



Nortel CS1000E Release 7.0 using SIP trunk to Cisco Unified Communications Manager Release 9.0

November 27, 2012 – Initial revision

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Introduction

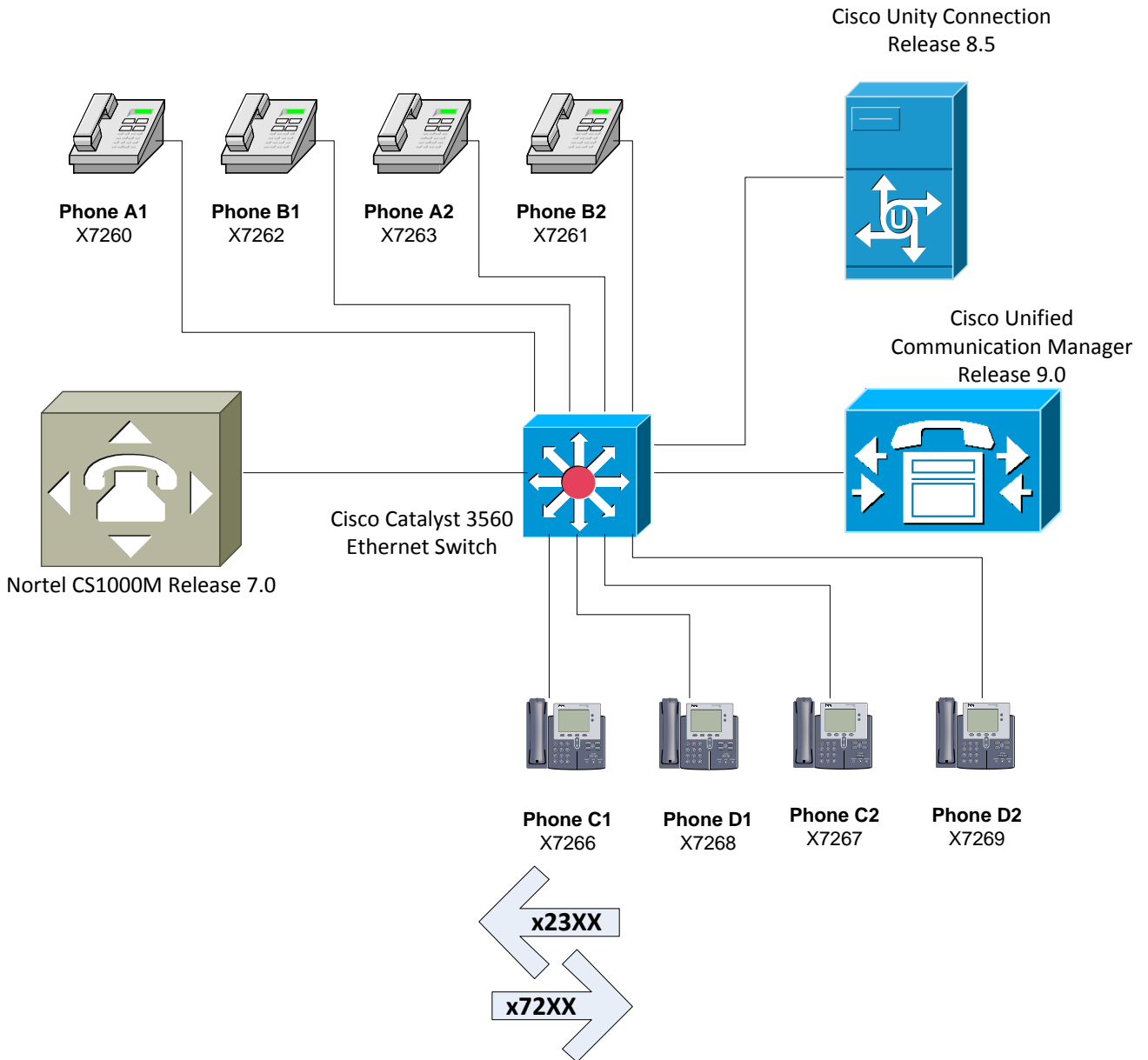
This document describes the steps and configurations necessary for Cisco Unified Communications Manager (Cisco UCM) release 9.0 to interoperate with the Nortel CS1000E using SIP Early-Offer.

The following items were tested:

- Basic call between the two systems and verification of voice path, using both IP phones and digital phones on the Nortel side, and SIP and SCCP IP phones on the Cisco side.
- CLIP/CLIR/CNIP/CNIR features: calling party name and number delivery (allowed and restricted).
- COLP/CONP/COLR/CONR features: connected name and number delivery (allowed and restricted).
- Call transfer: attended, and early attended.
- Call forwarding: call forward all (CFA), call forward busy (CFB), and call forward no answer (CFNA).
- Hold and resume with music on hold.
- Three-way conferencing.
- Voice messaging and MWI activation-deactivation.
- DTMF-relay via RFC2833.

Network Topology

Figure 1. Network Topology/Test bed Setup





Caveats and Limitations

These are the known caveats, limitations or integration issues:

- Basic calls worked from Cisco UCM to Nortel CS1000E and vice versa. The Nortel CS1000E only supports early offer to set its media attribute to send/receive mode. Thus, for calls from Cisco UCM to Nortel CS1000E, the Cisco UCM must be set to send SIP Invite with SDP. This will ensure two-way audio once the call is connected.
- CLIR/CNIR - Restriction of calling number on Nortel CS1000 Unistm phones is achieved by configuring the Nortel station's class of service. Setting the class of service (CLS) to DDGD sets the SIP P-Asserted Identity setting to privacy = id. This restricts the calling number information. Setting the class of service to NAMD sets the SIP P-Asserted Identity setting to privacy = user. Restriction of calling name and number on the Cisco UCM can be done on the Route Pattern or SIP Trunk page. Calling name and number restrictions are honored by both sides.
- COLR/CONR - as with calling name and number presentation restrictions, the Nortel CS1000 restricts connected name and number by configuring the Nortel station's class of service. The station CLS is set to DDGD and NAMD to restrict connected number and name respectively. When a Nortel phone is configured to restrict connected name and number, it was observed that the SIP response to Cisco UCM only sets the privacy=user. However, the Cisco UCM only recognizes privacy=id to restrict presentation of both connected name and number. Thus, Cisco UCM does not honor the restriction of both connected name and number intended by the Nortel side. On the other hand, Cisco UCM restricts the connected name and number information it sends in its SIP response by setting the SIP PRIVACY to "id" on the SIP trunk configuration page. With this setting, the SIP P-Asserted Identity setting within the SIP Response message back to Nortel has the privacy set to "id" only. This results in the Nortel phone restricting the presentation of the connected number only. The connected name is still presented.
- Alerting Name – Although the Cisco UCM sends P-Asserted Identity (PAI) header with the alerting name, this information is not displayed by the Nortel phone.
- Both systems support call forwarding and call transfer features. There are some call forward and transfer scenarios where the calling name and number and/or connected name and number are not updated after the call has been transferred or forwarded. This issue is found primarily when a Nortel phone is the forwarding or transferring party to a Cisco phone via the SIP trunk.
- The Nortel PBX uses the History-Info field to send redirecting number information, while the Cisco UCM uses the Diversion header. This affects how calls are treated when redirected to a voice mail system over an SIP trunk. Since release 8.5, Cisco UCM provides the ability to translate either Diversion headers into History-Info headers or History-Info headers to Diversion headers via SIP Normalization Script. Please refer to the configuration section of this document for more details on the actual normalization script used for this testing.
- For integration where Cisco Unity is the centralized voice messaging system, a SIP normalization script is required to enable/disable MWI on Nortel phones. Please refer to the configuration section of this document for more details on the actual normalization script used for this testing.
- When using G.729 codec between Nortel CS1000 and Cisco UCM, it is required to configure a conference bridge (CFB) resource to initiate a three-way conference between G729 media end-points. Please refer to Cisco UCM configuration section for details.



