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 (dialed 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 (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 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, 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
  - Media gateway
- Two Nortel digital stations 2616
- One analog station

Miscellaneous
- 2 – Fax Machines: HP Office Jet 5610xi

Software Requirements
- Cisco IOS Software releases: c3825adventureprisek9_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

Find and List Regions

1 matching record(s) for Region Name begins with ""

Find Regions where Region Name begins with [ ]
and show [20] items per page
To list all items, click Find without entering any search text.

Matching record(s) 1 to 1 of 1

Region

Default

Delete Selected

First Previous Next Last

Page 1 of 1
**Figure 3.** Default Region Detail (Cisco Unified Communications Manager Regional codec = G.711)

![Cisco CallManager Administration](image)

### Region Configuration

**Region:** Default  
**Status:** Ready

<table>
<thead>
<tr>
<th>Region Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Region Name*</td>
</tr>
</tbody>
</table>

### Call Information

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

<table>
<thead>
<tr>
<th>Region</th>
<th>Audio Codec</th>
<th>Video Call Bandwidth</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default (Within this Region)</td>
<td>G.711</td>
<td>None 384 kbps</td>
</tr>
</tbody>
</table>

*indicates required item
Figure 4. Region configuration. Ri_region set for G.729 Codec.
Figure 5. Gateway Configuration

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

Status: Ready
Update Delete Reset Gateway

Device Information

Device Name: 172.20.192.102
Description: gateway ip2ip
Device Poo: 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

### Multilevel Precendence and Preemption (MLPP) Information

| MLPP Domain (e.g., "0000FF") | Not available on this device |
| MLPP Indication | Not available on this device |
| MLPP Preemption | |

### Call Routing Information

#### Inbound Calls

- **Significant Digits**: All
- **Calling Search Space**: <None>
- **AAR Calling Search Space**: <None>
- **Prefix DN**
- [ ] Redirecting Number IE Delivery - Inbound
- [ ] Enable Inbound FastStart

#### Outbound Calls

- **Calling Party Selection**: Originator
- **Calling Party Presentation**: Default
- **Called party IE number type unknown**
- **Called party IE number type unknown**
- **Called Numbering Plan**: Cisco CallManager
- **Calling Numbering Plan**: Cisco CallManager
- **Caller ID DN**
- [ ] Display IE Delivery
- [ ] Redirecting Number IE Delivery - Outbound
- [ ] Enable Outbound FastStart
- **Codec For Outbound FastStart**: G729
Figure 6. Media Termination Point list

Find and List Media Termination Point

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

Find Media Termination Points where Name begins with

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.

<table>
<thead>
<tr>
<th>Media Termination Point</th>
<th>Description</th>
<th>Device Pool</th>
<th>Status</th>
<th>IP Address</th>
<th>Copy</th>
</tr>
</thead>
<tbody>
<tr>
<td>MTP_CM-GUANATOS</td>
<td>MTP_CM-GUANATOS</td>
<td>Default: CM-GUANATOS</td>
<td>172.20.8.254</td>
<td></td>
<td></td>
</tr>
<tr>
<td>MTP0013C4037300</td>
<td>MTP0013C4037300</td>
<td>Default: CM-GUANATOS</td>
<td>172.20.15.199</td>
<td></td>
<td></td>
</tr>
<tr>
<td>MTP0015F90D1590</td>
<td>MTP0015F90D1590</td>
<td>Default: CM-GUANATOS</td>
<td>172.20.192.102</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

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.
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.
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.
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:

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
 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
!
!
secc local GigabitEthernet0/0 2
secc ccm 172.20.8.254 identifier 1 version 4.1
secc
!
secc ccm group 1
  associate ccm 1 priority 1
  associate profile 1 register MTP0015f90d1590
!
dspfarm profile 1 mtp
  codec g729r8
  codec pass-through
  maximum sessions software 10
  associate application SCCP
!
!
dial-peer voice 7000 voip
  description dial peer digital toward CCM4.2
  destination-pattern 700[2-8]
  voice-class codec 1
  session target ipv4:172.20.8.254
  session transport tcp
  dtmf-relay h245-alphanumeric
  fax-relay ecm disable
  no vad
!
!
dial-peer voice 7200 pots
  destination-pattern 7055
  port 0/2/0
  forward-digits 0
!
!
dial-peer voice 22 voip
  description dial peer toward Nortel CS1000
  max-conn 2
  destination-pattern 2...

