Application Note

Nortel CS1000 Communications Server 4.0 to Cisco Unified Communications Manager 4.2 using Cisco Multi-service IP-to-IP Gateway with SIP-to-H.323

July 17, 2007 Initial Version

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Introduction

This is an application note for connectivity of Nortel CS1000 Communications Server 4.0 with Cisco Unified Communications Manager 4.2 using a Cisco Multiservice IP-to-IP Gateway via SIP and H.323 (10/100baseT).

The Multiservice IP-to-IP Gateway offers the following advantages:

Security - protects Cisco Unified Communications Manager from Nortel CS1000 & vice versa via IP topology hiding

Protocol inter-working from H.323 (Cisco Unified Communications Manager) to SIP (Nortel CS1000 PBX)

Co-resident Media Termination Point (MTP) with Cisco Unified Communications Manager (required for H.323 Fast Start calls with non-G711 codec)

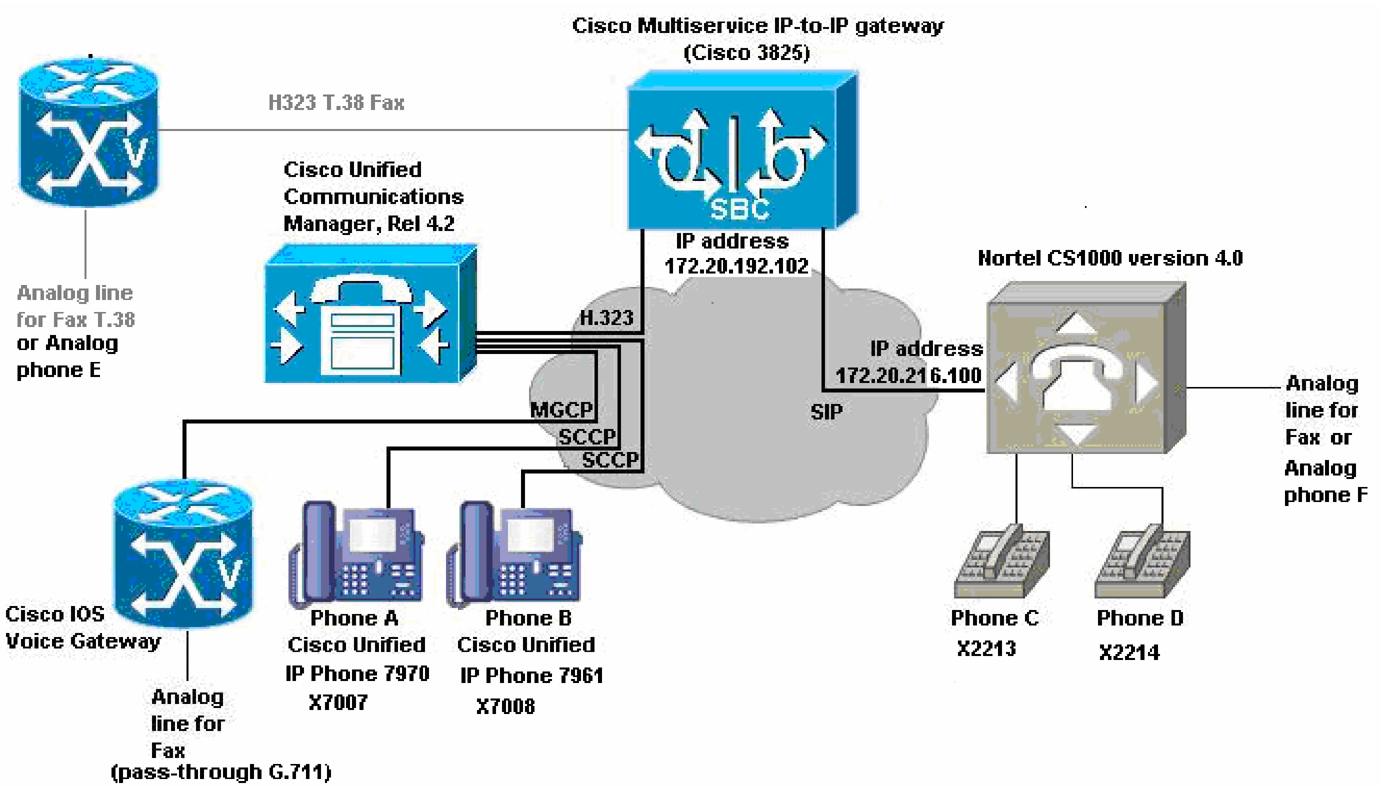
The network topology diagram (Figure 1) shows the test setup for end-to-end interoperability with the Cisco Multiservice IP-to-IP Gateway connected to the IP PBX via SIP (10/100baseT). Connectivity is achieved by using the SIP and H.323 protocols.

This Application Note uses the C3825 IOS-voice-gateway, however other Cisco voice gateways are also an option to use since IPIPGW implementation does not depend on the platform. Here is a list of Cisco Products capable of IPIPGW functionality:

Cisco 2800 Series Integrated Services Routers
Cisco 3800 Series Integrated Services Routers
Cisco 2600XM Series Multiservice Platforms
Cisco 3700 Series Routers
Cisco 7200VXR Routers
Cisco 7301 Routers
Cisco AS5350XM Universal Gateway
Cisco AS5400XM Universal Gateway

Network Topology

Figure 1. Network Topology or Test setup



Limitations

Connected Name and number (dialled number is presented) are not supported accross H323/SIP trunk. Nortel use SIP P-asserted-ID method to send name and number. Cisco IP-to-IP Gateway (IPIPGW) currently does not support SIP P-asserted ID feature

Basic Call using G.726 codec fails. Nortel does not support G.726 codec

For codec G.711 faststart, Cisco Unified Communications Manager (CUCM) can use Local software MTP, but for basic calls using G.729 faststart, it requires an IOS hardware MTP which is co-resident on the Multi-service IP-to-IP gateway.

Basic Call using G.723 codec fails. Cisco Unified Communications Manager 4.2 will not support other codec besides G.711 on local software MTP, and Cisco IOS hardware MTP will not support G.723 codec.

Call Transfer and Call Forward Name and Number updates do not occur consistently.

Nortel CS1000 release 4.0 requires INVITE with early offer for 2 way voice to be established. H323 call leg must be setup for faststart

DTMF in-band (G.711) does not inter-operate in the direction from Nortel succession PBX toward Cisco Unified Communications Manager 4.2. The Nortel PBX uses SIP INFO message to relay detected DTMF tone at the SIP trunk, CUCM does not support SIP INFO messages. DTMF relay using RFC2833 is not supported by Nortel. However, analog to analog calls support DTMF tone generation in both directions using in-band DTMFsignaling.

Fax Comparability using T.38 protocol requires an additional H.323 gateway with a FXS port between Fax device and IP to IP gateway. The H.323 gateway will communicate T.38 protocol across the network.

Three way conference call using G.729 between call leg CUCM and Nortel PBX requires a hardware MTP Transcoder for codec G.711 and G729 conversion, this requires a second IOS hardware MTP (additional DSP).

Caller ID restricted fails on the direction from CUCM to Nortel CS1000: The Nortel CS1000 does not support the “Remote-party-ID” SIP header. The Nortel looks into the “From” header to determine Calling ID presentation. IP-to-IP Gateway receives calling number restricted from CUCM H.225 message, IPIPGW sends the INVITE message to Nortel with “From” header encoded "anonymous", but still keeps the calling number in the URI-address and the SIP from address header. The Nortel CS1000 only restricts calling name, but because number is still visible the CS1000 presents it to called station.

System Components

Hardware Requirements

Cisco Hardware

Cisco MCS-7800 CCM (4.2 release)
Cisco 3845 Gateway
Cisco 3825 Gateway
Cisco 2801 Gateway
Cisco Catalyst 3550 Power Ethernet switch.
Two Cisco Unified IP phones 7960 and one analog phone.

Third vendor PBX

Nortel Communication System Succession 1000 which includes
Call Server
Signaling Server and
Media gateway
Two Nortel digital stations 2616
One analog station

Miscellaneous

2 – Fax Machines: HP Office Jet 5610xi

Software Requirements

Cisco IOS Software releases: c3825adventerprisek9_ivs-mz.124-11.T1
PBX Software: Nortel Succession 4.0 Release
Cisco Unified Communication Manager Software Release 4.2

Features

Features Supported

SIP call establishment with TCP or UDP

Codec G.711 Ulaw and Alaw and codec G.729

Calling name and number

Call Transfer blind and Call Transfer supervised

Call Conference

FAX integrity (T.38 FAX relay and G711 pass-through) T.38 requires external gateway connected to IP to IP gateway via H.323

Call on-hold

Call Forward No Reply

Call Forward all

Call Forward Busy

CAC threshold

DTMF - In-band (G.711). CUCM analog phone to PBX analog Phone only.

Features Not Supported

Codec G.723 and G.726

Connected Name

Calling Number Restriction (in the direction from CUCM to Nortel CS1000)

DTMF in-band (G711) with Nortel digital phones

DTMF (RFC2833)

Configuration

Configuring Cisco Unified Communications Manager 4.2

Figure 2. Default Region

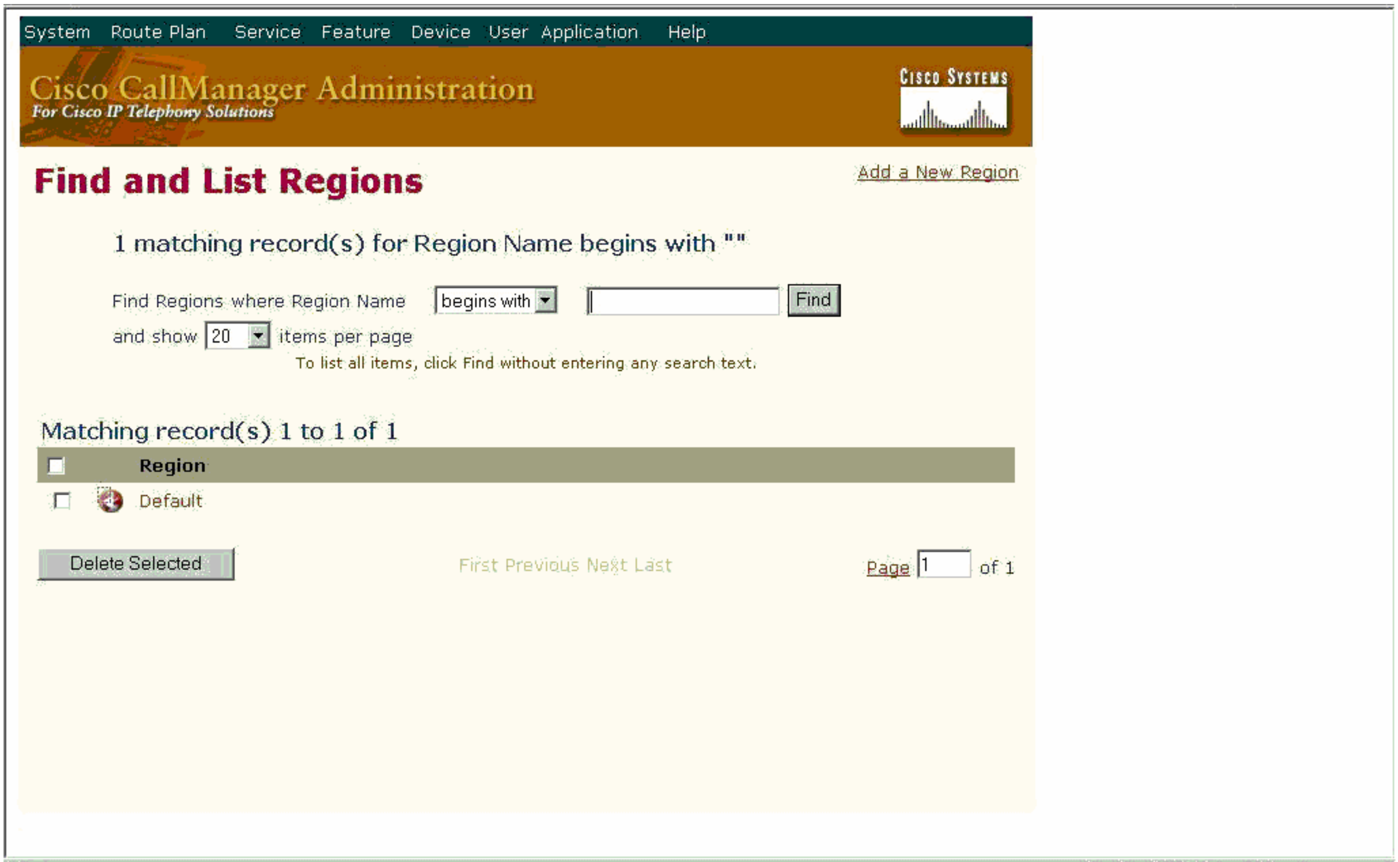


Figure 3. Default Region Detail (Cisco Unified Communications Manager Regional codec = G.711)

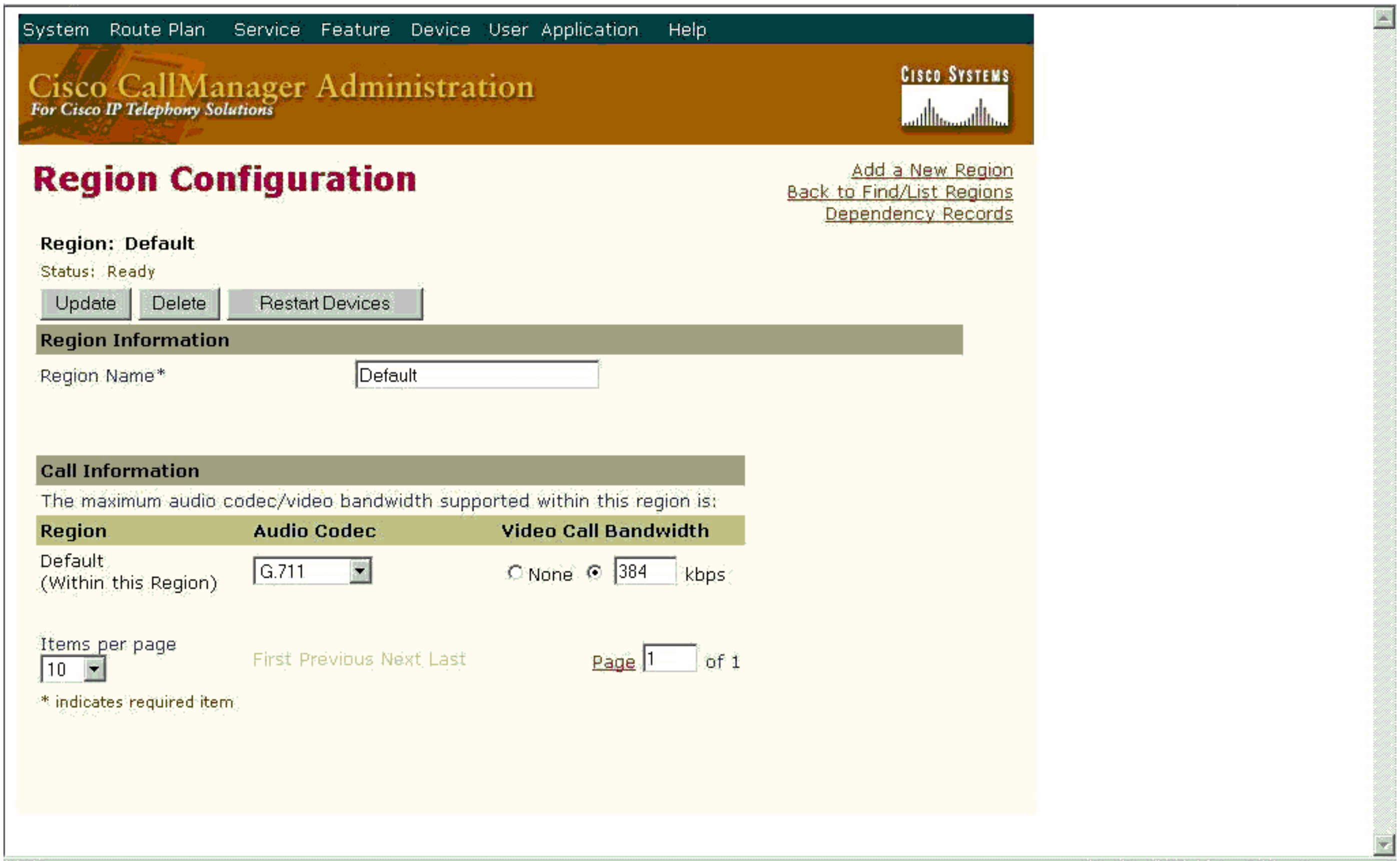


Figure 4. Region configuration. Ri_region set for G.729 Codec.

