

# Cisco Unified CallManager Release 4.1(3) - PBX Interoperability: Nortel Communication Server 1000M Release 4.0 Using SIP Trunk

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## Introduction

This is an application note for interoperability connectivity of Nortel Communication Server 1000 (formerly known as Succession 1000) PBX with Cisco Unified CallManager Release 4.1(3)SR2 using a SIP trunk.

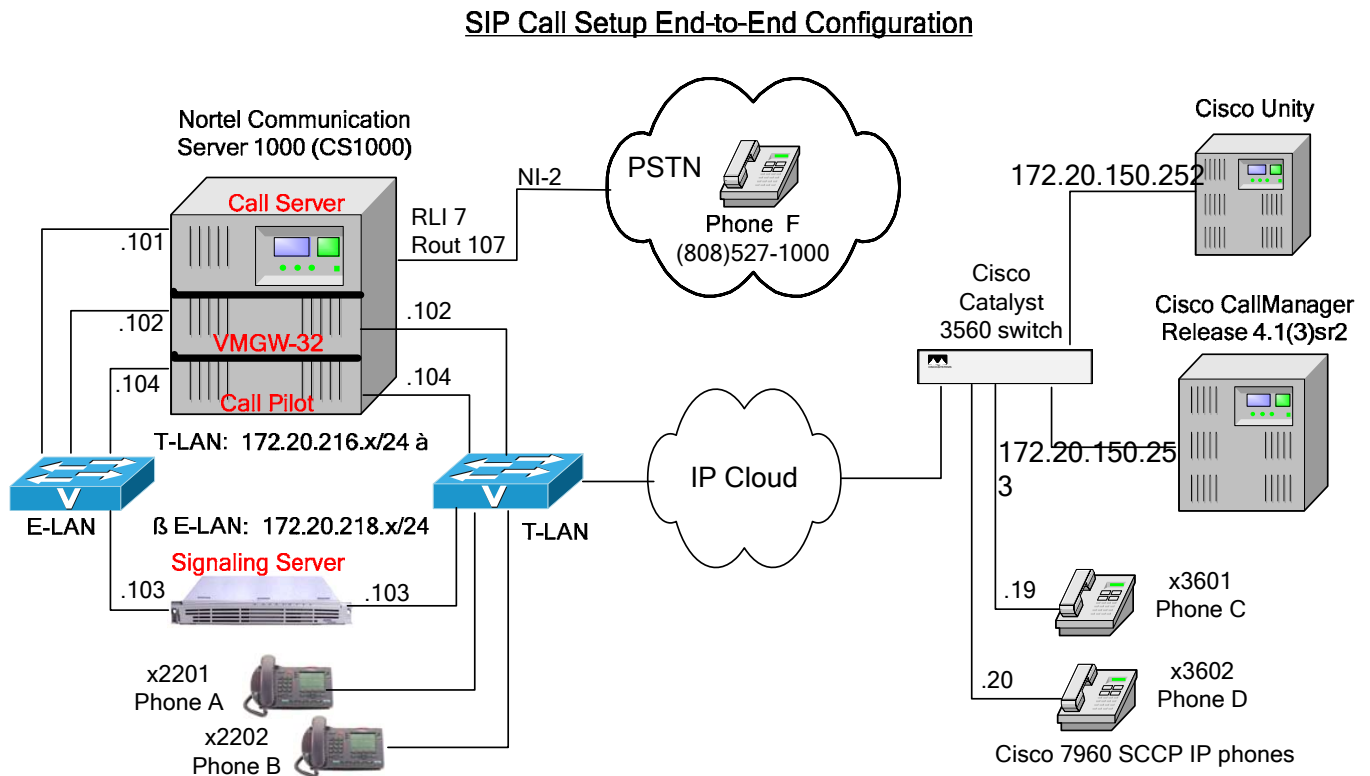
The network topology diagram (Figure 1) shows the test setup for end-to-end interoperability with the Nortel CS1000 PBX configured as a SIP trunk with the Cisco Unified CallManager 4.1(3)SR2 release.

On Cisco Unified CallManager, ensure that the SIP Trunk Configuration web page is defined for the Nortel PBX and make sure that the “Media Termination Point Required” box is checked.



## Network Topology

Figure 1. Network Topology



## Limitations

Cisco Unified CallManager and Nortel CS1K systems use different methods for passing the name and number information across the SIP trunk. Cisco Unified CallManager uses the Remote-Party-Id (RPID) field in the SIP header and Nortel uses the P-Asserted-Id field (PAI) in the SIP header. Because neither party understand the other's method, they use the name and number information in the SIP FROM: header instead.

For features such as CLIR, CNIR, COLR, and CONR, set the SIP FROM: header to be "Anonymous" and the privacy setting on the Remote-Party-Id or P-Asserted-Id field to "Restricted."

Cisco Unified CallManager supports Alerting/Connected Name and Number using the Remote-Party-Id tag in the SIP Header. Nortel supports the Connected Name and Number by using the P-Asserted-Id tag in the SIP Header. However, the Nortel PBX does not support Alerting name and number information (there is no P-Asserted-Id tag within the SIP 180/183 alerting message).

Call Transfer and Call Forward features work but the phone's name and number display update capability does not because the two systems use different methods of passing the name and number across the SIP trunk.

Call Completion (Callback) Feature is not supported on either system (Cisco Unified CallManager or Nortel CS1000M) using standard SIP protocol.

