



Nortel Communication Server 1000M Release 4.0 to Cisco IOS Voice Gateway Using SIP with E1 NET5

September 02, 2008 Initial Version

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Introduction

- This application note describes basic call interoperability and the documented steps and configurations necessary for SIP integration between Nortel Communication Server (CS) 1000M version 4.0 and a Cisco IOS voice gateway providing E1 NET5 PSTN connectivity.
- The SIP protocol is used between the Cisco IOS voice gateway and Nortel CS 1000M version 4.0. The connection between the Cisco IOS gateway and PSTN uses E1 with a switch-type NET5 protocol.
- Features tested include basic call; call transfer supervised; call transfer blind; call forward (all, busy, and no answer); three-way conference; DTMF tones; and caller ID functionality between Nortel CS1000M version 4.0 users and PSTN users.
- The Cisco IOS voice gateway offers the advantage of providing connectivity between Nortel CS 1000M version 4.0 and PSTN by offering SIP to ISDN inter-working functionality.
- The network topology diagram (Figure 1) shows the test setup for end-to-end interoperability with the Cisco IOS voice gateway connected to the Nortel CS 1000M version 4.0 and connected to the PSTN via E1 NET5 ISDN.
- This application note uses the C3825 IOS voice gateway. However, other Cisco voice gateways are also an option to use since the voice gateway implementation does not depend on the platform. Below is a list of Cisco platforms capable of voice gateway functionality. Be careful when you select a voice gateway platform; consider the capacity and capability required for the intended deployment.

[Cisco 1861 Integrated Services Router](#)

[Cisco IAD2400 Series Integrated Access Device](#)

[Cisco 2800 Series Integrated Services Routers](#)

[Cisco 3700 Series Multiservice Access Routers](#)

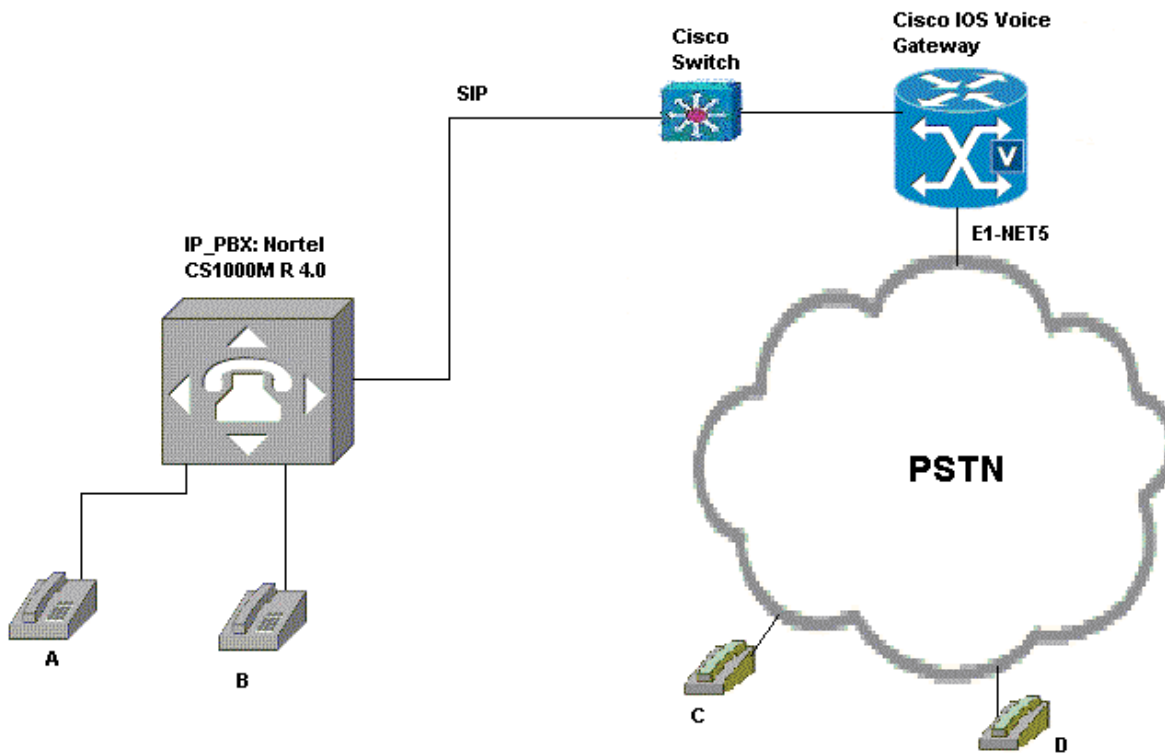
[Cisco 3800 Series Integrated Services Routers](#)