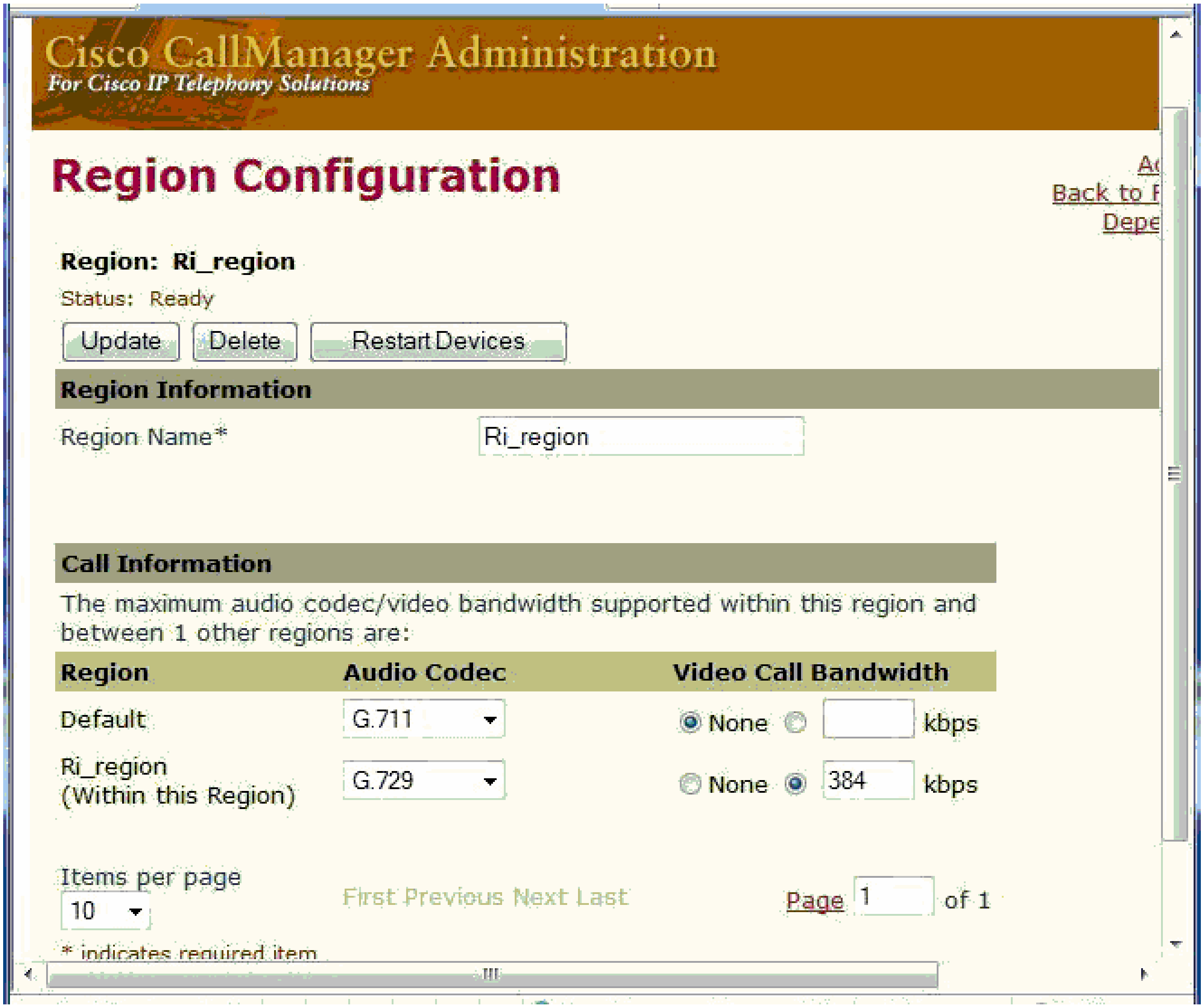
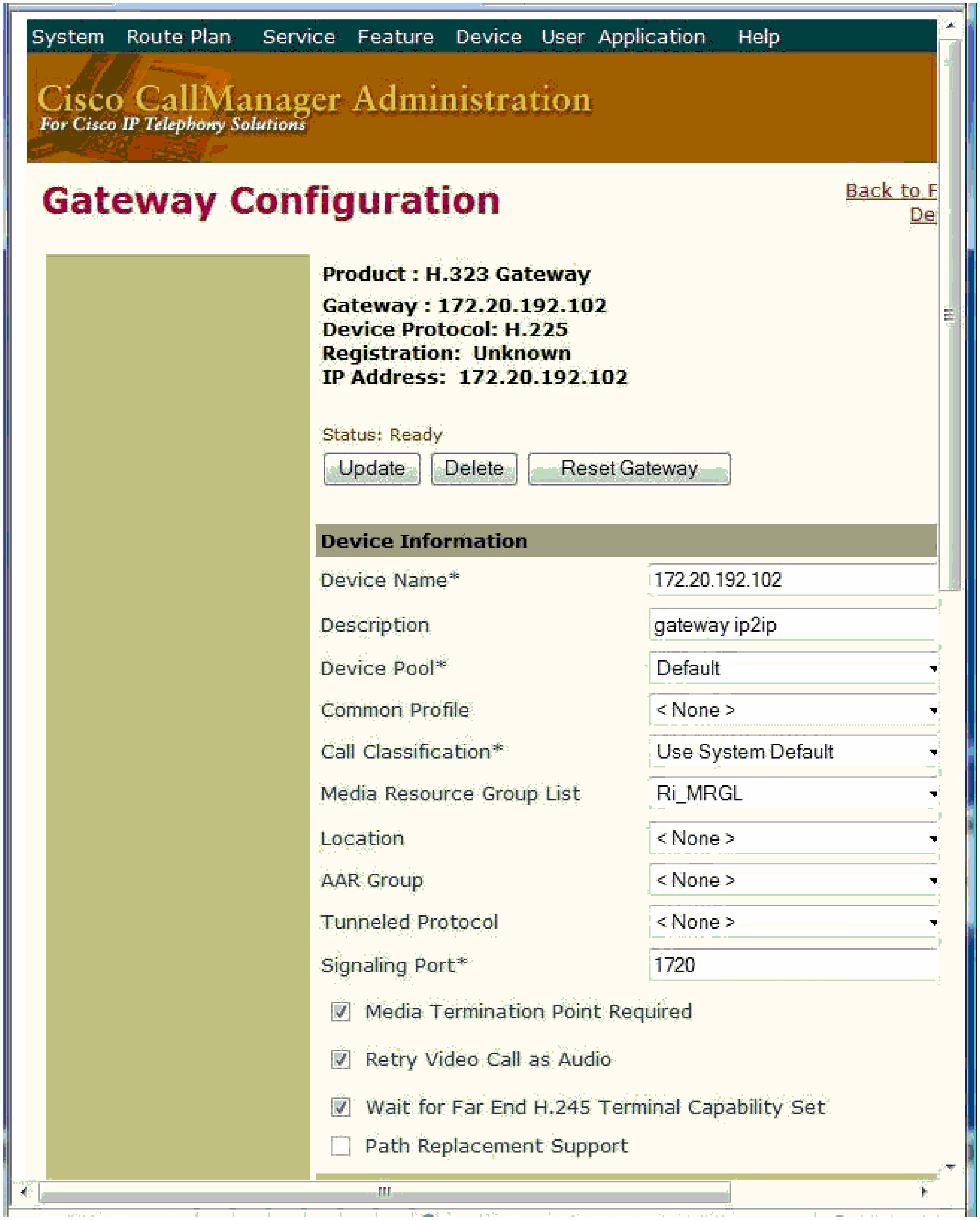


Figure 5. Configurations Page 1 of 2



Configuring the IP -to- IP gateway

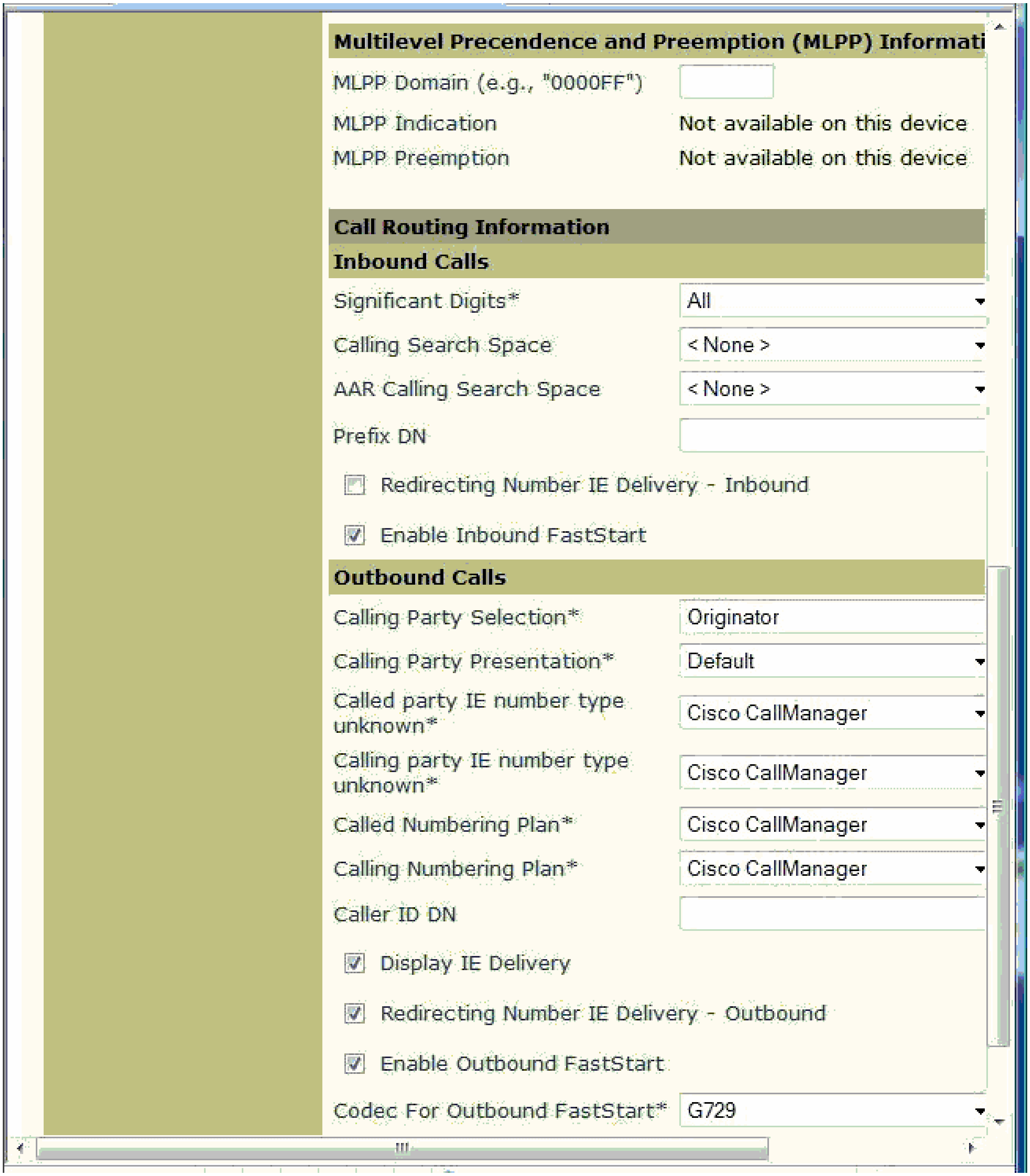


Figure 6. Media Termination Point list

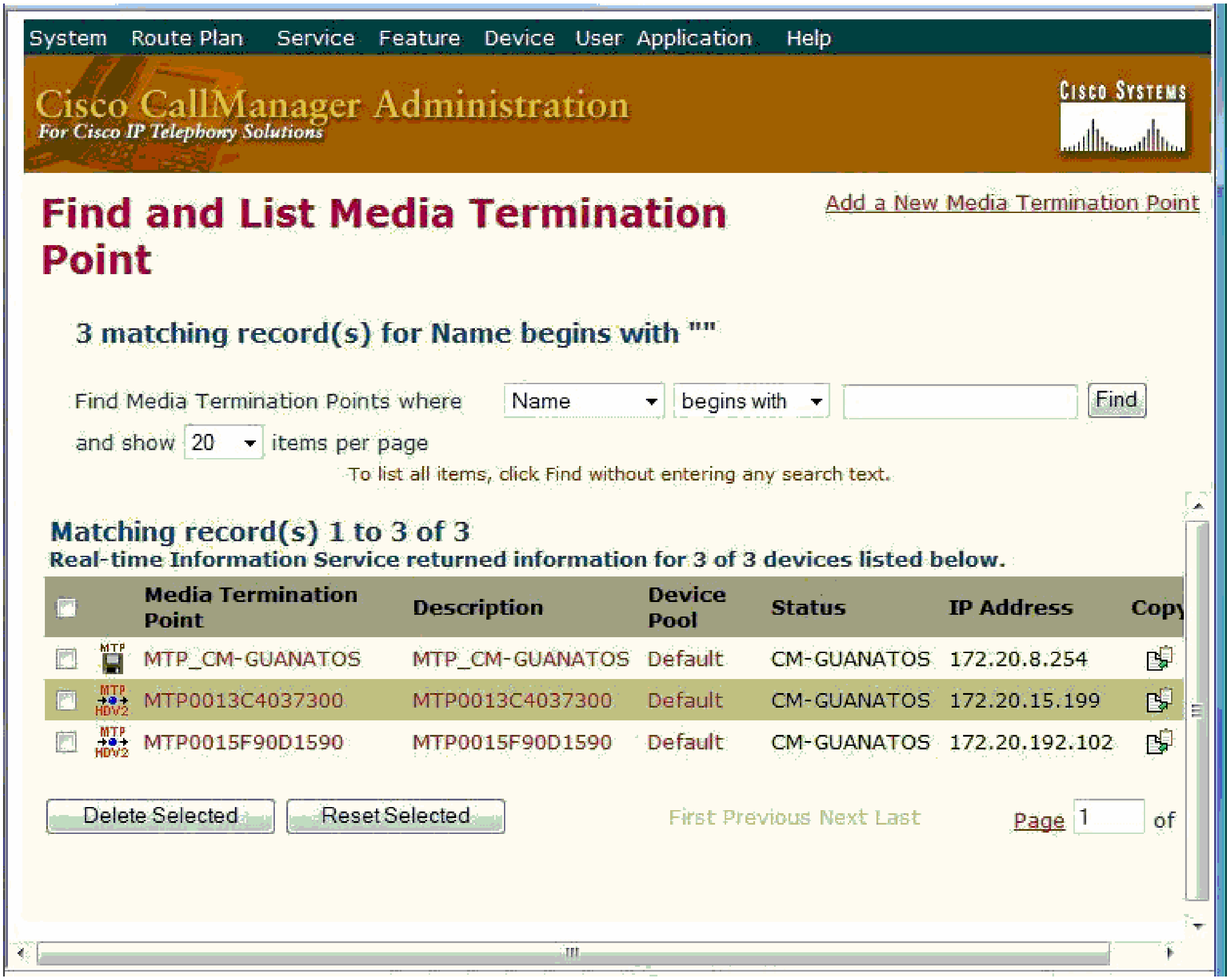


Figure 7. Local Media Termination Point (CCM MTP)

Notes: For codec G.711, Cisco Unified Communications Manager can use its own software MTP

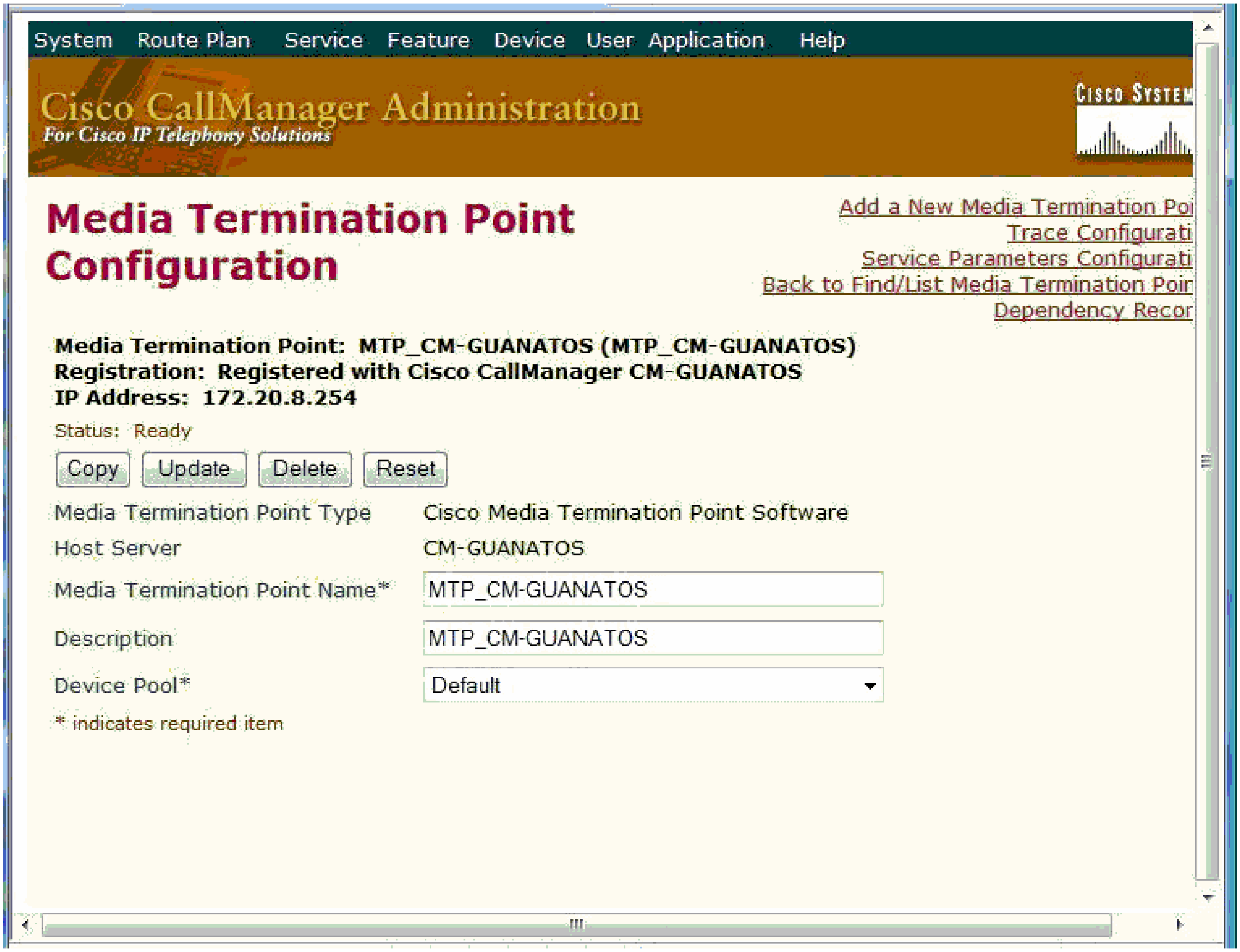


Figure 8. Hardware Media termination Point configuration. (IPIP GW MTP) for G.729 codec.

Notes: For basic calls using G.729, it requires an IOS hardware MTP which is co-resident on the Multi-service IP-to-IP gateway.

