



# 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

## Table of Contents

Introduction .....	2
Network Topology.....	3
Limitations.....	4
System Components .....	5
Hardware Requirements .....	5
Software Requirements .....	5
Features .....	5
Features Supported.....	5
Features Not Supported .....	5
Configuration.....	6
Configuring Cisco Unified Communications Manager 4.2 .....	6
Configuring the Cisco 3825 IP to IP Gateway.....	18
Configuring the Nortel Communication Server 1000 (CS1000) PBX.....	46
Acronyms .....	56



## 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](#)

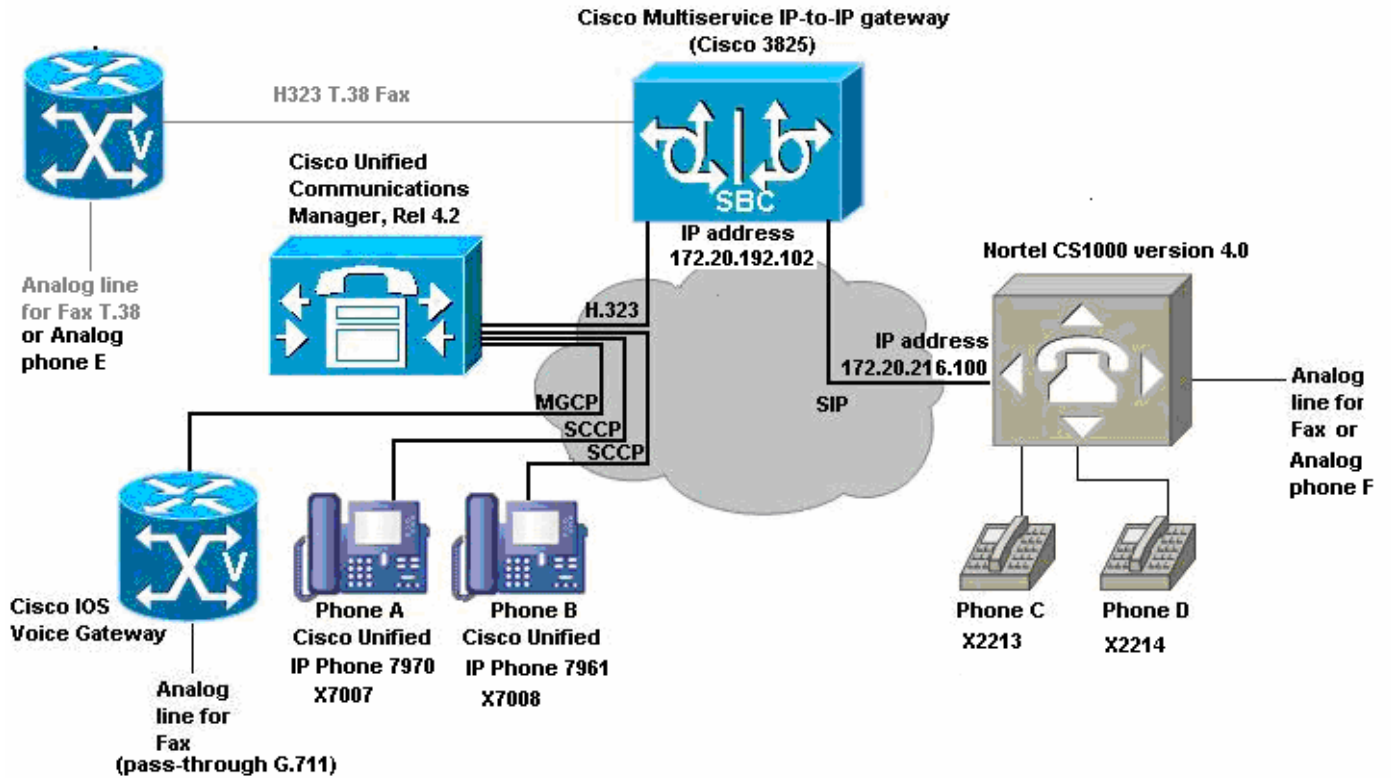
[Cisco 7301 Routers](#)

[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 across H323/SIP trunk. Nortel use SIP P-asserted-ID method to send name and number. Cisco IP-to-IP Gateway (IIPGW) 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 DTMF signaling.

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, IIPGW 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

The screenshot shows the Cisco CallManager Administration web interface. At the top, there is a navigation menu with links for System, Route Plan, Service, Feature, Device, User, Application, and Help. Below the menu is a header banner with the text "Cisco CallManager Administration For Cisco IP Telephony Solutions" and the Cisco Systems logo. The main content area is titled "Find and List Regions" and includes a link for "Add a New Region". A search result is displayed, indicating "1 matching record(s) for Region Name begins with ''". Below this, there is a search form with a dropdown menu set to "begins with", an empty text input field, and a "Find" button. The form also shows "and show 20 items per page" and a note: "To list all items, click Find without entering any search text." The search results are listed in a table with one entry: "Default" under the "Region" column. At the bottom of the results area, there is a "Delete Selected" button and navigation links: "First Previous Next Last". A pagination indicator shows "Page 1 of 1".