[Cisco AS5350XM Universal Gateway](#)

[Cisco AS5400XM Universal Gateway](#)

Network Topology

Figure 1. Basic Call Setup



Limitations

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

- Calling name and number are not updated consistently.
- Calling number cannot be restricted on network calls from PSTN. For example, restrict calling number and name on PSTN phone C. Place a call from phone C to PBX phone A. Calling number (phone C's number) is displayed on phone A.
- On network/external conference calls, the third party on the conference call does not have a talk path when a conference call is accomplished by a network/external call followed by a network/external call. For example, phone C calls phone A, phone A calls phone D. When phone D answers, activate "conference" on phone A; phone D can hear but has no talk path to C and A phones, while C phone and A phone can talk to all conference members.



System Components

Hardware Requirements

The following hardware is required:

Cisco equipment

- Cisco 3825 Gateway
- DSP Mod NM-HDV2-1T1E1
- Cisco Catalyst 3550 Power Ethernet switch

Nortel equipments

- Nortel CS1000M

Software Requirements

The following software is required:

- IOS Software releases: c3825-ipvoice ivs-mz.12.4 (15)xz.bin
- PBX Software: Nortel CS 1000M version 4.0

Features

This section lists supported and unsupported features.

Features Supported

- Calling Name Identification Restriction (see Limitations on page 3)
- Calling Number Identification Restriction (see Limitations)
- Codec G.711 Ulaw
- Codec G.729
- Codec G.723
- Calling name (see Limitations)
- Calling number (see Limitations)
- Call transfer blind
- Call transfer supervised
- Call conference (see Limitations)
- Call on hold
- Call forward no reply
- Call forward all
- Call forward busy
- Digit translation—The voice gateway can modify the digits of the called four-digit number sent by Nortel CS 1000M version 4.0 and PSTN.



Features Not Supported

- DTMF tones using RFC2833
- DTMF tones using inband signaling



Configuration

This section contains configuration menus and commands and describes configuration sequences and tasks.

Configuring Nortel CS 1000M Version 4.0

Nortel Communication Server 1000 (CS1000) Configuration

Call Server Setup:

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

```
>ld 22
PT2000
REQ prt
TYPE adan dch 0
DCH 0 IS UNDEFINED
REQ prt
TYPE adan
ADAN HIST
  SIZE 5000
  USER MTC SCH BUG
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
  OVLS NO
```

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

```
>ld 97
SCSYS000
MEM AVAIL: (U/P): 2717255  USED U P: 344386 51318  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 1 ----
036 STD LEFT 07 0 1 ----
040 STD LEFT 08 0 1 ----
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 ----
ARDS 61 - 64 81 - 84
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
```

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

```
>ld 20
PT0000
REQ: prt
TYPE: tnb
TN 000 0 00 00
CDEN
CUST
DATE
PAGE
DES

TN 000 0 00 00
TYPE DTR
CDEN 8D
DATE NO DATE

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

LD 11 – Configure for the Nortel 2616 phones