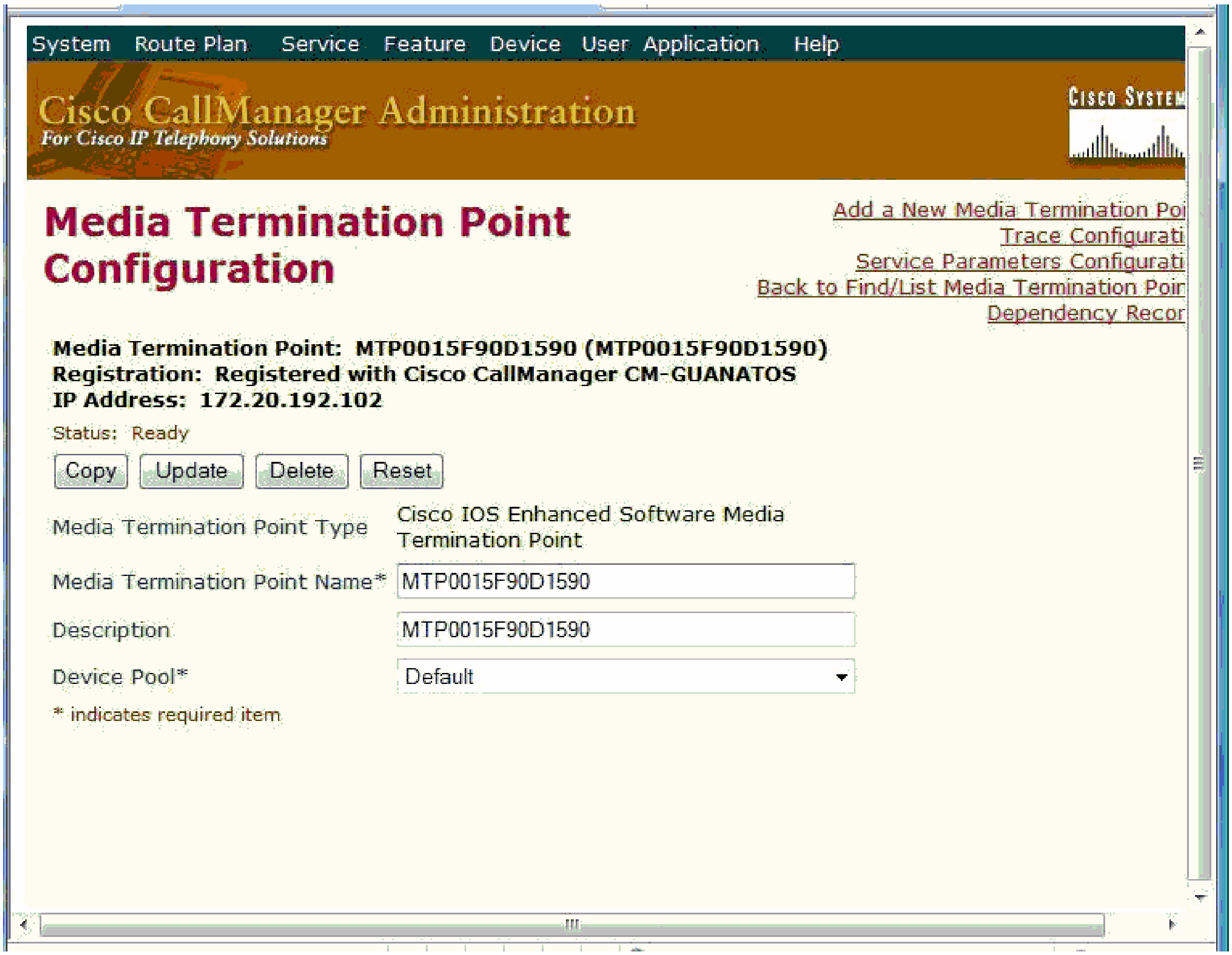


Figure 9. External hardware Media Termination Point Configuration. for T.38 FAX protocol and G.729 Codec.

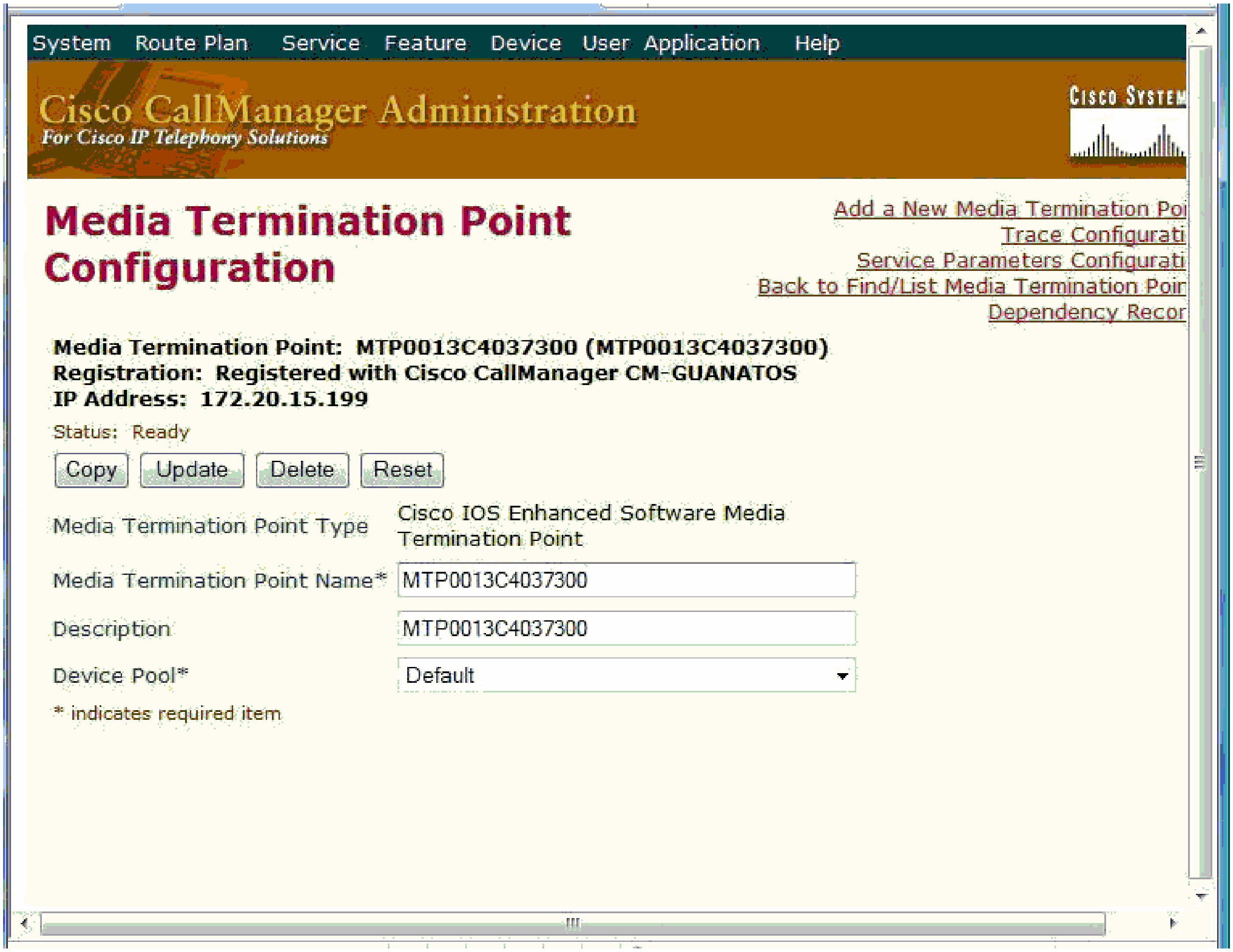


Figure 10. Media Resource Group Configuration.

Notes

1- Media Resource MTP0015F90d1590 (IPIP gateway hardware MTP) was used for G.729 Codec.

2- Media Resource MTP0015F90d159 and MTP0011936851409(xcode) was used for conference call using G.729 codec.

3- Local Media Resource MTP_CM-GUANATOS (MTP) for G.711 Codec.

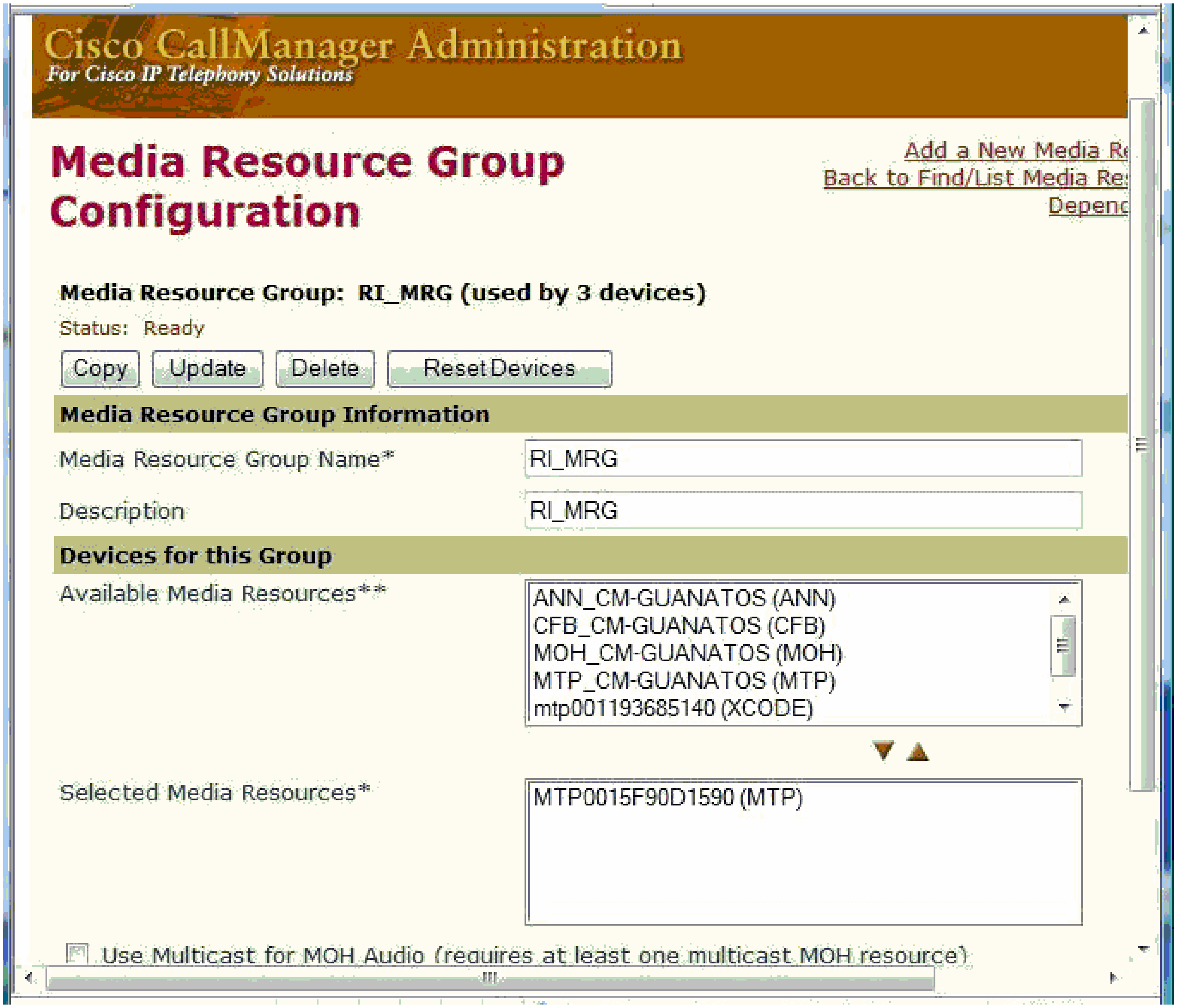
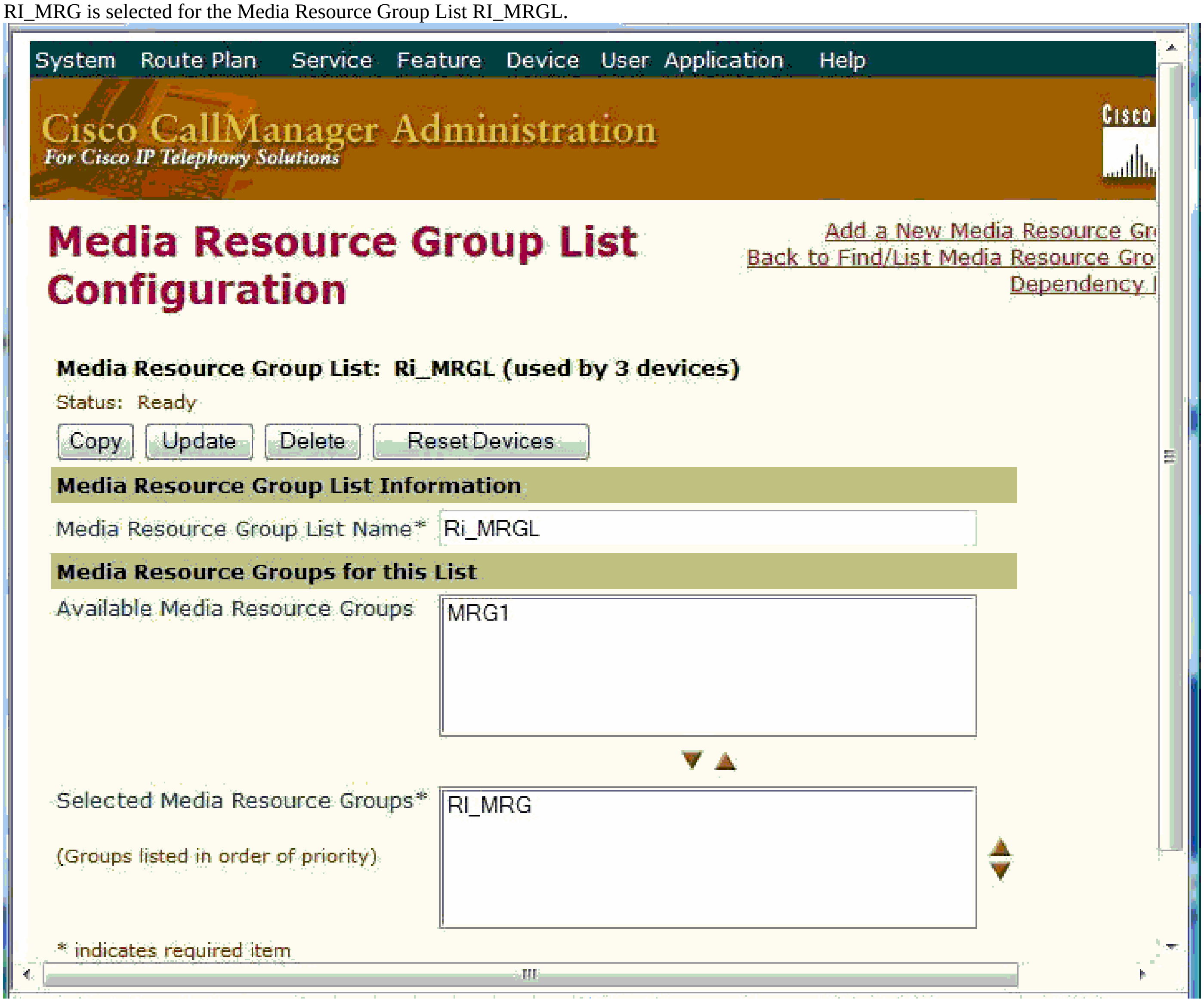


Figure 11. Media Resource Group Configuration. Selected Local MTP for G.711 Codec.

Figure 12. Media Resource Group List conf.

Notes: Select Media Resource Group for the Media Resource Group list for the specific Codec and protocol. The Media Resource Group RI_MRG is selected for the Media Resource Group List RI_MRGL.



Router#sh ver
Cisco IOS Software, 3800 Software (C3825-ADVENTERPRISEK9_IVS-M), Version 12.4(11
)T1, RELEASE SOFTWARE (fc5)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2007 by Cisco Systems, Inc.
Compiled Thu 25-Jan-07 17:16 by prod_rel_team

ROM: System Bootstrap, Version 12.3(11r)T2, RELEASE SOFTWARE (fc1)

Router uptime is 1 week, 2 days, 2 hours, 29 minutes
System returned to ROM by power-on
System image file is "flash:c3825-adventerprisek9_ivs-mz.124-11.T1.bin"

This product contains cryptographic features and is subject to United
States and local country laws governing import, export, transfer and
use. Delivery of Cisco cryptographic products does not imply
third-party authority to import, export, distribute or use encryption.
Importers, exporters, distributors and users are responsible for
compliance with U.S. and local country laws. By using this product you
agree to comply with applicable laws and regulations. If you are unable
to comply with U.S. and local laws, return this product immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at:
http://www.cisco.com/wwl/export/crypto/tool/stqrg.html

If you require further assistance please contact us by sending email to
export@cisco.com.

Cisco 3825 (revision 1.0) with 223232K/38912K bytes of memory.
Processor board ID FTX0946A1BV
2 Gigabit Ethernet interfaces
1 Channelized T1/PRI port
1 Virtual Private Network (VPN) Module
2 Voice FXS interfaces
DRAM configuration is 64 bits wide with parity enabled.
479K bytes of NVRAM.
62720K bytes of ATA System CompactFlash (Read/Write)

Configuration register is 0x2102

Router#sh run
Building configuration...

