Application Note

Nortel CS1000E Release 7.0 using SIP trunk to Cisco Unified Communications Manager Release 9.0

November 27, 2012 – Initial revision

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Introduction

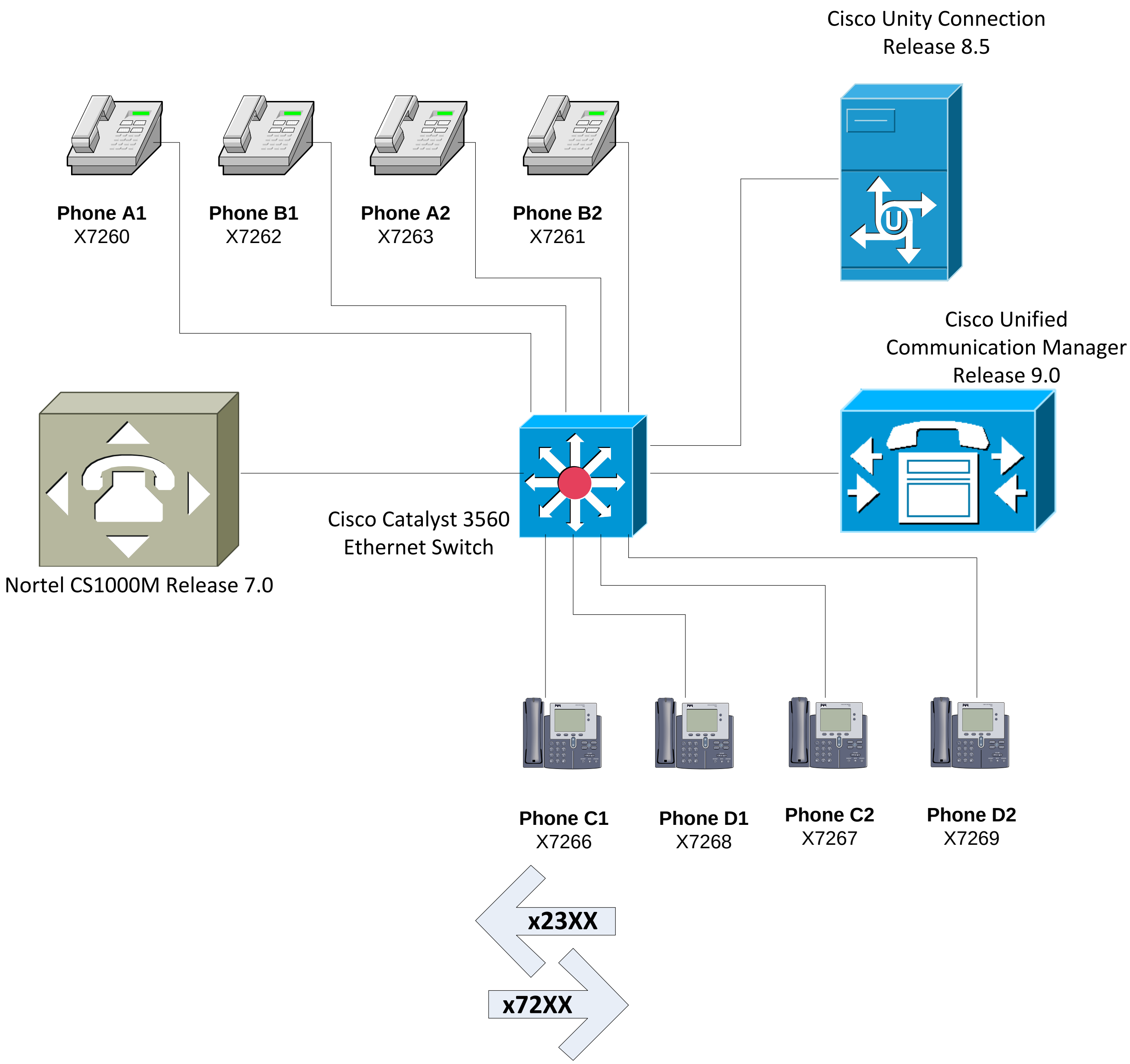
This document describes the steps and configurations necessary for Cisco Unified Communications Manager (Cisco UCM) release 9.0 to interoperate with the Nortel CS1000E using SIP Early-Offer.

The following items were tested:

- Basic call between the two systems and verification of voice path, using both IP phones and digital phones on the Nortel side, and 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.
- Call forwarding: call forward all (CFA), 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.
- DTMF-relay via RFC2833.

Network Topology

Figure 1. Network Topology/Test bed Setup



Caveats and Limitations

These are the known caveats, limitations or integration issues:

- Basic calls worked from Cisco UCM to Nortel CS1000E and vice versa. The Nortel CS1000E only supports early offer to set its media attribute to send/receive mode. Thus, for calls from Cisco UCM to Nortel 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 Nortel CS1000 Unistm phones is achieved by configuring the Nortel 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.
- COLR/CONR - as with calling name and number presentation restrictions, the Nortel CS1000 restricts connected name and number by configuring the Nortel station's class of service. The station CLS is set to DDGD and NAMD to restrict connected number and name respectively. When a Nortel 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. Thus, Cisco UCM does not honor the restriction of both connected name and number intended by the Nortel side. On the other hand, Cisco UCM restricts the connected name and number information it sends in its SIP response by setting the SIP PRIVACY to "id" on the SIP trunk configuration page. With this setting, the SIP P-Asserted Identity setting within the SIP Response message back to Nortel has the privacy set to "id" only. This results in the Nortel phone restricting the presentation of the connected number only. The connected name is still presented.
- Alerting Name – Although the Cisco UCM sends P-Asserted Identity (PAI) header with the alerting name, this information is not displayed by the Nortel phone.
- 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 a Nortel phone is the forwarding or transferring party to a Cisco phone via the SIP trunk.
- The Nortel 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.
- For integration where Cisco Unity is the centralized voice messaging system, a SIP normalization script is required to enable/disable MWI on Nortel phones. Please refer to the configuration section of this document for more details on the actual normalization script used for this testing.
- When using G.729 codec between Nortel CS1000 and Cisco UCM, it is required to configure a conference bridge (CFB) resource to initiate a three-way conference between G729 media end-points. Please refer to Cisco UCM configuration section for details.

System Components

Hardware Requirements

The following hardware is required:

- Cisco Unified Communications Manager (Cisco UCM) MCS server. MCS-7825-I4 was used for this testing.
- Cisco Unity Connection MCS server. MCS-7835 server was used for this testing
- Catalyst switch. Catalyst 3560 was used for this testing
- Cisco IP phones. Cisco 9971, 7960, 7962, 7971 & 7975 were used for this testing
- Nortel CS1000E
- Nortel IP and Digital station phones. Nortel-2004P2 IP Phones and M2616 series digital station phones were used for this testing.

Software Requirements

The following software is required:

- Cisco Unified Communications Manager release 9.0. (9.0.1.10000-37 was used for this testing)
- Catalyst 3560 release: 12.2(35)SE5
- Cisco Unity Connection release 8.5
- Nortel CS1000E release 7.0

Features

This section lists supported and unsupported features. Please see the Limitations section on page 4 for more information.

Features Supported

- CLIP—calling line (number) identification presentation.
- CLIR—calling line (number) identification restriction. (Refer to Caveats and Limitations Section)
- CNIP—calling name identification presentation.
- CNIR—calling name identification restriction. (Refer to Caveats and Limitations Section)
- Alerting name. (Refer to Caveats and Limitations Section)
- Attended call transfer. (Refer to Caveats and Limitations Section)
- Early attended call transfer. (Refer to Caveats and Limitations Section)
- CFU—call forwarding unconditional. (Refer to Caveats and Limitations Section)
- CFB—call forwarding busy. (Refer to Caveats and Limitations Section)
- CFNA—call forwarding no answer. (Refer to Caveats and Limitations Section)
- COLP—connected line (number) identification presentation.
- COLR—connected line (number) identification restriction. (Refer to Caveats and Limitations Section)
- CONP—connected name identification presentation.
- CONR—connected name identification restriction. (Refer to Caveats and Limitations Section)
- Hold and resume.
- Conference call. (Refer to Caveats and Limitations Section)
- DTMF-relay using RFC2833.

Features Not Supported

- Call completion (callback, automatic callback).

Configuration

Configuration Sequence and Tasks

Configuring the Nortel PBX

Configuring the Nortel Communication Server 1000 PBX

1. LD 17—Configure the IP D-channel (signaling channel) between the call server and the signaling server.
2. LD 14—Configure the SIP virtual trunks to the signal.
3. LD 11—Configure for the virtual lines for the Nortel IP phone (i200x series).
4. LD 16—Configure the SIP route.
5. LD 86—Configure the route list block for the virtual trunk route.
6. LD 87—Configure CDP steering codes.
7. LD 21—List trunk member.

Signaling Server Setup via the Nortel Node Summary
1. Configure a new IP telephony node summary.
2. Configure the VGW and IP phone codec profile section.
3. Configure the quality of service (QoS) section.
4. Configure the LAN configuration section.
5. Configure the SIP GW setting section.
6. Configure the SIP URI map section.
7. Configure the card section for the MC-32 VGMC card section.
8. Configure the signaling server 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 (Cisco UCM):

1. Cisco UCM software release.
2. Cisco UCM Regional configuration.
3. Cisco UCM Device Pool configuration.
4. Media Resource Group and Media Resource Group List.
5. Cisco UCM Media Termination Point.
6. SIP trunk security profile.
7. Device setting SIP profile.
8. SIP trunk to the Nortel CS1000E PBX.
9. SIP and SCCP phones device configuration.
10. Route pattern to the Nortel CS1000E PBX.
11. Cisco UCM Service Parameter “Duplex Streaming Enabled” set to “True”.
12. SIP Trunk Normalization Script

LD 17 – Configure the IP D-channel (signaling channel) between the Call Server and the Signaling Server

```
>LD 22
PT2000
REQ PRT
TYPE ADAN DCH 30
ADAN DCH 30
CTYP DCIP
DES SIP_Trunk
USR ISLD
ISLM 4000
SSRC 1800
OTBF 32
NASA YES
IFC SL1
CNEG 1
RLS ID 4
RCAP MWI ND3 CPK
MBGA NO
H323
OVLR NO
OVLS NO
```

LD 14 – Configure the SIP Virtual Trunks to the Signaling Server (One trunk = one line connection)

```
>LD 20
PT0000
REQ: PRT
TYPE: TNB
TN 020 0 3 7
DATE
PAGE
DES
DES SIP_TRUNK
TN 020 0 03 07 VIRTUAL
TYPE IPTI
CDEN 8D
CUST 0
XTRK VTRK
ZONE 000
LDOP BOP
TIMP 600
BIMP 600
AUTO_BIMP NO
NMUS NO
TRK ANLG
NCOS 0
RTMB 30 8
CHID 8
TGAR 1
STRI/STRO IMM IMM
SUPN YES
AST NO
IAPG 0
CLS CTD DTN CND ECD WTA LPR APN THFD SPCD MSBT
P10 NTC MID
TKID
AACR NO
```

LD 11 – Configure for the Virtual lines for the Nortel IP phones (phone A and phone B)
Phone A1 (i2004)

```
>ld 11
SL1000
MEM AVAIL: (U/P): 99165853 USED U P: 5049370 54598 TOT: 104269821
DISK SPACE NEEDED: 47 KBYTES
DIGITAL TELEPHONES AVAIL: 3 USED: 5 TOT: 8
IP USERS AVAIL: 18 USED: 6 TOT: 24
BASIC IP USERS AVAIL: 7 USED: 1 TOT: 8
TEMPORARY IP USERS AVAIL: 0 USED: 0 TOT: 0
```

```
ACD AGENTS AVAIL: 10 USED: 0 TOT: 10
PCA AVAIL: 4 USED: 1 TOT: 5
AST AVAIL: 4 USED: 1 TOT: 5
SIP CONVERGED DESKTOPS AVAIL: 0 USED: 0 TOT: 0
SIP CTI TR87 AVAIL: 4 USED: 1 TOT: 5
TNS AVAIL: 32546 USED: 214 TOT: 32760
DATA PORTS AVAIL: 32760 USED: 0 TOT: 32760
REQ: prt
TYPE: tnb
TN 020 0 0 03
DATE
PAGE
DES
DES Phone A1
TN 020 0 00 03 VIRTUAL
TYPE 2004P2
CDEN 8D
CTYP XDLC
CUST 0
NUID
NHTN
CFG_ZONE 000
CUR_ZONE 000
ERL 0
ECL 0
FDN 2327
TGAR 0
LDN NO
NCOS 0
SGRP 0
RNPG 0
SCI 0
SSU
XLST
CLS CTD FBA WTA LPR MTD FNA HTA TDD HFD CRPD
MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
POD DSX VMD SLKD CCSD SWD LND CNDD
CFTD 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 FLXD FTTC DNDY DNO3 MCBN
VOLA VOUD CDMR ICRD MCDD T87A KEM2 MSNV FRA PKCH
CPND_LANG ENG
RCO 0
HUNT 2327
LHK 0
PLEV 02
DANI NO
AST 00
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY 00 SCR 2326 0 MARP
CPND
NAME Phone_A1
XPLN 13
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 2500
20 RGA
21 PRK
22 RNP
23
24 PRS
25 CHG
26 CPN
27
28
29
30
31
DATE 29 SEP 200
```

Phone A2 (2616):

```
REQ: prt
TYPE: 2616
TN 0 0 7 2
DATE
PAGE
DES
DES Phone A2
TN 000 0 07 02 VIRTUAL
TYPE 2616
CDEN 8D
CTYP XDLC
CUST 0
AOM 0
ERL 0
FDN 2328
TGAR 1
LDN NO
NCOS 0
SGRP 0
RNPG 0
SCI 0
SSU
XLST
CLS CTD FBA WTA LPR MTD FNA HTA ADD HFD
MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
POD DSX VMD SLKD CCSD SWD LND CNDD
CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBD
ICDD CDMD LLCN MCTD CLBD AUTU
GPUD DPUD DNDD CFXA ARHD CLTD ASCD
CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
AHD DDGA NAMA
DRDD EXR0
USMD USRD ULAD RTDD RBDD RBHD PGND FLXD FTTC DNDY DNO3 MCBN
CDMR MCDD T87D PKCH
CPND_LANG ENG
RCO 0
HUNT 2326
```

```
LHK 0
PLEV 02
DANI NO
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
DNDR 0
KEY 00 SCR 2332 0 MARP
CPND
NAME Phone_A2
XPLN 13
DISPLAY_FMT FIRST,LAST
01
02
03 CFW 4 2500
04 AO6
05 TRN
06
07
08
09
10
11
12
13
14
15
DATE 29 SEP 2008
NACT
```

LD 16 – Configure the SIP route

```
>LD 21
PT1000
REQ: PRT
TYPE: RDB
CUST 0
ROUT 30
TYPE RDB
CUST 00
ROUT 30
DES SIP_TRUNK
TKTP TIE
M911P NO
ESN NO
CNVT NO
SAT NO
RCLS EXT
VTRK YES
ZONE 000
PCID SIP
CRID YES
NODE 104
DTRK NO
ISDN YES
MODE ISLD
DCH 30
IFC SL1
PNI 00001
NCNA YES
NCRD YES
TRO NO
FALT NO
CTYP UKWN
```

```
INAC NO
ISAR NO
DAPC NO
PTYP ATT
AUTO NO
DNIS NO
DCDR NO
ICOG IAO
SRCH LIN
TRMB YES
STEP
ACOD 430
TCPP NO
TARG 01
CLEN 1
BILN NO
OABS
INST
ANTK
SIGO STD
STYP SDAT
ICIS YES
TIMR ICF 512
OGF 512
EOD 13952
DSI 34944
NRD 10112
DDL 70
ODT 4096
RGV 640
GRD 896
SFB 3
NBS 2048
NBL 4096
IENB 5
TFD 0
PAGE 002
VSS 0
VGD 6
SST 5 0
NEDC ORG
FEDC ORG
CPDC NO
DLTN NO
HOLD 02 02 40
SEIZ 02 02
SVFL 02 02
DRNG NO
CDR NO
VRAT NO
MUS NO
MANO NO
OHQ NO
OHQT 00
CBQ NO
AUTH NO
TDET NO
TTBL 0
ATAN NO
OHTD NO
PLEV 2
ALRM NO
ART 0
SGRP 0
ARDN NO
AACR NO
```

LD 86 – Configure the Route List Block for the Virtual Trunk route

```
>LD 86
ESN000
MEM AVAIL: (U/P): 99165853 USED U P: 5049370 54598 TOT: 104269821
DISK SPACE NEEDED: 47 KBYTES
REQ PRT
CUST 0
FEAT RLB
RLI 3
RLI 3
ENTR 0
LTER NO
ROUT 30
TOD 0 ON 1 ON 2 ON 3 ON
4 ON 5 ON 6 ON 7 ON
VNS NO
SCNV NO
CNV NO
EXP NO
FRL 0
DMI 0
ISDM 0
FCI 0
FSNI 0
DORG NO
SBOC NRR
IDBB DBD
IOHQ NO
OHQ NO
CBQ NO
ISET 0
NALT 5
MFRL 0
OVLL 0
MEM AVAIL: (U/P): 99165853 USED U P: 5049370 54598 TOT: 104269821
DISK SPACE NEEDED: 47 KBYTES
```

LD 87 – Configure CDP steering codes

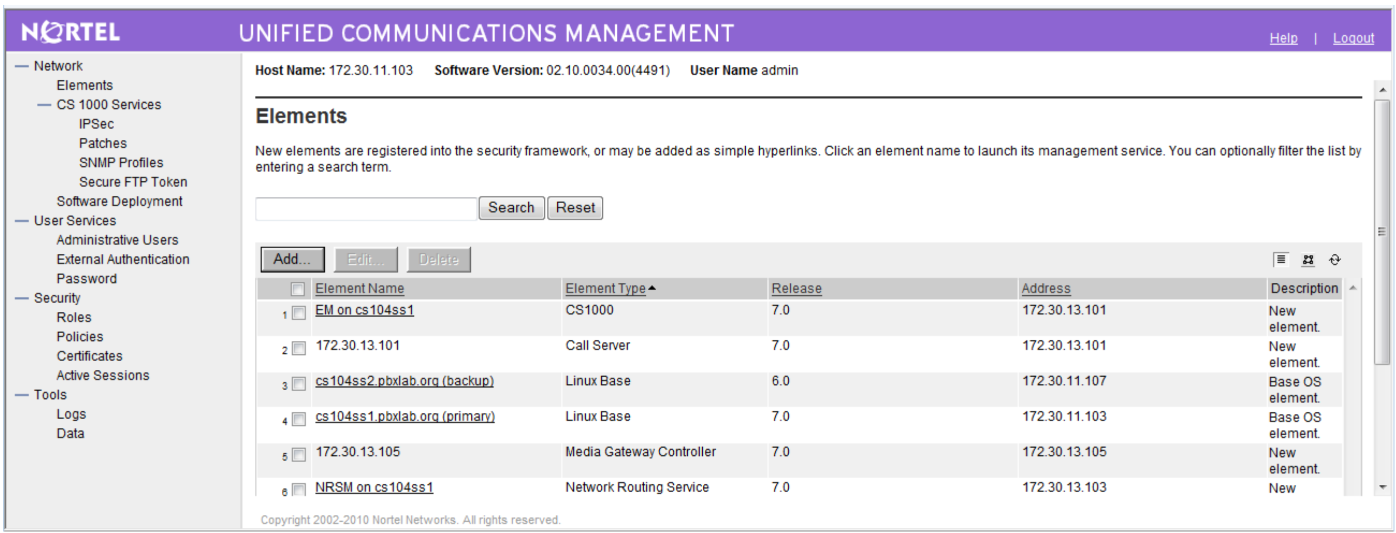
```
>LD 87
ESN000
MEM AVAIL: (U/P): 99165853 USED U P: 5049370 54598 TOT: 104269821
DISK SPACE NEEDED: 47 KBYTES
REQ PRT
CUST 0
FEAT CDP
TYPE DSC
DSC 37
DSC 37
FLEN 0
DSP LSC
RLI 3
NPA
NXX
```

LD 21 – List Trunk Members

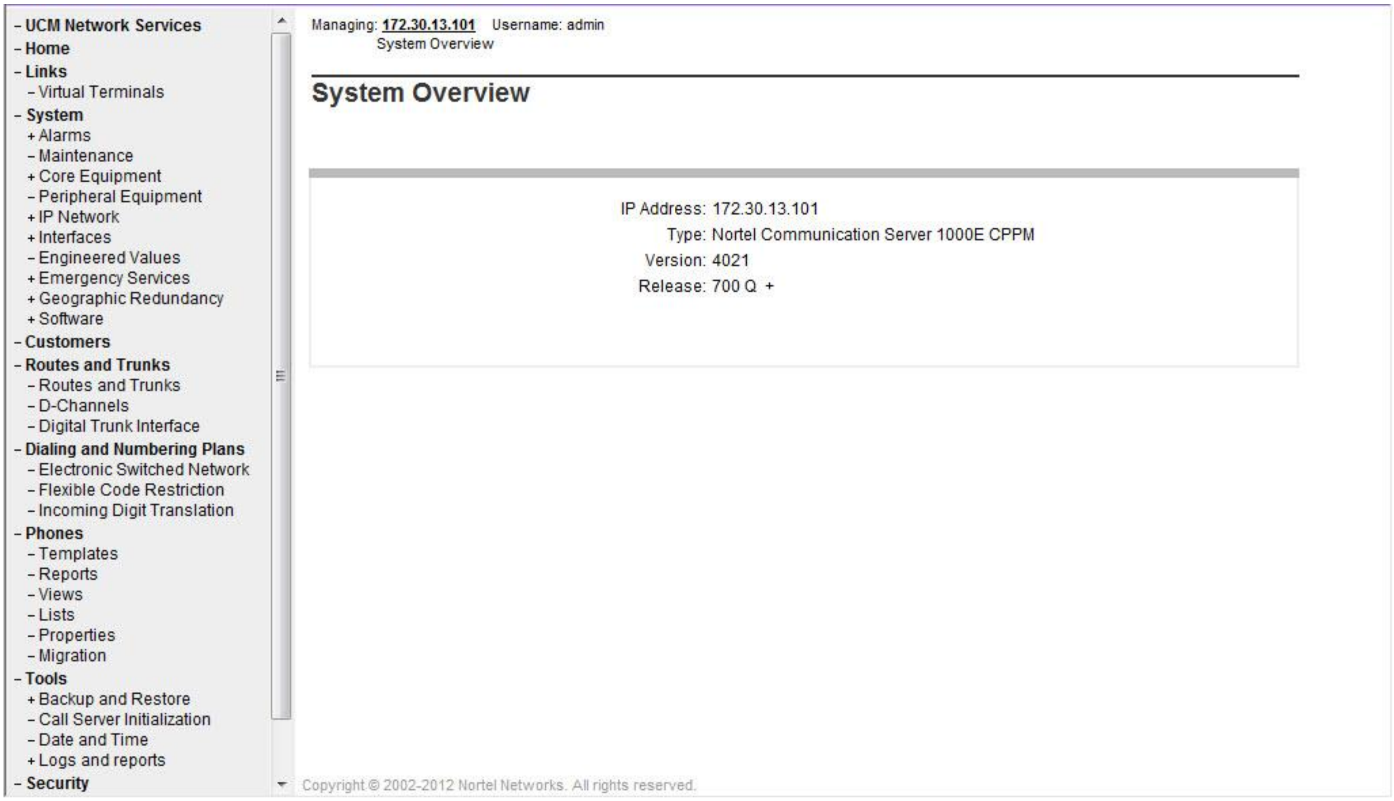
```
>LD 21
PT1000
REQ: LTM
CUST 0
ROUT 30
TYPE TLST
TKTP TIE
ROUT 30
DES SIP_TRUNK
TN 020 0 03 00 MBER 1 SIP_TRUNK
TN 020 0 03 01 MBER 2 SIP_TRUNK
TN 020 0 03 02 MBER 3 SIP_TRUNK
```

```
TN 020 0 03 03 MBER 4 SIP_TRUNK
TN 020 0 03 04 MBER 5 SIP_TRUNK
TN 020 0 03 05 MBER 6 SIP_TRUNK
TN 020 0 03 06 MBER 7 SIP_TRUNK
TN 020 0 03 07 MBER 8 SIP_TRUNK
```