System Components

Hardware Requirements

The following hardware is required:

- Cisco Unified Communications Manager (Cisco UCM) MCS server. MCS-7825-I4 was used for this testing.
- Cisco Unity Connection MCS server. MCS-7835 server was used for this testing
- Catalyst switch. Catalyst 3560 was used for this testing
- Cisco IP phones. Cisco 9971, 7960, 7962, 7971 & 7975 were used for this testing
- Nortel CS1000E
- Nortel IP and Digital station phones. Nortel-2004P2 IP Phones and M2616 series digital station phones were used for this testing.

Software Requirements

The following software is required:

- Cisco Unified Communications Manager release 9.0. (9.0.1.10000-37 was used for this testing)
- Catalyst 3560 release: 12.2(35)SE5
- Cisco Unity Connection release 8.5
- Nortel CS1000E release 7.0



Features

This section lists supported and unsupported features. Please see the Limitations section on page 4 for more information.

Features Supported

- CLIP—calling line (number) identification presentation.
- CLIR—calling line (number) identification restriction. (Refer to Caveats and Limitations Section)
- CNIP—calling name identification presentation.
- CNIR—calling name identification restriction. (Refer to Caveats and Limitations Section)
- Alerting name. (Refer to Caveats and Limitations Section)
- Attended call transfer. (Refer to Caveats and Limitations Section)
- Early attended call transfer. (Refer to Caveats and Limitations Section)
- CFU—call forwarding unconditional. (Refer to Caveats and Limitations Section)
- CFB—call forwarding busy. (Refer to Caveats and Limitations Section)
- CFNA—call forwarding no answer. (Refer to Caveats and Limitations Section)
- COLP—connected line (number) identification presentation.
- COLR—connected line (number) identification restriction. (Refer to Caveats and Limitations Section)
- CONP—connected name identification presentation.
- CONR—connected name identification restriction. (Refer to Caveats and Limitations Section)
- Hold and resume.
- Conference call. (Refer to Caveats and Limitations Section)
- DTMF-relay using RFC2833.

Features Not Supported

- Call completion (callback, automatic callback).



Configuration

Configuration Sequence and Tasks

Configuring the Nortel PBX

Configuring the Nortel Communication Server 1000 PBX

1. LD 17—Configure the IP D-channel (signaling channel) between the call server and the signaling server.
2. LD 14—Configure the SIP virtual trunks to the signal.
3. LD 11—Configure for the virtual lines for the Nortel IP phone (i200x series).
4. LD 16—Configure the SIP route.
5. LD 86—Configure the route list block for the virtual trunk route.
6. LD 87—Configure CDP steering codes.
7. LD 21—List trunk member.

Signaling Server Setup via the Nortel Node Summary

1. Configure a new IP telephony node summary.
2. Configure the VGW and IP phone codec profile section.
3. Configure the quality of service (QoS) section.
4. Configure the LAN configuration section.
5. Configure the SIP GW setting section.
6. Configure the SIP URI map section.
7. Configure the card section for the MC-32 VGMC card section.
8. Configure the signaling server section.

Network Routing Server

1. Configure the system-wide settings.
2. Configure the NRS server settings.
3. Configure a service domain.
4. Configure an L1 domain (UDP).
5. Configure an L0 domain (CDP).
6. Configure a gateway endpoint gateway.
7. Configure the routing entries.

Cisco Unified Communications Manager (Cisco UCM):

1. Cisco UCM software release.
2. Cisco UCM Regional configuration.
3. Cisco UCM Device Pool configuration.
4. Media Resource Group and Media Resource Group List.
5. Cisco UCM Media Termination Point.
6. SIP trunk security profile.
7. Device setting SIP profile.
8. SIP trunk to the Nortel CS1000E PBX.
9. SIP and SCCP phones device configuration.
10. Route pattern to the Nortel CS1000E PBX.
11. Cisco UCM Service Parameter “Duplex Streaming Enabled” set to “True”.
12. SIP Trunk Normalization Script



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 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
OVLN NO
OVLS NO
```

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

```
>LD 20
PT0000
REQ: PRT
TYPE: TNB
TN 020 0 3 7
DATE
PAGE
DES
DES SIP TRUNK
TN 020 0 03 07 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 8
CHID 8
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
```

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

Phone A1 (i2004)

```
>ld 11
SL1000
MEM AVAIL: (U/P): 99165853 USED U P: 5049370 54598 TOT: 104269821
DISK SPACE NEEDED: 47 KBYTES
DIGITAL TELEPHONES AVAIL: 3 USED: 5 TOT: 8
IP USERS AVAIL: 18 USED: 6 TOT: 24
BASIC IP USERS AVAIL: 7 USED: 1 TOT: 8
TEMPORARY IP USERS AVAIL: 0 USED: 0 TOT: 0
```




ACD AGENTS AVAIL: 10 USED: 0 TOT: 10
PCA AVAIL: 4 USED: 1 TOT: 5
AST AVAIL: 4 USED: 1 TOT: 5
SIP CONVERGED DESKTOPS AVAIL: 0 USED: 0 TOT: 0
SIP CTI TR87 AVAIL: 4 USED: 1 TOT: 5
TNS AVAIL: 32546 USED: 214 TOT: 32760
DATA PORTS AVAIL: 32760 USED: 0 TOT: 32760
REQ: prt
TYPE: tnb
TN 020 0 0 03
DATE
PAGE
DES
DES Phone A1
TN 020 0 00 03 VIRTUAL
TYPE 2004P2
CDEN 8D
CTYP XDLC
CUST 0
NUID
NHTN
CFG_ZONE 000
CUR_ZONE 000
ERL 0
ECL 0
FDN 2327
TGAR 0
LDN NO
NCOS 0
SGRP 0
RNPG 0
SCI 0
SSU
XLST
CLS CTD FBA WTA LPR MTD FNA HTA TDD HFD CRPD
MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
POD DSX VMD SLKD CCSD SWD LND CNDD
CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBF
ICDD CDMD LLCN MCTD CLBD AUTU
GPUD DPUD DNDD CFXD 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 T87A KEM2 MSNV FRA PKCH
CPND LANG ENG
RCO 0
HUNT 2327
LHK 0
PLEV 02
DANI NO
AST 00
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG_ENG
DNDR 0
KEY 00 SCR 2326 0 MARP
CPND
NAME Phone_A1
XPLN 13
DISPLAY_FMT FIRST, LAST
01
02
03
04



05
06
07
08
09
10
11
12
13
14
15
16
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 29 SEP 200

Phone A2 (2616):

REQ: prt
TYPE: 2616
TN 0 0 7 2
DATE
PAGE
DES
DES Phone A2
TN 000 0 07 02 VIRTUAL
TYPE 2616
CDEN 8D
CTYP XDLC
CUST 0
AOM 0
ERL 0
FDN 2328
TGAR 1
LDN NO
NCOS 0
SGRP 0
RNPG 0
SCI 0
SSU
XLST
CLS CTD FBA WTA LPR MTD FNA HTA ADD HFD
MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
POD DSX VMD SLKD CCSD SWD LND CNDD
CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBD
ICDD CDMD LLCN MCTD CLBD AUTU
GPUD DPUD DNDD 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
CDMR MCDD T87D PKCH
CPND LANG ENG
RCO 0
HUNT 2326



```
LHK 0
PLEV 02
DANI NO
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
DNDR_0
KEY 00 SCR 2332 0 MARP
CPND
NAME Phone_A2
XPLN 13
DISPLAY_FMT FIRST, LAST
01
02
03 CFW 4 2500
04 AO6
05 TRN
06
07
08
09
10
11
12
13
14
15
DATE 29 SEP 2008
NACT
```

LD 16 – Configure the SIP route