Current configuration : 2566 bytes
!
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Router
!
boot-start-marker
boot system flash:c3825-adventerprisek9_ivs-mz.124-11.T1.bin

```
boot-end-marker
!
logging buffered 10000000
no logging console
enable password cisco
!
no aaa new-model
ip cef
!
!
!
multilink bundle-name authenticated
!
voice-card 0
 dspfarm
 dsp services dspfarm
!
!
!
!
voice service voip
 allow-connections h323 to h323
 allow-connections h323 to sip
 allow-connections sip to h323
 allow-connections sip to sip
 h323
!
!
!
voice class codec 1 [1]
 codec preference 1 g729r8   ==→Notes: This is to set to G.729 or G.723 to test voice quality and/or initiate T.38
 codec preference 2 g711ulaw
!
!
!
interface GigabitEthernet0/0
 ip address 172.20.192.102 255.255.255.0
 duplex auto
 speed auto
 media-type rj45
 no keepalive
!
interface GigabitEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
 media-type rj45
 no keepalive
!
ip default-gateway 172.20.192.1
ip route 0.0.0.0 0.0.0.0 172.20.192.1
!
!
ip http server
no ip http secure-server
!
```

[1] This section was added, and voice codec hard-coding statements were removed from the dial peers, to check codec negotiation.

```
!
control-plane
!
!
voice-port 0/2/0
 station-id name Test Analog
 station-id number 7055
!
voice-port 0/2/1
 station-id name RI-NGUYEN
 station-id number 7044
!
!
!
sccp local GigabitEthernet0/0 [2]
sccp ccm 172.20.8.254 identifier 1 version 4.1[3]
sccp
!
sccp ccm group 1
 associate ccm 1 priority 1
 associate profile 1 register MTP0015f90d1590[4]
!
dspfarm profile 1 mtp
 codec g729r8
 codec pass-through
 maximum sessions software 10
 associate application SCCP
!
!
dial-peer voice 7000 voip  ==➔Notes: This is the H.323 signaling dial-peer.
 description dial peer digital toward CCM4.2
 destination-pattern 700[2-8]
 voice-class codec 1[5]
 session target ipv4:172.20.8.254
 session transport tcp[6]
 dtmf-relay h245-alphanumeric[7]
 fax-relay ecm disable[8]
 no vad[9]
!
dial-peer voice 7200 pots
 destination-pattern 7055
 port 0/2/0
 forward-digits 0
!
dial-peer voice 22 voip  =➔Notes: This is the SIP signaling dial-peer
 description dial peer toward Nortel CS1000
 max-conn 2[10]
 destination-pattern 2...
```

---

[2] This section was added for G.729B. Cisco Unify Communications Manager 4.2 does not support non G.711 MTP. Hardware MTP is required for Nortel to do early offer. MTP was placed on IP-to-IP gateway.
[3] Although 4.2 is not an available selection, the Cisco Unified Communications Manager version is critical for setting up the MTP on the IP-to-IP gateway.
[4] This must match the MTP name in Cisco Unified Communications Manager 4.2, and is in the format “MTP xxxx”, where “xxxx” is the MAC address of the Ethernet port of the device where the hardware MTP is located (the IP-to-IP gateway Gigabit Ethernet 0/0 in this case).
[5] Inserted for voice codec negotiation test. Codecs were specified in dial peers for other tests.
[6] Changed this line to “session transport udp” to test UDP session transport.
[7] Remove for G711 and G729 codec.
[8] This was removed at the time of G711 pass-through FAX testing.
[9] Removed when codec was set to G729B, as VAD is not optional with G729B
[10] This line item will only be added for CAC threshold test.

```
 voice-class codec 1
 session protocol sipv2
 session target ipv4:172.20.216.100
 session transport tcp
```

dtmf-relay h245-alphanumeric
Codec G711ulaw[11]

```
!
dial-peer voice 7044 pots
 destination-pattern 8000
 port 0/2/1
!
dial-peer voice 7009 voip
 description dial peer Analog FAX toward FaxGW
 destination-pattern 7009
 session target ipv4:172.20.15.199
!
dial-peer voice 7099 voip
 description dial peer Analog MGCP toward CCM4.2
 destination-pattern 7099
 voice-class codec 1
 session target ipv4:172.20.8.254
!
!
sip-ua
 no remote-party-id
 retry options 0
!
!
!
gatekeeper
 shutdown
!
!
line con 0
 exec-timeout 0 0
 password cisco
 login
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 exec-timeout 0 0
 password cisco
 login
line vty 5 10
 exec-timeout 0 0
 password cisco
 login
!
scheduler allocate 20000 1000
!
end
Router#
```

[11] Changed to "codec g729br8" when codec was set to G729B.
Removed this line for codec negotiation test. Use voice-class.

Cisco Second Gateway 3825 configuration for FAX using T.38. H.323 and G.729 transcoder to G.711 for conference calls

```
IPIPgw-3825#sho ver
Cisco IOS Software, 3800 Software (C3825-IPVOICE_IVS-M), Version 12.4(11)T, REL)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2006 by Cisco Systems, Inc.
Compiled Sat 18-Nov-06 23:16 by prod_rel_team

ROM: System Bootstrap, Version 12.3(11r)T2, RELEASE SOFTWARE (fc1)

IPIPgw-3825 uptime is 6 days, 23 hours, 59 minutes
System returned to ROM by reload at 00:12:47 UTC Fri Apr 20 2007
System image file is "flash:c3825-ipvoice_ivs-mz.124-11.T.bin"

  Cisco 3825 (revision 1.0) with 226304K/35840K bytes of memory.
Processor board ID FTX0925A0ST
2 Gigabit Ethernet interfaces
31 Serial interfaces
1 Serial(sync/async) interface
2 Channelized E1/PRI ports
2 Voice FXS interfaces
DRAM configuration is 64 bits wide with parity enabled.
479K bytes of NVRAM.
125184K bytes of ATA System CompactFlash (Read/Write)

Configuration register is 0x2102
```

ipipgw_3825#sh run

```
Building configuration...

Current configuration : 2574 bytes
!
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname ipipgw_3825
!
boot-start-marker
boot system flash:c3825-ipvoice_ivs-mz.124-11.T.bin
boot-end-marker
!
logging buffered 10000000
no logging console
enable password cisco
!
no aaa new-model
no network-clock-participate slot 1
no network-clock-participate slot 2
voice-card 0
 dspfarm
 dsp services dspfarm
!
voice-card 1
 dspfarm
!
voice-card 2
```

```
 dspfarm
!
ip cef
!
!
multilink bundle-name authenticated
!
isdn switch-type primary-net5
!
!
voice service voip
 allow-connections h323 to h323
 h323
!
!
controller E1 1/0/0
 pri-group timeslots 1-31
!
controller E1 1/0/1
!
interface GigabitEthernet0/0
 description $ETH-LAN$$ETH-SW-LAUNCH$$INTF-INFO-GE 0/0$
 ip address 172.20.15.199 255.255.255.0
 duplex auto
 speed auto
 media-type rj45
 no keepalive
!
interface GigabitEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
 media-type rj45
 no keepalive
!
interface Serial0/0/0
 no ip address
 shutdown
 clock rate 2000000
!
interface Serial1/0/0:15
 no ip address
 encapsulation hdlc
 isdn switch-type primary-net5
 isdn protocol-emulate network
 isdn incoming-voice voice
 isdn send-alerting
 isdn outgoing display-ie
 no cdp enable
!
ip route 0.0.0.0 0.0.0.0 GigabitEthernet0/0
!
ip http server
!
control-plane
!
voice-port 0/2/0
 station-id name Cecily
```

```
 station-id number 4001
 caller-id enable
!
voice-port 0/2/1
 station-id name riri
 station-id number 7009
 caller-id enable
!
voice-port 1/0/0:15
!
!
!
sccp local GigabitEthernet0/0
sccp ccm 172.20.8.254 identifier 1 version 4.1
sccp
!
sccp ccm group 1
 associate ccm 1 priority 1
 associate profile 1 register mtp001193685140[12]
!
dspfarm profile 1 transcode
 codec g711ulaw
 codec g711alaw
 codec g729ar8
 codec g729abr8
 codec gsmfr
 codec g729r8
 maximum sessions 20
 associate application SCCP

!
dial-peer voice 10015 pots
 destination-pattern 74..
 direct-inward-dial
 port 1/0/0:15
 forward-digits all
!
dial-peer voice 10016 pots
 destination-pattern 8...
 direct-inward-dial
 port 1/0/0:15
 forward-digits all
!
dial-peer voice 7000 pots
 destination-pattern 7009
 port 0/2/1
!
dial-peer voice 2200 voip
 destination-pattern 22..
 session target ipv4:172.20.192.102[13]
 no fax-relay sg3-to-g3
!
dial-peer voice 4001 pots
 destination-pattern 4155554001
```

[12] For three way conference call using G.729 between call leg CUCM and Nortel PBX required a hardware MTP Transcoder for codec G.711 and G729 conversion.

[13] The Second gateway is pointed toward the IP-to- IP gateway bi-passing Cisco Unify Communication Manager to negotiate and support T.38 protocol using G729 codec.

```
 port 0/2/0
!
!
!
gatekeeper
 shutdown
!
!
line con 0
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 password cisco
 login
!
scheduler allocate 20000 1000
!
end
```

Signaling Server Setup via the Nortel Element Manager:

Figure 13. Configure the Zones

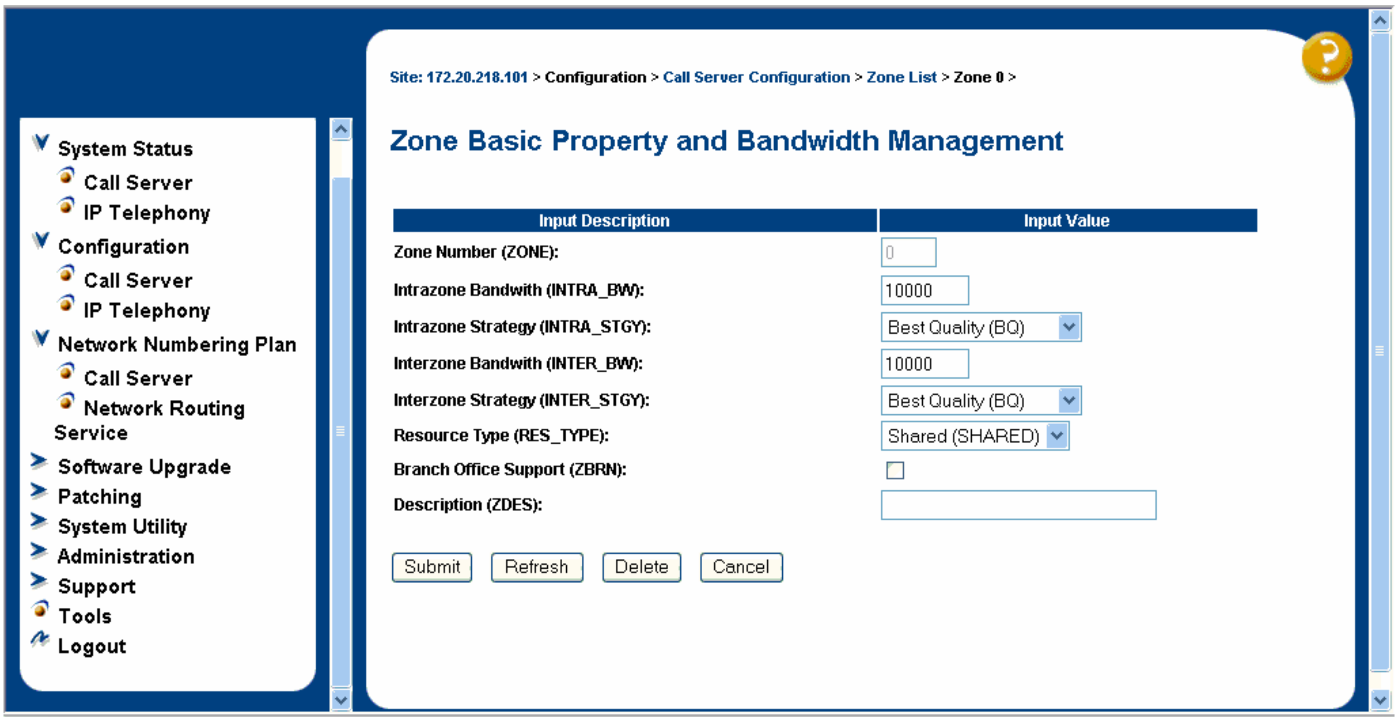


Figure 14. IP Telephony Configuration

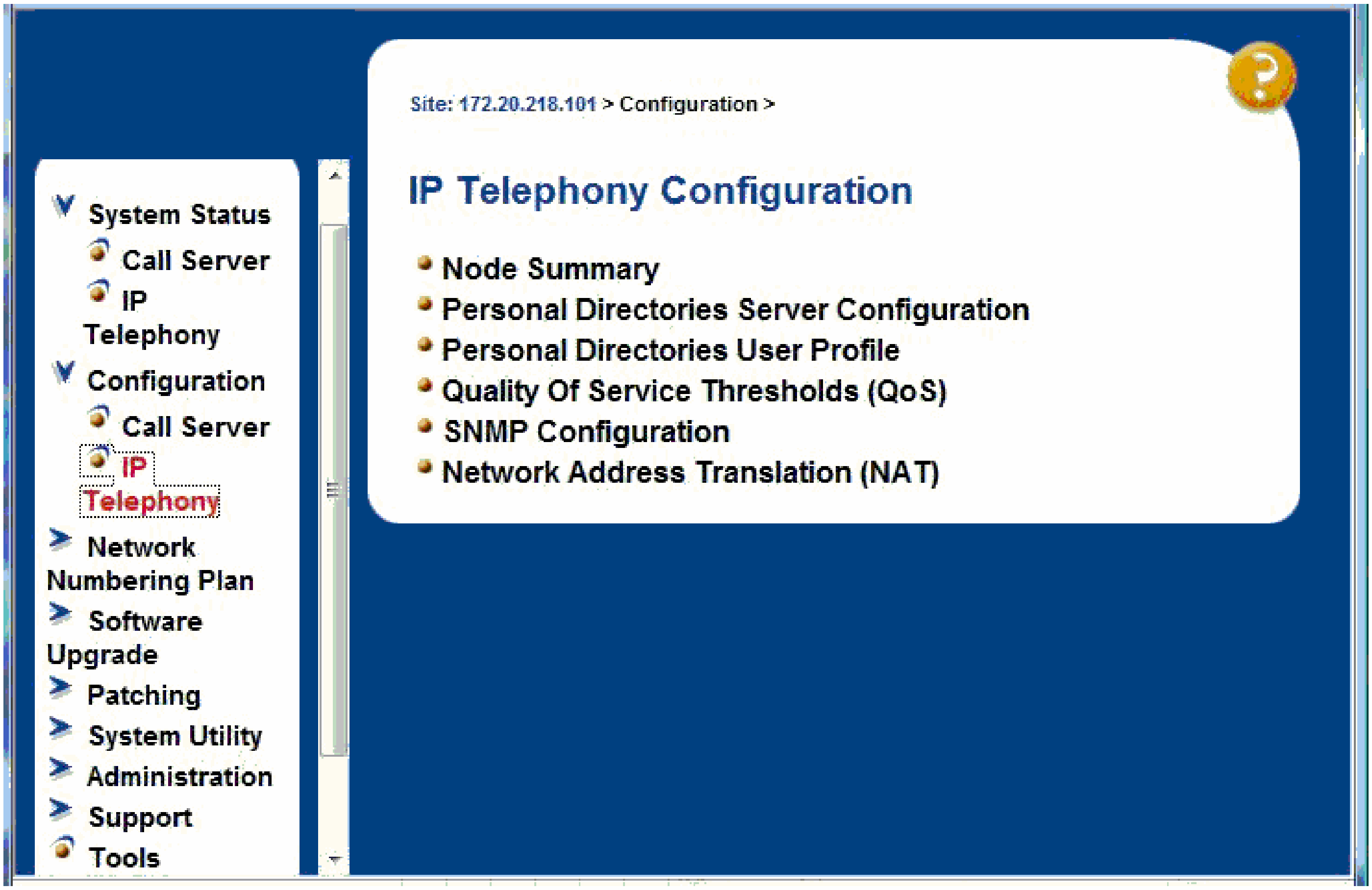


Figure 15. Configure a new IP Telephony Node summary

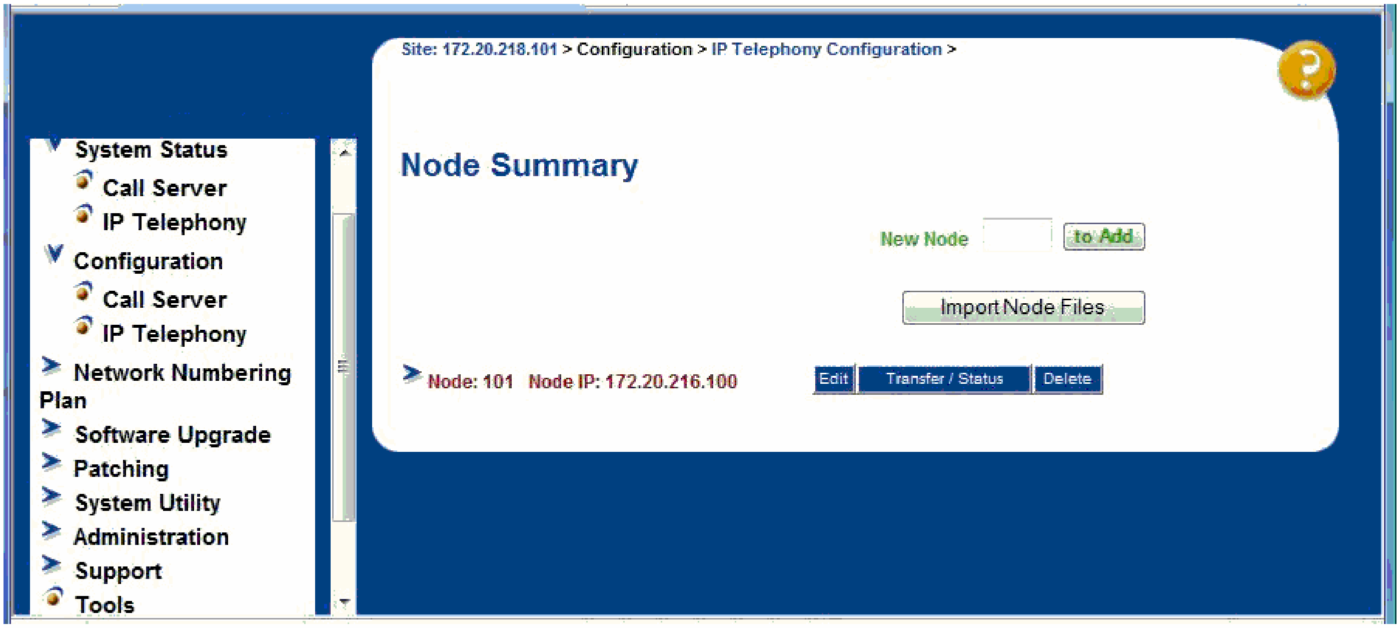


Figure 16. Configure the Node section

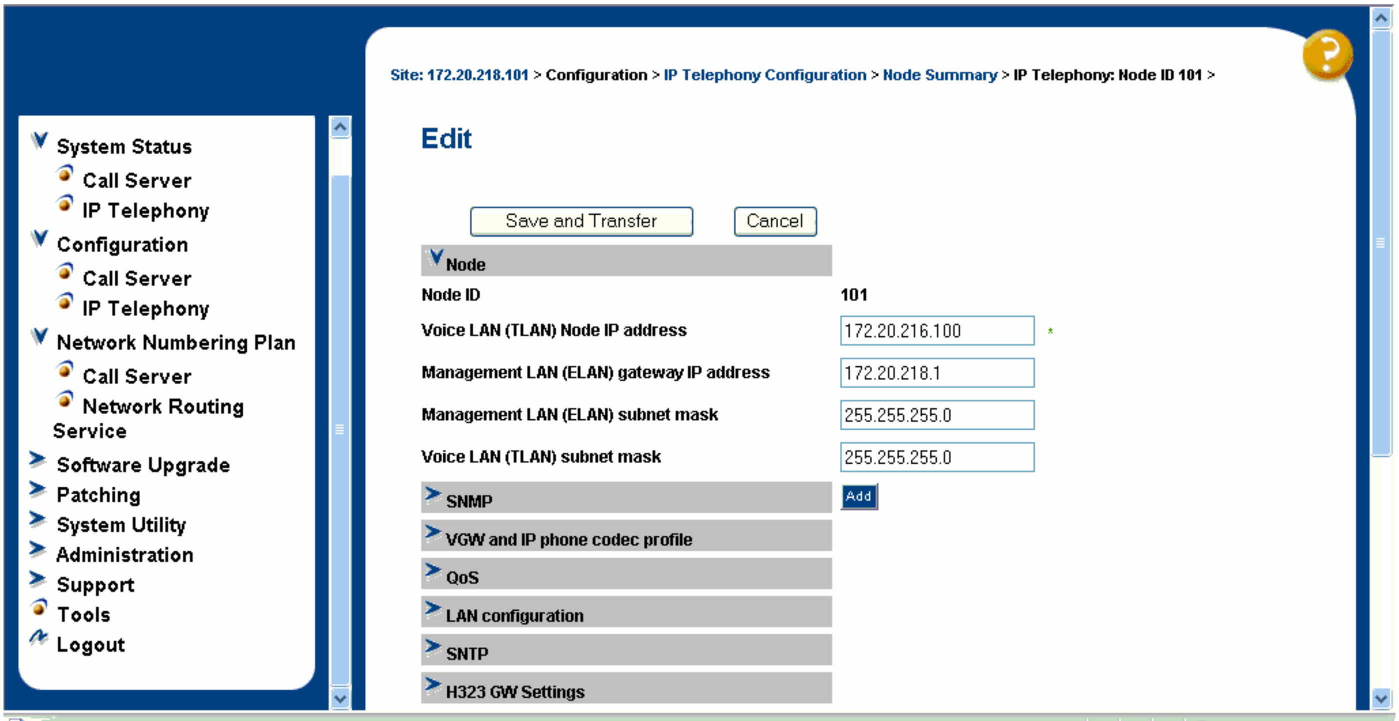


Figure 17. Configure the VGW and IP phone codec profile section

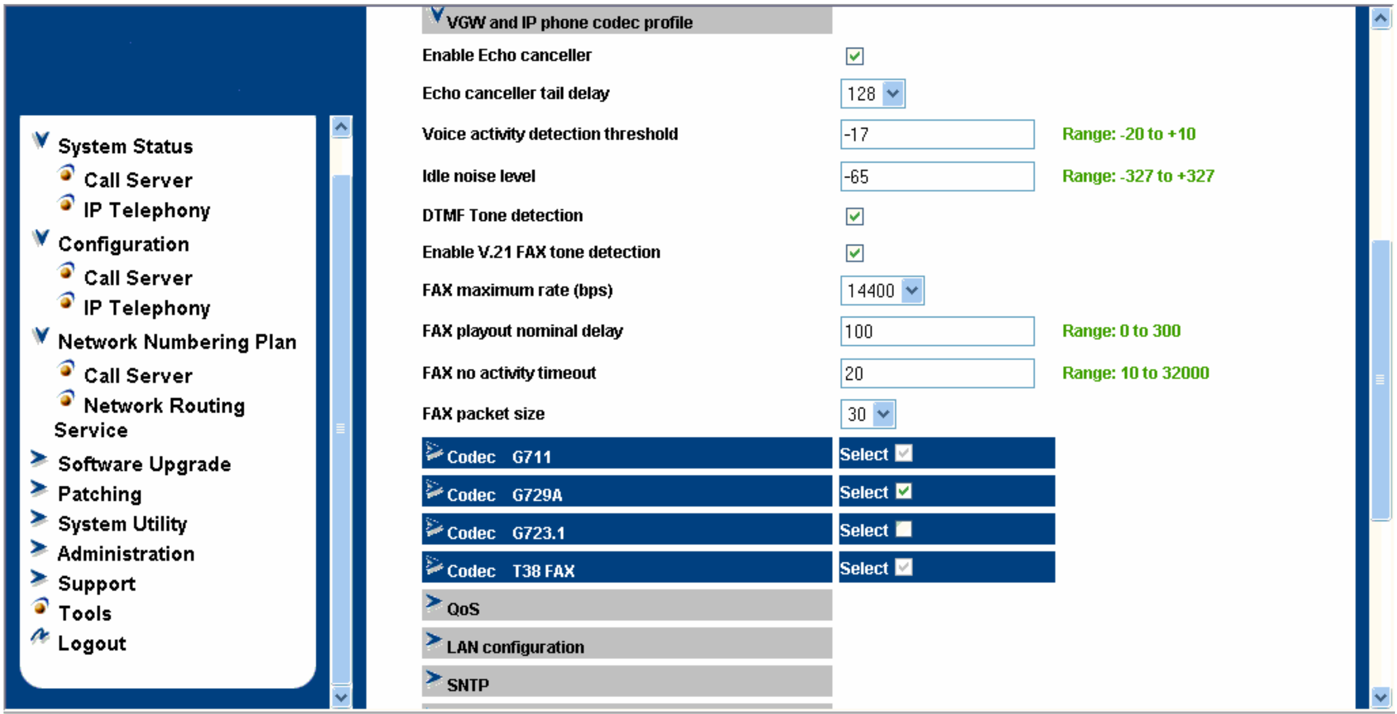


Figure 18. CODEC profile selection

Figure 19. Configure the QoS section

