Nortel CS1000 Release 5.0 to Cisco Unified Communications Manager Release 7.0 using Cisco Unified Border Element (CUBE) for SIP-to-SIP Interworking

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Introduction

- This application note describes the necessary steps and configurations for connectivity between a Nortel Communication Server 1000 (CS1000) Rel. 5.0 and Cisco Unified Communications Manager 7.0 using Cisco Unified Border Element 1.2 (CUBE) with SIP trunks.

- The network topology diagram (Figure 1) shows the test setup for end-to-end interoperability between Cisco Unified Communications Manager Release 7.0 connected to CS1000 Rel. 5.0 via CUBE 1.2 using SIP trunks. A gigabit Ethernet port on the CUBE served as the SIP trunking interface.

- Features tested are:
  1. Basic calls between the two systems and verification of voice path
  2. Calling/connected party name and number delivery (allowed and restricted)
  3. 3-way (ad-hoc) conference
  4. Call transfer (blind, early attended, attended)
  5. Call forward (all, busy and no answer)
  6. Hold/Resume
  7. Voice messaging with Message Waiting Indication (MWI) activation/deactivation, and
  8. DTMF-relay (RFC2833) – verification of DTMF-relay by accessing each other’s VM system and responding to prompts using the keypad to send RTP Telephone Event (RFC2833) of digits pressed.

- This application note describes the use of Cisco 2851 voice gateway to run the CUBE features set, however other Cisco voice gateways may be used. The selection of Cisco gateway model will be dependent on the capabilities and the capacity required by the planned network deployment. Following is a list of Cisco IOS products capable of running CUBE.

  Cisco 2800 Series Integrated Services Routers
  Cisco 3800 Series Integrated Services Routers
  Cisco AS5350XM Universal Gateway
  Cisco AS5400XM Universal Gateway
Network Topology

Figure 1. Network Topology for SIP trunk using G729
Limitations and Caveats

Early Offer Requirement

- For Nortel CS1000 and Cisco Unified Communications Manager (CUCM) to interoperate properly with each other (Basic Call and/or Supplementary features), the CS1000 requires SIP INVITE with early offer (i.e. the delivery of media attributes with the SIP Invite message). SIP INVITE with early offer is enabled on the CUCM by checking the “Media Termination Point Required” box under the SIP Trunk configuration page. However, as of CUCM Rel. 7.0, G711 is the only preferred originating codec option for this static Media Termination Point (MTP) insertion. For deployments requiring G729 as originating codec option, the CUCM will have to allocate MTP dynamically. Thus, the MTP required box is unchecked and the CUCM falls back to its default INVITE method of delay offer. The Cisco Unified Border Element (CUBE) corrects this by forcing the transmission of early offer SIP INVITE to the Nortel while maintaining the G729 codec option.

No Audio or One-Way Audio Issue

- There were audio issues (one way and no audio) found in some of the transfer, hold/resume and conference call scenarios when MTP is disabled and media resources are dynamically allocated on the CUCM. Trace analysis shows that there are differences on how both systems implement the SIP protocol in its SIP header and SDP audio attributes. As a workaround, the following voice-class SIP profile is applied to the dial-peer towards the Nortel PBX to “repair” or “modify” incoming SIP/SDP header values from the CUCM.

  - voice class sip-profiles 1
  - request ANY sip-header Allow-Header modify "OPTIONS," ""
  - response ANY sip-header Allow-Header modify "OPTIONS," ""
  - request ACK sdp-header Audio-Attribute modify "sendonly" "sendrecv"
  - request REINVITE sdp-header Audio-Attribute modify "inactive" "sendrecv"

- When the CUCM is the transferring node in a local or network early attended call transfers, ringback tone is not heard by the calling party (i.e. Nortel phone). This is because the CUCM keeps the audio port closed and RTP in inactive state (hold state).

Alerting Name Delivery and Presentation

- When calling from a CUCM IP Phone to a Nortel CS1000 phone, the CS1000 does not send alerting name information within its SIP 180 message back to the CUBE / CUCM. When calling from a Nortel phone to a CUCM IP Phone, the CUCM sends alerting name information using PAI header within its 180 message back to the CUBE, however the CUBE does not relay this information back to the Nortel. However, since the Nortel does not support alerting name delivery and presentation, this information is not displayed on the Nortel phone regardless of whether the CUBE relayed this information or not.

Name and Number Updates

- Although transferred, forwarded and conference calls are completed across the network, the calling/connected names and numbers are not updated accordingly.

Name and Number Presentation Restrictions

- Calling Line ID Restriction (CLIR) / Calling Name ID Restriction (CNIR): Aside from sending SIP Invite message with the SIP FROM: header set to “Anonymous”, both systems support P-Asserted Identity (PAI) method within the SIP INVITE message to send calling name / number information with the appropriate PRIVACY header setting to restrict the presentation of name and/or number. To restrict calling name and number, The Nortel CS1000 sets the privacy setting to id (for number) and user (for name). The CUCM sets the privacy setting to id to restrict both name and number. However, CUBE does not support relaying the PRIVACY header on a per user basis. The CUBE does not transparently pass the privacy information from the CUCM to the CS1000 and vice versa. The CUBE does, however, provide the option of restricting name and/or number for all calls by configuring the SIP UA privacy settings under the global SIP configuration.

  • Connected Line ID Restriction (COLR) / Connected Name ID Restriction (CONR): The Nortel does not support the delivery and presentation of connected name and number. On the other hand, the CUCM supports P-Asserted Identity (PAI) method within the SIP RINGING (180) message to send connected name and number information along with PRIVACY header to restrict the presentation of
name and number. Since the Nortel does not support the connected name and number feature, Nortel phones do not display this information regardless of the privacy settings sent by the CUCM/CUBE.

Blind Call Transfers

- Both Nortel phones and Cisco Unified Communications Manager TNP phones (7961, 7970, 7971 and 7911) phones do not support SIP Blind Call Transfer.

Call Conference

- To ensure conference calls are sustained, a Media Resource Group (MRG) selecting a Conference Bridge using the defined G729 device pool is created. This Conference Bridge is associated with CUBE DSPfarm and is registered to the CUCM. The Cisco IP phones are then assigned to this Media Resource Group (MRG). Please refer to CUCM / CUBE configuration.

Call Completion

- Call Completion (Callback) is not a supported feature on either systems using standard SIP protocol.