```
>ld 11
SL1000
MEM AVAIL: (U/P): 2717255  USED U P: 344386 51318  TOT: 3112959
DISK RECS AVAIL: 1152
DIGITAL TELEPHONES AVAIL: 0  USED: 8  TOT: 8
IP USERS AVAIL: 1  USED: 7  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: 2303  USED: 197  TOT: 2500
DATA PORTS AVAIL: 2500  USED: 0  TOT: 2500
```

REQ: prt
TYPE: 2616

TN 1 0
DATE
PAGE
DES

DES CS102
TN 001 0 00 00
TYPE 2616
CDEN 8D
CUST 0
AOM 0
FDN 5216
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
DRDD EXR0
USRD ULAD RTDD RBDD RBHD PGND FLXD FTTC DNDY DNO3 MCBN CDMR

CPND_LANG ENG
RCO 0
EFD 5216
HUNT 5216
EHT 5216
LHK 0
PLEV 02
CSDN
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
DNDR 0
KEY 00 SCR 5213 0 MARP
CPND
NAME ATHENA
XPLN 13
DISPLAY_FMT FIRST, LAST
01 PRK
02 MCR 5215 0 MARP
CPND
NAME ATHENA_5215
XPLN 13
DISPLAY_FMT FIRST, LAST
03 CFW 4 2324
04 AO6
05 TRN
06
07
08
09
10
11
12 MIK
13 MIK
14 MCK
15 RGA
DATE 17 JUN 2008

NACT
REQ: prt
TYPE: 2616
TN 1 2
DATE



PAGE
DES

DES CS102
TN 001 0 00 02
TYPE 2616
CDEN 8D
CUST 0
AOM 0
FDN 2321
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 CNID 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 FLXD FTTU DNDY DNO3 MCBN CDMR
CPND_LANG ENG
RCO 0
EFD 2321
HUNT 2321
EHT 2321
LHK 0
PLEV 02
CSDN
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
DNDR 0
KEY 00 SCR 5216 0 MARP
CPND
NAME ATHENA_5216
XPLN 13
DISPLAY_FMT FIRST, LAST
01
02
03 CFW 4 3502
04 AO6
05 TRN



06
07
08
09
10
11
12
13 MIK
14 MCK
15 TRN

DATE 17 JUN 2008

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 102
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 710
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
OHQ NO



OHQT 00
CBQ NO
AUTH NO
TTBL 0
ATAN NO
OHTD NO
PLEV 2
ALRM NO
ART 0
SGRP 0
AACR NO

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

>ld 86
ESN000

MEM AVAIL: (U/P): 2717255 USED U P: 344386 51318 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

LD 87 – Configure CDP steering codes



>ld 87
ESN000

MEM AVAIL: (U/P): 2717255 USED U P: 344386 51318 TOT: 3112959
DISK RECS AVAIL: 1152
REQ prt
CUST 0
FEAT cdp
TYPE dsc
DSC 232
DSC 232
FLEN 0
DSP LSC
RLI 10
NPA
NXX
OVL000

Signaling Server Setup
Configure the Zones

Site: 172.20.219.101 > 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"/> |



Configure a New IP Telephony Node Summary

Site: 172.20.219.101 > Configuration > IP Telephony Configuration >

Node Summary

New Node to Add

Import Node Files

Node: 102 Node IP: 172.20.217.100 Edit Transfer / Status Delete

| | |
|-----------------------------|----------------|
| Voice LAN (TLAN) IP address | TN |
| Signaling Server | 172.20.217.103 |
| Pentium Card | |
| Succession Media Card | 172.20.217.102 |
| | 3 0 |
| | VGW Channels |

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

Configure the Node Section

Site: 172.20.219.101 > Configuration > IP Telephony Configuration > Node Summary > IP Telephony: Node ID 102 >

Edit

Save and Transfer Cancel

Node

| | |
|--|----------------|
| Node ID | 102 |
| Voice LAN (TLAN) Node IP address | 172.20.217.100 |
