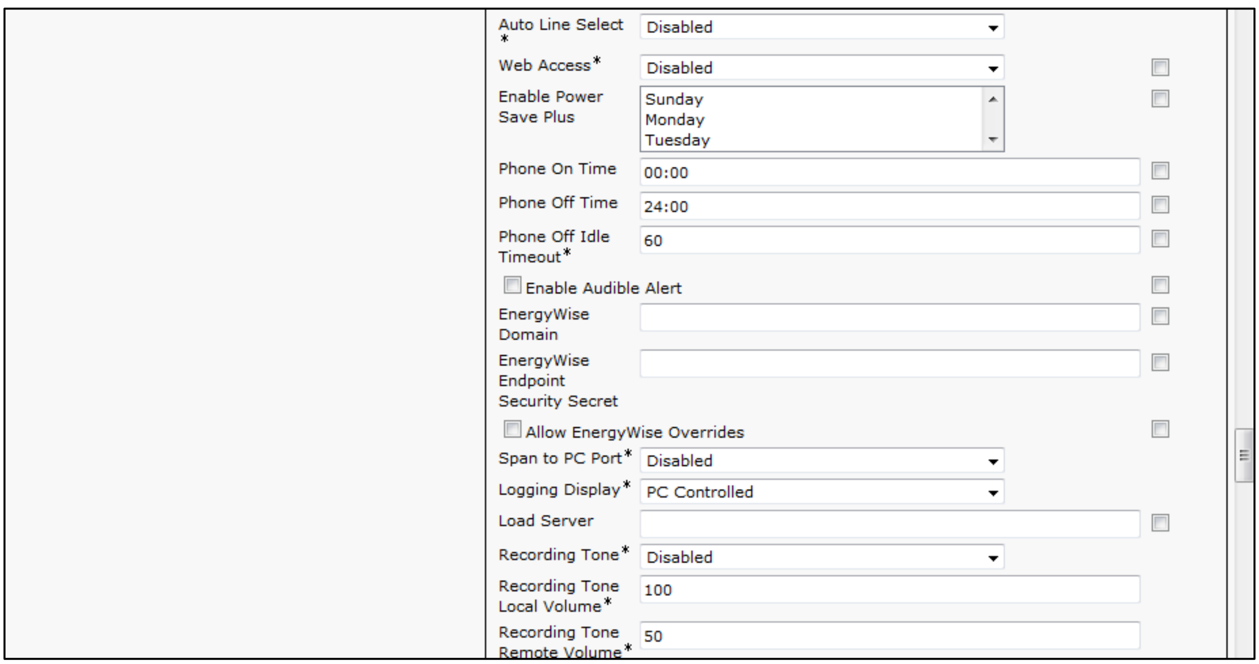
Cisco Unified Communications Manager SCCP Phone Ext. 2004 Device Level Configuration (Continued)



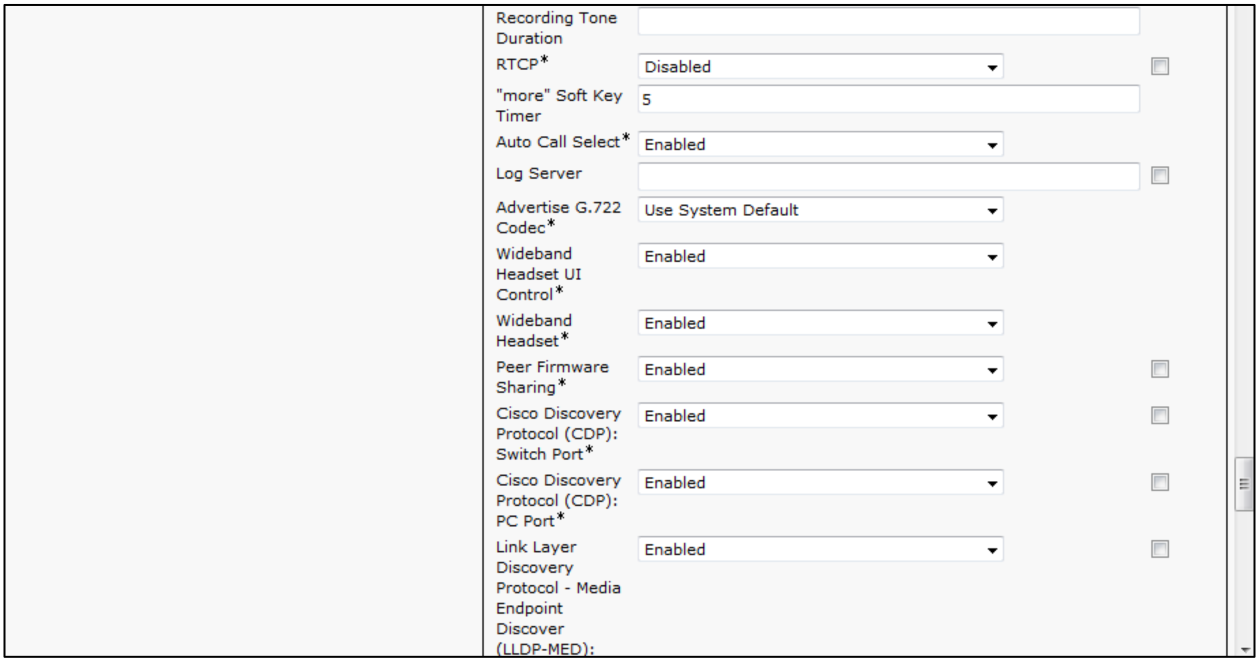


Cisco Unified Communications Manager SCCP Phone Ext. 2004 Device Level Configuration (Continued)

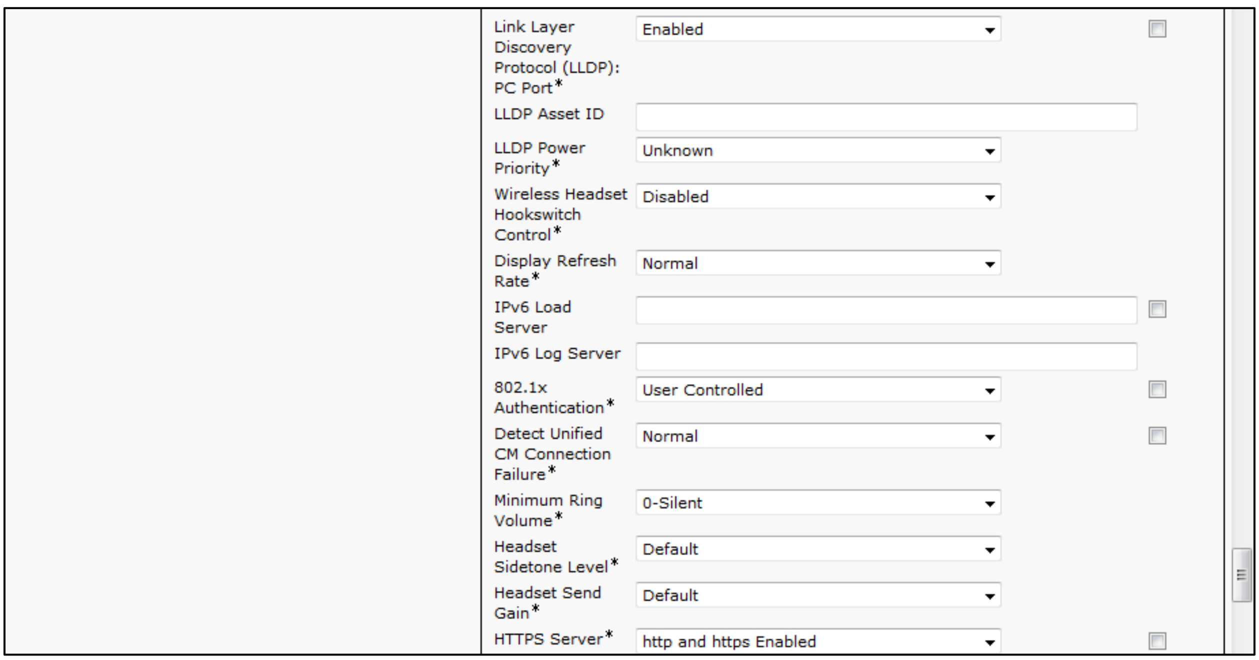




Cisco Unified Communications Manager SCCP Phone Ext. 2004 Device Level Configuration (Continued)

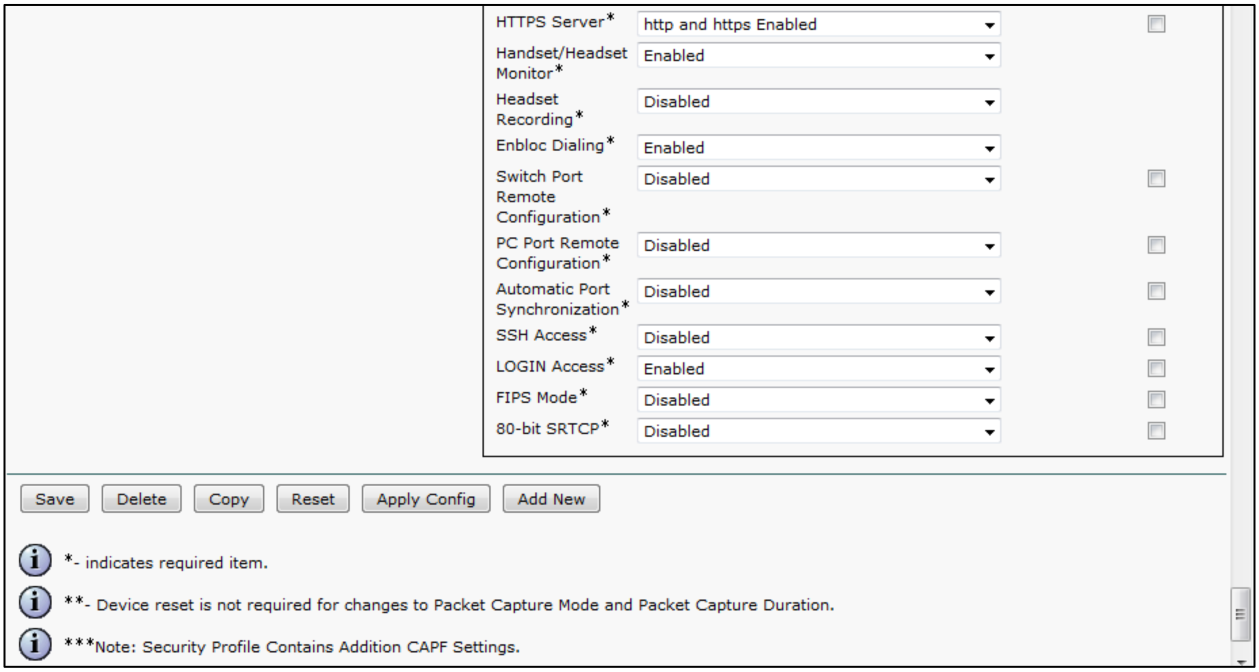


Recording Tone Duration	
RTCP*	Disabled
"more" Soft Key Timer	5
Auto Call Select*	Enabled
Log Server	
Advertise G.722 Codec*	Use System Default
Wideband Headset UI Control*	Enabled
Wideband Headset*	Enabled
Peer Firmware Sharing*	Enabled
Cisco Discovery Protocol (CDP): Switch Port*	Enabled
Cisco Discovery Protocol (CDP): PC Port*	Enabled
Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED):	Enabled



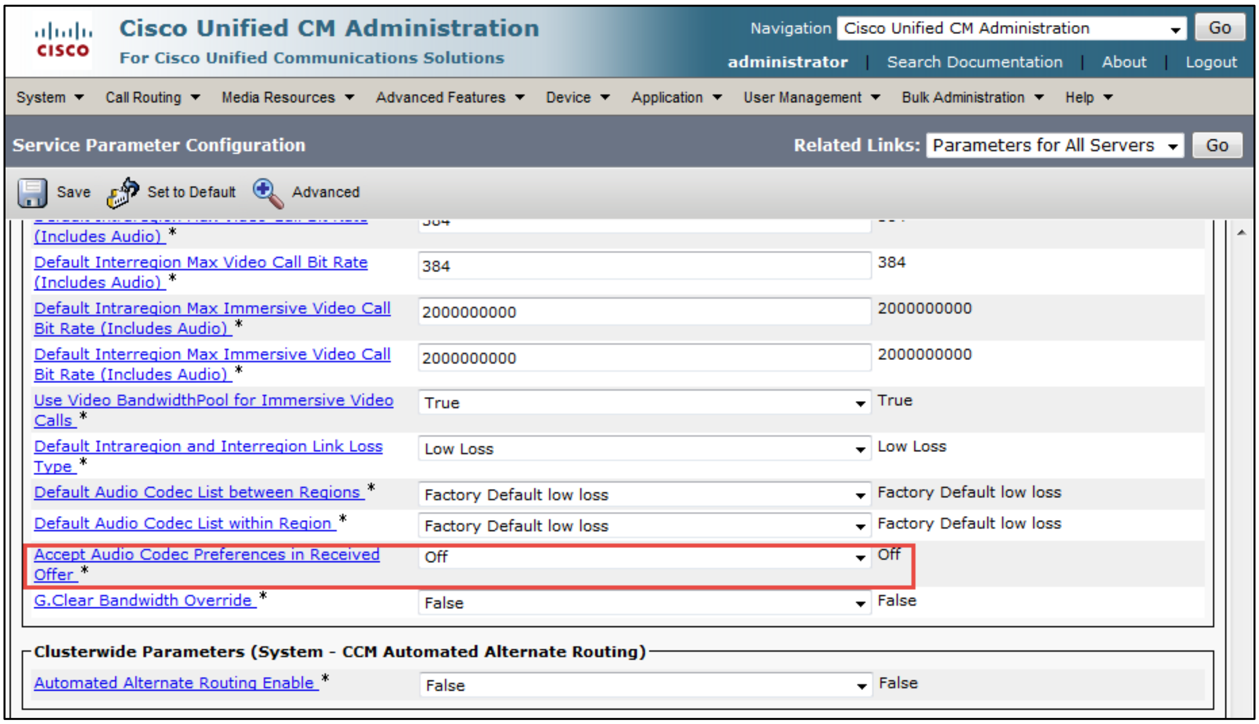
Link Layer Discovery Protocol (LLDP): PC Port*	Enabled
LLDP Asset ID	
LLDP Power Priority*	Unknown
Wireless Headset Hookswitch Control*	Disabled
Display Refresh Rate*	Normal
IPv6 Load Server	
IPv6 Log Server	
802.1x Authentication*	User Controlled
Detect Unified CM Connection Failure*	Normal
Minimum Ring Volume*	0-Silent
Headset Sidetone Level*	Default
Headset Send Gain*	Default
HTTPS Server*	http and https Enabled

Cisco Unified Communications Manager SCCP Phone Ext. 2004 Device Level Configuration (Continued)



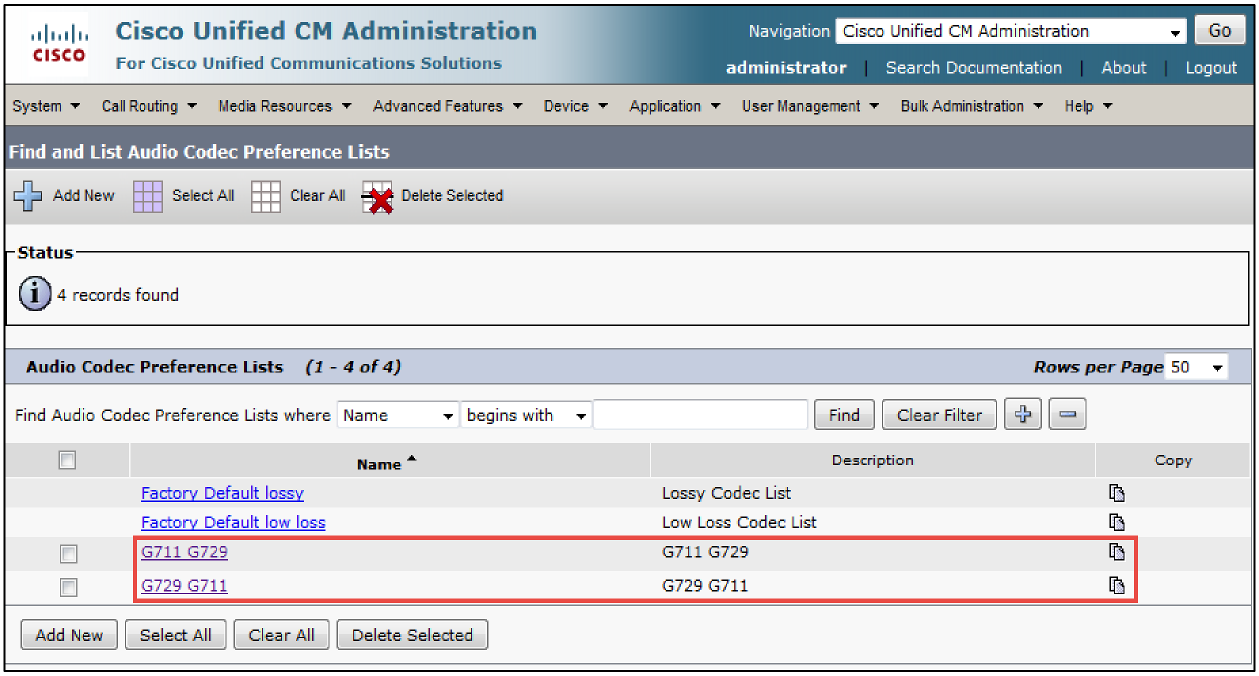
Cisco UCM Audio Codec Preference List Configuration

Set Accept Audio Codec Preference in Received Offer *= Off. This needs to be set when you are wanting to use the Codec Preference List created.



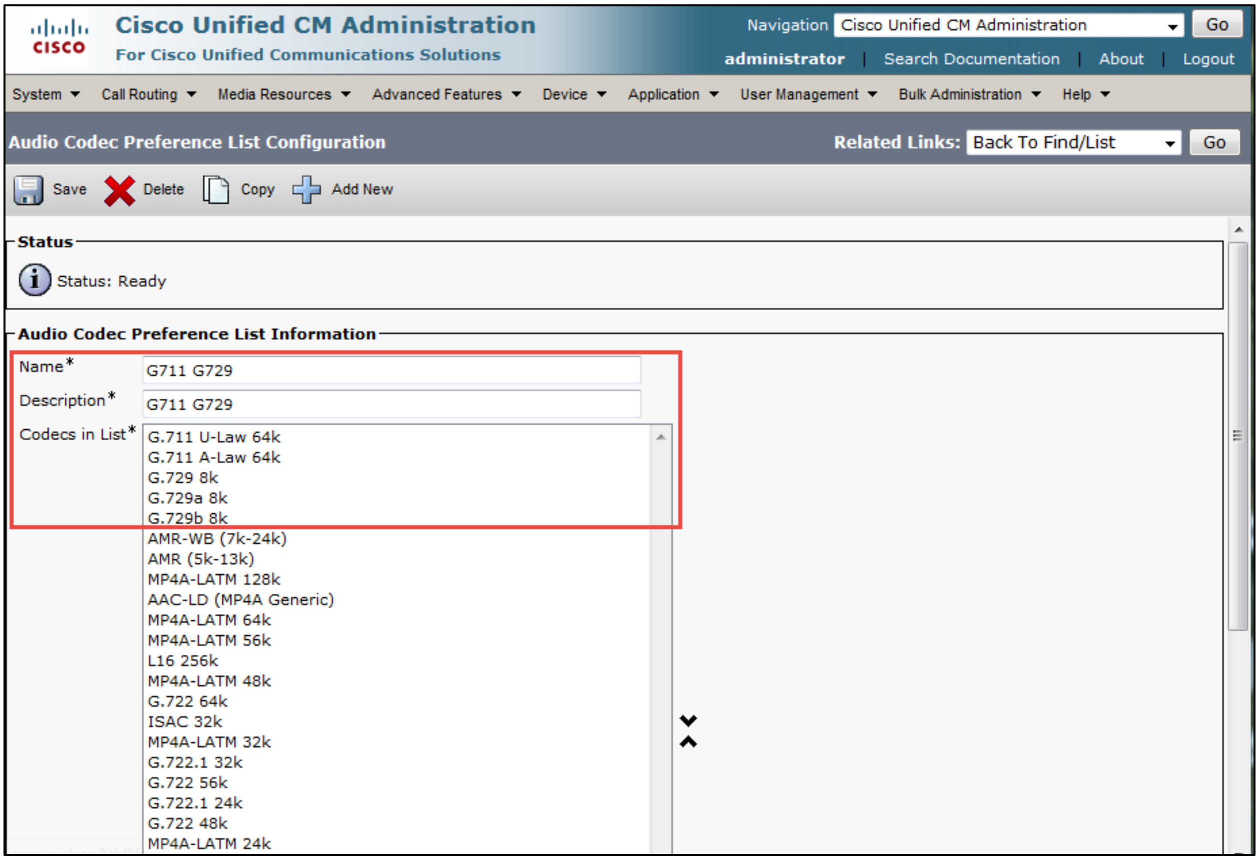
Cisco Unified Communications Manager Audio Codec Preference List Configuration (Continued)

G711 Preferred and G729 Preferred Audio Codec Preference List created in this example. All other values are default.