Notes: This is the H.323 signaling dial-peer.
Notes: This is the SIP signaling dial-peer.
voice-class codec 1
session protocol sipv2
session target ipv4:172.20.216.100
session transport tcp
dtmf-relay h245-alphanumeric
Codec G711ulaw 1

!  
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#

1 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
dsf

ip cef
!
!
multilink bundle-name authenticated
!
isdnswitch-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 SETH-LANSETHSW-LAUNCHSSINTF-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
isdnswitch-type primary-net5
isdnsend-alternate-network
isdnsend-alerting
isdnsend-alerting
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
!
!
secp local GigabitEthernet0/0
secp ccm 172.20.8.254 identifier 1 version 4.1
secp
!
secp ccm group 1
  associate ccm 1 priority 1
  associate profile 1 register mtp001193685140
!
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
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

Zone Basic Property and Bandwidth Management

<table>
<thead>
<tr>
<th>Input Description</th>
<th>Input Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Zone Number (ZONE):</td>
<td></td>
</tr>
<tr>
<td>Intrazone Bandwidth (INTRA BW):</td>
<td>10000</td>
</tr>
<tr>
<td>Intrazone Strategy (INTRA SIGY):</td>
<td>Best Quality (BO)</td>
</tr>
<tr>
<td>Interzone Bandwidth (INTER BW):</td>
<td>10000</td>
</tr>
<tr>
<td>Interzone Strategy (INTER SIGY):</td>
<td>Best Quality (BO)</td>
</tr>
<tr>
<td>Resource Type (RES_TYPE):</td>
<td>Shared (SHARED)</td>
</tr>
<tr>
<td>Branch Office Support (ZBRN):</td>
<td></td>
</tr>
<tr>
<td>Description (ZDES):</td>
<td></td>
</tr>
</tbody>
</table>

Submit  Refresh  Delete  Cancel
Figure 14. IP Telephony Configuration

IP Telephony Configuration

- Node Summary
- Personal Directories Server Configuration
- Personal Directories User Profile
- Quality Of Service Thresholds (QoS)
- SNMP Configuration
- Network Address Translation (NAT)
Figure 15. Configure a new IP Telephony Node summary
Figure 16. Configure the Node section

![Node Configuration Interface]

**Edit**

<table>
<thead>
<tr>
<th>Node</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Node ID</td>
<td>101</td>
</tr>
<tr>
<td>Voice LAN (TAN) Node IP address</td>
<td>172.20.215.100</td>
</tr>
<tr>
<td>Management LAN (ELAN) gateway IP address</td>
<td>172.20.218.1</td>
</tr>
<tr>
<td>Management LAN (ELAN) subnet mask</td>
<td>255.255.255.0</td>
</tr>
<tr>
<td>Voice LAN (ELAN) subnet mask</td>
<td>255.255.255.0</td>
</tr>
</tbody>
</table>

- **SNMP**
- **VGM and IP phone codec profile**
- **QoS**
- **LAN configuration**
- **SNTP**
- **H323 GW Settings**
Figure 17. Configure the VGW and IP phone codec profile section

<table>
<thead>
<tr>
<th>Codec</th>
<th>Select</th>
</tr>
</thead>
<tbody>
<tr>
<td>G.711</td>
<td></td>
</tr>
<tr>
<td>G.729A</td>
<td></td>
</tr>
<tr>
<td>G.723.1</td>
<td></td>
</tr>
<tr>
<td>T30 FAX</td>
<td></td>
</tr>
<tr>
<td>QoS</td>
<td></td>
</tr>
<tr>
<td>LAN config</td>
<td></td>
</tr>
<tr>
<td>SNTP</td>
<td></td>
</tr>
</tbody>
</table>

 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
  - Range: 0 to 300
- FAX playout nominal delay: 100
  - Range: 10 to 32800
- FAX no activity timeout: 20
- FAX packet size: 30

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Figure 18. CODEC profile selection

Figure 19. Configure the QoS section
### LAN Configuration

#### 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)

### SNTP

#### 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
Configure the SIP GW Settings section

Figure 21.
Figure 22. SIP CD Services

- System Status
  - Call Server
  - IP Telephony
- Configuration
  - Call Server
  - IP Telephony
- Network Numbering Plan
- Software Upgrade
- Patching
- System Utility
- Administration
- Support
- Tools
- Logout