MWI ON/OFF messages do not work across the SIP Trunk connection between the two systems.

End-to-end DTMF relay signaling does not work between the two systems. Cisco Unified CallManager uses the RFC2833 method of passing the DTMF digits across the SIP trunk and Nortel uses SIP INFO method of passing the DTMF digits across SIP Trunk.



For the Cisco Unified CallManager SIP Trunk configuration, the “Media Termination Point Required” box must be checked in order for the two systems to communicate.

## System Components

### Hardware Requirements

Cisco Unified CallManager MCS server, Unity server, and Cisco 7960 and 7940 phones

Nortel Communication System 1000M (which includes Call Server, Signaling Server and Media Gateway) and Nortel’s i2004/i2002 IP phones

### Software Requirements

Cisco Unified CallManager Release 4.1(3)sr2

Nortel Succession 4.0 Release

## Features

Note: See the “Limitations” section.

CLIP-Calling Line (Number) Identification Presentation

CLIR-Calling Line (Number) Identification Restriction

CNIP-Calling Name Identification Presentation

CNIR-Calling Name Identification Restriction

Alerting Name

CT-Call Transfer by Join

CFU-Call Forwarding Unconditional

CFB-Call Forwarding Busy

CFB-Call Forwarding Busy

COLP-Connected Line (Number) Identification Presentation

COLR- Connected Line (Number) Identification Restriction

CONP-Connected Name Identification Presentation

CONR-Connected Name Identification Restriction

## Features Not Supported

MWI- Message Waiting Indication (lamp ON, lamp OFF) across the SIP Trunk

Call Completion (Callback, Automatic Callback)

Call Completion

End-to-End DTMF signaling



## Configuration

### Nortel Communication Server 1000 PBX Configuration Sequence and Tasks

#### 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 11 – Configure for the Virtual lines for the Nortel IP phone (i200x series)
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

#### 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

#### NRS (Network Routing Server)

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

#### Cisco Unified CallManager Setup

25. Create the Media Resource Group and Media Resource Group List for the MTP requirement
26. Add an SIP Trunk for the Nortel CS1000 PBX under the Device pull-down menu
27. Add a Route Pattern to reach the Nortel's phone DN extensions
28. Add a Route Pattern to reach the Nortel CallPilot VoiceMail system
29. Add a Route Pattern to PSTN via Nortel CS1000M
30. Configure Cisco 7960 phone and line DN

## Configuration Menus and Commands

### Nortel Communication Server 1000 (CS1000) Configuration

Call Server Setup:

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

```
>ld 22  
PT2000
```

```
REQ prt
```



TYPE adan dch 3

ADAN DCH 3  
CTYP DCIP  
DES IP\_Trunk\_DCH  
**USR ISLD**  
ISLM 4000  
SSRC 1800  
OTBF 32  
NASA NO  
**IFC SL1**  
CNEG 1  
RLS ID 4  
**RCAP ND2**  
MBGA NO  
H323  
OVLN NO  
OVLS NO

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

```
>ld 97
SCSYS000
MEM AVAIL: (U/P): 2854769  USED U P: 182454 59352  TOT: 3096575
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
```



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

```
>ld 20
REQ: prt
TYPE: tnb
TN 62 0 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
```

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

```
>ld 20
REQ: prt
TYPE: tnb
TN 3
CDEN
CUST
DATE
PAGE
DES

DES 192.168.1.2
TN 003 0 00 00           → 1st channel define on the gateway
```



**TYPE VGW**  
CUST 0  
**XTRK MC32**  
**ZONE 000**

DES 192.168.1.2  
**TN 003 0 00 01**  
TYPE VGW  
CUST 0  
XTRK MC32  
ZONE 000

→ 2nd channel define on the gateway

5. LD 11 – Configure for the Virtual lines for the Nortel IP phones (phone A and phone B)

Phone A (i2004)

```
>ld 11
SL1000
MEM AVAIL: (U/P): 2854769  USED U P: 182454 59352  TOT: 3096575
DISK RECS AVAIL: 1152
DIGITAL TELEPHONES AVAIL: 6  USED: 2  TOT: 8
IP USERS AVAIL: 6  USED: 2  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: 2405  USED: 95  TOT: 2500
DATA PORTS AVAIL: 2500  USED: 0  TOT: 2500
```

```
REQ: prt
TYPE: tnb
```

```
TN 61 0 0 02
DATE
PAGE
DES
```

```
DES I2004
TN 061 0 00 02 VIRTUAL
TYPE I2004
CDEN 8D
CUST 0
ZONE 000
FDN 2500
TGAR 1
LDN NO
NCOS 0
SGRP 0
RNPG 0
SCI 0
SSU
LNRS 16
XLST
CLS CTD FBA WTA LPR MTD FNA HTA TDD HFD CRPD
```