```
>LD 21
PT1000
REQ: PRT
TYPE: RDB
CUST 0
ROUT 30
TYPE RDB
CUST 00
ROUT 30
DES SIP_TRUNK
TKTP TIE
M911P NO
ESN NO
CNVT NO
SAT NO
RCLS EXT
VTRK YES
ZONE 000
PCID SIP
CRID YES
NODE 104
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
PTYT ATT
AUTO NO
DNIS NO
DCDR NO
ICOG IAO
SRCH LIN
TRMB YES
STEP
ACOD 430
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
TFD 0
PAGE 002
VSS 0
VGD 6
SST 5 0
NEDC ORG
FEDC ORG
CPDC NO
DLTN NO
HOLD 02 02 40
SEIZ 02 02
SVFL 02 02
DRNG NO
CDR NO
VRAT NO
MUS NO
MANO NO
OHQ NO
OHQT 00
CBQ NO
AUTH NO
TDET NO
TTBL 0
ATAN NO
OHTD NO
PLEV 2
ALRM NO
ART 0
SGRP 0
ARDN NO
AACR NO



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

```
>LD 86
ESN000
MEM AVAIL: (U/P): 99165853 USED U P: 5049370 54598 TOT: 104269821
DISK SPACE NEEDED: 47 KBYTES
REQ PRT
CUST 0
FEAT RLB
RLI 3
RLI 3
ENTR 0
LTER NO
ROUT 30
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
DORG NO
SBOC NRR
IDBB DBD
IOHQ NO
OHQ NO
CBQ NO
ISET 0
NALT 5
MFRL 0
OVLL 0
MEM AVAIL: (U/P): 99165853 USED U P: 5049370 54598 TOT: 104269821
DISK SPACE NEEDED: 47 KBYTES
```

LD 87 – Configure CDP steering codes

```
>LD 87
ESN000
MEM AVAIL: (U/P): 99165853 USED U P: 5049370 54598 TOT: 104269821
DISK SPACE NEEDED: 47 KBYTES
REQ PRT
CUST 0
FEAT CDP
TYPE DSC
DSC 37
DSC 37
FLEN 0
DSP LSC
RLI 3
NPA
NXX
```

LD 21 – List Trunk Members

```
>LD 21
PT1000
REQ: LTM
CUST 0
ROUT 30
TYPE TLST
TKTP TIE
ROUT 30
DES SIP TRUNK
TN 020 0 03 00 MBER 1 SIP_TRUNK
TN 020 0 03 01 MBER 2 SIP_TRUNK
TN 020 0 03 02 MBER 3 SIP_TRUNK
```



```
TN 020 0 03 03 MBER 4 SIP_TRUNK
TN 020 0 03 04 MBER 5 SIP_TRUNK
TN 020 0 03 05 MBER 6 SIP_TRUNK
TN 020 0 03 06 MBER 7 SIP_TRUNK
TN 020 0 03 07 MBER 8 SIP_TRUNK
```

Nortel CS1000E Element Manager

The screenshot shows the Nortel Unified Communications Management interface. The top navigation bar includes the Nortel logo, the title "UNIFIED COMMUNICATIONS MANAGEMENT", and links for "Help" and "Logout". The main content area is titled "Elements" and displays a table of registered elements. The table has columns for "Element Name", "Element Type", "Release", "Address", and "Description".

	Element Name	Element Type	Release	Address	Description
1	EM on cs104ss1	CS1000	7.0	172.30.13.101	New element.
2	172.30.13.101	Call Server	7.0	172.30.13.101	New element.
3	cs104ss2.pbxlab.org (backup)	Linux Base	6.0	172.30.11.107	Base OS element.
4	cs104ss1.pbxlab.org (primary)	Linux Base	7.0	172.30.11.103	Base OS element.
5	172.30.13.105	Media Gateway Controller	7.0	172.30.13.105	New element.
6	NRSM on cs104ss1	Network Routing Service	7.0	172.30.13.103	New



Element Manager-System Overview

- UCM Network Services
- Home
- Links
 - Virtual Terminals
- System
 - + Alarms
 - Maintenance
 - + Core Equipment
 - Peripheral Equipment
 - + IP Network
 - + Interfaces
 - Engineered Values
 - + Emergency Services
 - + Geographic Redundancy
 - + Software
- Customers
- Routes and Trunks
 - Routes and Trunks
 - D-Channels
 - Digital Trunk Interface
- Dialing and Numbering Plans
 - Electronic Switched Network
 - Flexible Code Restriction
 - Incoming Digit Translation
- Phones
 - Templates
 - Reports
 - Views
 - Lists
 - Properties
 - Migration
- Tools
 - + Backup and Restore
 - Call Server Initialization
 - Date and Time
 - + Logs and reports
- Security

Managing: **172.30.13.101** Username: admin
System Overview

System Overview

IP Address: 172.30.13.101
Type: Nortel Communication Server 1000E CPPM
Version: 4021
Release: 700 Q +

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Element Manager - Node Details

- UCM Network Services
- Home
- Links
- Virtual Terminals
- System
 - + Alarms
 - Maintenance
 - + Core Equipment
 - Peripheral Equipment
 - IP Network
 - [Nodes: Servers, Media Cards](#)
 - Maintenance and Reports
 - Media Gateways
 - Zones
 - Host and Route Tables
 - Network Address Translation
 - QoS Thresholds
 - Personal Directories
 - Unicode Name Directory
 - + Interfaces
 - Engineered Values
 - + Emergency Services
 - + Geographic Redundancy
 - + Software
- Customers
- Routes and Trunks
 - Routes and Trunks
 - D-Channels
 - Digital Trunk Interface
- Dialing and Numbering Plans
 - Electronic Switched Network
 - Flexible Code Restriction
 - Incoming Digit Translation
- Phones
 - Templates
 - Reports

Node Details (ID: 104 - LTPS, Gateway (SIPGw, H323Gw))

Node ID: * (0-9999)

Call server IP address: * TLAN address type: IPv4 only
 IPv4 and IPv6

Embedded LAN (ELAN) **Telephony LAN (TLAN)**

Gateway IP address: * Node IPv4 address: *

Subnet mask: * Subnet mask: *

Node IPv6 address:

* Required Value.

Associated Signaling Servers & Cards

Select to add ▼ [Print](#) | [Refresh](#)

☐ Hostname ▲	Type	Deployed Applications	ELAN IP	TLAN IPv4	Role
☐ cs104ss1	Signaling_Server	LTPS, Gateway, PD, Presence Publisher, IP Media Services	172.30.13.103	172.30.11.103	Leader

Show: IPv6 address

Note: Only server(s) that are not part of any other IP telephony node and deployed application(s) that match the service(s) selected for this node are available in the servers list

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Element Manager - Voice Gateway and Codec

Managing: 172.30.13.101 Username: admin
System » IP Network » IP Telephony Nodes » Node Details » VGW and Codecs

Node ID: 104 - Voice Gateway (VGW) and Codecs

General | Voice Codecs | Fax

General

Echo cancellation: Use canceller, with tail delay: 128 ▾
 Dynamic attenuation

Voice activity detection threshold: -17 (-20 - +10 DBM)
Idle noise level: -65 (-327 - +327 DBM)

Signaling options: DTMF tone detection
 Low latency mode
 Remove DTMF delay (squelch DTMF from TDM to IP)
 Modem/Fax pass-through
 V.21 Fax tone detection
 R factor calculation

Voice Codecs

Codec G711: Enabled (required)
Voice payload size: 20 (milliseconds per frame)
Voice playout (litter buffer) delay: 40 80 (milliseconds)

* Required Value. Note: Changes made on this page will NOT be transmitted until the Node is also saved.

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Managing: 172.30.13.101 Username: admin
System » IP Network » IP Telephony Nodes » Node Details » VGW and Codecs

Node ID: 104 - Voice Gateway (VGW) and Codecs

General | Voice Codecs | Fax

Codec G723.1: Enabled

Voice payload size: 30 (milliseconds per frame)

Voice playout (jitter buffer) delay: 60 120 (milliseconds)

Nominal Maximum
Maximum delay may be automatically adjusted based on nominal settings.