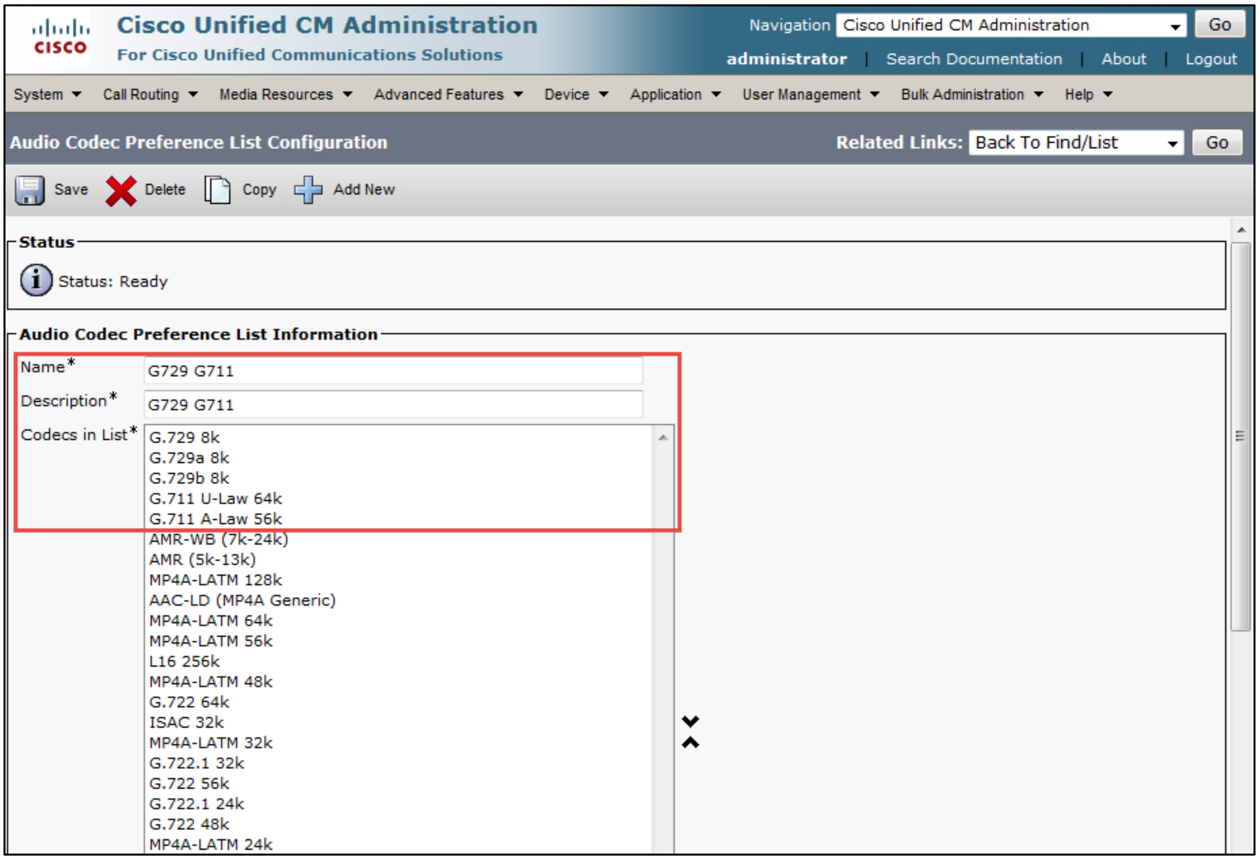
Cisco Unified Communications Manager Audio Codec Preference List Configuration (Continued)

Set Name*= G711 G729. This is used for this example.
Set Description*= G711 G729.This text is used to identify this Audio Codec Preference List.
Set Codec in List*= G.711 U-Law 64k. First choice in this example.
Set Codec in List*= G.711 A-Law 64k. Second choice in this example.



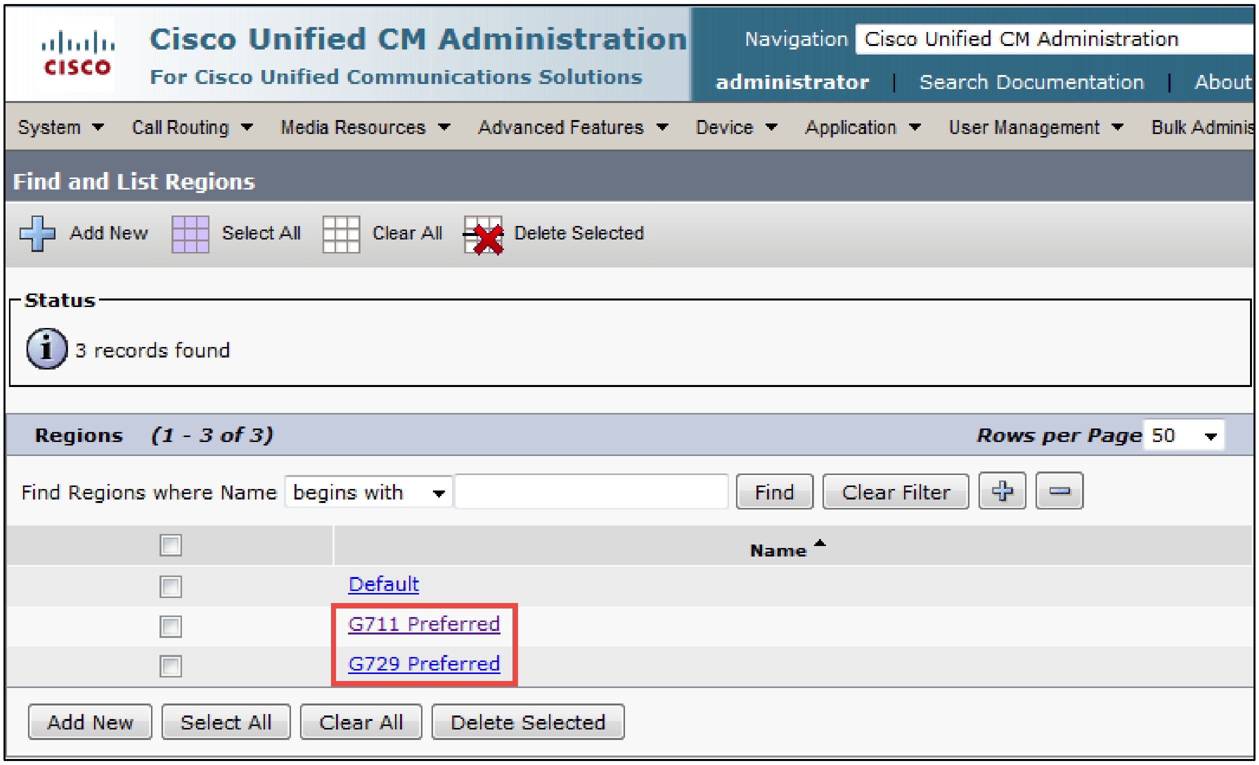
Cisco Unified Communications Manager Audio Codec Preference List Configuration (Continued)

Set Name*= G729 G711. This is used for this example.
Set Description* = G729 G711.This text is used to identify this Audio Codec Preference List.
Set Codec in List*= G.729 8k. First choice for this example.
Set Codec in List*= G.729a 8k. Second choice for this example.
Set Codec in List*= G.729b 8k. Second choice for this example.



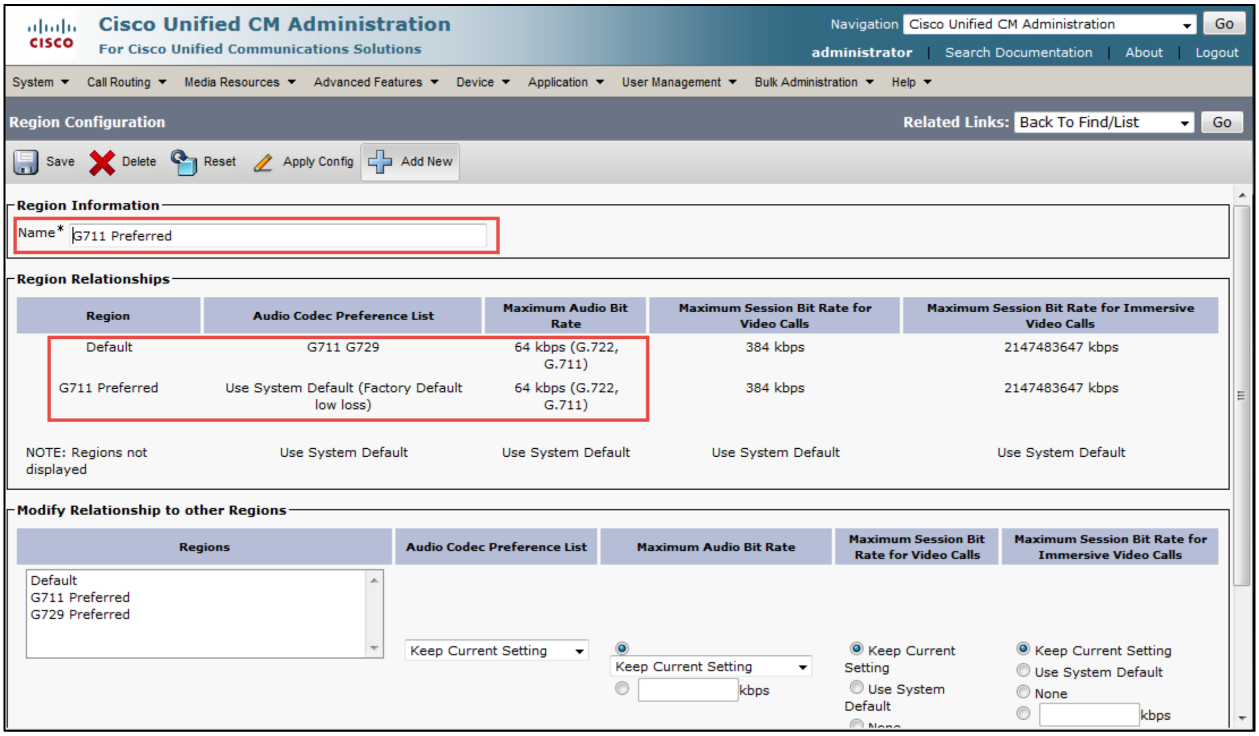
Cisco UCM Region Configuration

G711 Region and G729 Region created in this example.
All other values are default.



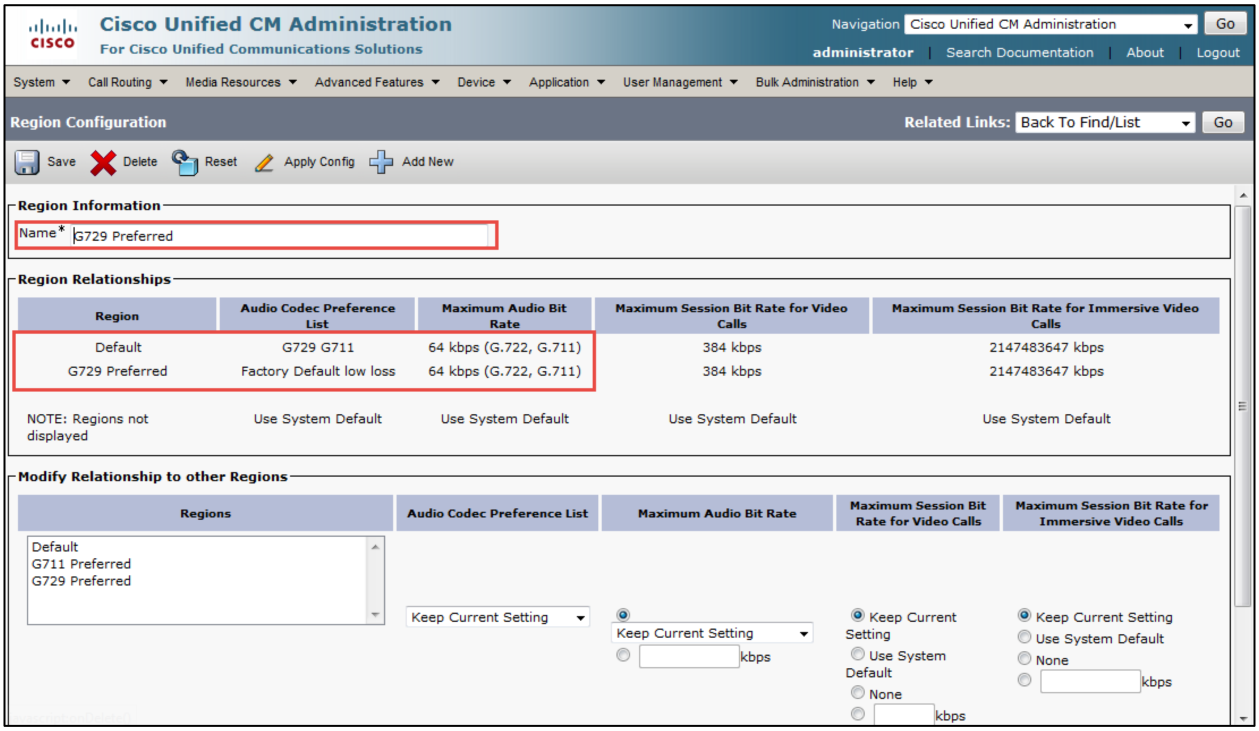
Cisco Unified Communications Manager Region Configuration (Continued)

Set Name*= G711 Preferred. This is used in this example.
Set Region= G711 Preferred. This is used in this example
Set Audio Codec Preference List= G711 Preferred.
Set Maximum Audio Bit Rate= 64 Kbps (G7.22, G7.11). This is used in this example.



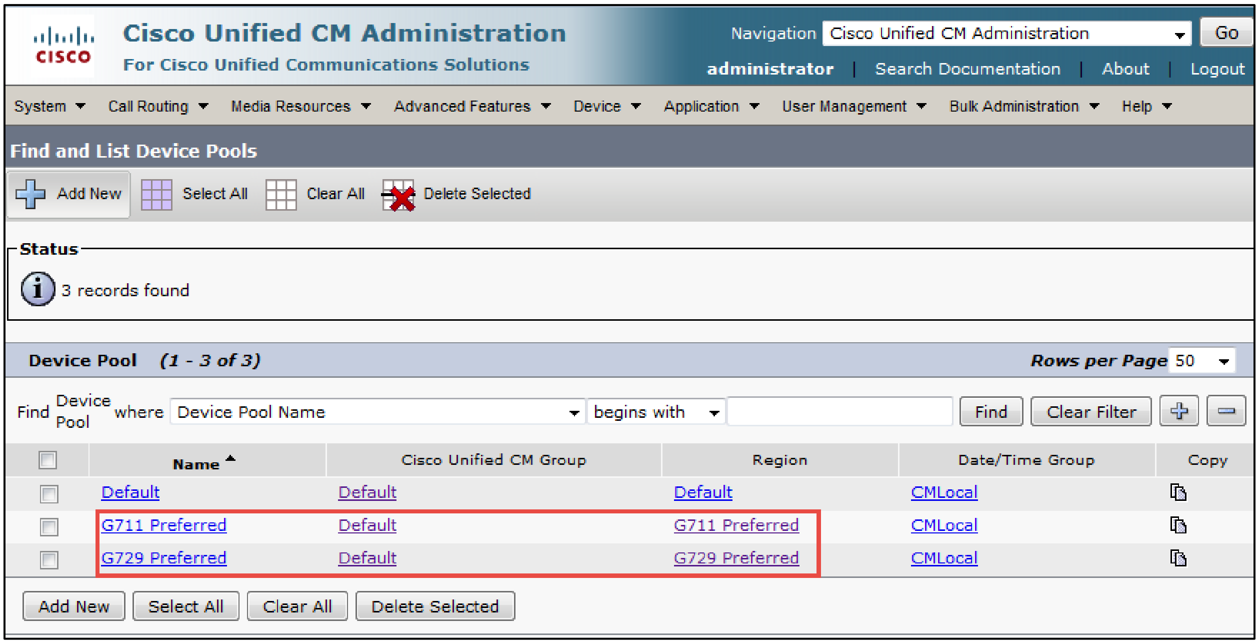
Cisco Unified Communications Manager Region Configuration (Continued)

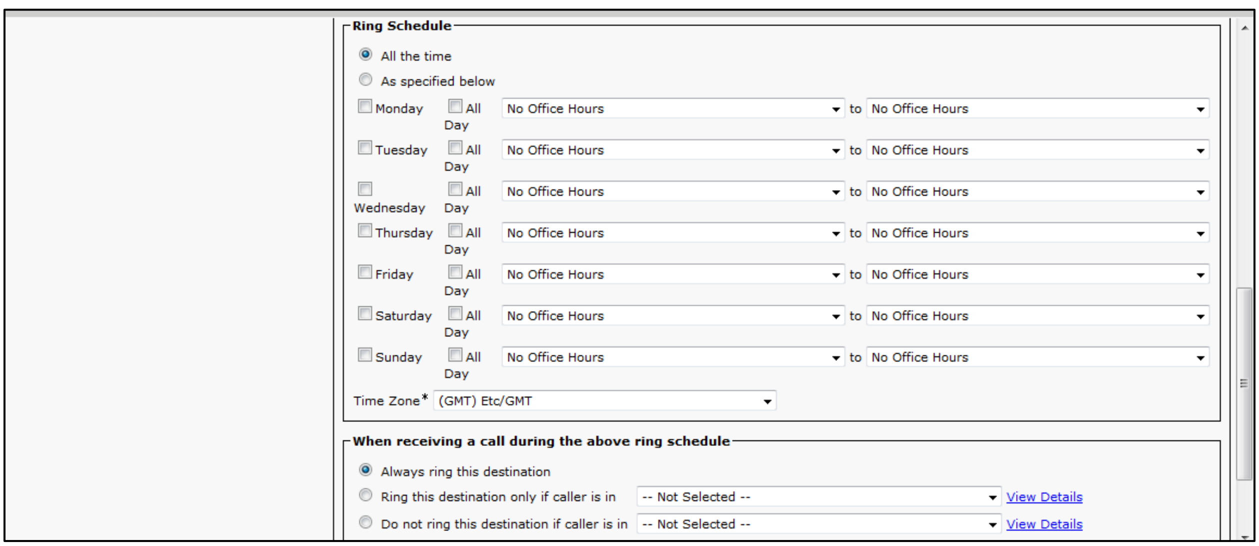
Set Name*= G729 Region. This is used in this example.
Set Audio Codec Preference List= G729 Preferred. This is used in this example
Set Maximum Audio Bit Rate= 64 Kbps (G7.22, G7.11). This is used in this example.



Cisco UCM Device Pool Configuration

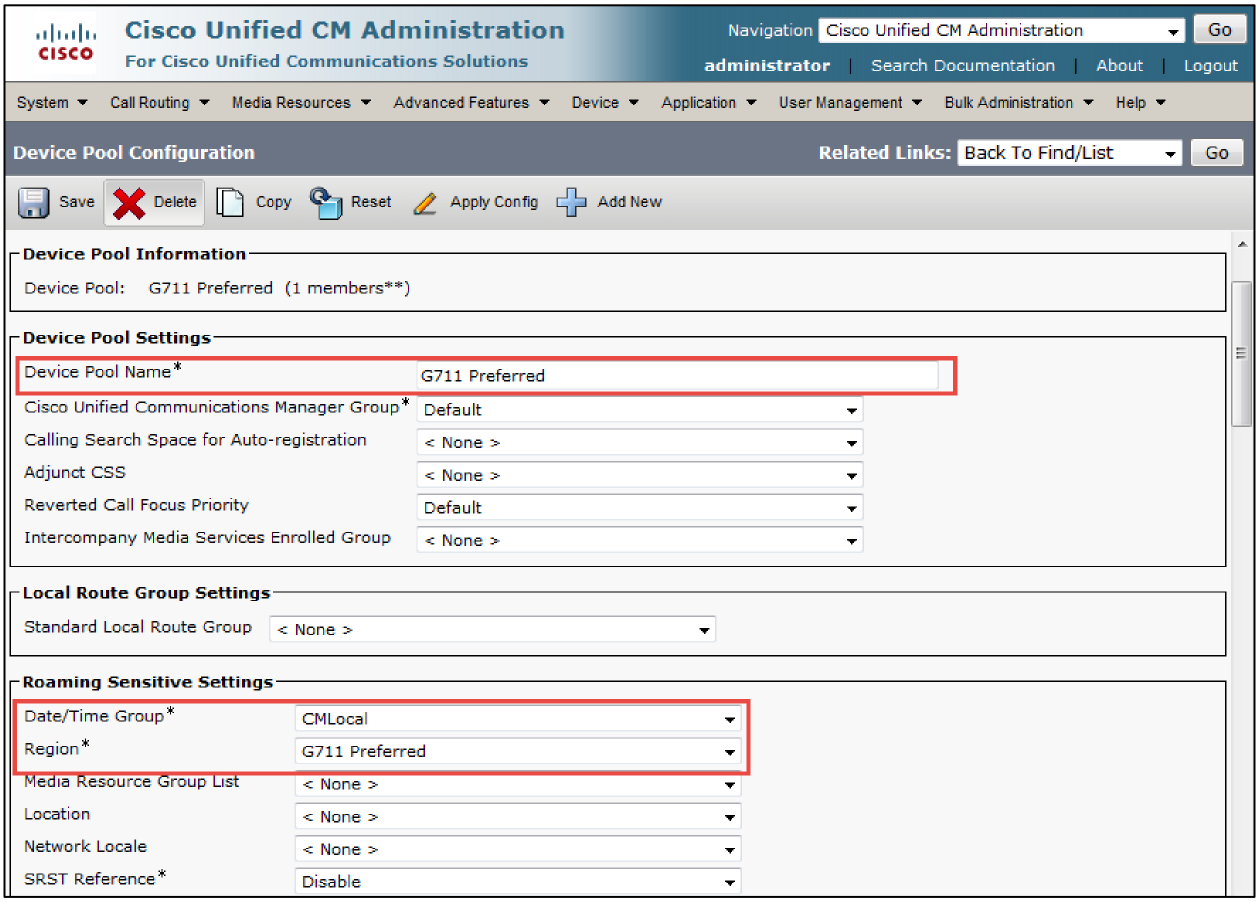
G711 Preferred Pool and G729 Preferred Pool created in this example.
All other values are default.





Cisco Unified Communications Manager Device Pool Configuration (Continued)

Set Device Pool Name*= G711 Preferred. This is used in this example.
Set Cisco Unified Communications Manager Group*= Default.
Set Date/Time Group* = CMLocal.
Set Region* =G711 Preferred. This is used in this example.
All other values are default.



