



Cisco Unified Communications Manager Release 6.0 - PBX Interoperability: Nortel CS1000M Release 4.0 Using SIP Trunk.

August 1, 2007 Initial Version

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Introduction

The purpose of this document is to detail the steps and configurations necessary for Cisco Unified Communications Manager 6.0 to interoperate with the Nortel Communication Server 1000M (CS1000M) running software release 4.0

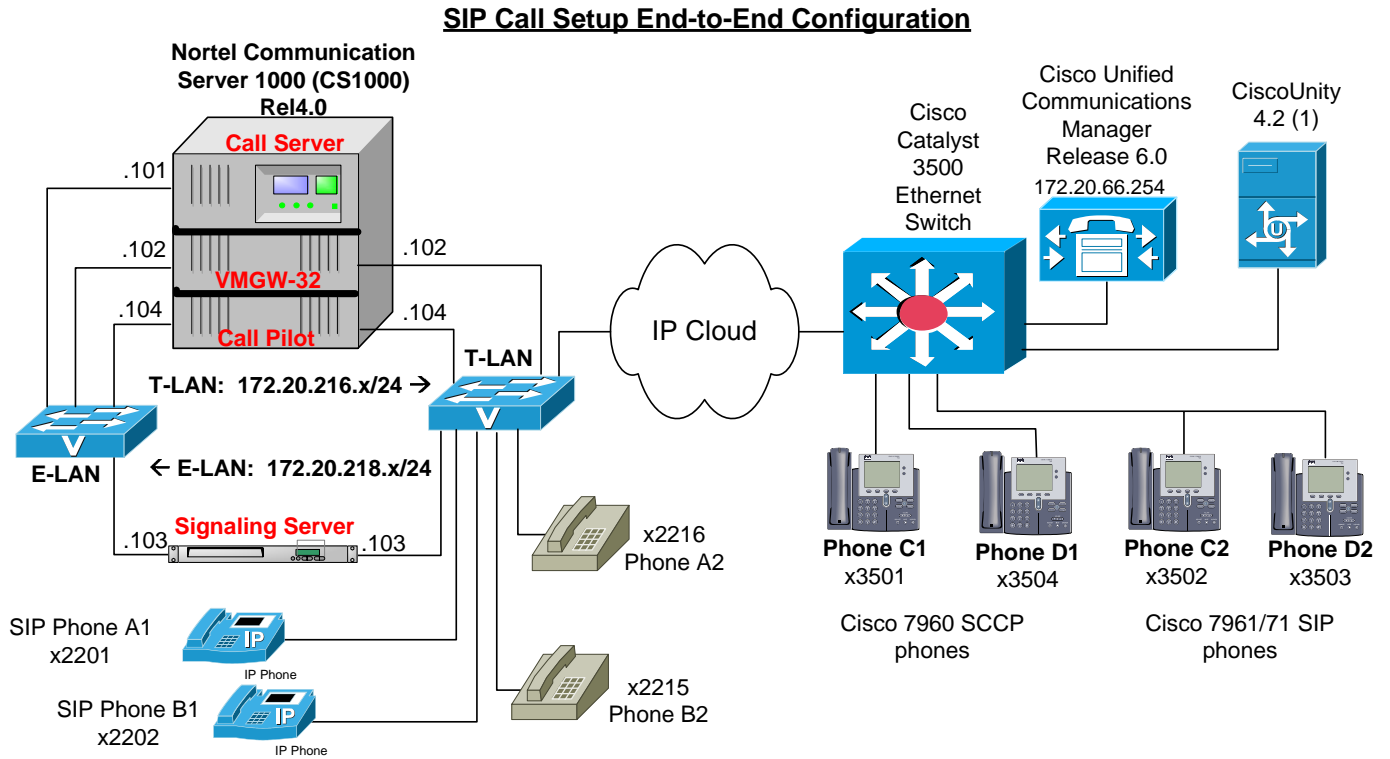
The following items were tested:

1. SIP and SCCP phone registration to the Cisco Unified Communications Manager
2. Basic call between the two systems and verification of voice path
3. CLIP/CLIR/CNIP/CNIR features - Calling Party Name and Number delivery (Allowed and Restricted)
4. COLP/CONP/COLR/CONR features - Connected Name and Number delivery (Allowed and Restricted)
5. Call Transfer (Blind, Attended, Early Attended)
6. Call Forwarding (CFA - Call Forward All, CFB - Call Forward Busy, CFNA - Call Forward No Answer)
7. Hold and Resume with Music On-Hold
8. Voice Messaging and MWI activation-deactivation
9. DTMF-relay (RFC2833, SIP INFO, or KPML)



Network Topology

Figure 1. Network Topology or Test Setup





Limitations

For Nortel CS1000M and Cisco Unified Communications Manager (CUCM) to interoperate with each other (Basic Call and/or Supplementary features), the CUCM “Media Termination Point Required” box must be checked under the SIP Trunk configuration. This causes the CUCM SIP trunk to support Early Offer (SIP Invite with SDP) by pre-allocating MTP resources. The Nortel PBX requires early offer to set its media attribute to send/receive mode. If this box is not checked, CUCM will do delay offer (SIP Invite without SDP, SDP included in 200 ACK message instead). The Nortel PBX responds to delay offer by setting its media attribute to receive only mode. Thus, although calls are completed, there is one way audio from Cisco to Nortel.