MWA LMPN RMMD SMWD AAD IMD XHD IRA NID OLD VCE DRG1  
POD DSX VMD CMSD SLKD CCSD SWD LNA **CNDA**  
**CFTA** SFD MRD DDV **CNIA** CDCA MSID DAPA BFED RCBF  
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  
VOLA VOUD CDMR  
CPND\_LANG ENG  
RCO 0  
**EFD 2500**  
**HUNT 2500**  
**EHT 2500**  
LHK 0  
LPK 1  
PLEV 02  
CSDN  
AST  
IAPG 0  
AACS NO  
ITNA NO  
DGRP  
MLWU\_LANG 0  
DNDR 0  
**KEY 00 SCR 2201 0 MARP**  
CPND  
NAME **ZEUS\_2201**  
XPLN 8  
DISPLAY\_FMT FIRST, LAST  
01  
02  
03 MIK  
04 MCK  
05  
06  
07  
08  
09  
10  
11  
12  
13  
14  
15  
16 MWK 2500  
17 TRN  
18 AO6  
19 CFW 16 2500  
20 RGA  
21 PRK  
22 RNP  
23





24 PRS  
25 CHG  
26 CPN  
27  
28  
29  
30  
31  
DATE 30 NOV 2005  
NACT

Phone B (i2002):

REQ: prt  
TYPE: tnb  
**TN 61001**  
DATE  
PAGE  
DES

DES I2002  
TN 06100001 VIRTUAL

**TYPE I2002**

CDEN 8D

CUST 0

**ZONE 000**

**FDN 3690**

TGAR 1

LDN NO

NCOS 0

SGRP 0

RNPG 0

SCI 0

SSU

LNRS 16

XLST

CLS CTD **FBD** WTA LPR MTD **FNA HTA** TDD HFD CRPD

**MWA** LMPN RMMD SMWD AAD IMD XHD IRA NID OLD VCE DRG1

POD DSX VMD CMSD SLKD CCSD SWD LNA **CNDA**

**CFTA** SFD MRD DDV **CNIA** CDCA MSID DAPA BFED RCBF

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 OCBF FLXD FTTC DNDY DNO3 MCBN

VOLA VOUD CDMR

CPND\_LANG ENG

RCO 0

**EFD 3690**

**HUNT 3690**

**EHT 3690**

LHK 0



```
LPK 1
PLEV 02
CSDN
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
DNDR 0
KEY 00 SCR 2201 0  MARP
  CPND
    NAME ZEUS_2201
    XPLN 8
    DISPLAY_FMT FIRST, LAST
  01
  02
  03 MIK
  04 MCK
  05
  06
  07
  08
  09
  10
  11
  12
  13
  14
  15
  16 MWK 3690
  17 TRN
  18 AO6
  19 CFW 16 3690
  20 RGA
  21 PRK
  22 RNP
  23
  24 PRS
  25 CHG
  26 CPN
  27
  28
  29
  30
  31
DATE 30 NOV 2005
NACT
```

6. LD 16 – Configure the SIP route

```
>ld 21
PT1000
```



REQ: prt  
TYPE: rdb  
CUST 0  
ROUT 10

**TYPE RDB**

CUST 00

DMOD

**ROUT 10**

DES SIP\_TIE

**TKTP TIE**

NPID\_TBL\_NUM 0

ESN NO

CNVT NO

SAT NO

RCLS EXT

**VTRK YES**

**ZONE 000**

**PCID SIP**

CRID YES

NODE 101

**DTRK NO**

**ISDN YES**

MODE ISLD

**DCH 3**

**IFC SL1**

PNI 00001

NCNA YES

NCRD YES

TRO NO

FALT NO

CTYP UKWN

INAC NO

ISAR NO

DAPC NO

PTYP ATT

AUTO NO

DNIS NO

DCDR NO

**ICOG IAO**

**SRCH LIN**

**TRMB YES**

STEP

**ACOD 2310**

TCPP NO

TARG 01

CLEN 1

BILN NO

OABS

INST

ANTK

SIGO STD

STYP SDAT

ICIS YES



TIMR ICF 512  
OGF 512  
EOD 13952  
DSI 34944  
NRD 10112  
DDL 70  
ODT 4096  
RGV 640  
GRD 896  
SFB 3  
NBS 2048  
NBL 4096

IENB 5

PAGE 002

TFD 0  
VSS 0  
VGD 6  
SST 5 0  
NEDC ORG  
FEDC ORG  
CPDC NO  
DLTN NO  
HOLD 02 02 40  
SEIZ 02 02  
SVFL 02 02  
DRNG NO  
CDR NO  
VRAT NO  
MUS NO  
MANO NO  
FRL 0 0  
FRL 1 0  
FRL 2 0  
FRL 3 0  
FRL 4 0  
FRL 5 0  
FRL 6 0  
FRL 7 0  
OHQ NO  
OHQT 00  
CBQ NO  
AUTH NO  
TTBL 0  
ATAN NO  
OHTD NO  
PLEV 2  
ALRM NO  
ART 0  
SGRP 0  
AACR NO



REQ:

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

```
>ld 86  
ESN000
```

```
MEM AVAIL: (U/P): 2819994  USED U P: 223389 69576  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
```

8. LD 87 – Configure CDP steering codes

```
>ld 87  
ESN000
```

```
MEM AVAIL: (U/P): 2819994  USED U P: 223389 69576  TOT: 3112959  
DISK RECS AVAIL: 1152
```

```
REQ prt
```

```
CUST 0
```

```
FEAT cdp
```

```
TYPE dsc
```

```
DSC
```

```
DSC 35
```

➔ **35xx Route to Unity VM send out via SIP route**



FLEN 0  
DSP LSC  
**RLI 10** → SIP Route List  
NPA  
NXX  
**DSC 36** → 36xx Route to CCM extension, sent out via SIP route  
FLEN 0  
DSP LSC  
**RLI 10** → SIP Route List  
NPA  
NXX