Call Diversion

- For diverted calls, the CUCM uses the SIP Diversion header to pass the redirect information across the SIP Trunk while the Nortel CS1000 uses the SIP History-Info field. The CUBE may be configured to support the History-Info field to translate Diversion header information from the CUCM. However, the way CUBE constructs the History-Info header is not interoperable with how Nortel CS1000 implements this header information. Thus, redirected calls from the CUCM to the Nortel/CallPilot side are treated as direct access calls instead of being redirected to the called party’s voice mailbox. Since the CUCM does not support History-Info field, redirected calls from the Nortel are not recognized by the CUCM. The CUBE does not translate the History-Info field to Diversion header. Thus, again, these calls are treated as direct access calls. As a result, Cisco Unity and/or Nortel Call Pilot will not work as a centralized voice mail system for both systems. Users from one system cannot be a subscriber to the VM system hosted by the other system. (Refer to DDTS CSCta16365)

Message Waiting Indicator

- Due the reasons provided in the Call Diversion section, the delivery of MWI activation and deactivation messages across the SIP trunk is irrelevant at this time as a user of one system (CUCM or CS1000) cannot be a subscriber to the other systems VM. To test Message Waiting Indication (MWI) interoperability, activation and deactivation messages were sent manually. Even though both sides sent activate/deactivate message via SIP NOTIFY message, the subscribed phone on the other side does not light or extinguish MWI. When the Nortel CS1000 sends SIP NOTIFY activate/deactivate message to the CUCM via CUBE, the CUBE responds with SIP 500 – Internal Server Error. On the other hand, when the CUCM/Unity sends SIP Notify activate/deactivate messages to the CS1000, the CUBE responds with a SIP 481 – Call Leg/Transaction does not exist.

DTMF Relay

- To ensure the proper delivery of DTMF digits across the SIP trunk, the telephony-event payload type is set to “101” for both systems. DTMF interworking was verified by successfully entering DTMF digits as response to VM prompts.

System Components

Hardware Requirements

- Cisco Unified Communications Manager MCS -7835H server,
- Unity server MCS-7835H
- Catalyst switch 3560 PoE-48
- Cisco 7971, 7961 and 7960 IP phones
- Cisco 2851 serving as Cisco Unified Border Element (CUBE)
- Nortel CS1000
- Nortel digital (2616) and IP (i2004P2) phones

**Software Requirements**

- Cisco Unified Communications Manager Release 7.0(1)
- Cisco Unity Release 7.1(1)
- CUBE (C2851) IOS image: c c2800nm-ipvoice_ivs-mz.124-22.T.bin
- CS1000 Release 5.0 (VERSION 4021 RELEASE 5 ISSUE 00 W)

**Features**

- CLIP-Calling Line (Number) Identification Presentation
- CLIR-Calling Line (Number) Identification Restriction (Please see Limitations and Caveats section)
- CNIP-Calling Name Identification Presentation
- CNIR-Calling Name Identification Restriction (Please see Limitations and Caveats section)
- Alerting Name (Please see Limitations and Caveats section)
- Attended Call Transfer (Please see Limitations and Caveats section)
- Early Attended Call Transfer (Please see Limitations and Caveats section)
- CFU-Call Forwarding Unconditional (Please see Limitations and Caveats section)
- CFB-Call Forwarding Busy (Please see Limitations and Caveats section)
- CFNA-Call Forwarding No Answer (Please see Limitations and Caveats section)
- COLP-Connected Line (Number) Identification Presentation (Please see Limitations and Caveats section)
- COLR-Connected Line (Number) Identification Restriction (Please see Limitations and Caveats section)
- CONP-Connected Name Identification Presentation (Please see Limitations and Caveats section)
- CONR-Connected Name Identification Restriction (Please see Limitations and Caveats section)
- Hold and Resume (Please see Limitations and Caveats section)
- Conference Call (Please see Limitations and Caveats section)
- DTMF-relay – (Please see Limitations and Caveats section)

**Features Not Supported**

- MWI-Message Waiting Indication (lamp ON, lamp OFF) across the SIP Trunk
- Call Completion (Callback; Automatic Callback)
- Blind Call Transfer
Configuration

Configuration Sequence and Tasks

Nortel Communication Server 1000 PBX Configuration Sequence and Tasks

Call Server Setup

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
9. LD 21 – List Trunk Member

Signaling Server Setup via the Nortel Element Manager

1. Configure the Zones
2. Configure a new IP Telephony Node summary
3. Configure the Node section
4. Configure the VGW and IP phone codec profile section
5. Configure the Quality of Service (QoS) section
6. Configure LAN Configuration section
7. Configure the SIP GW Setting section
8. Configure the Card section for the MC-32 VGMC card section
9. Configure the Signaling Server section

NRS (Network Routing Server)

10. Configure the System Wide Settings
11. Configure the NRS Server Settings
12. Configure a Service Domain
13. Configure a L1 Domain (UDP)
14. Configure a L0 Domain (CDP)
15. Configure a SIP gateway
16. Configure the Routing Entries
Configuration Menus and Commands

Nortel Communication Server 1000 (CS1000) Configuration

Call Server Setup:

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

REQ: prt
TYPE: adan dch 30

ADAN     DCH 30
CTYP DCIP
DES  SIP_Trunk
USR  ISLD
ISLM 4000
SSRC 1800
OTBF 32
NASA YES
IFC  SL1
CNEG 1
RLS  ID 4
RCAP MWI ND3 CPK
MBGA NO
H323
OVLR NO
OVLS NO

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

REQ: prt
TYPE: suppl

SUPL.
SUPT SLOT XPEC0  XPEC1  SHLF  ZONE0/1 IPR0/1

000  IPMG ---- ---- 0 000 172.30.13.105 1 --- -----------
020  ---- ---- VIRTUAL ---- --- --- --- -----------

LD 14 – Configure the SIP Virtual Trunks to the Signaling Server

REQ: prt
TYPE: ipti
TN  020 0 3 0
DATE
PAGE

DES  SIP_TRUNK
TN  020 0 03 00 VIRTUAL
TYPE IPTI
CDEN 8D
CUST 0
XTRK VTRK
ZONE 000
LDOP BOP
TIMP 600
BIMP 600
AUTO_BIMP NO
NMUS NO
TRK ANLG
NCOS 0
RTMB 30 1
CHID 1
TGAR 1
STRI/STRO IMM IMM
SUPN YES
AST NO
IAPG 0
CLS CTD DTN CND ECD WTA LPR APN THFD SPCD MSBT
P10 NTC MID
TKID
AACR NO
DATE 27 JUN 2008

*****************************************************************************************************
LD 11 –Virtual lines for the Nortel IP phone (i200x series)
*****************************************************************************************************
REQ PRT
TYPE: tnbl
TN 020 0 0 0
DATE
PAGE
DES

DES TESTIP
TN 020 0 00 00 VIRTUAL
TYPE 2004P2
CDEN 8D
CTYP XDLC
CUST 0
NUID
NHTN
CFG_ZONE 000
CUR_ZONE 000
ERL 0
ECL 0
FDN
TGAR 0
LDN NO
NCOS 0
SGRP 0
RNPG 0
SCI 0
SSU
XLST
CLS UNR FBA WTA LPR MTD FNA HTA TDD HFA CRPD
MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
POD DSX VMD 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
USMD USRD ULAD RTDD RBDD RBHD PGND FLXD FTTC DNDY DNO3 MCBN
VOLA VOUD CDMR ICRD MCDD T87D KEM2 MSNV FRA PKCH
CPND_LANG ENG
RCO 0
EFD
HUNT
EHT
LHK 0
PLEV 02
DANI NO
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG LANG ENG
DNDR 0
KEY 00 SCR 2330 0 MARP
CPND
   NAME Apollo_30
   XPLN 13
   DISPLAY_FMT FIRST,LAST
   01 MIK
   02
   03
   04
   05
   06
   07
   08
   09
   10
   11
   12
   13
   14
   15
   16
   17 TRN
   18 A06
   19 CFW 16
   20 RGA
   21 PRK
   22 RNP
   23
   24 PRS
   25 CHG
LD 11 –Nortel Digital Station Phone (2616 series)