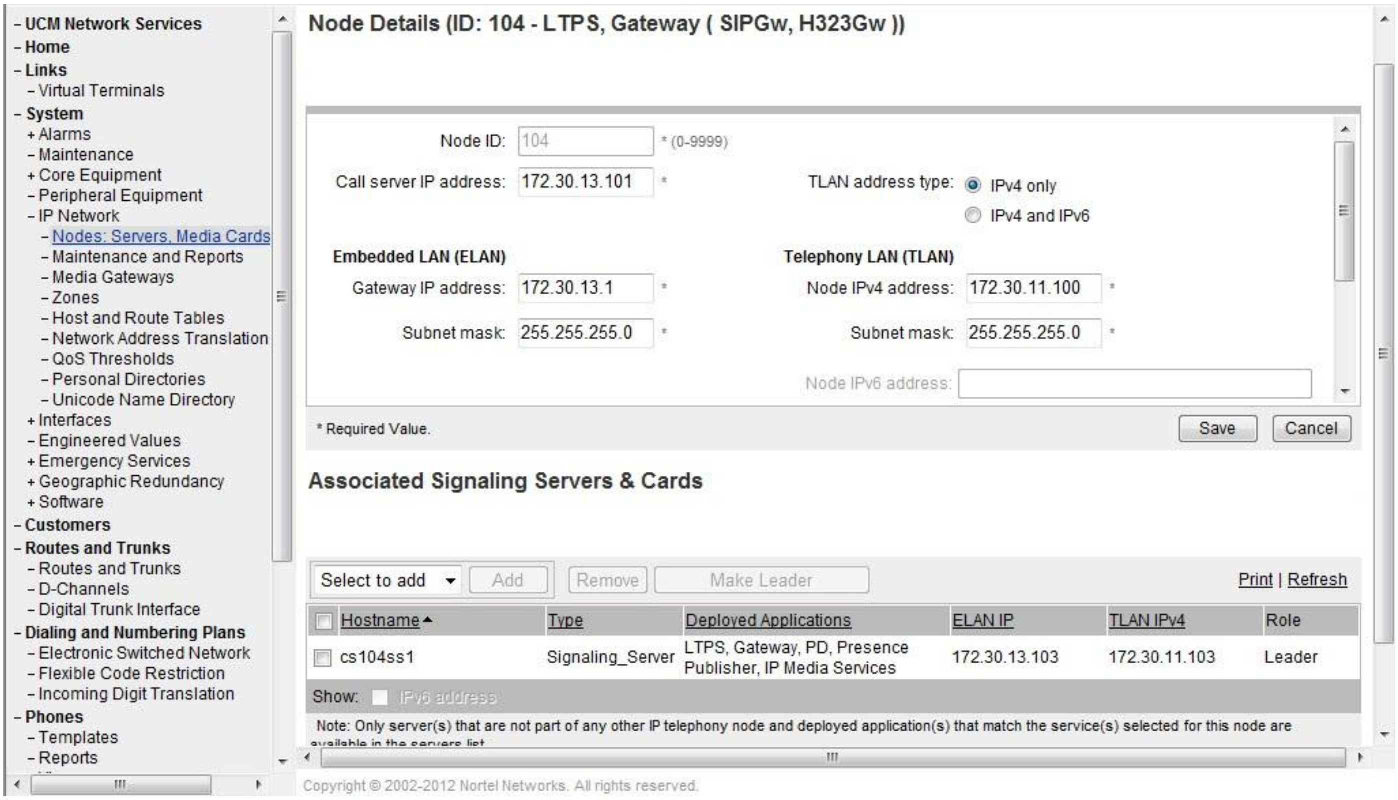
Nortel CS1000E Element Manager



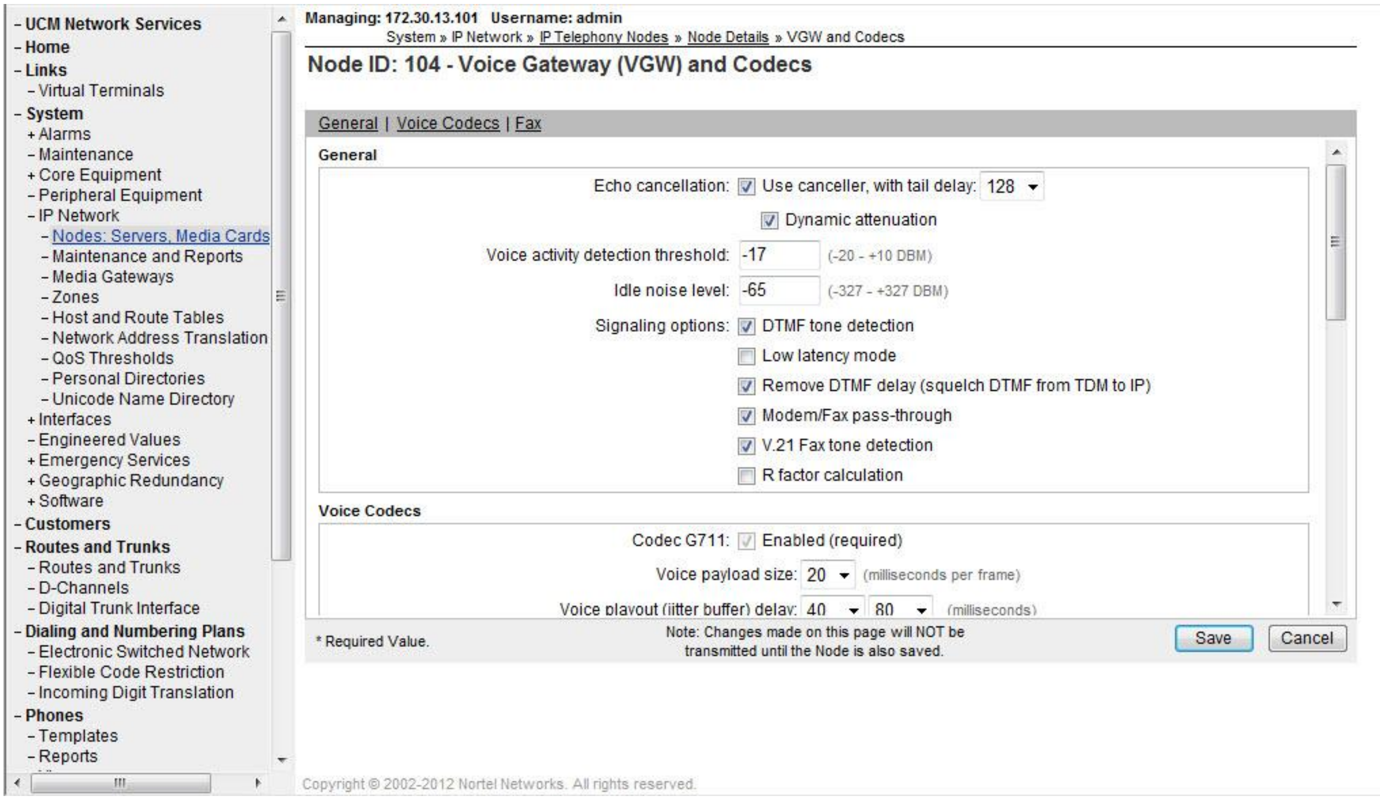
Element Manager-System Overview

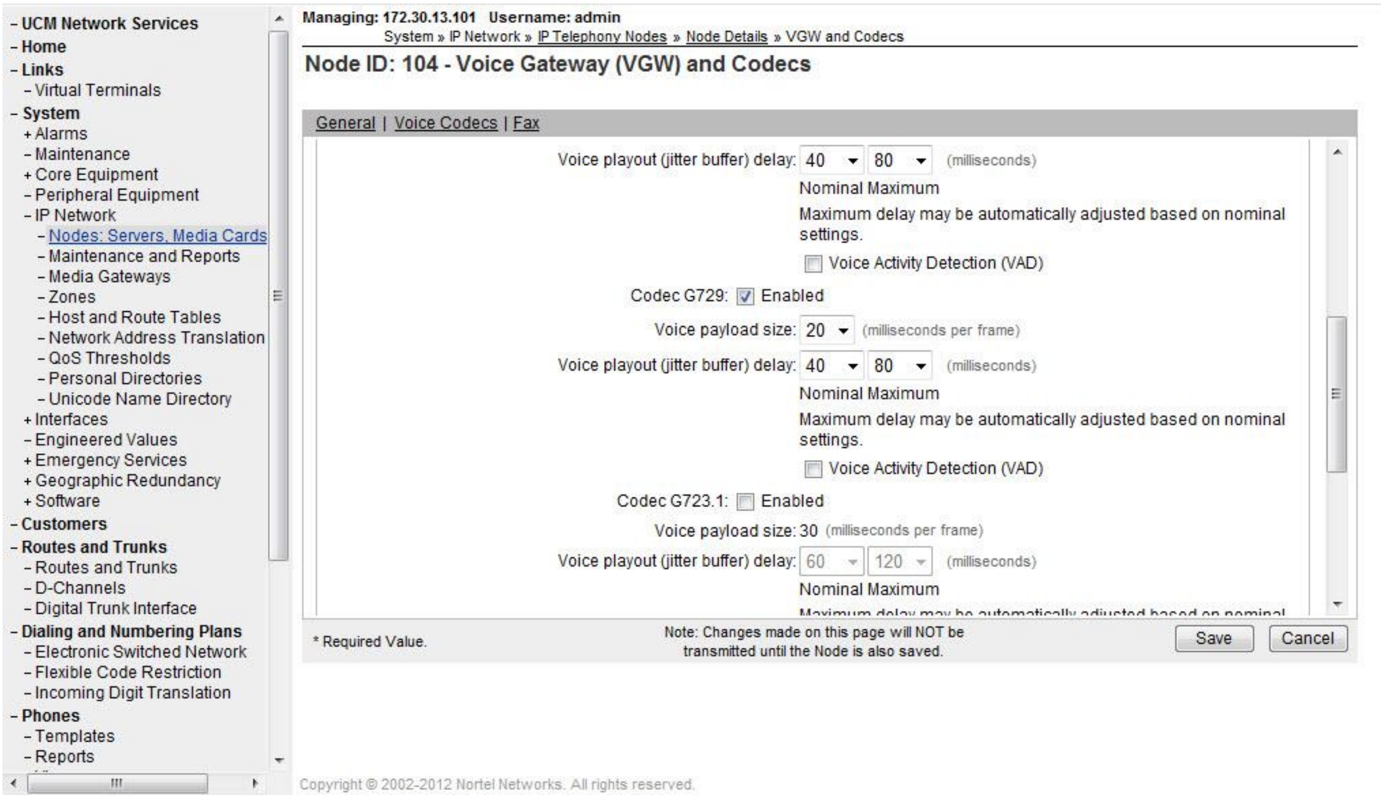


Element Manager - Node Details



Element Manager - Voice Gateway and Codec




- UCM Network Services
- Home
- Links
- Virtual Terminals
- System
+ Alarms
- Maintenance
+ Core Equipment
- Peripheral Equipment
- IP Network
- Nodes: Servers, Media Cards
- Maintenance and Reports
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+ 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
Managing: 172.30.13.101 Username: admin
System » IP Network » IP Telephony Nodes » Node Details » VGW and Codecs
Node ID: 104 - Voice Gateway (VGW) and Codecs
General | Voice Codecs | Fax
Voice playout (jitter buffer) delay: 40 80 (milliseconds)
Nominal Maximum
Maximum delay may be automatically adjusted based on nominal settings.
Voice Activity Detection (VAD)
Codec G729: Enabled
Voice payload size: 20 (milliseconds per frame)
Voice playout (jitter buffer) delay: 40 80 (milliseconds)
Nominal Maximum
Maximum delay may be automatically adjusted based on nominal settings.
Voice Activity Detection (VAD)
Codec G723.1: Enabled
Voice payload size: 30 (milliseconds per frame)
Voice playout (jitter buffer) delay: 60 120 (milliseconds)
Nominal Maximum
* Required Value.
Note: Changes made on this page will NOT be transmitted until the Node is also saved.
Save
Cancel
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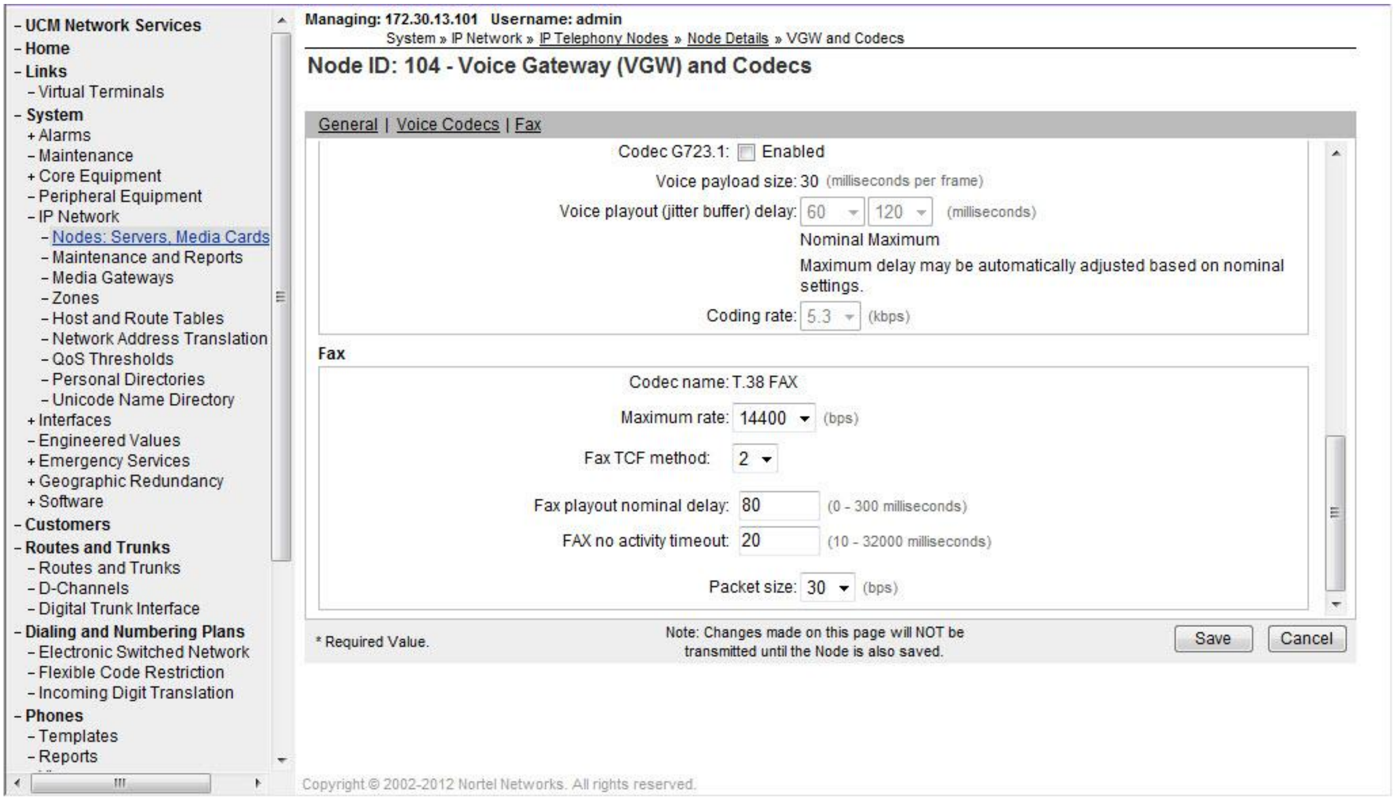