The method used by each system to pass the phone name and number information across the SIP trunk is different. Cisco Unified Communications Manager used the “Remote-Party-Id” field while Nortel CS1000M used the “P-Asserted-Id” field. Since both parties do not support each other method, they used the information from the SIP From header as the calling party name and number information

For features such as CLIR, CNIR, COLR and CONR, both systems set the SIP FROM: header to be “Anonymous” and have the proper restriction set with the RPID and PAI fields. Since both parties do not support each other method, they used the information within the SIP FROM: header instead which is “Anonymous”. Cisco Unified Communications Manager use “Remote-Party-Id” and “Privacy” whereas Nortel use “P-Asserted-Id” and “Privacy”.

Both systems support Attended and Early Attended Call Transfer feature. However, they are not able to update the phone displays properly after the transfer is completed. This is due to the differences between the two systems method of passing the name and number information across SIP Trunk.

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

Both systems support Call Forwarding (CFU, CFB, and CFNA) features. However, they are not able to update the phone displays properly after forwarding the call. This is due to the differences between the two systems method of passing the name and number information.

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

Voice Messaging does not work across SIP Trunk between CUCM and Nortel CS1000M PBX. The CUCM uses the SIP Diversion header to pass the redirect information across the SIP Trunk to the system that hosts the VM system. On the other hand, Nortel uses the SIP History-Info field. Due to the difference in the method of sending redirect information, the voice mail system cannot recognize a forwarded call and thus, will treat the call as a direct access call. As a result, Cisco Unity and/or Nortel Call Pilot will not work as a centralized voice mail system for both systems. Each voice mail system (Cisco Unity or Nortel Call Pilot) cannot have subscribers belonging to the one another. For example, Cisco Unity can be a voice mail system for Cisco Unified Communications Manager subscribers but not Nortel CS1000M subscribers using SIP Trunk.

MWI Activation and De-activation message do not work across SIP Trunk. The CUCM uses SIP Notify message for MWI notification. However, Nortel does not support this method.

End-to-end DTMF relay signaling does not work between the two systems and are incompatible with one another. The CUCM supports both RFC2833 and KPML methods of DTMF-relay. As of Release 4.0, the Nortel CS1000 uses SIP INFO message to relay DTMF.

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

Nortel CS1000M

Nortel digital (2616) and IP (i2004, i2002) phones



Software Requirements

Cisco Unified Communications Manager Release 6.0.1

Cisco Unity Release 4.2(1)

CS1000M Release 4.0

Features

CLIP-Calling Line (Number) Identification Presentation (Please see the Limitation section)

CLIR-Calling Line (Number) Identification Restriction (Please see the Limitation section)

CNIP-Calling Name Identification Presentation (Please see the Limitation section)

CNIR-Calling Name Identification Restriction (Please see the Limitation section)

Alerting Name (Please see the Limitation section)

Attended Call Transfer (Please see the Limitation section)

Early Attended Call Transfer (Please see the Limitation section)

CFU-Call Forwarding Unconditional (Please see the Limitation section)

CFB-Call Forwarding Busy (Please see the Limitation section)

CFNA-Call Forwarding No Answer (Please see the Limitation section)

COLP-Connected Line (Number) Identification Presentation (Please see the Limitation section)

COLR- Connected Line (Number) Identification Restriction (Please see the Limitation section)

CONP-Connected Name Identification Presentation (Please see the Limitation section)

CONR-Connected Name Identification Restriction (Please see the Limitation section)

Hold and Resume

Conference Call (Please see the Limitation section)

Features Not Supported

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

Call Completion (Callback; Automatic Callback)

Blind Call Transfer

DTMF-relay

Configuration

Configuring the Nortel Communication Server 1000 PBX

Call Server Setup via SSC Card Console

1. LD 17 – Configure the IP D-channel (signaling channel) between the Call Server and the Signaling Server
2. LD 97 – Configure the Super-loop for the Virtual Trunks
3. LD 14 – Configure the SIP Virtual Trunks to the Signaling Server
4. LD 14 – Configure the Virtual Gateway Trunks
5. LD 11 – Configure for the Virtual lines for the Nortel IP phone (i200x series)
6. LD 16 – Configure the SIP route
7. LD 86 – Configure the Route List Block for the Virtual Trunk route



8. LD 87 – Configure CDP steering codes
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)

1. Configure the System Wide Settings
2. Configure the NRS Server Settings
3. Configure a Service Domain
4. Configure a L1 Domain (UDP)
5. Configure a L0 Domain (CDP)
6. Configure a SIP gateway
7. Configure the Routing Entries