9. LD 90 – Configure BARS/NARS route to PSTN

>ld 90  
ESN000

MEM AVAIL: (U/P): 2819994 USED U P: 223389 69576 TOT: 3112959  
DISK RECS AVAIL: 1152  
REQ prt  
CUST 0  
**FEAT net**  
**TRAN ac1**  
**TYPE npa**

NPA

NPA 1212  
RLI 2  
SDRR NONE  
ITEI NONE

NPA 1408  
RLI 2  
SDRR NONE  
ITEI NONE

NPA 1800  
RLI 2  
SDRR NONE  
ITEI NONE

**NPA 1808**  
**RLI 7**  
**SDRR NONE**  
**ITEI NONE**

MEM AVAIL: (U/P): 2819994 USED U P: 223389 69576 TOT: 3112959  
DISK RECS AVAIL: 1152  
REQ



## Signaling Server Setup:

### 10. 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"/>
Intrazone Bandwidth (INTRA_BW):	<input type="text" value="10000"/>
Intrazone Strategy (INTRA_STGY):	Best Quality (BQ) <input type="button" value="v"/>
Interzone Bandwidth (INTER_BW):	<input type="text" value="10000"/>
Interzone Strategy (INTER_STGY):	Best Quality (BQ) <input type="button" value="v"/>
Resource Type (RES_TYPE):	Shared (SHARED) <input type="button" value="v"/>
Branch Office Support (ZBRN):	<input type="checkbox"/>
Description (ZDES):	<input type="text"/>



## 11. Configure a new IP Telephony Node summary

Site: 172.20.218.101 > Configuration > IP Telephony Configuration >

### Node Summary

New Node

Node: 101 Node IP: 172.20.216.100

Voice LAN (TLAN) IP address TN

**Signaling Server**  
172.20.216.103

**Pentium Card**  
**Succession Media Card**  
172.20.216.102 3 0

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





## 12. 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



### 13. 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 tree 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, Administration, Support, Tools, and Logout. The main content area is titled "VGW and IP phone codec profile" and contains the following 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	Selection
G711	Select <input checked="" type="checkbox"/>
G729A	Select <input checked="" type="checkbox"/>
G723.1	Select <input type="checkbox"/>
T38 FAX	Select <input checked="" type="checkbox"/>

At the bottom of the configuration area, there are three expandable sections: QoS, LAN configuration, and SNTP.



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

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



## 14. Configure the QoS section

The screenshot displays the Cisco configuration interface for QoS. On the left is 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, Administration, Support, Tools, and Logout. The main configuration area is divided into several sections:

- Codec**: A list of codecs with selection checkboxes:
  - Codec G711: Select
  - Codec G729A: Select
  - Codec G723.1: Select
  - Codec T38 FAX: Select
- QoS**:
  - Diffserv Codepoint(DSCP) Control packets:  Range: 0 to 63
  - Diffserv Codepoint(DSCP) Voice packets:  Range: 0 to 63
  - Enable 802.1Q support:
  - 802.1Q Bits value (802.1p):  Range: 0 to 7
- Configuration Options**: A list of expandable sections:
  - LAN configuration
  - SNTP
  - H323 GW Settings
  - Firmware
  - SIP GW Settings
  - SIP URI Map
  - SIP CD Services
  - Cards:
  - Signaling Servers:



## 15. Configure LAN Configuration section

The screenshot displays the Cisco configuration interface for LAN settings. On the left is a navigation tree with categories like System Status, Configuration, Network Numbering Plan, Software Upgrade, Patching, System Utility, Administration, Support, Tools, and Logout. The main area is divided into sections: QoS, LAN configuration, and Routes.

**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="172.20.218.101"/>	
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**

IP address	Subnet mask	
<input type="text" value="172.20.216.1"/>	<input type="text" value="255.255.255.0"/>	<input type="button" value="Remove"/>



## 16. Configure the SIP GW Setting section

<ul style="list-style-type: none"><li>System Status<ul style="list-style-type: none"><li>Call Server</li><li>IP Telephony</li></ul></li><li>Configuration<ul style="list-style-type: none"><li>Call Server</li><li>IP Telephony</li></ul></li><li>Network Numbering Plan</li><li>Software Upgrade</li><li>Patching</li><li>System Utility</li><li>Administration</li><li>Support</li><li>Tools</li><li>Logout</li></ul>	<b>SIP GW Settings</b>	
	Primary Proxy / Re-direct IP address	<input type="text" value="172.20.216.103"/>
	Primary Proxy / Re-direct IP Port	<input type="text" value="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	<input type="text" value="0.0.0.0"/>
	Secondary Proxy / Re-direct IP Port	<input type="text" value="5060"/>
	Secondary Proxy Supports Registration	<input type="checkbox"/>
	Secondary CDS Proxy or Re-direct server flag	<input type="checkbox"/>
	<b>SIP URI Map</b>	
	Public E.164/National domain name	<input type="text" value="+1"/>
	Public E.164/Subscriber domain name	<input type="text" value="+1314"/>
	Public E.164/Unknown domain name	<input type="text"/>
	Public E.164/Special Number domain name	<input type="text"/>
	Private/UDP domain name	<input type="text" value="rtp"/>
	Private/CDP domain name	<input type="text" value="interop.rtp"/>
	Private/Special Number domain name	<input type="text" value="SPN.rtp"/>
	Private/Unknown (vacant number routing) domain name	<input type="text"/>
	Unknown/Unknown domain name	<input type="text"/>
	<b>SIP CD Services</b>	
<b>Cards</b>	<input type="button" value="Add"/>	