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- Personal Directories
- Unicode Name Directory
+ Interfaces
- Engineered Values
+ Emergency Services
+ Geographic Redundancy
+ Software
- Customers
- Routes and Trunks
- Routes and Trunks
- D-Channels
- Digital Trunk Interface
- Dialing and Numbering Plans
- Electronic Switched Network
- Flexible Code Restriction
- Incoming Digit Translation
- Phones
- Templates
- Reports
Managing: 172.30.13.101 Username: admin
System » IP Network » IP Telephony Nodes » Node Details » VGW and Codecs
Node ID: 104 - Voice Gateway (VGW) and Codecs
General | Voice Codecs | Fax
Codec G723.1: Enabled
Voice payload size: 30 (milliseconds per frame)
Voice playout (jitter buffer) delay: 60 120 (milliseconds)
Nominal Maximum
Maximum delay may be automatically adjusted based on nominal settings.
Coding rate: 5.3 (kbps)
Fax
Codec name: T.38 FAX
Maximum rate: 14400 (bps)
Fax TCF method: 2
Fax playout nominal delay: 80 (0 - 300 milliseconds)
FAX no activity timeout: 20 (10 - 32000 milliseconds)
Packet size: 30 (bps)
* Required Value.
Note: Changes made on this page will NOT be transmitted until the Node is also saved.
Save
Cancel
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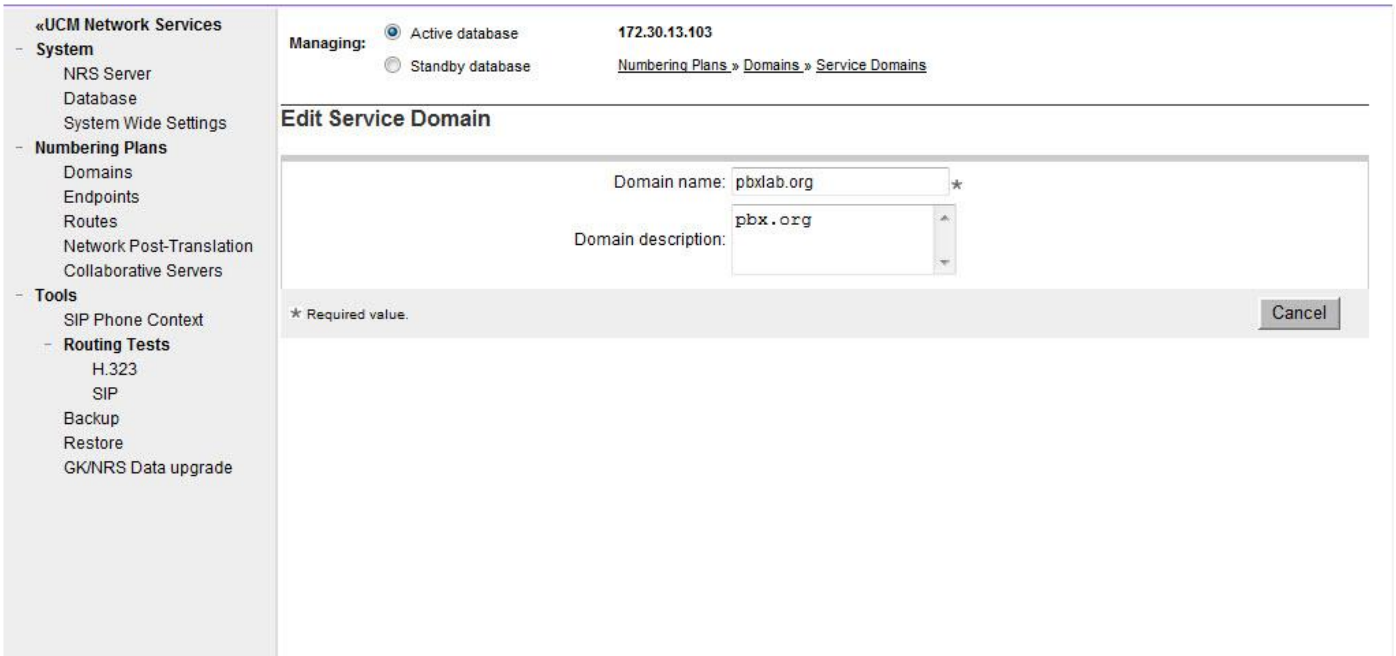
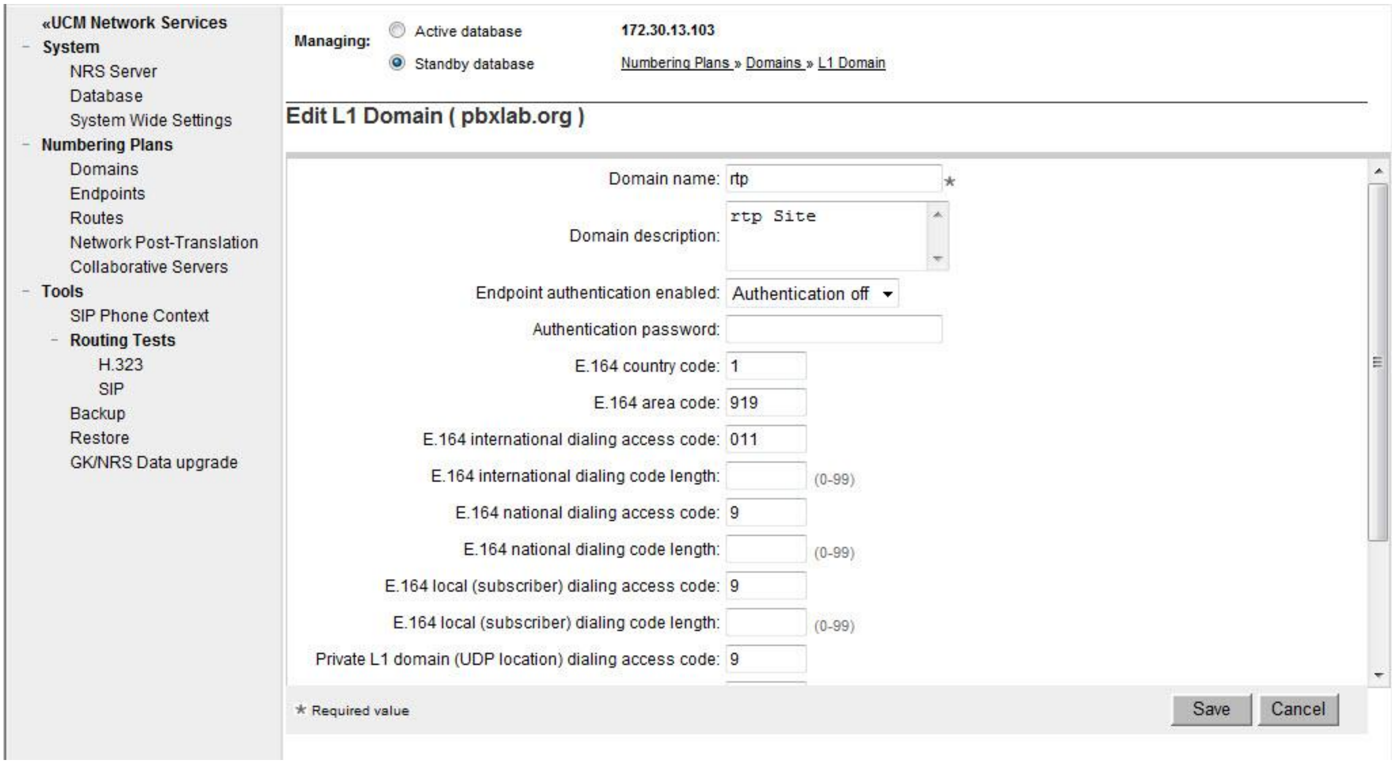
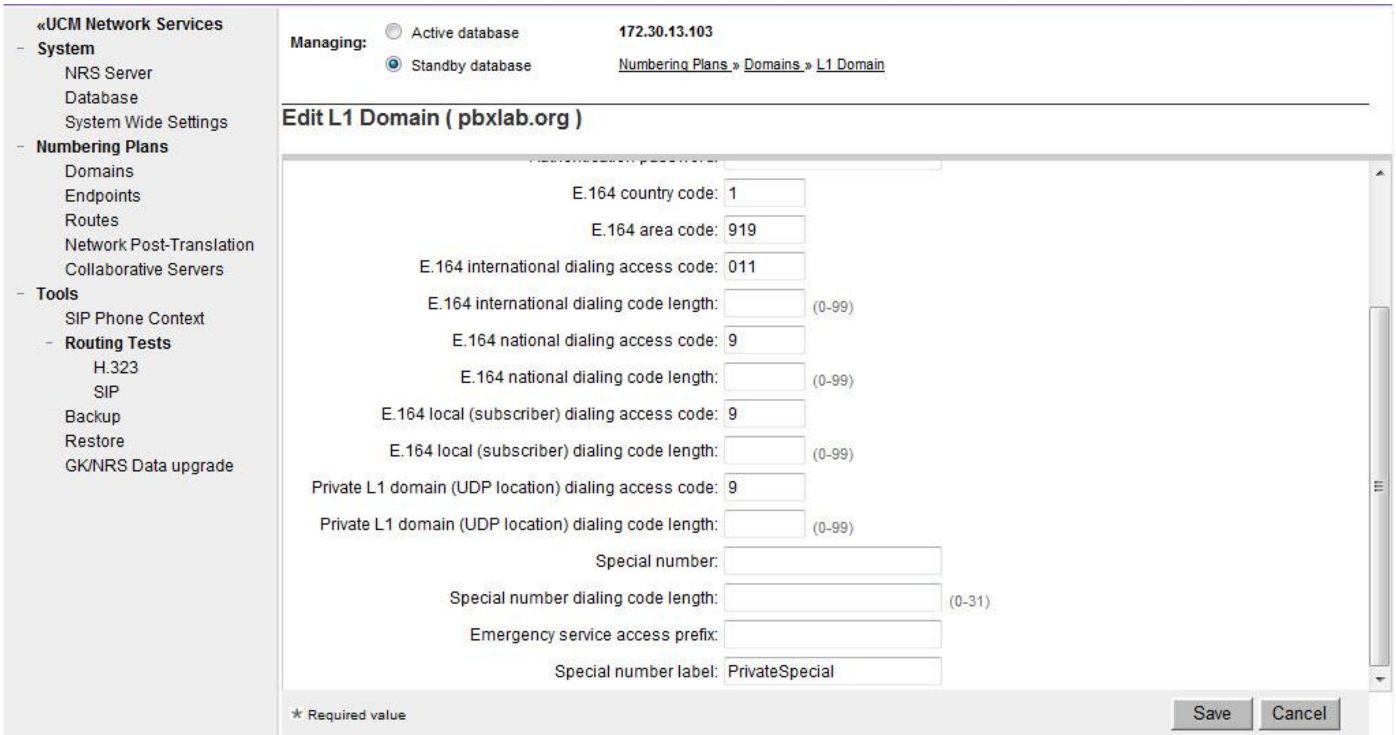
Element Manager - Service Domain



Domain L1



«UCM Network Services
System
NRS Server
Database
System Wide Settings
Numbering Plans
Domains
Endpoints
Routes
Network Post-Translation
Collaborative Servers
Tools
SIP Phone Context
Routing Tests
H.323
SIP
Backup
Restore
GK/NRS Data upgrade
Managing:
Active database
Standby database
172.30.13.103
Numbering Plans » Domains » L1 Domain
Edit L1 Domain (pbxlab.org)
Domain name: rtp
Domain description: rtp Site
Endpoint authentication enabled: Authentication off
Authentication password:
E.164 country code: 1
E.164 area code: 919
E.164 international dialing access code: 011
E.164 international dialing code length: (0-99)
E.164 national dialing access code: 9
E.164 national dialing code length: (0-99)
E.164 local (subscriber) dialing access code: 9
E.164 local (subscriber) dialing code length: (0-99)
Private L1 domain (UDP location) dialing access code: 9
* Required value
Save
Cancel


«UCM Network Services
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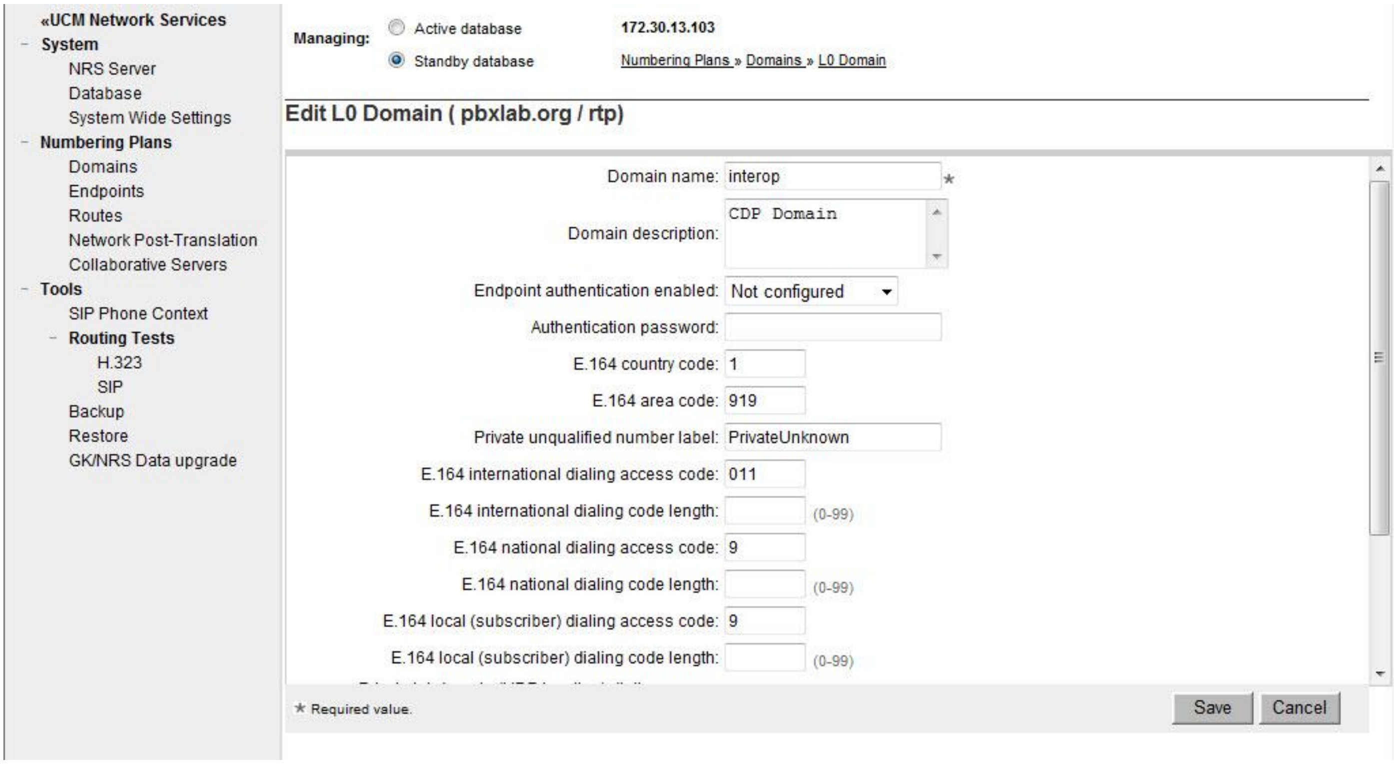
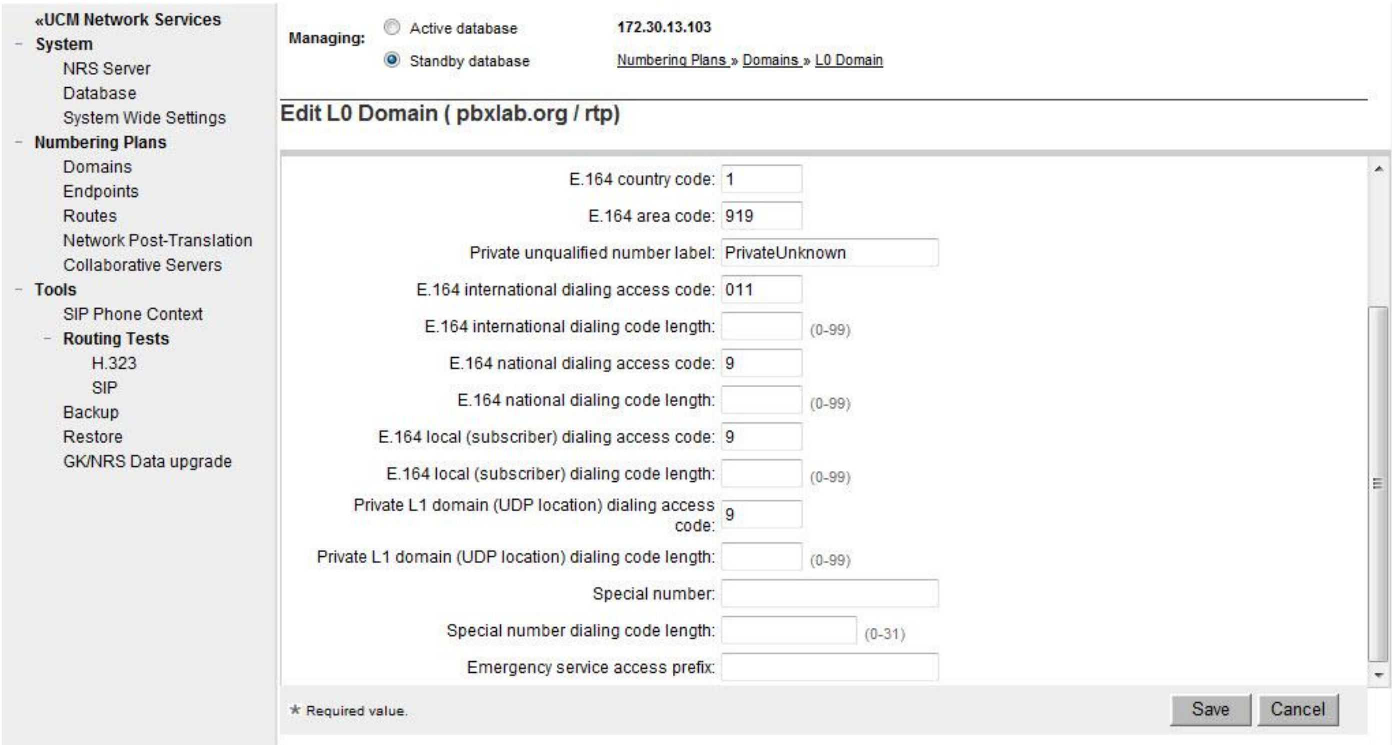
Tools
SIP Phone Context
Routing Tests
H.323
SIP
Backup
Restore
GK/NRS Data upgrade
Managing:
Active database
Standby database
172.30.13.103
Numbering Plans » Domains » L1 Domain
Edit L1 Domain (pbxlab.org)
E.164 country code: 1
E.164 area code: 919
E.164 international dialing access code: 011
E.164 international dialing code length: (0-99)
E.164 national dialing access code: 9
E.164 national dialing code length: (0-99)
E.164 local (subscriber) dialing access code: 9
E.164 local (subscriber) dialing code length: (0-99)
Private L1 domain (UDP location) dialing access code: 9
Private L1 domain (UDP location) dialing code length: (0-99)
Special number:
Special number dialing code length: (0-31)
Emergency service access prefix:
Special number label: PrivateSpecial
* Required value
Save
Cancel

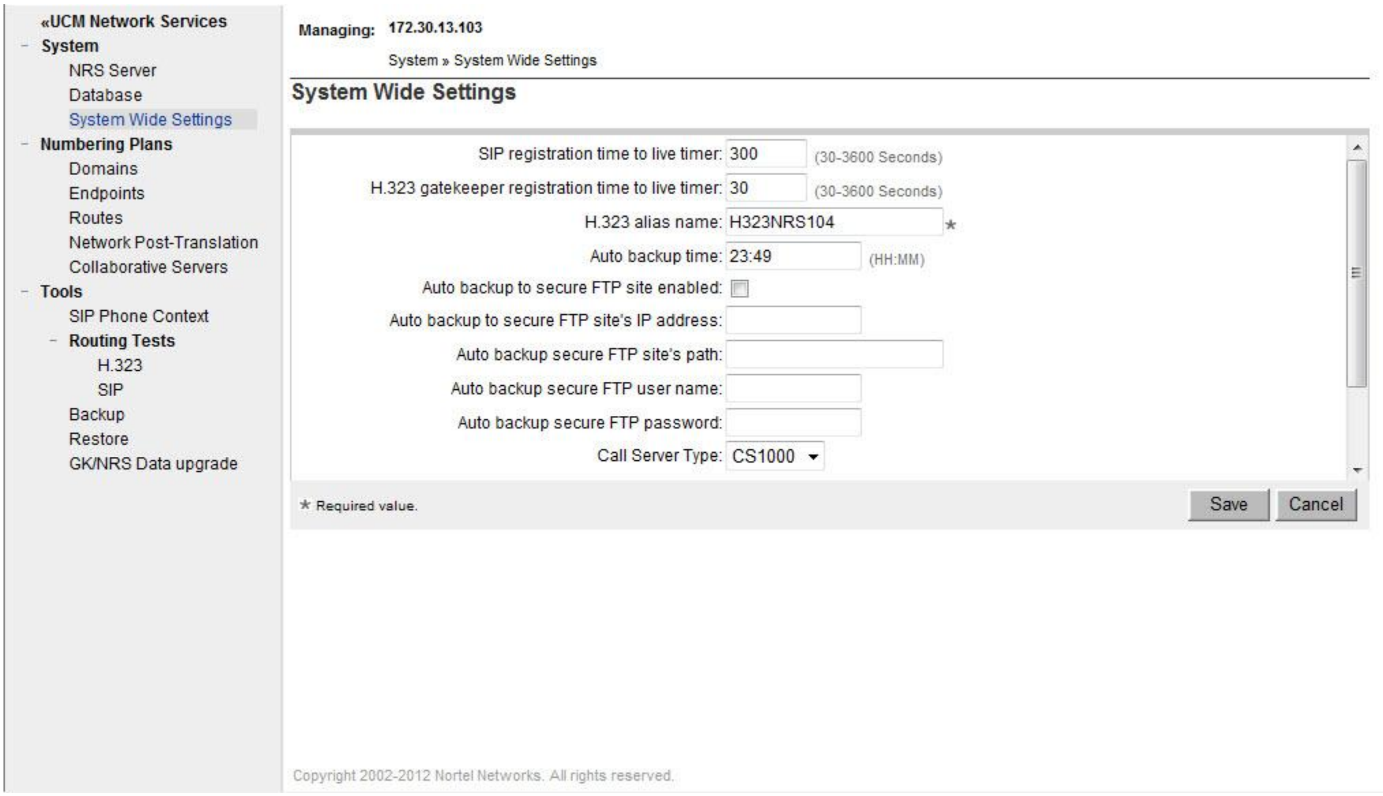
Domain L0


«UCM Network Services
System
NRS Server
Database
System Wide Settings
Numbering Plans
Domains
Endpoints
Routes
Network Post-Translation
Collaborative Servers
Tools
SIP Phone Context
Routing Tests
H.323
SIP
Backup
Restore
GK/NRS Data upgrade
Managing:
Active database
Standby database
172.30.13.103
Numbering Plans » Domains » L0 Domain
Edit L0 Domain (pbxlab.org / rtp)
Domain name: interop
Domain description: CDP Domain
Endpoint authentication enabled: Not configured
Authentication password:
E.164 country code: 1
E.164 area code: 919
Private unqualified number label: PrivateUnknown
E.164 international dialing access code: 011
E.164 international dialing code length: (0-99)
E.164 national dialing access code: 9
E.164 national dialing code length: (0-99)
E.164 local (subscriber) dialing access code: 9
E.164 local (subscriber) dialing code length: (0-99)
* Required value.
Save
Cancel

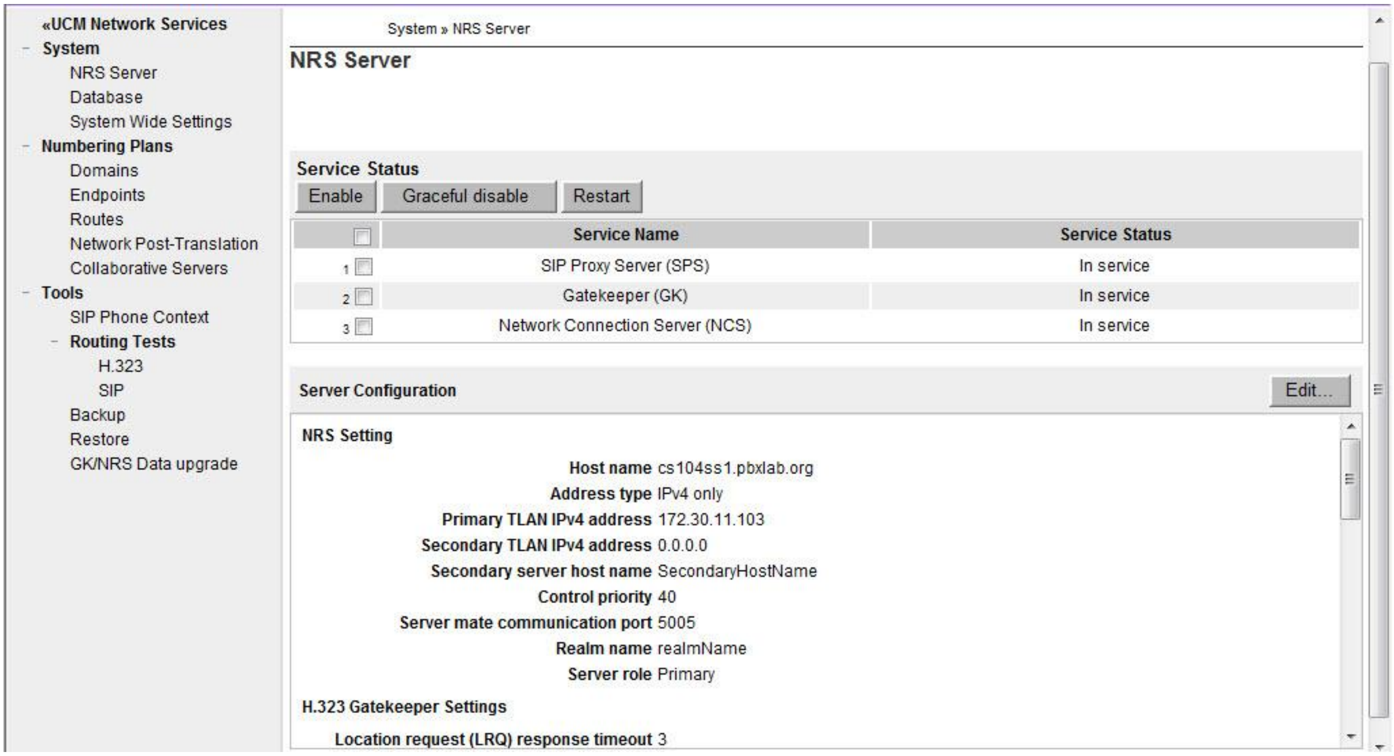

«UCM Network Services
System
NRS Server
Database
System Wide Settings
Numbering Plans
Domains
Endpoints
Routes
Network Post-Translation
Collaborative Servers
Tools
SIP Phone Context
Routing Tests
H.323
SIP
Backup
Restore
GK/NRS Data upgrade
Managing:
Active database
Standby database
172.30.13.103
Numbering Plans » Domains » L0 Domain
Edit L0 Domain (pbxlab.org / rtp)
E.164 country code: 1
E.164 area code: 919
Private unqualified number label: PrivateUnknown
E.164 international dialing access code: 011
E.164 international dialing code length: (0-99)
E.164 national dialing access code: 9
E.164 national dialing code length: (0-99)
E.164 local (subscriber) dialing access code: 9
E.164 local (subscriber) dialing code length: (0-99)
Private L1 domain (UDP location) dialing access code: 9
Private L1 domain (UDP location) dialing code length: (0-99)
Special number:
Special number dialing code length: (0-31)
Emergency service access prefix:
* Required value.
Save
Cancel

Network Routing Services (NRS).