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

System Route Plan Service Feature Device User Application Help

Cisco CallManager Administration  
For Cisco IP Telephony Solutions

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## Region Configuration

[Add a New Region](#)  
[Back to Find/List Regions](#)  
[Dependency Records](#)

**Region: Default**  
Status: Ready

### Region Information

Region Name\*

### Call Information

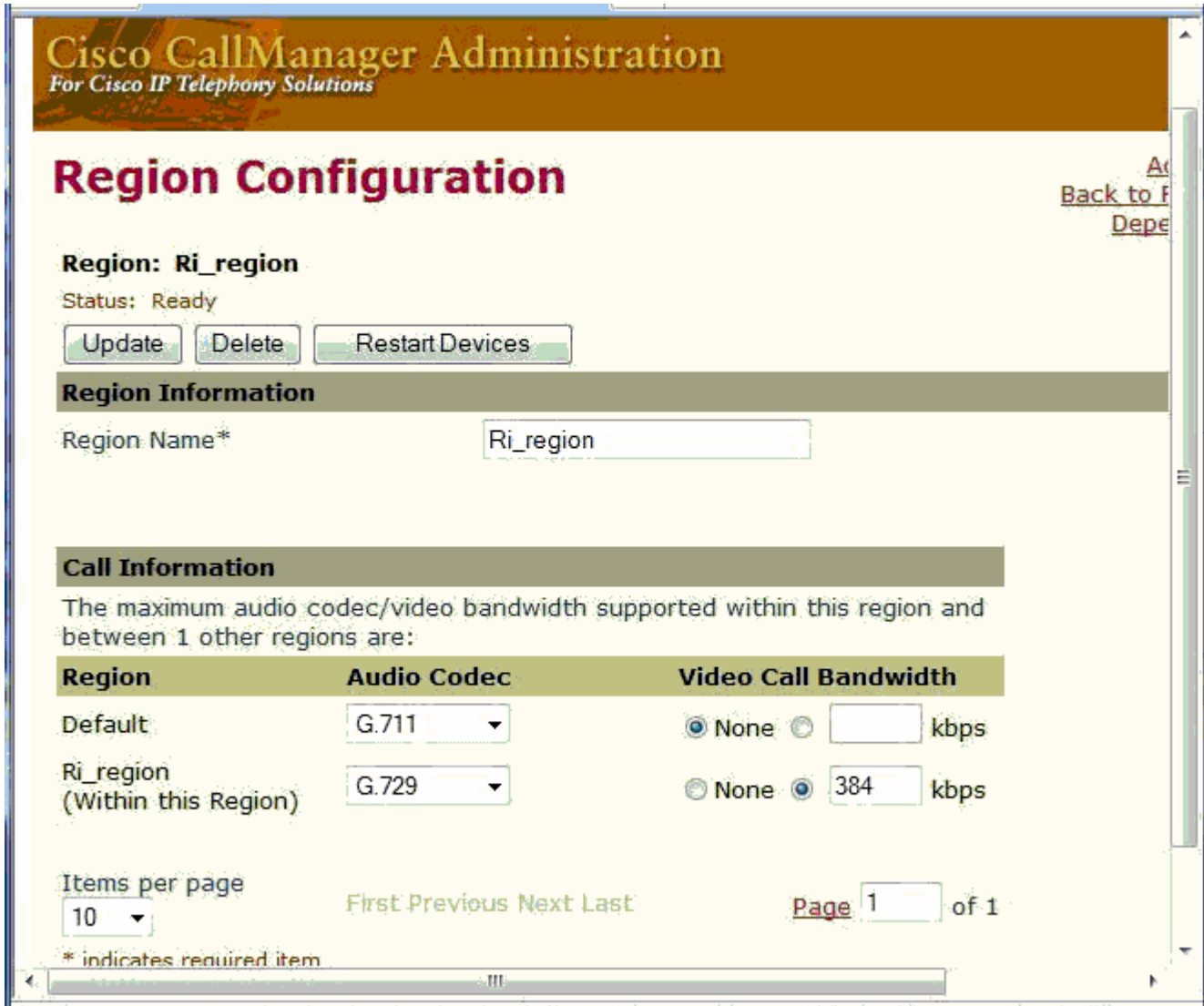
The maximum audio codec/video bandwidth supported within this region is:

Region	Audio Codec	Video Call Bandwidth
Default (Within this Region)	<input type="text" value="G.711"/>	<input type="radio"/> None <input checked="" type="radio"/> <input type="text" value="384"/> kbps

Items per page:  First Previous Next Last Page  of 1

\* indicates required item.

Figure 4. Region configuration. Ri\_region set for G.729 Codec.



**Cisco CallManager Administration**  
For Cisco IP Telephony Solutions

## Region Configuration

Region: **Ri\_region**  
Status: Ready

Update Delete Restart Devices

### Region Information

Region Name\*

### Call Information

The maximum audio codec/video bandwidth supported within this region and between 1 other regions are:

Region	Audio Codec	Video Call Bandwidth
Default	<input type="text" value="G.711"/>	<input checked="" type="radio"/> None <input type="radio"/> <input type="text" value=""/> kbps
Ri_region (Within this Region)	<input type="text" value="G.729"/>	<input type="radio"/> None <input checked="" type="radio"/> <input type="text" value="384"/> kbps

Items per page:  First Previous Next Last Page  of 1

\* indicates required item





Figure 5. Configurations Page 1 of 2

System Route Plan Service Feature Device User Application Help

**Cisco CallManager Administration**  
For Cisco IP Telephony Solutions

## Gateway Configuration [Back to F](#) [De](#)

**Product : H.323 Gateway**  
**Gateway : 172.20.192.102**  
**Device Protocol: H.225**  
**Registration: Unknown**  
**IP Address: 172.20.192.102**

Status: Ready

### Device Information

Device Name*	172.20.192.102
Description	gateway ip2ip
Device Pool*	Default
Common Profile	< None >
Call Classification*	Use System Default
Media Resource Group List	Ri_MRGL
Location	< None >
AAR Group	< None >
Tunneled Protocol	< None >
Signaling Port*	1720

Media Termination Point Required

Retry Video Call as Audio

Wait for Far End H.245 Terminal Capability Set

Path Replacement Support



Configuring the IP -to- IP gateway

<b>Multilevel Precedence and Preemption (MLPP) Information</b>	
MLPP Domain (e.g., "0000FF")	<input type="text"/>
MLPP Indication	Not available on this device
MLPP Preemption	Not available on this device

<b>Call Routing Information</b>	
<b>Inbound Calls</b>	
Significant Digits*	All
Calling Search Space	< None >
AAR Calling Search Space	< None >
Prefix DN	<input type="text"/>
<input type="checkbox"/> Redirecting Number IE Delivery - Inbound	
<input checked="" type="checkbox"/> Enable Inbound FastStart	