Cisco Unified Communications Manager Device Pool Configuration (Continued)

Connection Monitor Duration***	
Single Button Barge*	Default
Join Across Lines*	Default
Physical Location	< None >
Device Mobility Group	< None >
Wireless LAN Profile Group	< None > View Details

Device Mobility Related Information****

Device Mobility Calling Search Space	< None >
AAR Calling Search Space	< None >
AAR Group	< None >
Calling Party Transformation CSS	< None >
Called Party Transformation CSS	< None >

Geolocation Configuration

Geolocation	< None >
Geolocation Filter	< None >

Call Routing Information

Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings | Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	Default		< None >
International Number	Default		< None >
Unknown Number	Default		< None >
Subscriber Number	Default		< None >

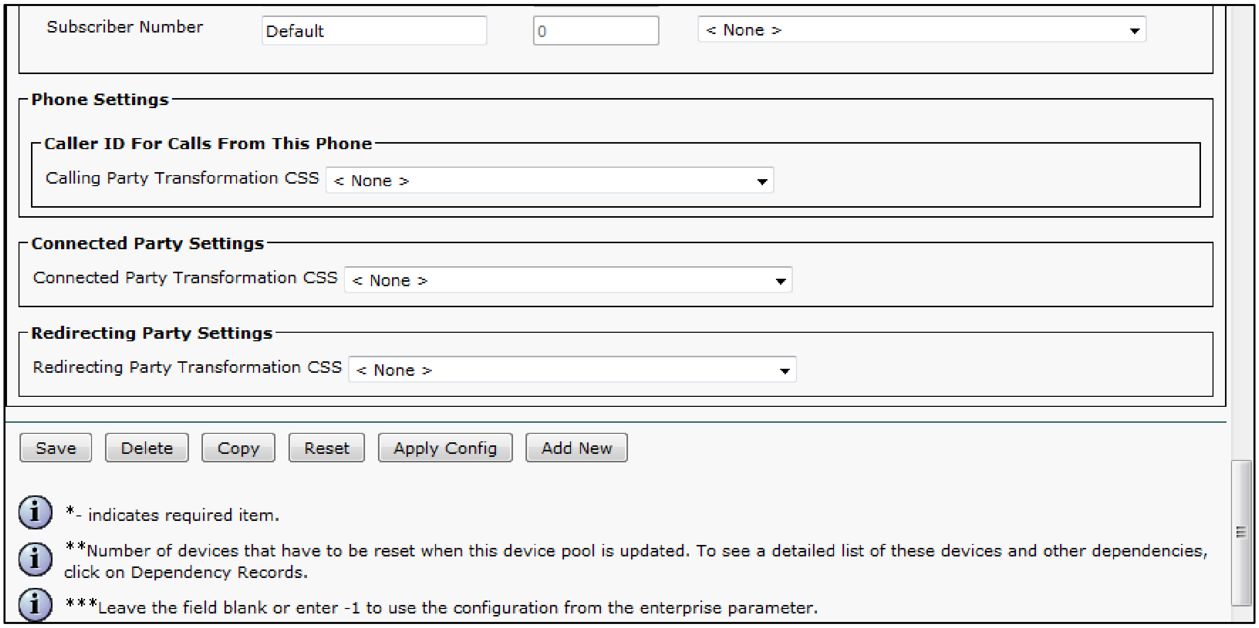
Incoming Called Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings | Default Prefix Settings

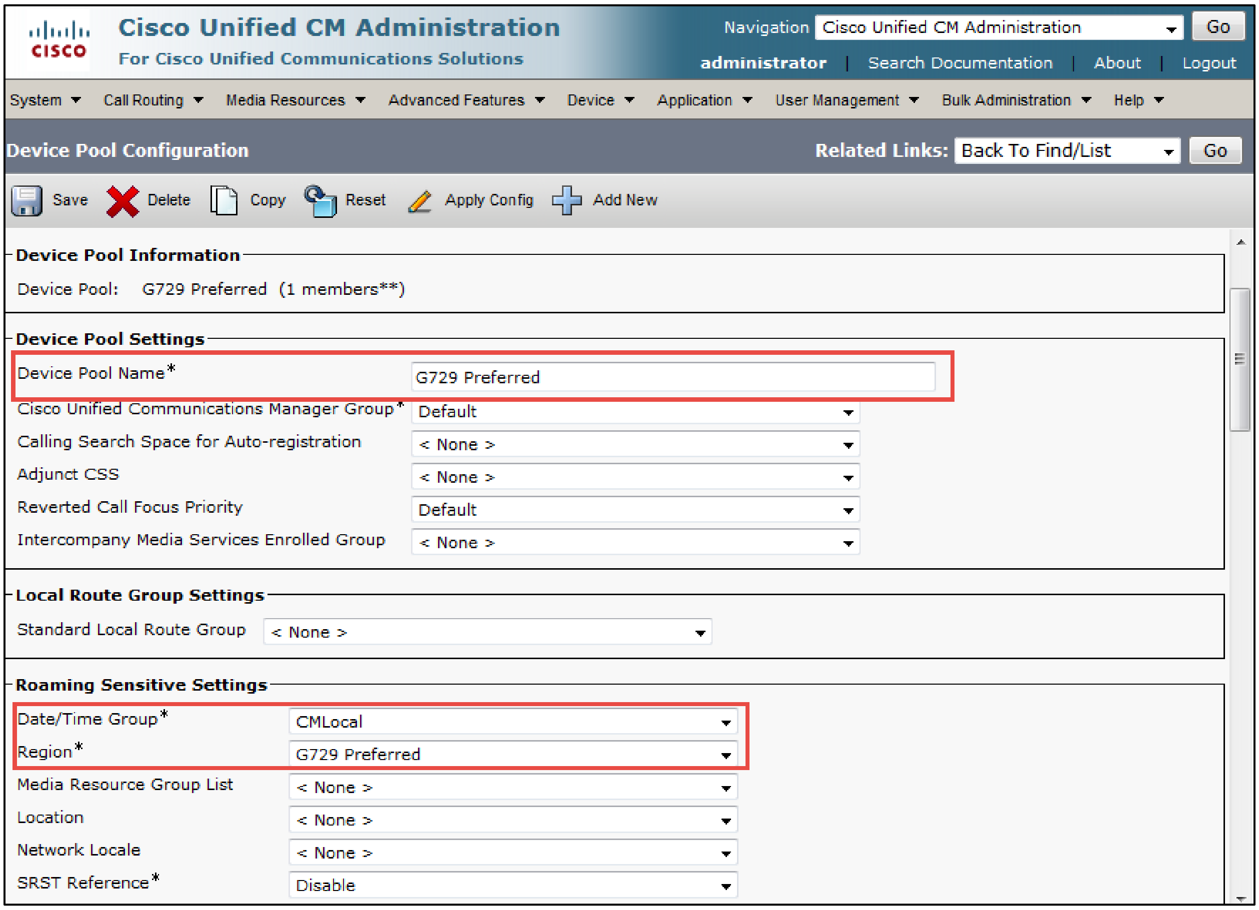
Number Type	Prefix	Strip Digits	Calling Search Space
National Number	Default	0	< None >
International Number	Default	0	< None >
Unknown Number	Default	0	< None >

Cisco Unified Communications Manager Device Pool Configuration (Continued)

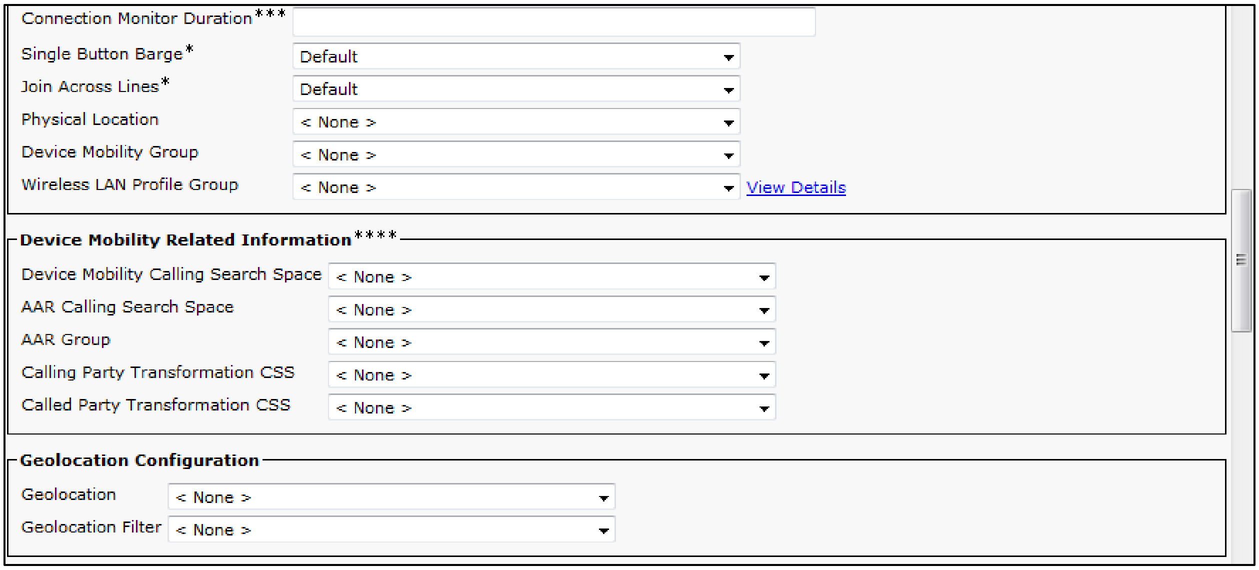


Cisco Unified Communications Manager Device Pool Configuration (Continued)

Set Device Pool Name*= G729 Preferred. This is used in this example.
Set Cisco Unified Communications Manager Group*= Default.
Set Date/Time Group* = CMLocal.
Set Region* =G729 Preferred. This is used in this example.
All other values are default.



Cisco Unified Communications Manager Device Pool Configuration (Continued)



Connection Monitor Duration***

Single Button Barge* Default

Join Across Lines* Default

Physical Location < None >

Device Mobility Group < None >

Wireless LAN Profile Group < None > View Details

Device Mobility Related Information****

Device Mobility Calling Search Space < None >

AAR Calling Search Space < None >

AAR Group < None >

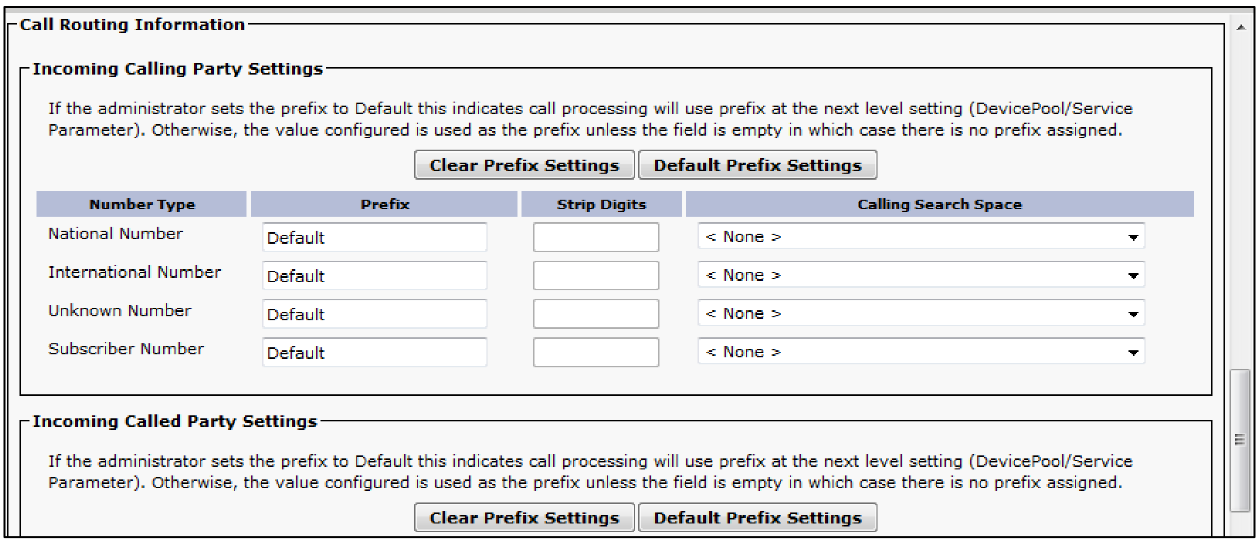
Calling Party Transformation CSS < None >

Called Party Transformation CSS < None >

Geolocation Configuration

Geolocation < None >

Geolocation Filter < None >



Call Routing Information

Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings | Default Prefix Settings

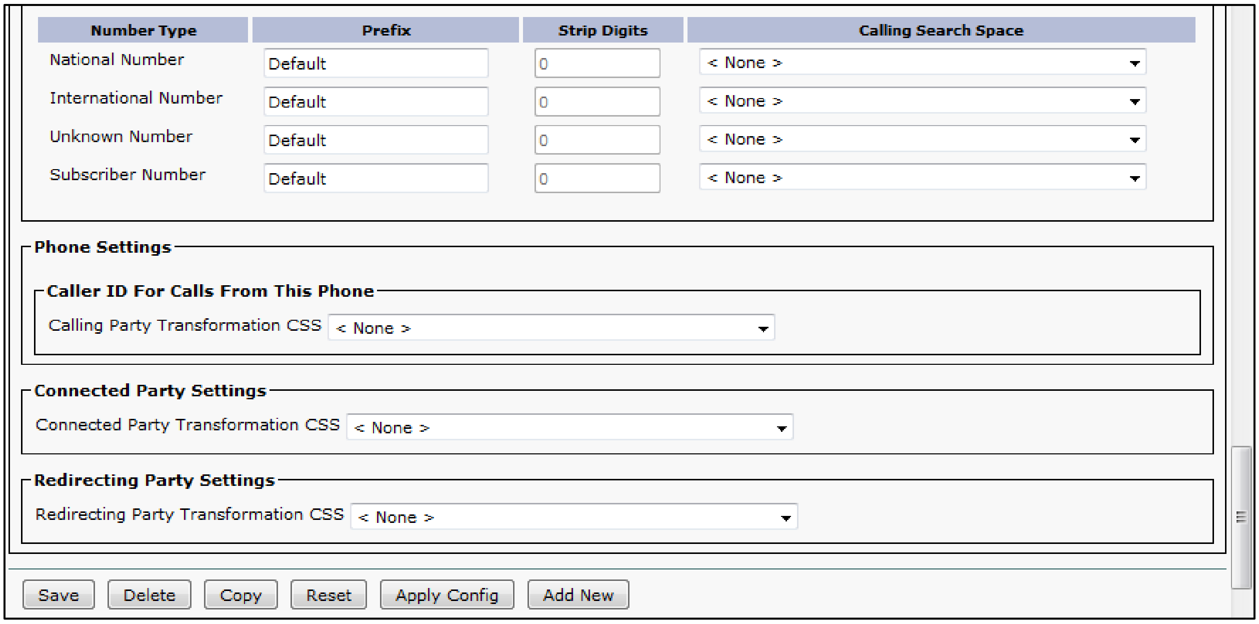
Number Type	Prefix	Strip Digits	Calling Search Space
National Number	Default		< None >
International Number	Default		< None >
Unknown Number	Default		< None >
Subscriber Number	Default		< None >

Incoming Called Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings | Default Prefix Settings

Cisco Unified Communications Manager Device Pool Configuration (Continued)



Cisco UCM Extent and Connect

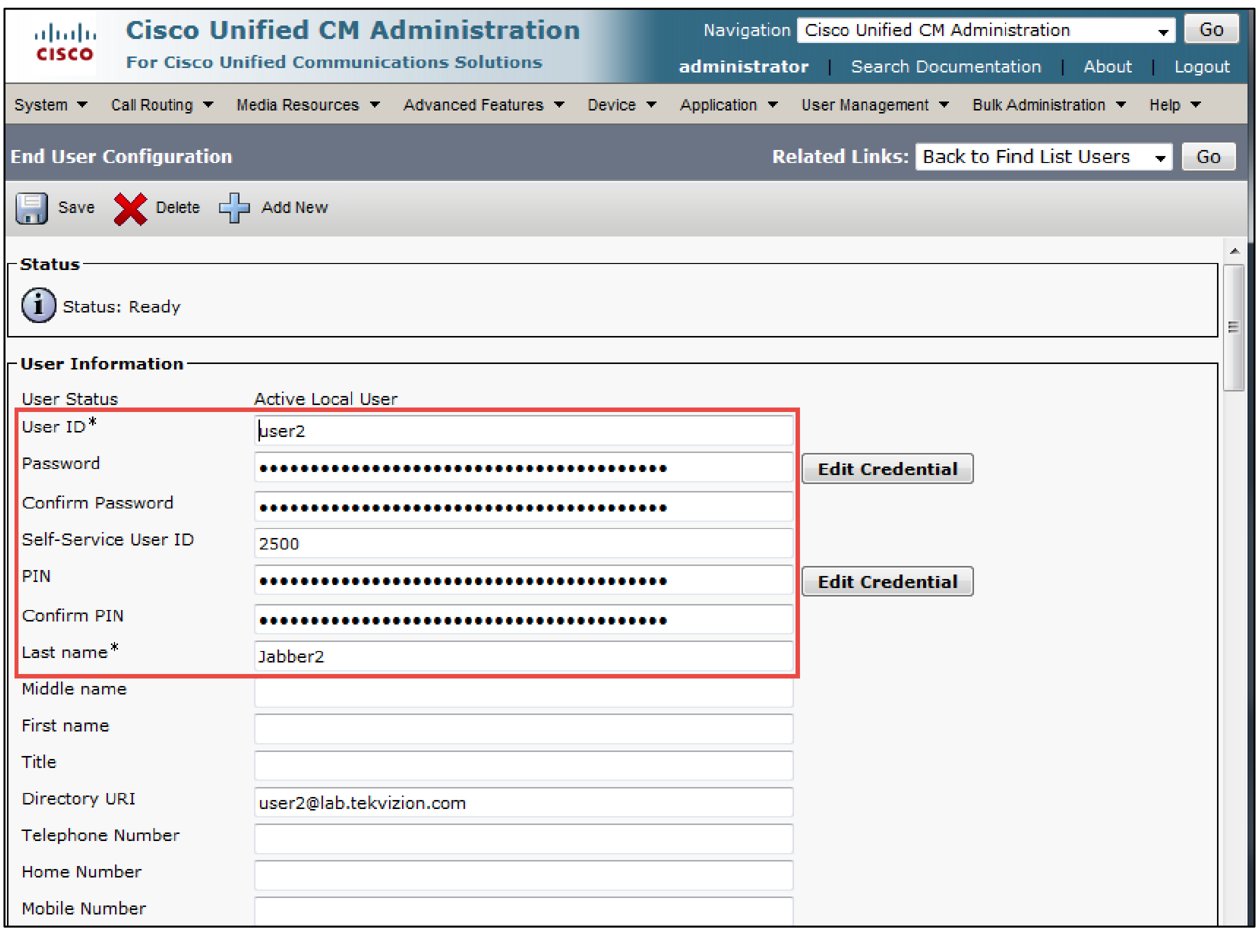
Extend and Connect is a feature that allows administrators to rapidly deploy UC Computer Telephony Integration (CTI) applications which interoperate with any endpoint. With Extend and Connect, users can Leverage the benefits of UC applications from any location using any device. This feature also allows Interoperability between newer UC solutions and legacy systems, so customers can migrate to newer UC Solutions over time as existing hardware is deprecated.

Cisco UCM end user configuration

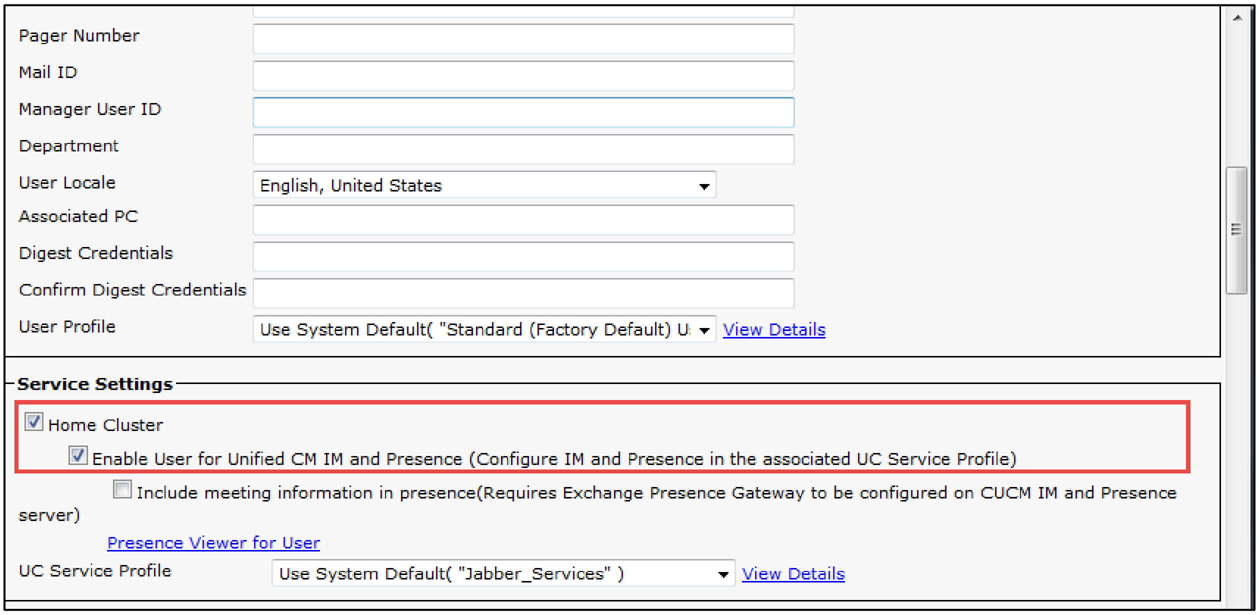
Add user to Cisco UCM

Navigation Path: User management→ End user

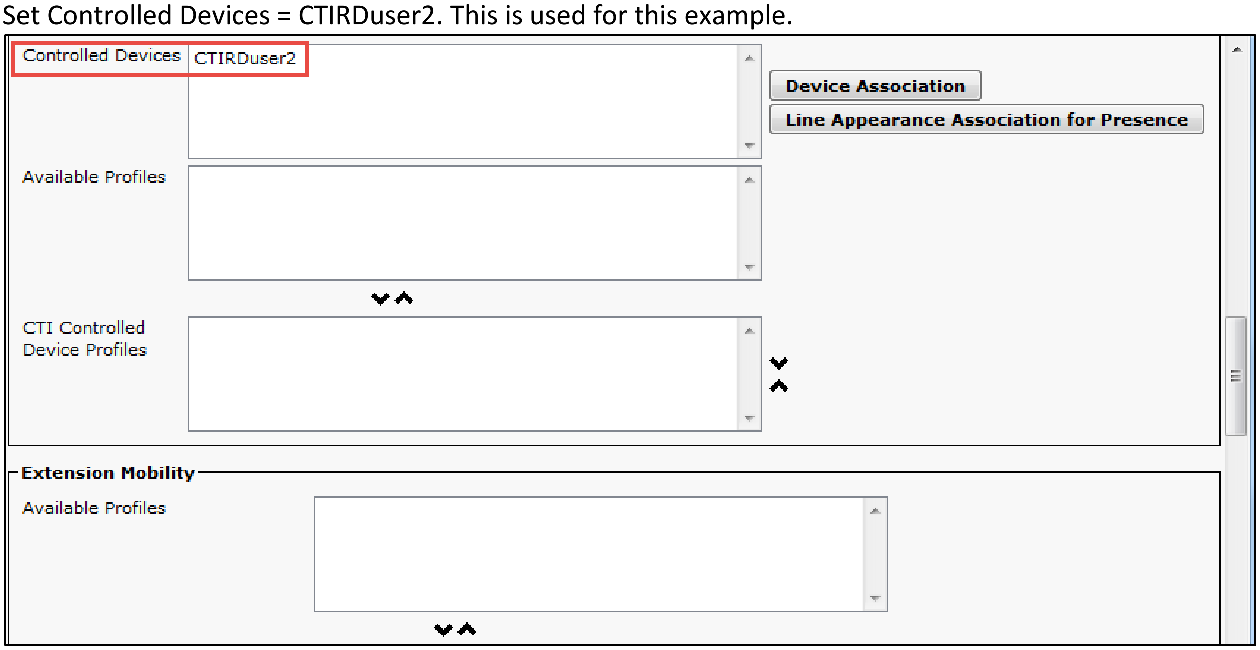
Set User ID*= user2. This is used for this example.
Set Last Name = Jabber2. This is used for this example.
Check Home Cluster.