| Management LAN (ELAN) gateway IP address | 172.20.219.1 |
| Management LAN (ELAN) subnet mask | 255.255.255.0 |
| Voice LAN (TLAN) subnet mask | 255.255.255.0 |

- SNMP Add
- VGW and IP phone codec profile
- QoS
- LAN configuration
- SNTP
- H323 GW Settings
- Firmware
- SIP GW Settings
- SIP URI Map
- 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



Configure the VGW and IP Phone Codec Profile Section¹

VGW and IP phone codec profile

| | | |
|--|-------------------------------------|---|
| Enable Echo canceller | <input checked="" type="checkbox"/> | |
| 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 | <input checked="" type="checkbox"/> | |
| Enable V.21 FAX tone detection | <input checked="" type="checkbox"/> | |
| 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 | |
| Codec | G711 | Select <input checked="" type="checkbox"/> |
| Codec Name | G711 | |
| Voice payload size (ms/frame) | 20 | |
| Voice playout (jitter buffer) nominal delay | 40 | |
| <small>Modifications may cause changes to dependent settings</small> | | |
| Voice playout (jitter buffer) maximum delay | 80 | |
| <small>Modifications may cause changes to dependent settings</small> | | |
| VAD | <input type="checkbox"/> | |
| Codec | G729A | Select <input type="checkbox"/> |
| Codec | G723.1 | Select <input type="checkbox"/> |
| Codec | T38 FAX | Select <input checked="" type="checkbox"/> |

Codec G711 **Select**

| | |
|--|--------------------------|
| Codec Name | G711 |
| Voice payload size (ms/frame) | 20 |
| Voice playout (jitter buffer) nominal delay | 40 |
| <small>Modifications may cause changes to dependent settings</small> | |
| Voice playout (jitter buffer) maximum delay | 80 |
| <small>Modifications may cause changes to dependent settings</small> | |
| VAD | <input type="checkbox"/> |
| Codec | G729A |
| Codec Name | G729A |
| Voice payload size (ms/frame) | 20 |
| Voice playout (jitter buffer) nominal delay | 40 |
| <small>Modifications may cause changes to dependent settings</small> | |
| Voice playout (jitter buffer) maximum delay | 80 |
| <small>Modifications may cause changes to dependent settings</small> | |
| VAD | <input type="checkbox"/> |
| Codec | G723.1 |
| Codec | T38 FAX |
| Codec Name | T38 FAX |

¹ Change audio codec to match the ISR media gateway when testing codec.



Configure the QoS Section

| | | | |
|--|--|---------------------------------------|--|
| <ul style="list-style-type: none"> System Status Call Server IP Telephony Configuration <ul style="list-style-type: none"> Call Server IP Telephony Network Numbering Plan Software Upgrade Patching System Utility Administration Support Tools Logout | VAD | <input type="checkbox"/> | |
| | Codec | T38 FAX | Select <input checked="" type="checkbox"/> |
| | Codec Name | T38 FAX | |
| | QoS | | |
| | Diffserv Codepoint(DSCP) Control packets | <input type="text" value="40"/> | Range: 0 to 63 |
| | Diffserv Codepoint(DSCP) Voice packets | <input type="text" value="46"/> | Range: 0 to 63 |
| | Enable 802.1Q support | <input type="checkbox"/> | |