<b>Outbound Calls</b>	
Calling Party Selection*	Originator
Calling Party Presentation*	Default
Called party IE number type unknown*	Cisco CallManager
Calling party IE number type unknown*	Cisco CallManager
Called Numbering Plan*	Cisco CallManager
Calling Numbering Plan*	Cisco CallManager
Caller ID DN	<input type="text"/>
<input checked="" type="checkbox"/> Display IE Delivery	
<input checked="" type="checkbox"/> Redirecting Number IE Delivery - Outbound	
<input checked="" type="checkbox"/> Enable Outbound FastStart	
Codec For Outbound FastStart*	G729



Figure 6. Media Termination Point list

System Route Plan Service Feature Device User Application Help

**Cisco CallManager Administration**  
For Cisco IP Telephony Solutions

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## Find and List Media Termination Point

[Add a New Media Termination Point](#)

3 matching record(s) for Name begins with ""







Find Media Termination Points where Name begins with  Find

and show 20 items per page

To list all items, click Find without entering any search text.

### Matching record(s) 1 to 3 of 3

Real-time Information Service returned information for 3 of 3 devices listed below.

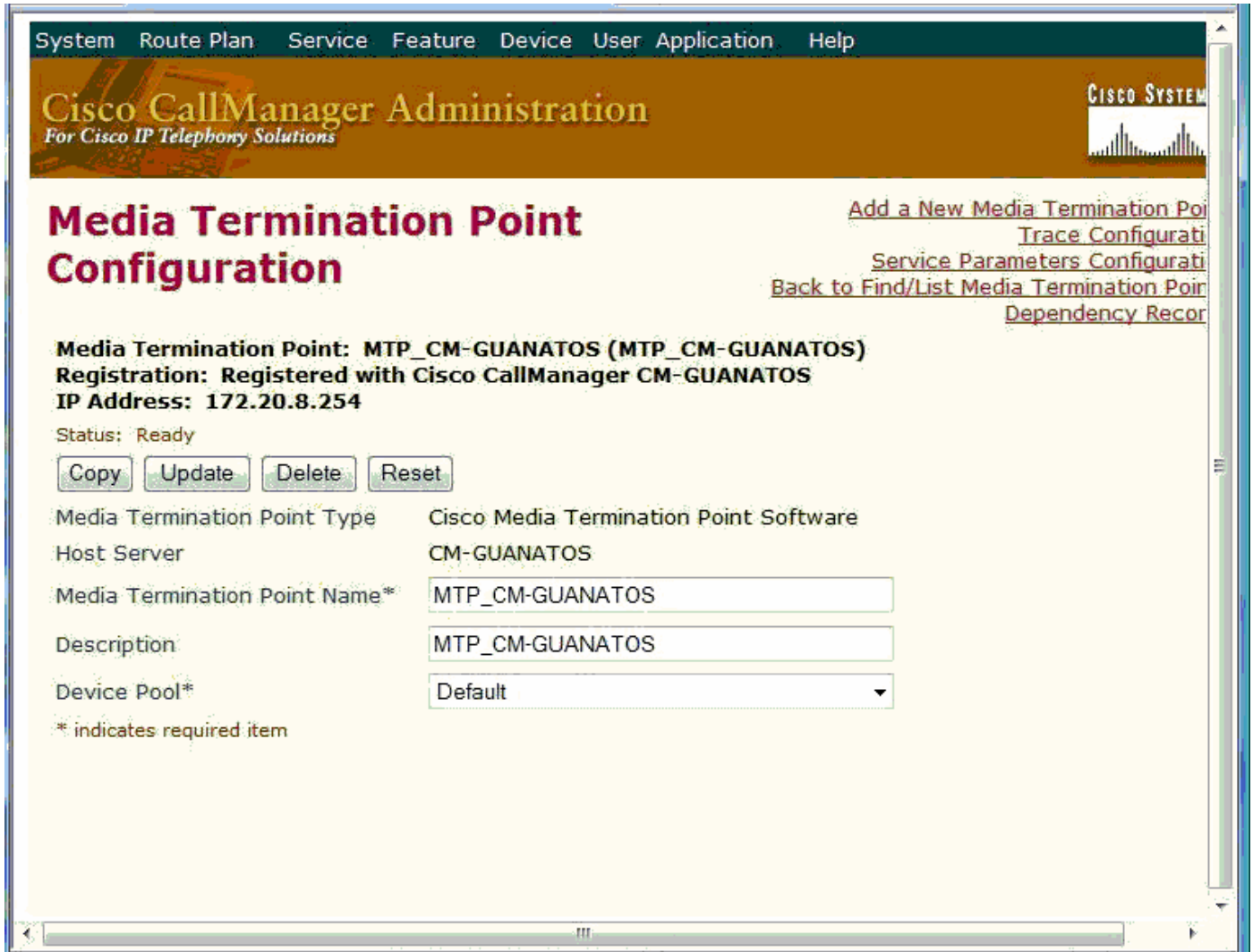
<input type="checkbox"/>	Media Termination Point	Description	Device Pool	Status	IP Address	Copy
<input type="checkbox"/>	 MTP_CM-GUANATOS	MTP_CM-GUANATOS	Default	CM-GUANATOS	172.20.8.254	
<input type="checkbox"/>	 MTP0013C4037300	MTP0013C4037300	Default	CM-GUANATOS	172.20.15.199	
<input type="checkbox"/>	 MTP0015F90D1590	MTP0015F90D1590	Default	CM-GUANATOS	172.20.192.102	

Delete Selected Reset Selected First Previous Next Last Page 1 of



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.