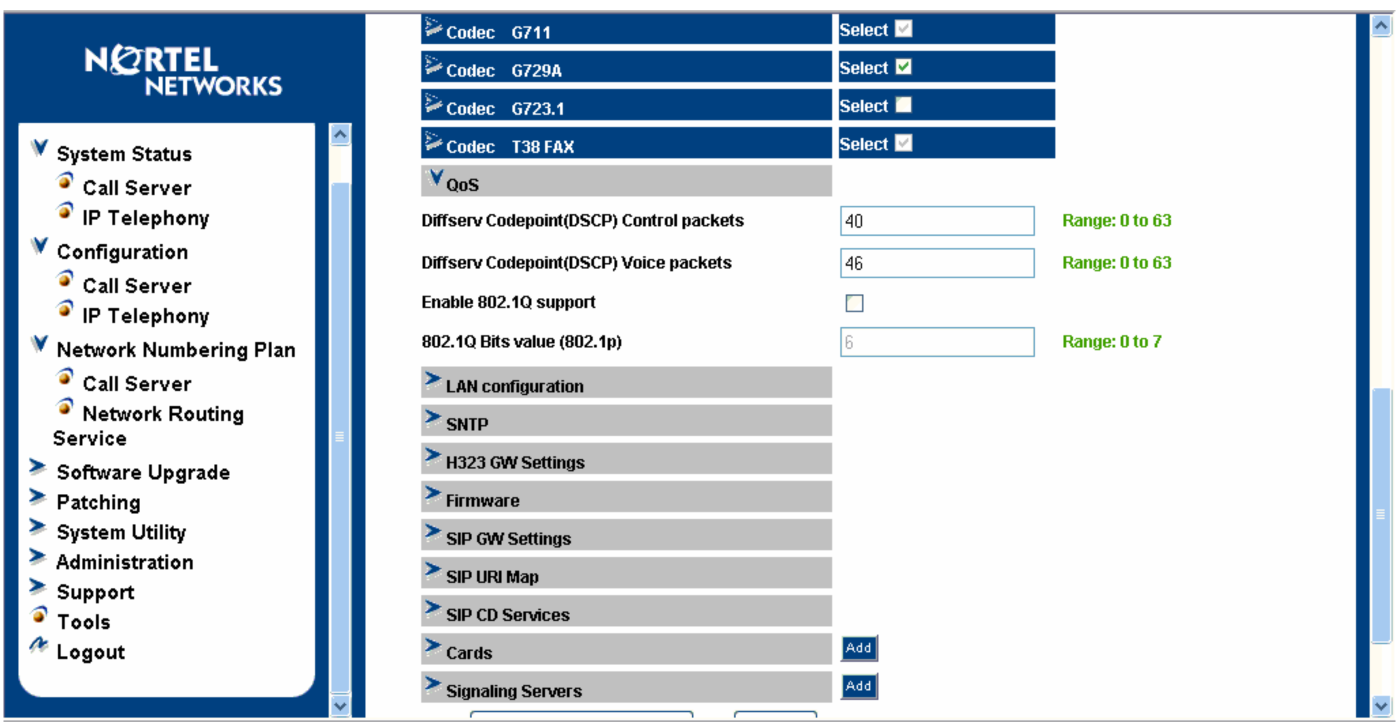


Figure 20. Configure LAN and SNTP Configuration section

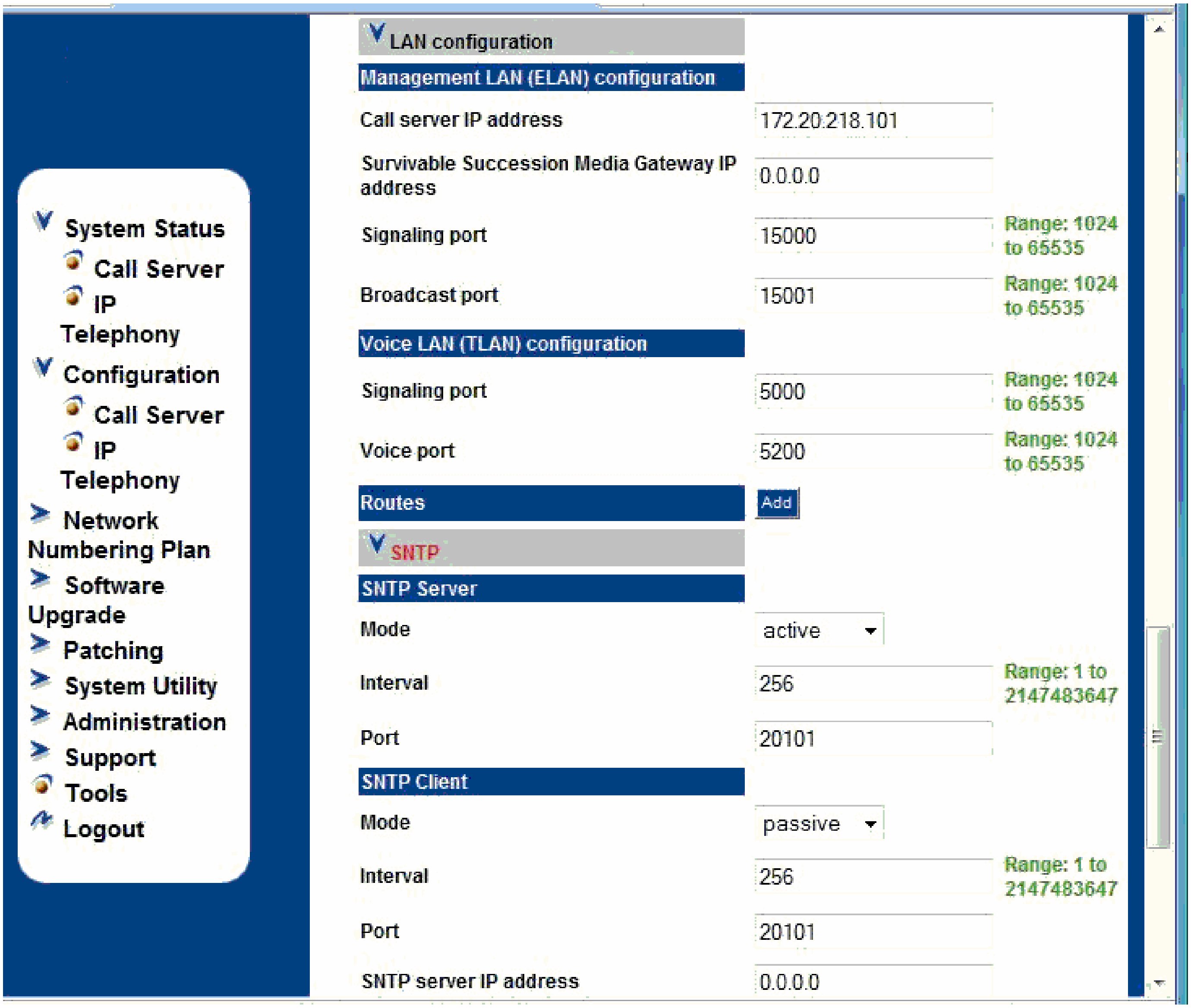


Figure 21. Configure the SIP GW Settings section

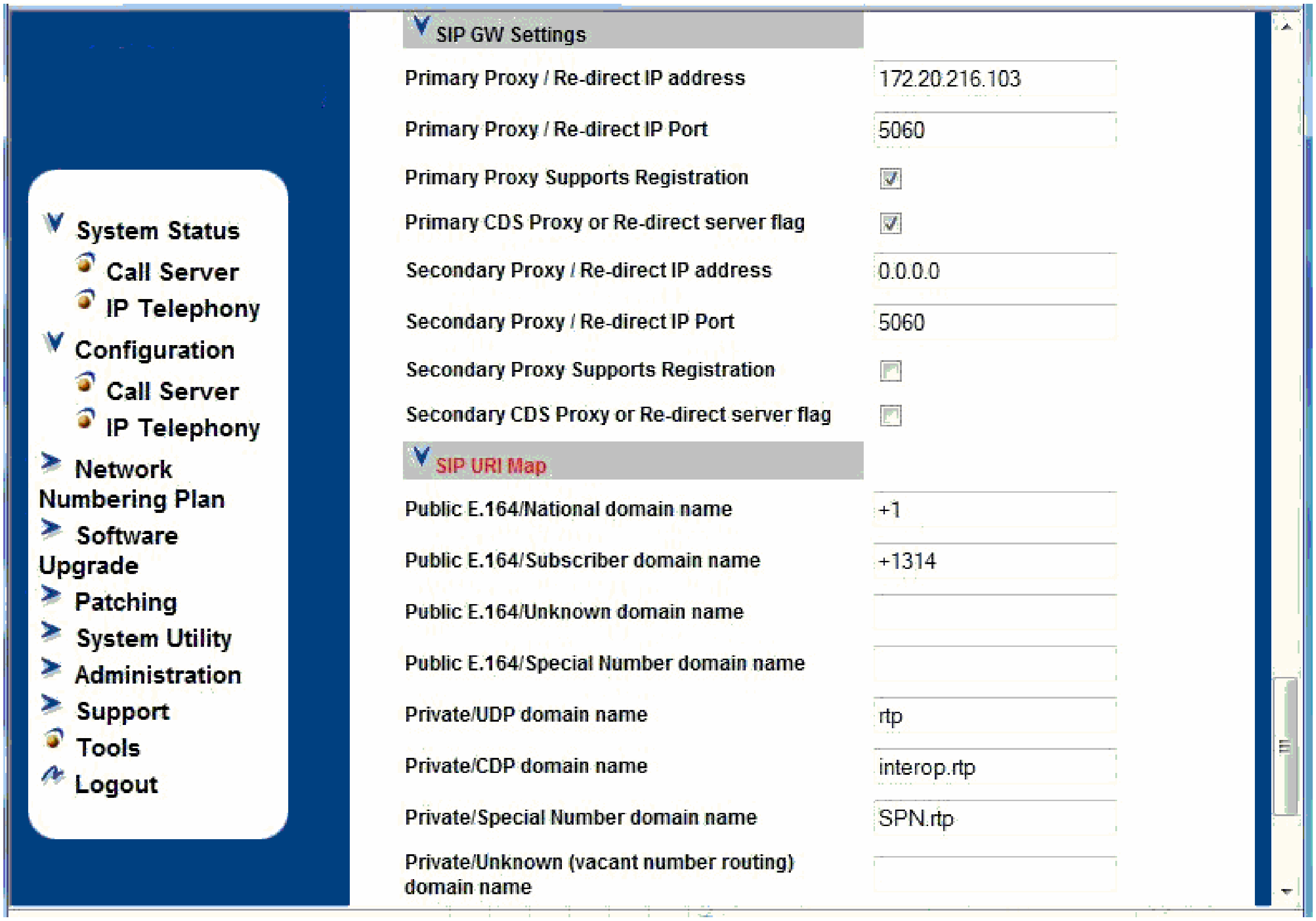


Figure 22. SIP CD Services

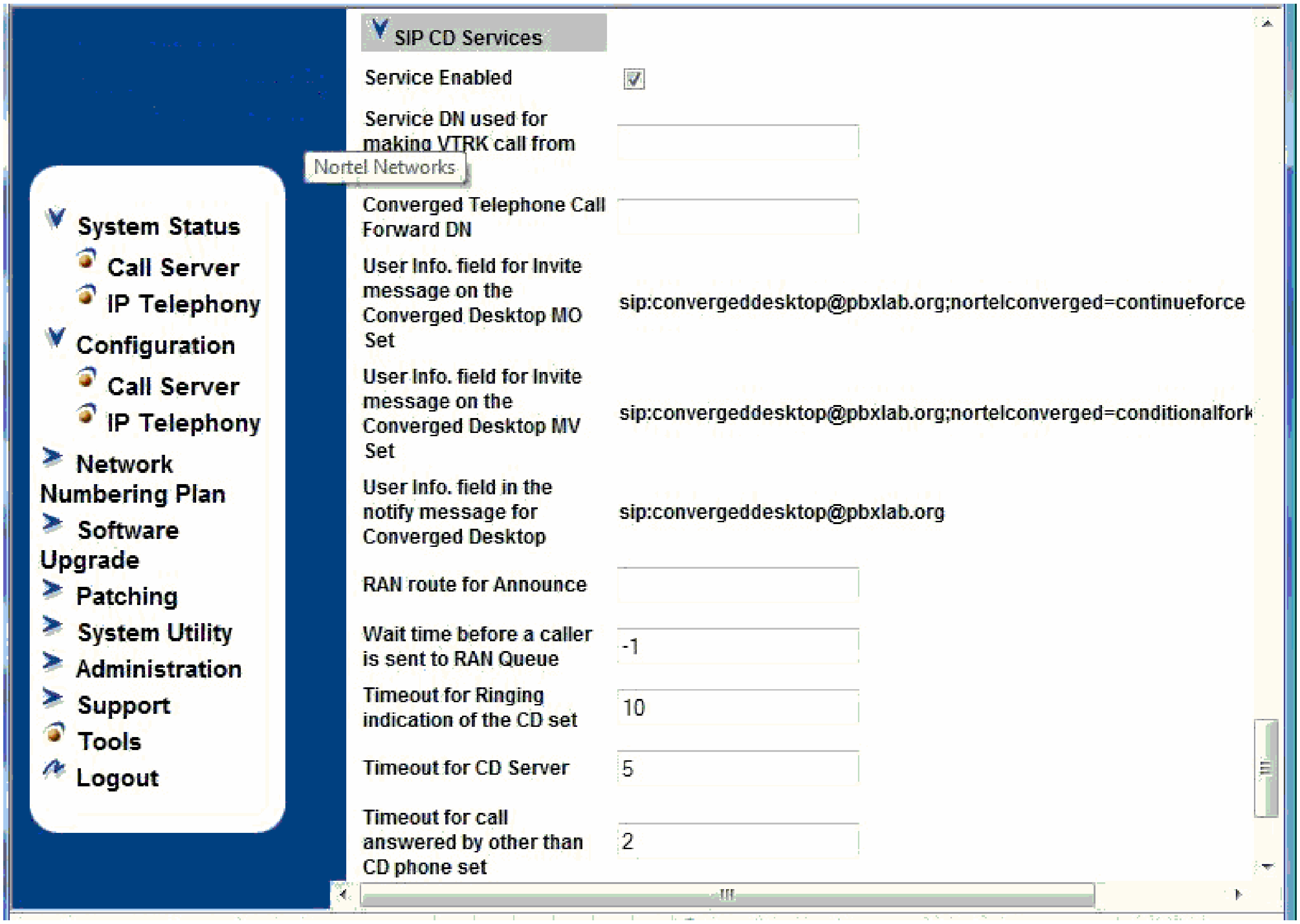


Figure 23. Configure the Card section for the MC-32 VGMC card section

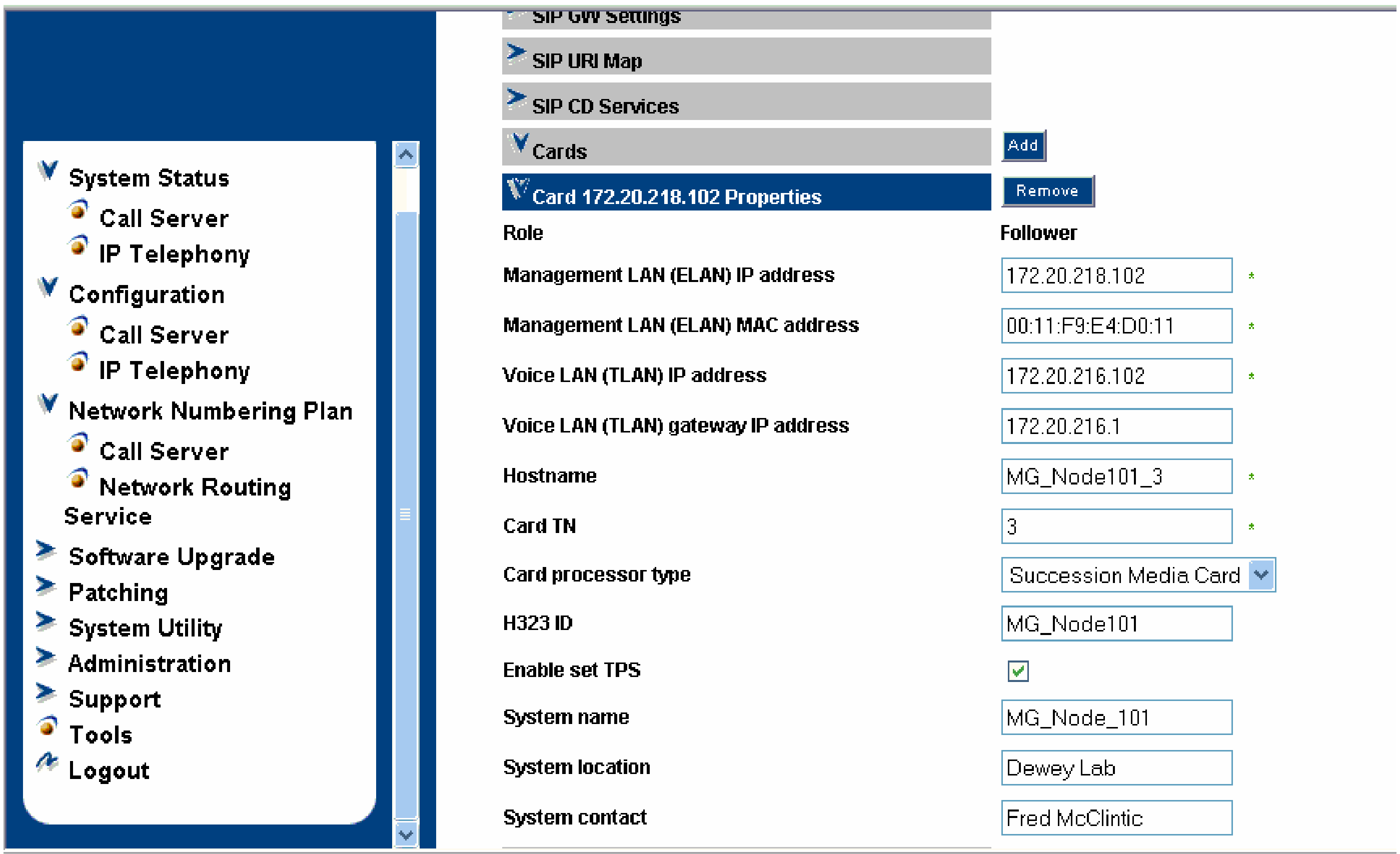
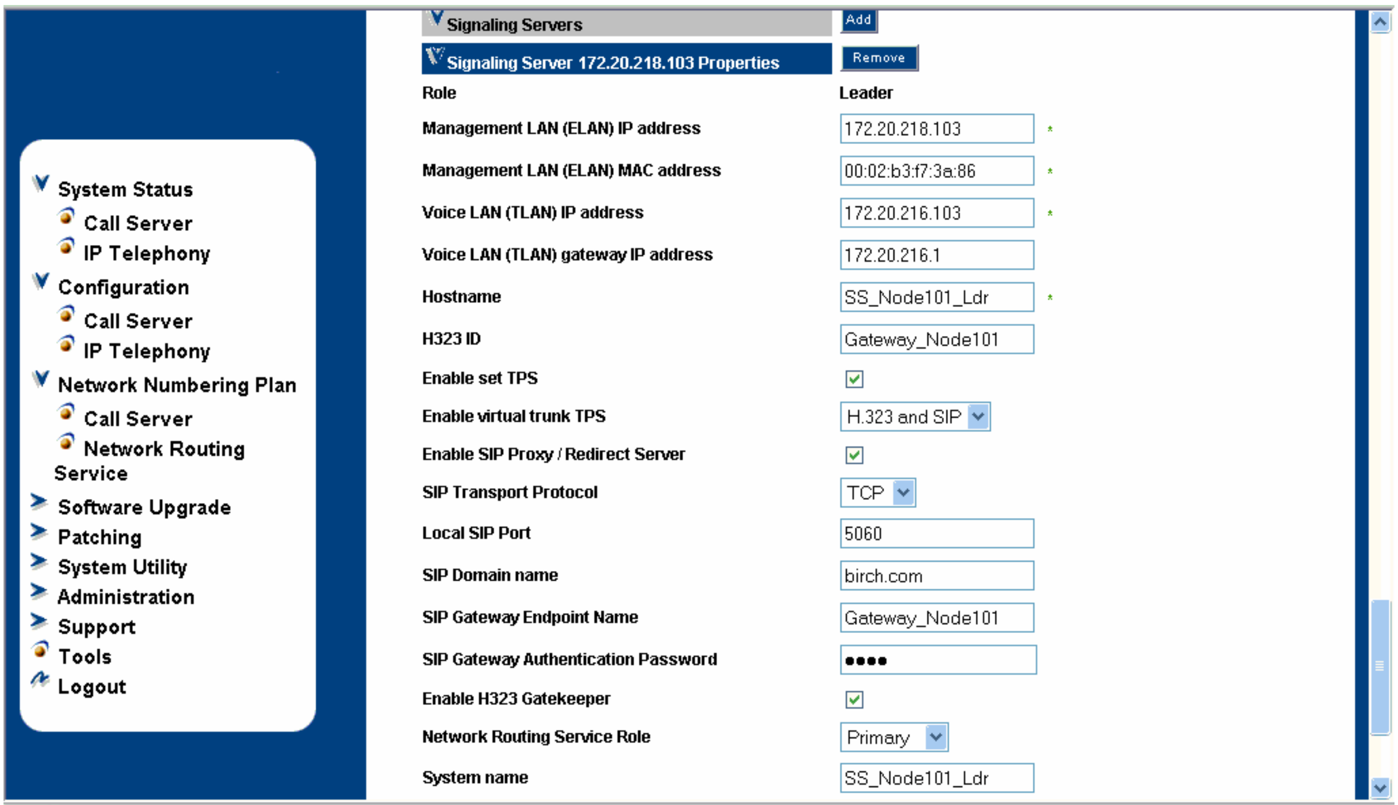


Figure 24. Configure the Signaling Server section



NRS (Network Routing Server):