Coding rate: 5.3 (kbps)

Fax

Codec name: T.38 FAX

Maximum rate: 14400 (bps)

Fax TCF method: 2

Fax playout nominal delay: 80 (0 - 300 milliseconds)

FAX no activity timeout: 20 (10 - 32000 milliseconds)

Packet size: 30 (bps)

* Required Value. Note: Changes made on this page will NOT be transmitted until the Node is also saved.

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- UCM Network Services
- Home
- Links
 - Virtual Terminals
- System
 - + Alarms
 - Maintenance
 - + Core Equipment
 - Peripheral Equipment
 - IP Network
 - Nodes: Servers, Media Cards
 - Maintenance and Reports
 - Media Gateways
 - Zones
 - Host and Route Tables
 - Network Address Translation
 - QoS Thresholds
 - Personal Directories
 - Unicode Name Directory
 - + Interfaces
 - Engineered Values
 - + Emergency Services
 - + Geographic Redundancy
 - + Software
- Customers
- Routes and Trunks
 - Routes and Trunks
 - D-Channels
 - Digital Trunk Interface
- Dialing and Numbering Plans
 - Electronic Switched Network
 - Flexible Code Restriction
 - Incoming Digit Translation
- Phones
 - Templates
 - Reports



Element Manager - Service Domain

«UCM Network Services

- System
 - NRS Server
 - Database
 - System Wide Settings
- Numbering Plans
 - Domains
 - Endpoints
 - Routes
 - Network Post-Translation
 - Collaborative Servers
- Tools
 - SIP Phone Context
 - Routing Tests
 - H.323
 - SIP
 - Backup
 - Restore
 - GK/NRS Data upgrade

Managing: Active database 172.30.13.103
 Standby database [Numbering Plans » Domains » Service Domains](#)

Edit Service Domain

Domain name: *

Domain description:

* Required value. Cancel

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Domain L1

«UCM Network Services

- System
 - NRS Server
 - Database
 - System Wide Settings
- Numbering Plans
 - Domains
 - Endpoints
 - Routes
 - Network Post-Translation
 - Collaborative Servers
- Tools
 - SIP Phone Context
 - Routing Tests
 - H.323
 - SIP
 - Backup
 - Restore
 - GK/NRS Data upgrade

Managing: Active database 172.30.13.103
 Standby database [Numbering Plans » Domains » L1 Domain](#)

Edit L1 Domain (pbxlab.org)

Domain name: *

Domain description:

Endpoint authentication enabled:

Authentication password:

E.164 country code:

E.164 area code:

E.164 international dialing access code:

E.164 international dialing code length: (0-99)

E.164 national dialing access code:

E.164 national dialing code length: (0-99)

E.164 local (subscriber) dialing access code:

E.164 local (subscriber) dialing code length: (0-99)

Private L1 domain (UDP location) dialing access code:

* Required value



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Managing: Active database 172.30.13.103
 Standby database [Numbering Plans » Domains » L1 Domain](#)

Edit L1 Domain (pbxlab.org)

E.164 country code:	<input type="text" value="1"/>
E.164 area code:	<input type="text" value="919"/>
E.164 international dialing access code:	<input type="text" value="011"/>
E.164 international dialing code length:	<input type="text"/> (0-99)
E.164 national dialing access code:	<input type="text" value="9"/>
E.164 national dialing code length:	<input type="text"/> (0-99)
E.164 local (subscriber) dialing access code:	<input type="text" value="9"/>
E.164 local (subscriber) dialing code length:	<input type="text"/> (0-99)
Private L1 domain (UDP location) dialing access code:	<input type="text" value="9"/>
Private L1 domain (UDP location) dialing code length:	<input type="text"/> (0-99)
Special number:	<input type="text"/>
Special number dialing code length:	<input type="text"/> (0-31)
Emergency service access prefix:	<input type="text"/>
Special number label:	<input type="text" value="PrivateSpecial"/>

* Required value



Domain L0

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Managing: Active database 172.30.13.103
 Standby database [Numbering Plans » Domains » L0 Domain](#)

Edit L0 Domain (pbxlab.org / rtp)

Domain name: *

Domain description:

Endpoint authentication enabled:

Authentication password:

E.164 country code:

E.164 area code:

Private unqualified number label:

E.164 international dialing access code:

E.164 international dialing code length: (0-99)

E.164 national dialing access code:

E.164 national dialing code length: (0-99)

E.164 local (subscriber) dialing access code:

E.164 local (subscriber) dialing code length: (0-99)

* Required value.



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Managing: Active database 172.30.13.103
 Standby database [Numbering Plans](#) » [Domains](#) » [L0 Domain](#)

Edit L0 Domain (pbxlab.org / rtp)

E.164 country code:	<input type="text" value="1"/>
E.164 area code:	<input type="text" value="919"/>
Private unqualified number label:	<input type="text" value="PrivateUnknown"/>
E.164 international dialing access code:	<input type="text" value="011"/>
E.164 international dialing code length:	<input type="text"/> (0-99)
E.164 national dialing access code:	<input type="text" value="9"/>
E.164 national dialing code length:	<input type="text"/> (0-99)
E.164 local (subscriber) dialing access code:	<input type="text" value="9"/>
E.164 local (subscriber) dialing code length:	<input type="text"/> (0-99)
Private L1 domain (UDP location) dialing access code:	<input type="text" value="9"/>
Private L1 domain (UDP location) dialing code length:	<input type="text"/> (0-99)
Special number:	<input type="text"/>
Special number dialing code length:	<input type="text"/> (0-31)
Emergency service access prefix:	<input type="text"/>

* Required value.



Network Routing Services (NRS).

System Wide Setting

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Managing: 172.30.13.103

System » System Wide Settings

System Wide Settings

SIP registration time to live timer:	<input type="text" value="300"/>	(30-3600 Seconds)
H.323 gatekeeper registration time to live timer:	<input type="text" value="30"/>	(30-3600 Seconds)
H.323 alias name:	<input type="text" value="H323NRS104"/>	*
Auto backup time:	<input type="text" value="23:49"/>	(HH:MM)
Auto backup to secure FTP site enabled:	<input type="checkbox"/>	
Auto backup to secure FTP site's IP address:	<input type="text"/>	
Auto backup secure FTP site's path:	<input type="text"/>	
Auto backup secure FTP user name:	<input type="text"/>	
Auto backup secure FTP password:	<input type="text"/>	
Call Server Type:	<input type="text" value="CS1000"/>	

* Required value.