17. Configure the Card section for the MC-32 VGMC card section

**SIP CD Services**

**▼ Cards** Add

**▼ Card 172.20.218.102 Properties** Remove

<b>Role</b>	<b>Follower</b>
Management LAN (ELAN) IP address	172.20.218.102 *
Management LAN (ELAN) MAC address	00:11:F9:E4:D0:11 *
Voice LAN (TLAN) IP address	172.20.216.102 *
Voice LAN (TLAN) gateway IP address	172.20.216.1
Hostname	MC_Node101 *
Card TN	3 *
Card processor type	Succession Media Card ▼
H323 ID	MC_Node101
Enable set TPS	<input checked="" type="checkbox"/>
System name	SS_Node101
System location	
System contact	

**▼ Signaling Servers** Add

**▼ Signaling Server 172.20.218.103 Properties** Remove

Save and Transfer Cancel

*\* Mandatory fields of current configuration*



## 18. Configure the Signaling Server section

Signaling Server 172.20.218.103 Properties		Remove
Role	Leader	
Management LAN (ELAN) IP address	<input type="text" value="172.20.218.103"/>	*
Management LAN (ELAN) MAC address	<input type="text" value="00:02:b3:f7:3a:86"/>	*
Voice LAN (TLAN) IP address	<input type="text" value="172.20.216.103"/>	*
Voice LAN (TLAN) gateway IP address	<input type="text" value="172.20.216.1"/>	
Hostname	<input type="text" value="SS_Node101"/>	*
H323 ID	<input type="text" value="SS_Node101"/>	
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="pbxlab.org"/>	
SIP Gateway Endpoint Name	<input type="text" value="SS_Node101"/>	
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_Node101"/>	
System location	<input type="text" value=""/>	
System contact	<input type="text" value=""/>	

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





## Network Routing Server Setup:

### 19. 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



## 20. Configure the NRS Server Settings

**Network Routing Service**

[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)



## Network Routing Service

[Home](#)[Configuration](#)[Tools](#)[Reports](#)[Administration](#)[Help](#) | [Logout](#)[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] \*Mandatory field indicator



## 21. Configure a Service Domain

### Network Routing Service

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	<input type="text" value="pbxlab.org"/> *
Domain description	<input type="text" value="Cisco Interop Lab Domain"/>

*\*Mandatory field indicator*

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



## 22. Configure a L1 Domain (UDP)

**Network Routing Service**

**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 (pbxlab.org)**

<b>Service Domains</b>	Domain name	<input type="text" value="rtp"/>
<b>=&gt; L1 Domains (UDP)</b>	Domain description	<input type="text" value="RTP Site"/>
<b>L0 Domains (CDP)</b>	Endpoint authentication enabled	<input type="text" value="Authentication off"/>
<b>Gateway Endpoints</b>	Authentication password	<input type="text"/>
<b>User Endpoints</b>	E.164 country code	<input type="text" value="1"/>
<b>Routing Entries</b>	E.164 area code	<input type="text" value="919"/>
<b>Default Routes</b>	International dialing access code	<input type="text" value="011"/>
<b>Collaborative Servers</b>	L1 domain dialing access code	<input type="text" value="9"/>
	National dialing access code	<input type="text" value="9"/>
	Local dialing access code	<input type="text" value="9"/>
	Special number 1	<input type="text"/>
	Special number 2	<input type="text"/>



## 23. Configure a L0 Domain (CDP)

Network Routing Service

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

Location: Configuration > L0 Domains (CDP) > View L0 Domain Property >

**View L0 Domain Property (pbxlab.org / rtp)**

Service Domains	Domain name	<input type="text" value="interop"/>	*
L1 Domains (UDP)	Domain description	<input type="text" value="CDP"/>	
=> L0 Domains (CDP)	Special number label	<input type="text"/>	
Gateway Endpoints	Unqualified number label	<input type="text"/>	
User Endpoints	Endpoint authentication enabled	<input type="text" value="Not configured"/>	
Routing Entries	Authentication password	<input type="text"/>	
Default Routes	E.164 country code	<input type="text" value="1"/>	
Collaborative Servers	E.164 area code	<input type="text" value="919"/>	
	International dialing access code	<input type="text" value="011"/>	
	L1 domain dialing access code	<input type="text" value="9"/>	
	National dialing access code	<input type="text" value="9"/>	
	Local dialing access code	<input type="text" value="9"/>	



## 24. 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 / rtp / interop)**

Service Domains

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

Endpoint name  \*

Endpoint description

Tandem endpoint name  [Look up](#)

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



## Network Routing Service

[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

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 Network Connection Server enabled 

*\*Mandatory field indicator*





## 25. 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):

/  /  /  [Look up](#)

Showing 1 - 8 of 8 < Previous | Next >

#	DN Prefix	DN Type	Route Cost	SIP URI Phone Context
1	<u>30</u>	Level0 regional	1	interop.rtp
2	<u>31</u>	Level0 regional	1	interop.rtp
3	<u>331</u>	Level1 regional	1	rtp
4	<u>331</u>	E.164 local	1	+1919
5	<u>332</u>	Level1 regional	1	rtp
6	<u>332</u>	E.164 local	1	+1919
7	<u>36</u>	Level0 regional	1	interop.rtp
8	<u>415</u>	E.164 national	1	+1