Cisco UCM end user Configuration (Continued)

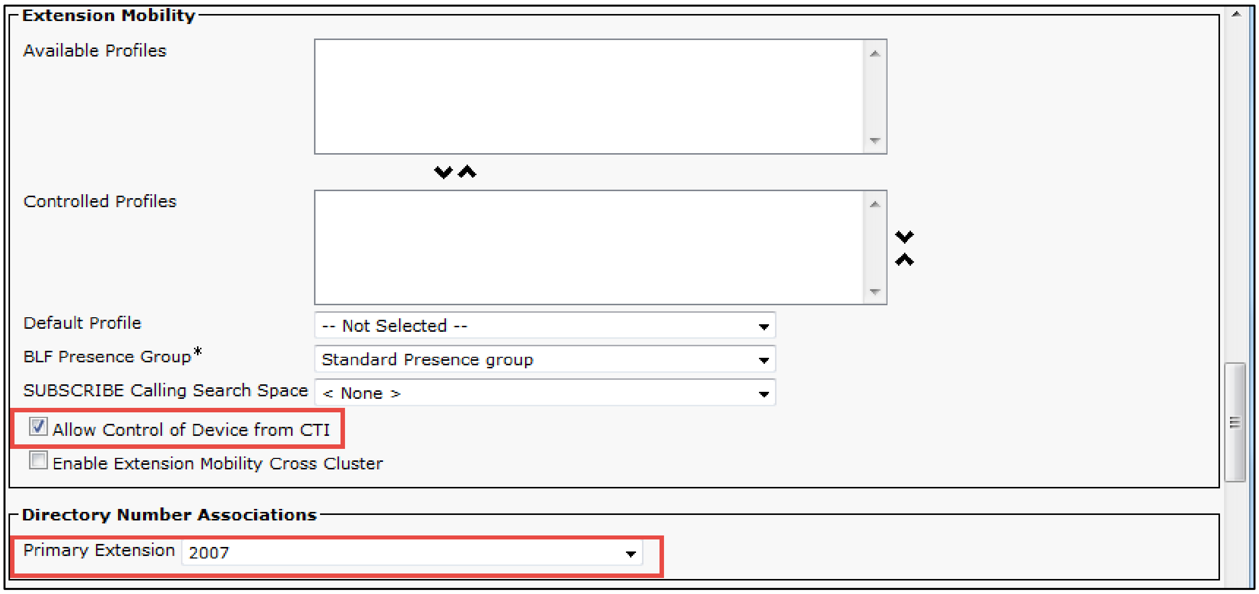


Set Controlled Devices = CTIRDuser2. This is used for this example.

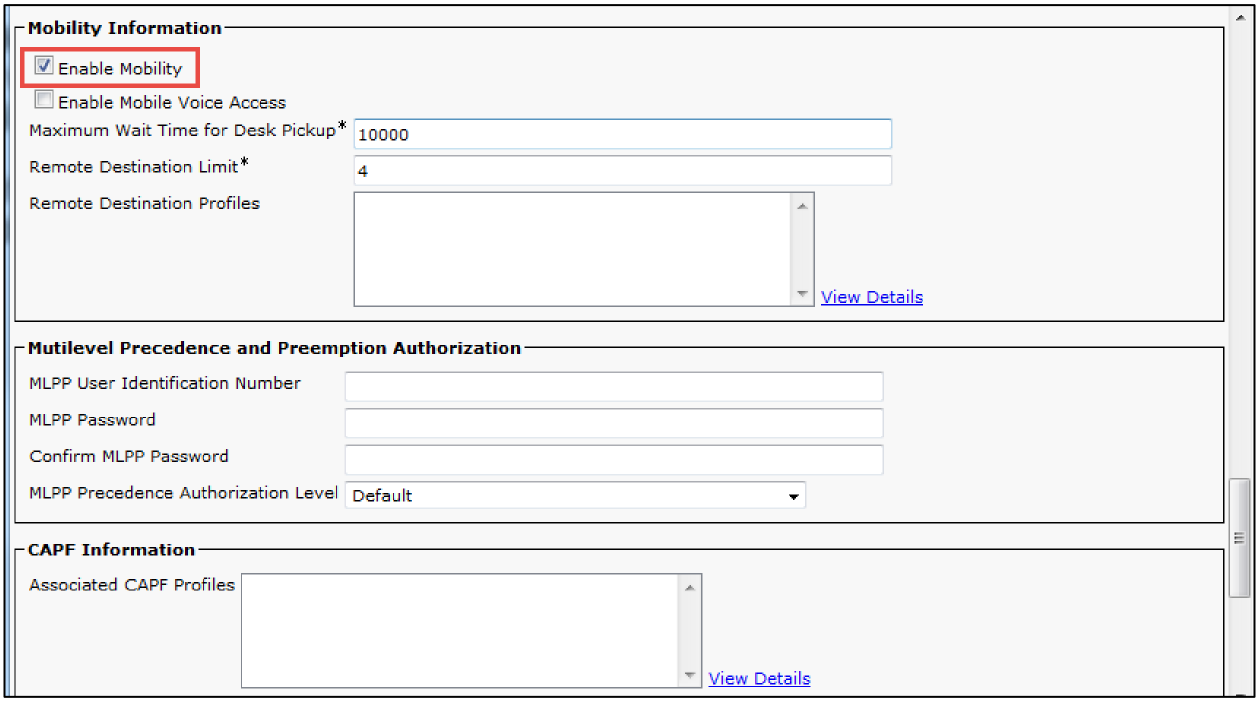


Cisco UCM end user Configuration (Continued)

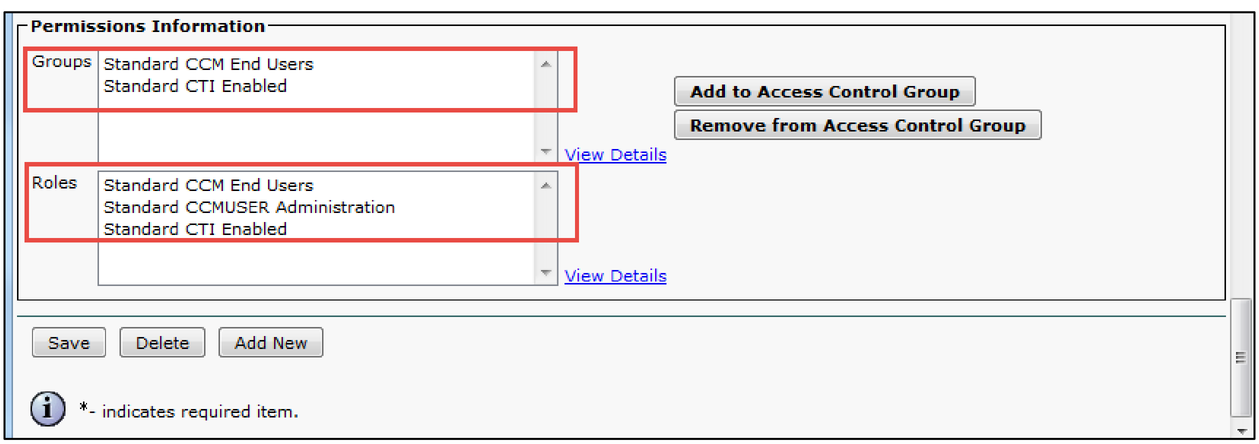
Check Allow Control of Device from CTI
Select the Primary Extension for this user.2007 is used for this example.



Check Enable Mobility

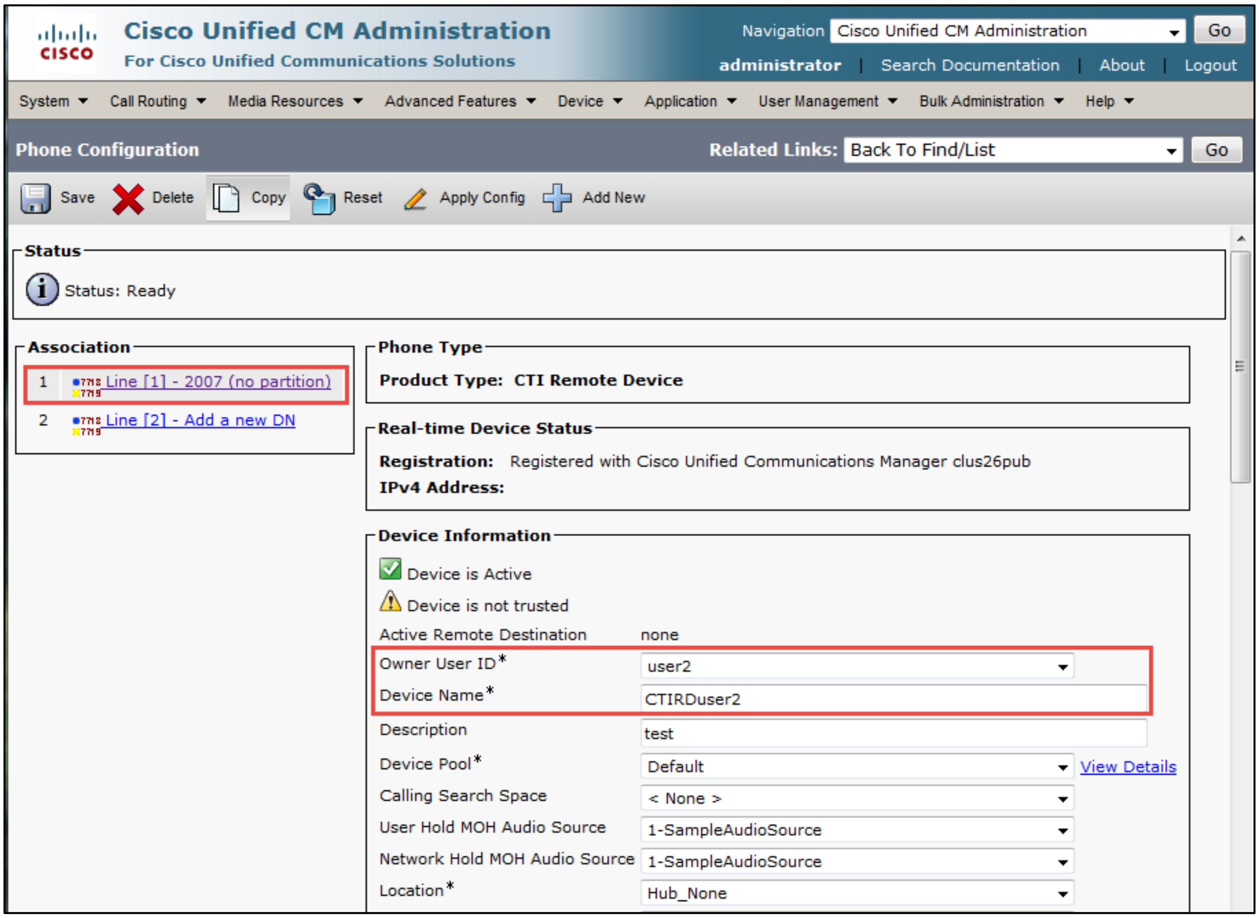


Add the following permissions for Standard Users:
–Standard CCM End-Users
–Standard CTI Enabled
–Standard CCMUSER Administration

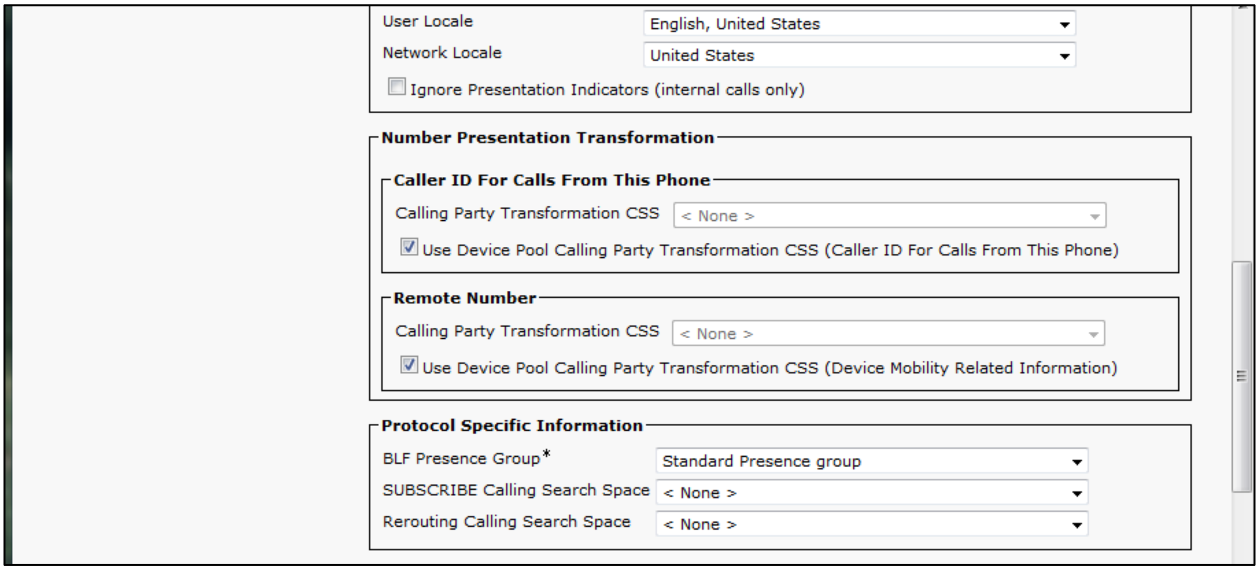


Add Phone: CTI Remote Device

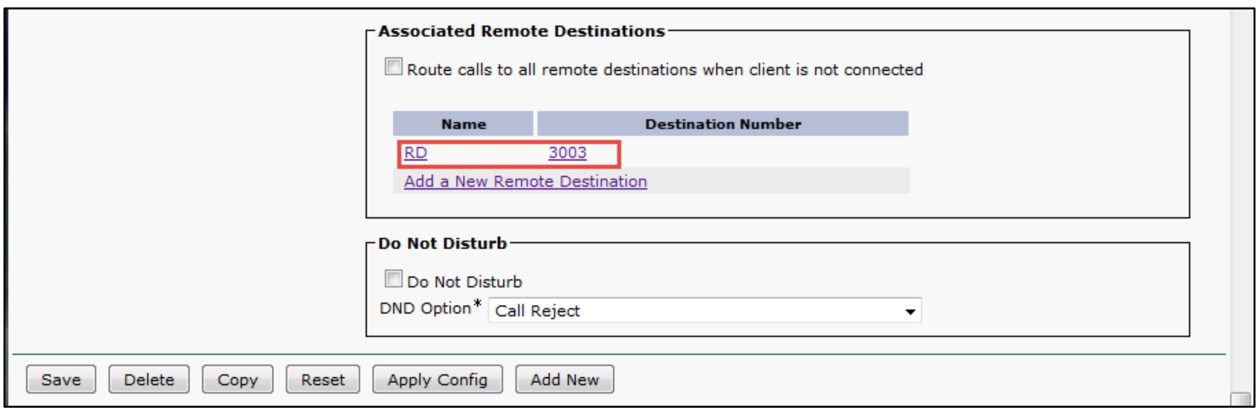
The CTI Remote Device type represents the user's remote device(s) .
Select the desired Owner User ID .User2 is used in this example.
Set the Device name populated automatically. Modify if desired - CTIRDuser2 used this example.
Set Device Pool: Default. This is used in this example.



Cisco UCM CTI remote device Configuration (Continued)

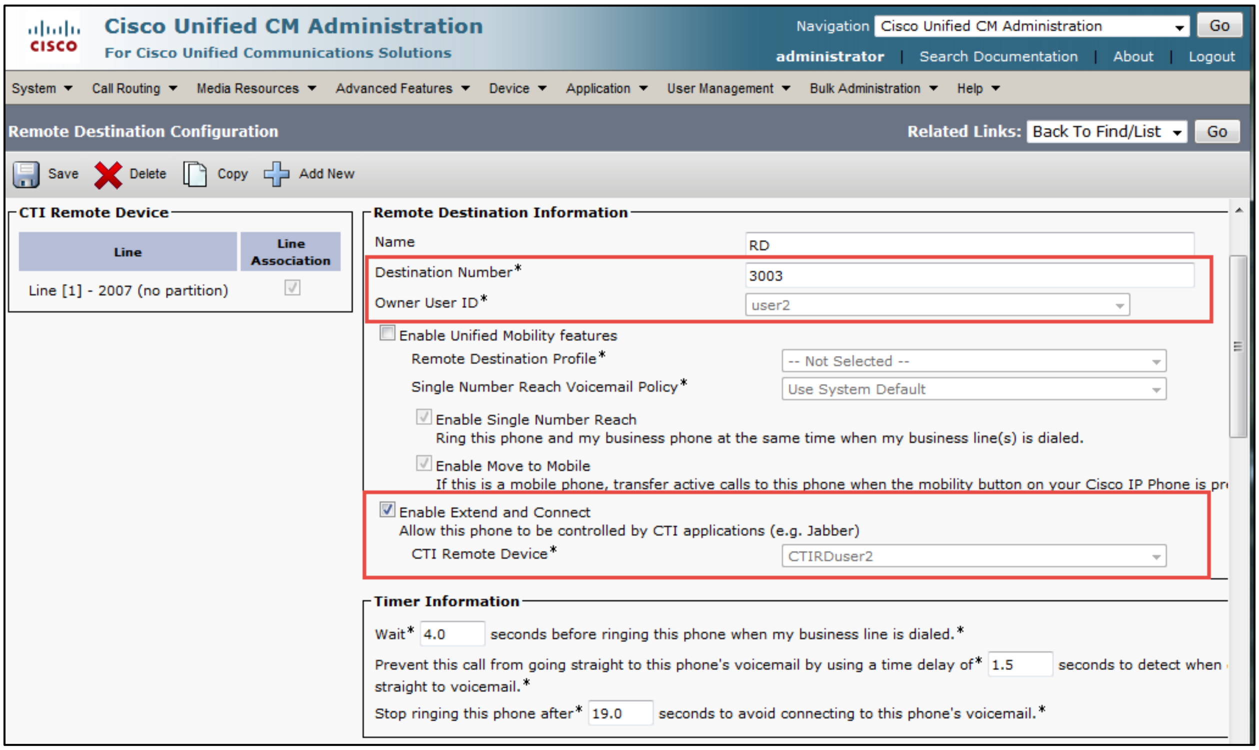


Set RD*= 3003. This is used for this example.3003 is the Avaya extension.



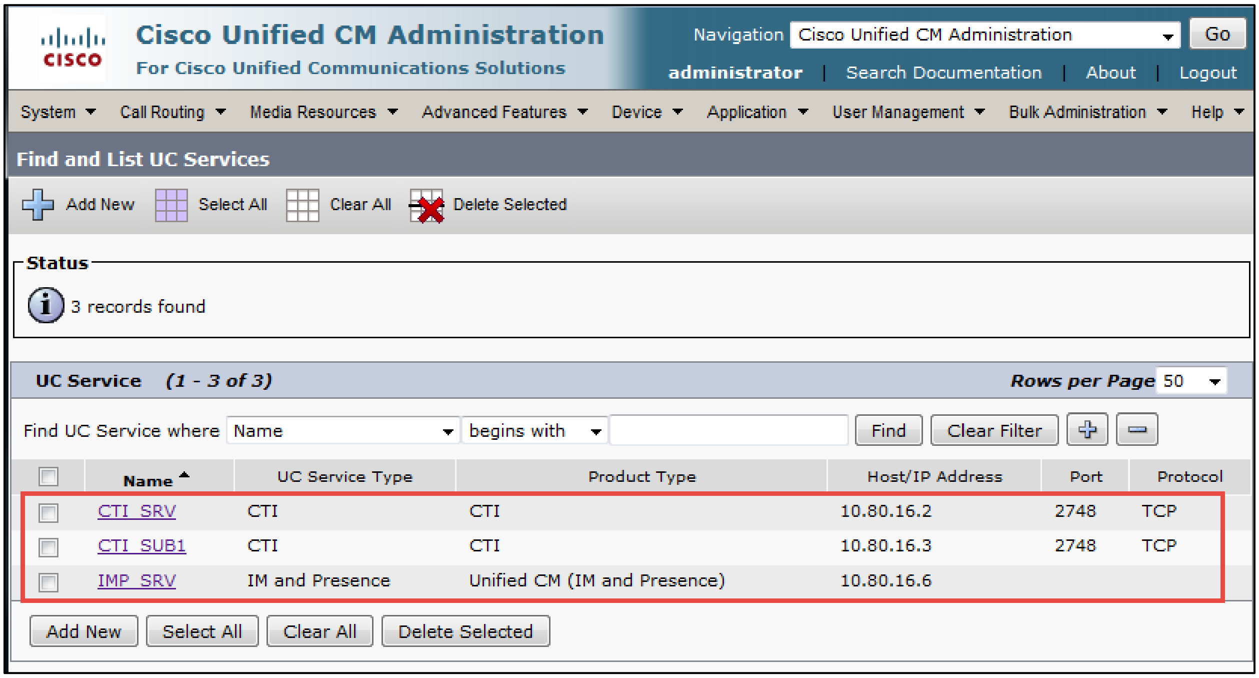
Remote Destination Configuration

Set Destination Number*= 3003. This is used for this example.
Check Enable Extend and Connect.