Configuring the Nortel Communication Server 1000 (CS1000)

Call Server Setup:

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

```
>ld 22
PT2000

REQ prt
TYPE adan dch 3

ADAN DCH 3
CTYP DCIP
DES IP_Trunk_DCH
USR ISLD
ISLM 4000
SSRC 1800
OTBF 32
NASA NO
IFC SL1
CNEG 1
RLS ID 4
RCAP ND2
MBGA NO
H323
OVLN NO
OVLN NO
```

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

```
>ld 97
SCSYS000
```



MEM AVAIL: (U/P): 2854769 USED U P: 182454 59352 TOT: 3096575

DISK RECS AVAIL: 1152

REQ prt

TYPE supl

SUPL

SUPL SUPT SLOT XPEC0 XPEC1

000 STD LEFT 01 0 1 ----
004 STD LEFT 02 0 1 ----
008 STD LEFT 03 0 1 ----
012 STD LEFT 04 0 1 ----
016 STD LEFT 05 0 1 ----
032 STD LEFT 06 0 3 ----
036 STD LEFT 07 0 3 ----
040 STD LEFT 08 0 3 ----
044 STD LEFT 10 0 3 ----
048 STD LEFT 09 0 3 ----
064 STD LEFT 11 0 3 ----
068 STD LEFT 12 0 3 ----
072 STD LEFT 13 0 3 ----
096 VIRTUAL CARDS 61 - 64 81 - 84
100 VIRTUAL CARDS 65 - 68 85 - 88
128 STD LEFT 32 0 1 33 2 3
132 STD LEFT 34 0 1 35 2 3
136 STD LEFT 36 0 1 37 2 3
140 STD LEFT 38 0 1 39 2 3
144 STD LEFT 40 0 1 41 2 3
148 STD LEFT 42 0 1 43 2 3
152 STD LEFT 44 0 1 45 2 3
156 STD LEFT 46 0 1 47 2 3

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

>ld 20

REQ: prt

TYPE: tnb

TN 62 0 0 0

→ SIP Virtual trunk to Signaling Server

DATE

PAGE

DES

DES SIP_IP_VTRK

TN 062 0 00 00 VIRTUAL

TYPE IPTI

CDEN 8D

CUST 0

XTRK VTRK

ZONE 000

LDOP BOP

TIMP 600

BIMP 600

AUTO_BIMP NO

TRK ANLG

NCOS 0

RTMB 10 1

CHID 1

TGAR 1

STRI/STRO IMM IMM



SUPN YES
AST NO
IAPG 0
CLS CTD DTN WTA LPR APN THFD
P10 NTC MID
TKID
AACR NO
DATE 25 FEB 2005

NACT

4. LD 14 – Configure the Virtual Gateway Trunks (up to 32 trunks per MC-32)

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

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

→ 1st channel define on the gateway

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

→ 2nd channel define on the gateway

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

Phone A1 (i2004)

>ld 11
SL1000
MEM AVAIL: (U/P): 2854769 USED U P: 182454 59352 TOT: 3096575
DISK RECS AVAIL: 1152
DIGITAL TELEPHONES AVAIL: 6 USED: 2 TOT: 8
IP USERS AVAIL: 6 USED: 2 TOT: 8
BASIC IP USERS AVAIL: 7 USED: 1 TOT: 8
ACD AGENTS AVAIL: 10 USED: 0 TOT: 10
PCA AVAIL: 0 USED: 0 TOT: 0
AST AVAIL: 1 USED: 0 TOT: 1
TNS AVAIL: 2405 USED: 95 TOT: 2500
DATA PORTS AVAIL: 2500 USED: 0 TOT: 2500

REQ: prt
TYPE: tnb

TN 61 0 0 02



DATE
PAGE
DES

DES I2004
TN 061 0 00 02 VIRTUAL
TYPE I2004

CDEN 8D
CUST 0
ZONE 000

FDN
TGAR 1
LDN NO
NCOS 0
SGRP 0
RNPG 0
SCI 0
SSU
LNRS 16
XLST

CLS CTD **FBA** WTA LPR MTD **FNA HTA** TDD HFD CRPD
MWA LMPN RMMD SMWD AAD IMD XHD IRA NID OLD VCE DRG1
POD DSX VMD CMSD SLKD CCSD SWD LNA **CNDA**
CFTA SFD MRD DDV **CNIA** CDCA MSID DAPA BFED RCBD
ICDD CDMD LLCN MCTD CLBD AUTU
GPUD DPUD **DNDA CFXA** ARHD CLTD ASCD
CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
AHD **DDGA NAMA**
DRDD EXR0
USRD ULAD RTDD RBDD RBHD PGND OCBF FLXD FTTC DNDY DNO3 MCBN
VOLA VOUD CDMR