SIP CD Services

- Service Enabled
  - Service DN used for making VTIRK call from

Nortel Networks

Converged Telephone Call Forward DN
- User Info. field for Invite message on the Converged Desktop MO Set
  - sip:convergeddesktop@pbxlab.org:nortelconverged=continueforcall
- User Info. field for Invite message on the Converged Desktop MV Set
  - sip:convergeddesktop@pbxlab.org:nortelconverged=conditionalforcall
- User Info. field in the notify message for Converged Desktop
  - sip:convergeddesktop@pbxlab.org

- RAI route for Announce

Wait time before a caller is sent to RAN Queue
- 1

Timeout for Ringing indication of the CD set
- 10

Timeout for CD Server
- 5

Timeout for call answered by other than CD phone set
- 2
Figure 23. Configure the Card section for the MC-32 VGMC card section

<table>
<thead>
<tr>
<th>Role</th>
<th>Follower</th>
</tr>
</thead>
<tbody>
<tr>
<td>Management LAN (ELAN) IP address</td>
<td>172.20.218.102</td>
</tr>
<tr>
<td>Management LAN (ELAN) MAC address</td>
<td>00:11:F9:E4:D0:11</td>
</tr>
<tr>
<td>Voice LAN (TLAN) IP address</td>
<td>172.20.216.102</td>
</tr>
<tr>
<td>Voice LAN (TLAN) gateway IP address</td>
<td>172.20.216.1</td>
</tr>
<tr>
<td>Hostname</td>
<td>MG_Node101_0</td>
</tr>
<tr>
<td>Card TN</td>
<td>3</td>
</tr>
<tr>
<td>Card processor type</td>
<td>Succession Media Card</td>
</tr>
<tr>
<td>H323 ID</td>
<td>MG_Node101</td>
</tr>
<tr>
<td>Enable set TPS</td>
<td>☑</td>
</tr>
<tr>
<td>System name</td>
<td>MG_Node_101</td>
</tr>
<tr>
<td>System location</td>
<td>Dewey Lab</td>
</tr>
<tr>
<td>System contact</td>
<td>Fred McClintic</td>
</tr>
</tbody>
</table>
Figure 24. Configure the Signaling Server section

<table>
<thead>
<tr>
<th>Role</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Management LAN (ELAN) IP address</td>
<td>172.20.218.103</td>
</tr>
<tr>
<td>Management LAN (ELAN) MAC address</td>
<td>00:02:B3:17:3a:36</td>
</tr>
<tr>
<td>Voice LAN (ELAN) IP address</td>
<td>172.20.216.103</td>
</tr>
<tr>
<td>Voice LAN (ELAN) gateway IP address</td>
<td>172.20.216.1</td>
</tr>
<tr>
<td>Hostname</td>
<td>SS_Node101_Ldr</td>
</tr>
<tr>
<td>H323 ID</td>
<td>Gateway_Node101</td>
</tr>
<tr>
<td>Enable 101TSP</td>
<td>✓</td>
</tr>
<tr>
<td>Enable virtual trunk TSP</td>
<td>H.323 and SIP</td>
</tr>
<tr>
<td>Enable SIP Proxy/Redirect Server</td>
<td>✓</td>
</tr>
<tr>
<td>SIP Transport Protocol</td>
<td>TCP</td>
</tr>
<tr>
<td>Local SIP Port</td>
<td>5000</td>
</tr>
<tr>
<td>SIP Domain name</td>
<td>birch.com</td>
</tr>
<tr>
<td>SIP Gateway Endpoint Name</td>
<td>Gateway_Node101</td>
</tr>
<tr>
<td>SIP Gateway Authentication Password</td>
<td>****</td>
</tr>
<tr>
<td>Enable H323 Gatekeeper</td>
<td>✓</td>
</tr>
<tr>
<td>Network Routing Service Role</td>
<td>Primary</td>
</tr>
<tr>
<td>System name</td>
<td>SS_Node101_Ldr</td>
</tr>
</tbody>
</table>
NRS (Network Routing Server):