Figure 25. NRS Overview

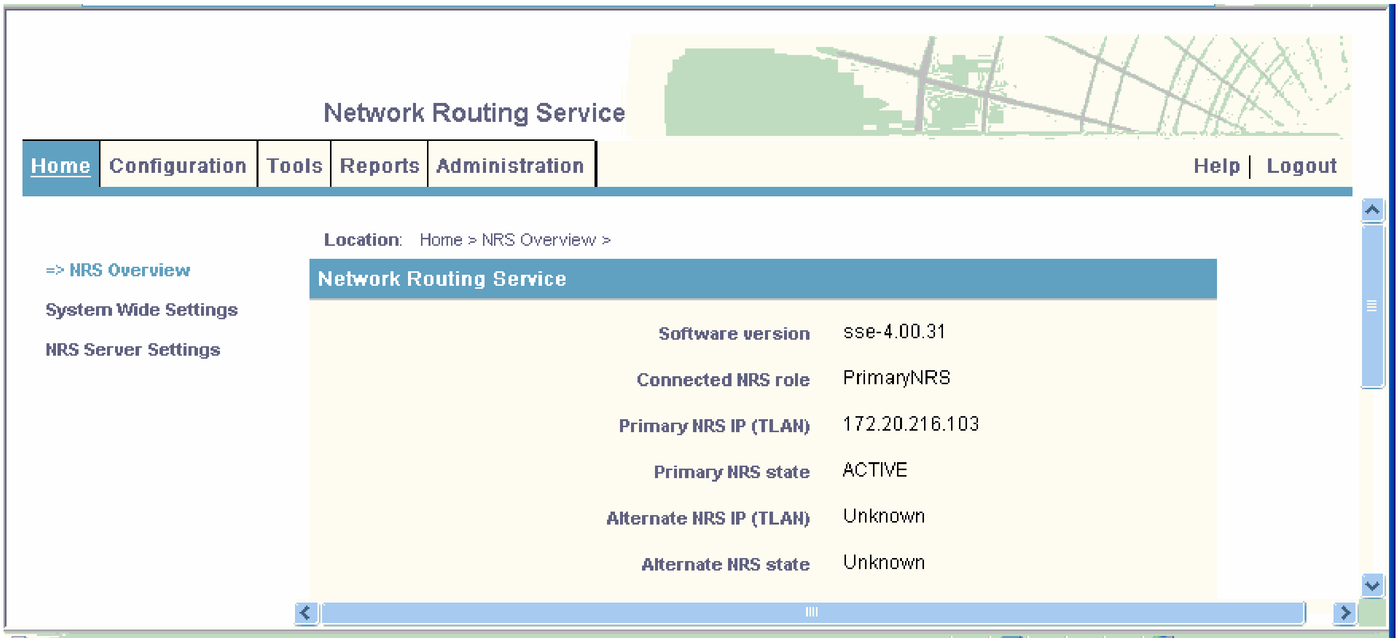


Figure 26. Configure the System Wide Settings

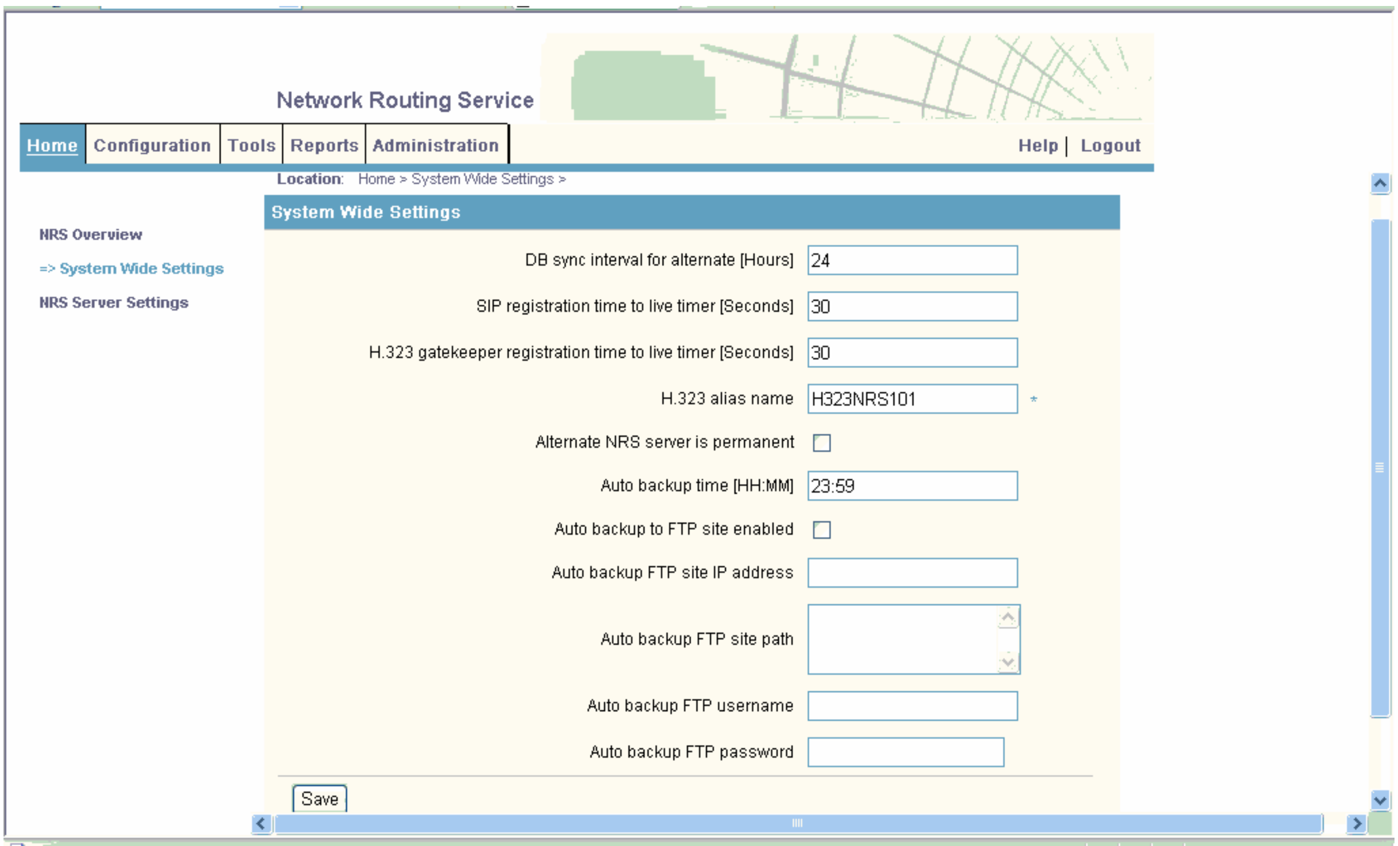


Figure 27. Configure the NRS Server Settings

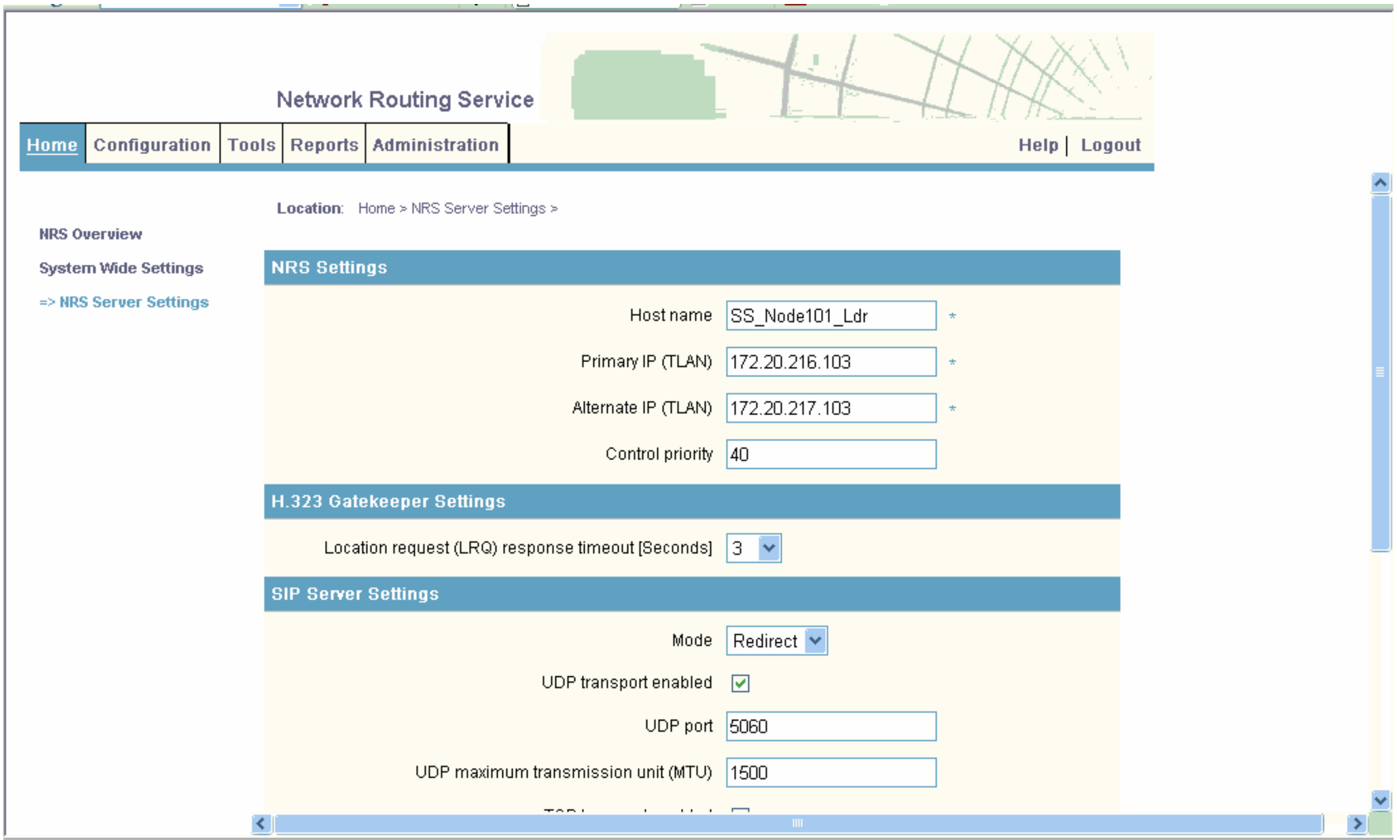


Figure 28. SIP Server Setting Cont'd

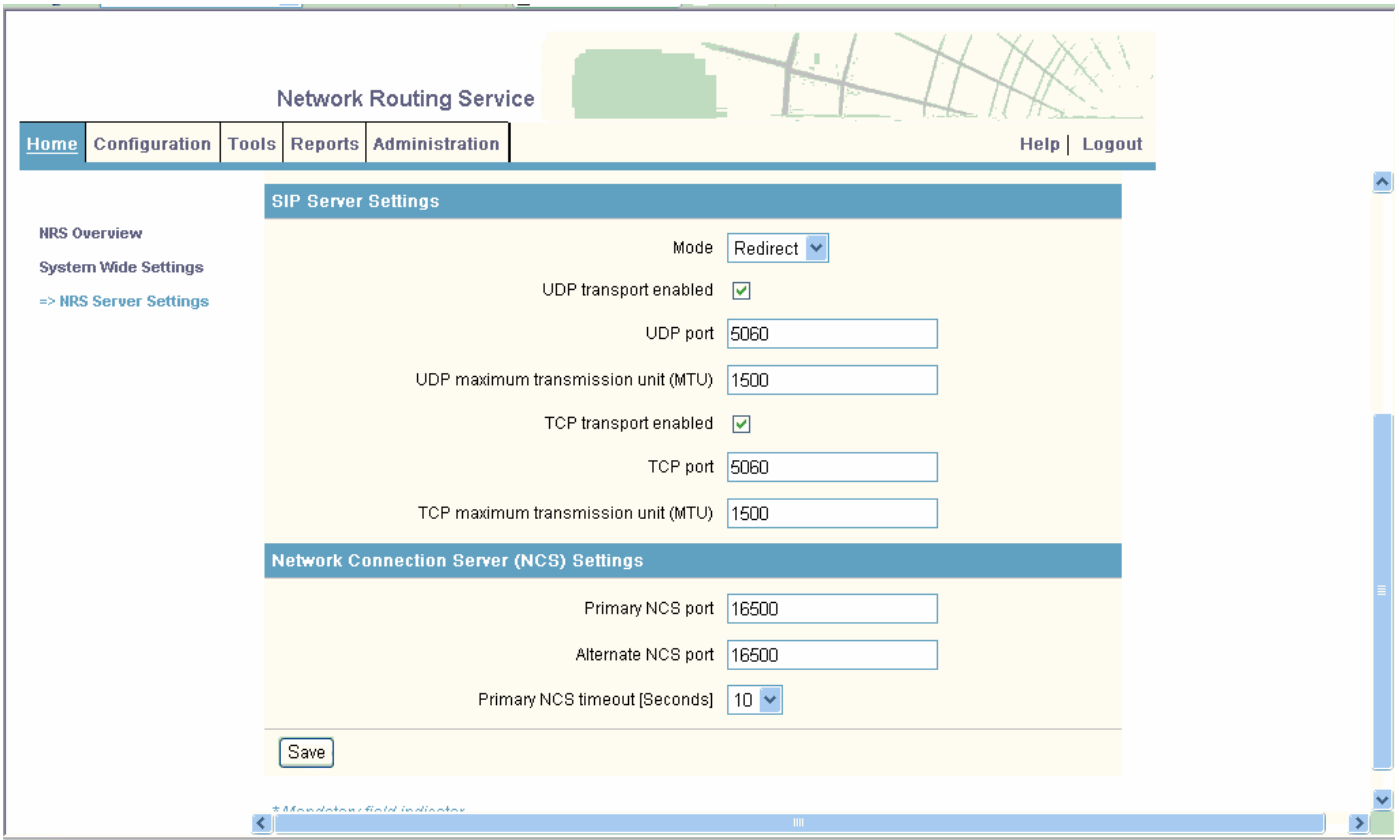


Figure 29. Configure a Service Domain

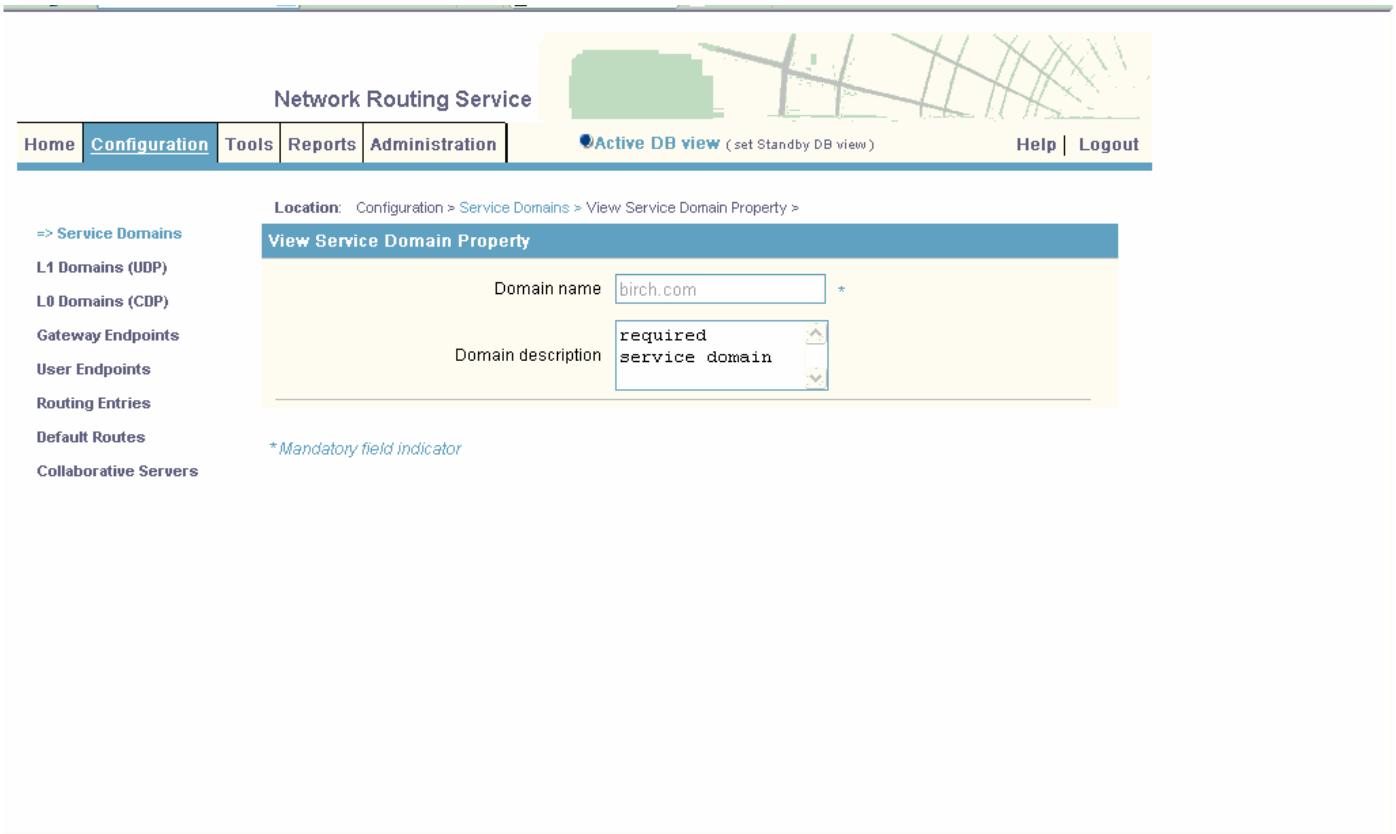


Figure 30. Configure a L1 Domain (UDP)

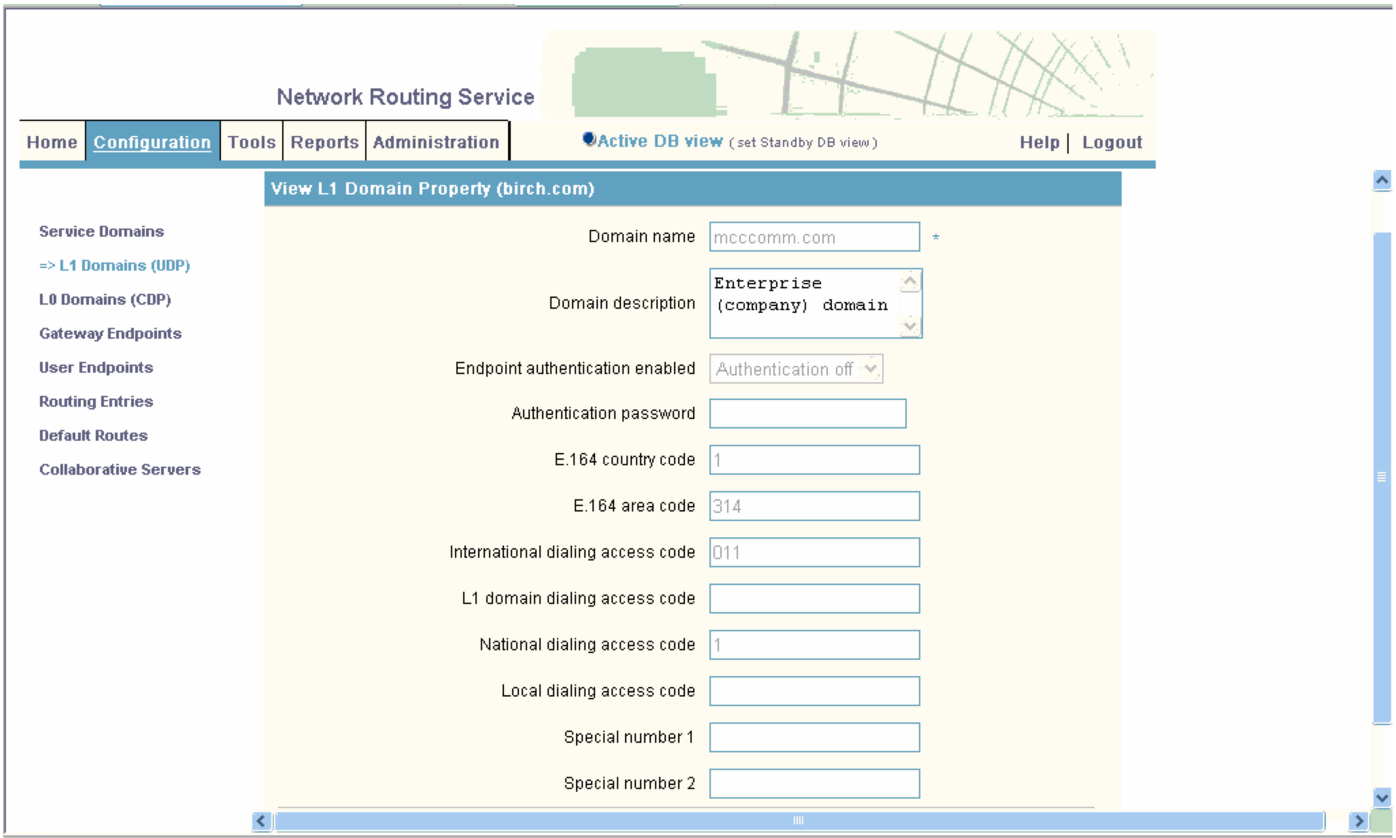


Figure 31. Configure a L0 Domain (CDP)