The screenshot displays the Cisco CallManager Administration web interface. At the top, there is a navigation menu with links for System, Route Plan, Service, Feature, Device, User, Application, and Help. Below the menu is a header banner for "Cisco CallManager Administration" with the tagline "For Cisco IP Telephony Solutions" and the Cisco logo. The main content area is titled "Media Termination Point Configuration". On the right side of this area, there are several navigation links: "Add a New Media Termination Point", "Trace Configuration", "Service Parameters Configuration", "Back to Find/List Media Termination Point", and "Dependency Recorder". The configuration details for a specific MTP are shown: "Media Termination Point: MTP0015F90D1590 (MTP0015F90D1590)", "Registration: Registered with Cisco CallManager CM-GUANATOS", and "IP Address: 172.20.192.102". The status is "Ready". Below this, there are four buttons: "Copy", "Update", "Delete", and "Reset". The configuration fields include: "Media Termination Point Type" set to "Cisco IOS Enhanced Software Media Termination Point", "Media Termination Point Name\*" with the value "MTP0015F90D1590", "Description" with the value "MTP0015F90D1590", and "Device Pool\*" set to "Default". A note at the bottom left states "\* indicates required item".



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

The screenshot displays the Cisco CallManager Administration web interface. At the top, there is a navigation menu with links for System, Route Plan, Service, Feature, Device, User, Application, and Help. Below the menu is a header banner with the text "Cisco CallManager Administration For Cisco IP Telephony Solutions" and the Cisco logo. The main content area is titled "Media Termination Point Configuration" in large red font. To the right of the title are several links: "Add a New Media Termination Point", "Trace Configuration", "Service Parameters Configuration", "Back to Find/List Media Termination Points", and "Dependency Recorder".

The configuration details for the selected Media Termination Point are as follows:

- Media Termination Point:** MTP0013C4037300 (MTP0013C4037300)
- Registration:** Registered with Cisco CallManager CM-GUANATOS
- IP Address:** 172.20.15.199
- Status:** Ready

Below the status, there are four buttons: Copy, Update, Delete, and Reset. The configuration fields are:

- Media Termination Point Type:** Cisco IOS Enhanced Software Media Termination Point
- Media Termination Point Name\*:** MTP0013C4037300
- Description:** MTP0013C4037300
- Device Pool\*:** Default

A note at the bottom left states: "\* indicates required item".





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.

**Cisco CallManager Administration**  
For Cisco IP Telephony Solutions

## Media Resource Group Configuration

[Add a New Media Resource](#)  
[Back to Find/List Media Resources](#)  
[Dependencies](#)

**Media Resource Group: RI\_MRG (used by 3 devices)**  
Status: Ready

### Media Resource Group Information

Media Resource Group Name\*: RI\_MRG  
Description: RI\_MRG

### Devices for this Group

Available Media Resources\*\*

- ANN\_CM-GUANATOS (ANN)
- CFB\_CM-GUANATOS (CFB)
- MOH\_CM-GUANATOS (MOH)
- MTP\_CM-GUANATOS (MTP)
- mtp001193685140 (XCODE)

Selected Media Resources\*

- MTP0015F90D1590 (MTP)

Use Multicast for MOH Audio (requires at least one multicast MOH resource)

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



The screenshot displays the Cisco CallManager Administration web interface. At the top, there is a navigation menu with links for System, Route Plan, Service, Feature, Device, User, Application, and Help. Below the menu is a header banner for "Cisco CallManager Administration For Cisco IP Telephony Solutions". The main heading is "Media Resource Group Configuration". To the right of the heading are links for "Add a New Media", "Back to Find/List Media R", and "Deper".

The configuration details for "Media Resource Group: MRG1 (used by 25 devices)" are shown. The status is "Ready". There are four buttons: "Copy", "Update", "Delete", and "Reset Devices".

**Media Resource Group Information**

Media Resource Group Name*	MRG1
Description	MRG1

**Devices for this Group**

Available Media Resources\*\*

- ANN\_CM-GUANATOS (ANN)
- CFB\_CM-GUANATOS (CFB)
- MOH\_CM-GUANATOS (MOH)
- mtp001193685140 (XCODE)
- MTP0013C4037300 (MTP)

Selected Media Resources\*

- MTP\_CM-GUANATOS (MTP)

Use Multicast for MOH Audio (requires at least one multicast MOH resource)





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\_MRGL is selected for the Media Resource Group List RI\_MRGL.

The screenshot shows the Cisco CallManager Administration web interface. At the top, there is a navigation menu with links for System, Route Plan, Service, Feature, Device, User, Application, and Help. Below the menu is a header banner with the text "Cisco CallManager Administration For Cisco IP Telephony Solutions" and the Cisco logo. The main heading is "Media Resource Group List Configuration". To the right of the heading are three links: "Add a New Media Resource Gr...", "Back to Find/List Media Resource Gro...", and "Dependency I...".

The configuration page is for a "Media Resource Group List: Ri\_MRGL (used by 3 devices)". The status is "Ready". There are four buttons: "Copy", "Update", "Delete", and "Reset Devices".

Under "Media Resource Group List Information", the "Media Resource Group List Name\*" field contains "Ri\_MRGL".

Under "Media Resource Groups for this List", there are two sections:

- "Available Media Resource Groups" with a list box containing "MRG1".
- "Selected Media Resource Groups\*" with a list box containing "RI\_MRGL". Below this list box is the text "(Groups listed in order of priority)".

At the bottom left, there is a note: "\* indicates required item".



## Configuring the Cisco 3825 IP to IP Gateway

### 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](mailto: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 11
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
!
```

<sup>1</sup> 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 GigabitEthernet0/02  
sccp ccm 172.20.8.254 identifier 1 version 4.13  
sccp  
!  
sccp ccm group 1  
associate ccm 1 priority 1  
associate profile 1 register MTP0015f90d15904  
!  
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 15  
session target ipv4:172.20.8.254  
session transport tcp6  
dtmf-relay h245-alphanumeric7  
fax-relay ecm disable8  
no vad9  
!  
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 210  
destination-pattern 2...
```

<sup>2</sup> This section was added for G.729B. Cisco Unified 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.

<sup>3</sup> 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.

<sup>4</sup> 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).