System Wide Setting



NRS – Server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Tools
SIP Phone Context
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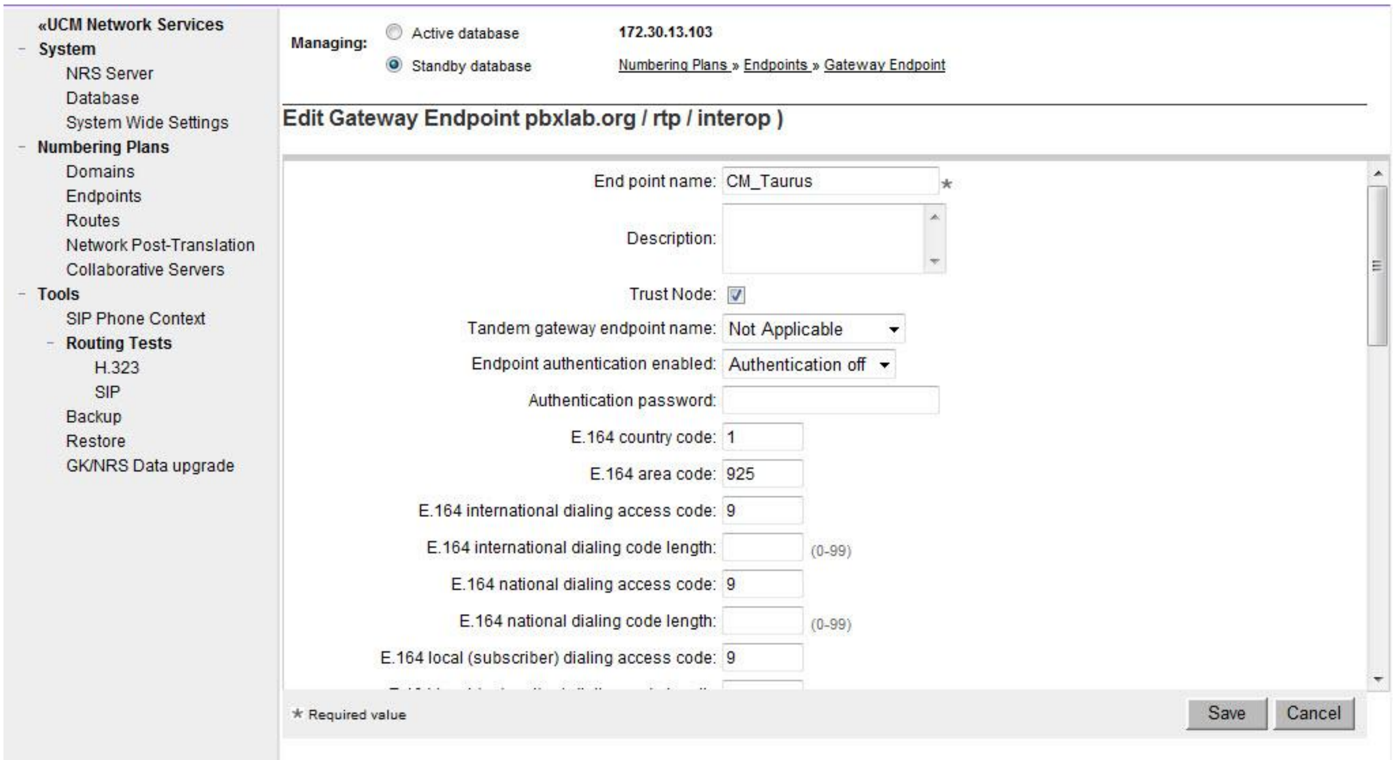
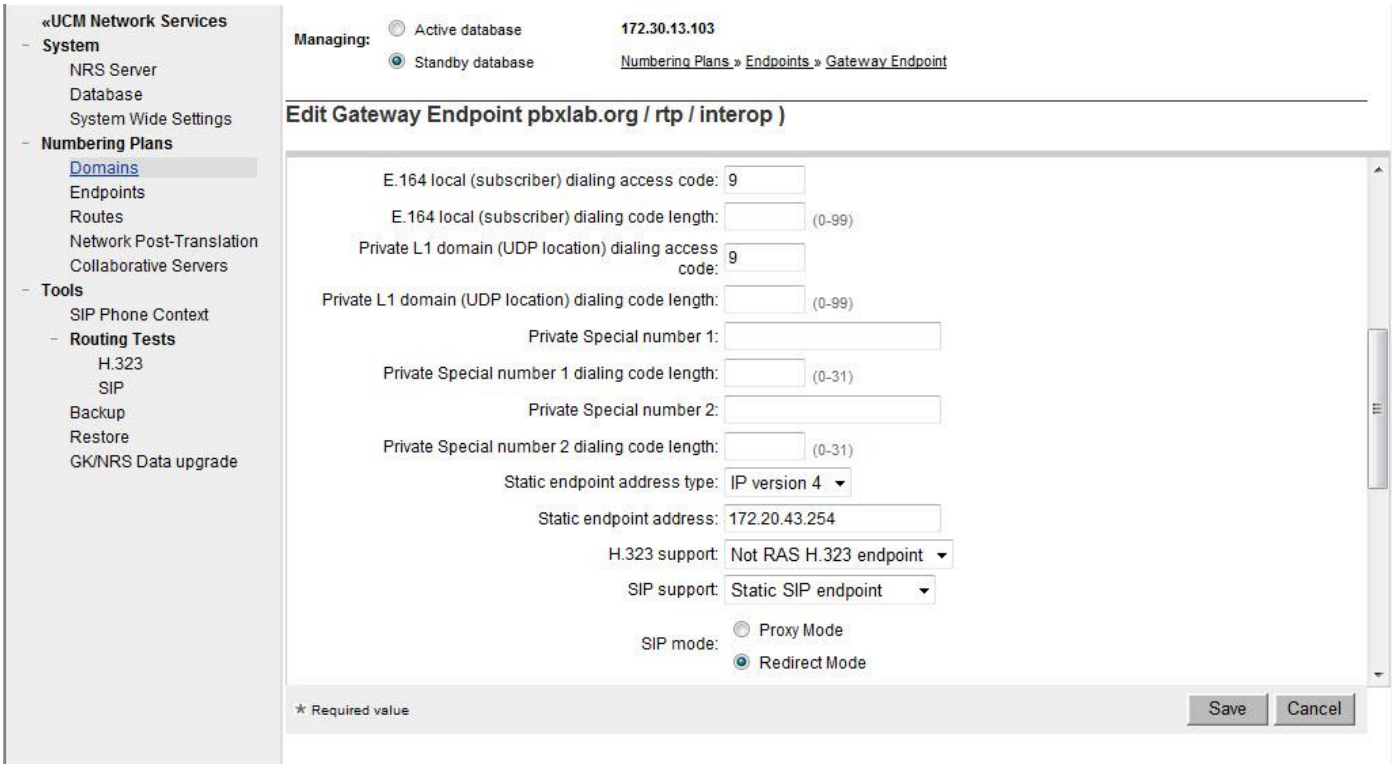
H.323
SIP
Backup
Restore
GK/NRS Data upgrade
System » NRS Server
NRS Server
Service Status
Enable
Graceful disable
Restart
Service Name
Service Status
1 SIP Proxy Server (SPS) In service
2 Gatekeeper (GK) In service
3 Network Connection Server (NCS) In service
Server Configuration
Edit...
NRS Setting
Host name cs104ss1.pbxlab.org
Address type IPv4 only
Primary TLAN IPv4 address 172.30.11.103
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
H.323 Gatekeeper Settings
Location request (LRQ) response timeout 3

NRS- Gateway End Points

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NRS Server
Database
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Routes
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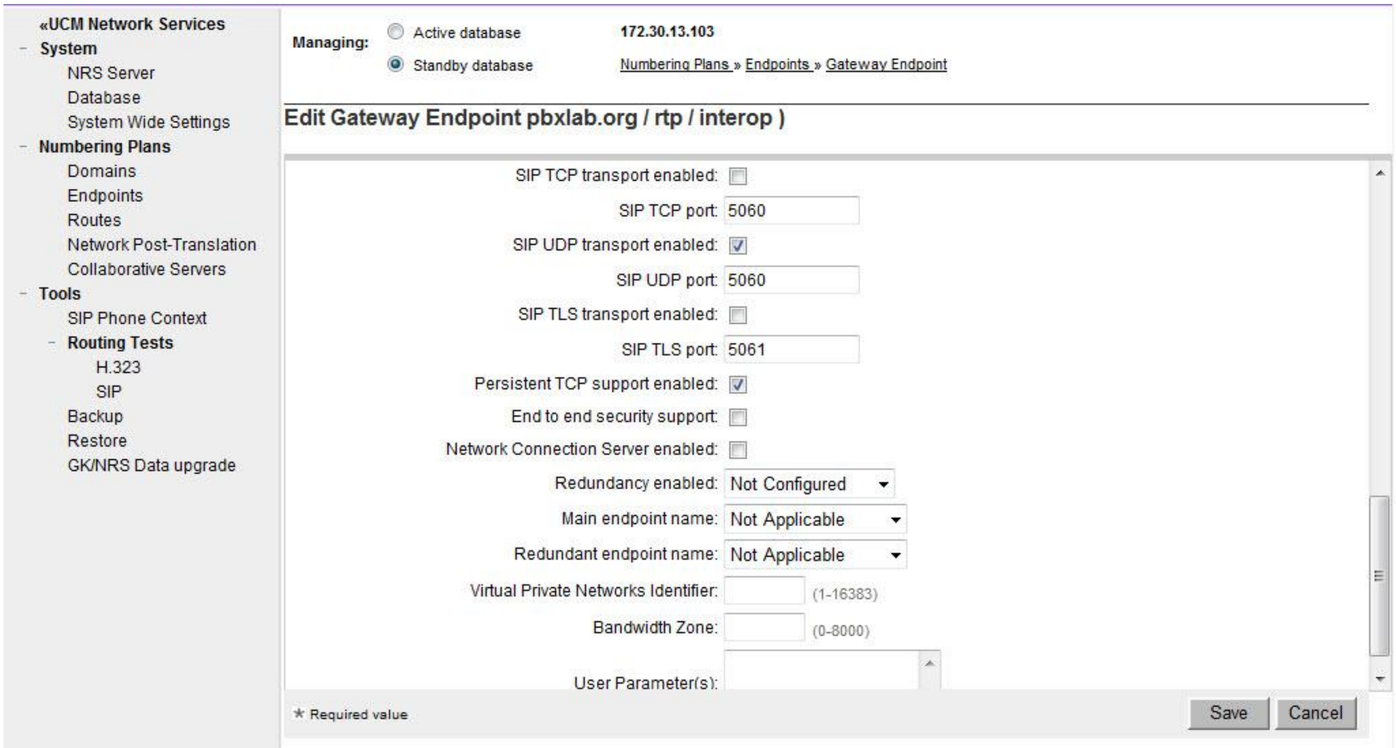
Collaborative Servers
Tools
SIP Phone Context
Routing Tests
H.323
SIP
Backup
Restore
GK/NRS Data upgrade
Managing:
Active database
Standby database
172.30.13.103
Numbering Plans » Endpoints » Gateway Endpoint
Edit Gateway Endpoint pbxlab.org / rtp / interop)
E.164 local (subscriber) dialing access code: 9
E.164 local (subscriber) dialing code length: (0-99)
Private L1 domain (UDP location) dialing access code: 9
Private L1 domain (UDP location) dialing code length: (0-99)
Private Special number 1:
Private Special number 1 dialing code length: (0-31)
Private Special number 2:
Private Special number 2 dialing code length: (0-31)
Static endpoint address type: IP version 4
Static endpoint address: 172.20.43.254
H.323 support: Not RAS H.323 endpoint
SIP support: Static SIP endpoint
SIP mode:
Proxy Mode
Redirect Mode
* Required value
Save
Cancel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System Wide Settings
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Endpoints
Routes
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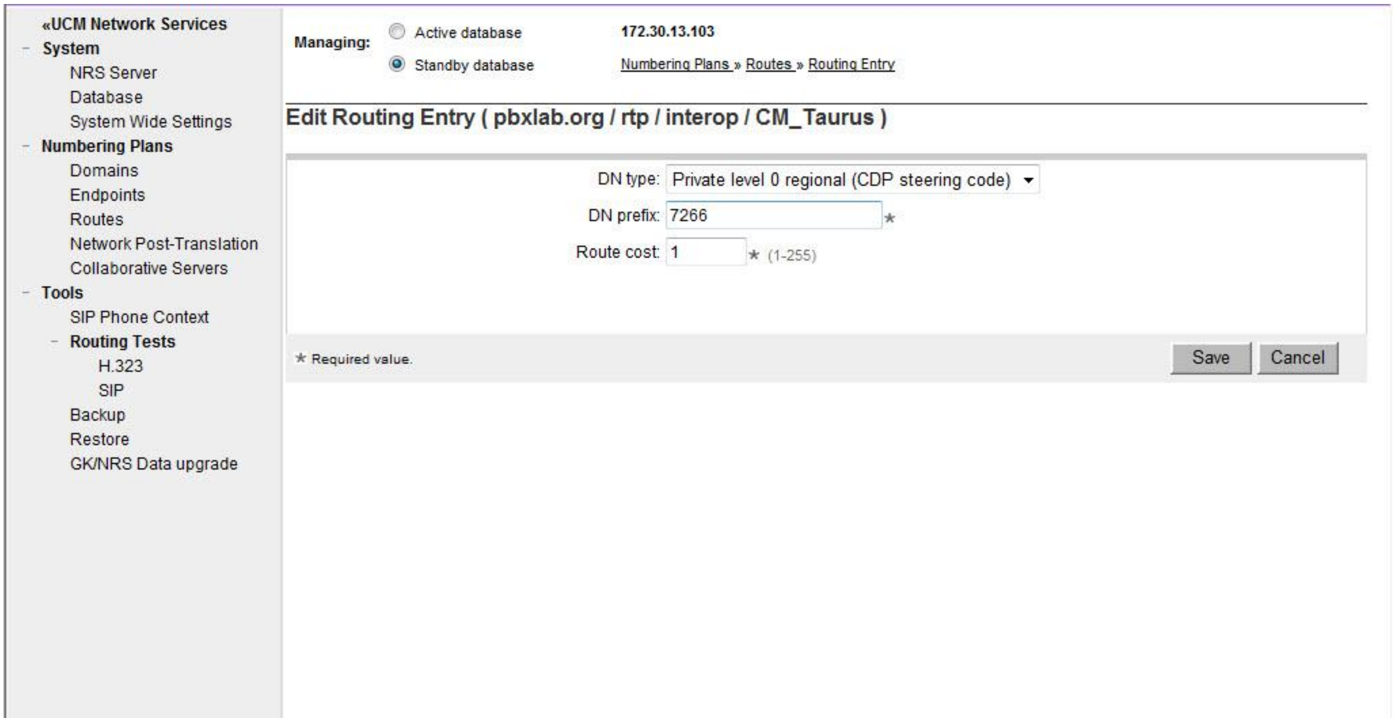
Collaborative Servers
Tools
SIP Phone Context
Routing Tests
H.323
SIP
Backup
Restore
GK/NRS Data upgrade
Managing:
Active database
Standby database
172.30.13.103
Numbering Plans » Endpoints » Gateway Endpoint
Edit Gateway Endpoint pbxlab.org / rtp / interop)
SIP TCP transport enabled:
SIP TCP port: 5060
SIP UDP transport enabled:
SIP UDP port: 5060
SIP TLS transport enabled:
SIP TLS port: 5061
Persistent TCP support enabled:
End to end security support:
Network Connection Server enabled:
Redundancy enabled: Not Configured
Main endpoint name: Not Applicable
Redundant endpoint name: Not Applicable
Virtual Private Networks Identifier: (1-16383)
Bandwidth Zone: (0-8000)
User Parameter(s):
* Required value
Save
Cancel

NRS Routes



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Tools
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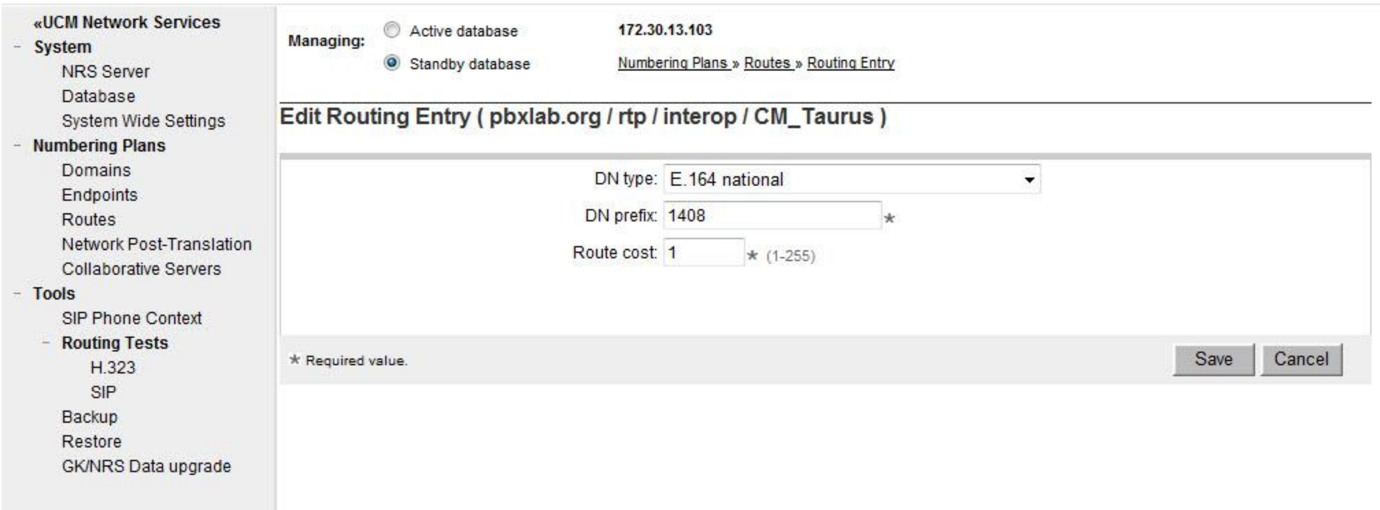
H.323
SIP
Backup
Restore
GK/NRS Data upgrade
Managing:
Active database
Standby database
172.30.13.103
Numbering Plans » Routes » Routing Entry
Edit Routing Entry (pbxlab.org / rtp / interop / CM_Taurus)
DN type: Private level 0 regional (CDP steering code)
DN prefix: 7266
Route cost: 1
(1-255)
★ Required value.
Save
Cancel



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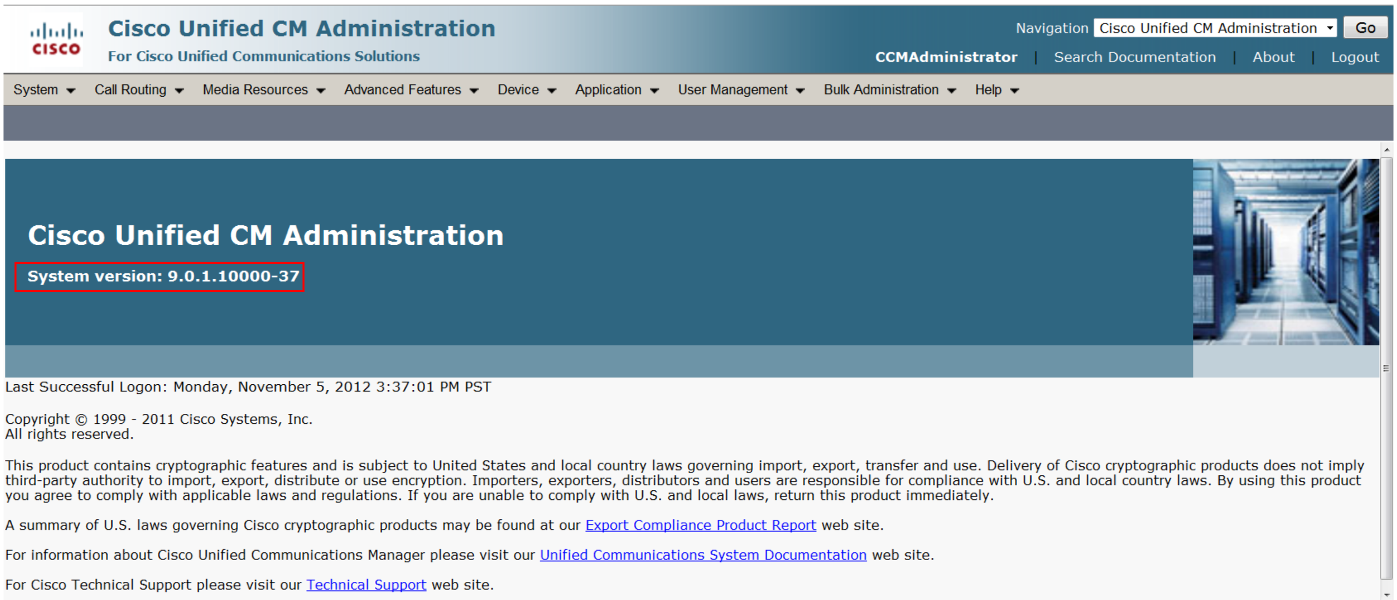
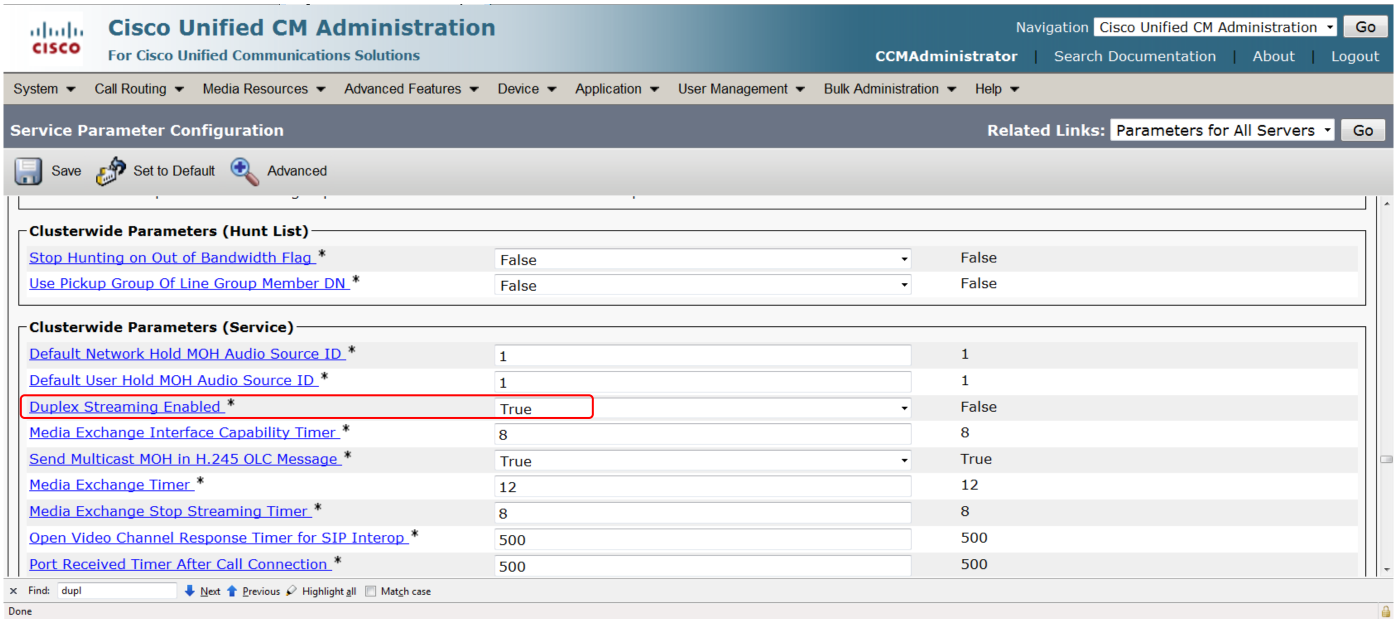
Routing Tests
H.323
SIP
Backup
Restore
GK/NRS Data upgrade
Managing:
Active database
Standby database
172.30.13.103
Numbering Plans » Routes » Routing Entry
Edit Routing Entry (pbxlab.org / rtp / interop / CM_Taurus)
DN type: E.164 national
DN prefix: 1408 *
Route cost: 1 * (1-255)
* Required value.
Save
Cancel