REQ:prt
TYPE: 2616

TN 000 0 7 1
DATE
PAGE
DES

DES NE
TN 000 0 07 01 VIRTUAL
TYPE 2616
CDEN 8D
CTYP XDLC
CUST 0
AOM 0
ERL 0
FDN
TGAR 0
LDN NO
NCOS 0
SGRP 0
RNPG 0
SCI 0
SSU
XLST

CLS CTD FBA WTA LPR MTD FNA HTA ADD HFD
   MWD LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
   POD DSX VMD SLKD CCSD SWD LND CNDA
   CFTD 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 DGDA NAMA
   DRDD EXR0
   USMD USRD ULAD RTDD RBDD RBHD PGND FLXD FTTC DNDY DNO3 MCBN
   CDMR MCDD T87D PKCH
   CPND_LANG ENG
   RCO 0
   HUNT
   LHK 0
   PLEV 02
   DANI NO
   AST
   IAPG 0
   AACS NO
   ITNA NO
   DGRP
MLWU_LANG 0
DNDR 0
KEY 00 SCR 2332 MARP
CPND
    NAME Apollo_32
    XPLN 13
    DISPLAY_FMT FIRST, LAST
    01
    02
    03 CFW 4 4205
    04 AO6
    05 TRN
    06
    07
    08
    09
    10
    11
    12
    13
    14
    15
DATE 10 APR 2008

NACT
REQ: prt
TYPE: 2616
TN 000 0 7 3
DATE
PAGE
DES

DES 2616
TN 000 0 07 03 VIRTUAL
TYPE 2616
CDEN 8D
CTYP XDLC
CUST 0
AOM 0
ERL 0
FDN
TGAR 0
LDN NO
NCOS 0
SGRP 0
RNPG 0
SCI 0
SSU
XLST
CLS CTD FBD WTA LPR MTD FND HTD ADD HFD
    MWD LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
    POD DSX VMD SLKD CCSD SWD LND CNDA
    CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBD
    ICDD CDMD LLCN MCTD CLBD AUTU
    GPUD DPUD DNDA CFXD ARHD CLTD ASCD
LD 16 – Configure the SIP route

>ld 21
PT1000

REQ: prt
TYPE: rdb
CUST 0
ROUT 31

TYPE RDB
CUST 00
ROUT 31
DES SIP_TRUNK
TKTP TIE
M911P NO
ESN NO
CNVT NO
SAT NO
RCLS EXT
VTRK YES
ZONE 001
PCID SIP
CRID YES
NODE 101
DTRK NO
ISDN YES
MODE ISLD
DCH 30
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 530
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
LD 86 – Configure the Route List Block for the Virtual Trunk route

>ld 86
ESN000

MEM AVAIL: (U/P): 99163053  USED U P: 5052246 54522  TOT: 104269821
DISK SPACE NEEDED: 46 KBYTES
REQ  prt
CUST 0
FEAT rlb
RLI  3

RLI  3
ENTR 0
LTER NO
ROUT 31
TOD  0 ON  1 ON  2 ON  3 ON
  4 ON  5 ON  6 ON  7 ON
VNS  NO
SCNV NO
CNV  NO
LD 87 – Configure CDP steering codes
******************************************************************************
>ld 87
ESN000
MEM AVAIL: (U/P): 99163053 USED U P: 5052246 54522 TOT: 104269821
DISK SPACE NEEDED: 46 KBYTES
REQ prt
CUST 0
FEAT cdp
TYPE dsc
DSC 42
DSC 42
FLEN 0
DSP LSC
RLI 3
NPA
NXX
MEM AVAIL: (U/P): 99163053 USED U P: 5052246 54522 TOT: 104269821
DISK SPACE NEEDED: 46 KBYTES
REQ
******************************************************************************
LD 21 – List Trunk Member
******************************************************************************
>ld 21
PT1000
REQ: ltm
CUST 0
ROUT 30
TYPE TLST
TKTP TIE
ROUT 31
DES SIP_TRUNK
TN 020 0 03 00 MBER 1 SIP_TIE
TN 020 0 03 01 MBER 2 SIP_TIE
TN 020 0 03 02 MBER 3 SIP_TIE
TN 020 0 03 03 MBER 4 SIP_TIE
TN 020 0 03 04 MBER 5 SIP_TIE
TN 020 0 03 05 MBER 6 SIP_TIE
TN 020 0 03 06 MBER 7 SIP_TIE
TN 020 0 03 07 MBER 8 SIP_TIE
Signaling Server Setup:
Configure the Zone(s)
Configure a new IP Telephony Node (1 of 10)
Configure a new IP Telephony Node (2 of 10)
Configure a new IP Telephony Node (3 of 10)
Configure a new IP Telephony Node (4 of 10)

![Image of CS 1000 Element Manager interface](image-url)

### SNTP Server Configuration
- **Alternate DNS Server 1 IP address**: 0.0.0.0
- **Alternate DNS Server 2 IP address**: 0.0.0.0

#### SNTP Server Settings
- **Mode**: Active
- **Interval**: 255
- **Port**: 20104

### SNTP Client Configuration
- **Mode**: Passive
- **Interval**: 255
- **Port**: 20104

### H323 GW Settings
- **Primary Gatekeeper (ILAN) IP address**: 172.30.11.103
- **Alternate Gatekeeper (ILAN) IP address**: 0.0.0.0
- **Primary Network Connect Server (ILAN) IP address**: 172.30.11.103
- **Primary Network Connect Server Port number**: 16500
- **Alternate Network Connect Server (ILAN) IP address**: 0.0.0.0

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EDCS # 777115 Rev # Initial Version
### CS 1000 ELEMENT MANAGER

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<tr>
<td>- Links</td>
<td>- Alternate Network Connect Server Port number</td>
</tr>
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<tr>
<td>- Zones</td>
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<td>- Supports Registration</td>
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<tr>
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<tr>
<td>- Geographic Redundancy</td>
<td>- Transport Protocol</td>
</tr>
<tr>
<td>- Software</td>
<td>TCP</td>
</tr>
</tbody>
</table>

**Firmware**
- Firmware download server IP address: 0.0.0.0
- Firmware file path: uploads/firmware/
- User ID: (default)
- Password: (default)

**TLS Security**
- Security Policy: Security Disabled
- TLS Security Port: 4431
- Client Authentication: (default)
- Re-negotiation: (default)
- X.509 Certificate Authentication: (default)

**Primary Proxy/Redirect Server**
- Primary Proxy/Redirect (T2) IP address: 172.30.11.103
- Port: 5660
- Supports Registration: (default)
- Primary CNAME Prefix/Redirect server flag: (default)
- Transport Protocol: TCP
Configure a new IP Telephony Node (6 of 10)
Configure a new IP Telephony Node (7 of 10)

![Cisco CS 1000 Element Manager](image)

**Private/UDP domain name**: rp

**Private/CDP domain name**: interop_rp

**Private/Special Number domain name**: PrivateSpecial

**Private/Unknown (vacant number routing) domain name**: PrivateUnknown

**Unknown/Unknown domain name**: UnknownUnknown

**SIP CD Services**

- **Service Enabled**: [ ]