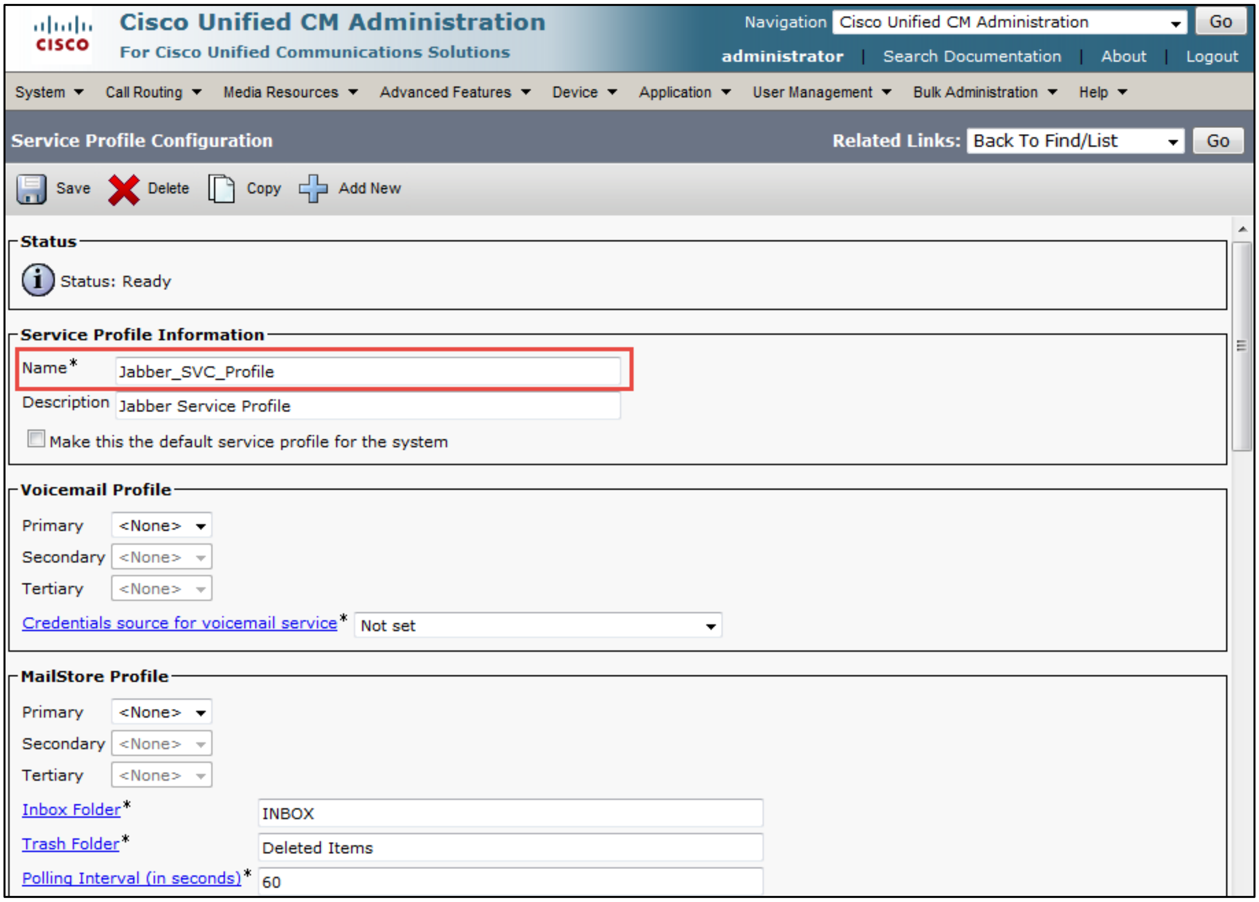
Cisco UCM UC service Configuration

Navigation Path: User management→ User setting → UC Service

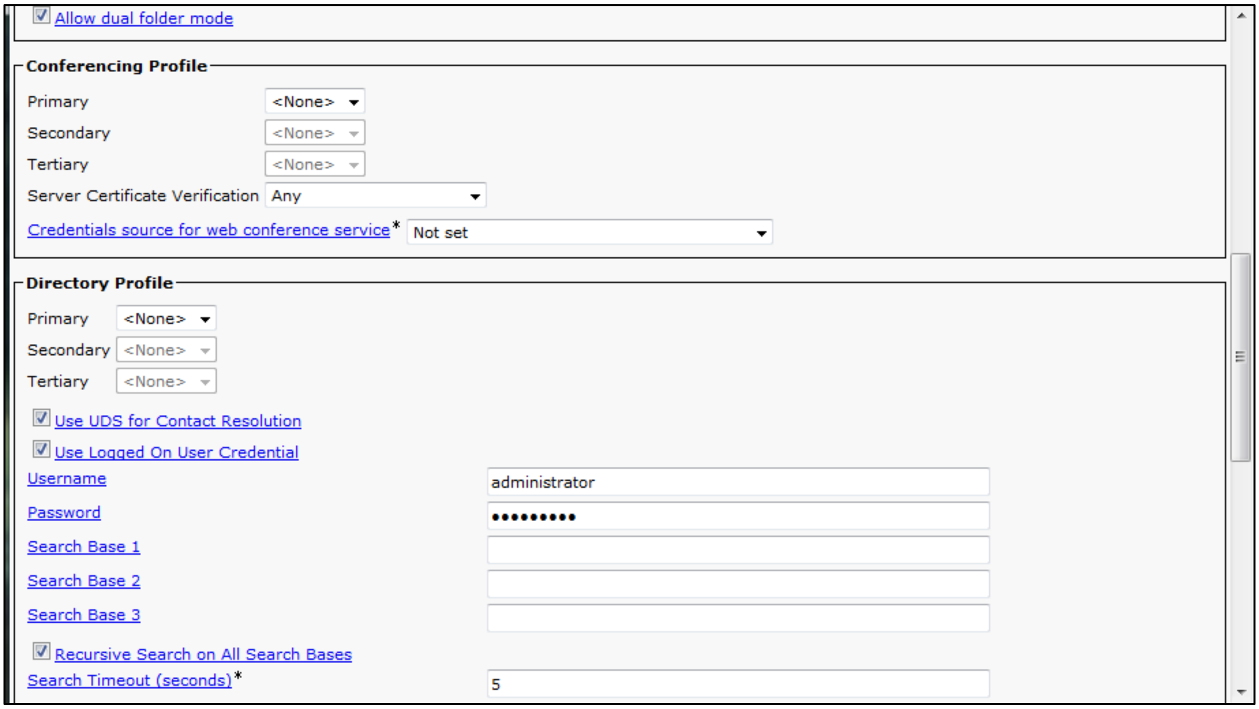


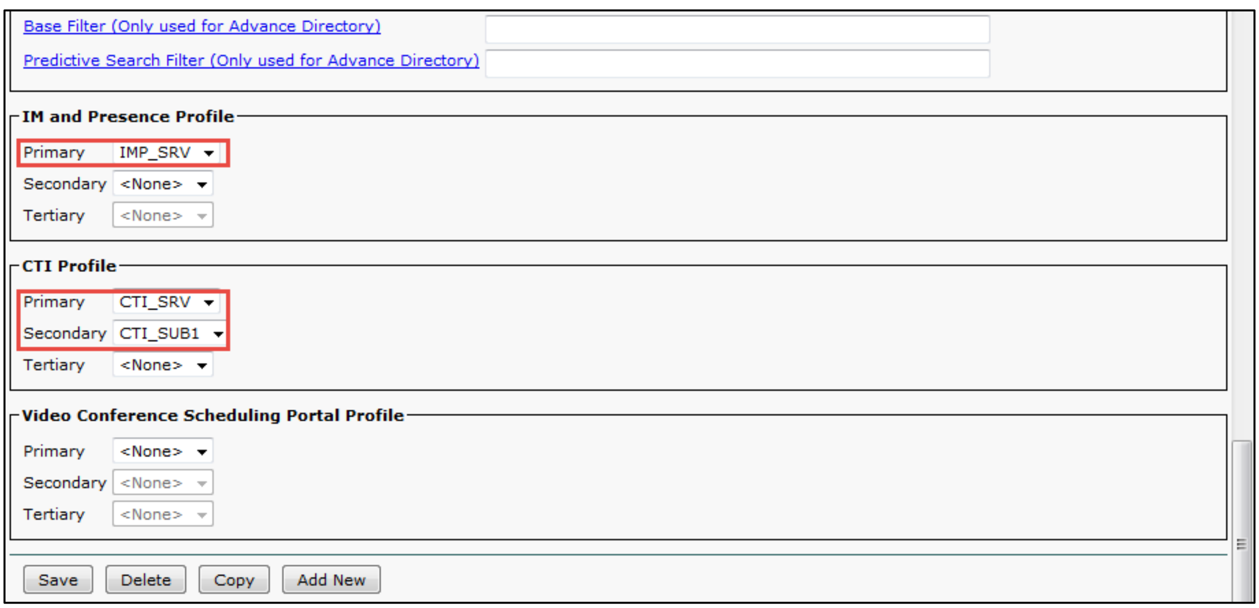
Cisco UCM service Profile Configuration

Navigation Path: User management→ User setting → Service Profile



Cisco UCM service profile Configuration (Continued)

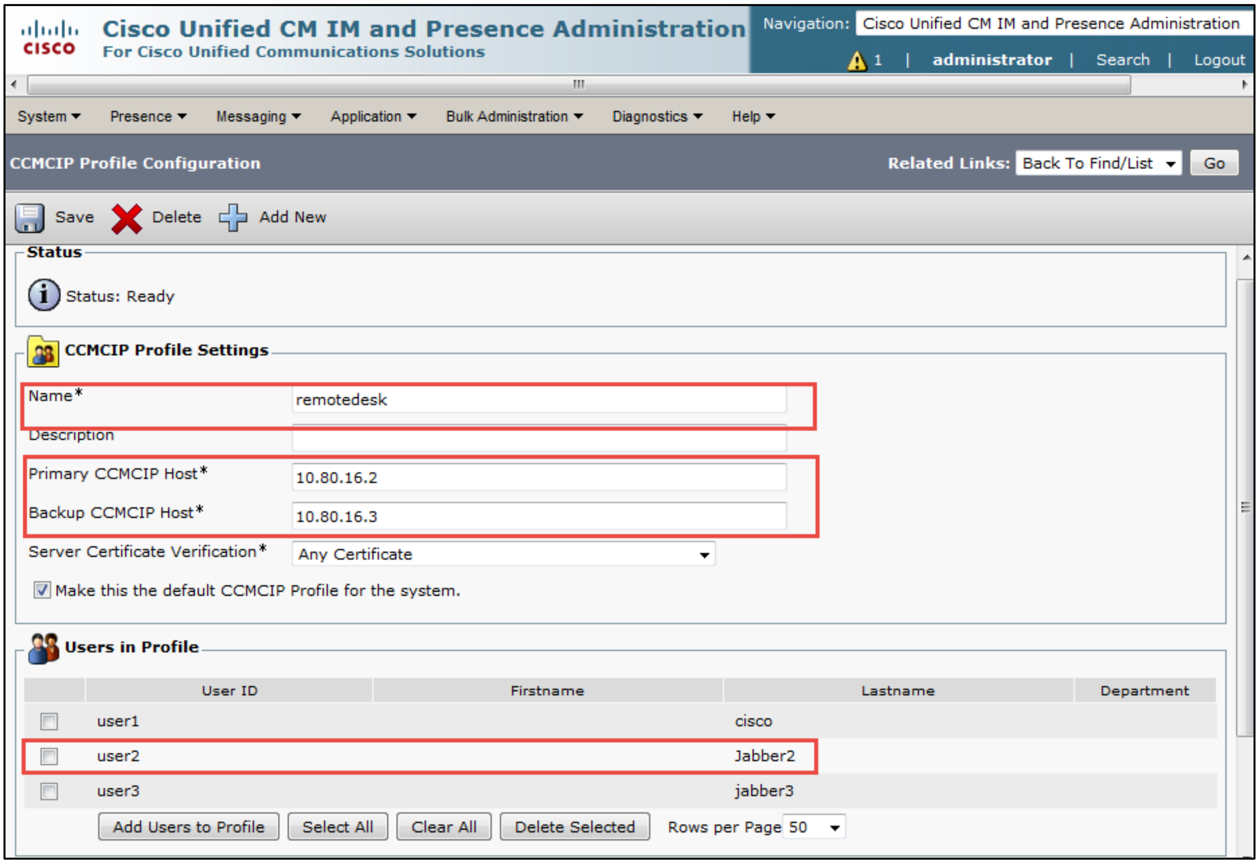




Cisco Unified CM IM Presence – CCMCIP Profile Configuration

Navigation Path: Application→ Legacy Clients → CCMCIP Profile

Set Name *: remotedesk, this is used in this example.
Set Primary CCMCIP Host *: 10.80.16.2.Cisco Publisher IP. This is used in this example.
Set Backup CCMCIP Host *: 10.80.16.3.Cisco Publisher IP. This is used in this example.
Add Users to Profile: user2.This is used in this example.

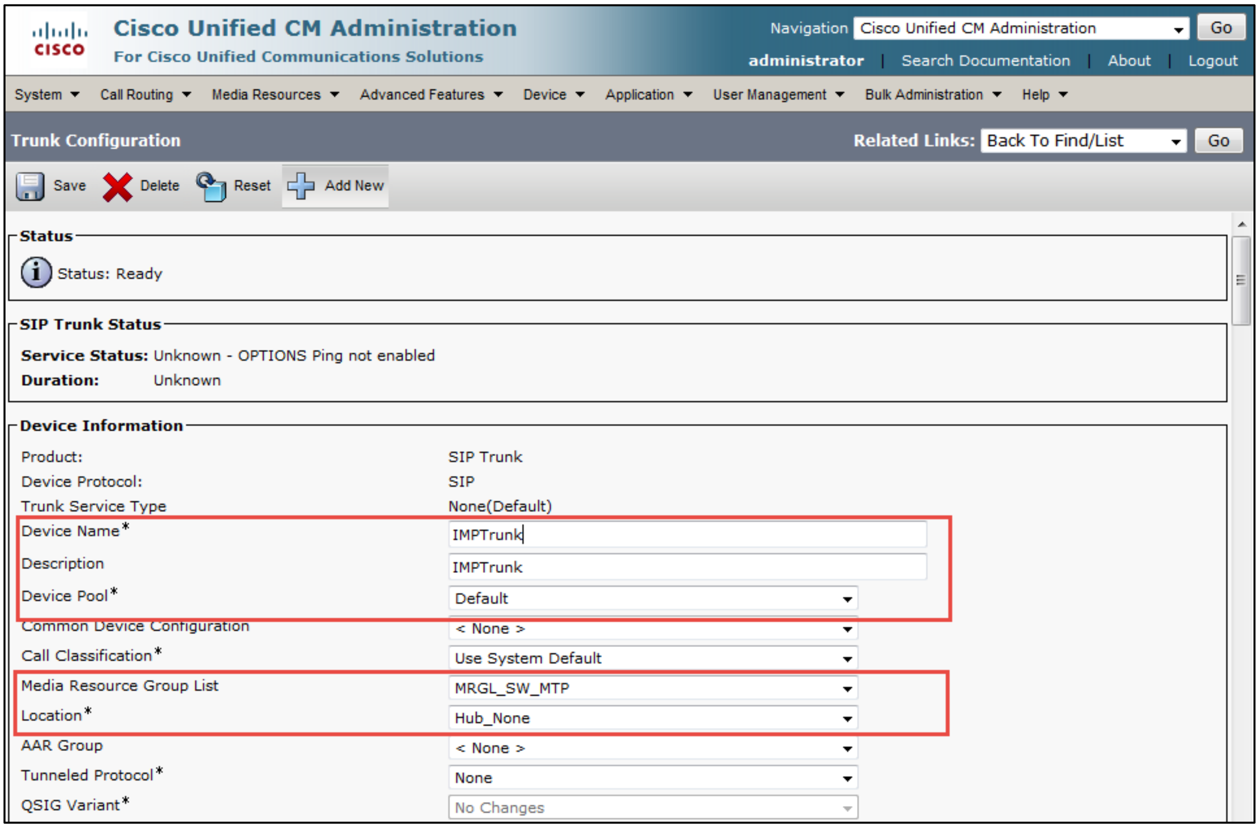


Cisco UCM – SIP trunk to Cisco IM&Presence Trunk Configuration

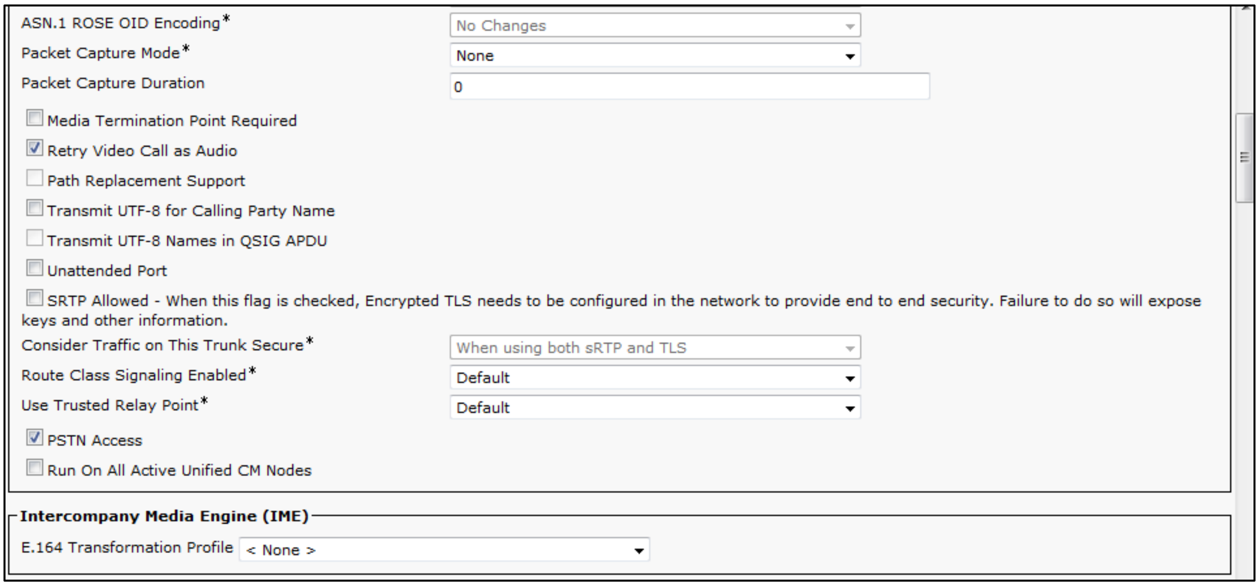
Navigation Path: Device→ Trunk

Set Device Name*= IMPTrunk. This is used for this example.
Set Description = This text is used to identify this Trunk Group.
Set Device Pool* = Default. This is used for this example.
Set Media Resource Group List = MRGL_SW_MTP. This is used for this example.

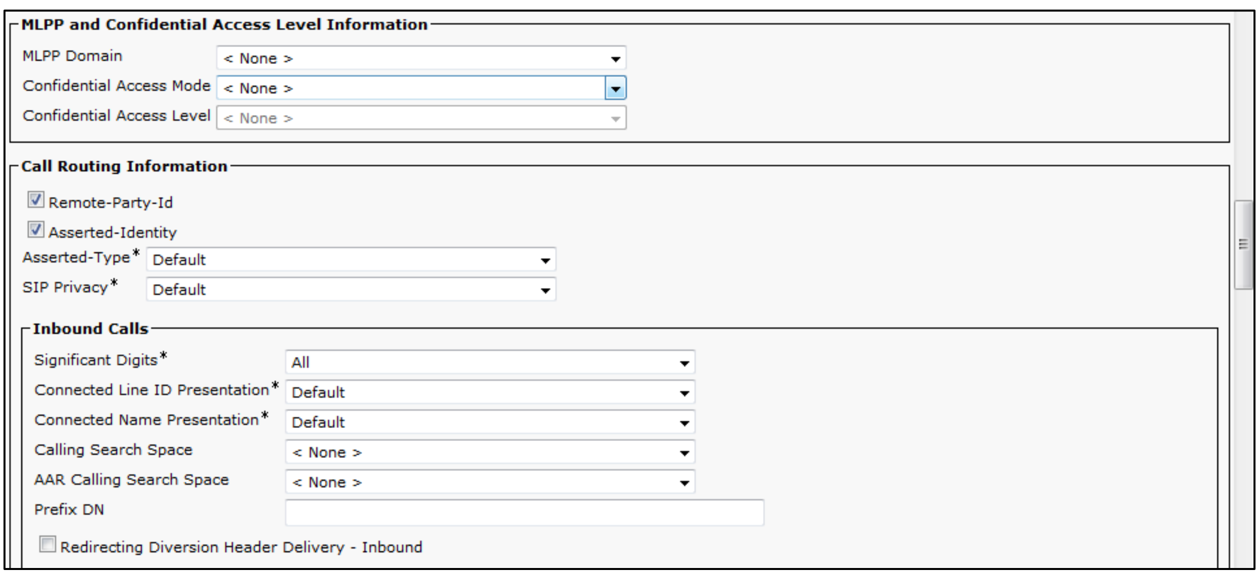
All other values are default.



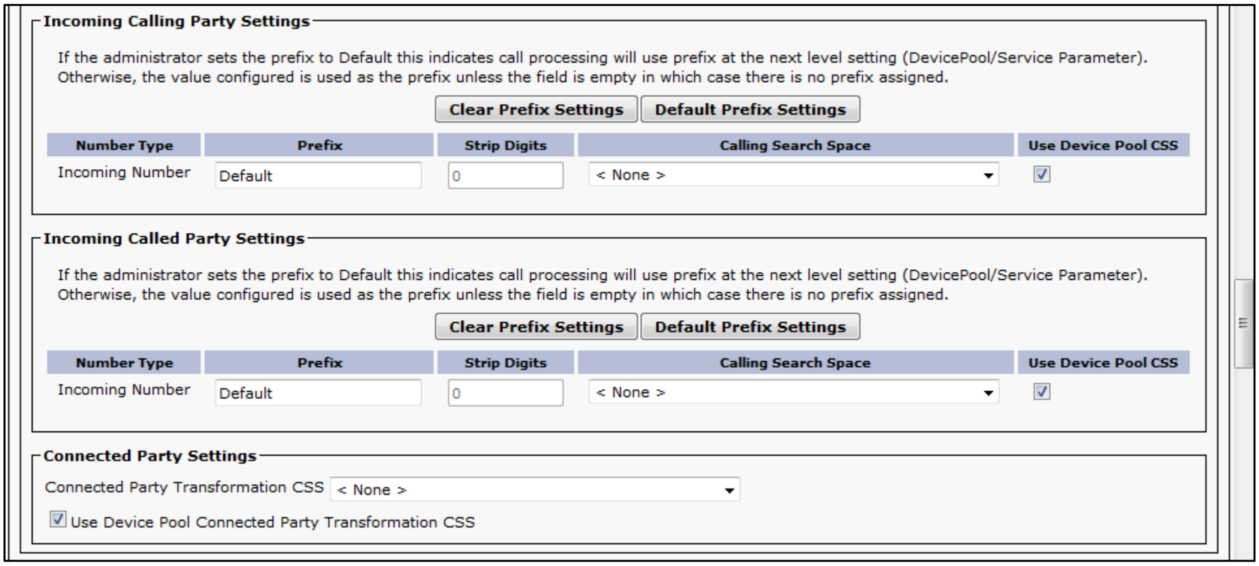
Cisco Unified Communications Manager SIP Trunk to CUP Configuration (Continued)

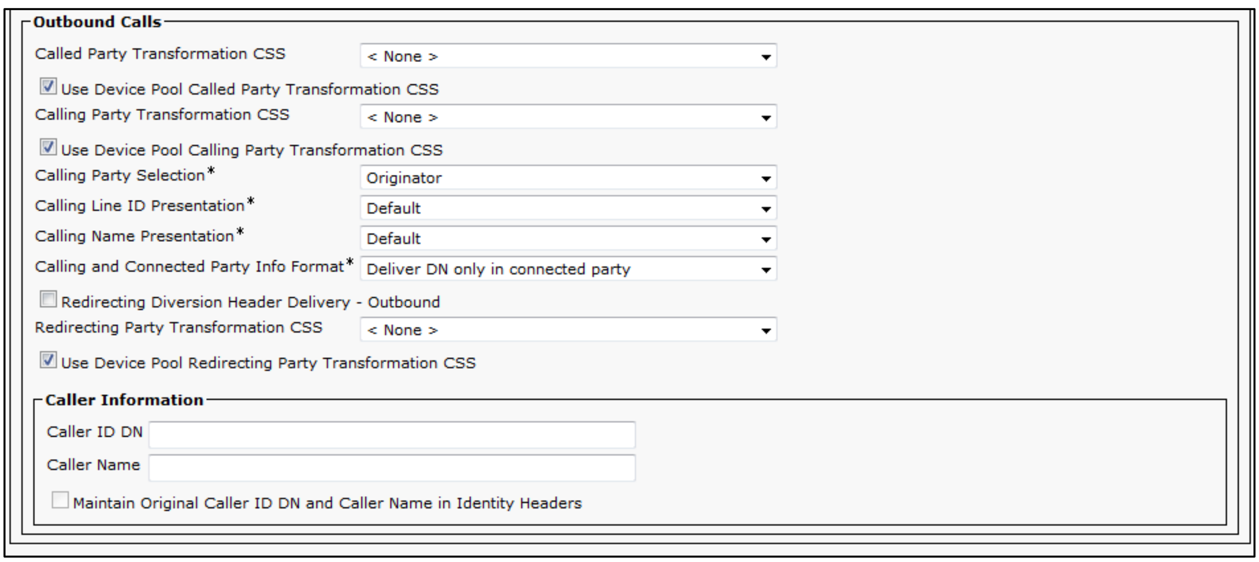


All other values are default.