Save Cancel

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NRS – Server

System » NRS Server

NRS Server

Service Status

<input type="checkbox"/>	Service Name	Service Status
1 <input type="checkbox"/>	SIP Proxy Server (SPS)	In service
2 <input type="checkbox"/>	Gatekeeper (GK)	In service
3 <input type="checkbox"/>	Network Connection Server (NCS)	In service

Server Configuration

NRS Setting

Host name cs104ss1.pbxlab.org
Address type IPv4 only
Primary TLAN IPv4 address 172.30.11.103
Secondary TLAN IPv4 address 0.0.0.0
Secondary server host name SecondaryHostName
Control priority 40
Server mate communication port 5005
Realm name realmName
Server role Primary

H.323 Gatekeeper Settings

Location request (LRQ) response timeout 3



NRS- Gateway End Points

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Managing: Active database 172.30.13.103
 Standby database [Numbering Plans » Endpoints » Gateway Endpoint](#)

Edit Gateway Endpoint pbxlab.org / rtp / interop)

End point name: *

Description:

Trust Node:

Tandem gateway endpoint name:

Endpoint authentication enabled:

Authentication password:

E.164 country code:

E.164 area code:

E.164 international dialing access code:

E.164 international dialing code length: (0-99)

E.164 national dialing access code:

E.164 national dialing code length: (0-99)

E.164 local (subscriber) dialing access code:

* Required value



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Managing: Active database 172.30.13.103
 Standby database [Numbering Plans](#) » [Endpoints](#) » [Gateway Endpoint](#)

Edit Gateway Endpoint pbxlab.org / rtp / interop)

E.164 local (subscriber) dialing access code:

E.164 local (subscriber) dialing code length: (0-99)

Private L1 domain (UDP location) dialing access code:

Private L1 domain (UDP location) dialing code length: (0-99)

Private Special number 1:

Private Special number 1 dialing code length: (0-31)

Private Special number 2:

Private Special number 2 dialing code length: (0-31)

Static endpoint address type: IP version 4 ▾

Static endpoint address:

H.323 support: Not RAS H.323 endpoint ▾

SIP support: Static SIP endpoint ▾

SIP mode: Proxy Mode
 Redirect Mode

* Required value

Save Cancel



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Managing: Active database 172.30.13.103
 Standby database [Numbering Plans » Endpoints » Gateway Endpoint](#)

Edit Gateway Endpoint pbxlab.org / rtp / interop)

SIP TCP transport enabled:
SIP TCP port: 5060

SIP UDP transport enabled:
SIP UDP port: 5060

SIP TLS transport enabled:
SIP TLS port: 5061

Persistent TCP support enabled:
End to end security support:
Network Connection Server enabled:

Redundancy enabled: Not Configured
Main endpoint name: Not Applicable
Redundant endpoint name: Not Applicable

Virtual Private Networks Identifier: (1-16383)
Bandwidth Zone: (0-8000)

User Parameter(s):

* Required value

Save Cancel



NRS Routes

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Managing: Active database 172.30.13.103
 Standby database [Numbering Plans](#) » [Routes](#) » [Routing Entry](#)

Edit Routing Entry (pbxlab.org / rtp / interop / CM_Taurus)

DN type: Private level 0 regional (CDP steering code) ▾

DN prefix: 7266 *

Route cost: 1 * (1-255)

* Required value.

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Managing: Active database 172.30.13.103
 Standby database [Numbering Plans](#) » [Routes](#) » [Routing Entry](#)

Edit Routing Entry (pbxlab.org / rtp / interop / CM_Taurus)

DN type:	E.164 national	▼
DN prefix:	1408	*
Route cost:	1	* (1-255)

* Required value.

Save Cancel



Configuring the Cisco Unified Communications Manager

Cisco Unified Communications Manager –Software Release

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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Cisco Unified CM Administration

System version: 9.0.1.10000-37

Last Successful Logon: Monday, November 5, 2012 3:37:01 PM PST

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A summary of U.S. laws governing Cisco cryptographic products may be found at our [Export Compliance Product Report](#) web site.

For information about Cisco Unified Communications Manager please visit our [Unified Communications System Documentation](#) web site.

For Cisco Technical Support please visit our [Technical Support](#) web site.



Cisco Unified Communication Manager – Service Parameter setting for Duplex Streaming

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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Service Parameter Configuration

Related Links: Parameters for All Servers

Clusterwide Parameters (Hunt List)

Stop Hunting on Out of Bandwidth Flag *	False	False
Use Pickup Group Of Line Group Member DN *	False	False

Clusterwide Parameters (Service)

Default Network Hold MOH Audio Source ID *	1	1
Default User Hold MOH Audio Source ID *	1	1
Duplex Streaming Enabled *	True	False
Media Exchange Interface Capability Timer *	8	8
Send Multicast MOH in H.245 OLC Message *	True	True
Media Exchange Timer *	12	12
Media Exchange Stop Streaming Timer *	8	8
Open Video Channel Response Timer for SIP Interop *	500	500
Port Received Timer After Call Connection *	500	500

x Find: dupl Next Previous Highlight all Match case

Done



Cisco Unified Communication Manager – Audio Codec Preference Setting

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For Cisco Unified Communications Solutions

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Audio Codec Preference List Configuration

Related Links:

Save Delete Copy Add New

Status: Ready

Audio Codec Preference List Information

Name*

Description*

Codecs in List*

- G.729 8k
- G.729b 8k
- G.711 U-Law 64k
- AMR-WB (7K-24K)
- AMR (5k-13k)
- MP4A-LATM 128k
- AAC-LD (MP4A Generic)
- MP4A-LATM 64k
- MP4A-LATM 56k
- L16 256k
- MP4A-LATM 48k
- ISAC 32k
- MP4A-LATM 32k
- G.722 64k



Cisco Unified Communications Manager –Regional Configuration

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Find and List Regions

Status
 2 records found

Regions (1 - 2 of 2)

Rows per Page 50

Find Regions where Name begins with

<input type="checkbox"/>	Name ^
<input type="checkbox"/>	Default
<input type="checkbox"/>	Phones



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Region Configuration Related Links:

Save Reset

Region Information

Name* Default

Region Relationships

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls
Default	Use System Default (Factory Default low loss)	64 kbps (G.722, G.711)	384
Phones	G729_ Factory Default low loss	64 kbps (G.722, G.711)	384
NOTE: Regions not displayed	Use System Default	Use System Default	Use System Default

Modify Relationship to other Regions

Regions	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls
Default Phones	Keep Current Setting	Keep Current Setting	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None

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Region Configuration Related Links:

Save Reset

Region Information

Name* Phones

Region Relationships

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls
Default	G729_ Factory Default low loss	64 kbps (G.722, G.711)	384
Phones	Use System Default (Factory Default low loss)	64 kbps (G.722, G.711)	384
NOTE: Regions not displayed	Use System Default	Use System Default	Use System Default

Modify Relationship to other Regions

Regions	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls
Default Phones	Keep Current Setting	Keep Current Setting	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None



Cisco Unified Communications Manager – Device Pool Configuration

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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Find and List Device Pools

Add New

Status
2 records found

Device Pool (1 - 2 of 2)

Rows per Page 50

Find Device Pool where Device Pool Name begins with

<input type="checkbox"/>	Name	Cisco Unified CM Group	Region	Date/Time Group	Copy
<input type="checkbox"/>	Default	Default	Default	CMLocal	<input type="button" value="Copy"/>
<input type="checkbox"/>	Phones	Default	Phones	CMLocal	<input type="button" value="Copy"/>



Cisco Unified Communications Manager – Device Pool Configuration (Default)

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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Device Pool Configuration Related Links:

Status

Status: Ready

Device Pool Information

Device Pool: Default (18 members**)

Device Pool Settings

Device Pool Name*	<input type="text" value="Default"/>
Cisco Unified Communications Manager Group*	<input type="text" value="Default"/>
Calling Search Space for Auto-registration	<input type="text" value="< None >"/>
Adjunct CSS	<input type="text" value="< None >"/>
Reverted Call Focus Priority	<input type="text" value="Default"/>
Local Route Group	<input type="text" value="< None >"/>
Intercompany Media Services Enrolled Group	<input type="text" value="< None >"/>

Roaming Sensitive Settings

Date/Time Group*	<input type="text" value="CMLocal"/>
Region*	<input type="text" value="Default"/>
Media Resource Group List	<input type="text" value="< None >"/>
Location	<input type="text" value="< None >"/>
Network Locale	<input type="text" value="< None >"/>
SRST Reference*	<input type="text" value="Disable"/>
Connection Monitor Duration***	<input type="text"/>
Single Button Barge*	<input type="text" value="Default"/>
Join Across Lines*	<input type="text" value="Default"/>
Physical Location	<input type="text" value="< None >"/>
Device Mobility Group	<input type="text" value="< None >"/>



Device Pool Configuration

Related Links:

Device Mobility Related Information****

Device Mobility Calling Search Space
 AAR Calling Search Space
 AAR Group
 Calling Party Transformation CSS
 Called Party Transformation CSS