- **Service DN used for making VTRK call from Agent**

- **Converged Telephone Call Forward DN**

- **User Info. field for Invite message on the Converged Desktop MO Set**

- **User Info. field for Invite message on the Converged Desktop MO Set**

- **User Info. field in the notify message for Converged Desktop**

**SIP CTI Services**

- **Service Enabled**: [ ]
Configure a new IP Telephony Node (8 of 10)
Configure a new IP Telephony Node (9 of 10)

![Configuration Interface](image)

<table>
<thead>
<tr>
<th>Configuration Details</th>
<th>Value</th>
</tr>
</thead>
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<tr>
<td>Subscriber Number (SN)</td>
<td>0</td>
</tr>
<tr>
<td>National Number (NN)</td>
<td>0</td>
</tr>
<tr>
<td>International Number</td>
<td></td>
</tr>
<tr>
<td>Cards Add/Remove</td>
<td></td>
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</tr>
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<td>172.30.13.103</td>
</tr>
<tr>
<td>Embedded LAN (ELAN) MAC address</td>
<td>00:0e:0c:18:b15</td>
</tr>
<tr>
<td>Telephone LAN (TM) IP address</td>
<td>172.30.11.103</td>
</tr>
<tr>
<td>Telephone LAN (TM) gateway IP address</td>
<td>172.30.11.1</td>
</tr>
<tr>
<td>Hostname</td>
<td>SS_Node104_LDR</td>
</tr>
<tr>
<td>H323 ID</td>
<td>SS_Node104_LDR</td>
</tr>
<tr>
<td>Enable Line TPS</td>
<td>✓</td>
</tr>
<tr>
<td>Enable IP Peer Gateway (Virtual Trunk TPS)</td>
<td>H323 and SIP</td>
</tr>
<tr>
<td>Enable SIP Proxy/Redirect Server</td>
<td>✓</td>
</tr>
<tr>
<td>Local SIP TCP/UDP Port to Listen to</td>
<td>5660</td>
</tr>
</tbody>
</table>

*If Telephone LAN (TM) IP address and Telephone LAN (TM) gateway IP address are not in the same subnet as Telephone LAN (TM) Node IP address, when Line TPS or IP Peer Gateway is enabled, then the TPS and/or VTRK applications will not run.*
Configure a new IP Telephony Node (10 of 10)
Network Routing Server Setup:

NRS Overview
NRS System Wide Settings

```
Network Routing Service Manager

System Wide Settings

- DB sync interval for alternate [Seconds]
- 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

Save

* Mandatory field indicator
```
### NRS Server Settings (1 of 2)

#### NRS Settings

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Host name</td>
<td>SS_Node104_LDR</td>
</tr>
<tr>
<td>Primary IP (TLAN)</td>
<td>172.30.11.103</td>
</tr>
<tr>
<td>Alternate IP (TLAN)</td>
<td>0.0.0.0</td>
</tr>
<tr>
<td>Control prefix</td>
<td>40</td>
</tr>
</tbody>
</table>

#### 1.323 Ostekeeper Settings

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Protocol type</td>
<td></td>
</tr>
<tr>
<td>Time to live (TTL)</td>
<td></td>
</tr>
<tr>
<td>Rejected packet count</td>
<td></td>
</tr>
</tbody>
</table>

#### SIP Server Settings

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Mode</td>
<td>Redirect</td>
</tr>
<tr>
<td>UDP transport enabled</td>
<td></td>
</tr>
<tr>
<td>UDP port</td>
<td>5060</td>
</tr>
<tr>
<td>UDP maximum transmission unit (MTU)</td>
<td>1500</td>
</tr>
<tr>
<td>TCP transport enabled</td>
<td></td>
</tr>
<tr>
<td>TCP port</td>
<td>ANY</td>
</tr>
<tr>
<td>TCP maximum transmission unit (MTU)</td>
<td>1500</td>
</tr>
</tbody>
</table>
NRS Server Settings (2 of 2)

H.323 Gatekeeper Settings

Location request (LRQ) response timeout (Seconds) 3

SIP Server Settings

Mode: Redirect

UDP transport enabled: [ ]

UDP port: 5060

UDP maximum transmission unit (MTU): 1500

TCP transport enabled: [ ]

TCP port: 5060

TCP maximum transmission unit (MTU): 1500

Network Connection Server (NCS) Settings

Primary NCS port: 16380

Alternate NCS port: 16500

Primary NCS timeout (Seconds) 30

Save
Configure a Service Domain
Configure L1 Sub-Domain (UDP).

![Network Routing Service Manager](image)

**View L1 Domain Property (pbxlab.org)**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Domain name</td>
<td>NPI</td>
</tr>
<tr>
<td>Domain description</td>
<td>RTP Site</td>
</tr>
<tr>
<td>Endpoint authentication enabled</td>
<td>Off</td>
</tr>
<tr>
<td>Authentication password</td>
<td></td>
</tr>
<tr>
<td>E.164 country code</td>
<td>1</td>
</tr>
<tr>
<td>E.164 area code</td>
<td>408</td>
</tr>
<tr>
<td>E.164 international dialing access code</td>
<td>011</td>
</tr>
<tr>
<td>E.164 national dialing access code</td>
<td>9</td>
</tr>
<tr>
<td>E.164 local (subscriber) dialing access code</td>
<td>9</td>
</tr>
<tr>
<td>Private L1 domain (UDP location) dialing access code</td>
<td>9</td>
</tr>
<tr>
<td>Special number</td>
<td></td>
</tr>
<tr>
<td>Emergency service access prefix</td>
<td></td>
</tr>
<tr>
<td>Special number label</td>
<td>PrivateSpecial</td>
</tr>
</tbody>
</table>
Configure L0 Sub-Domain (CDP)
SIP gateway to CUCM via CUBE (1 of 2)
SIP gateway to CUCM via CUBE (2 of 2)
### Routing Entries (1 of 2)

![Network Routing Service Manager](image)

**Table: 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>24</td>
<td>Private level 0 regional (CDP steering code)</td>
<td>1</td>
<td>interop.rtp</td>
</tr>
<tr>
<td>2</td>
<td>33</td>
<td>Private level 0 regional (CDP steering code)</td>
<td>1</td>
<td>interop.rtp</td>
</tr>
<tr>
<td>3</td>
<td>24</td>
<td>Private level 0 regional (CDP steering code)</td>
<td>1</td>
<td>interop.rtp</td>
</tr>
<tr>
<td>4</td>
<td>44</td>
<td>Private level 0 regional (CDP steering code)</td>
<td>1</td>
<td>interop.rtp</td>
</tr>
<tr>
<td>5</td>
<td></td>
<td>Private level 0 regional (CDP steering code)</td>
<td>1</td>
<td>interop.rtp</td>
</tr>
</tbody>
</table>
Cisco 2851 Configuration – Cisco Unified Border Element (CUBE)

CUBE_G729#sh ver
Cisco IOS Software, 2800 Software (C2800NM-IPVOICE_IVS-M), Version 12.4(22)T, RE
LEASE SOFTWARE (fc1)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2008 by Cisco Systems, Inc.
Compiled Fri 10-Oct-08 00:06 by prod_rel_team

ROM: System Bootstrap, Version 12.3(8r)T7, RELEASE SOFTWARE (fc1)

CUBE_G729 uptime is 19 hours, 36 minutes
System returned to ROM by reload at 14:14:38 PST Tue May 12 2009
System image file is "flash:c2800nm-ipvoice_ivs-mz.124-22.T.bin"

Cisco 2851 (revision 53.51) with 247808K/14336K bytes of memory.
Processor board ID FHK0847F03X
2 Gigabit Ethernet interfaces
2 Channelized/Clear T1/PRI ports
4 Voice FXO interfaces
2 Voice FXS interfaces
DRAM configuration is 64 bits wide with parity enabled.
239K bytes of non-volatile configuration memory.
62592K bytes of ATA CompactFlash (Read/Write)

Configuration register is 0x2102

CUBE_G729#

CUBE_G729#sh run
Building configuration...