## Cisco Unified CallManager Configuration

25. Create the Media Resource Group and Media Resource Group List for the MTP requirement

System Route Plan Service Feature Device User Application Help

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

CISCO SYSTEMS

### Media Resource Group Configuration

[Add a New Media Resource Group](#)  
[Back to Find/List Media Resource Groups](#)  
[Dependency Records](#)

**Media Resource Group: GCM\_MTP (used by 16 devices)**  
Status: Ready

#### Media Resource Group Information

Media Resource Group Name\*

Description

#### Devices for this Group

Available Media Resources\*\*

ANN_CCMPUB (ANN) MOH_CCMPUB (MOH)
--------------------------------------

▼ ▲

Selected Media Resources\*

CFB_CCMPUB (CFB) MTP_CCMPUB (MTP)
--------------------------------------

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



## Media Resource Group List Configuration

[Add a New Media Resource Group List](#)  
[Back to Find/List Media Resource Group Lists](#)  
[Dependency Records](#)

Media Resource Group List: CCM\_MRGL (used by 16 devices)

Status: Ready

### Media Resource Group List Information

Media Resource Group List Name\*

### Media Resource Groups for this List

Available Media Resource Groups



Selected Media Resource Groups\*

(Groups listed in order of priority)



\* indicates required item



26. Add SIP Trunk for the Nortel CS1000M PBX under the Device pull-down menu

System Route Plan Service Feature Device User Application Help

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

CISCO SYSTEMS

## Trunk Configuration

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

**Product: SIP Trunk**  
**Device Protocol: SIP**  
Status: Ready

### Device Information

Device Name*	<input type="text" value="SS_Node101"/>
Description	<input type="text" value="Nortel CS101_SIP"/>
Device Pool*	<input type="text" value="Default"/>
Call Classification*	<input type="text" value="Use System Default"/>
Media Resource Group List	<input type="text" value="CCM_MRGL"/>
Location	<input type="text" value="&lt; None &gt;"/>
AAR Group	<input type="text" value="&lt; None &gt;"/>
<input checked="" type="checkbox"/> Media Termination Point Required	
Destination Address*	<input type="text" value="172.20.216.100"/>
<input type="checkbox"/> Destination Address is an SRV	
Destination Port	<input type="text" value="5060"/>



Destination Port	<input type="text" value="5060"/>
Incoming Port*	<input type="text" value="5061"/>
Outgoing Transport Type*	<input type="text" value="TCP"/>
Preferred Originating Codec*	<input type="text" value="711ulaw"/>

### Call Routing Information

#### Inbound Calls

Significant Digits*	<input type="text" value="All"/>
Connected Line ID Presentation*	<input type="text" value="Default"/>
Connected Name Presentation*	<input type="text" value="Default"/>
Calling Search Space	<input type="text" value=" &lt; None &gt;"/>
AAR Calling Search Space	<input type="text" value=" &lt; None &gt;"/>
Prefix DN	<input type="text"/>

Redirecting Number Delivery - Inbound

#### Outbound Calls

Calling Party Selection*	<input type="text" value="Originator"/>
Calling Line ID Presentation*	<input type="text" value="Default"/>
Calling Name Presentation*	<input type="text" value="Default"/>
Caller ID DN	<input type="text"/>
Caller Name	<input type="text"/>

Redirecting Number Delivery - Outbound

#### Multilevel Precedence and Preemption (MLPP) Information



27. Add a Route Pattern to reach the Nortel's phone DN extensions

System Route Plan Service Feature Device User Application Help

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

CISCO SYSTEMS

## Route Pattern Configuration

[Add a New Route Pattern](#)  
[Back to Find/List Route Patterns](#)

**Route Pattern: 210X**  
Status: Ready  
Note: Any update to this Route Pattern automatically resets the associated gateway or Route List

### Pattern Definition

Route Pattern*	<input type="text" value="210X"/>	
Partition	<input type="text" value="&lt; None &gt;"/>	
Description	<input type="text" value="To Nortel H323"/>	
Numbering Plan*	<input type="text" value="North American Numbering Plan"/>	
Route Filter	<input type="text" value="&lt; None &gt;"/>	
MLPP Precedence	<input type="text" value="Default"/>	
Gateway or Route List*	<input type="text" value="172.20.216.100"/> <a href="#">(Edit)</a>	
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern <input type="text" value="— Not Selected —"/>	
Call Classification*	<input type="text" value="OnNet"/> <input type="checkbox"/> Allow Device Override	
<input type="checkbox"/> Provide Outside Dial Tone	<input type="checkbox"/> Allow Overlap Sending	<input type="checkbox"/> Urgent Priority
<input type="checkbox"/> Require Forced Authorization Code		



Require Forced Authorization Code

Authorization Level

Require Client Matter Code

#### Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation

Calling Name Presentation

#### Connected Party Transformations

Connected Line ID Presentation

Connected Name Presentation

#### Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

#### ISDN Network-Specific Facilities Information Element

Carrier Identification Code

Network Service Protocol

Network Service

Service Parameter Name

Service Parameter Value

\* indicates required item.