CPND_LANG ENG
RCO 0

EFD
HUNT
EHT

LHK 0
LPK 1
PLEV 02
CSDN

AST
IAPG 0
AACs NO
ITNA NO

DGRP
MLWU_LANG 0
DNDR 0

KEY 00 SCR 2201 0 MARP

CPND
NAME **ZEUS_2201**
XPLN 8
DISPLAY_FMT FIRST, LAST

01
02
03 MIK
04 MCK
05
06
07



08
09
10
11
12
13
14
15
16 MWK 2500
17 TRN
18 AO6
19 CFW 16 2500
20 RGA
21 PRK
22 RNP
23
24 PRS
25 CHG
26 CPN
27
28
29
30
31
DATE 24 MAY 2006
NACT

Phone A2 (2616):

REQ: prt
TYPE: 2616

TN 1 0
DATE
PAGE
DES

DES CS101A
TN **001 0 00 03**
TYPE **2616**

CDEN 8D
CUST 0
AOM 0
FDN
TGAR 1
LDN NO
NCOS 0
SGRP 0
RNPG 0
0 I
SSU
XLST
CLS CTD **FBA** WTA LPR MTD **FNA HTA** ADD HFD
MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
POD DSX VMD CMSD SLKD CCSD SWD LND **CNDA**
CFTA SFD MRD DDV CNID CDCA MSID DAPA BFED RCBF
ICDD CDMD LLCN MCTD CLBD AUTU



GPUD DPUD **DNDA CFXA** ARHD CLTD ASCD
CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD AHD
DDGA NAMA
DRDD EXR0
USRD ULAD RTDD RBDD RBHD PGND OCBF FLXD FTTC DNDY DNO3 MCBN CDMR
CPND_LANG ENG
RCO 0
EFD
HUNT
EHT
LHK 0
PLEV 02
CSDN
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
DNDR 0
KEY 00 SCR 2216 0 MARP
CPND
NAME **ZEUS_2216**
XPLN 9
DISPLAY_FMT FIRST, LAST
01
02
03 CFW 4 6100
04 AO6
05 TRN
06
07
08
09
10
11
12
13 MIK
14 MCK
15 RGA
DATE 25 MAY 2006

NACT

Phone B1 (i2002):

REQ: prt
TYPE: tnb
TN 61 0 0 1
DATE
PAGE
DES

DES I2002
TN 061 0 00 01 VIRTUAL
TYPE I2002
CDEN 8D
CUST 0



ZONE 000

FDN

TGAR 1
LDN NO
NCOS 0
SGRP 0
RNPG 0
SCI 0
SSU
LNRS 16
XLST

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

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

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

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

ICDD CDMD LLCN MCTD CLBD AUTU

GPUD DPUD **DNDA CFXA** ARHD CLTD ASCD

CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD AHD

DDGA NAMA

DRDD EXR0

USRD ULAD RTDD RBDD RBHD PGND OCBF FLXD FTTC DNDY DNO3 MCBN

VOLA VOUD CDMR

CPND_LANG ENG

RCO 0

EFD

HUNT

EHT

LHK 0

LPK 1

PLEV 02

CSDN

AST

IAPG 0

AACS NO

ITNA NO

DGRP

MLWU_LANG 0

DNDR 0

KEY 00 SCR 2202 0 MARP

CPND

NAME ZEUS_2202

XPLN 8

DISPLAY_FMT FIRST, LAST

01

02

03 MIK

04 MCK

05

06

07

08

09

10

11

12

13

14

15

16 MWK 6100



17 TRN
18 AO6
19 CFW 16 6100
20 RGA
21 PRK
22 RNP
23
24 PRS
25 CHG
26 CPN
27
28
29
30
31
DATE 24 MAY 2006
NACT

Phone B2 (2616):

REQ PRT
TYPE: 2616
TN 1 1
DATE
PAGE
DES

DES CS101A
TN 001 0 00 02
TYPE 2616
CDEN 8D
CUST 0
AOM 0
FDN
TGAR 1
LDN NO
NCOS 0
SGRP 0
RNPG 0
SCI 0
SSU
XLST
CLS CTD **FBA** WTA LPR MTD FND **HTA** ADD HFD
DRG1MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE
POD DSX VMD CMSD SLKD CCSD SWD LND **CNDA**
CFTD SFD MRD DDV **CNIA** CDCA MSID DAPA BFED RCBD
ICDD CDMD LLCN MCTD CLBD AUTU
GPUD DPUD DNDA **CFXA** ARHD CLTD ASCD
CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD AHD
DDGA NAMA
DRDD EXR0
CDMRUSRD ULAD RTDD RBDD RBHD PGND OCBF FLXD FTTC DNDY DNO3 MCBN
CPND_LANG ENG
RCO 0
HUNT
LHK 0
PLEV 02
CSDN



```
AST
IAPG 0
AACs NO
ITNA NO
DGRP
MLWU_LANG 0
DNDR 0
KEY 00 SCR 2215 0  MARP
  CPND
  NAME ZEUS_2215
  XPLN 7
  DISPLAY_FMT FIRST, LAST
01
02
03 CFW 4 2500
04 AO6
05 TRN
06
07
08
09
10
11
12
13
14
15 RGA
DATE 4 MAY 2006
```