Current configuration : 5203 bytes
!
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
service sequence-numbers
!
hostname CUBE_G729
!
boot-start-marker
boot-end-marker
!
logging message-counter syslog
logging queue-limit 10000
logging buffered 1000000
logging rate-limit 10000
enable secret 5 $1$v0tv$DYoywWasCG5us.lpzy6Th.
enable password cisco
!
no aaa new-model
clock timezone PST -8
no network-clock-participate slot 1
voice-card 0
dspfarm
dsp services dspfarm
!
voice-card 1
dspfarm
!
ip source-route
!
ip cef
!
no ipv6 cef
multilink bundle-name authenticated
!
!
isdn switch-type primary-net5
!
!
voice service voip
allow-connections sip to sip
sip
bind control source-interface GigabitEthernet0/0
bind media source-interface GigabitEthernet0/0
header-passing
asserted-id pai
early-offer forced
history-info
midcall-signaling passthru
!
!
voice class codec 1
codec preference 1 g729r8
!
!
voice class sip-profiles 1
request ANY sip-header Allow-Header modify " OPTIONS, " ""
response ANY sip-header Allow-Header modify " OPTIONS, " ""
response 180 sip-header P-Asserted-Identity modify ".*" ".*"
request ACK sd-header Audio-Attribute modify "sendonly" ":sendrecv"
request REINVITE sd-header Audio-Attribute modify "inactive" ":sendrecv"

Global SIP configuration.
SIP Invite Early –offer method is forced
History-info is supported as a way to pass redirect information

Voice-class SIP profile is applied to the dial-peer towards the Nortel PBX to “repair” or “modify” incoming SIP/SDP header values from the CUCM.
archive
log config
hidekeys
controller T1 1/0/0
controller T1 1/0/1
no ip ftp passive
translation-rule 1
interface GigabitEthernet0/0
  ip address 172.20.66.55 255.255.255.0
duplex auto
speed auto
no keepalive
interface GigabitEthernet0/1
  ip address 192.168.89.3 255.255.255.0
shutdown
duplex auto
speed auto
  ip default-gateway 172.20.66.1
  ip forward-protocol nd
  ip route 0.0.0.0 0.0.0.0 172.20.66.1
ip http server
ip dns server view-group DNS
control-plane
voice-port 0/0/0
voice-port 0/0/1
voice-port 0/2/0
voice-port 0/2/1
! voice-port 0/2/2
! voice-port 0/2/3
! no ccm-manager fax protocol cisco
! no mgcp package-capability res-package
no mgcp package-capability fxr-package
mgcp fax t38 ecm
! sccp local GigabitEthernet0/0
sccp ccm 172.20.215.254 identifier 1 version 7.0
sccp
! sccp ccm group 1
  associate ccm 1 priority 1
  associate profile 1 register CFB000F352F26E9
! dspfarm profile 2 transcode
  codec g711ulaw
  codec g711alaw
  codec g729ar8
  codec g729abr8
  codec g729r8
  codec g729br8
  maximum sessions 3
  associate application SCCP
! dspfarm profile 1 conference
  codec g711ulaw
  codec g711alaw
  codec g729ar8
  codec g729abr8
  codec g729r8
  codec g729br8
  maximum sessions 6
  associate application SCCP
! 
dial-peer voice 22 voip
  description dial-peer to Nortel CS1000
  destination-pattern 23..
  voice-class codec 1
  voice-class sip g729 annexb-all
  voice-class sip profiles 1
  session protocol sipv2
  session target ipv4:172.30.11.100
  session transport udp
  incoming called-number .T
dtmf-relay rtp-nte
! 
dial-peer voice 42 voip
  description DIAL-PEER TO CM-MERCURY
  destination-pattern 42..
rtp payload-type cisco-codec-fax-ind 98

Conference Bridge configured in CUCM
voice-class codec 1
voice-class sip g729 annexb-all
session protocol sipv2
session target ipv4:172.20.215.254
session transport udp
dtmf-relay rtp-nte
no vad
!
dial-peer voice 4210 pots
destination-pattern 4210
port 0/0/0
!
dial-peer voice 4211 pots
destination-pattern 2401
port 0/0/1
!
dial-peer voice 70 voip
description dial-peer to UNITY
destination-pattern 70..
voice-class codec 1
voice-class sip g729 annexb-all
session protocol sipv2
session target ipv4:172.20.215.254
session transport udp
dtmf-relay rtp-nte
!
dial-peer voice 23 voip
description dial-peer to Nortel CS104
shutdown
destination-pattern 23..
voice-class codec 1
voice-class sip g729 annexb-all
session protocol sipv2
session target ipv4:172.30.11.100
dtmf-relay rtp-nte
!
dial-peer voice 223 voip
description dial-peer to Nortel CS104-2
shutdown
destination-pattern 23..
voice-class codec 1
voice-class sip outbound-proxy ipv4:172.30.11.103
session protocol sipv2
session target dns:pbxlab.org
!
dial-peer voice 45 voip
!
dial-peer voice 3300 voip
description DIAL-PEER TO CM-MERCURY
destination-pattern 34..
rtp payload-type cisco-codec-fax-ind 98
voice-class codec 1
voice-class sip g729 annexb-all
session protocol sipv2
session target ipv4:172.20.215.254
session transport udp
dtmf-relay rtp-nte
no vad
!

dial-peer voice 25 voip
description dial-peer to Nortel CS1000
destination-pattern 25..
voice-class codec 1
voice-class sip g729 annexb-all
voice-class sip profiles 1
session protocol sipv2
session target ipv4:172.30.11.100
session transport udp
incoming called-number .T
dtmf-relay rtp-nte
!
!
sip-ua
mwi-server ipv4:172.30.11.100 expires 3600 port 5060 transport udp unsolicited
!
!
gatekeeper
shutdown
!
!
telephony-service
em logout 0:0 0:0 0:0
mwi relay
max-conferences 8 gain -6
transfer-system full-consult
create cnf-files version-stamp Jan 01 2002 00:00:00
!
!
line con 0
line aux 0
line vty 0 4
exec-timeout 0 0
password cisco
login
line vty 5 10
exec-timeout 0 0
password 7 0822455D0A16
login
!
exception data-corruption buffer truncate
scheduler allocate 20000 1000
end