<sup>5</sup> Inserted for voice codec negotiation test. Codecs were specified in dial peers for other tests.

<sup>6</sup> Changed this line to "session transport udp" to test UDP session transport.

<sup>7</sup> Remove for G711 and G729 codec.

<sup>8</sup> This was removed at the time of G711 pass-through FAX testing.

<sup>9</sup> Removed when codec was set to G729B, as VAD is not optional with G729B

<sup>10</sup> 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 G711ulaw11

!
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#
```

---

<sup>11</sup> 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 (fcl)

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 1000000  
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 mtp00119368514012
!
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 ipv4:172.20.192.10213
no fax-relay sg3-to-g3
!
dial-peer voice 4001 pots
destination-pattern 4155554001
```

<sup>12</sup> 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.

<sup>13</sup> 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:

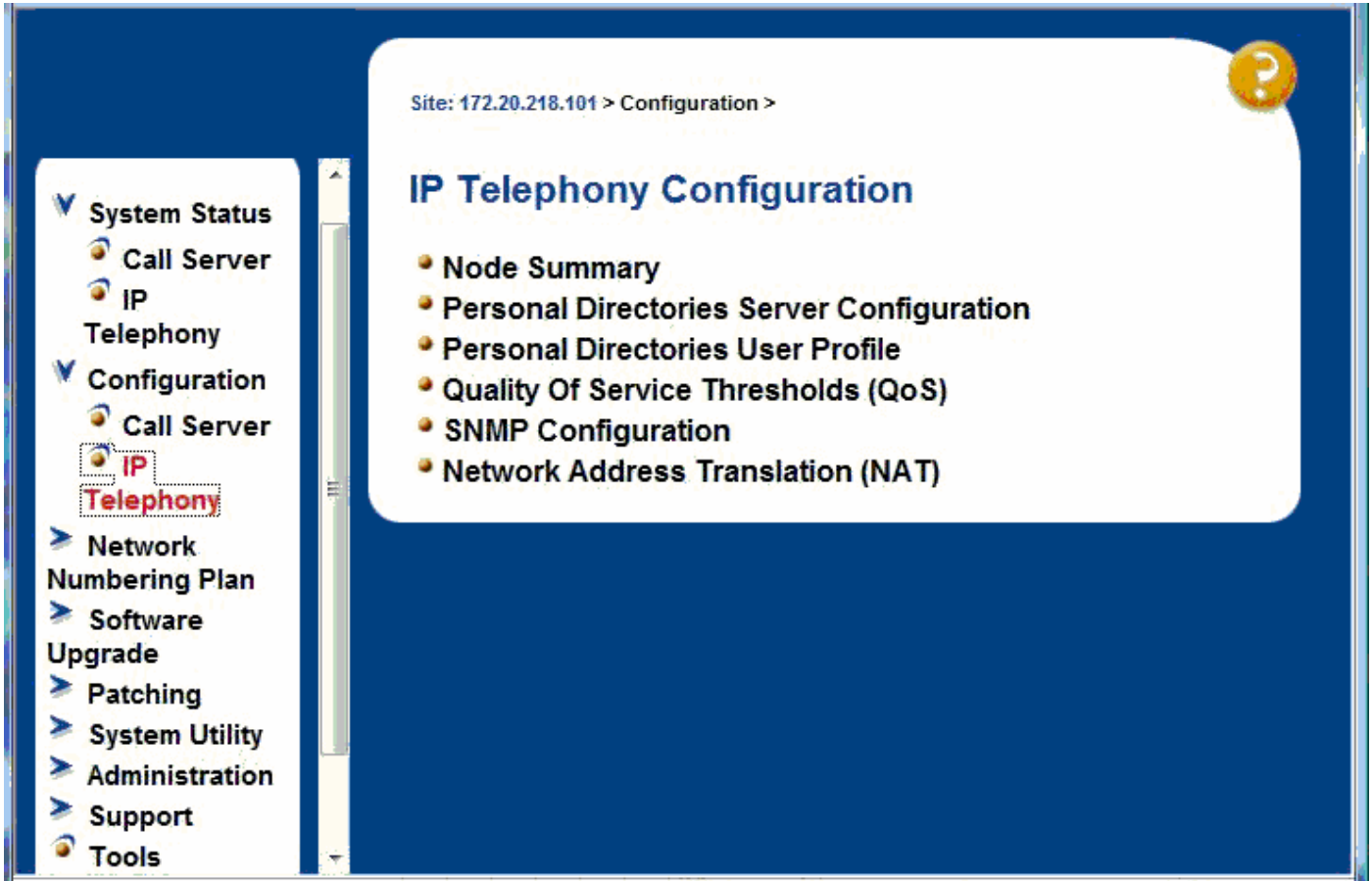
Figure 13. Configure the Zones

Site: 172.20.218.101 > Configuration > Call Server Configuration > Zone List > Zone 0 >

### Zone Basic Property and Bandwidth Management

Input Description	Input Value
Zone Number (ZONE):	<input type="text" value="0"/>
Intrazone Bandwidth (INTRA_BW):	<input type="text" value="10000"/>
Intrazone Strategy (INTRA_STGY):	<input type="text" value="Best Quality (BQ)"/>
Interzone Bandwidth (INTER_BW):	<input type="text" value="10000"/>
Interzone Strategy (INTER_STGY):	<input type="text" value="Best Quality (BQ)"/>
Resource Type (RES_TYPE):	<input type="text" value="Shared (SHARED)"/>
Branch Office Support (ZBRN):	<input type="checkbox"/>
Description (ZDES):	<input type="text"/>