Configuring the Cisco Unified Communications Manager

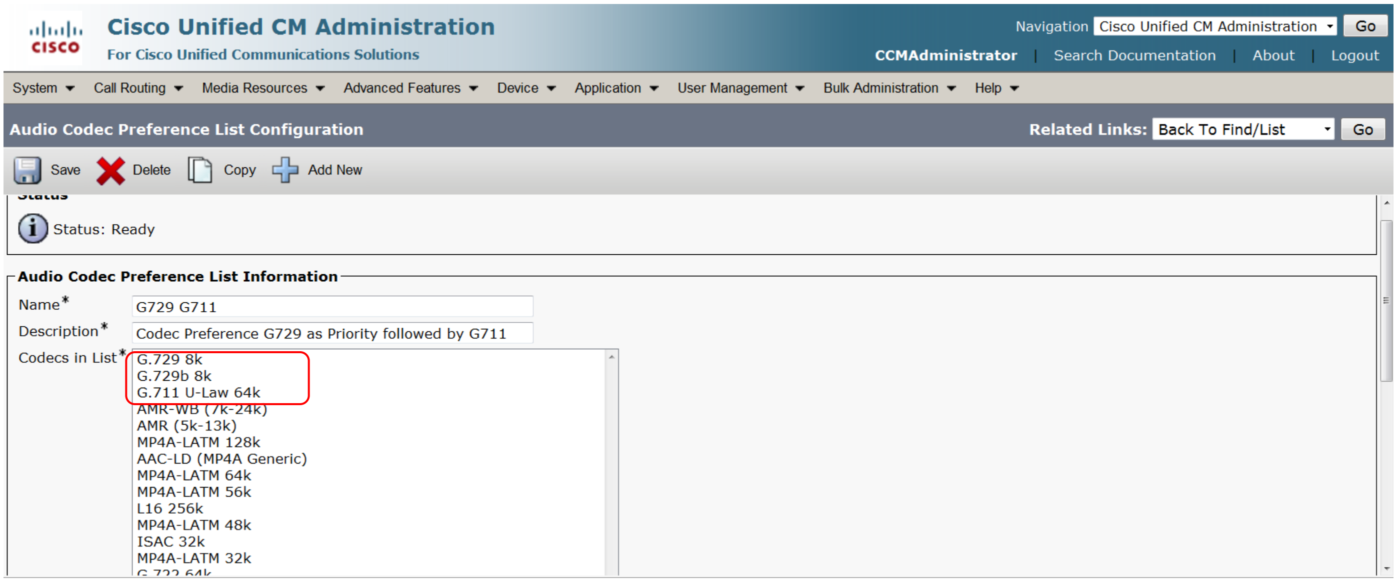
Cisco Unified Communications Manager –Software Release



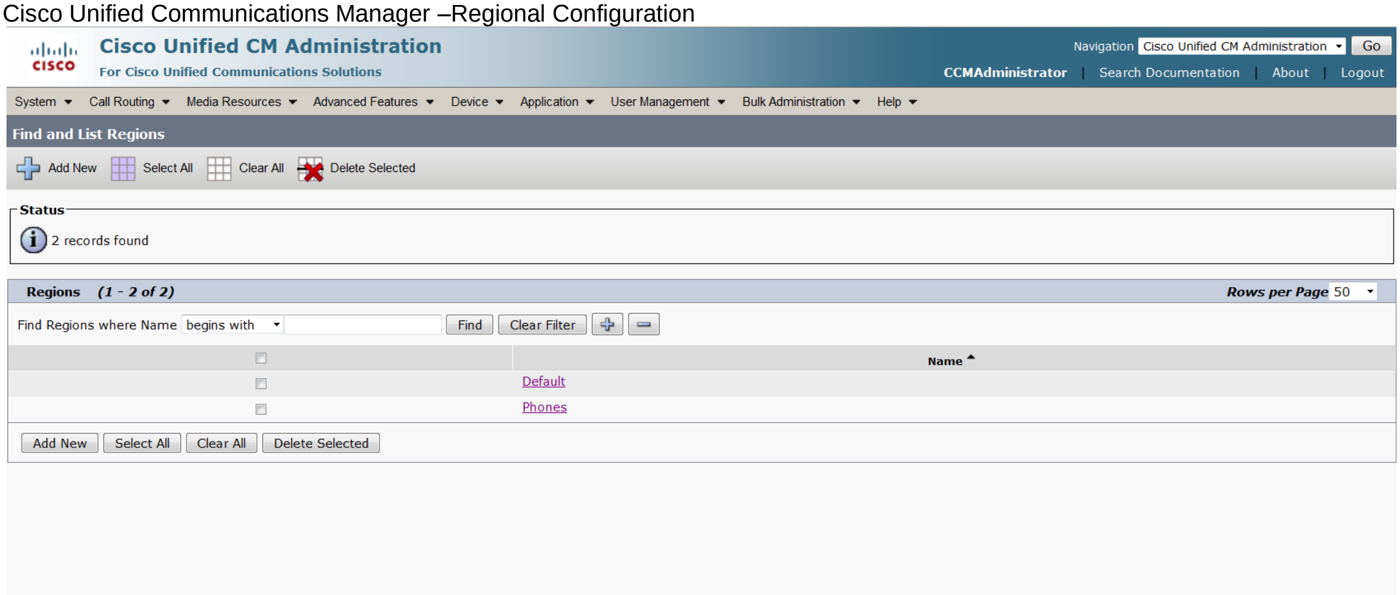
Cisco Unified Communication Manager – Service Parameter setting for Duplex Streaming

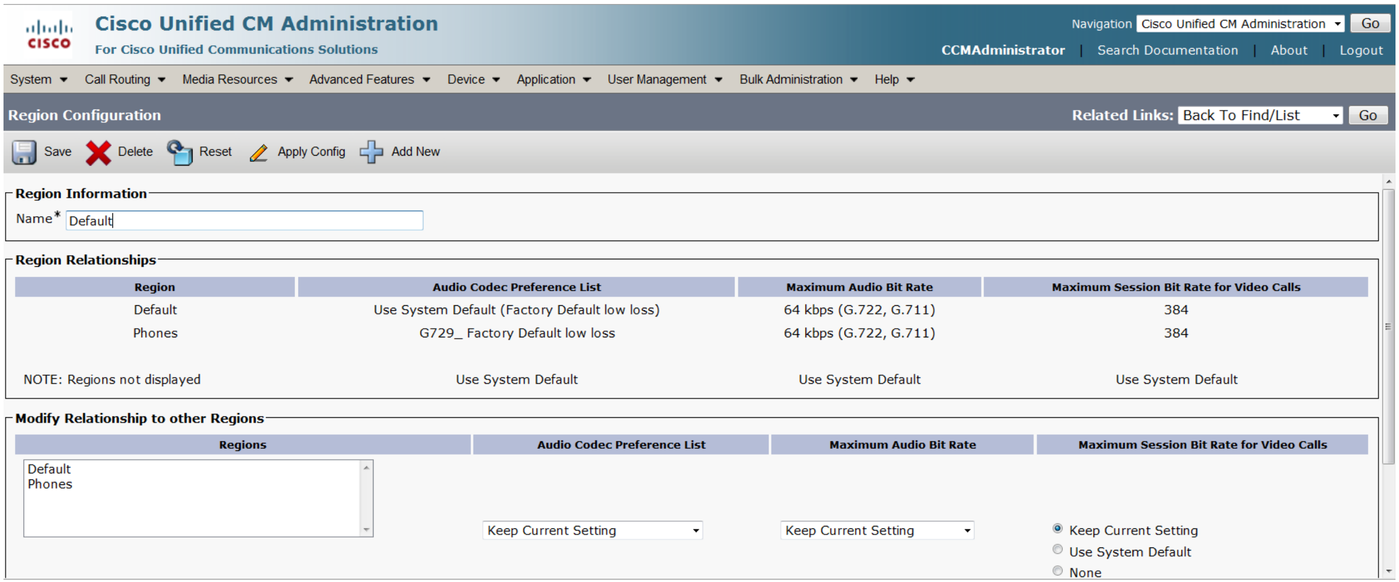


Cisco Unified Communication Manager – Audio Codec Preference Setting



Cisco Unified Communications Manager –Regional Configuration

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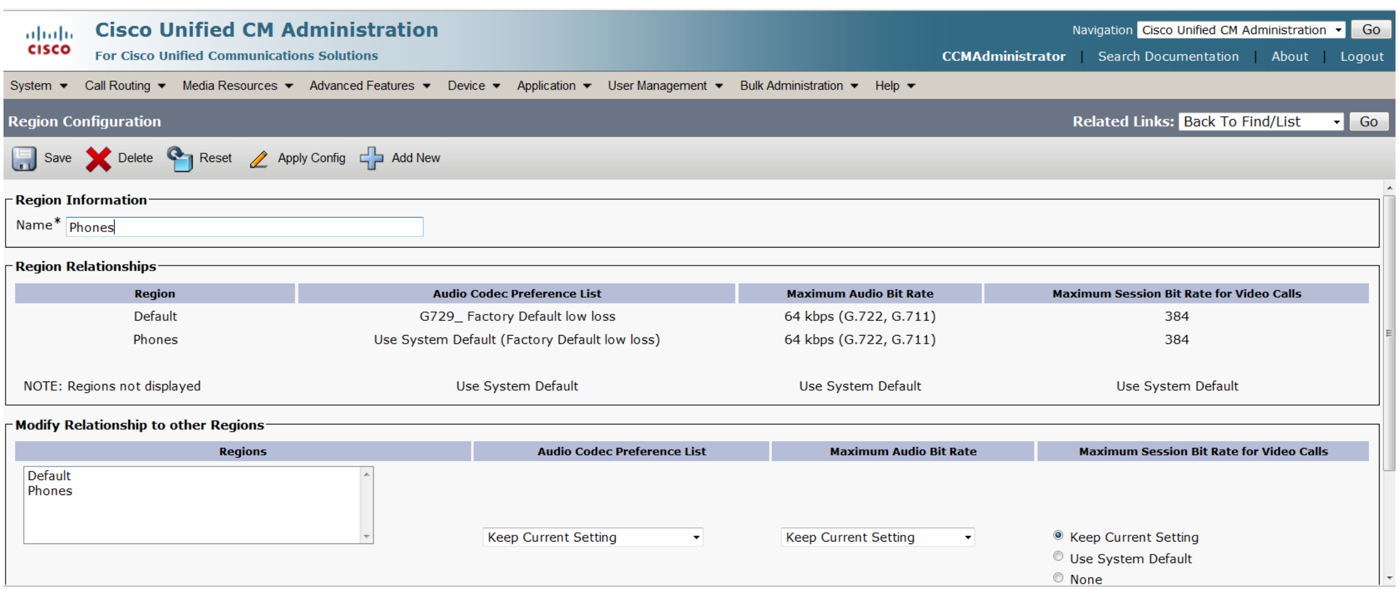
Region Configuration
Related Links: Back To Find/List Go
Save Delete Reset Apply Config Add New
Region Information
Name* Default
Region Relationships
Region | Audio Codec Preference List | Maximum Audio Bit Rate | Maximum Session Bit Rate for Video Calls
Default | Use System Default (Factory Default low loss) | 64 kbps (G.722, G.711) | 384
Phones | G729_ Factory Default low loss | 64 kbps (G.722, G.711) | 384
NOTE: Regions not displayed | Use System Default | Use System Default | Use System Default
Modify Relationship to other Regions
Regions | Audio Codec Preference List | Maximum Audio Bit Rate | Maximum Session Bit Rate for Video Calls
Default
Phones
Keep Current Setting
Keep Current Setting
Keep Current Setting
Use System Default
None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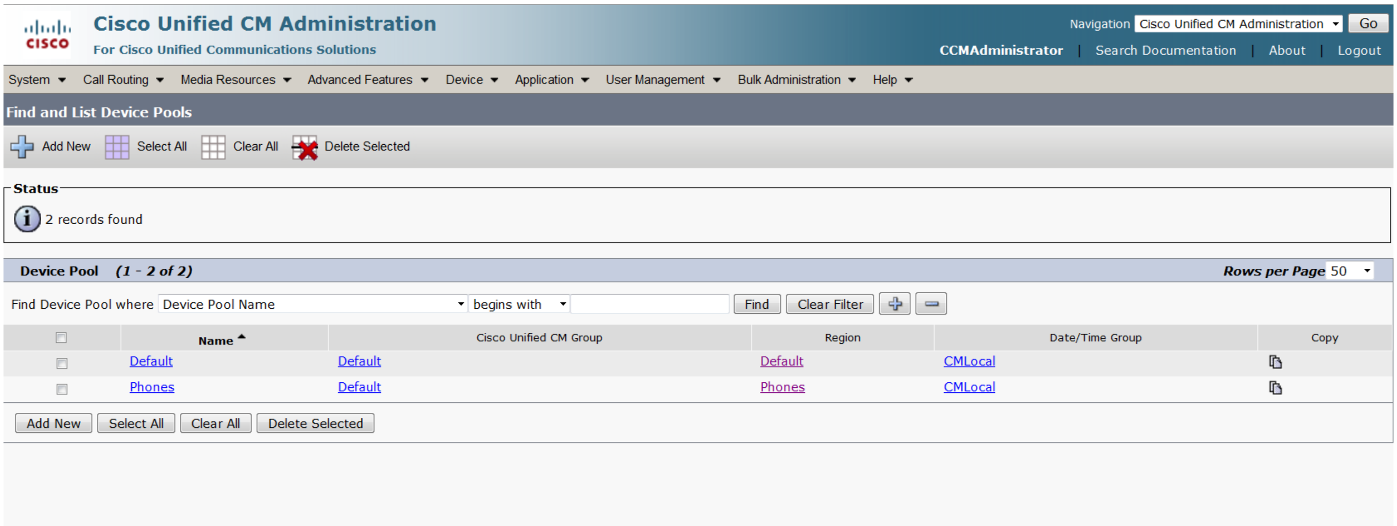
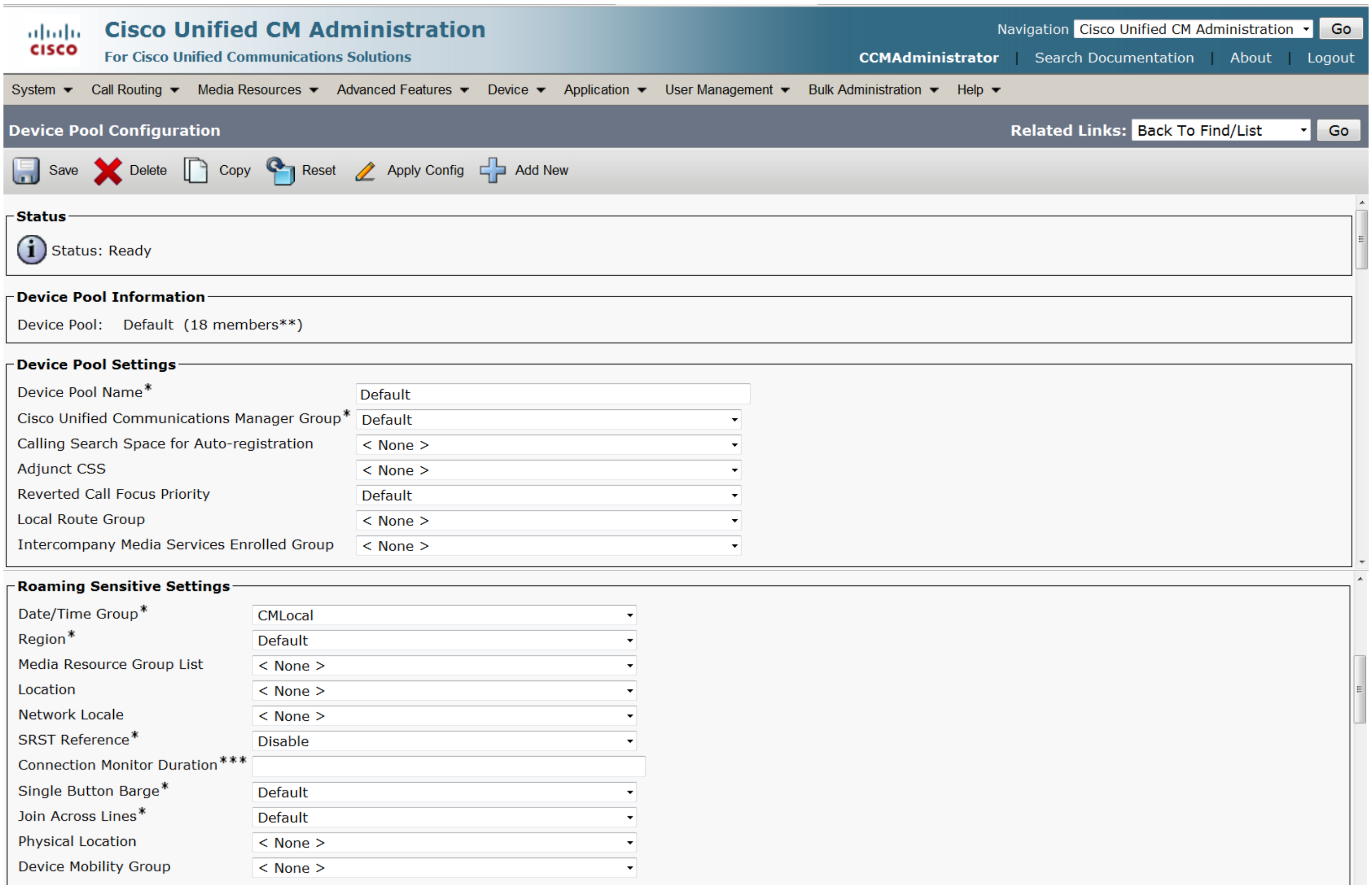
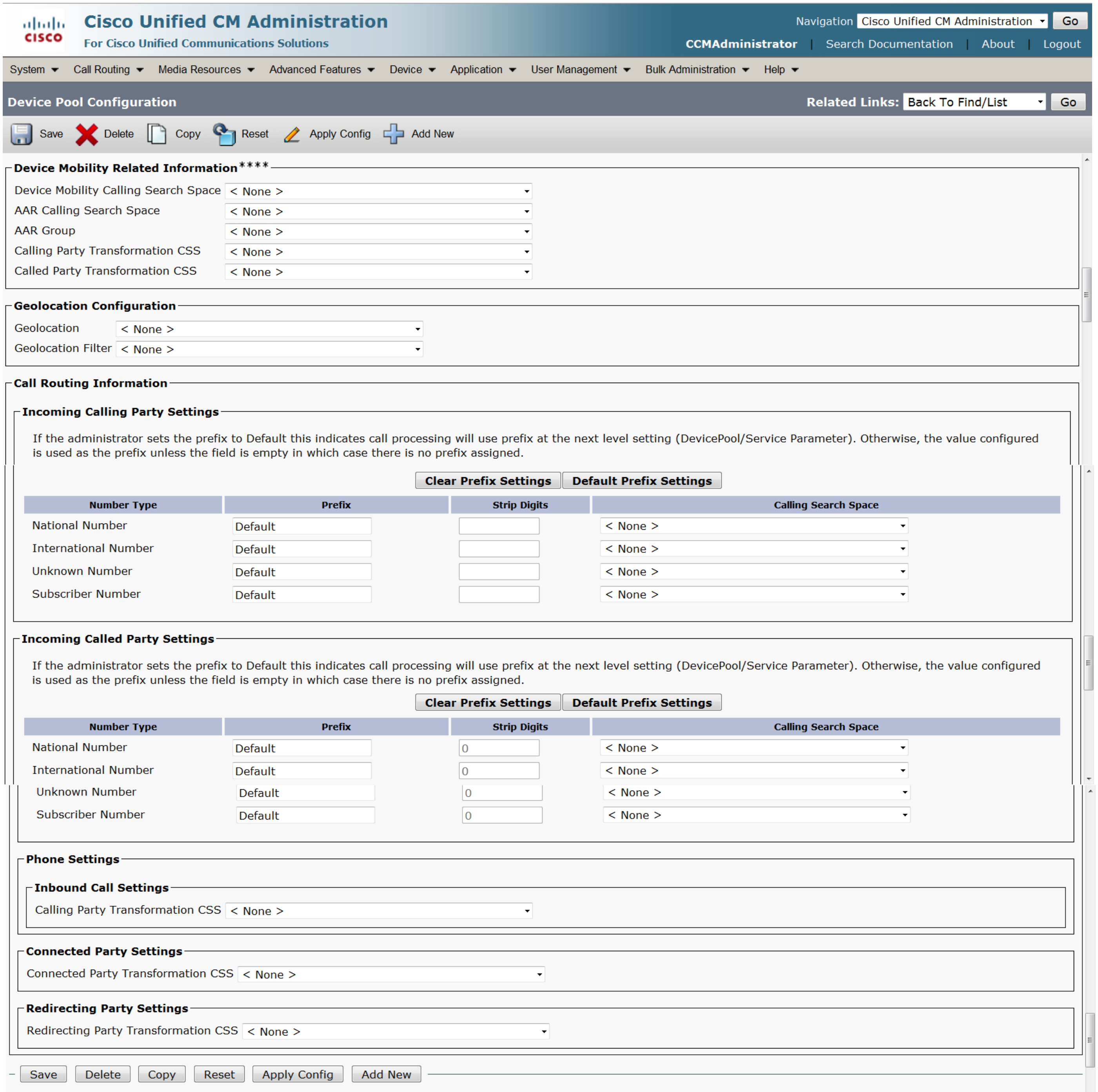
Region Configuration
Related Links: Back To Find/List Go
Save Delete Reset Apply Config Add New
Region Information
Name* Phones
Region Relationships
Region | Audio Codec Preference List | Maximum Audio Bit Rate | Maximum Session Bit Rate for Video Calls
Default | G729_ Factory Default low loss | 64 kbps (G.722, G.711) | 384
Phones | Use System Default (Factory Default low loss) | 64 kbps (G.722, G.711) | 384
NOTE: Regions not displayed | Use System Default | Use System Default | Use System Default
Modify Relationship to other Regions
Regions | Audio Codec Preference List | Maximum Audio Bit Rate | Maximum Session Bit Rate for Video Calls
Default
Phones
Keep Current Setting
Keep Current Setting
Keep Current Setting
Use System Default
None