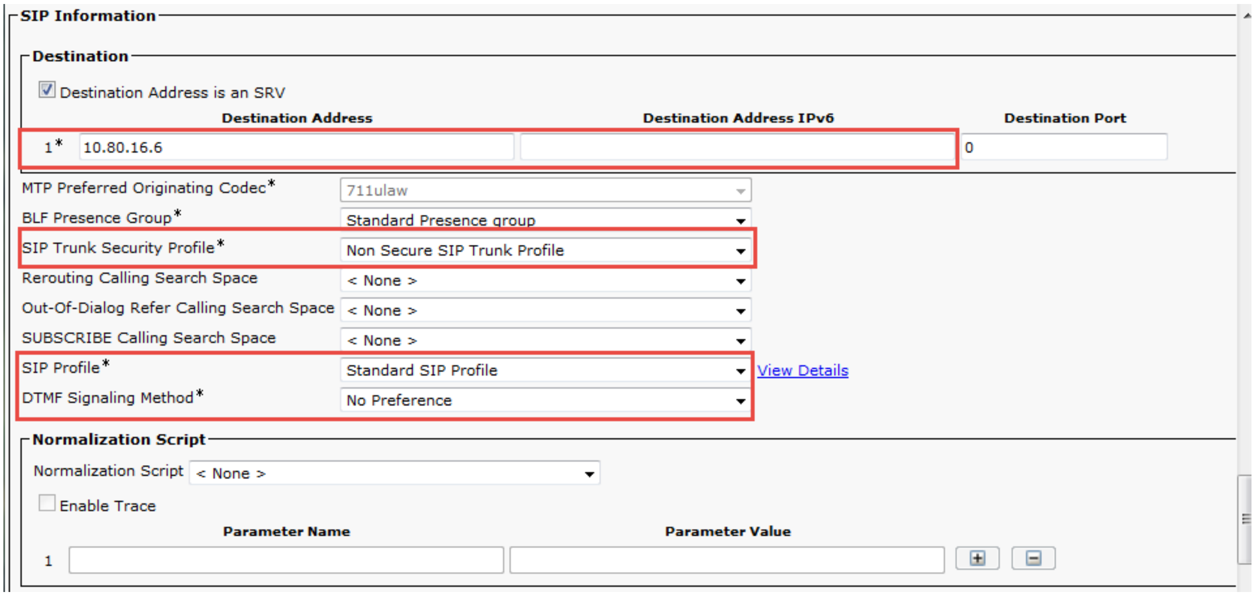
Cisco UCM SIP Trunk to CUP Configuration (Continued)

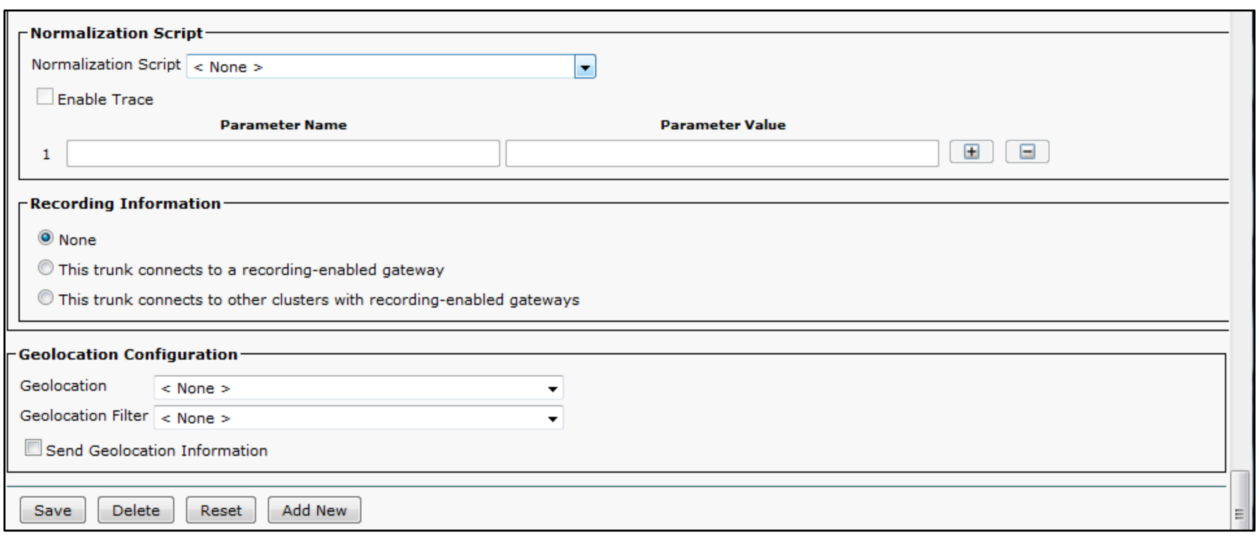




Cisco UCM SIP Trunk to CUP Configuration (Continued)

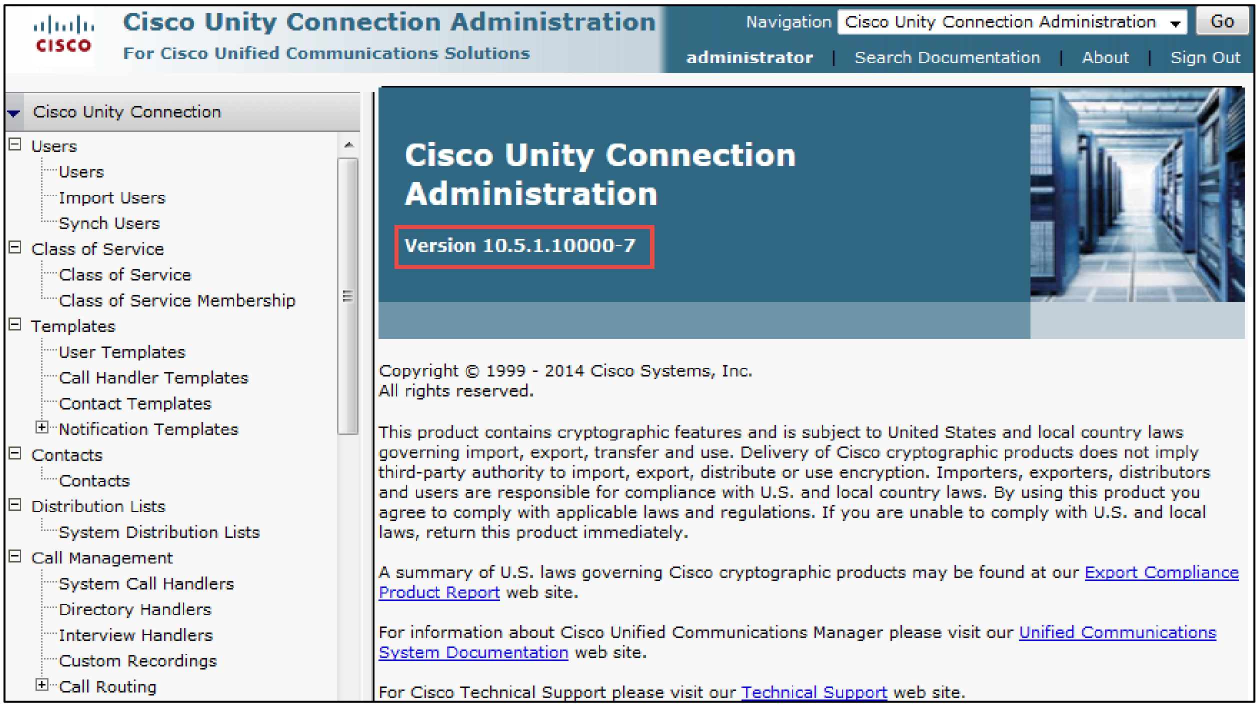
Set Destination Address = 10.80.16.6. This is used in this example.
Set SIP Trunk Security Profile*= Non Secure SIP Trunk Profile.
Set SIP Profile*= Standard SIP Profile.
Set DTMF Signaling Method*= No Preference.
All other values are default.





Cisco UCM SCCP Integration with Cisco Unity Connection (CUC)

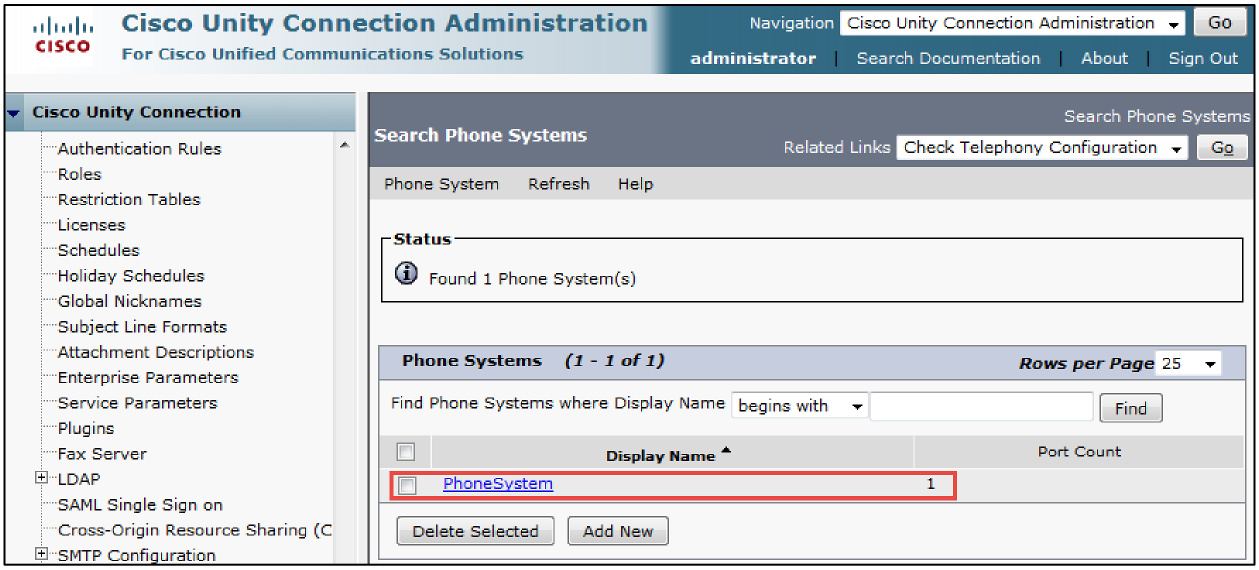
CUC Version



CUC Telephony Integration with Cisco UCM

Navigation: Telephony Integrations → Phone system

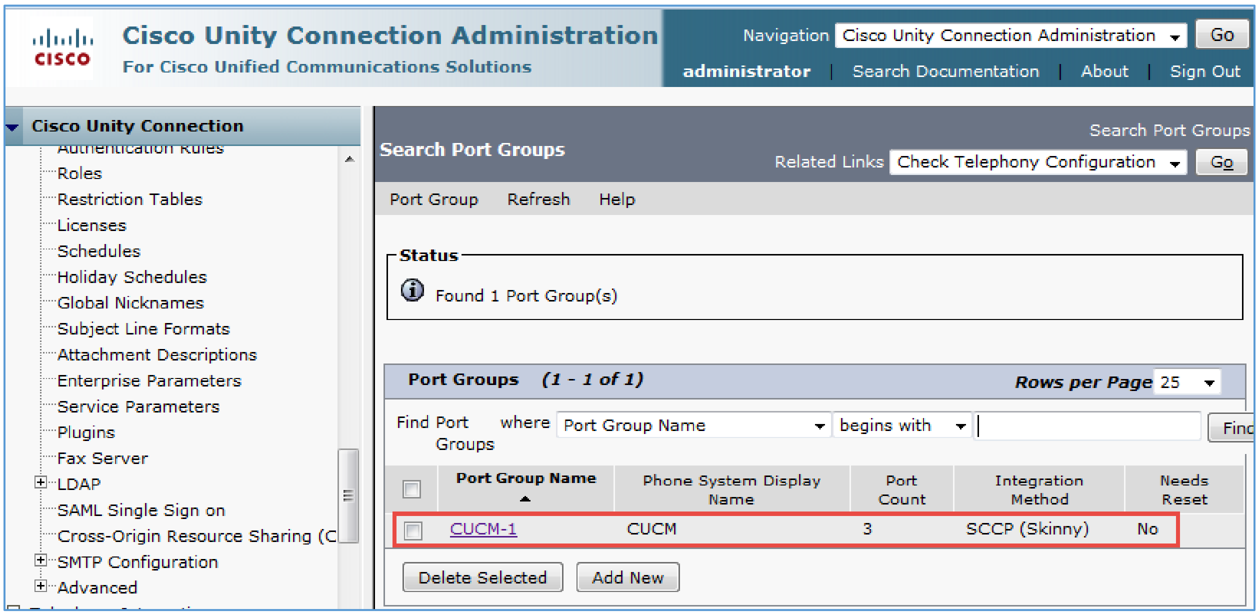
Phone System Name* = PhoneSystem. This is used for this example



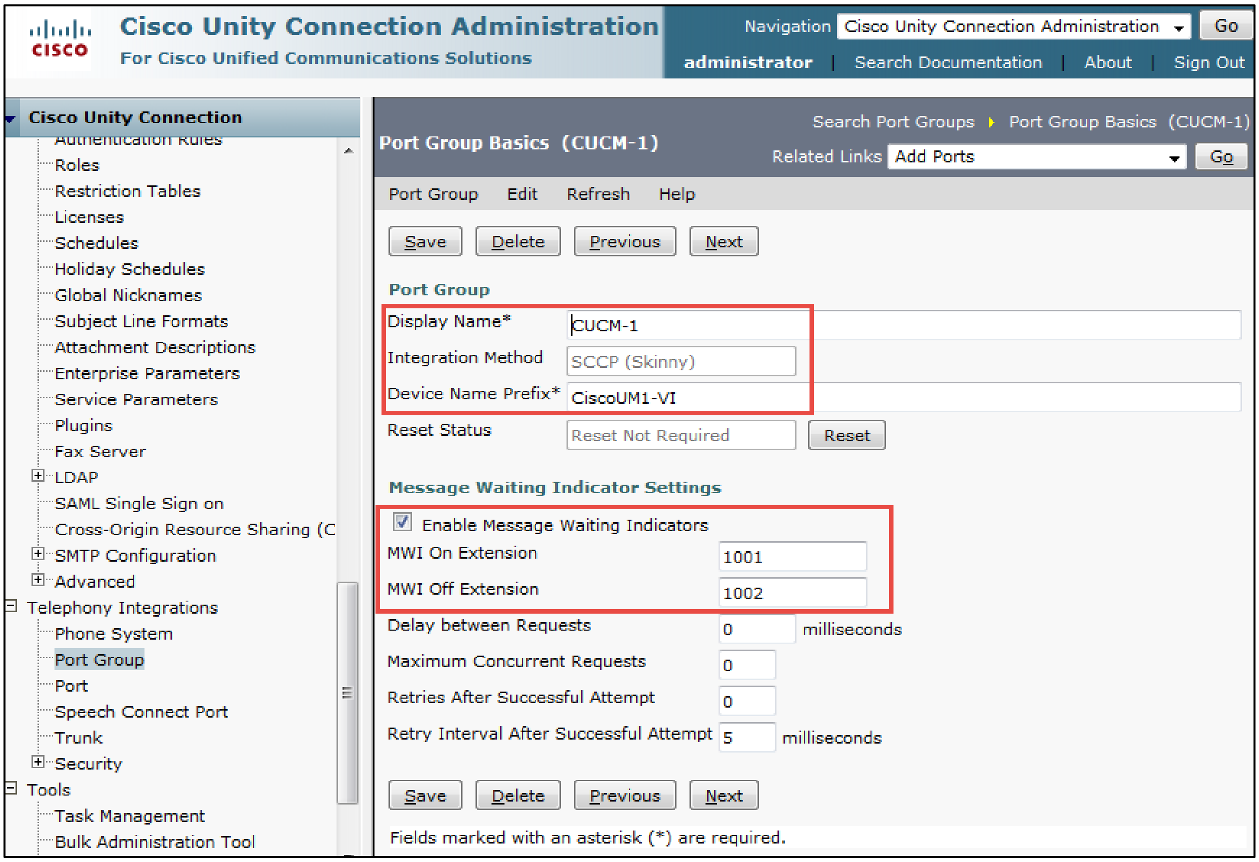
CUC Port Group

Navigation: Telephony Integration → Port Group

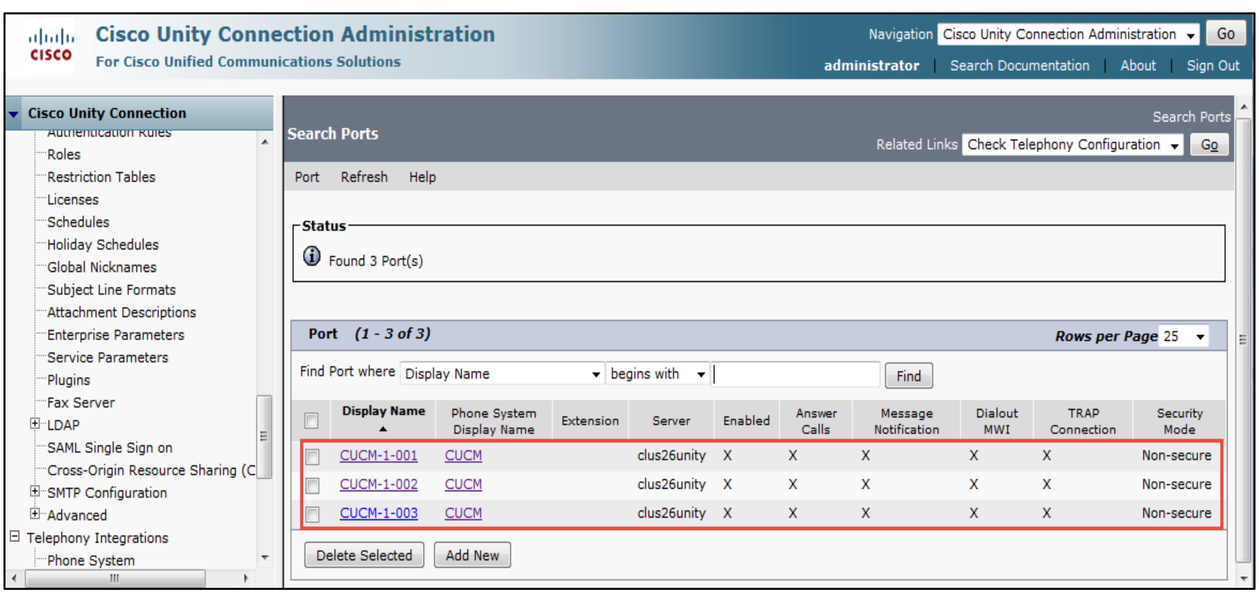
Set Display Name* = CUCM-1. This is used in this example.
Check Enable Message waiting indicators.
Set MWI on Extension = 1001. This is used in this example.
Set MWI off Extension= 1002. This is used in this example.