Geolocation Configuration

Geolocation
 Geolocation Filter

Call Routing Information

Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	<input type="text" value="Default"/>	<input type="text"/>	<input type="text" value=" < None >"/>
International Number	<input type="text" value="Default"/>	<input type="text"/>	<input type="text" value=" < None >"/>
Unknown Number	<input type="text" value="Default"/>	<input type="text"/>	<input type="text" value=" < None >"/>
Subscriber Number	<input type="text" value="Default"/>	<input type="text"/>	<input type="text" value=" < None >"/>

Incoming Called Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value=" < None >"/>
International Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value=" < None >"/>
Unknown Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value=" < None >"/>
Subscriber Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value=" < None >"/>

Phone Settings

Inbound Call Settings

Calling Party Transformation CSS

Connected Party Settings

Connected Party Transformation CSS

Redirecting Party Settings

Redirecting Party Transformation CSS



Cisco Unified Communications Manager – Device Pool Configuration (Phones)

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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CCMAdministrator | [Search Documentation](#) | [About](#) | [Logout](#)

Device Pool Configuration

Related Links: Back To Find/List Go

Save ✖ Delete Copy Reset Apply Config + Add New

Status

i Status: Ready

Device Pool Information

Device Pool: Phones (5 members**)

Device Pool Settings

Device Pool Name*

Cisco Unified Communications Manager Group* Default ▾

Calling Search Space for Auto-registration < None > ▾

Adjunct CSS < None > ▾

Reverted Call Focus Priority Default ▾

Local Route Group < None > ▾

Intercompany Media Services Enrolled Group < None > ▾

Roaming Sensitive Settings

Date/Time Group* CMLocal ▾

Region* Phones ▾

Media Resource Group List < None > ▾

Location < None > ▾

Network Locale < None > ▾

SRST Reference* Disable ▾

Connection Monitor Duration***

Single Button Barge* Default ▾

Join Across Lines* Default ▾

Physical Location < None > ▾

Device Mobility Group < None > ▾

Device Mobility Related Information****

Device Mobility Calling Search Space < None > ▾

AAR Calling Search Space < None > ▾

AAR Group < None > ▾

Calling Party Transformation CSS < None > ▾

Called Party Transformation CSS < None > ▾

Geolocation Configuration

Geolocation < None > ▾

Geolocation Filter < None > ▾

Call Routing Information

Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	<input style="width: 80px;" type="text" value="Default"/>	<input style="width: 40px;" type="text"/>	< None > ▾
International Number	<input style="width: 80px;" type="text" value="Default"/>	<input style="width: 40px;" type="text"/>	< None > ▾
Unknown Number	<input style="width: 80px;" type="text" value="Default"/>	<input style="width: 40px;" type="text"/>	< None > ▾
Subscriber Number	<input style="width: 80px;" type="text" value="Default"/>	<input style="width: 40px;" type="text"/>	< None > ▾



Device Pool Configuration Related Links: [Back To Find/List](#)

Incoming Called Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value=" < None >"/>
International Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value=" < None >"/>
Unknown Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value=" < None >"/>
Subscriber Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value=" < None >"/>

Phone Settings

Inbound Call Settings

Calling Party Transformation CSS

Connected Party Settings

Connected Party Transformation CSS

Redirecting Party Settings

Redirecting Party Transformation CSS



Cisco Unified Communications Manager – Media Resource Group Lists Configuration

The screenshot shows the 'Media Resource Group List Configuration' page in Cisco Unified CM Administration. The page title is 'Media Resource Group List Configuration' and the status is 'Ready'. The configuration details are as follows:

- Media Resource Group List Status:** Media Resource Group List: MRGL_Aries (used by 3 devices)
- Media Resource Group List Information:** Name: MRGL_Taurus
- Media Resource Groups for this List:**
 - Available Media Resource Groups: MRG_Unused
 - Selected Media Resource Groups: MRG_Taurus

Cisco Unified Communications Manager – Media Resource Group Configuration

The screenshot shows the 'Media Resource Group Configuration' page in Cisco Unified CM Administration. The page title is 'Media Resource Group Configuration' and the status is 'Ready'. The configuration details are as follows:

- Media Resource Group Status:** Media Resource Group: MRG_Taurus (used by 3 devices)
- Media Resource Group Information:** Name: MRG_Taurus, Description: (empty)
- Devices for this Group:**
 - Available Media Resources: CFB_2, MTP_2 (MTP)
 - Selected Media Resources: ANN_2 (ANN), CFB000F352F26E9 (CFB), MOH_2 (MOH), MTP000F352F26E9 (XCODE), TMP000F352F26E9 (MTP)
- Use Multi-cast for MOH Audio (If at least one multi-cast MOH resource is available)



Cisco Unified Communications Manager - Media Termination Point Configuration

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Find and List Media Termination Points

Add New

Status
 2 records found

Media Termination Point (1 - 2 of 2) Rows per Page 50 ▾

Find Media Termination Point where Name ▾ begins with ▾

<input type="checkbox"/>	Name ^	Description	Device Pool	Status	IP Address	Copy
<input type="checkbox"/>	MTP_2	MTP_CM	Default	Registered with CM-Aries	172.20.66.254	Not Allowed
<input type="checkbox"/>	TMP000F352F26E9		Default	Registered with CM-Aries	172.20.66.55	



Media Termination Point Configuration

Related Links: Back To Find/List Go

Save Reset Apply Config

Status

Status: Ready

Media Termination Point Information

Registration Registered with Cisco Unified Communications Manager CM-Taurus
IP Address 172.20.43.254
IPv6 Address 0000:0000:0000:0000:0000:0000:0000:0000
Media Termination Point Type* Cisco Media Termination Point Software
Host Server* CM-Taurus
Media Termination Point Name* MTP_2
Description MTP_CM-Taurus
Device Pool* Default
 Trusted Relay Point

Save Reset Apply Config



Media Termination Point Configuration

Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

Status

Status: Ready

Media Termination Point Information

Registration Registered with Cisco Unified Communications Manager CM-Taurus
IP Address 172.20.66.55
IPv6 Address 0000:0000:0000:0000:0000:0000:0000:0000
Media Termination Point Type* Cisco IOS Enhanced Software Media Termination Point
Media Termination Point Name* TMP000F352F26E9
Description
Device Pool* Default
 Trusted Relay Point

Save Delete Copy Reset Apply Config Add New



Cisco Unified Communications Manager – Conference Bridge Configuration

Cisco Unified CM Administration
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Conference Bridge Configuration Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

Status
Status: Ready

Conference Bridge Information
Conference Bridge : CFB000F352F26E9 (Conference Bridge on IOS DSP Farm)
Registration Registered with Cisco Unified Communications Manager CM-Taurus
IP Address [172.20.66.55](#)

IOS Conference Bridge Info

Conference Bridge Type*	Cisco IOS Conference Bridge
⚠ Device is not trusted	
Conference Bridge Name*	CFB000F352F26E9
Description	Conference Bridge on IOS DSP Farm
Device Pool*	Default
Common Device Configuration	< None >
Location*	Hub_None
Use Trusted Relay Point*	Default

Save Delete Copy Reset Apply Config Add New

Sample IOS configuration for conference bridge registered to Cisco UCM

```
voice-card 0
dspfarm
dsp services dspfarm
!
sccp local GigabitEthernet0/1
sccp ccm 172.X.X.X identifier 1 version 7.0
sccp

sccp ccm group 1
associate ccm1 priority 1
associate profile 1 register CFB000F352F26E9

dspfarm profile 1 conference
codec g711ulaw
codec g711alaw
codec g729ar8
codec g729abr8
codec g729r8
codec g729br8
codec g722-64
maximum sessions 4
associate application SCCP
```



Cisco Unified Communications Manager – Transcoder Configuration

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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Transcoder Configuration

Related Links:

Transcoder Information

Transcoder: MTP000F352F26E9 (MTP000F352F26E9)
Registration: Registered with Cisco Unified Communications Manager CM-Aries
IP Address: [172.20.66.55](#)
IPv6 Address: 0000:0000:0000:0000:0000:0000:0000:0000

Media Termination Point Hardware Info

Transcoder Type*	Cisco Media Termination Point Hardware
Description	<input type="text" value="MTP000F352F26E9"/>
MAC Address*	<input type="text" value="000F352F26E9"/>
Device Pool*	<input type="text" value="Default"/> View Details
Common Device Configuration	<input type="text" value="< None >"/> View Details
Special Load Information	<input type="text" value=""/> Leave blank to use default

Trusted Relay Point

*- indicates required item.