CUBE_G729#
Cisco Unified Communications Manager Default Region (associating G729)

![Cisco Unified CM Administration](image)

**Status**
- Update successful
- Click on the Reset button to have the changes take effect.

**Region Information**
- **Name**: Default

**Region Relationships**
<table>
<thead>
<tr>
<th>Region</th>
<th>Audio Codes</th>
<th>Video Call Bandwidth</th>
<th>Link Loss Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td>G.711</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>G729_Region</td>
<td>G.729</td>
<td>384</td>
<td>Use System Default</td>
</tr>
</tbody>
</table>

**Modify Relationship to other Regions**
<table>
<thead>
<tr>
<th>Region</th>
<th>Audio Codes</th>
<th>Video Call Bandwidth</th>
<th>Link Loss Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td>G.729</td>
<td></td>
<td>Keep Current Setting</td>
</tr>
</tbody>
</table>

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## Cisco Unified Communications Manager G729 Region

### Region Information

<table>
<thead>
<tr>
<th>Name</th>
<th>G729 Region</th>
</tr>
</thead>
</table>

### Region Relationships

<table>
<thead>
<tr>
<th>Region</th>
<th>Audio Codec</th>
<th>Video Call Bandwidth</th>
<th>Link Loss Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td>G.729</td>
<td>364</td>
<td>Use System Default</td>
</tr>
<tr>
<td>G729 Region</td>
<td>G.729</td>
<td>364</td>
<td>Use System Default</td>
</tr>
</tbody>
</table>

**NOTE:** Regions(s) not displayed

Use System Default  
Use System Default  
Use System Default

### Modify Relationship to other Regions

- **Region**
  - **Audio Codec**
    - **Keep Current Setting**
  - **Video Call Bandwidth**
    - **Keep Current Setting**
  - **Link Loss Type**
    - **Keep Current Setting**

---

**Important Notice:**
- Indicates required item.
- The Audio Codec selection determines bandwidth only. The G.711 and G.722 codecs both result in a maximum bandwidth of 64 Kbps between regions and can be used interchangeably.
Cisco Unified Communications Manager G729 Device Pool Configuration (1 of 2)

**Device Pool Information:**
Device Pool: G729_DP (10 members)

**Device Pool Settings:**
- Device Pool Name: G729_DP
- Cisco Unified Communications Manager Group: Default
- Calling Search Space for Auto-registration: <None>
- Reverted Call Focus Priority: Default
- Local Route Group: <None>

**Roaming Sensitive Settings:**
- CMCategory Group: CMLocal
- CMRegion: G729 Region

**Notes:**
- Uses defined G729 Region
Cisco Unified Communications Manager Enterprise Parameters (Organization Top Level Domain) Configuration

User Management Parameters
- Effective Access Privileges for Overlapping User Groups and Roles
  - Maximum

Service Manager TCP ports parameters
- Service Manager TCP Server communication port number
  - 5000
- Service Manager TCP Client communication port number
  - 8888

CRS Application Parameters
- Busy Awaited Task Failed
  - false
- DCC Express Installed
  - false

Clusterwide Domain Configuration
- Organization Top Level Domain
  - nortel.com
- Cluster Fully Qualified Domain Name
  - nortel.com

Denial-of-Service Protection
- Denial-of-Service Protection Flag
  - TRUE

Cisco Support Use
- Cisco Support Use
  - 

Cisco Syslog Agent
- Remote Cisco Server Name
  - 

Matches Nortel Service Domain
### Cisco Unified Communications Manager Conference Bridge Configuration

#### Conference Bridge Information
- Conference Bridge: CUBE (CFB000F352F24E9)
- Registration: Registered with Cisco Unified Communications Manager CM-MERCURY
- IP Address: 172.25.46.96

#### Hardware Conference Bridge Info
- **MAC Address**: 000F352F24E9
- **Device Pool**: G729_DP

#### Instructions
- Use G729 Device Pool
- MAC Address of CUBE
- Gigabit Ethernet port

---

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EDCS # 777115 Rev # Initial Version
Cisco Unified Communications Manager Media Resource Group List

<table>
<thead>
<tr>
<th>Media Resource Group List Configuration</th>
</tr>
</thead>
<tbody>
<tr>
<td>Status: Ready</td>
</tr>
</tbody>
</table>

**Media Resource Group List Status**
Media Resource Group List: MRGL_G729 (used by 4 devices)

**Media Resource Group List Information**
Name: MRGL_G729

**Media Resource Groups for this List**
Available Media Resource Groups:
- MRG CM MERCURY

Selected Media Resource Groups:
- MRGL_G729

- Indicates required item.
Cisco Unified Communications Manager SIP Trunk Security Profile Configuration

<table>
<thead>
<tr>
<th>SIP Trunk Security Profile Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Name*</td>
</tr>
<tr>
<td>G729_None Secure TCP Profile</td>
</tr>
<tr>
<td>Description</td>
</tr>
<tr>
<td>for G729 Testbed - to CUBE</td>
</tr>
<tr>
<td>Device Security Mode</td>
</tr>
<tr>
<td>Non Secure</td>
</tr>
<tr>
<td>Incoming Transport Type*</td>
</tr>
<tr>
<td>TCP+UDP</td>
</tr>
<tr>
<td>Outgoing Transport Type</td>
</tr>
<tr>
<td>TCP</td>
</tr>
<tr>
<td>Enable Digest Authentication</td>
</tr>
<tr>
<td>☐</td>
</tr>
<tr>
<td>Renew Validity Term (minutes)*</td>
</tr>
<tr>
<td>600</td>
</tr>
<tr>
<td>X.509 Subject Name</td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td>Incoming Port*</td>
</tr>
<tr>
<td>5060</td>
</tr>
<tr>
<td>Enable Application Level Authorization</td>
</tr>
<tr>
<td>☐</td>
</tr>
<tr>
<td>Accept Presence Subscription</td>
</tr>
<tr>
<td>☐</td>
</tr>
<tr>
<td>Accept Out-of-Diallog REFER</td>
</tr>
<tr>
<td>☐</td>
</tr>
<tr>
<td>Accept Unsoldicated Notification</td>
</tr>
<tr>
<td>☐</td>
</tr>
<tr>
<td>Accept Replaces reader</td>
</tr>
<tr>
<td>☐</td>
</tr>
<tr>
<td>Transmit Security Status</td>
</tr>
<tr>
<td></td>
</tr>
</tbody>
</table>
Payload Type = 101  
Must be same as Nortel Payload Type for proper DTMF relay
Cisco Unified Communications Manager SIP Profile for Nortel CS1000M PBX Configuration (2 of 3)

![Cisco Unified CM Administration](image)