Cisco Unified Connections Port Configuration (Continued)



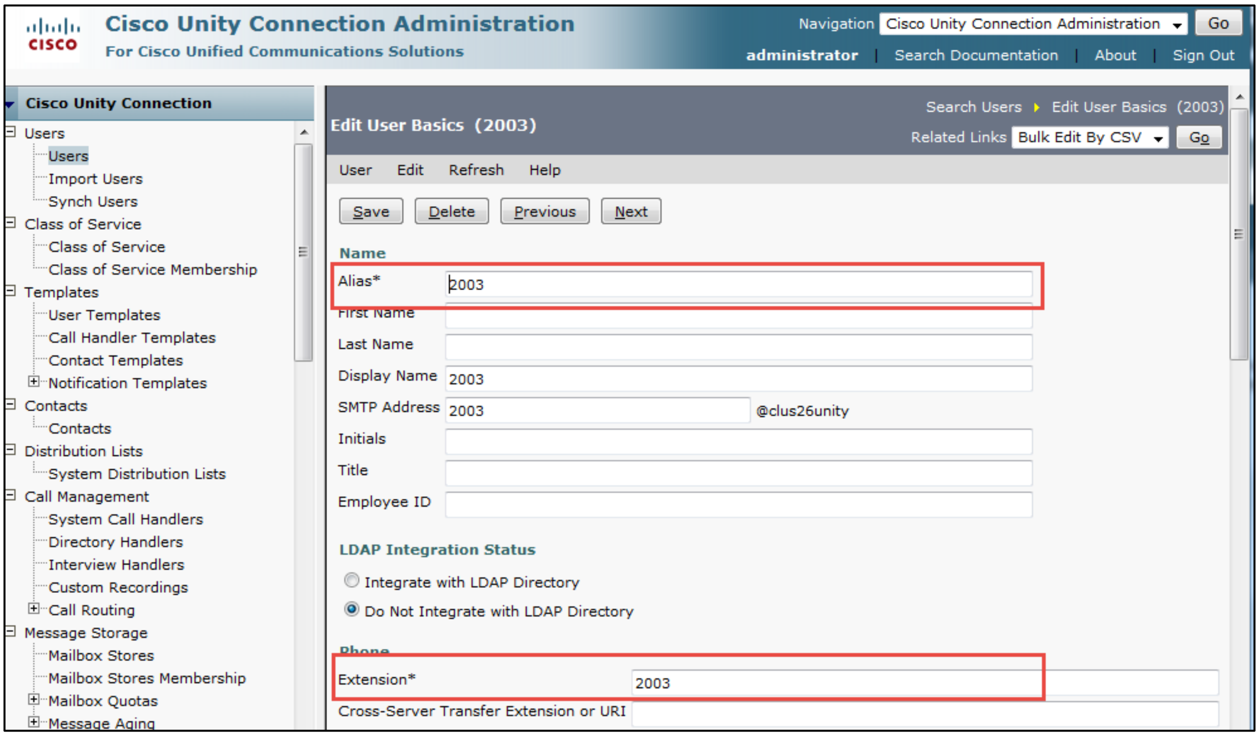
CUC Port Settings



CUC Sample User Basic Settings

Navigation: Cisco Unity connection → Users → Users

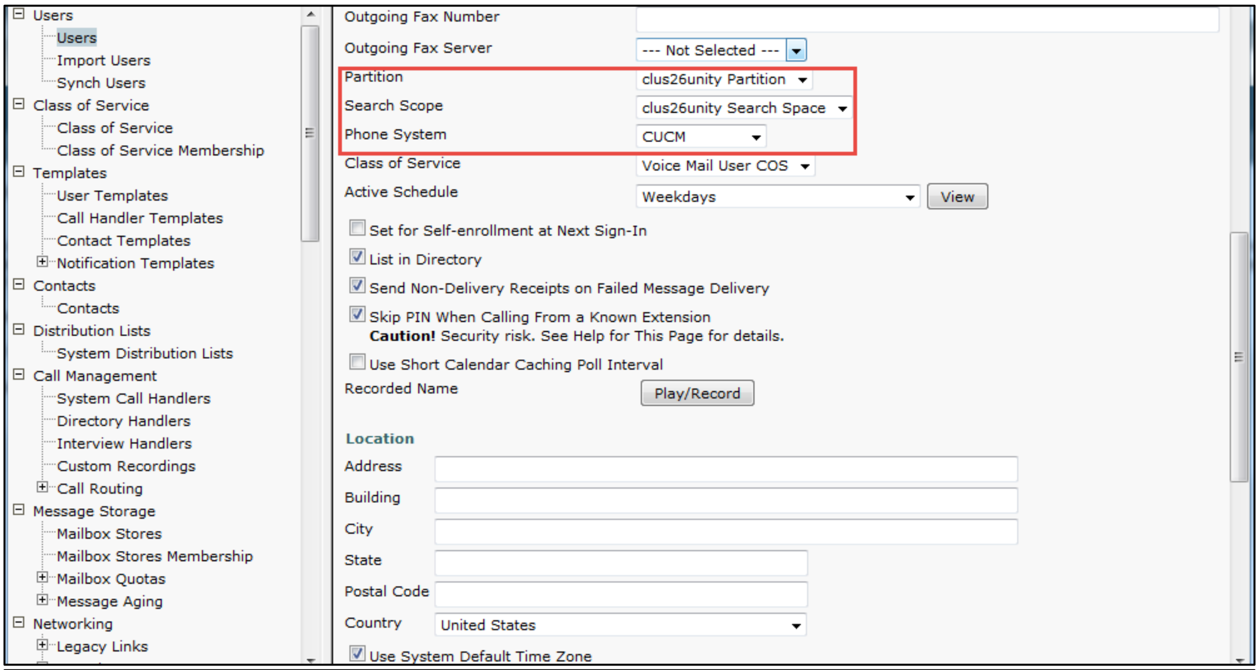
Set Alias = 2003.This is one of the extension used for this testing.
Set Extension = 2003. This is used for this example.

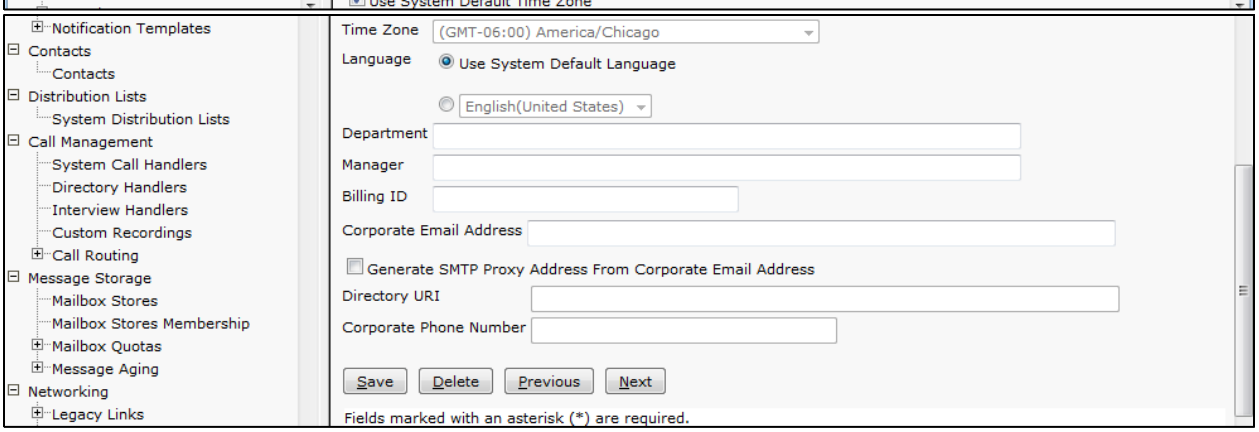


Note: Need to configure Alternate extension for Cisco Extend and connect enabled remote destination DN to retrieve successful mail access from Avaya CS1000E.

CUC Sample User Basic Settings (Contd…)

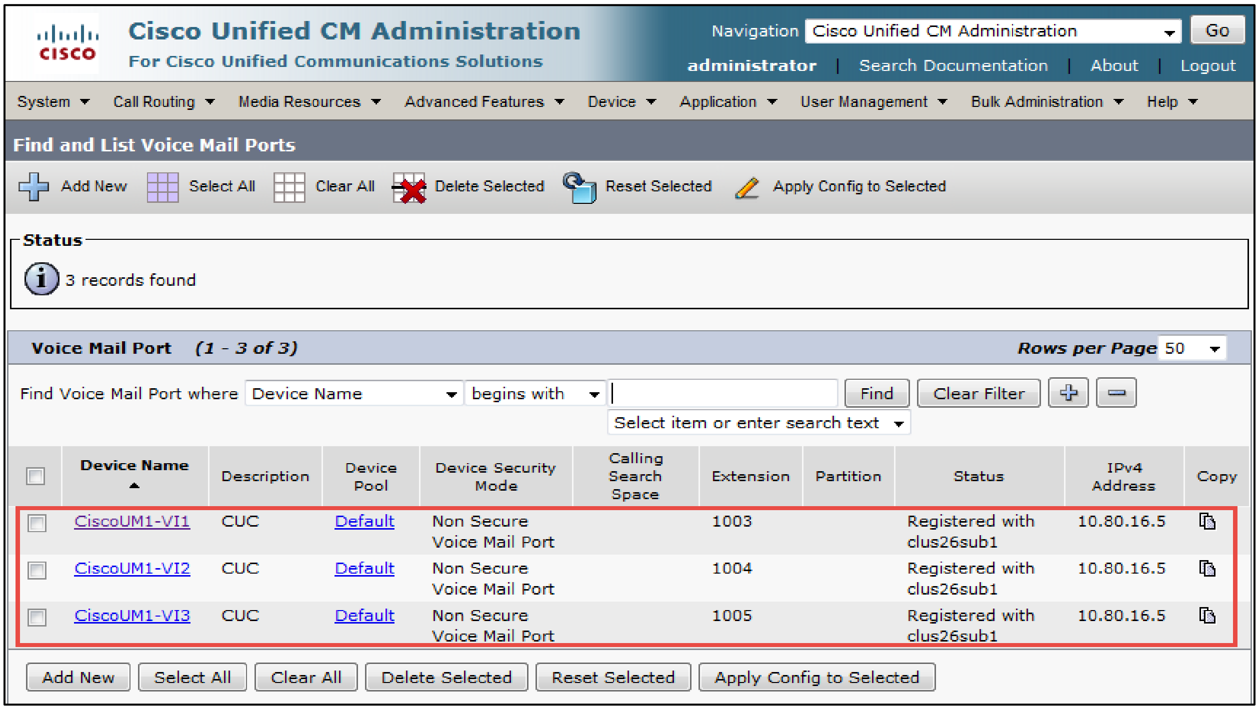
Set Partition = clus23-unity partition. This is used for this example.
Select Search Scope = clus23-unity Search Scope.
Select Phone System = CUCM.

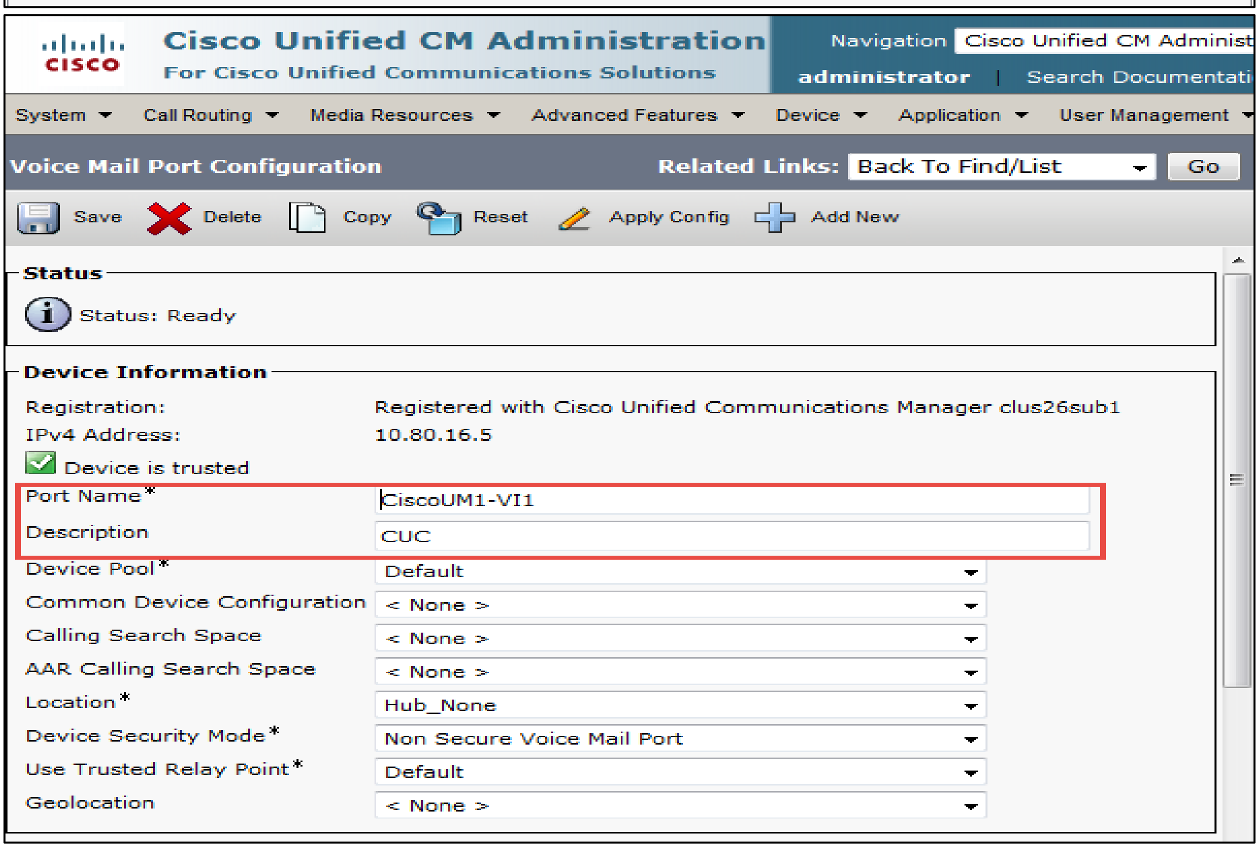




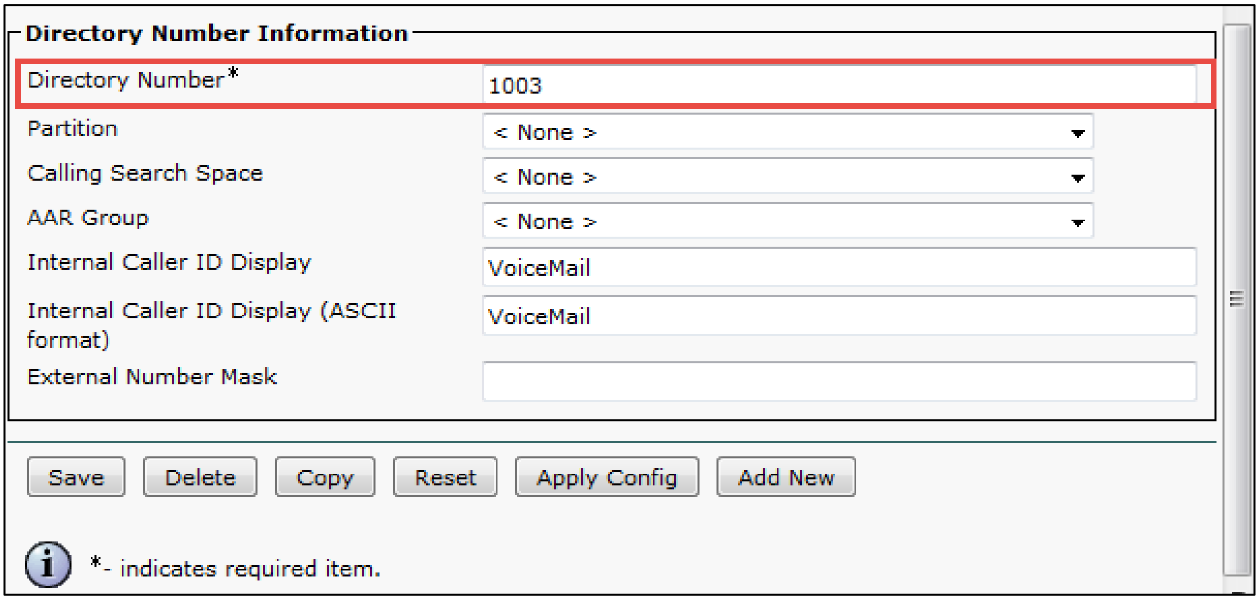
Cisco UCM Voice Mail Port

Navigation: Advanced Feature → Voice Mail → Voice Mail Port



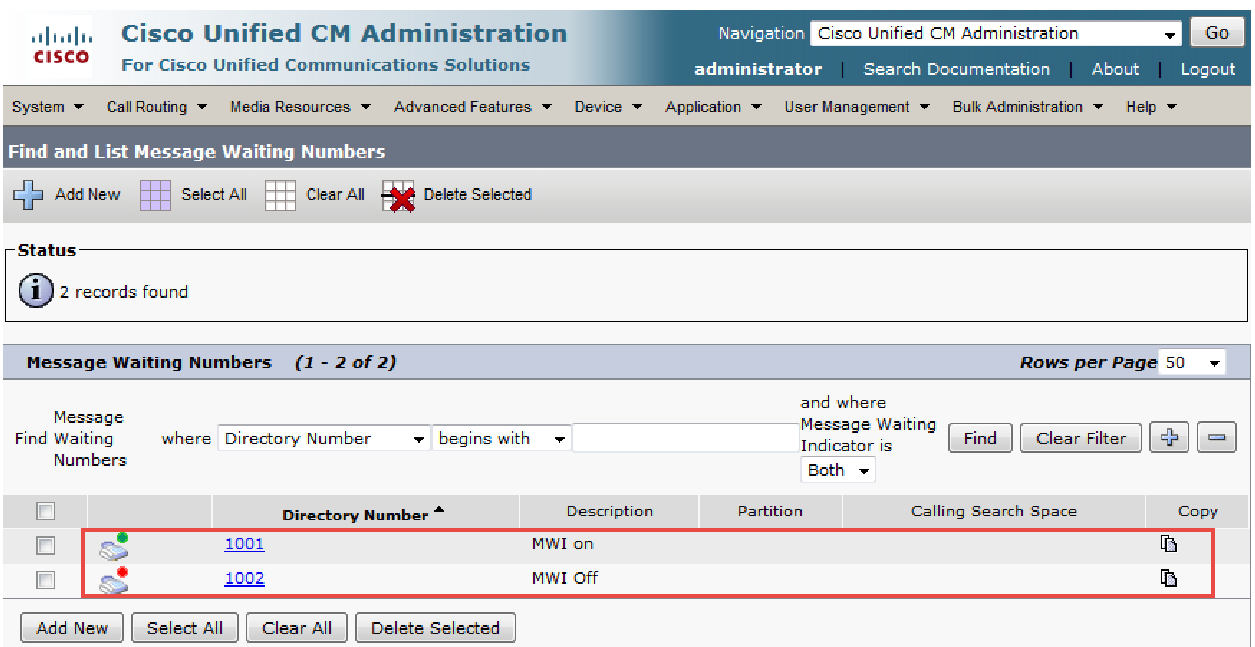


Cisco Unified Connections Voice mail port configuration Settings (Contd…)