Cisco Unified Communications Manager – 726(0-5) Route Pattern to Nortel

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Route Pattern Configuration Related Links: Back To Find/List - Go

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Status
Status: Ready

Pattern Definition

Route Pattern*	726[0-5]
Route Partition	< None >
Description	To SIP trunk to Nortel
Numbering Plan	-- Not Selected --
Route Filter	< None >
MLPP Precedence*	Default
<input type="checkbox"/> Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	CS104_Nortel (Edit)
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error

Call Classification* OffNet

Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code

Authorization Level* 0

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*	Default
Calling Name Presentation*	Default
Calling Party Number Type*	Cisco CallManager
Calling Party Numbering Plan*	Cisco CallManager

Connected Party Transformations

Connected Line ID Presentation*	Default
Connected Name Presentation*	Default

Called Party Transformations

Discard Digits	< None >
Called Party Transform Mask	
Prefix Digits (Outgoing Calls)	
Called Party Number Type*	Cisco CallManager
Called Party Numbering Plan*	Cisco CallManager

ISDN Network-Specific Facilities Information Element

Network Service Protocol	-- Not Selected --	
Carrier Identification Code		
Network Service	Service Parameter Name	Service Parameter Value
-- Not Selected --	< Not Exist >	

Save Delete Copy Add New



Cisco Unified Communications Manager – SIP Profile (DO Standard SIP Profile with PRACK)

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For Cisco Unified Communications Solutions

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SIP Profile Configuration Related Links:

Status

- Status: Ready
- All SIP devices using this profile must be restarted before any changes will take affect.

SIP Profile Information

Name*	Early Offer SIP Standard Profile
Description	Default SIP Profile
Default MTP Telephony Event Payload Type*	101
Early Offer for G.Clear Calls*	Disabled
SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*	TIAS and AS
User-Agent and Server header information*	Send Unified CM Version Information as User-Agent ▾
Accept Audio Codec Preferences in Received Offer*	Default
Dial String Interpretation*	Phone number consists of characters 0-9, *, #, and + ▾

Redirect by Application

- Disable Early Media on 180
- Outgoing T.38 INVITE include audio mline
- Enable ANAT
- Require SDP Inactive Exchange for Mid-Call Media Change
- Use Fully Qualified Domain Name in SIP Requests
- Assured Services SIP conformance

Parameters used in Phone

Timer Invite Expires (seconds)*	180
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10



Start Media Port*	16384
Stop Media Port*	32766
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
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*	Off
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
Resource Priority Namespace	< None >
Timer Keep Alive Expires (seconds)*	120

SIP Profile Configuration Related Links:

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Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70
Off Hook To First Digit Timer (milliseconds)*	15000
Call Forward URI*	x-cisco-serviceuri-cfwdall
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial
<input checked="" type="checkbox"/> Conference Join Enabled	
<input type="checkbox"/> RFC 2543 Hold	
<input checked="" type="checkbox"/> Semi Attended Transfer	
<input type="checkbox"/> Enable VAD	
<input type="checkbox"/> Stutter Message Waiting	
<input type="checkbox"/> MLPP User Authorization	

Normalization Script

Normalization Script

Enable Trace

	Parameter Name	Parameter Value	
1	<input type="text"/>	<input type="text"/>	<input type="button" value="+"/> <input type="button" value="-"/>

Incoming Requests FROM URI Settings

Caller ID DN

Caller Name



SIP Profile Configuration

Save Delete Copy Reset Apply Config Add New

Trunk Specific Configuration

Route Incoming Request to new Trunk based on* Never
RSVP Over SIP* Local RSVP
Resource Priority Namespace List < None >
 Fall back to local RSVP
SIP Rel1XX Options* Disabled
Video Call Traffic Class* Mixed
Calling Line Identification Presentation* Default
 Deliver Conference Bridge Identifier
 Early Offer support for voice and video calls (insert MTP if needed)
 Send send-receive SDP in mid-call INVITE
 Allow Presentation Sharing using BFCP
 Allow IX Application Media
 Allow Passthrough of Configured Line Device Caller Information

- Reject Anonymous Incoming Calls
- Reject Anonymous Outgoing Calls

SIP OPTIONS Ping

Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"
Ping Interval for In-service and Partially In-service Trunks (seconds)* 60
Ping Interval for Out-of-service Trunks (seconds)* 120
Ping Retry Timer (milliseconds)* 500
Ping Retry Count* 6

Save Delete Copy Reset Apply Config Add New



Cisco Unified Communications Manager - SIP trunk to Nortel PBX Configuration

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Trunk Configuration Related Links: **Back To Find/List**

Status

Status: Ready

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	<input type="text" value="CS104_Nortel"/>
Description	<input type="text" value="SIP Trunk to Nortel CS1K"/>
Device Pool*	<input type="text" value="Default"/>
Common Device Configuration	<input type="text" value="< None >"/>
Call Classification*	<input type="text" value="Use System Default"/>
Media Resource Group List	<input type="text" value="MRGL_Taurus"/>
Location*	<input type="text" value="Hub_None"/>
AAR Group	<input type="text" value="< None >"/>

Tunneled Protocol*

QSIG Variant*

ASN.1 ROSE OID Encoding*

Packet Capture Mode*

Packet Capture Duration

Media Termination Point Required

Retry Video Call as Audio

Path Replacement Support

Transmit UTF-8 for Calling Party Name

Transmit UTF-8 Names in QSIG APDU

Unattended Port

SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.

Consider Traffic on This Trunk Secure*

Route Class Signaling Enabled*

Use Trusted Relay Point*

PSTN Access

Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)

E.164 Transformation Profile

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain

Call Routing Information

Remote-Party-Id

Asserted-Identity

Asserted-Type*

SIP Privacy*



Inbound Calls

Significant Digits* All

Connected Line ID Presentation* Default

Connected Name Presentation* Default

Calling Search Space < None >

AAR Calling Search Space < None >

Prefix DN

Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

Connected Party Settings

Connected Party Transformation CSS < None >

Use Device Pool Connected Party Transformation CSS

Outbound Calls

Called Party Transformation CSS < None >

Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS < None >

Use Device Pool Calling Party Transformation CSS

Calling Party Selection* Originator

Calling Line ID Presentation* Default

Calling Name Presentation* Default

Calling and Connected Party Info Format* Deliver DN only in connected party

Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS < None >

Use Device Pool Redirecting Party Transformation CSS

Caller Information

Caller ID DN

Caller Name

Maintain Original Caller ID DN and Caller Name in Identity Headers

SIP Information

Destination

Destination Address is an SRV

Destination Address	Destination Address IPv6	Destination Port
1* 172.30.11.100		5060

MTP Preferred Originating Codec* 711ulaw

SIP Trunk Security Profile* Non Secure SIP Trunk Profile

Rerouting Calling Search Space < None >

Out-Of-Dialog Refer Calling Search Space < None >

SUBSCRIBE Calling Search Space < None >

SIP Profile* Early Offer SIP Standard Profile

DTMF Signaling Method* No Preference

Normalization Script

Normalization Script Nortel_Script_As_Is

Enable Trace

Parameter Name	Parameter Value
1	



Geolocation Configuration

Geolocation

Geolocation Filter

Send Geolocation Information



Cisco UCM- SIP Normalization Configuration

- 1-Copy the SIP Normalization Script (below) and paste it to a note pad and save as text. Make sure that the script is in text format and not corrupted.
- 2- Access to the SIP Normalization Script Configuration in Cisco UCM and upload to the Script. Give the loaded script a short name so that the name will be assigned to the SIP Trunk.
- 3- Assign the script name to the SIP Trunk in the SIP Trunk configuration-SIP Normalization section.