### SIP Profile Configuration

- **Reply-Invite**: 6
- **Start Media Port**: 16384
- **Stop Media Port**: 9776
- **Call Pickup URI**: `x-cisco-serviceuri-pickup`
- **Call Pickup Group URI**: `x-cisco-serviceuri-gppickup`
- **Meet Me Service URI**: `x-cisco-serviceuri-meetme`
- **User Info**: None
- **DTMF DB Level**: Nominal
- **Call Hold Ring Back**: Off
- **Anonymous Call Block**: Off
- **Caller ID Blocking**: User
- **Do Not Disturb Control**: User
- **Telnet Level (7900 and 7976)**: Disabled
- **Timer Keep AliveExpires (seconds)**: 120
- **Timer Subscribe Expires (seconds)**: 120
- **Timer Subscribe Delta (seconds)**: 5
- **Maximum Redirects**: 70
- **Off Hook To First Digit Timer (milliseconds)**: 15000
Use G729 Device Pool

Media Termination Point
Required box left unchecked to enable G729 as preferred codec option
Check Asserted-Identity box and select PAI option to match Nortel method of name delivery and presentation
Cisco Unified Communications Manager SIP Trunk Configuration to Nortel CS1000M PBX Configuration (3 of 4)

IP address of Cisco Unified Border Element (CUBE)
Set to RFC2833 to match Nortel DTMF Signaling Method
Use G729 Device Pool
Use Media Resource Group List defined for G729 codec operation
Cisco Unified Communications Manager SCCP Phone Level Configuration (2 of 6)

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Network Hold INOW Audio Source Location</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>AAR Group</td>
<td>Hub_None</td>
</tr>
<tr>
<td>User Locale</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Network Locale</td>
<td>&lt; none &gt;</td>
</tr>
<tr>
<td>Built In Bridge</td>
<td>Default</td>
</tr>
<tr>
<td>Privacy</td>
<td>Default</td>
</tr>
<tr>
<td>Device Mobility Mode</td>
<td>Default</td>
</tr>
<tr>
<td>Owner User ID</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Phone Load Name</td>
<td></td>
</tr>
<tr>
<td>Join Across Lines</td>
<td>Default</td>
</tr>
<tr>
<td>Use Trusted Relay Point</td>
<td>Default</td>
</tr>
<tr>
<td>RIF audible alert setting (Phone Busy)</td>
<td>Default</td>
</tr>
<tr>
<td>BLF audible alert setting (Phone Busy)</td>
<td>Default</td>
</tr>
<tr>
<td>Calling Party Transformation CSS</td>
<td>&lt; none &gt;</td>
</tr>
<tr>
<td>Use Device Pool Calling Party Transformation CSS</td>
<td></td>
</tr>
<tr>
<td>IS Active</td>
<td></td>
</tr>
<tr>
<td>Retype Video Call as Audio</td>
<td></td>
</tr>
<tr>
<td>Tones Representation Indicator (internal calls only)</td>
<td></td>
</tr>
</tbody>
</table>
Cisco Unified Communications Manager SCCP Phone Level Configuration (3 of 6)
Cisco Unified Communications Manager SCCP Phone Level Configuration (5 of 6)

<table>
<thead>
<tr>
<th>Extension Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enable Extension Mobility</td>
</tr>
<tr>
<td>Login Out Profile: Use Current Device Settings</td>
</tr>
<tr>
<td>Log In Time: &lt; None &gt;</td>
</tr>
<tr>
<td>Log Out Time: &lt; None &gt;</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>MLPP Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>MLPP Domain: &lt; None &gt;</td>
</tr>
<tr>
<td>MLPP Indication: Default</td>
</tr>
<tr>
<td>MLPP Preemption: Default</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Do Not Disturb</th>
</tr>
</thead>
<tbody>
<tr>
<td>Do Not Disturb</td>
</tr>
<tr>
<td>DND Option: Ringer Off</td>
</tr>
<tr>
<td>DND Incoming Call Alert: &lt; None &gt;</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Product Specific Configuration Layout</th>
</tr>
</thead>
<tbody>
<tr>
<td>Disable Speakerphone</td>
</tr>
<tr>
<td>Disable Speakerphone and Headset</td>
</tr>
</tbody>
</table>
Cisco Unified Communications Manager SCCP Phone Directory Number (Ext 4200) Level Configuration (2 of 4)

![Cisco Unified CM Administration Interface](image)

**Directory Number Configuration**
- **Save**, **Delete**, **Reset**, **Add New**
- **Network mail Gamma Audio Source** (None)
- **Auto Answer** (Auto Answer: Off)

**AAR Settings**
- **Voice Mail**: [ ]
- **AAR Destination Mask**: <None>
- **AAR Group**: [ ]

- **Retain this destination in the call forwarding history**

**Call Forward and Call Pickup Settings**
- **Calling Search Space Activation Policy**
  - **Forward All**: [ ]
  - **Secondary Calling Search Space for Forward All**: [ ]
  - **Forward Busy Internal**: [ ]
  - **Forward Busy External**: [ ]
  - **Forward No Answer Internal**: [ ]
  - **Forward No Answer External**: [ ]
  - **Forward No Coverage Internal**: [ ]
  - **Forward No Coverage External**: [ ]
  - **Forward on CTI Failure**: [ ]
  - **Forward Unregistered Internal**: [ ]
  - **Forward Unregistered External**: [ ]

- **Calling Search Space**
  - **UCC System Default**: [ ]
  - **Phones**: [ ]
Cisco Unified Communications Manager SCCP Phone Directory Number (Ext 4200) Level Configuration (3 of 4)
Cisco Unified Communications Manager SCCP Phone Directory Number (Ext 4200) Level Configuration (4 of 4)

![Cisco Unified CM Administration](image-url)

**Directory Number Configuration**

<table>
<thead>
<tr>
<th>Settings</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ring Setting (Phone Idle)</td>
<td>Ring</td>
</tr>
<tr>
<td>Ring Setting (Phone Active)</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Call Pickup Group Audio Alert Setting (Phone Idle)</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Call Pickup Group Audio Alert Setting (Phone Active)</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Monitoring Calling Search Space</td>
<td>None</td>
</tr>
</tbody>
</table>

**Multiple Call/Call Waiting Settings on Device SEP0006D7AA8D30**

Note: The range to select the Max. Number of calls is: 1-200

- Maximum Number of Calls: 
- Busy Trigger: 0 (Less than or equal to Max. Calls)

**Forwarded Call Information Display on Device SEP0006D7AA8D30**

- Caller Name
- Caller number
- Redirected Number
- Dialed Number

**Users Associated with Line**

- Associate End Users
Cisco Unified Communications Manager SIP Phone Level Configuration (1 of 7)

Use G729 Device Pool

Use Media Resource Group List defined for G729 codec operation
Cisco Unified Communications Manager SIP Phone Level Configuration (3 of 7)
Cisco Unified Communications Manager SIP Phone Level Configuration (4 of 7)
Cisco Unified Communications Manager SIP Phone Level Configuration (5 of 7)
Cisco Unified Communications Manager SIP Phone Level Configuration (6 of 7)

### Phone Configuration

- **Web Access**
- **Open to PC Port**
- **Logging Display**
- **Load Server**
- **Recording Tone**
- **Recording Tone Local Volume**
- **Recording Tone Remote Volume**
- **Recording Tone Duration**
- **RTP**
- **RTP**
- **Soft Key Timer**
- **Auto Call Select**
- **Log Server**
- **Advertise G.722 Codec**
- **Wideband Headset UI Control**
- **Wideband Handset UI Control**
- **Wideband Headset**
- **Wideband Handset**
- **Peer Firmware Sharing**
- **Cisco Discovery Protocol (CDP): Switch Port**
- **Cisco Discovery Protocol (CDP): FC Port**
- **Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port**
- **Link Layer Discovery Protocol (LLDP): PC Port**
- **LLDP Asset ID**