Cisco Unified Communications Manager – Device Pool Configuration



Cisco Unified Communications Manager – Device Pool Configuration (Default)

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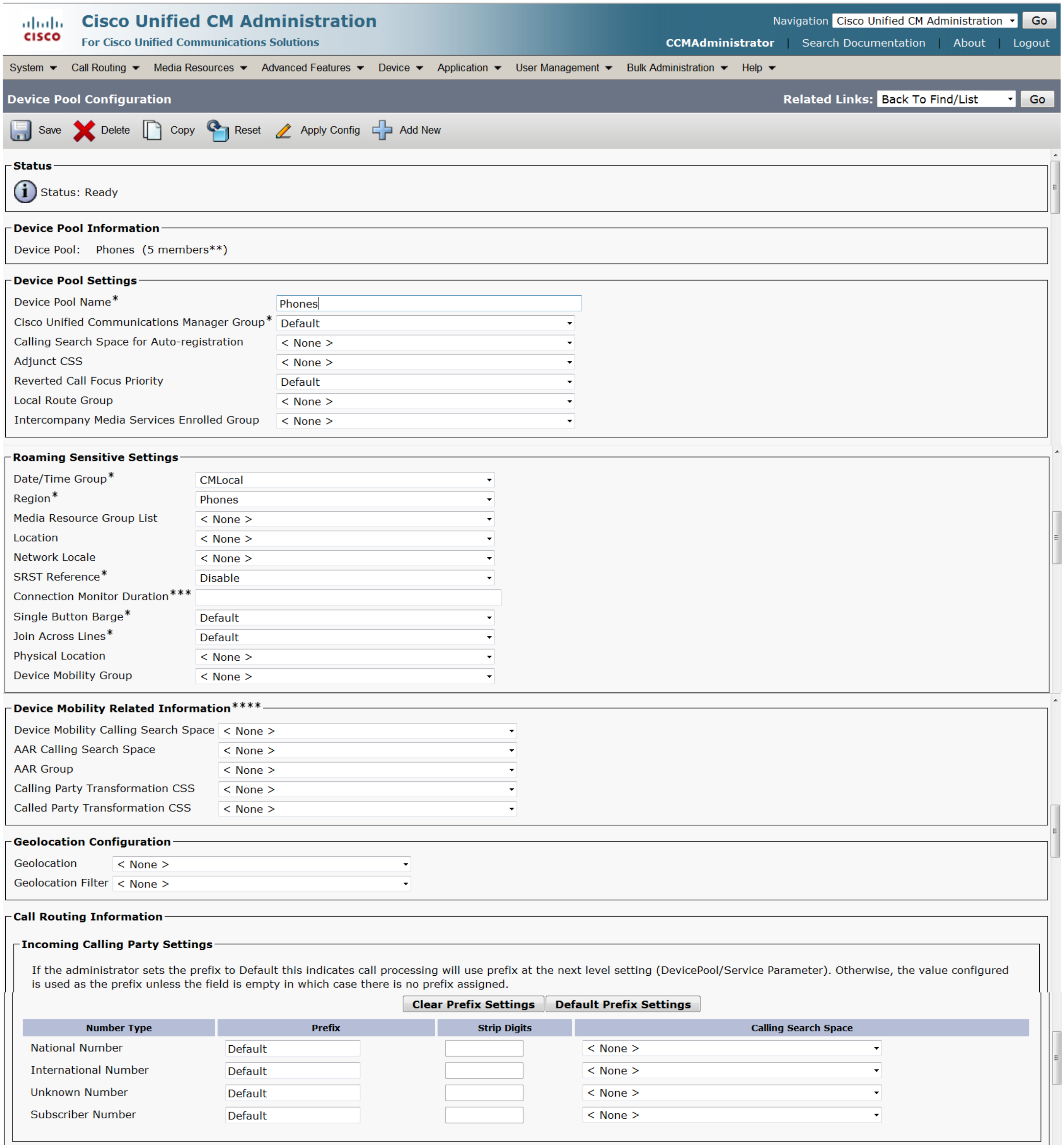
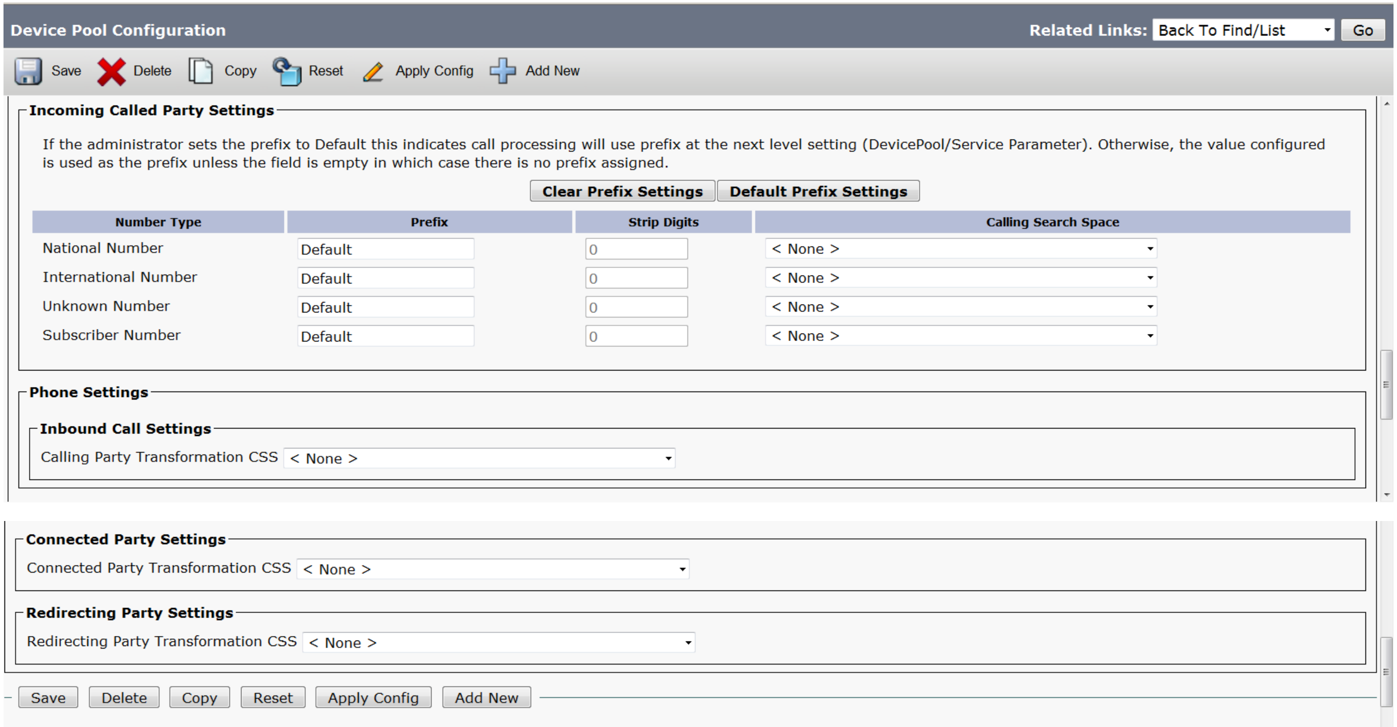
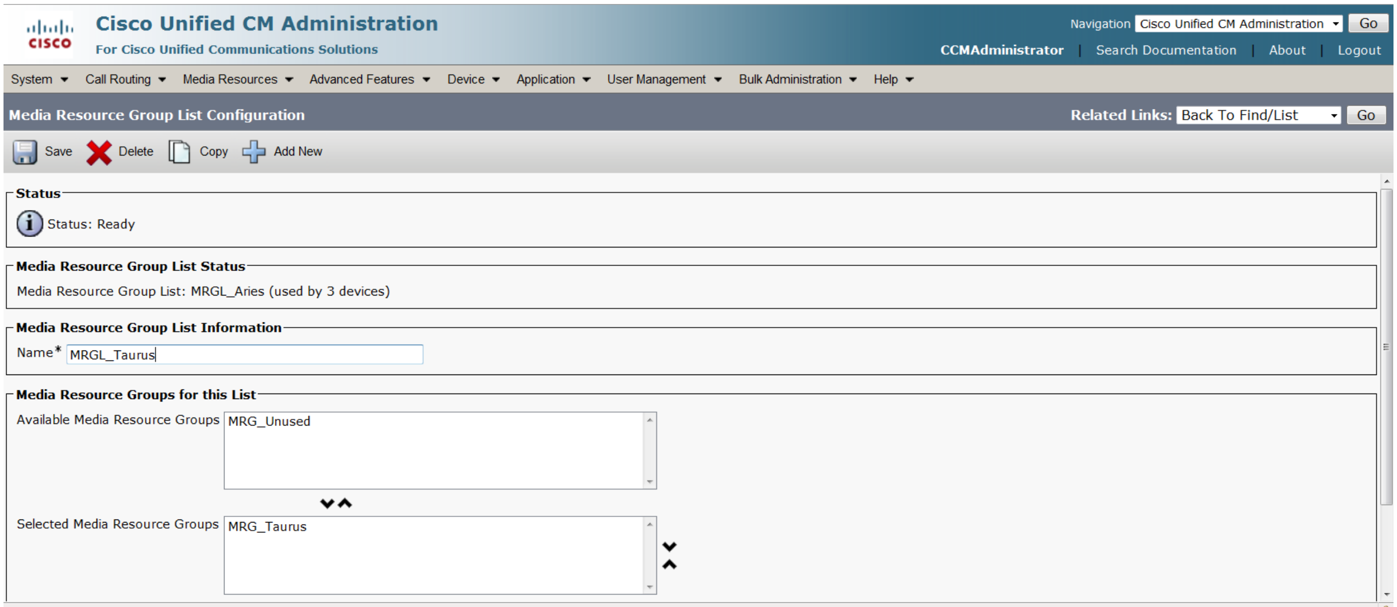
Device Pool Configuration
Related Links: Back To Find/List Go
Save Delete Copy Reset Apply Config Add New
Device Mobility Related Information****
Device Mobility Calling Search Space < None >
AAR Calling Search Space < None >
AAR Group < None >
Calling Party Transformation CSS < None >
Called Party Transformation 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 >
Subscriber Number Default 0 < None >
Phone Settings
Inbound Call Settings
Calling Party Transformation CSS < None >
Connected Party Settings
Connected Party Transformation CSS < None >
Redirecting Party Settings
Redirecting Party Transformation CSS < None >
Save Delete Copy Reset Apply Config Add New

Cisco Unified Communications Manager – Device Pool Configuration (Phones)

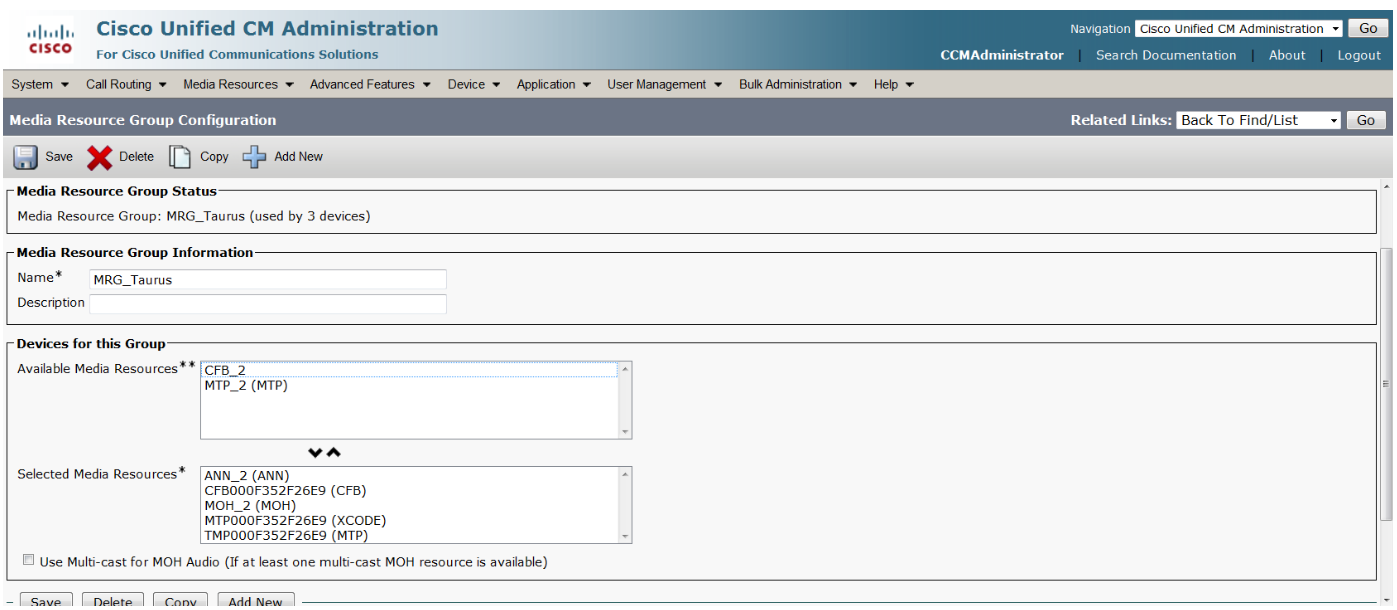



Device Pool Configuration
Related Links: Back To Find/List Go
Save Delete Copy Reset Apply Config Add New
Incoming Called Party Settings
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, 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
International Number
Unknown Number
Subscriber Number
Default
0
< None >
Phone Settings
Inbound Call Settings
Calling Party Transformation CSS < None >
Connected Party Settings
Connected Party Transformation CSS < None >
Redirecting Party Settings
Redirecting Party Transformation CSS < None >
Save Delete Copy Reset Apply Config Add New

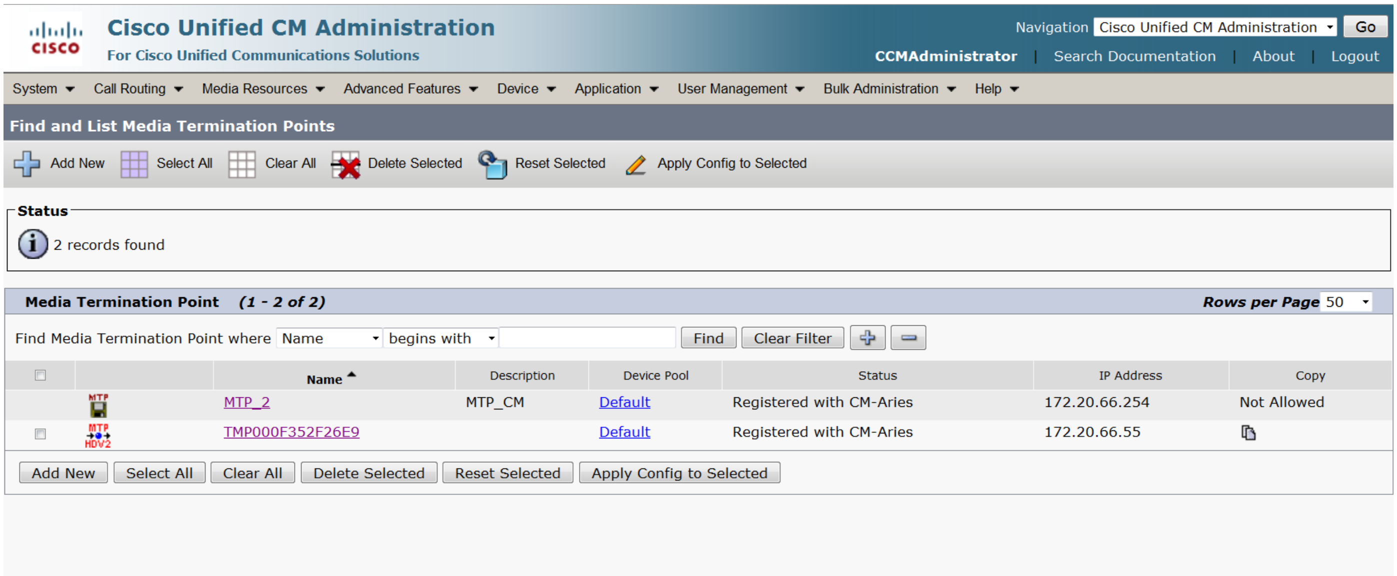
Cisco Unified Communications Manager – Media Resource Group Lists Configuration