NACT

6. LD 16 – Configure the SIP route

```
>ld 21
PT1000

REQ: prt
TYPE: rdb
CUST 0
ROUT 10

TYPE RDB
CUST 00
DMOD
ROUT 10
DES SIP_TIE
TKTP TIE
NPID_TBL_NUM 0
ESN NO
CNVT NO
SAT NO
RCLS EXT
VTRK YES
ZONE 000
PCID SIP
CRID YES
NODE 101
DTRK NO
```



ISDN YES
MODE ISLD
DCH 3
IFC SL1
PNI 00001
NCNA YES
NCRD YES
TRO NO
FALT NO
CTYP UKWN
INAC NO
ISAR NO
DAPC NO
PTYP ATT
AUTO NO
DNIS NO
DCDR NO
ICOG IAO
SRCH LIN
TRMB YES
STEP
ACOD 2310
TCPP NO
TARG 01
CLEN 1
BILN NO
OABS
INST
ANTK
SIGO STD
STYP SDAT
ICIS YES
TIMR ICF 512
OGF 512
EOD 13952
DSI 34944
NRD 10112
DDL 70
ODT 4096
RGV 640
GRD 896
SFB 3
NBS 2048
NBL 4096

IENB 5

PAGE 002

TFD 0
VSS 0
VGD 6
SST 5 0
NEDC ORG
FEDC ORG
CPDC NO
DLTN NO
HOLD 02 02 40



SEIZ 02 02
SVFL 02 02
DRNG NO
CDR NO
VRAT NO
MUS NO
MANO NO
FRL 0 0
FRL 1 0
FRL 2 0
FRL 3 0
FRL 4 0
FRL 5 0
FRL 6 0
FRL 7 0
OHQ NO
OHQT 00
CBQ NO
AUTH NO
TTBL 0
ATAN NO
OHTD NO
PLEV 2
ALRM NO
ART 0
SGRP 0
AACR NO

REQ:

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

>ld 86
ESN000

MEM AVAIL: (U/P): 2819994 USED U P: 223389 69576 TOT: 3112959

DISK RECS AVAIL: 1152

REQ prt

CUST 0

FEAT rlb

RLI 10

RLI 10

ENTR 0

LTER NO

ROUT 10

TOD 0 ON 1 ON 2 ON 3 ON

4 ON 5 ON 6 ON 7 ON

VNS NO

SCNV NO

CNV NO

EXP NO

FRL 0

DMI 0

ISDM 0

FCI 0

FSNI 0

SBOC NRR

IDBB DBD



IOHQ NO
OHQ NO
CBQ NO

ISET 0
NALT 5
MFRL 0
OVLL 0

8. LD 87 – Configure CDP steering codes

>ld 87
ESN000

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

REQ prt
CUST 0
FEAT cdp
TYPE dsc
DSC

DSC 600 → 600x Route to CCM extension, sent out via SIP route

FLEN 0
DSP LSC

RLI 10 → SIP Route List which point to Rout 10

NPA
NXX

DSC 610 → 610x Route to Unity VM send out via SIP route

FLEN 0
DSP LSC

RLI 10 → SIP Route List which point to Rout 10

NPA
NXX

9. LD 21 – List Trunk Members

>Ld 21
PT1000

REQ: ltm
CUST 0
ROUT 10

TYPE TLST
TKTP TIE
ROUT 10
DES SIP_TIE

TN 062 0 00 00 MBER 1 SIP_IP_VTRK
TN 062 0 00 01 MBER 2 SIP_IP_VTRK
TN 062 0 00 02 MBER 3 SIP_IP_VTRK
TN 062 0 00 03 MBER 4 SIP_IP_VTRK
TN 062 0 00 04 MBER 5 SIP_IP_VTRK
TN 062 0 00 05 MBER 6 SIP_IP_VTRK
TN 062 0 00 06 MBER 7 SIP_IP_VTRK
TN 062 0 00 07 MBER 8 SIP_IP_VTRK



Signaling Server Setup:

10. Configure the Zones

Site: 172.20.249.101 > Configuration > Call Server Configuration > Zone List > Zone 0 >

Zone Basic Property and Bandwidth Management

Input Description	Input Value
Zone Number (ZONE):	<input type="text"/>
Intrazone Bandwidth (INTRA_BW):	<input type="text" value="10300"/>
Intrazone Strategy (INTRA_STGY):	Best Quality (BQ) ▾
Interzone Bandwidth (INTER_BW):	<input type="text" value="10300"/>
Interzone Strategy (INTER_STGY):	Best Quality (BQ) ▾
Resource Type (RES_TYPE):	Shared (SHARED) ▾
Branch Office Support (ZBRN):	<input type="checkbox"/>
Description (ZDCS):	<input type="text"/>