| | 802.1Q Bits value (802.1p) | <input type="text" value="6"/> | Range: 0 to 7 |
| | LAN configuration | | |
| | SNTP | | |
| | H323 GW Settings | | |
| | Firmware | | |
| | SIP GW Settings | | |
| | SIP URI Map | | |
| | SIP CD Services | | |
| Cards | <input type="button" value="Add"/> | | |
| Signaling Servers | <input type="button" value="Add"/> | | |
| <input type="button" value="Save and Transfer"/> | | <input type="button" value="Cancel"/> | |

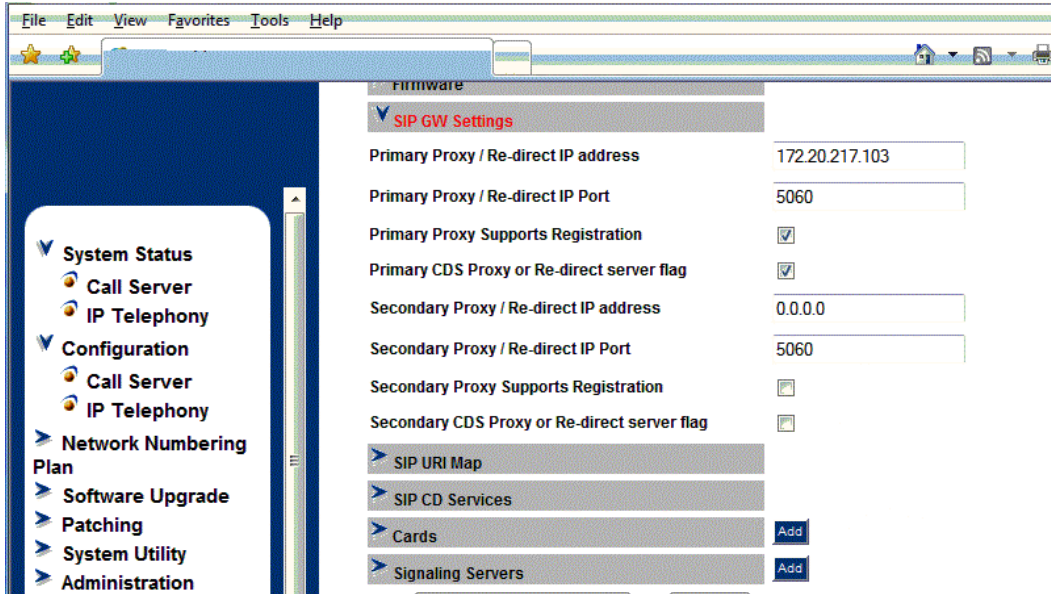
**Mandatory fields of current configuration*

Configure the LAN Configuration Section

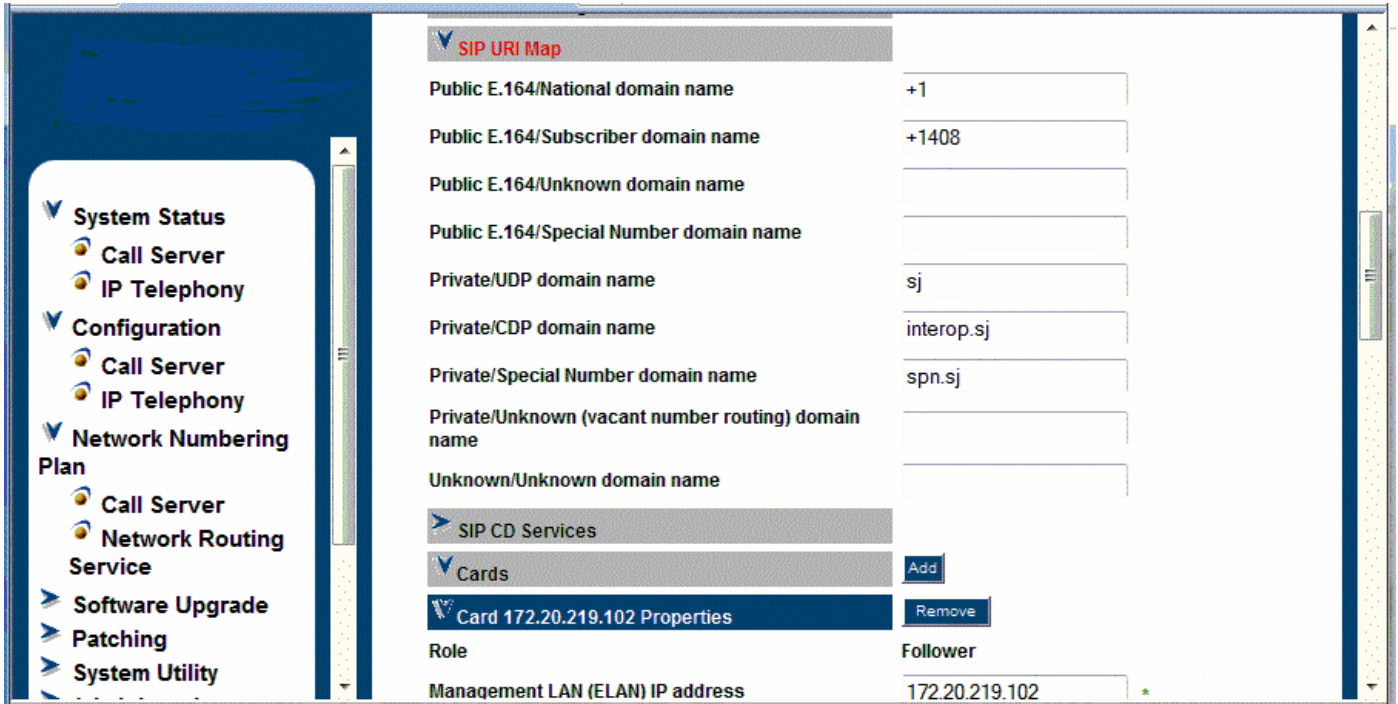
| | | |
|--|--|---|
| <ul style="list-style-type: none"> System Status Call Server IP Telephony Configuration <ul style="list-style-type: none"> Call Server IP Telephony Network Numbering Plan Software Upgrade Patching System Utility Administration Support Tools Logout | QoS | |
| | Diffserv Codepoint(DSCP) Control packets | <input type="text" value="40"/> Range: 0 to 63 |
| | Diffserv Codepoint(DSCP) Voice packets | <input type="text" value="46"/> Range: 0 to 63 |
| | Enable 802.1Q support | <input type="checkbox"/> |
| | 802.1Q Bits value (802.1p) | <input type="text" value="6"/> Range: 0 to 7 |
| | LAN configuration | |
| | Management LAN (ELAN) configuration | |
| | Call server IP address | <input type="text" value="172.20.219.101"/> |
| | Survivable Succession Media Gateway IP address | <input type="text" value="0.0.0.0"/> |
| | Signaling port | <input type="text" value="15000"/> Range: 1024 to 65535 |
| | Broadcast port | <input type="text" value="15001"/> Range: 1024 to 65535 |
| | Voice LAN (TLAN) configuration | |
| | Signaling port | <input type="text" value="5000"/> Range: 1024 to 65535 |
| | Voice port | <input type="text" value="5200"/> Range: 1024 to 65535 |
| | Routes | <input type="button" value="Add"/> |
| SNTP | | |
| H323 GW Settings | | |
| Firmware | | |
| SIP GW Settings | | |
| SIP URI Map | | |



Configure the SIP GW Setting Section



SIP URL Setting



The screenshot shows the configuration interface for SIP URI Map and Card 172.20.219.102 Properties. The left sidebar contains a navigation menu with the following items:

- System Status
- Call Server
- IP Telephony
- Configuration
 - Call Server
 - IP Telephony
- Network Numbering Plan
 - Call Server
 - Network Routing Service
- Software Upgrade
- Patching
- System Utility

The main content area is divided into two sections:

SIP URI Map

| | |
|---|------------|
| Public E.164/National domain name | +1 |
| Public E.164/Subscriber domain name | +1408 |
| Public E.164/Unknown domain name | |
| Public E.164/Special Number domain name | |
| Private/UDP domain name | sj |
| Private/CDP domain name | interop.sj |
| Private/Special Number domain name | spn.sj |
| Private/Unknown (vacant number routing) domain name | |
| Unknown/Unknown domain name | |

SIP CD Services

Cards [Add](#)

Card 172.20.219.102 Properties [Remove](#)

| | |
|----------------------------------|----------------|