Figure 25. NRS Overview
Figure 26. Configure the System Wide Settings

<table>
<thead>
<tr>
<th>System Wide Settings</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>DB sync interval for alternate [Hours]</td>
<td>24</td>
</tr>
<tr>
<td>SIP registration time to live timer [Seconds]</td>
<td>30</td>
</tr>
<tr>
<td>H.323 gatekeeper registration time to live timer [Seconds]</td>
<td>90</td>
</tr>
<tr>
<td>H.323 alias name</td>
<td>HO23NRS101</td>
</tr>
<tr>
<td>Alternate NRS server is permanent</td>
<td></td>
</tr>
<tr>
<td>Auto backup time [HH:MM]</td>
<td>23:59</td>
</tr>
<tr>
<td>Auto backup to FTP site enabled</td>
<td></td>
</tr>
<tr>
<td>Auto backup FTP site IP address</td>
<td></td>
</tr>
<tr>
<td>Auto backup FTP site path</td>
<td></td>
</tr>
<tr>
<td>Auto backup FTP username</td>
<td></td>
</tr>
<tr>
<td>Auto backup FTP password</td>
<td></td>
</tr>
</tbody>
</table>

Save
Figure 27. Configure the NRS Server Settings

<table>
<thead>
<tr>
<th>NRS Settings</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Host name</td>
<td>SS_Node101_Lot</td>
</tr>
<tr>
<td>Primary IP (T1/0)</td>
<td>172.30.216.103</td>
</tr>
<tr>
<td>Alternate IP (T1/0)</td>
<td>172.30.217.103</td>
</tr>
<tr>
<td>Control priority</td>
<td>40</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>H.323 Gatekeeper Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>Localice request (LRQ) response timeout (Seconds)</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>SIP Server Settings</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Node</td>
<td>Redirect</td>
</tr>
<tr>
<td>UDP transport enabled</td>
<td>✔</td>
</tr>
<tr>
<td>UDP port</td>
<td>5000</td>
</tr>
<tr>
<td>UDP maximum transmission unit (MTU)</td>
<td>1500</td>
</tr>
<tr>
<td>TCP transport enabled</td>
<td>✔</td>
</tr>
</tbody>
</table>
Figure 28. SIP Server Setting Cont'd
Configure a Service Domain

Network Routing Service

Location: Configuration > Service Domains > View Service Domain Property

View Service Domain Property

- Domain name: birch.com
- Domain description: service domain

* Mandatory field indicator
Figure 30. Configure a L1 Domain (UDP)
**Figure 31.** Configure a L0 Domain (CDP)

The image shows a network routing service interface with a section titled "View L0 Domain Property (birch.com/meccomm.com)". The interface includes fields for domain name, domain description, special number label, unqualified number label, endpoint authentication enabled, authentication password, E.164 country code, E.164 area code, international dialing access code, L1 domain dialing access code, and local dialing access code.
Configure a SIP Gateway

Figure 32.
Figure 33. Configure the Routing Entries

![Network Routing Service Screen](image)

### Routing Entries

<table>
<thead>
<tr>
<th>#</th>
<th>DN Prefix</th>
<th>DN Type</th>
<th>Route Cost</th>
<th>SIP URI Phone Context</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>1</td>
<td>Level1 regional</td>
<td>1</td>
<td>CDP.mccomm.com</td>
</tr>
</tbody>
</table>
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
2. LD 97 – Configure the Super-loop for the Virtual Trunks
3. LD 14 – Configure the SIP Virtual Trunks to the Signaling Server
4. LD 14 – Configure the Virtual Gateway Trunks
5. LD 16 – Configure the SIP route
6. LD 86 – Configure the Route List Block for the Virtual Trunk route
7. LD 87 – Configure CDP steering codes
8. Configure Digital Stations (Phones)

Signaling Server Setup Using the Nortel Element Manager:

9. Configure the Zones
10. Configure a new IP Telephony Node summary
11. Configure the Node section
12. Configure the VGW and IP phone codec profile section
13. Configure the Quality of Service (QoS) section
14. Configure LAN Configuration section
15. Configure the SIP GW Setting section
16. Configure the Card section for the MC-32 VGMC card section
17. Configure the Signaling Server section
18. Configure the NRS Server Settings
19. Configure a Service Domain
20. Configure a L0 Domain (CDP)
21. Configure a L1 Domain (UDP)
22. Configure the System Wide Settings