Figure 14. IP Telephony Configuration



**Figure 15.** Configure a new IP Telephony Node summary

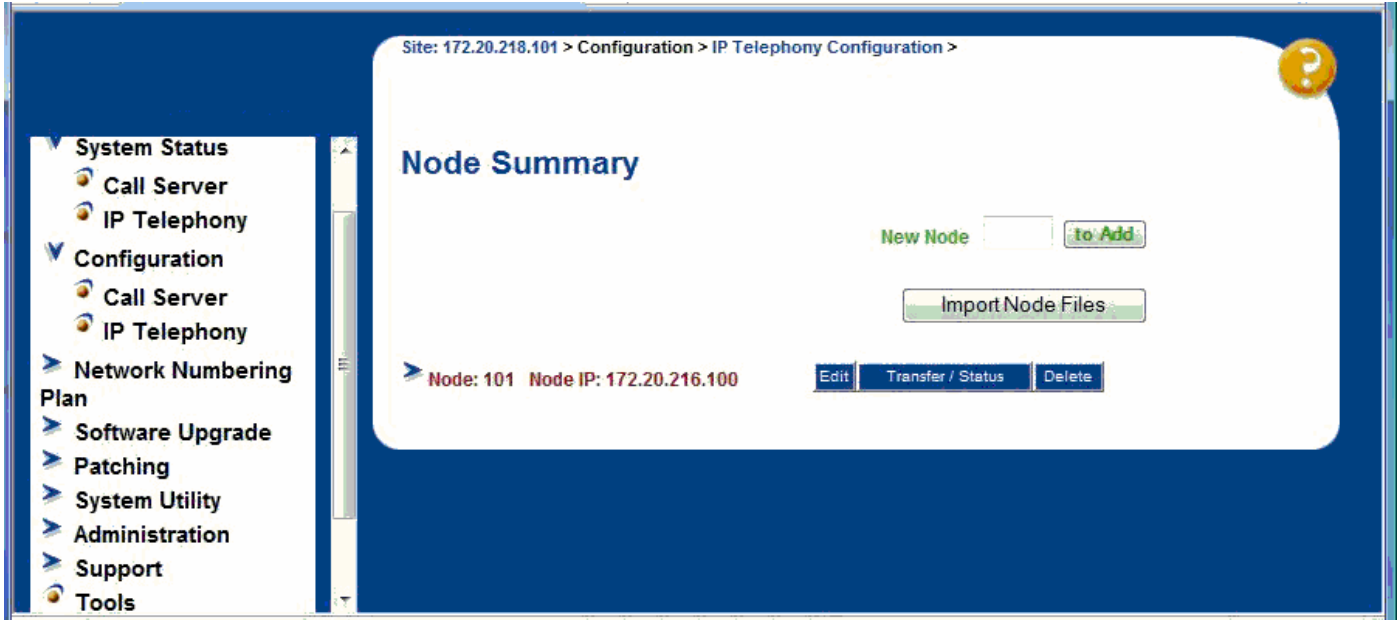




Figure 16. Configure the Node section

Site: 172.20.218.101 > Configuration > IP Telephony Configuration > Node Summary > IP Telephony: Node ID 101 >

### Edit

**Node**

Node ID	101
Voice LAN (TLAN) Node IP address	<input type="text" value="172.20.216.100"/>
Management LAN (ELAN) gateway IP address	<input type="text" value="172.20.218.1"/>
Management LAN (ELAN) subnet mask	<input type="text" value="255.255.255.0"/>
Voice LAN (TLAN) subnet mask	<input type="text" value="255.255.255.0"/>

**SNMP**

**VGW and IP phone codec profile**

**QoS**

**LAN configuration**

**SNTP**

**H323 GW Settings**



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

The screenshot shows the configuration page for the VGW and IP phone codec profile. On the left is a navigation menu with categories like System Status, Configuration, Network Numbering Plan, Software Upgrade, Patching, System Utility, Administration, Support, Tools, and Logout. The main content area is titled 'VGW and IP phone codec profile' and contains several settings:

- Enable Echo canceller:
- Echo canceller tail delay: 128
- Voice activity detection threshold: -17 (Range: -20 to +10)
- Idle noise level: -65 (Range: -327 to +327)
- DTMF Tone detection:
- Enable V.21 FAX tone detection:
- FAX maximum rate (bps): 14400
- FAX playout nominal delay: 100 (Range: 0 to 300)
- FAX no activity timeout: 20 (Range: 10 to 32000)
- FAX packet size: 30

Below these settings is a table for selecting codecs:

Codec	G711	Select	<input checked="" type="checkbox"/>
Codec	G729A	Select	<input checked="" type="checkbox"/>
Codec	G723.1	Select	<input type="checkbox"/>
Codec	T38 FAX	Select	<input checked="" type="checkbox"/>
QoS			
LAN configuration			
SNTP			



Figure 18. CODEC profile selection

The screenshot shows a configuration page for CODEC profiles. On the left is a navigation menu with categories like System Status, Configuration, Network Numbering Plan, Software Upgrade, Patching, System Utility, Administration, Support, Tools, and Logout. The main content area displays three CODEC profiles:

- Codec G711**: Select . Settings include Voice payload size (ms/frame) at 20, Voice playback (jitter buffer) nominal delay at 40, and Voice playback (jitter buffer) maximum delay at 80. A red note states: "Modifications may cause changes to dependent settings". VAD is disabled.
- Codec G729A**: Select . Settings include Voice payload size (ms/frame) at 20, Voice playback (jitter buffer) nominal delay at 40, and Voice playback (jitter buffer) maximum delay at 80. A red note states: "Modifications may cause changes to dependent settings". VAD is disabled.
- Codec G723.1**: Select . No settings are visible.
- Codec T38 FAX**: Select . Settings include Codec Name: T38 FAX.