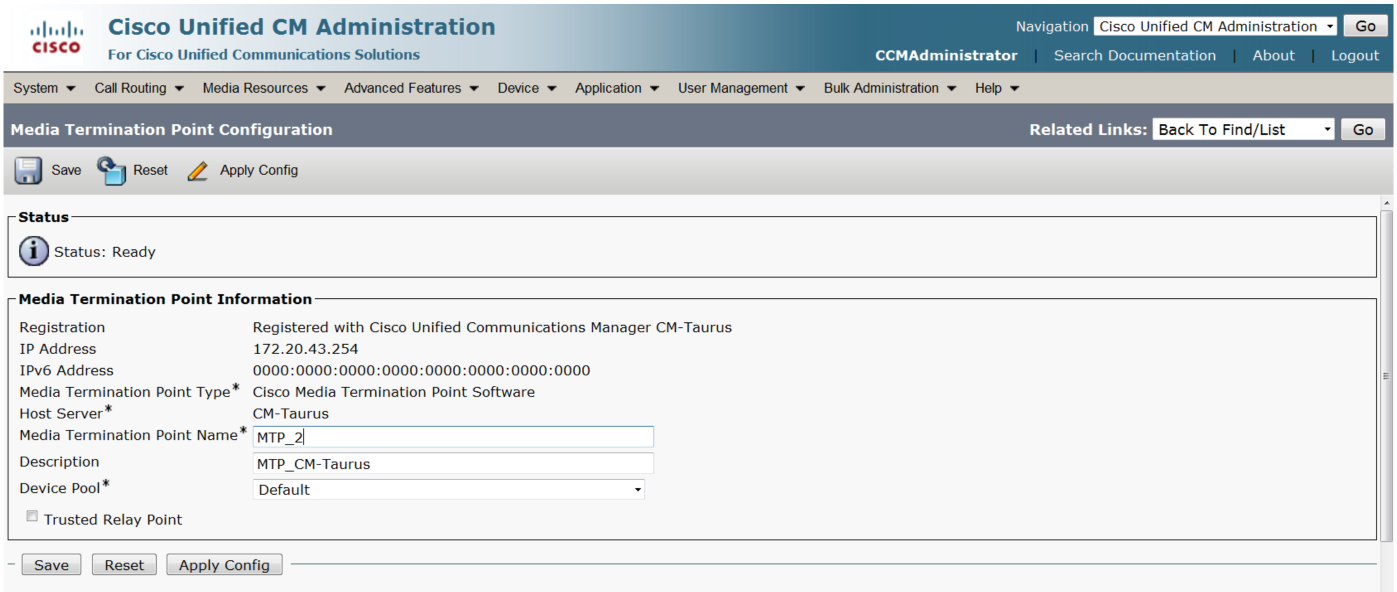


Cisco Unified Communications Manager – Media Resource Group Configuration



Cisco Unified Communications Manager - Media Termination Point Configuration

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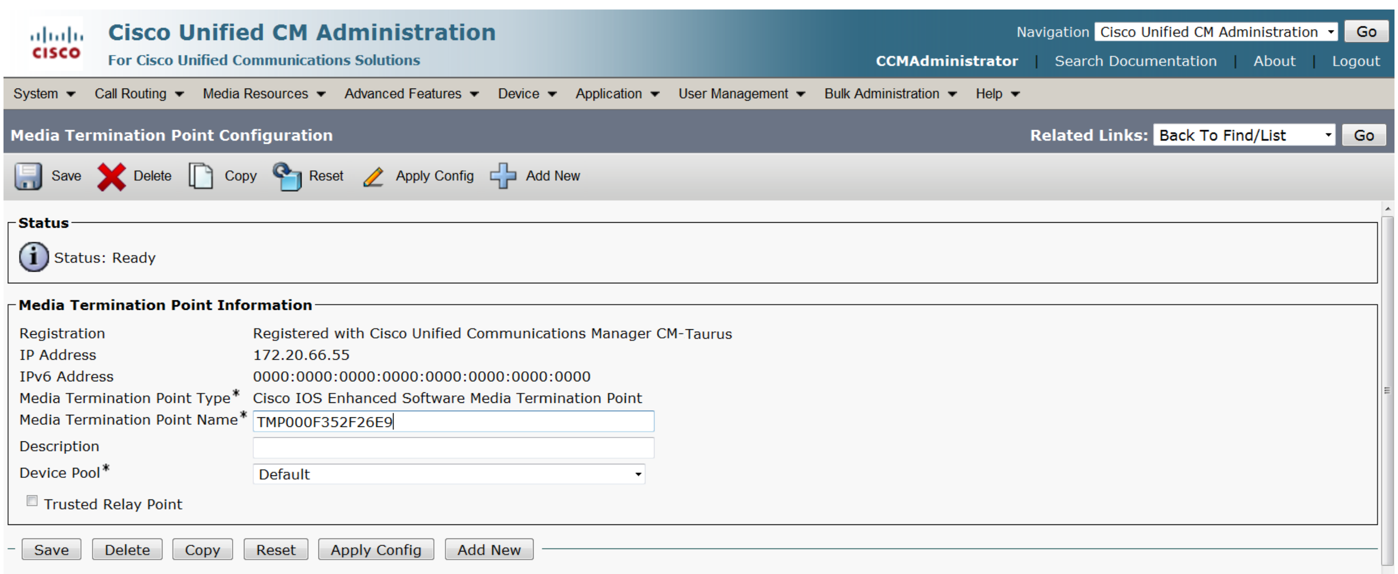
Media Termination Point Configuration
Related Links: Back To Find/List Go
Save Reset Apply Config
Status
Status: Ready
Media Termination Point Information
Registration Registered with Cisco Unified Communications Manager CM-Taurus
IP Address 172.20.43.254
IPv6 Address 0000:0000:0000:0000:0000:0000:0000:0000
Media Termination Point Type* Cisco Media Termination Point Software
Host Server* CM-Taurus
Media Termination Point Name* MTP_2
Description MTP_CM-Taurus
Device Pool* Default
Trusted Relay Point
Save Reset Apply Config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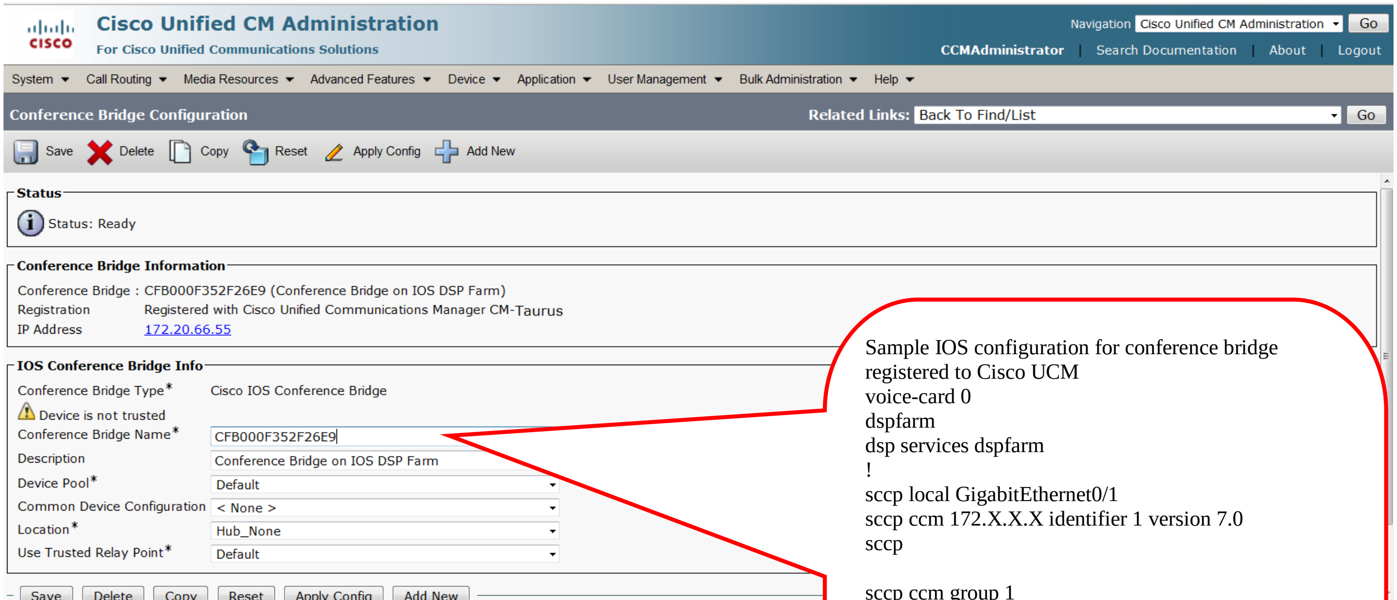
Media Termination Point Configuration
Related Links: Back To Find/List Go
Save Delete Copy Reset Apply Config Add New
Status
Status: Ready
Media Termination Point Information
Registration Registered with Cisco Unified Communications Manager CM-Taurus
IP Address 172.20.66.55
IPv6 Address 0000:0000:0000:0000:0000:0000:0000:0000
Media Termination Point Type* Cisco IOS Enhanced Software Media Termination Point
Media Termination Point Name* TMP000F352F26E9
Description
Device Pool* Default
Trusted Relay Point
Save Delete Copy Reset Apply Config Add New

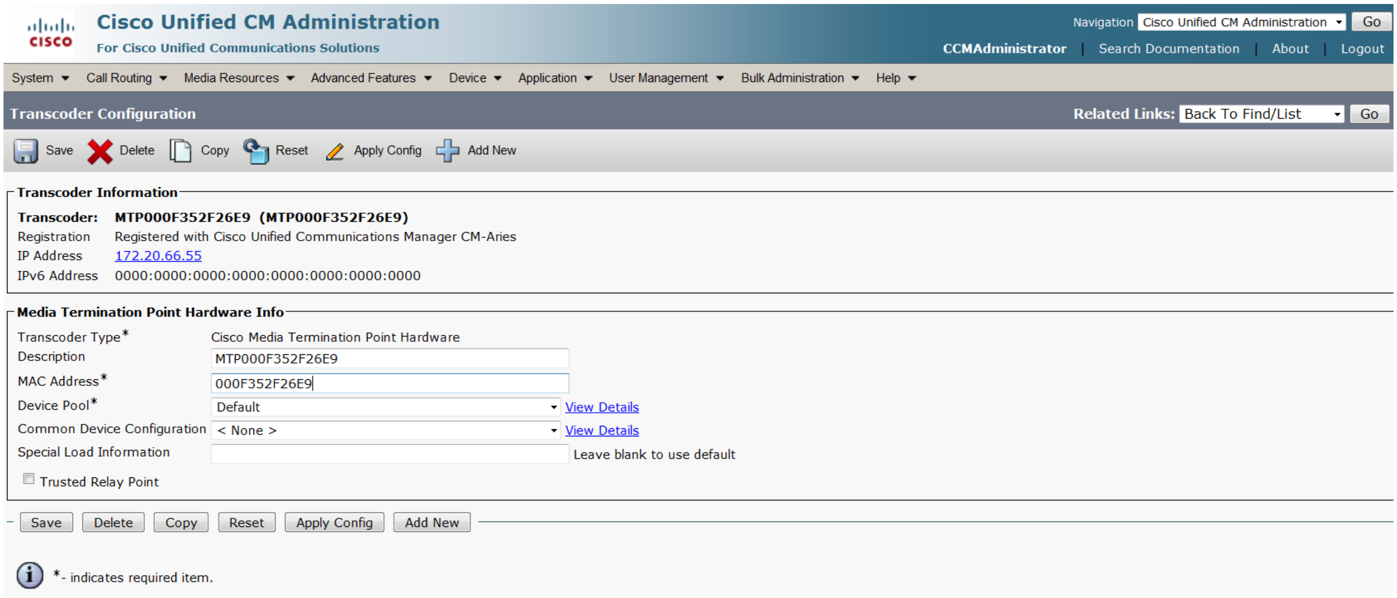
Cisco Unified Communications Manager – Conference Bridge Configuration



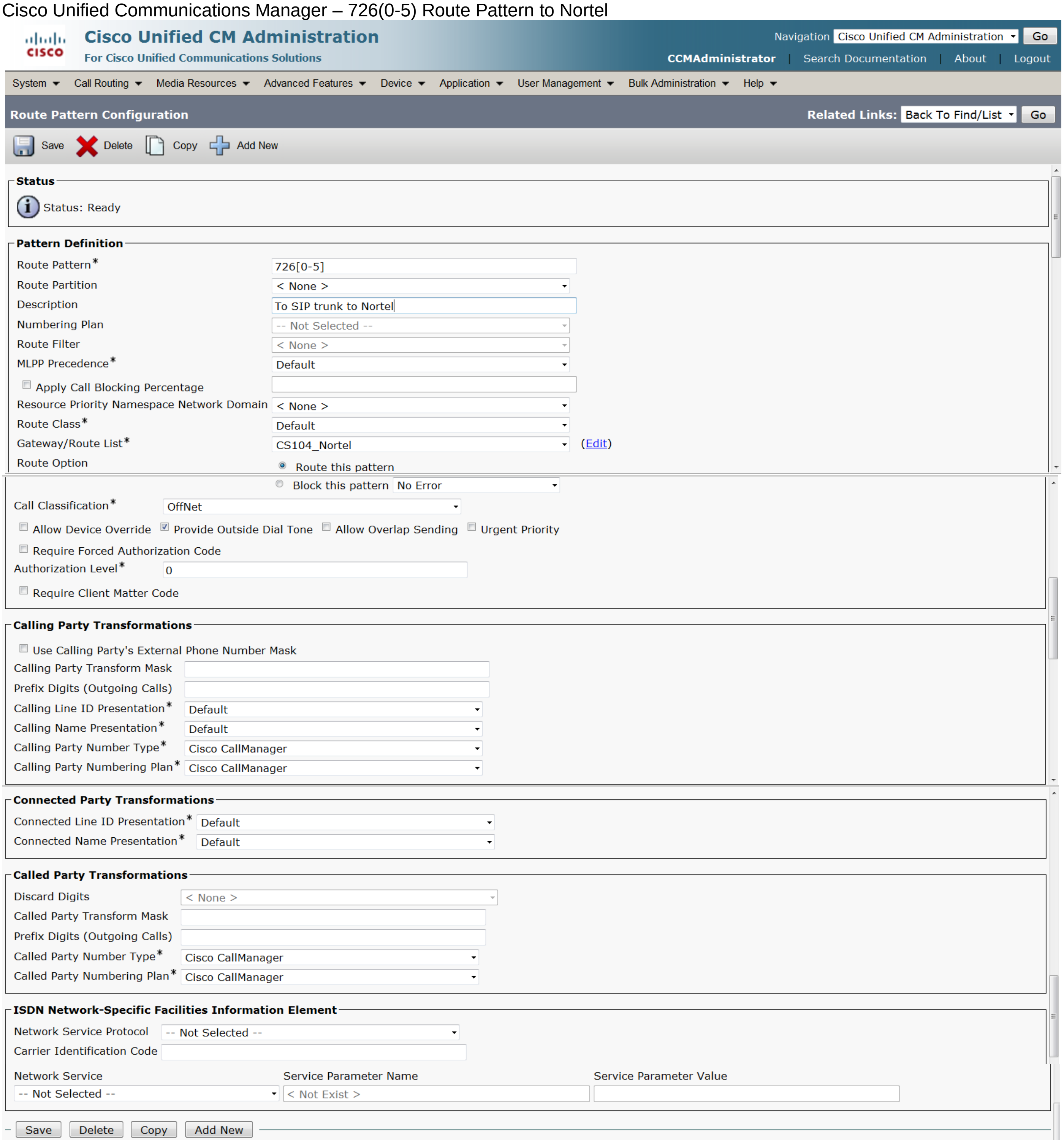
Sample IOS configuration for conference bridge registered to Cisco UCM

```
voice-card 0
dspfarm
dsp services dspfarm
!
sccp local GigabitEthernet0/1
sccp ccm 172.X.X.X identifier 1 version 7.0
sccp

sccp ccm group 1
 associate ccm1 priority 1
 associate profile 1 register CFB000F352F26E9

dspfarm profile 1 conference
 codec g711ulaw
 codec g711alaw
 codec g729ar8
 codec g729abr8
 codec g729r8
 codec g729br8
codec g722-64
maximum sessions 4
associate application SCCP
```

Cisco Unified Communications Manager – Transcoder Configuration



Cisco Unified Communications Manager – 726(0-5) Route Pattern to Nortel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Route Pattern Configuration — Related Links: Back To Find/List | Go

Save | Delete | Copy | Add New

Status

Status: Ready

Pattern Definition

Route Pattern*	726[0-5]
Route Partition	< None >
Description	To SIP trunk to Nortel
Numbering Plan	-- Not Selected --
Route Filter	< None >
MLPP Precedence*	Default
Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	CS104_Nortel (Edit)
Route Option	(selected) Route this pattern / Block this pattern: No Error
Call Classification*	OffNet
Allow Device Override	unchecked
Provide Outside Dial Tone	checked
Allow Overlap Sending	unchecked
Urgent Priority	unchecked
Require Forced Authorization Code	unchecked
Authorization Level*	0
Require Client Matter Code	unchecked

Calling Party Transformations

Use Calling Party's External Phone Number Mask	unchecked
Calling Party Transform Mask	
Prefix Digits (Outgoing Calls)	
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Calling Party Number Type*	Cisco CallManager
Calling Party Numbering Plan*	Cisco CallManager

Connected Party Transformations

Connected Line ID Presentation*	Default
Connected Name Presentation*	Default

Called Party Transformations

Discard Digits	< None >
Called Party Transform Mask	
Prefix Digits (Outgoing Calls)	
Called Party Number Type*	Cisco CallManager
Called Party Numbering Plan*	Cisco CallManager

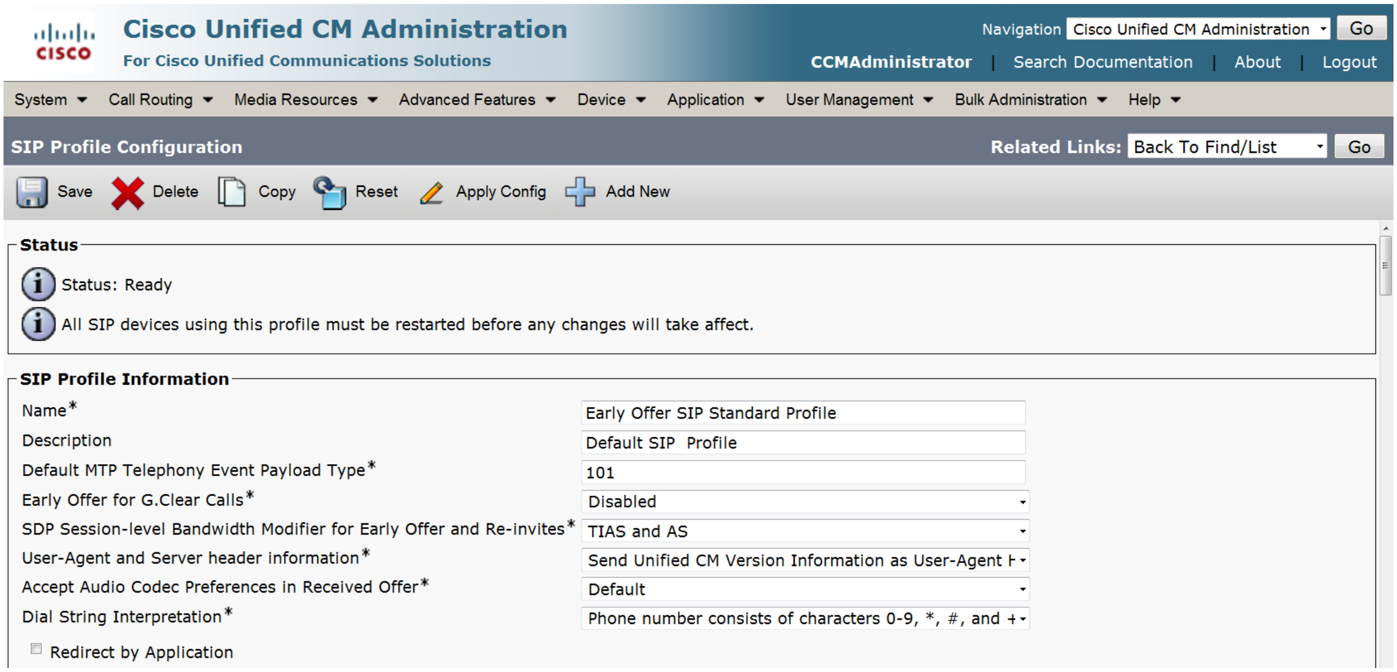
ISDN Network-Specific Facilities Information Element

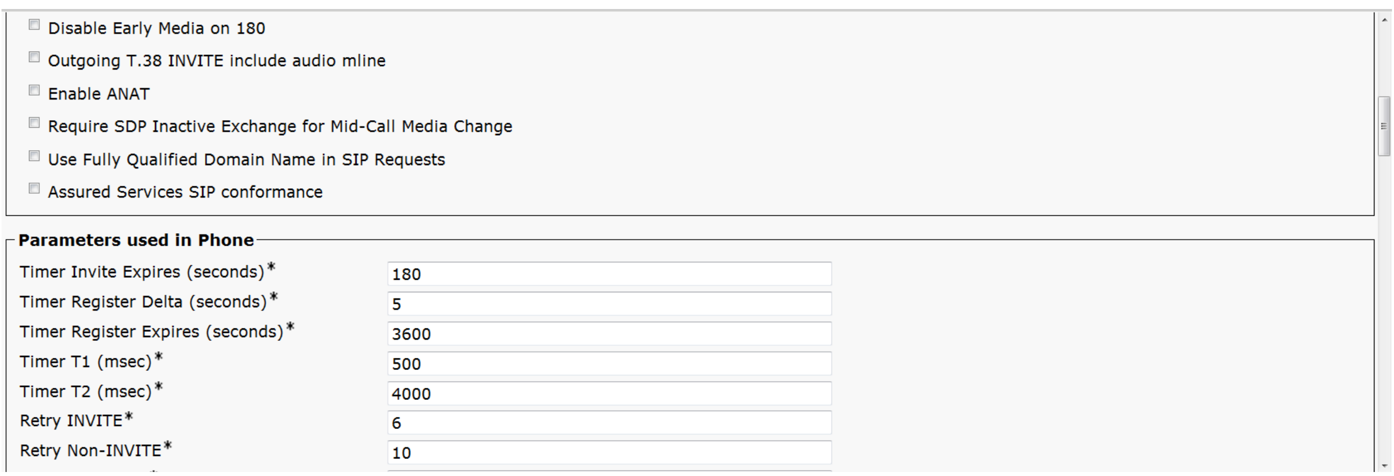
Network Service Protocol	-- Not Selected --
Carrier Identification Code	

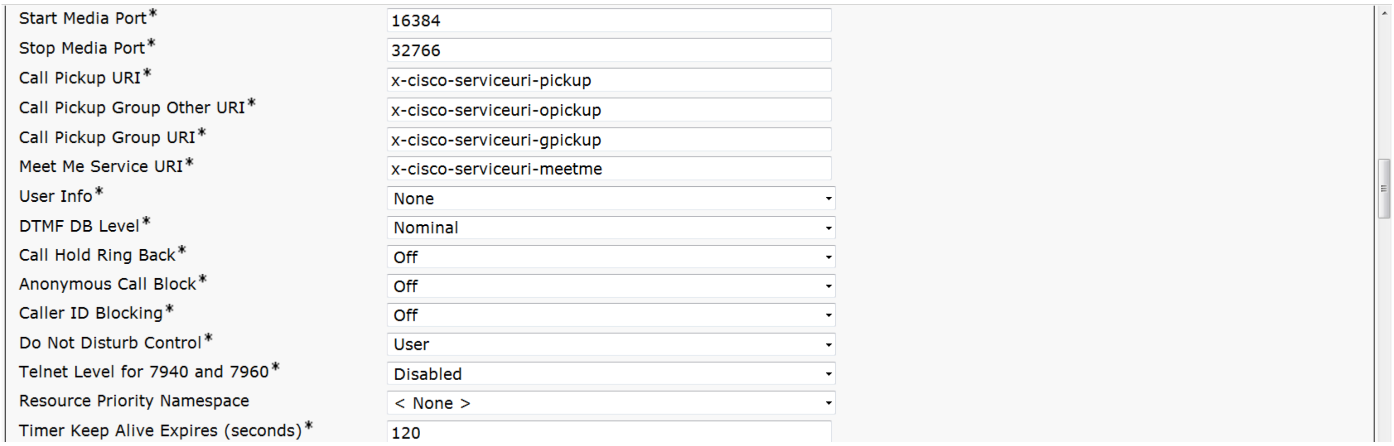
Network Service	Service Parameter Name	Service Parameter Value
-- Not Selected --	< Not Exist >	

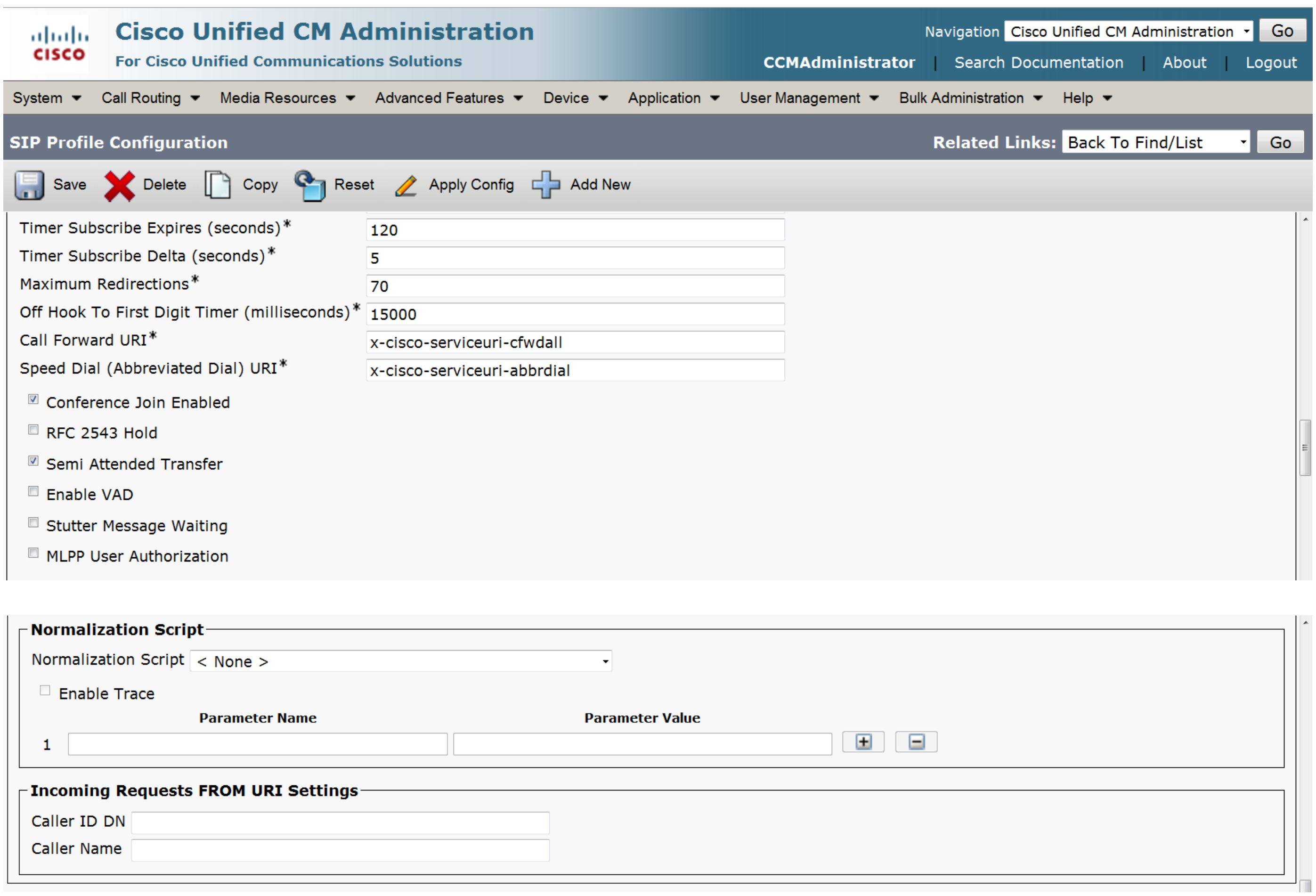
Save | Delete | Copy | Add New

Cisco Unified Communications Manager – SIP Profile (DO Standard SIP Profile with PRACK)