The screenshot shows the Cisco Unified CM Administration web interface. The page title is "SIP Normalization Script Configuration". The status is "Ready". The "SIP Normalization Script Info" section contains the following fields:

Name*	Nortel_Script_As_Is
Description	
Content*	<pre>Nortel = {} trace.enable() -- Tested with Nortel CS1000E release 7,0 Nortel.allowHeaders = {"History-Info"} local mwi_number = scriptParameters.getValue("mwi-number") if not mwi_number then mwi_number = "1000" --</pre>



Cisco UCM- Software Script

The Cisco UCM-Software Script should be applied at SIP trunk toward Nortel PBX. This script normalizes the SIP messaging to/from the Nortel for UC Voice Mail center MWI, History-Info to Diversion Header conversion, Diversion Header to History-Info header conversion, Omitting Option and Update from Allow header. Copy and paste to note pad and save it using txt format. Review the script to ensure that it is not corrupted before upload to the normalization section.

```
Nortel = { }
```

```
trace.enable()
```

```
-- Tested with Nortel CS1000E release 7.0
```

```
Nortel.allowHeaders = {"History-Info" }
```

```
local mwi_number = scriptParameters.getValue("mwi-number")
```

```
if not mwi_number
```

```
then
```

```
    mwi_number = "1000"
```

```
end
```

```
local function adjustRedirectInfo(msg)
```

```
    local di = msg.getHeader("Diversion")
```

```
    if not di
```

```
    then
```

```
        return
```

```
    end
```

```
    msg:convertDiversionToHI()
```

```
    msg.removeHeader("Diversion")
```

```
    local historyInfos = msg.getHeaderValues("History-Info")
```

```
    msg.removeHeader("History-Info")
```

```
-- For debugging purposes, dump out what the Diversion header contained and dump out the list of History-Info headers
```

```
-- produced by msg:convertDiversionToHI. These extra headers will help debug but should be ignored by Nortel. Trace
```

```
-- should be disabled via Admin UI unless a problem is being debugged. Therefore, under normal operating conditions,
```

```
-- the debug headers won't be included in the message.
```

```
if trace.enabled()
```

```
then
```

```
    msg:addHeader("X-Debug-Diversion", di)
```

```
    for i, hi in ipairs(historyInfos)
```

```
    do
```

```
        msg:addHeader("X-Debug-History-Info", hi)
```

```
    end
```

```
end
```

```
-- Example:
```

```
-- Original Diversion header generated natively by CUCM might have been this:
```

```
-- Diversion: <sip:1002@10.10.10.100>;reason=unconditional;privacy=off;screen=yes
```

```
--
```

```
-- The call to convertDiversionToHI will produce these:
```



```
-- History-Info: <sip:1002@10.10.10.100:5060?Reason=sip;cause=302;text="unconditional">;index=1
-- History-Info: <sip:2400@10.10.10.200:5060>;index=1.1
--
-- However, Nortel needs something that looks like this:
-- History-Info:<sip:1002@10.10.10.100?reason=sip%3Bcause%3D302%3Btext%3D%22Moved%20Temporarily%22>;index=1
-- History-Info: <sip:2400@10.10.10.200>;index=2
--
-- This loop generates the additional History-Info header and uses the index value for the first header generated by
-- convertDiversionToHI. Each header uses the index from the next. The last header uses the last value plus one.
-- While processing each header, it also removes the port number from the URI and does any necessary conversion of
-- to special characters to the escaped value for the embedded header.
for i, hi in ipairs(historyInfos)
do
    local uri = string.match(hi, '<(.*?)>') or string.match(hi, "<(.*?)>;index=.*") or ""
-- Strip out the port number.
    uri = string.gsub(uri, "@(.*?)%d+", "@%1")

-- Get the embedded header but without the ?Reason=sip part.
    local embed_header = string.match(hi, '%?Reason=sip(.*?)>')

    if embed_header
    then
        embed_header = string.gsub(embed_header, "unconditional", "Moved Temporarily")
        embed_header = string.gsub(embed_header, ";", "%3B")
        embed_header = string.gsub(embed_header, "=", "%3D")
        embed_header = string.gsub(embed_header, "\"", "%22")
        embed_header = string.gsub(embed_header, " ", "%20")

        embed_header = string.format("?reason=sip%s", embed_header)
    end

    hi = string.format("<%s%>;index=%s", uri, embed_header or "", i)

    msg:addHeader("History-Info", hi)
end
end

-- Remove OPTIONS from outbound INVITE requests.
-- Convert Diversion to History-Info.
function Nortel.outbound_INVITE(msg)
    msg:removeHeaderValue("Allow", "OPTIONS")

    adjustRedirectInfo(msg)
end

-- Remove OPTIONS from any outbound request
function Nortel.outbound_ANY(msg)
    msg:removeHeaderValue("Allow", "OPTIONS")
end

-- Remove OPTIONS from any ourbound response to any request
function Nortel.outbound_ANY_ANY(msg)
    msg:removeHeaderValue("Allow", "OPTIONS")
end

-- Modify the From header so that the userpart is numeric. CUCM will natively send
-- 'voicemail' as the userpart. Nortel does not handle that. This code changes
-- the user part to 1000 or the value of the configured script parameter: mwi-number.
```




```
function Nortel.outbound_NOTIFY(msg)
  msg:removeHeaderValue("Allow", "OPTIONS")

  local from = msg:getHeader("From")
  if from
  then
    from = from:gsub("voicemail", mwi_number)
    msg:modifyHeader("From", from)
    msg:addHeaderUriParameter("From", "user", "phone")
  end
end

-- Convert History-Info to Diversion for inbound invites. Also, remove the
-- phone-context userpart parameter and user=phone URI parameter if either
-- is present.
function Nortel.inbound_INVITE(msg)
  msg:removeHeaderValue("Allow", "UPDATE")

  local hist = msg:getHeader("History-Info")
  if not hist
  then
    return
  end

  msg:convertHIToDiversion()
  msg:removeHeader("History-Info")

  local diversion = msg:getHeader("Diversion")
  if diversion
  then
    -- This first regex will remove the phone-context userpart parameter if there
    -- are other parameters after it but before the @.

    diversion = diversion:gsub(";phone%-context=[^;]*;([^@]*)@", ";%1@")

    -- This second regex will remove the phone-context userpart parameter if it
    -- is immediately before the @.

    diversion = diversion:gsub(";phone%-context=[^@]*@", "@")
    -- Remove user=phone URI parameter.
    diversion = diversion:gsub(";user=phone", "")
    diversion = diversion:gsub(";reason=deflection", ";reason=no-answer")
    -- Save the changes.
    msg:modifyHeader("Diversion", diversion)
  end
end

function Nortel.inbound_ANY_INVITE(msg)
  msg:removeHeaderValue("Allow", "UPDATE")
end

function Nortel.outbound_ANY_INVITE(msg)
  msg:removeHeaderValue("Allow", "OPTIONS")
end

return Nortel
```



Acronyms

Acronym	Definition
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
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
Cisco UCM	Cisco Unified Communications Manager
DNS	Domain Name Server
FQDN	Fully Qualified Domain Name
MWI	Message Waiting Indicator
MRGL	Media Resource Group List
MTP	Media Termination Point
PSTN	Public Switched Telephone Network
SIP	Session Initiated Protocol



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