| Role | Follower |
| Management LAN (ELAN) IP address | 172.20.219.102 |

Configure the Card Section for the MC-32 VGMC Card Section



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

Cards Add

Card 172.20.219.102 Properties Remove

Role **Follower**

Management LAN (ELAN) IP address *

Management LAN (ELAN) MAC address *

Voice LAN (TLAN) IP address *

Voice LAN (TLAN) gateway IP address

Hostname *

Card TN *

Card processor type ▾

H323 ID

Enable set TPS

System name

System location

System contact

Signaling Servers Add

Save and Transfer Cancel

Configure the Signaling Server Section



▼ System Status
● Call Server
● IP Telephony

▼ Configuration
● Call Server
● IP Telephony

▶ Network Numbering Plan
▶ Software Upgrade
▶ Patching
▶ System Utility
▶ Administration
▶ Support
● Tools
▶ Logout

▼ Signaling Server 172.20.219.103 Properties Remove

| | |
|-------------------------------------|--|
| Role | Leader |
| Management LAN (ELAN) IP address | <input type="text" value="172.20.219.103"/> * |
| Management LAN (ELAN) MAC address | <input type="text" value="00:02:b3:f7:33:76"/> * |
| Voice LAN (TLAN) IP address | <input type="text" value="172.20.217.103"/> * |
| Voice LAN (TLAN) gateway IP address | <input type="text" value="172.20.217.1"/> |
| Hostname | <input type="text" value="SS_Node102"/> * |
| H323 ID | <input type="text" value="SS_Node102"/> |
| Enable set TPS | <input checked="" type="checkbox"/> |
| Enable virtual trunk TPS | <input type="text" value="H.323 and SIP"/> |
| Enable SIP Proxy / Redirect Server | <input checked="" type="checkbox"/> |
| SIP Transport Protocol | <input type="text" value="TCP"/> |
| Local SIP Port | <input type="text" value="5060"/> |
| SIP Domain name | <input type="text" value="cisco.com"/> |
| SIP Gateway Endpoint Name | <input type="text" value="SS_Node102"/> |
| SIP Gateway Authentication Password | <input type="password" value=""/> |
| Enable H323 Gatekeeper | <input checked="" type="checkbox"/> |
| Network Routing Service Role | <input type="text" value="Primary"/> |
| System name | <input type="text" value="SS_Node102"/> |
| System location | <input type="text" value=""/> |
| System contact | <input type="text" value=""/> |



Network Routing Server Setup

Configure the NRS Server Settings

Home Configuration Tools Reports Administration Help Logout

Location: Home > NRS Server Settings >

NRS Overview

System Wide Settings

=> NRS Server Settings

NRS Settings

Host name *

Primary IP (TLAN) *

Alternate IP (TLAN) *

Control priority

H.323 Gatekeeper Settings

Location request (LRQ) response timeout [Seconds]

SIP Server Settings

Mode

UDP transport enabled

UDP port

UDP maximum transmission unit (MTU)

TCP transport enabled

TCP port

SIP Server Setting

Home Configuration Tools Reports Administration Help Logout

Location request (LRQ) response timeout [Seconds]

NRS Overview

System Wide Settings

=> NRS Server Settings

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]

Save



Configure a Service Domain