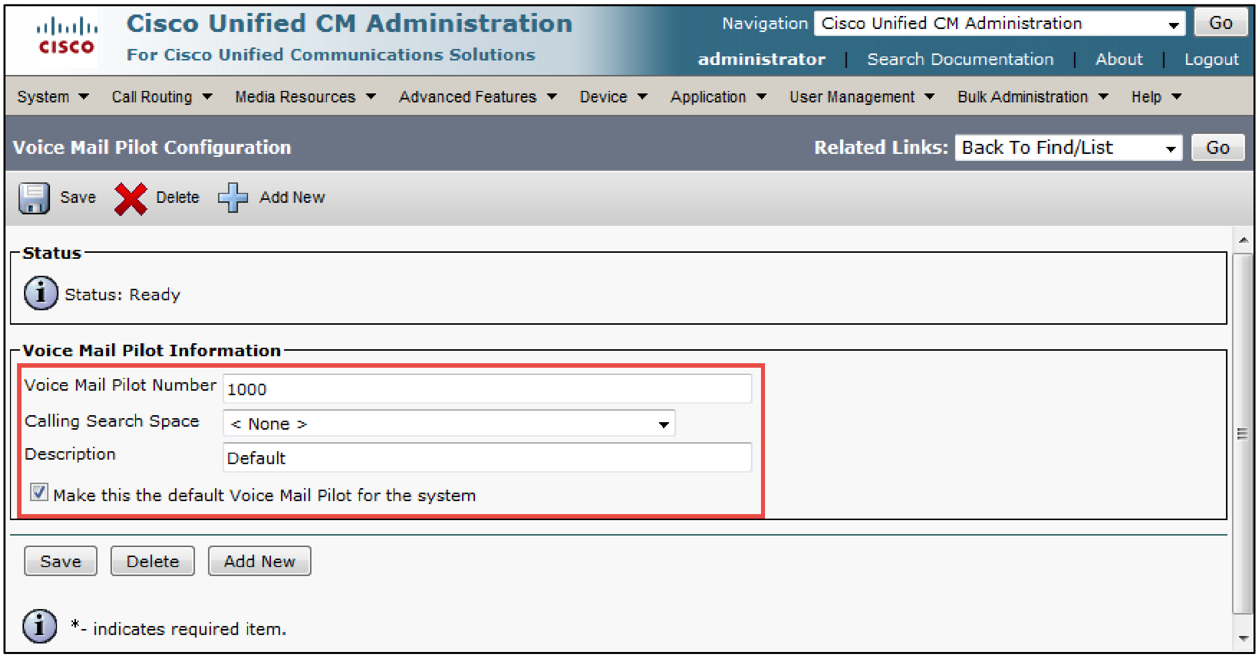
Cisco UCM Message Waiting Numbers

Navigation: Advanced Feature → Voice Mail → Messaged Waiting



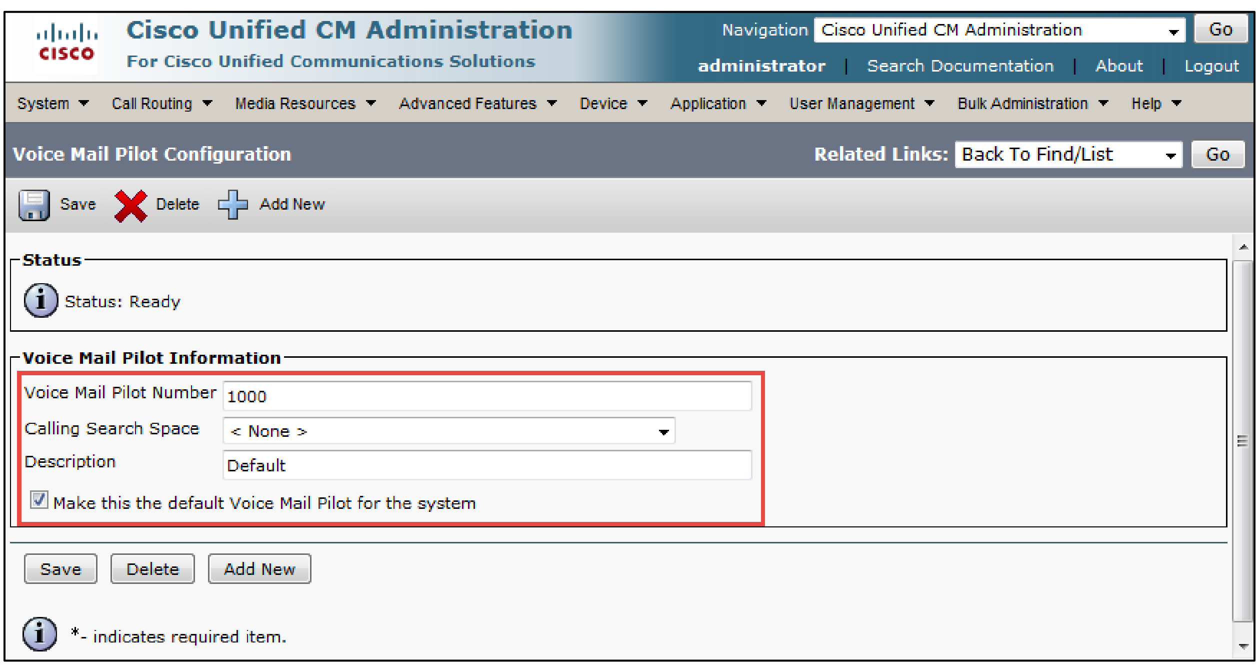
Cisco UCM Voice Mail Pilot

Navigation: Advanced Feature → Voice Mail → Voice Mail Pilot



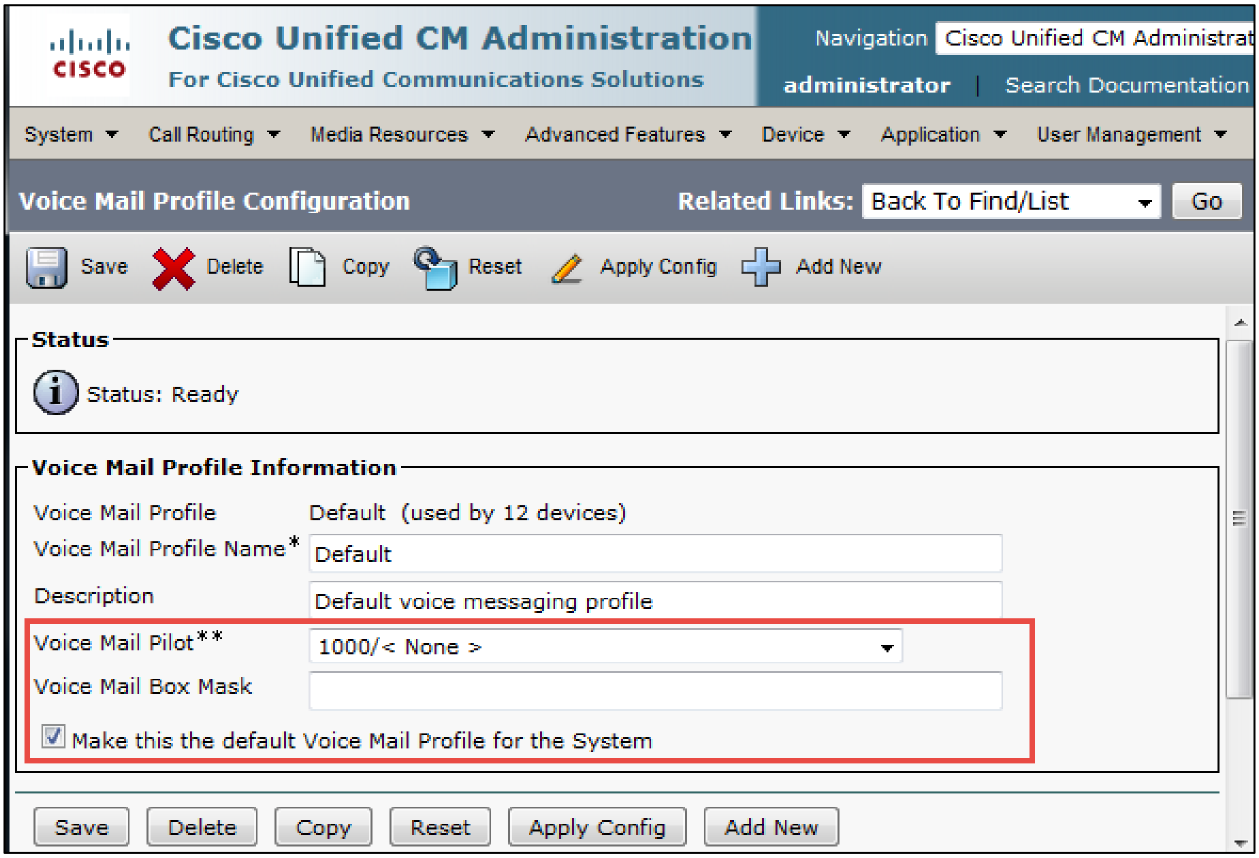
Cisco UCM Voice Mail Pilot

Navigation: Advanced Feature → Voice Mail → Voice Mail Pilot



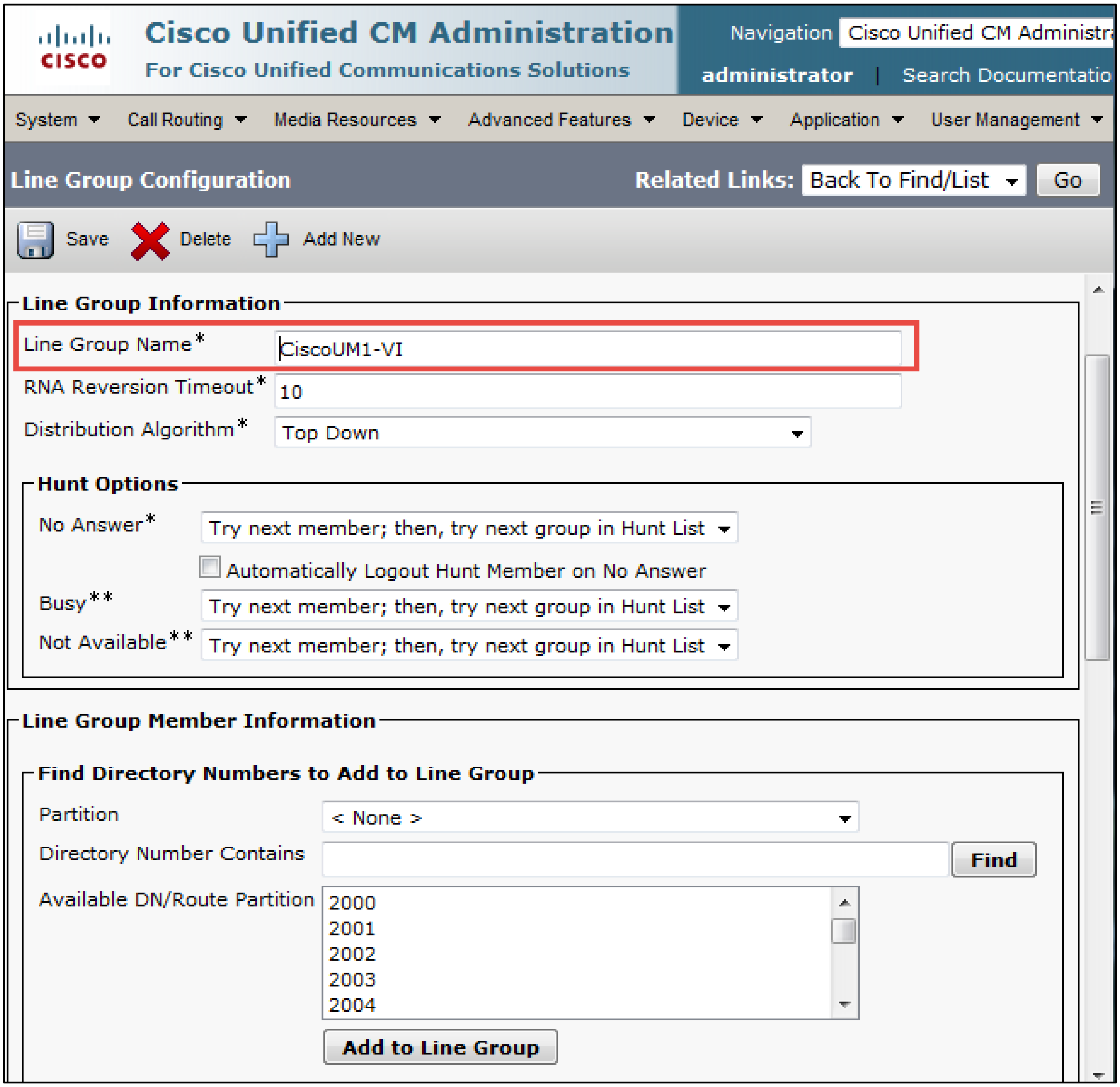
Cisco UCM Voice Mail Profile

Navigation: Advanced Feature → Voice Mail → Voice Mail Pilot

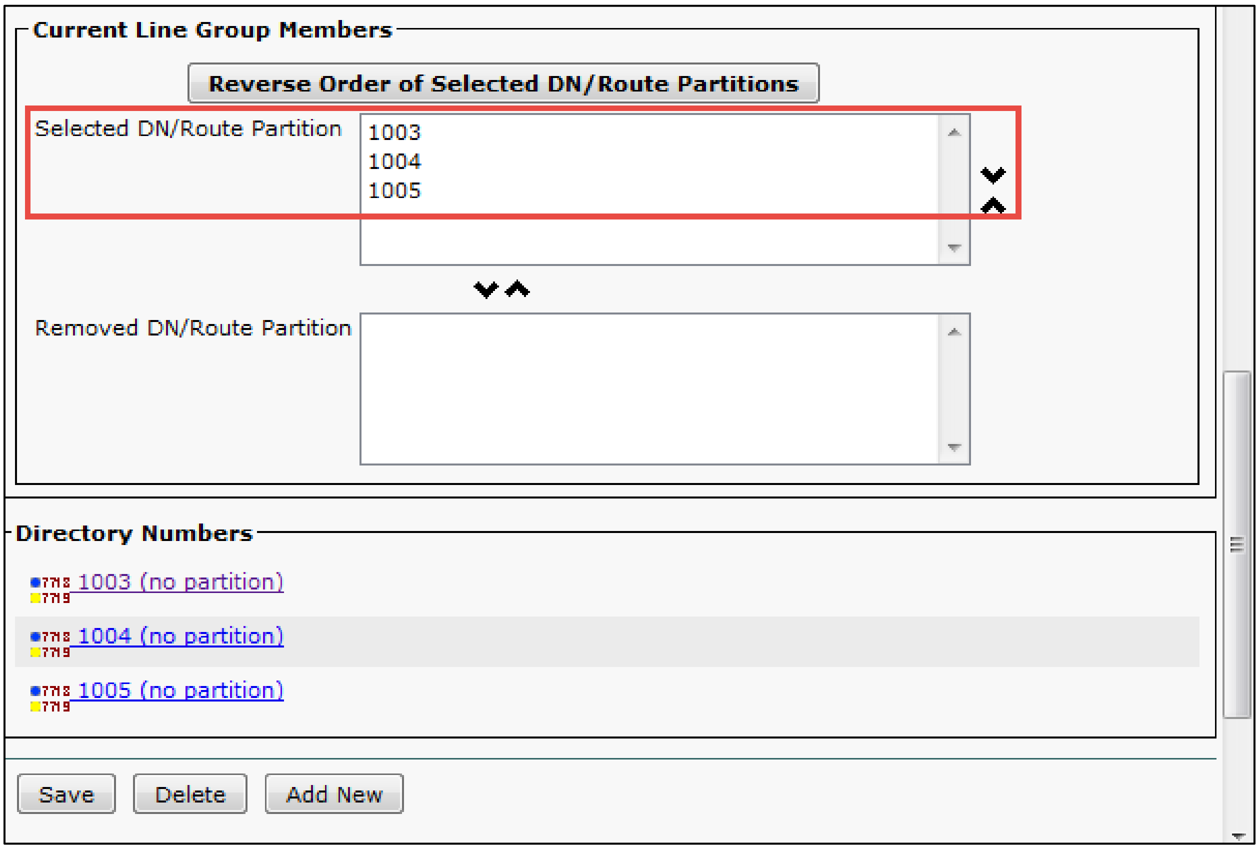


Cisco UCM Line Group

Navigation: Call Routing → Route/Hunt → Line Group

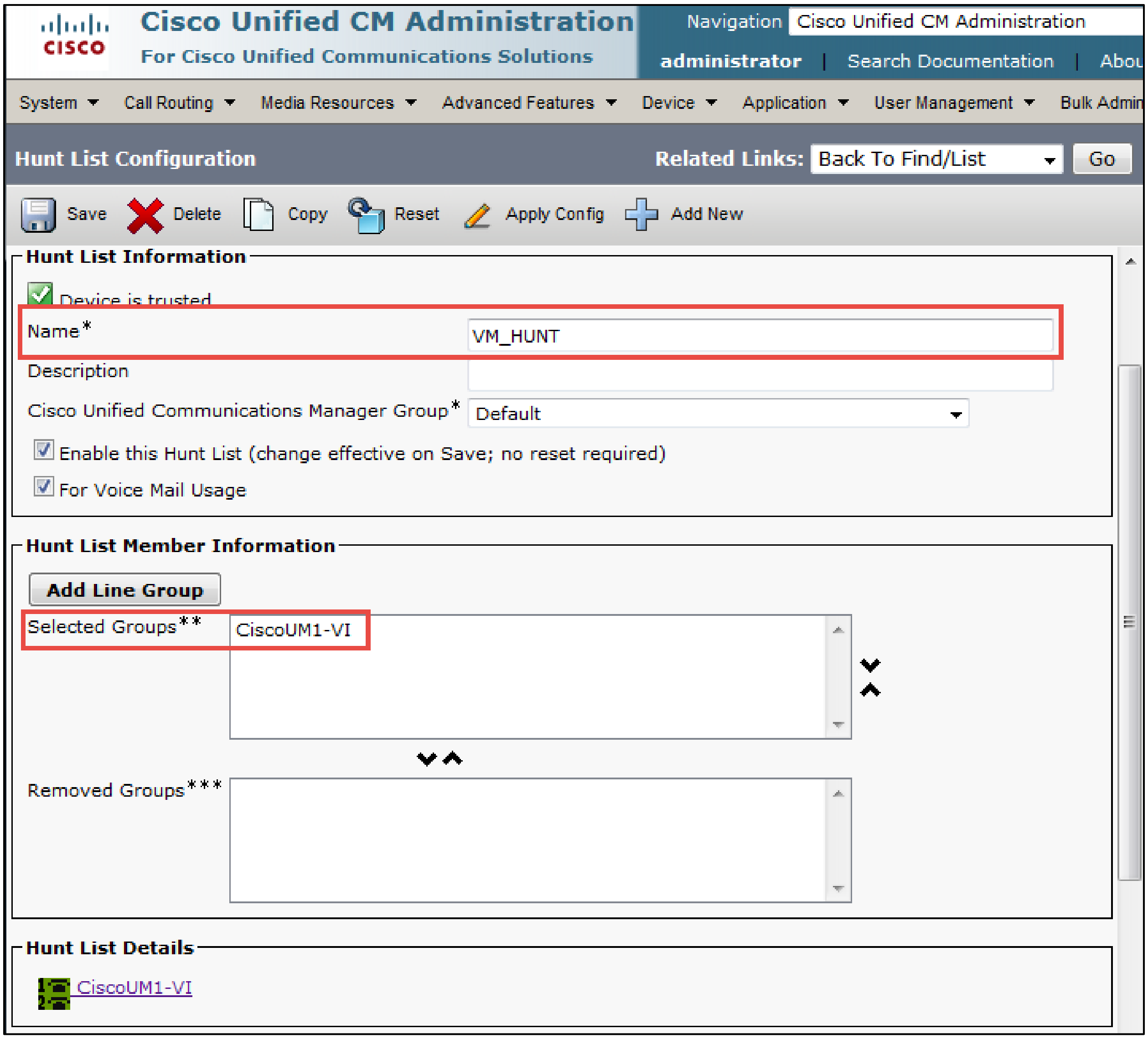


Cisco Unified Connections Line Group Member configuration (Contd…)



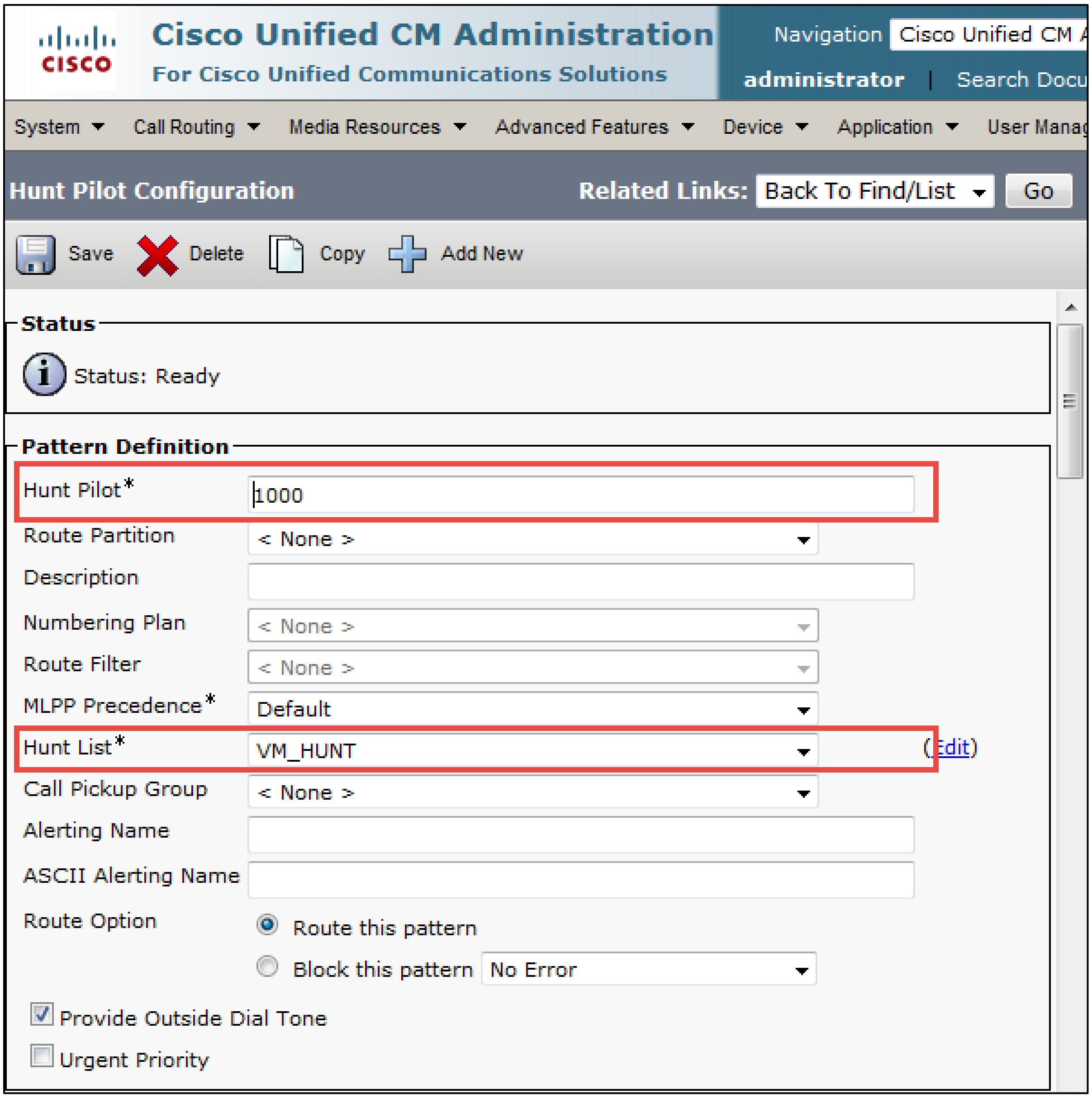
Cisco UCM Hunt List

Navigation: Call Routing → Route/Hunt → Hunt List



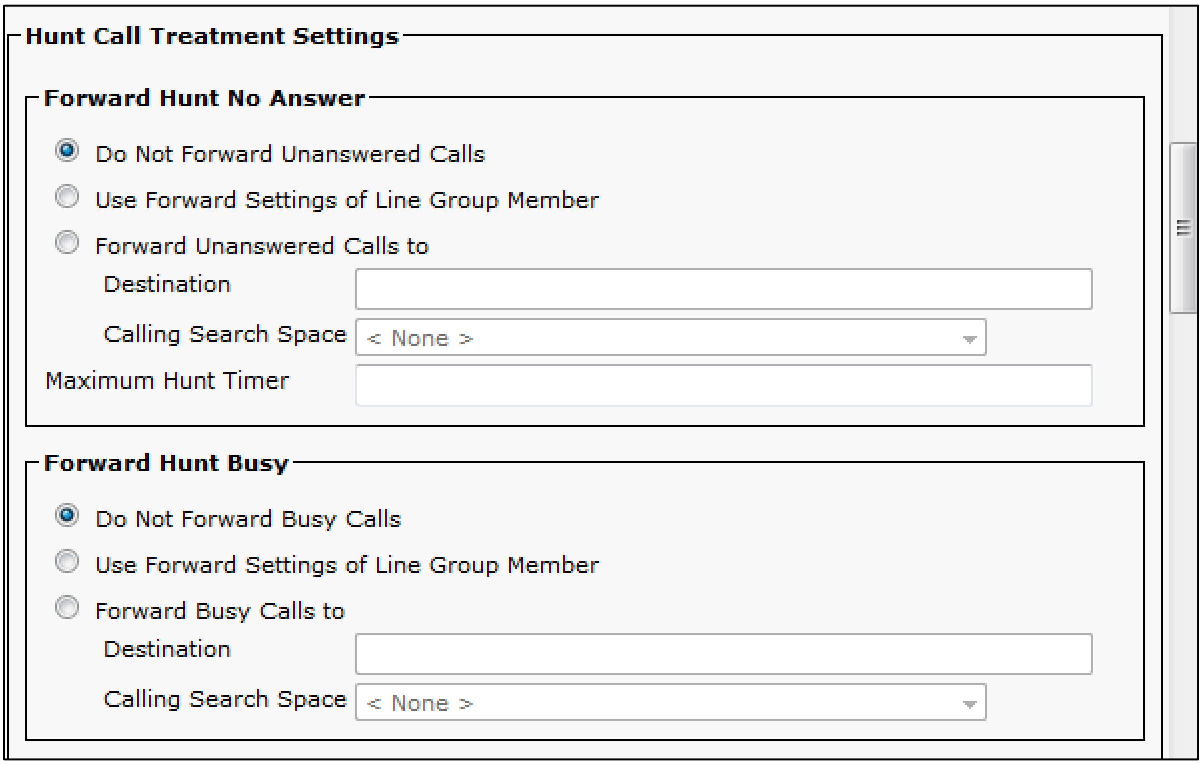
Cisco UCM Hunt Pilot

Navigation: Call Routing → Route/Hunt → Hunt Pilot



Cisco Unified Hunt Pilot configuration (Contd…)

All Other Values are Default



Acronyms

Acronym	Definition
CCBS	Call Completion to Busy Subscriber
CCNR	Call Completion on No Reply
CFB	Call Forwarding on Busy
CFNR	Call Forwarding No Reply
CFU	Call Forwarding Unconditional
CLIP	Calling Line (Number) Identification Presentation
CLIR	Calling Line (Number) Identification Restriction
CNIP	Calling Name Identification Presentation
CNIR	Calling Name Identification Restriction
COLP	Connected Line (Number) Identification Presentation
COLR	Connected Line (Number) Identification Restriction
CONP	Connected Name Identification Presentation
CONR	Connected Name Identification Restriction
CT	Call Transfer
Cisco UCM	Cisco Unified Communications Manager
CUP	Cisco Unified IM and Presence Server
DNS	Domain Name Server
FQDN	Fully Qualified Domain Name
MWI	Message Waiting Indicator
MRGL	Media Resource Group List
MTP	Media Termination Point
PSTN	Public Switched Telephone Network
SIP	Session Initiated Protocol

Important Information

THE SPECIFICATIONS AND INFORMATION REGARDING THE PRODUCTS IN THIS MANUAL ARE SUBJECT TO CHANGE WITHOUT NOTICE. ALL STATEMENTS, INFORMATION, AND