11. Configure a new IP Telephony Node summary

Site: 172.20.218.101 > Configuration > IP Telephony Configuration >

Node Summary

New Node

▼ **Node: 101 Node IP: 172.20.216.100**

Voice LAN (TLAN) IP address TN

Signaling Server
172.20.216.103

Pentium Card
Succession Media Card
172.20.216.102 3 0

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



12. Configure the Node section

Site: 172.20.218.101 > Configuration > IP Telephony Configuration > Node Summary > IP Telephony: Node ID 101 >

Edit

▼ Node	
Node ID	101
Voice LAN (TLAN) Node IP address	<input type="text" value="172.20.216.100"/>
Management LAN (ELAN) gateway IP address	<input type="text" value="172.20.218.1"/>
Management LAN (ELAN) subnet mask	<input type="text" value="255.255.255.0"/>
Voice LAN (TLAN) subnet mask	<input type="text" value="255.255.255.0"/>
> SNMP	<input type="button" value="Add"/>
> VGW and IP phone codec profile	
> QoS	
> LAN configuration	
> SNTP	
> H323 GW Settings	



13. Configure the VGW and IP phone codec profile section

The screenshot shows the Cisco configuration interface for the 'VGW and IP phone codec profile' section. On the left is a navigation menu with categories like System Status, Configuration, Network Numbering Plan, Software Upgrade, Patching, System Utility, Administration, Support, Tools, and Logout. The main content area is titled 'VGW and IP phone codec profile' and contains several settings:

- Enable Echo canceller:
- Echo canceller tail delay: 128 (dropdown)
- Voice activity detection threshold: -17 (text input) Range: -20 to +10
- Idle noise level: -65 (text input) Range: -327 to +327
- DTMF Tone detection:
- Enable V.21 FAX tone detection:
- FAX maximum rate (bps): 14400 (dropdown)
- FAX payout nominal delay: 100 (text input) Range: 0 to 300
- FAX no activity timeout: 20 (text input) Range: 10 to 32000
- FAX packet size: 30 (dropdown)

Below these settings is a list of codecs with 'Select' buttons and checkboxes:

Codec G711	Select <input checked="" type="checkbox"/>
Codec G729A	Select <input checked="" type="checkbox"/>
Codec G723.1	Select <input type="checkbox"/>
Codec T38 FAX	Select <input checked="" type="checkbox"/>

At the bottom of the configuration area, there are expandable sections for QoS, LAN configuration, and SNMP.



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

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



14. Configure the QoS section

▼ Codec G711 Select

▼ Codec G729A Select

▼ Codec G723.1 Select

▼ Codec T38 FAX Select

▼ QoS

Diffserv Codepoint(DSCP) Control packets Range: 0 to 63

Diffserv Codepoint(DSCP) Voice packets Range: 0 to 63

Enable 802.1Q support

802.1Q Bits value (802.1p) Range: 0 to 7

> LAN configuration

> SNTP

> H323 GW Settings

> Firmware

> SIP GW Settings

> SIP URI Map

> SIP CD Services

> Cards

> Signaling Servers



15. Configure LAN Configuration section

<ul style="list-style-type: none"> System Status <ul style="list-style-type: none"> Call Server IP Telephony Configuration <ul style="list-style-type: none"> Call Server IP Telephony Network Numbering Plan <ul style="list-style-type: none"> Call Server Network Routing Service Software Upgrade Patching System Utility Administration Support Tools Logout 	Codec	T38 FAX	Select <input checked="" type="checkbox"/>
	QoS		
	Diffserv Codepoint(DSCP) Control packets	<input type="text" value="40"/>	Range: 0 to 63
	Diffserv Codepoint(DSCP) Voice packets	<input type="text" value="46"/>	Range: 0 to 63
	Enable 802.1Q support	<input type="checkbox"/>	
	802.1Q Bits value (802.1p)	<input type="text" value="6"/>	Range: 0 to 7
	LAN configuration		
	Management LAN (ELAN) configuration		
	Call server IP address	<input type="text" value="172.20.218.101"/>	
	Survivable Succession Media Gateway IP address	<input type="text" value="0.0.0.0"/>	
	Signaling port	<input type="text" value="15000"/>	Range: 1024 to 65535
	Broadcast port	<input type="text" value="15001"/>	Range: 1024 to 65535
	Voice LAN (TLAN) configuration		
	Signaling port	<input type="text" value="5000"/>	Range: 1024 to 65535
	Voice port	<input type="text" value="5200"/>	Range: 1024 to 65535
Routes			
<input type="button" value="Add"/>			
IP address	<input type="text" value="172.20.216.1"/>	Subnet mask	
	<input type="text" value="255.255.255.0"/>	<input type="button" value="Remove"/>	



16. Configure the SIP GW Setting section

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

▼ SIP GW Settings

Primary Proxy / Re-direct IP address	172.20.216.103
Primary Proxy / Re-direct IP Port	5060
Primary Proxy Supports Registration	<input checked="" type="checkbox"/>
Primary CDS Proxy or Re-direct server flag	<input checked="" type="checkbox"/>
Secondary Proxy / Re-direct IP address	0.0.0.0
Secondary Proxy / Re-direct IP Port	5060
Secondary Proxy Supports Registration	<input type="checkbox"/>
Secondary CDS Proxy or Re-direct server flag	<input type="checkbox"/>