Start Media Port* 16384
Stop Media Port* 32766
Call Pickup URI* x-cisco-serviceuri-pickup
Call Pickup Group Other URI* x-cisco-serviceuri-opickup
Call Pickup Group URI* x-cisco-serviceuri-gpickup
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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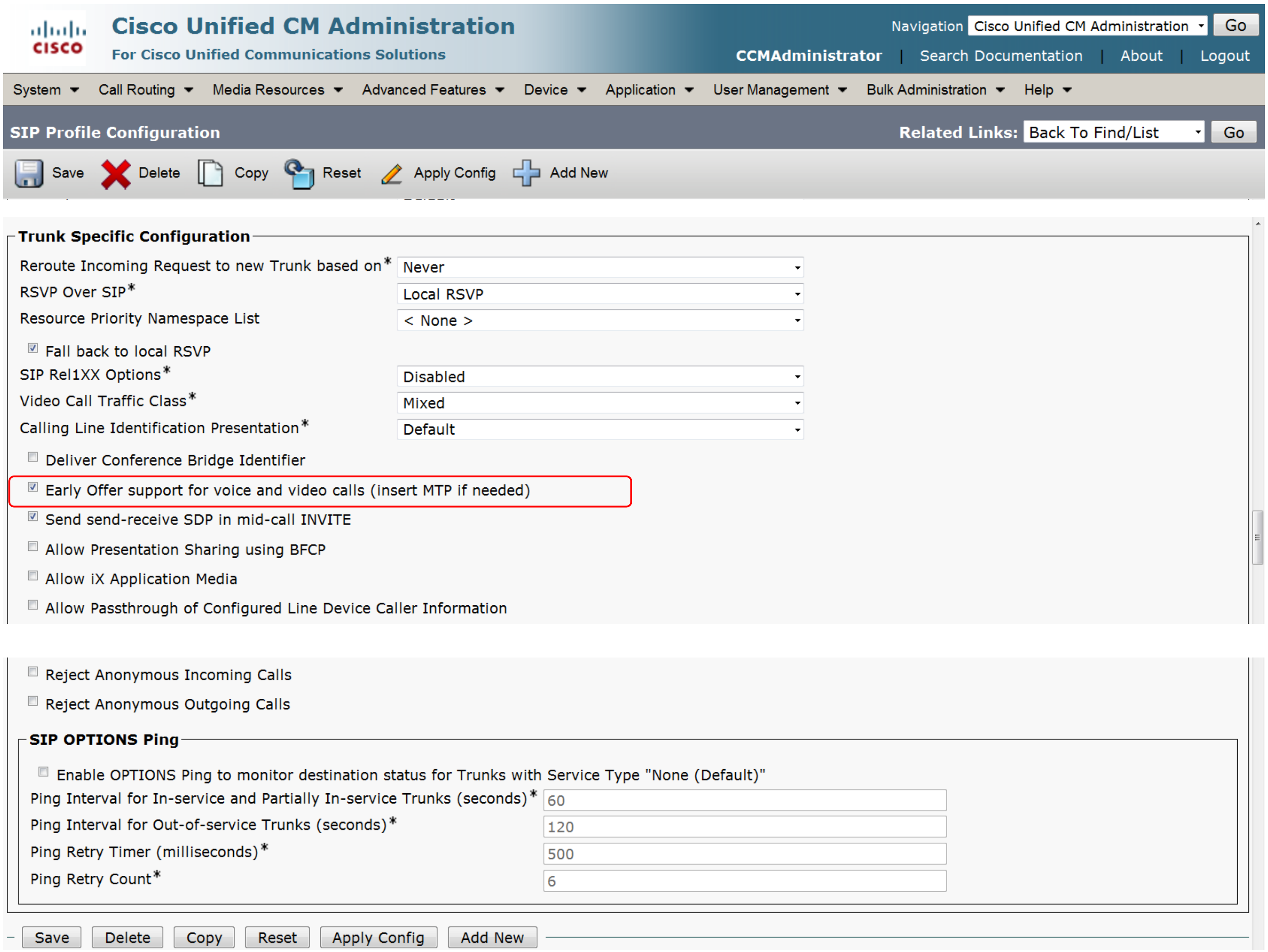
SIP Profile Configuration
Related Links: Back To Find/List Go
Save Delete Copy Reset Apply Config Add New
Timer Subscribe Expires (seconds)* 120
Timer Subscribe Delta (seconds)* 5
Maximum Redirections* 70
Off Hook To First Digit Timer (milliseconds)* 15000
Call Forward URI* x-cisco-serviceuri-cfwdall
Speed Dial (Abbreviated Dial) URI* x-cisco-serviceuri-abbrdial
Conference Join Enabled
RFC 2543 Hold
Semi Attended Transfer
Enable VAD
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


Cisco Unified CM Administration
For Cisco Unified Communications Solutions
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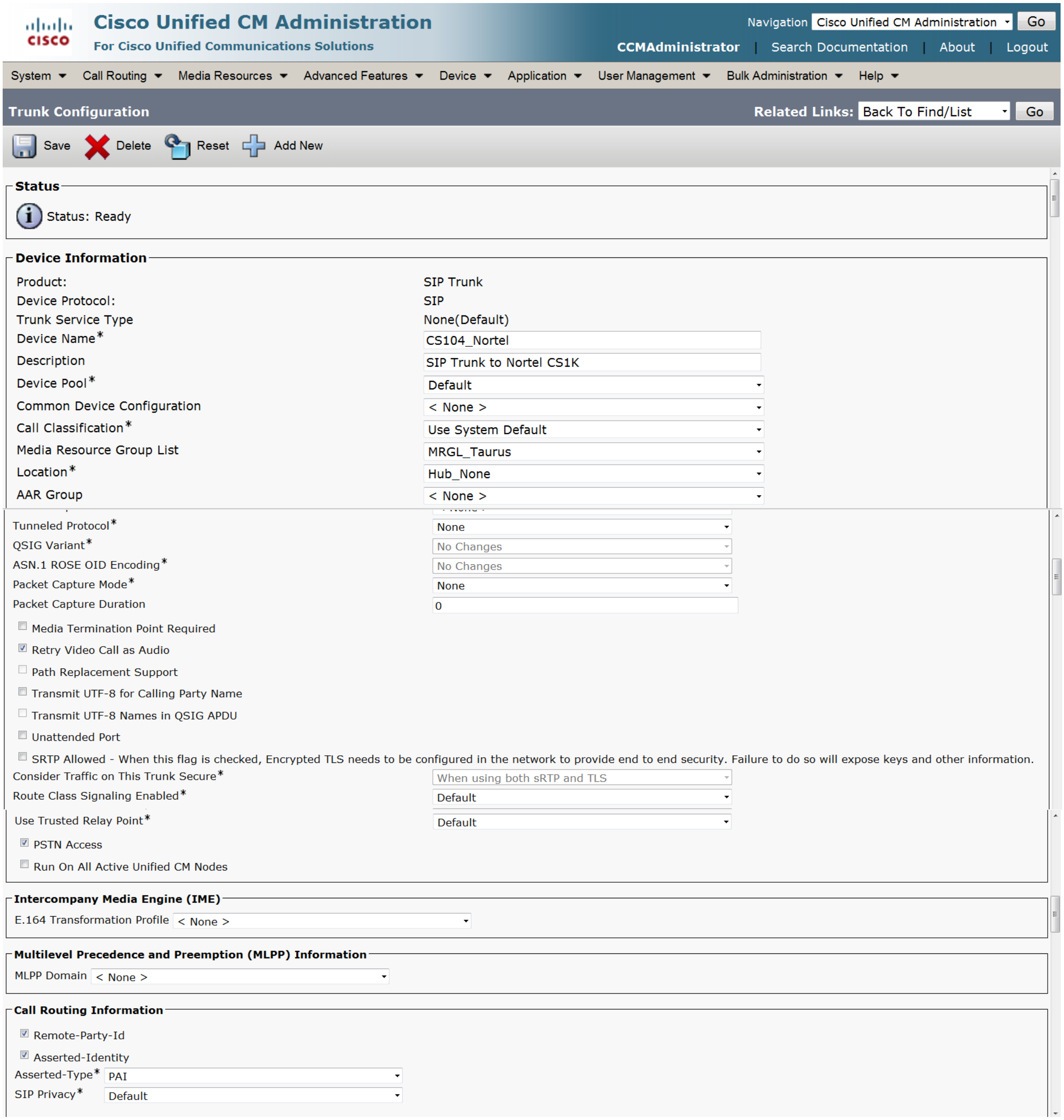
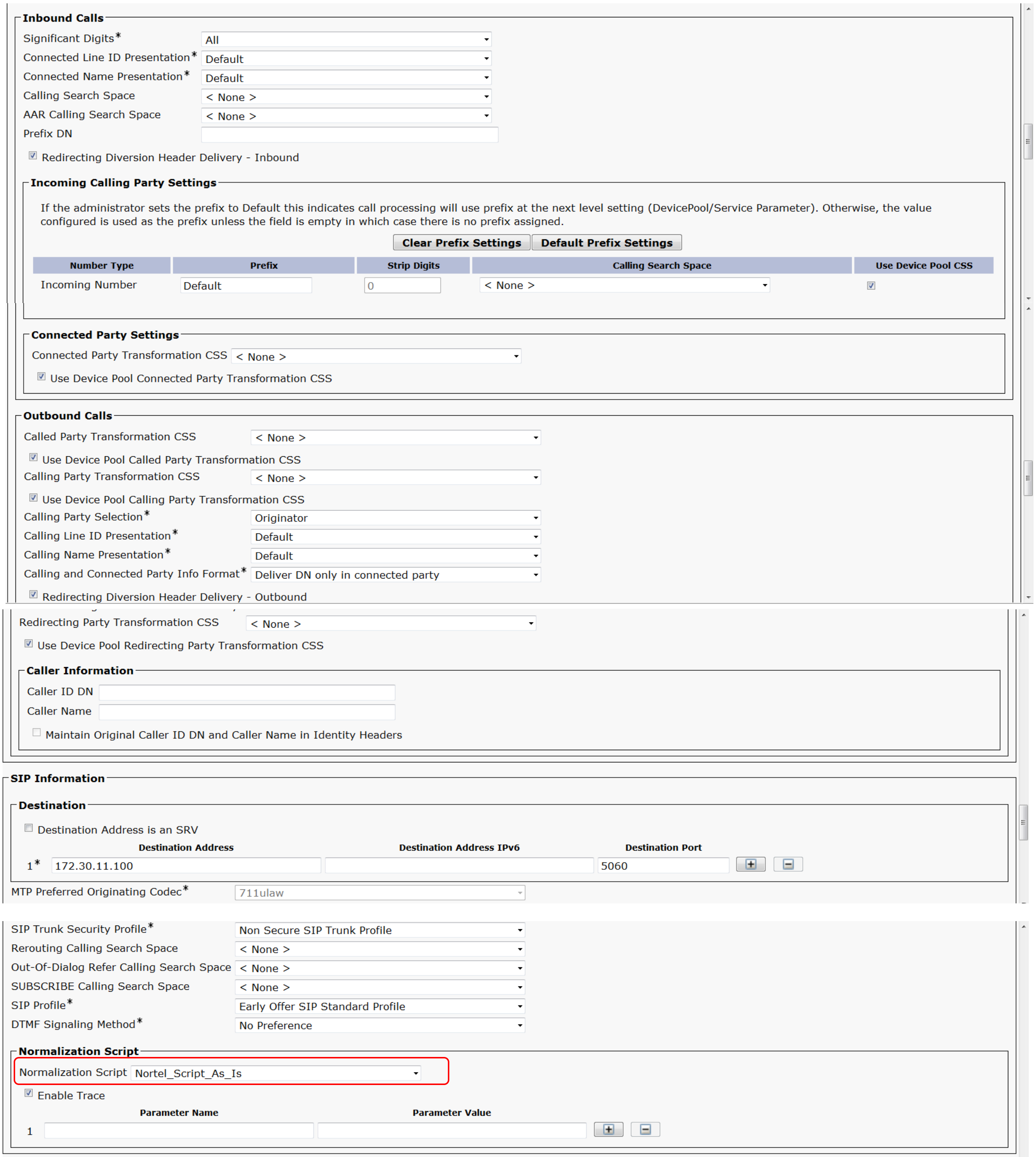
CCMAdministrator | Search Documentation | About | Logout
System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help
SIP Profile Configuration
Related Links: Back To Find/List Go
Save Delete Copy Reset Apply Config Add New
Trunk Specific Configuration
Reroute Incoming Request to new Trunk based on* Never
RSVP Over SIP* Local RSVP
Resource Priority Namespace List < None >
Fall back to local RSVP
SIP Rel1XX Options* Disabled
Video Call Traffic Class* Mixed
Calling Line Identification Presentation* Default
Deliver Conference Bridge Identifier
Early Offer support for voice and video calls (insert MTP if needed)
Send send-receive SDP in mid-call INVITE
Allow Presentation Sharing using BFCP
Allow iX Application Media
Allow Passthrough of Configured Line Device Caller Information
Reject Anonymous Incoming Calls
Reject Anonymous Outgoing Calls
SIP OPTIONS Ping
Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"
Ping Interval for In-service and Partially In-service Trunks (seconds)* 60
Ping Interval for Out-of-service Trunks (seconds)* 120
Ping Retry Timer (milliseconds)* 500
Ping Retry Count* 6
Save Delete Copy Reset Apply Config Add New

Cisco Unified Communications Manager - SIP trunk to Nortel PBX Configuration





Inbound Calls

Significant Digits* All
Connected Line ID Presentation* Default
Connected Name Presentation* Default
Calling Search Space < None >
AAR Calling Search Space < None >
Prefix DN

☑ Redirecting Diversion Header Delivery - Inbound

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	Use Device Pool CSS
Incoming Number	Default	0	< None >	☑

Connected Party Settings

Connected Party Transformation CSS < None >

☑ Use Device Pool Connected Party Transformation CSS

Outbound Calls

Called Party Transformation CSS < None >
☑ Use Device Pool Called Party Transformation CSS
Calling Party Transformation CSS < None >
☑ Use Device Pool Calling Party Transformation CSS
Calling Party Selection* Originator
Calling Line ID Presentation* Default
Calling Name Presentation* Default
Calling and Connected Party Info Format* Deliver DN only in connected party
☑ Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS < None >
☑ Use Device Pool Redirecting Party Transformation CSS

Caller Information

Caller ID DN
Caller Name
☐ Maintain Original Caller ID DN and Caller Name in Identity Headers

SIP Information

Destination

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1*	172.30.11.100		5060

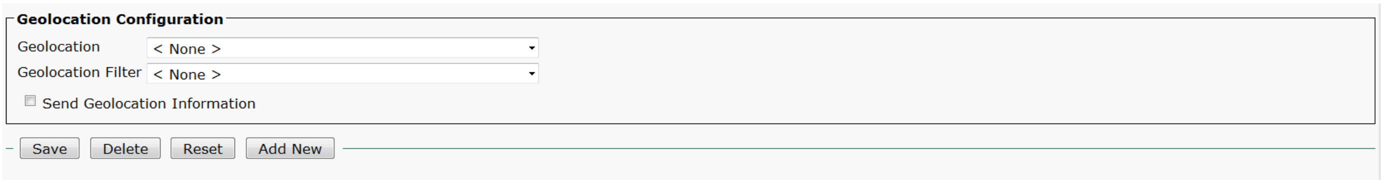
MTP Preferred Originating Codec* 711ulaw

SIP Trunk Security Profile* Non Secure SIP Trunk Profile
Rerouting Calling Search Space < None >
Out-Of-Dialog Refer Calling Search Space < None >
SUBSCRIBE Calling Search Space < None >
SIP Profile* Early Offer SIP Standard Profile
DTMF Signaling Method* No Preference

Normalization Script

Normalization Script Nortel_Script_As_Is
☑ Enable Trace

	Parameter Name	Parameter Value
1		

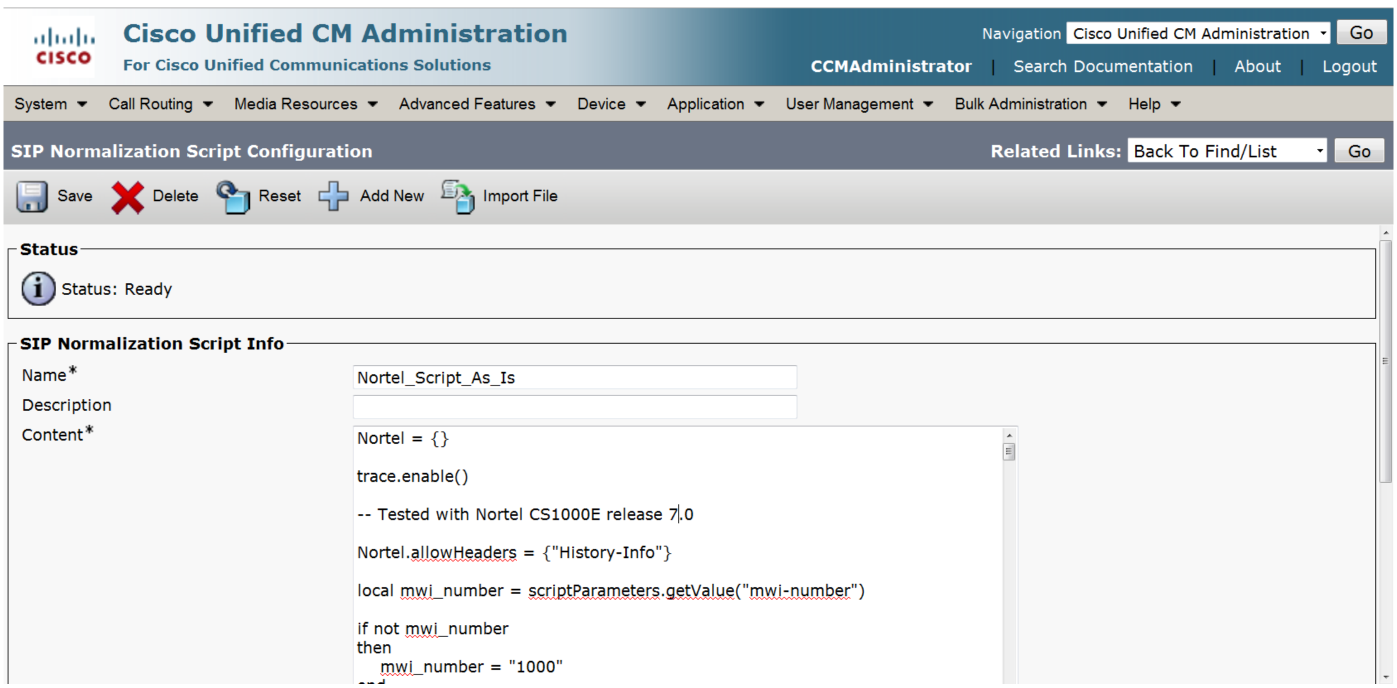

Geolocation Configuration
Geolocation
< None >
Geolocation Filter
< None >
Send Geolocation Information
Save
Delete
Reset
Add New

Cisco UCM- SIP Normalization Configuration

1-Copy the SIP Normalization Script (below) and paste it to a note pad and save as text. Make sure that the script is in text format and not corrupted.

2- Access to the SIP Normalization Script Configuration in Cisco UCM and upload to the Script. Give the loaded script a short name so that the name will be assigned to the SIP Trunk.

3- Assign the script name to the SIP Trunk in the SIP Trunk configuration-SIP Normalization section.



Cisco UCM- Software Script

The Cisco UCM-Software Script should be applied at SIP trunk toward Nortel PBX. This script normalizes the SIP messaging to/from the Nortel 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. Copy and paste to note pad and save it using txt format. Review the script to ensure that it is not corrupted before upload to the normalization section.

****

```
Nortel = {}

trace.enable()

-- Tested with Nortel CS1000E release 7.0

Nortel.allowHeaders = {"History-Info"}

local mwi_number = scriptParameters.getValue("mwi-number")

if not mwi_number
then
   mwi_number = "1000"
end

local function adjustRedirectInfo(msg)
   local di = msg:getHeader("Diversion")
   if not di
   then
      return
   end

   msg:convertDiversionToHI()

   msg:removeHeader("Diversion")

   local historyInfos = msg:getHeaderValues("History-Info")
   msg:removeHeader("History-Info")

-- For debugging purposes, dump out what the Diversion header contained and dump out the list of History-Info headers
-- produced by msg:convertDiversionToHI. These extra headers will help debug but should be ignored by Nortel. Trace
-- should be disabled via Admin UI unless a problem is being debugged. Therefore, under normal operating conditions,
-- the debug headers won't be included in the message.
   if trace.enabled()
   then
      msg:addHeader("X-Debug-Diversion", di)
      for i, hi in ipairs(historyInfos)
      do
         msg:addHeader("X-Debug-History-Info", hi)
      end
   end

-- Example:
--    Original Diversion header generated natively by CUCM might have been this:
--       Diversion: <sip:1002@10.10.10.100>;reason=unconditional;privacy=off;screen=yes
--
--    The call to convertDiversionToHI will produce these:
```

```
--        History-Info: <sip:1002@10.10.10.100:5060?Reason=sip;cause=302;text="unconditional">;index=1
--        History-Info: <sip:2400@10.10.10.200:5060>;index=1.1
--
--     However, Nortel needs something that looks like this:
--        History-Info:<sip:1002@10.10.10.100?reason=sip%3Bcause%3D302%3Btext%3D%22Moved%20Temporarily%22>;index=1
--        History-Info: <sip:2400@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.
        local embed_header = string.match (hi, '%?Reason=sip(.*)>')

        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.
```

```
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

return Nortel
```

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
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

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