28. Add a Route Pattern to reach the Nortel CallPilot VoiceMail system via SIP trunk

System Route Plan Service Feature Device User Application Help

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

CISCO SYSTEMS

## Route Pattern Configuration

[Add a New Route Pattern](#)  
[Back to Find/List Route Patterns](#)

**Route Pattern: 2500**  
Status: Ready  
Note: Any update to this Route Pattern automatically resets the associated gateway or Route List

### Pattern Definition

Route Pattern*	<input type="text" value="2500"/>	
Partition	<input type="text" value=" &lt; None &gt;"/>	
Description	<input type="text" value="CCM to CS101 Call Pilot"/>	
Numbering Plan*	<input type="text" value="North American Numbering Plan"/>	
Route Filter	<input type="text" value=" &lt; None &gt;"/>	
MLPP Precedence	<input type="text" value="Default"/>	
Gateway or Route List*	<input type="text" value="SS_Node101"/> <a href="#">(Edit)</a>	
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern <input type="text" value=" — Not Selected —"/>	
Call Classification*	<input type="text" value="OffNet"/> <input type="checkbox"/> Allow Device Override	
<input type="checkbox"/> Provide Outside Dial Tone	<input type="checkbox"/> Allow Overlap Sending	<input type="checkbox"/> Urgent Priority
<input type="checkbox"/> Require Forced Authorization Code		





Require Forced Authorization Code

Authorization Level

Require Client Matter Code

#### Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation

Calling Name Presentation

#### Connected Party Transformations

Connected Line ID Presentation

Connected Name Presentation

#### Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

#### ISDN Network-Specific Facilities Information Element

Carrier Identification Code

Network Service Protocol

Network Service

Service Parameter Name

Service Parameter Value

\* indicates required item.



## 29. Add a Route Pattern to the PSTN via Nortel CS1000M

System Route Plan Service Feature Device User Application Help

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

CISCO SYSTEMS

### Route Pattern Configuration

[Add a New Route Pattern](#)  
[Back to Find/List Route Patterns](#)

**Route Pattern: 9.@**

Status: Ready  
Note: Any update to this Route Pattern automatically resets the associated gateway or Route List

#### Pattern Definition

Route Pattern*	<input type="text" value="9.@"/>	
Partition	<input type="text" value="&lt; None &gt;"/>	
Description	<input type="text"/>	
Numbering Plan*	<input type="text" value="North American Numbering Plan"/>	
Route Filter	<input type="text" value="&lt; None &gt;"/>	
MLPP Precedence	<input type="text" value="Default"/>	
Gateway or Route List*	<input type="text" value="SS_Node101"/> <a href="#">(Edit)</a>	
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern <input type="text" value="-- Not Selected --"/>	
Call Classification*	<input type="text" value="OnNet"/> <input type="checkbox"/> Allow Device Override	
<input checked="" type="checkbox"/> Provide Outside Dial Tone	<input type="checkbox"/> Allow Overlap Sending	<input type="checkbox"/> Urgent Priority
<input type="checkbox"/> Require Forced Authorization Code		



Require Forced Authorization Code

Authorization Level

Require Client Matter Code

#### Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation

Calling Name Presentation

#### Connected Party Transformations

Connected Line ID Presentation

Connected Name Presentation

#### Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

#### ISDN Network-Specific Facilities Information Element

Carrier Identification Code

Network Service Protocol

Network Service

Service Parameter Name

Service Parameter Value

\* indicates required item.



30. Configure Cisco 7960 and 7940 phones with 3601 and 3602 DN.

Cisco SCCP 7960 extension 3601:

## Directory Number Configuration

[Configure Device \(SEP00146A3C1A08\)](#)  
[Dependency Records](#)

**Associated With**  
SEP00146A3C1A08  
7960 (Line 1)

**Directory Number: 3601**  
Status: Ready  
Note: Any update to this Directory Number automatically resets the associated devices

**Directory Number**

Directory Number\*

Partition

**Directory Number Settings**

Voice Mail Profile   
(Choose <None> to use default)

Calling Search Space

AAR Group

User Hold Audio Source

Network Hold Audio Source

Auto Answer

**Call Forward and Pickup Settings**

	Voice Mail	Coverage/ Destination	Calling Search Space
Forward All	<input type="checkbox"/>	<input type="text" value=""/>	<input type="text" value=" &lt; None &gt;"/>
Forward Busy Internal	<input type="checkbox"/>	<input type="text" value=" 3602"/>	<input type="text" value=" &lt; None &gt;"/>
Forward Busy External	<input type="checkbox"/>	<input type="text" value=" 3602"/>	<input type="text" value=" &lt; None &gt;"/>
Forward No Answer Internal	<input type="checkbox"/>	<input type="text" value=" 3602"/>	<input type="text" value=" &lt; None &gt;"/>
Forward No Answer External	<input type="checkbox"/>	<input type="text" value=" 3602"/>	<input type="text" value=" &lt; None &gt;"/>