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

Location: Configuration > Service Domains > View Service Domain Property >

View Service Domain Property

Domain name *

Domain description

* Mandatory field indicator

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

Configure an L1 Domain (UDP)

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

Location: Configuration > L1 Domains (UDP) > View L1 Domain Property >

View L1 Domain Property (cisco.com)

Domain name *

Domain description

Endpoint authentication enabled

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

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



Configure an L0 Domain (CDP)

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

View L0 Domain Property (cisco.com / SanJose)

| | | |
|-----------------------|-----------------------------------|----------------|
| Service Domains | Domain name | pbx * |
| L1 Domains (UDP) | Domain description | PBX Interop |
| => L0 Domains (CDP) | Special number label | |
| Gateway Endpoints | Unqualified number label | |
| User Endpoints | Endpoint authentication enabled | Not configured |
| Routing Entries | Authentication password | |
| Default Routes | E.164 country code | 1 |
| Collaborative Servers | E.164 area code | 408 |
| | International dialing access code | 011 |
| | L1 domain dialing access code | |
| | National dialing access code | |
| | Local dialing access code | |
| | Special number 1 | |
| | Special number 2 | |

Configure an SIP Gateway

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

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

| | | |
|-----------------------|-----------------------------------|-------------------------|
| Service Domains | Endpoint name | ri19221 * |
| L1 Domains (UDP) | Endpoint description | r119221 |
| L0 Domains (CDP) | Tandem endpoint name | Look up |
| => Gateway Endpoints | Endpoint authentication enabled | Not configured |
| User Endpoints | Authentication password | |
| Routing Entries | E.164 country code | 1 |
| Default Routes | E.164 area code | 415 |
| Collaborative Servers | International dialing access code | 011 |
| | L1 domain dialing access code | 9 |
| | National dialing access code | 9 |
| | Local dialing access code | 9 |
| | Special number 1 | |
| | Special number 2 | |
| | Static endpoint address type | IP version 4 |



Home Configuration Tools Reports Administration **Active DB view** (set Standby DB view) Help | Logo

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

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
Static endpoint address
H.323 Support
SIP support
SIP transport
SIP port

Configure the Routing Entries

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

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

Location: Configuration > Routing Entries > View Routing Entry Property >

View Routing Entry Property (cisco.com / SanJose / pbx / Tala_CME1)

DN type
Default route
DN prefix
Route cost (1 -255) *

* Mandatory field indicator



Cisco 3825 configuration