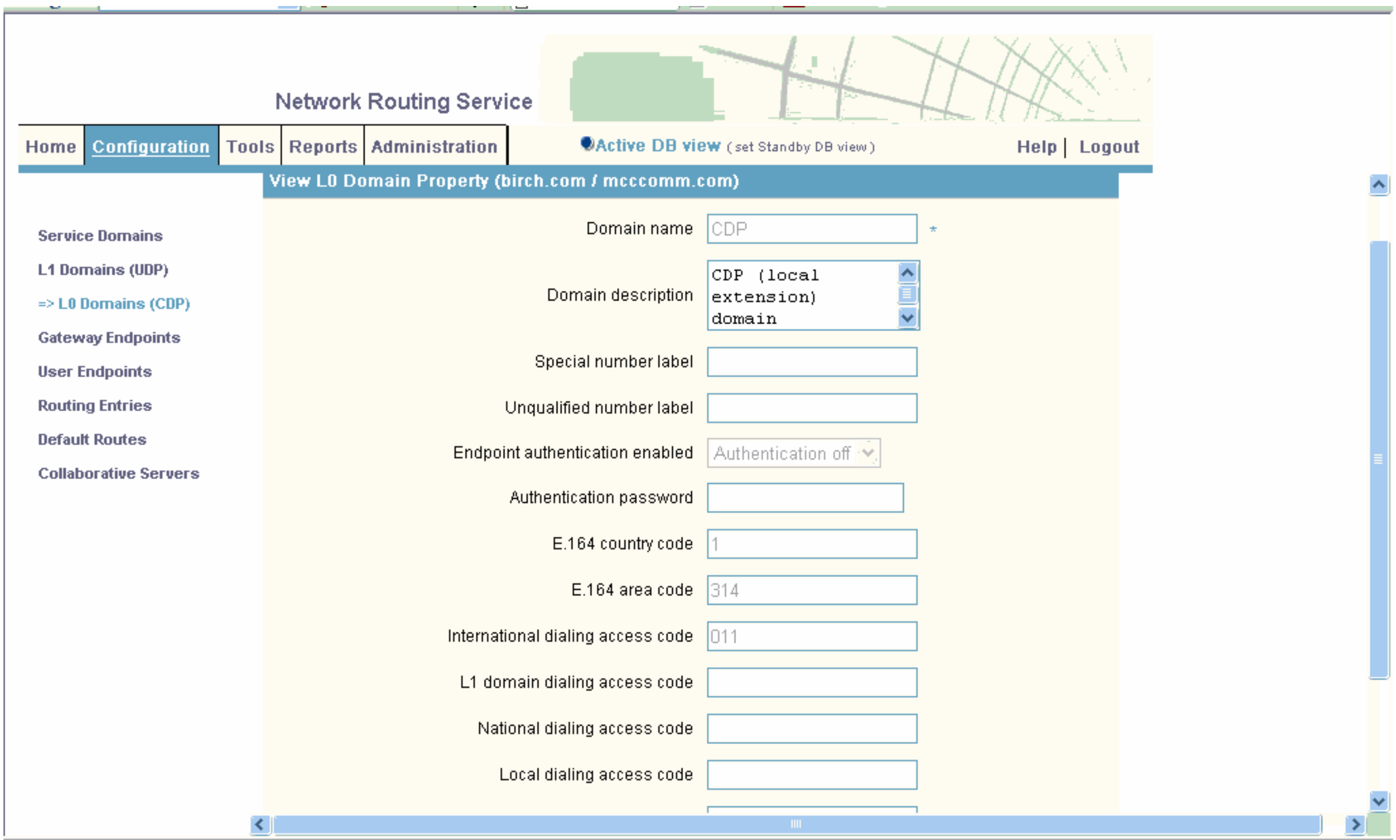


Figure 32. Configure a SIP Gateway

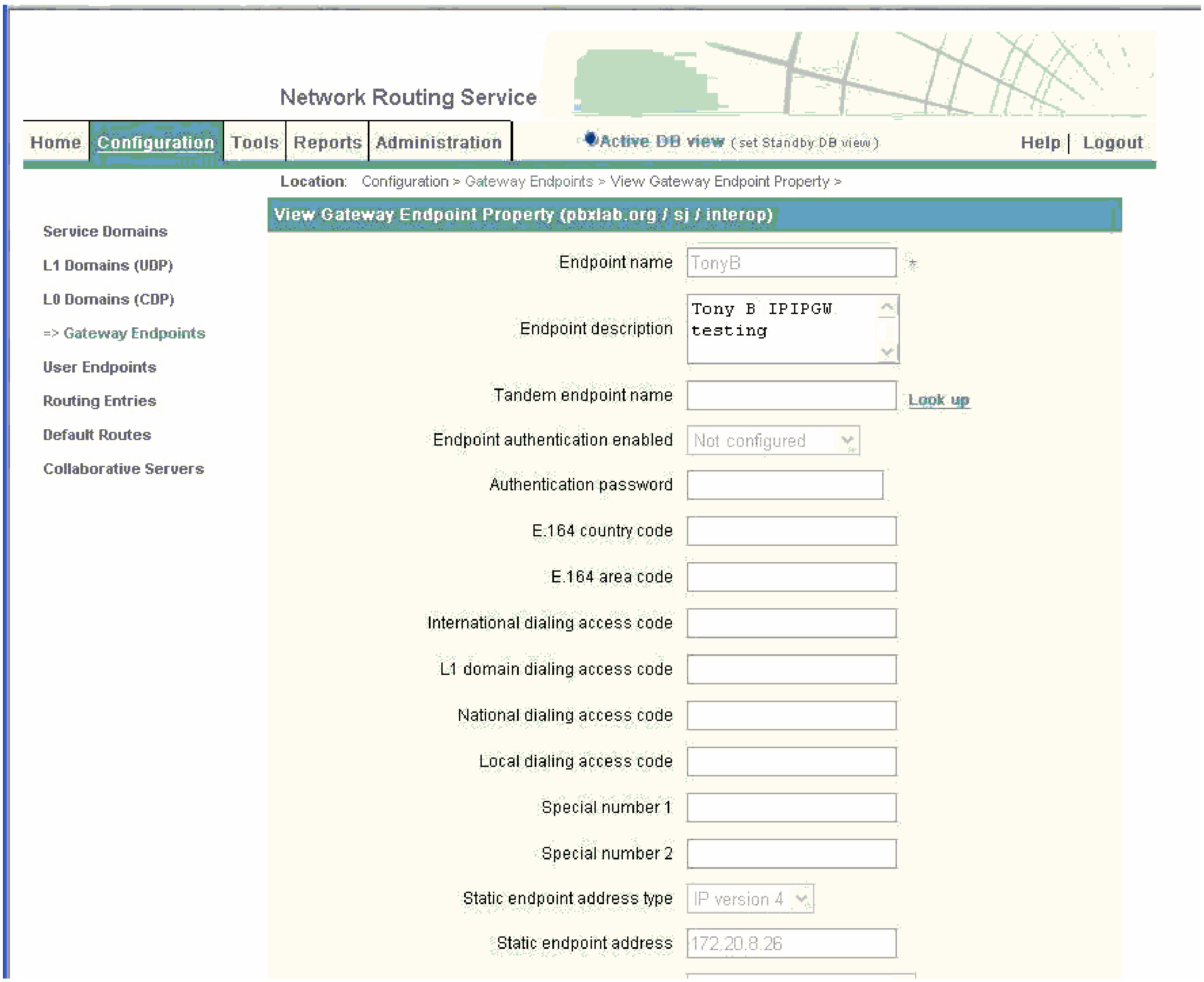
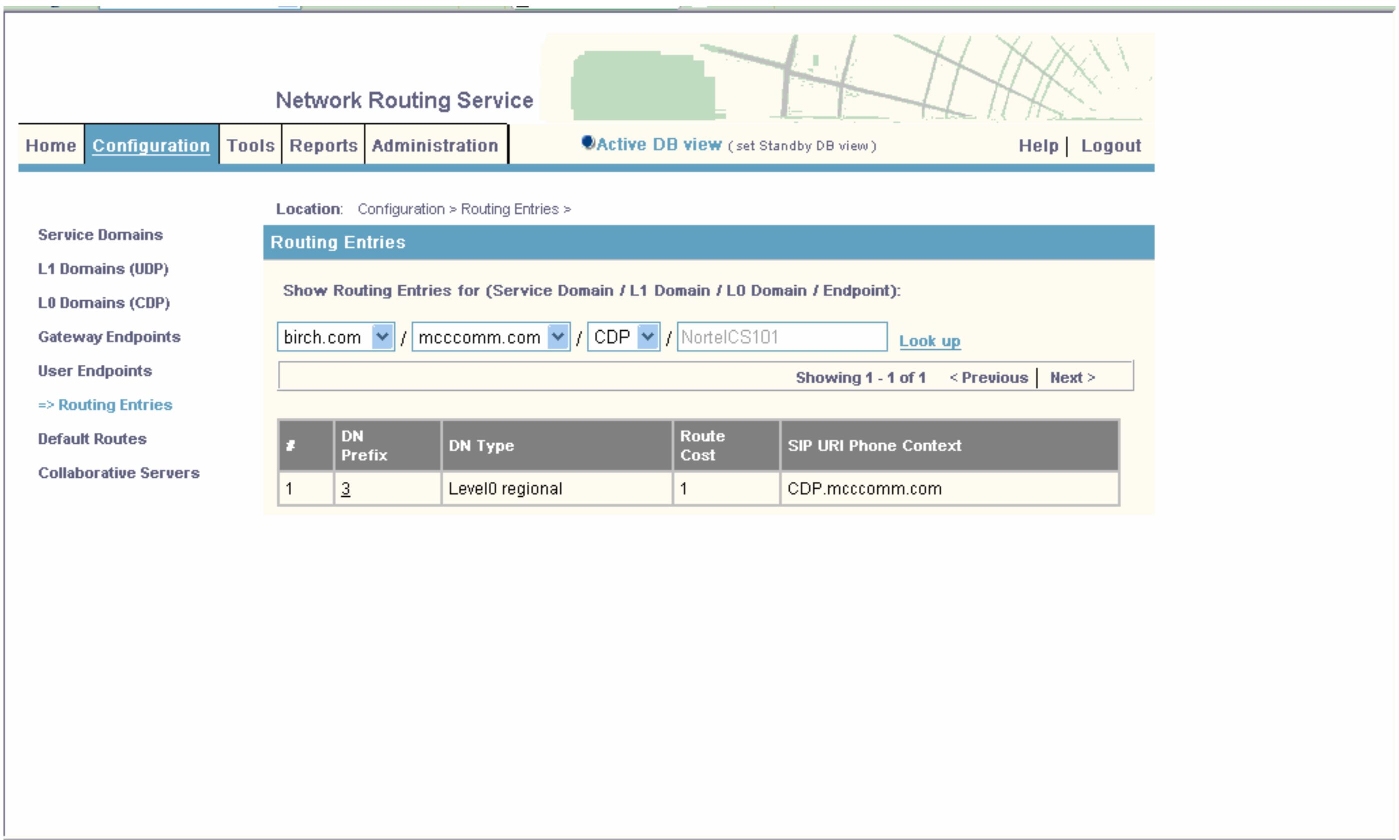


Figure 33. Configure the Routing Entries



Configuring the Nortel Communication Server 1000 (CS1000) PBX

SIP trunk

Call Server Setup Using SSC Card Console:

1. LD 17 – Configure the IP D-channel (signaling channel) between the Call Server and the Signaling Server
3. LD 97 – Configure the Super-loop for the Virtual Trunks
4. LD 14 – Configure the SIP Virtual Trunks to the Signaling Server
5. LD 14 – Configure the Virtual Gateway Trunks
6. LD 16 – Configure the SIP route
7. LD 86 – Configure the Route List Block for the Virtual Trunk route
8. LD 87 – Configure CDP steering codes
9. Configure Digital Stations (Phones)

Signaling Server Setup Using the Nortel Element Manager:

10. Configure the Zones
11. Configure a new IP Telephony Node summary
12. Configure the Node section
13. Configure the VGW and IP phone codec profile section
14. Configure the Quality of Service (QoS) section
15. Configure LAN Configuration section
16. Configure the SIP GW Setting section
17. Configure the Card section for the MC-32 VGMC card section
18. Configure the Signaling Server section

NRS (Network Routing Server):

19. Configure the System Wide Settings
20. Configure the NRS Server Settings
21. Configure a Service Domain
22. Configure a L1 Domain (UDP)
23. Configure a L0 Domain (CDP)
24. Configure a SIP gateway
25. Configure the Routing Entries

Call Server Setup using SSC Card Console:

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

```
>ld 22
REQ prt
TYPE
adna dch 3
TYPE adan dch 3

ADAN     DCH 3
  CTYP DCIP
  DES  IP_Trunk_DCH
  USR  ISLD
  ISLM 4000
```

```
SSRC 1800
OTBF 32

NASA YES
IFC  SL1
CNEG 1
RLS  ID  4
RCAP ND2 MWI CPK
MBGA NO
H323
  OVLR NO
  OVLS NO
```

2. LD 97 – Configure the Super-loop for the Virtual Trunks

```
>ld 97
SCSYS000
MEM AVAIL: (U/P): 2825281    USED U P: 218518 69160    TOT: 3112959
DISK RECS AVAIL: 1152
REQ  prt
TYPE supl
SUPL


SUPL  SUPT SLOT XPEC0   XPEC1

 000  STD  LEFT 01 0 1  -- - -
 004  STD  LEFT 02 0 1  -- - -
 008  STD  LEFT 03 0 1  -- - -
 012  STD  LEFT 04 0 1  -- - -
 016  STD  LEFT 05 0 1  -- - -
 032  STD  LEFT 06 0 3  -- - -
 036  STD  LEFT 07 0 3  -- - -
 040  STD  LEFT 08 0 3  -- - -
 044  STD  LEFT 10 0 3  -- - -
 048  STD  LEFT 09 0 3  -- - -
 064  STD  LEFT 11 0 3  -- - -
 068  STD  LEFT 12 0 3  -- - -
 072  STD  LEFT 13 0 3  -- - -
 096  VIRTUAL CARDS 61 - 64  81 - 84
 100  VIRTUAL CARDS 65 - 68  85 - 88
 128  STD  LEFT 32 0 1  33 2 3
 132  STD  LEFT 34 0 1  35 2 3
 136  STD  LEFT 36 0 1  37 2 3
 140  STD  LEFT 38 0 1  39 2 3
 144  STD  LEFT 40 0 1  41 2 3
 148  STD  LEFT 42 0 1  43 2 3
 152  STD  LEFT 44 0 1  45 2 3
 156  STD  LEFT 46 0 1  47 2 3
```

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

```
>ld 20
```

PT0000
REQ: prt
TYPE: tnb
TN 620 0 0=> SIP Virtual trunk to Signaling Server
DATE
PAGE
DES

DES SIP_IP_VTRK
TN   062 0 00 00 VIRTUAL
TYPE IPTI
CDEN 8D
CUST 0
XTRK VTRK
ZONE 000
LDOP BOP
TIMP 600
BIMP 600
AUTO_BIMP NO
TRK ANLG
NCOS 0
RTMB 10 1
CHID 1
TGAR 1
STRI/STRO IMM IMM
SUPN YES
AST NO
IAPG 0
CLS CTD DTN WTA LPR APN THFD
     P10 NTC MID
TKID
AACR NO
DATE 25 FEB 2005

NACT

4. LD 14 – Configure the Virtual Gateway Trunks (upto 32 trunks per MC-32)

>ld 20

PT0000
REQ: prt
TYPE: tnb
TN   3
CDEN
CUST
DATE
PAGE
DES

DES

```
TN   003 0 00 00
TYPE VGW
CUST 0
XTRK MC32
ZONE 000

DES
TN   003 0 00 01
TYPE VGW
CUST 0
XTRK MC32
ZONE 000
```

5. LD 16 – Configure the SIP route

```
>ld 21
PT1000

REQ: prt
TYPE: rdb
CUST 0
ROUT 10

TYPE RDB
CUST 00
DMOD
ROUT 10
DES  SIP_TIE
TKTP TIE
NPID_TBL_NUM   0
ESN  NO
CNVT NO
SAT  NO
RCLS EXT
VTRK YES
ZONE 000
PCID SIP
CRID YES
NODE 101
DTRK NO
ISDN YES
   MODE ISLD
   DCH  3
   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 2310
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


PAGE 002

    TFD  0
    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
FRL  0 0
FRL  1 0
FRL  2 0
FRL  3 0
FRL  4 0
FRL  5 0
FRL  6 0
FRL  7 0
OHQ  NO
OHQT 00
CBQ  NO
AUTH NO
TTBL 0
ATAN NO
OHTD NO
PLEV 2
ALRM NO
DTA015 2

ART  0
SGRP 0
AACR NO
```

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

```
>ld 86
ESN000

MEM AVAIL: (U/P): 2825281    USED U P: 218518 69160    TOT: 3112959
DISK RECS AVAIL: 1152
REQ prt
CUST 0
FEAT rlb
RLI 10

RLI  10
ENTR 0
LTER NO
ROUT 10
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
SBOC NRR
```

IDBB DBD
IOHQ NO
OHQ NO
CBQ NO

ISET 0
NALT 5
MFRL 0
OVLL 0

MEM AVAIL: (U/P): 2825281   USED U P: 218518 69160   TOT: 3112959
DISK RECS AVAIL: 1152

7. LD 87 – Configure CDP steering codes

>ld 87
ESN000

MEM AVAIL: (U/P): 2825281   USED U P: 218518 69160   TOT: 3112959
DISK RECS AVAIL: 1152
REQ prt
CUST 0
FEAT cdp
TYPE dsc
DSC

DSC 70 => Note: Dialing plan
FLEN 0
DSP LSC
RLI 10 => Note: SIP Route list used for DSC dialed numbers
NPA
NXX

8. LD 11 – Configure Digital Stations (Phones)

>ld 11
SL1000
MEM AVAIL: (U/P): 2718718   USED U P: 327039 50818   TOT: 3096575
DISK RECS AVAIL: 1152
DIGITAL TELEPHONES AVAIL: 0   USED: 8   TOT: 8
IP USERS AVAIL: 2   USED: 6   TOT: 8
BASIC IP USERS AVAIL: 7   USED: 1   TOT: 8
ACD AGENTS AVAIL: 10   USED: 0   TOT: 10
PCA AVAIL: 0   USED: 0   TOT: 0
AST   AVAIL: 1   USED: 0   TOT: 1
TNS   AVAIL: 2296   USED: 204   TOT: 2500
DATA PORTS AVAIL: 2500   USED: 0   TOT: 2500

REQ: prt
TYPE: 2616
TN   10

```
DATE
PAGE
DES

DES  CS101A
TN   001 0 00 00
TYPE 2616
CDEN 8D
CUST 0
AOM  0
FDN  7008
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 CMSD SLKD CCSD SWD LND CNDA
     CFTA SFD MRD DDV CNIA CDCA MSID DAPA BFED RCBD
     ICDD CDMD LLCN MCTD CLBD AUTU
     GPUD DPUD DNDA CFXA ARHD CLTD ASCD
     CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD AHD
     DDGA NAMA
DTA015 2

     DRDD EXR0
     USRD ULAD RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3 MCBN CDMR
CPND_LANG ENG
RCO  0
EFD  7008
HUNT 7008
EHT  7008
LHK  0
PLEV 02
CSDN
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
DNDR 0
KEY  00 SCR 2213 0     MARP
        CPND
          NAME ZEUS13
          XPLN 9
          DISPLAY_FMT FIRST,LAST
     01 SCR 2212 0     MARP
        CPND
```

```
        NAME ZEUS12
        XPLN 6
        DISPLAY_FMT FIRST,LAST
    02
    03 CFW 4  7008
    04 AO6
    05 TRN
    06
    07
    08
    09
    10
    11
    12 XMWK 2217 2212
    13 MIK
    14 MCK
    15 RGA
DATE  9 MAY 2007

NACT


REQ: PRT
TYPE:2616
TN1 1
DATE
PAGE
DES

DES  CS101A
TN   001 0 00 01
TYPE 2616
CDEN 8D
CUST 0
AOM  0
FDN  4000
TGAR 1
LDN  NO
NCOS 0
SGRP 0
RNPG 0
SCI  0
SSU
XLST
CLS  CTD FBD WTA LPR MTD FND HTA ADD HFD
    MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
    POD DSX VMD CMSD SLKD CCSD SWD LND CNDA
    CFTA SFD MRD DDV CNIA CDCA MSID DAPA BFED RCBD
    ICDD CDMD LLCN MCTD CLBD AUTU
    GPUD DPUD DNDA CFXA ARHD CLTD ASCD
    CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD AHD
    DDGA NAMA
    DRDD EXR0
```

```
    USRD ULAD RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3 MCBN CDMR
CPND_LANG ENG
RCO  0
EFD  4000
HUNT 4000
EHT  4000
LHK  0
PLEV 02
CSDN
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
DNDR 0
KEY  00 SCR 2214 0     MARP
       CPND
DTA015 2

         NAME Zeus14
         XPLN 7
         DISPLAY_FMT FIRST,LAST
    01
    02
    03 CFW 4  7008
    04 AO6
    05 TRN
    06
    07
    08
    09
    10
    11
    12
    13 MIK
    14 MCK
    15 RGA
DATE  9 MAY 2007
```

Acronyms

Acronym	Definitions
ANF-PR	Additional Network Feature Path Replacement
AOC	Advice-of-charge. Information element is sent with the connection setup information for incoming Euro-ISDN connections. The AOC IE is used for call charge calculation.
CUCM	Cisco Unified Communication Manager
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
CMM	Communication Media Module (CMM) is a Cisco Catalyst® 6500 Series and Cisco 7600 Series line card that provides flexible and high-density T1/E1 gateways
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
MWI	Message Waiting Indicator

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