Figure 19. Configure the QoS section

The screenshot shows a configuration page for QoS settings. On the left is a navigation menu with categories like System Status, Configuration, Network Numbering Plan, Software Upgrade, Patching, System Utility, Administration, Support, Tools, and Logout. The main content area displays the following QoS settings:

- Codec G711**: Select
- Codec G729A**: Select
- Codec G723.1**: Select
- Codec T38 FAX**: Select
- QoS**:
  - Diffserv Codepoint(DSCP) Control packets: 40 (Range: 0 to 63)
  - Diffserv Codepoint(DSCP) Voice packets: 46 (Range: 0 to 63)
  - Enable 802.1Q support:
  - 802.1Q Bits value (802.1p): 6 (Range: 0 to 7)
  - LAN configuration:
  - SNTP:
  - H323 GW Settings:
  - Firmware:
  - SIP GW Settings:
  - SIP URI Map:
  - SIP CD Services:
  - Cards:
  - Signaling Servers:



Figure 20. Configure LAN and SNTP Configuration section

The screenshot displays the Cisco configuration interface for LAN and SNTP settings. On the left is a navigation menu with categories like System Status, Configuration, Network, and Support. The main area is divided into sections: LAN configuration, Management LAN (ELAN) configuration, Voice LAN (TLAN) configuration, Routes, SNTP, and SNTP Client. Each section contains various configuration fields with values and range warnings.

Section	Field	Value	Range
Management LAN (ELAN) configuration	Call server IP address	172.20.218.101	
	Survivable Succession Media Gateway IP address	0.0.0.0	
	Signaling port	15000	Range: 1024 to 65535
	Broadcast port	15001	Range: 1024 to 65535
Voice LAN (TLAN) configuration	Signaling port	5000	Range: 1024 to 65535
	Voice port	5200	Range: 1024 to 65535
Routes		<a href="#">Add</a>	
SNTP Server	Mode	active	
	Interval	256	Range: 1 to 2147483647
	Port	20101	
SNTP Client	Mode	passive	
	Interval	256	Range: 1 to 2147483647
	Port	20101	
	SNTP server IP address	0.0.0.0	





Figure 21. Configure the SIP GW Settings section

SIP GW Settings	
Primary Proxy / Re-direct IP address	172.20.216.103
Primary Proxy / Re-direct IP Port	5060
Primary Proxy Supports Registration	<input checked="" type="checkbox"/>
Primary CDS Proxy or Re-direct server flag	<input checked="" type="checkbox"/>
Secondary Proxy / Re-direct IP address	0.0.0.0
Secondary Proxy / Re-direct IP Port	5060
Secondary Proxy Supports Registration	<input type="checkbox"/>
Secondary CDS Proxy or Re-direct server flag	<input type="checkbox"/>

SIP URI Map	
Public E.164/National domain name	+1
Public E.164/Subscriber domain name	+1314
Public E.164/Unknown domain name	
Public E.164/Special Number domain name	
Private/UDP domain name	rtp
Private/CDP domain name	interop.rtp
Private/Special Number domain name	SPN.rtp
Private/Unknown (vacant number routing) domain name	



Figure 22. SIP CD Services

The screenshot shows the 'SIP CD Services' configuration page. On the left is a navigation sidebar with the following items: System Status, Call Server, IP Telephony, Configuration, Call Server, IP Telephony, Network Numbering Plan, Software Upgrade, Patching, System Utility, Administration, Support, Tools, and Logout. The main content area is titled 'SIP CD Services' and contains the following configuration options:

- Service Enabled:**
- Service DN used for making VTRK call from:**
- Converged Telephone Call Forward DN:**
- User Info. field for Invite message on the Converged Desktop MO Set:**
- User Info. field for Invite message on the Converged Desktop MV Set:**
- User Info. field in the notify message for Converged Desktop:**
- RAN route for Announce:**
- Wait time before a caller is sent to RAN Queue:**
- Timeout for Ringing indication of the CD set:**
- Timeout for CD Server:**
- Timeout for call answered by other than CD phone set:**



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

Role	Follower
Management LAN (ELAN) IP address	<input type="text" value="172.20.218.102"/> *
Management LAN (ELAN) MAC address	<input type="text" value="00:11:F9:E4:D0:11"/> *
Voice LAN (TLAN) IP address	<input type="text" value="172.20.216.102"/> *
Voice LAN (TLAN) gateway IP address	<input type="text" value="172.20.216.1"/>
Hostname	<input type="text" value="MG_Node101_3"/> *
Card TN	<input type="text" value="3"/> *
Card processor type	<input type="text" value="Succession Media Card"/> ▼
H323 ID	<input type="text" value="MG_Node101"/>
Enable set TPS	<input checked="" type="checkbox"/>
System name	<input type="text" value="MG_Node_101"/>
System location	<input type="text" value="Dewey Lab"/>
System contact	<input type="text" value="Fred McClintic"/>



Figure 24. Configure the Signaling Server section

Signaling Servers		Add
Signaling Server 172.20.218.103 Properties		
Role	Leader	
Management LAN (ELAN) IP address	172.20.218.103 *	
Management LAN (ELAN) MAC address	00:02:b3:f7:3a:86 *	
Voice LAN (TLAN) IP address	172.20.216.103 *	
Voice LAN (TLAN) gateway IP address	172.20.216.1	
Hostname	SS_Node101_Ldr *	
H323 ID	Gateway_Node101	
Enable set TPS	<input checked="" type="checkbox"/>	
Enable virtual trunk TPS	H.323 and SIP	
Enable SIP Proxy / Redirect Server	<input checked="" type="checkbox"/>	
SIP Transport Protocol	TCP	
Local SIP Port	5060	
SIP Domain name	birch.com	
SIP Gateway Endpoint Name	Gateway_Node101	
SIP Gateway Authentication Password	••••	
Enable H323 Gatekeeper	<input checked="" type="checkbox"/>	
Network Routing Service Role	Primary	
System name	SS_Node101_Ldr	



## NRS (Network Routing Server):

Figure 25. NRS Overview

**Network Routing Service**

Home | Configuration | Tools | Reports | Administration | Help | Logout

Location: Home > NRS Overview >

**Network Routing Service**

Software version	sse-4.00.31
Connected NRS role	PrimaryNRS
Primary NRS IP (TLAN)	172.20.216.103
Primary NRS state	ACTIVE
Alternate NRS IP (TLAN)	Unknown
Alternate NRS state	Unknown