```
CUBE21#sh ver
Cisco IOS Software, 3800 Software (C3825-IPVOICE_IVS-M), Version 12.4(15)XZ, REL
EASE SOFTWARE (fc2)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2008 by Cisco Systems, Inc.
Compiled Fri 11-Apr-08 21:10 by prod_rel_team
```

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

CUBE21 uptime is 9 weeks, 5 days, 17 hours, 39 minutes
System returned to ROM by reload at 22:56:36 UTC Wed Apr 23 2008
System image file is "flash:c3825-ipvoice_ivs-mz.124-15.XZ.bin"

Cisco 3825 (revision 1.0) with 224256K/37888K bytes of memory.
Processor board ID FTX0946A1BT
2 Gigabit Ethernet interfaces
11 Serial interfaces
1 Channelized E1/PRI port
2 Channelized T1/PRI ports
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

```
CUBE21#sh run
Building configuration...
```

```
Current configuration : 2225 bytes
!
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname CUBE21
!
boot-start-marker
boot-end-marker
!
logging message-counter syslog
logging buffered 10000000
!
no aaa new-model
no network-clock-participate slot 1
network-clock-participate slot 2
!
voice-card 0
no dspfarm
!
```



```
voice-card 1
 dspfarm
 !
voice-card 2
 dspfarm
 !
ip cef
 !
 !
 !
no ip domain lookup
multilink bundle-name authenticated
 !
isdn switch-type primary-net52
 !
 !
 !
voice service voip
 allow-connections sip to sip
 sip
 !
 !
voice translation-rule 13
 rule 1 /400/ /521\1/
 !
voice translation-rule 2
 rule 2 /65/ /23\1/
 !
 !
voice translation-profile pots
 translate called 1
 !
voice translation-profile voip
 translate called 2
 !
 !
 !
 !
archive
 log config
  hidekeys
 !
 !
controller T1 1/0/0
 framing esf
```

² Set ISDN switch types.

³ The voice gateway manipulates the called digits to match configured dial-peers and to route calls appropriately. For example, digit manipulation rule 1 of voice translation rule 1 instructs the IOS gateway that when it receives “400,” the IOS gateway is to strip 400 and add digit 521 as leading numbers to the remaining digits x (x in this case are either 3 or 6) and send them to the appropriate dial-peer.



```
linecode b8zs
!
controller T1 1/0/1
framing esf
linecode b8zs
!
controller E1 2/0
pri-group timeslots 1-10,16
!
!
!
!
interface GigabitEthernet0/0
ip address 172.20.192.21 255.255.255.0
duplex auto
speed auto
media-type rj45
!
interface GigabitEthernet0/1
no ip address
shutdown
duplex auto
speed auto
media-type rj45
!
interface Serial2/0/0:15
no ip address
encapsulation hdlc
isdn switch-type primary-net52
isdn protocol-emulate network
isdn incoming-voice voice
isdn supp-service name calling ie 40 cs04
no cdp enable
!
ip default-gateway 172.20.192.1
ip forward-protocol nd
ip route 0.0.0.0 0.0.0.0 172.20.192.1
!
!
ip http server
!
!
!
!
control-plane
!
!
!
voice-port 0/1/0
!
voice-port 0/1/1
```

⁴ This command enables calling name.



```
!  
voice-port 2/0/0:15  
!  
!  
!  
dial-peer voice 5300 voip5  
translation-profile incoming voip  
destination-pattern 52..  
session protocol sipv26  
session target ipv4:172.20.217.100  
session transport tcp  
.dtmf-relay rtp-nte7  
codec g711alaw8  
!  
dial-peer voice 2000 pots9  
translation-profile incoming pots  
destination-pattern 232.  
incoming called-number 5...  
direct-inward-dial  
port 2/0/0:15  
forward-digits all  
!  
dial-peer voice 2325 pots  
destination-pattern 2325  
port 0/1/1  
!  
!  
!  
gatekeeper  
shutdown  
!  
!  
line con 0  
password cisco  
logging synchronous  
login  
no activation-character  
line aux 0  
line vty 0 4  
password cisco  
login  
!  
scheduler allocate 20000 1000  
end
```

⁵ VOIP dial-peer toward PBX.

⁶ This configures the dial-peer IETF SIP.

⁷ Insert this command for DTMF RFC2833. Removing this command will do DTMF in-band.

⁸ Specify codec here when testing codec. Also change codec settings at the PBX end to match the specified codec at the IOS Media Gateway.

⁹ POTS dial-peer toward PSTN.



Acronyms

| Acronym | Definitions |
|---------|-------------------------------------|
| CDP | Cisco Discovery Protocol |
| DSP | digital signal processor |
| DTMF | dual-tone multi-frequency |
| IOS | internetwork Operating System |
| ISDN | integrated services digital network |
| PSTN | public switched telephone network |
| SIP | session initiation protocol |

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