NRS (Network Routing Server):

23. Configure the Routing Entries
24. Configure the Routing Entries

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
```
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 00=> 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
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

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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
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 001 000 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 RRMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
  POD DSX VMD CMSD SLKD CSD SWD LND CND
  CFTA SFD MRD DDV CNTA CDCA MSID DAPA BFED RCBD
  ICDD CDMD LLCN MCTD CLBD AUTU
  GPUD DPUD DNDA CFXARHDLTD ASCD
  CPTA ABDD CFHD FICD NAI BUZZ AGRD MOAD AHD
  DDGA NAIR
  DRDD EXR0

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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
<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definitions</th>
</tr>
</thead>
<tbody>
<tr>
<td>ANF-PR</td>
<td>Additional Network Feature Path Replacement</td>
</tr>
<tr>
<td>AOC</td>
<td>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.</td>
</tr>
<tr>
<td>CUCM</td>
<td>Cisco Unified Communication Manager</td>
</tr>
<tr>
<td>CCBS</td>
<td>Call Completion to Busy Subscriber</td>
</tr>
<tr>
<td>CCNR</td>
<td>Call Completion on No Reply</td>
</tr>
<tr>
<td>CFB</td>
<td>Call Forwarding on Busy</td>
</tr>
<tr>
<td>CFNR</td>
<td>Call Forwarding No Reply</td>
</tr>
<tr>
<td>CFU</td>
<td>Call Forwarding Unconditional</td>
</tr>
<tr>
<td>CLIP</td>
<td>Calling Line (Number) Identification Presentation</td>
</tr>
<tr>
<td>CLIR</td>
<td>Calling Line (Number) Identification Restriction</td>
</tr>
<tr>
<td>CMM</td>
<td>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</td>
</tr>
<tr>
<td>CNIP</td>
<td>Calling Name Identification Presentation</td>
</tr>
<tr>
<td>CNIR</td>
<td>Calling Name Identification Restriction</td>
</tr>
<tr>
<td>COLP</td>
<td>Connected Line (Number) Identification Presentation</td>
</tr>
<tr>
<td>COLR</td>
<td>Connected Line (Number) Identification Restriction</td>
</tr>
<tr>
<td>CONP</td>
<td>Connected Name Identification Presentation</td>
</tr>
<tr>
<td>CONR</td>
<td>Connected Name Identification Restriction</td>
</tr>
<tr>
<td>CT</td>
<td>Call Transfer</td>
</tr>
<tr>
<td>MWI</td>
<td>Message Waiting Indicator</td>
</tr>
</tbody>
</table>
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<thead>
<tr>
<th>Corporate Headquarters</th>
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<th>Americas Headquarters</th>
<th>Asia Pacific Headquarters</th>
</tr>
</thead>
<tbody>
<tr>
<td>170 West Tasman Drive</td>
<td>Haarlerberg park</td>
<td>170 West Tasman Drive</td>
<td>Capital Tower</td>
</tr>
<tr>
<td>San Jose, CA 95134-1706</td>
<td>Haarlerbergweg 13-19</td>
<td>San Jose, CA 95134-1706</td>
<td>168 Robinson Road</td>
</tr>
<tr>
<td>USA</td>
<td>1101 CH Amsterdam</td>
<td>USA</td>
<td>#22-01 to #29-01</td>
</tr>
<tr>
<td><a href="http://www.cisco.com">www.cisco.com</a></td>
<td>The Netherlands</td>
<td><a href="http://www.cisco.com">www.cisco.com</a></td>
<td>Singapore 068912</td>
</tr>
<tr>
<td>Tel: 408 526-4000</td>
<td>www-europe.cisco.com</td>
<td>Tel: 408 526-7660</td>
<td><a href="http://www.cisco.com">www.cisco.com</a></td>
</tr>
<tr>
<td>Fax: 408 526-4100</td>
<td></td>
<td>Fax: 408 527-0883</td>
<td>Tel: +65 317 7777</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>Fax: +65 317 7799</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Cisco Systems has more than 200 offices in the following countries and regions. Addresses, phone numbers, and fax numbers are listed on the Cisco Web site at <a href="http://www.cisco.com/go/offices">www.cisco.com/go/offices</a>.</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

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