Options available:
- **Enabled**
- **Disabled**
- **PC Controlled**
- **100**
- **50**
- **Use System Default**
- **Enabled**
- **Disabled**
- **Enabled**
- **Enabled**
- **Enabled**
- **Enabled**
- **Enabled**
- **Enabled**
- **Enabled**
- **Enabled**
- **Enabled**
- **Enabled**
Cisco Unified Communications Manager SIP Phone Level Configuration (7 of 7)

- Auto Call Select: Enabled
- Log Server: Use System Default
- Admission Control: Use System Default
- Wideband Headset UI Control: Enabled
- Wideband Handset UI Control: Enabled
- Wideband Headset: Enabled
- Wideband Handset: Disabled
- Peer Firmware Sharing: Disabled
- Cisco Discovery Protocol (CDP): Switch Port: Enabled
- Cisco Discovery Protocol (CDP): PC Port: Enabled
- Link Layer Discovery Protocol (LLDP) - Media Endpoint Discover (LLDP-MED): Switch Port: Enabled
- Link Layer Discovery Protocol (LLDP): PC Port: Enabled
- LLDP Asset ID: Unknown
- LLDP Power Priority: Normal
- Uplink Refresh Rate: Normal

* indicates required items.
** Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.
*** Note: Security Profile Contains Addition CAPF Settings.
**** Note: A new softkey template without supplementary service softkeys must be used for a protected phone.
Cisco Unified Communications Manager SIP Phone Directory Number (Ext 4207) Level Configuration (1 of 5)
Cisco Unified Communications Manager SIP Phone Directory Number (Ext 4207) Level Configuration (2 of 5)

![Cisco Unified Communications Manager interface](image)

**AAR Settings**

<table>
<thead>
<tr>
<th>AAR</th>
<th>Voice Mail</th>
<th>AAR Destination Mask</th>
<th>AAR Group</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>40852</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

- Retain this destination in the call forwarding history

**Call Forward and Call Pickup Settings**

<table>
<thead>
<tr>
<th>Voice Mail</th>
<th>Destination</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>Forward All:</td>
<td>2229</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Secondary Call Search Space for Forward All</td>
<td>2229</td>
<td>phones</td>
</tr>
<tr>
<td>Forward Busy Internal</td>
<td>2229</td>
<td>phones</td>
</tr>
<tr>
<td>Forward Busy External</td>
<td>2229</td>
<td>phones</td>
</tr>
<tr>
<td>Forward No Answer Internal</td>
<td>2229</td>
<td>phones</td>
</tr>
<tr>
<td>Forward No Answer External</td>
<td>2229</td>
<td>phones</td>
</tr>
<tr>
<td>Forward No Coverage Internal</td>
<td>2229</td>
<td>phones</td>
</tr>
<tr>
<td>Forward No Coverage External</td>
<td>2229</td>
<td>phones</td>
</tr>
<tr>
<td>Forward on CTI Failure</td>
<td>2229</td>
<td>phones</td>
</tr>
<tr>
<td>Forward Unregistered Internal</td>
<td>2229</td>
<td>phones</td>
</tr>
<tr>
<td>Forward Unregistered External</td>
<td>2229</td>
<td>phones</td>
</tr>
</tbody>
</table>
Cisco Unified Communications Manager SIP Phone Directory Number (Ext 4207) Level Configuration (3 of 5)
Cisco Unified Communications Manager SIP Phone Directory Number (Ext 4207) Level Configuration (4 of 5)

### Directory Number Configuration

<table>
<thead>
<tr>
<th>Visual Message Waiting Indicator Policy</th>
<th>Use System Policy</th>
</tr>
</thead>
<tbody>
<tr>
<td>Audible Message Waiting Indicator Policy</td>
<td>Default</td>
</tr>
<tr>
<td>Ring Setting (Phone Idle)</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Ring Setting (Phone Active)</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Call Pickup Group Audio Alert Setting (Phone Idle)</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Call Pickup Group Audio Alert Setting (Phone Active)</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Recording Option</td>
<td>Call Recording Disabled</td>
</tr>
<tr>
<td>Recording Profile</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Monitoring Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

### Multiple Call/Call Waiting Settings on Device SEP001528BF36B1

Note: The range to select the Max Number of calls is: 1-50
Maximum Number of Calls: 4
Busy Trigger: 2 (Less than or equal to Max. Calls)

### Forwarded Call Information Display on Device SEP001528BF36B1

- [ ] Caller Name
- [ ] Caller Number
- [ ] Redirected Number
- [ ] Dialed Number
Cisco Unified Communications Manager SIP Phone Directory Number (Ext 4207) Level Configuration (4 of 5)
Cisco Unified Communications Manager Route Pattern (23XX) to Nortel PBX extensions Configuration (1 of 2)

**Cisco Unified CM Administration**

- **Route Pattern**: 23XX
- **Route Partition**: < None >
- **Description**: Route to Nortel via CUBE (SIP)
- **Numbering Plan**: Not Selected
- **Route Option**: Route this pattern
- **Call Classification**: OffNet
- **Gateway/Route List**: SIP_G729_CUBE
- **Calling Party Transformations**: Use Calling Party's External Phone Number Mask

**SIP Trunk to Nortel via CUBE**
Acronyms

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definitions</th>
</tr>
</thead>
<tbody>
<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>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>CUCM</td>
<td>Cisco Unified Communications Manager</td>
</tr>
<tr>
<td>DNS</td>
<td>Domain Name Server</td>
</tr>
<tr>
<td>FQDN</td>
<td>Fully Qualified Domain Name</td>
</tr>
<tr>
<td>MWI</td>
<td>Message Waiting Indicator</td>
</tr>
<tr>
<td>PSTN</td>
<td>Public Switched Telephone Network</td>
</tr>
<tr>
<td>SIP</td>
<td>Session Initiated Protocol</td>
</tr>
</tbody>
</table>

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<th>Americas Headquarters</th>
<th>Asia Pacific Headquarters</th>
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<tr>
<td>170 West Tasman Drive</td>
<td>BV</td>
<td>170 West Tasman Drive</td>
<td>Capital Tower</td>
</tr>
<tr>
<td>San Jose, CA 95134-1706</td>
<td>Haarlerbergpark</td>
<td>San Jose, CA 95134-1706</td>
<td>168 Robinson Road</td>
</tr>
<tr>
<td>USA</td>
<td>Haarlerbergweg 13-19</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>1101 CH Amsterdam</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>The Netherlands</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>www-europe.cisco.com</td>
<td>Fax: 408 527-0883</td>
<td>Tel: +65 317 7777</td>
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<tr>
<td></td>
<td>Tel: 31 0 20 357 1000</td>
<td></td>
<td>Fax: +65 317 7799</td>
</tr>
<tr>
<td></td>
<td>Fax: 31 0 20 357 1100</td>
<td></td>
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</tr>
</tbody>
</table>

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