▼ SIP URI Map

Public E.164/National domain name	+1
Public E.164/Subscriber domain name	+1314
Public E.164/Unknown domain name	
Public E.164/Special Number domain name	
Private/UDP domain name	rtp
Private/CDP domain name	interop.rtp
Private/Special Number domain name	SPN.rtp
Private/Unknown (vacant number routing) domain name	
Unknown/Unknown domain name	

▶ SIP CD Services

▶ Cards Add



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

SIP CD Services

▼ Cards Add

▼ Card 172.20.218.102 Properties Remove

Role: **Follower**

Management LAN (ELAN) IP address: 172.20.218.102 *

Management LAN (ELAN) MAC address: 00:11:F9:E4:D0:11 *

Voice LAN (TLAN) IP address: 172.20.216.102 *

Voice LAN (TLAN) gateway IP address: 172.20.216.1

Hostname: MC_Node101 *

Card TN: 3 *

Card processor type: Succession Media Card ▼

H323 ID: MC_Node101

Enable set TPS:

System name: SS_Node101

System location:

System contact:

▼ Signaling Servers Add

▼ Signaling Server 172.20.218.103 Properties Remove

Save and Transfer Cancel

**Mandatory fields of current configuration*



18. Configure the Signaling Server section

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

Signaling Server 172.20.218.103 Properties Remove

Role	Leader
Management LAN (ELAN) IP address	<input type="text" value="172.20.218.103"/> *
Management LAN (ELAN) MAC address	<input type="text" value="00:02:b3:f7:3a:86"/> *
Voice LAN (TLAN) IP address	<input type="text" value="172.20.216.103"/> *
Voice LAN (TLAN) gateway IP address	<input type="text" value="172.20.216.1"/>
Hostname	<input type="text" value="SS_Node101"/> *
H323 ID	<input type="text" value="SS_Node101"/>
Enable set TPS	<input checked="" type="checkbox"/>
Enable virtual trunk TPS	<input type="text" value="H.323 and SIP"/>
Enable SIP Proxy / Redirect Server	<input checked="" type="checkbox"/>
SIP Transport Protocol	<input type="text" value="TCP"/>
Local SIP Port	<input type="text" value="5060"/>
SIP Domain name	<input type="text" value="pbxlab.org"/>
SIP Gateway Endpoint Name	<input type="text" value="SS_Node101"/>
SIP Gateway Authentication Password	<input type="text" value="password"/>
Enable H323 Gatekeeper	<input checked="" type="checkbox"/>
Network Routing Service Role	<input type="text" value="Primary"/>
System name	<input type="text" value="SS_Node101"/>
System location	<input type="text"/>
System contact	<input type="text"/>



Network Routing Server Setup:

19. Configure the System Wide Settings

Network Routing Service

[Home](#) | [Configuration](#) | [Tools](#) | [Reports](#) | [Administration](#) | [Help](#) | [Logout](#)

Location: Home > System Wide Settings >

System Wide Settings

NRS Overview
=> [System Wide Settings](#)
NRS Server Settings

DB sync interval for alternate [Hours]

SIP registration time to live timer [Seconds]

H.323 gatekeeper registration time to live timer [Seconds]

H.323 alias name *

Alternate NRS server is permanent

Auto backup time [HH:MM]

Auto backup to FTP site enabled

Auto backup FTP site IP address

Auto backup FTP site path

Auto backup FTP username

Auto backup FTP password



20. Configure the NRS Server Settings

Network Routing Service

Home | Configuration | Tools | Reports | Administration | Help | Logout

Location: Home > NRS Server Settings >

NRS Overview

System Wide Settings

=> NRS Server Settings

NRS Settings

Host name *

Primary IP (TLAN) *

Alternate IP (TLAN) *

Control priority

H.323 Gatekeeper Settings

Location request (LRQ) response timeout [Seconds]

SIP Server Settings

Mode

UDP transport enabled

UDP port

UDP maximum transmission unit (MTU)



Network Routing Service

[Home](#) | [Configuration](#) | [Tools](#) | [Reports](#) | [Administration](#)

[Help](#) | [Logout](#)

NRS Overview

System Wide Settings

=> NRS Server Settings

SIP Server Settings

Mode

UDP transport enabled

UDP port

UDP maximum transmission unit (MTU)

TCP transport enabled

TCP port

TCP maximum transmission unit (MTU)

Network Connection Server (NCS) Settings

Primary NCS port

Alternate NCS port

Primary NCS timeout [Seconds]

**Mandatory field indicator*



21. Configure a Service Domain

Network Routing Service

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

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

View Service Domain Property

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

**Mandatory field indicator*

Service Domains

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



22. Configure a L1 Domain (UDP)

Network Routing Service

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

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

View L1 Domain Property (pbxlab.org)