=> NRS Overview  
System Wide Settings  
NRS Server Settings



Figure 26. Configure the System Wide Settings

Network Routing Service

Home Configuration Tools Reports Administration Help | Logout

Location: Home > System Wide Settings >

**System Wide Settings**

NRS Overview  
=> System Wide Settings  
NRS Server Settings

DB sync interval for alternate [Hours]

SIP registration time to live timer [Seconds]

H.323 gatekeeper registration time to live timer [Seconds]

H.323 alias name  \*

Alternate NRS server is permanent

Auto backup time [HH:MM]

Auto backup to FTP site enabled

Auto backup FTP site IP address

Auto backup FTP site path

Auto backup FTP username

Auto backup FTP password



Figure 27. Configure the NRS Server Settings

The screenshot displays the configuration interface for the Network Routing Service (NRS). The page title is "Network Routing Service" and the navigation menu includes Home, Configuration, Tools, Reports, Administration, Help, and Logout. The current location is "Home > NRS Server Settings >".

**NRS Settings**

Host name	SS_Node101_Ldr	*
Primary IP (TLAN)	172.20.216.103	*
Alternate IP (TLAN)	172.20.217.103	*
Control priority	40	

**H.323 Gatekeeper Settings**

Location request (LRQ) response timeout [Seconds]	3
---	---

**SIP Server Settings**

Mode	Redirect
UDP transport enabled	<input checked="" type="checkbox"/>
UDP port	5060
UDP maximum transmission unit (MTU)	1500



Figure 28. SIP Server Setting Cont'd

Network Routing Service

Home Configuration Tools Reports Administration Help | Logout

**SIP Server Settings**

Mode

UDP transport enabled

UDP port

UDP maximum transmission unit (MTU)

TCP transport enabled

TCP port

TCP maximum transmission unit (MTU)

**Network Connection Server (NCS) Settings**

Primary NCS port

Alternate NCS port

Primary NCS timeout [Seconds]

\* Mandatory field indicator





**Figure 29.** Configure a Service Domain

The screenshot shows the Cisco Network Routing Service configuration interface. At the top, there is a navigation bar with tabs for Home, Configuration, Tools, Reports, and Administration. The Configuration tab is active, and the page title is "Network Routing Service". Below the navigation bar, there is a breadcrumb trail: "Location: Configuration > Service Domains > View Service Domain Property >". The main content area is titled "View Service Domain Property" and contains two fields: "Domain name" with the value "birch.com" and a mandatory field indicator (\*), and "Domain description" with a dropdown menu showing "required" and "service domain". A note below the fields states "\*Mandatory field indicator". On the left side, there is a sidebar menu with the following items: "=> Service Domains", "L1 Domains (UDP)", "L0 Domains (CDP)", "Gateway Endpoints", "User Endpoints", "Routing Entries", "Default Routes", and "Collaborative Servers".



Figure 30. Configure a L1 Domain (UDP)

The screenshot displays the 'View L1 Domain Property (birch.com)' configuration page in the Network Routing Service. The page includes a navigation menu with 'Home', 'Configuration', 'Tools', 'Reports', and 'Administration'. The 'Configuration' tab is active, and the 'Active DB view' is selected. The configuration fields are as follows:

Field	Value
Domain name	mccomm.com *
Domain description	Enterprise (company) domain
Endpoint authentication enabled	Authentication off
Authentication password	
E.164 country code	1
E.164 area code	314
International dialing access code	011
L1 domain dialing access code	
National dialing access code	1
Local dialing access code	
Special number 1	
Special number 2	



**Figure 31.** Configure a L0 Domain (CDP)

The screenshot displays the 'View L0 Domain Property' configuration page for a CDP domain. The interface includes a navigation menu with 'Configuration' selected, and a sidebar with options like 'Service Domains', 'L1 Domains (UDP)', and 'Gateway Endpoints'. The main configuration area contains the following fields:

- Domain name: CDP \*
- Domain description: CDP (local extension) domain
- Special number label: [Empty]
- Unqualified number label: [Empty]
- Endpoint authentication enabled: Authentication off
- Authentication password: [Empty]
- E.164 country code: 1
- E.164 area code: 314
- International dialing access code: 011
- L1 domain dialing access code: [Empty]
- National dialing access code: [Empty]
- Local dialing access code: [Empty]



Figure 32. Configure a SIP Gateway

Network Routing Service

Home Configuration Tools Reports Administration Active DB view (set Standby DB view) Help Logout

Location: Configuration > Gateway Endpoints > View Gateway Endpoint Property >

View Gateway Endpoint Property (pbxlab.org / sj / interop)

Service Domains

- L1 Domains (UDP)
- L0 Domains (CDP)
- => Gateway Endpoints
- User Endpoints
- Routing Entries
- Default Routes
- Collaborative Servers

Endpoint name: TonyB

Endpoint description: Tony B IPIPGW testing

Tandem endpoint name: [Look up](#)

Endpoint authentication enabled: Not configured

Authentication password:

E.164 country code:

E.164 area code:

International dialing access code:

L1 domain dialing access code:

National dialing access code:

Local dialing access code:

Special number 1:

Special number 2:

Static endpoint address type: IP version 4

Static endpoint address: 172.20.8.26



Figure 33. Configure the Routing Entries

Network Routing Service

Home Configuration Tools Reports Administration Active DB view (set Standby DB view) Help Logout

Location: Configuration > Routing Entries >

**Routing Entries**

Show Routing Entries for (Service Domain / L1 Domain / L0 Domain / Endpoint):

birch.com / mccomm.com / CDP / NortelCS101 [Look up](#)

Showing 1 - 1 of 1 < Previous | Next >

#	DN Prefix	DN Type	Route Cost	SIP URI Phone Context
1	3	Level0 regional	1	CDP.mccomm.com



## 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  
OVLN NO  
OVLN 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  
PTYT 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 00  
FRL 10  
FRL 20  
FRL 30  
FRL 40  
FRL 50  
FRL 60  
FRL 70  
OHQ NO  
OHQT00  
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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