Forward No Coverage Internal	<input type="checkbox"/>	<input type="text" value="3602"/>	< None > ▾
Forward No Coverage External	<input type="checkbox"/>	<input type="text" value="3602"/>	< None > ▾
No Answer Ring Duration	<input type="text" value="5"/>	(seconds)	
Call Pickup Group	<input type="text" value="&lt; None &gt;"/>	<a href="#">(View Details)</a>	
<b>MLPP Alternate Party Settings</b>			
Target (Destination)	<input type="text"/>		
Calling Search Space	< None > ▾		
No Answer Ring Duration	<input type="text"/>	(seconds)	
<b>Line Settings for all Devices</b>			
Alerting Name	<input type="text" value="SCCP_3601A"/>		
<b>Line Settings for this Device</b>			
Display (Internal Caller ID)	<input type="text" value="SCCP_3601"/>		
Line Text Label	<input type="text" value="SCCP_3601"/>		
External Phone Number Mask	<input type="text"/>		
Message Waiting Lamp Policy	Use System Policy ▾		
Ring Setting (Phone Idle)	Use System Default ▾		
Ring Setting (Phone Active)**	Use System Default ▾		
<b>Multiple Call / Call Waiting Settings</b>			
Maximum Number of Calls*	<input type="text" value="4"/>	(1 - 200)	
Busy Trigger*	<input type="text" value="1"/>	(<= Max. Calls)	
<b>Forwarded Call Information Display</b>			
<input checked="" type="checkbox"/> Caller Name		<input checked="" type="checkbox"/> Caller Number	
<input checked="" type="checkbox"/> Redirected Number		<input checked="" type="checkbox"/> Dialed Number	



Cisco SCCP 7940 extension 3602:

<b>Associated With</b> SEP001120DFE7D8 7940 (Line 1)	<b>Directory Number: 3602</b>		
	Status: Ready Note: Any update to this Directory Number automatically resets the associated devices		
	<input type="button" value="Update"/> <input type="button" value="Remove from Device"/> <input type="button" value="Reset Devices"/>		
	<b>Directory Number</b>		
	Directory Number* <input type="text" value="3602"/>		
	Partition <input type="text" value="&lt; None &gt;"/>		
	<b>Directory Number Settings</b>		
	Voice Mail Profile <input type="text" value="2500_CallPilot"/> (Choose <None> to use default)		
	Calling Search Space <input type="text" value="&lt; None &gt;"/>		
	AAR Group <input type="text" value="&lt; None &gt;"/>		
User Hold Audio Source <input type="text" value="1 - SampleAudioSource"/>			
Network Hold Audio Source <input type="text" value="1 - SampleAudioSource"/>			
Auto Answer <input type="text" value="Auto Answer Off"/>			
<b>Call Forward and Pickup Settings</b>			
	<b>Voice Mail</b>	<b>Coverage/ Destination</b>	<b>Calling Search Space</b>
Forward All	<input type="checkbox"/>	<input type="text"/>	<input type="text" value="&lt; None &gt;"/>
Forward Busy Internal	<input checked="" type="checkbox"/>	<input type="text"/>	<input type="text" value="&lt; None &gt;"/>
Forward Busy External	<input checked="" type="checkbox"/>	<input type="text"/>	<input type="text" value="&lt; None &gt;"/>
Forward No Answer Internal	<input checked="" type="checkbox"/>	<input type="text"/>	<input type="text" value="&lt; None &gt;"/>
Forward No Answer External	<input checked="" type="checkbox"/>	<input type="text"/>	<input type="text" value="&lt; None &gt;"/>
Forward No Coverage Internal	<input checked="" type="checkbox"/>	<input type="text"/>	<input type="text" value="&lt; None &gt;"/>
Forward No Coverage External	<input checked="" type="checkbox"/>	<input type="text"/>	<input type="text" value="&lt; None &gt;"/>



Forward No Coverage Internal	<input checked="" type="checkbox"/>	<input type="text"/>	< None > ▾
Forward No Coverage External	<input checked="" type="checkbox"/>	<input type="text"/>	< None > ▾
No Answer Ring Duration	<input type="text" value="20"/>	(seconds)	
Call Pickup Group	<input type="text" value="&lt; None &gt;"/>	<a href="#">(View Details)</a>	
<b>MLPP Alternate Party Settings</b>			
Target (Destination)	<input type="text"/>		
Calling Search Space	<input type="text" value="&lt; None &gt;"/>		
No Answer Ring Duration	<input type="text"/>	(seconds)	
<b>Line Settings for all Devices</b>			
Alerting Name	<input type="text" value="SCCP_3602A"/>		
<b>Line Settings for this Device</b>			
Display (Internal Caller ID)	<input type="text" value="SCCP_3602"/>		
Line Text Label	<input type="text" value="SCCP_3602"/>		
External Phone Number Mask	<input type="text"/>		
Message Waiting Lamp Policy	<input type="text" value="Use System Policy"/>		
Ring Setting (Phone Idle)	<input type="text" value="Use System Default"/>		
Ring Setting (Phone Active)**	<input type="text" value="Use System Default"/>		
<b>Multiple Call / Call Waiting Settings</b>			
Maximum Number of Calls*	<input type="text" value="4"/>	(1 - 200)	
Busy Trigger*	<input type="text" value="1"/>	(<= Max. Calls)	
<b>Forwarded Call Information Display</b>			
<input checked="" type="checkbox"/> Caller Name		<input checked="" type="checkbox"/> Caller Number	
<input checked="" type="checkbox"/> Redirected Number		<input checked="" type="checkbox"/> Dialed Number	



## Acronyms

Acronym	Definitions
ANF-PR	Additional Network Feature Path Replacement
CCM	Cisco Unified CallManager
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
PSTN	Public Switched Telephone Network

## Important Information

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