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



23. Configure a L0 Domain (CDP)

Network Routing Service

[Home](#) | [Configuration](#) | [Tools](#) | [Reports](#) | [Administration](#) | [Active DB view](#) (set Standby DB view) | [Help](#) | [Logout](#)

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

View L0 Domain Property (pbxlab.org / rtp)

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

Domain name	<input type="text" value="interop"/> *
Domain description	<input type="text" value="CDP"/>
Special number label	<input type="text"/>
Unqualified number label	<input type="text"/>
Endpoint authentication enabled	<input type="text" value="Not configured"/>
Authentication password	<input type="text"/>
E.164 country code	<input type="text" value="1"/>
E.164 area code	<input type="text" value="919"/>
International dialing access code	<input type="text" value="011"/>
L1 domain dialing access code	<input type="text" value="9"/>
National dialing access code	<input type="text" value="9"/>
Local dialing access code	<input type="text" value="9"/>



24. Configure a SIP gateway

Network Routing Service

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

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

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

Endpoint name	<input type="text" value="CM_Aries"/>	*
Endpoint description	<input type="text" value="CCM_Aries"/> <input type="text" value="IP Addr"/> <input type="text" value="172.20.66.254"/>	<input type="button" value="↑"/> <input type="button" value="↓"/>
Tandem endpoint name	<input type="text"/>	Look up
Endpoint authentication enabled	<input type="text" value="Not configured"/>	▼
Authentication password	<input type="text"/>	
E.164 country code	<input type="text" value="1"/>	
E.164 area code	<input type="text" value="650"/>	
International dialing access code	<input type="text" value="011"/>	
L1 domain dialing access code	<input type="text" value="9"/>	
National dialing access code	<input type="text" value="9"/>	
Local dialing access code	<input type="text" value="9"/>	
Special number 1	<input type="text"/>	
Special number 2	<input type="text"/>	
Static endpoint address type	<input type="text" value="IP version 4"/>	▼
Static endpoint address	<input type="text" value="172.20.66.254"/>	
H.323 Support	<input type="text" value="Not RAS H.323 endpoint"/>	▼
SIP support	<input type="text" value="Static SIP endpoint"/>	▼
SIP transport	<input type="text" value="TCP"/>	▼



Network Routing Service



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

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

Endpoint description	IP Addr
	172.20.66.254
Tandem endpoint name	<input type="text"/> Look up
Endpoint authentication enabled	Not configured
Authentication password	<input type="text"/>
E.164 country code	1
E.164 area code	650
International dialing access code	011
L1 domain dialing access code	9
National dialing access code	9
Local dialing access code	9
Special number 1	<input type="text"/>
Special number 2	<input type="text"/>
Static endpoint address type	IP version 4
Static endpoint address	172.20.66.254
H.323 Support	Not RAS H.323 endpoint
SIP support	Static SIP endpoint
SIP transport	TCP
SIP port	5060
Network Connection Server enabled	<input type="checkbox"/>

*Mandatory field indicator



25. Configure the Routing Entries

Network Routing Service

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

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

View Routing Entry Property (pbxlab.org / rtp / interop / CM_Aries)

DN type	Level0 regional
Default route	<input type="checkbox"/>
DN prefix	350
Route cost (1 -255)	1

**Mandatory field indicator*



Configuring Cisco Unified Communications Manager

Cisco Unified Communications Manager Software Version

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go

Cisco Unified CM Administration
System version: 6.0.1.1000-37

Username
Password

Login Reset

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This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco cryptographic products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with U.S. and local country laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

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If you require further assistance please contact us by sending email to export@cisco.com.



Cisco Unified Communications Manager Group Configuration

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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Cisco Unified CM Group Configuration Related Links: [Back To Find/List](#)

Save Delete Copy Reset Add New

Cisco Unified Communications Manager Group Information
Cisco Unified Communications Manager Group: Default (used by 24 devices)

Cisco Unified Communications Manager Group Settings
Name*
 Auto-registration Cisco Unified Communications Manager Group

Cisco Unified Communications Manager Group Members
Available Cisco Unified Communications Managers
Selected Cisco Unified Communications Managers*

*- indicates required item.

**- indicates required item.



Cisco Unified Communications Manager Default Device Pool Configuration

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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

Save Delete Copy Reset Add New

Device Pool Information

Device Pool: Default (20 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 >"/>
Reverted Call Focus Priority	<input type="text" value="Default"/>

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="Aries_MRGL"/>
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"/>
Physical Location	<input type="text" value="< None >"/>
Device Mobility Group	<input type="text" value="< None >"/>



Device Pool Configuration

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

Save Delete Copy Reset Add New

Location	< None >
Network Locale	< None >
SRST Reference*	Disable
Connection Monitor Duration***	
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 >

- *- indicates required item.
- **Number of devices that have to be reset when this device pool is updated. To see a detailed list of these devices and other dependencies, click on Dependence Records.
- ***leave blank to use default.
- ****These three parameters will overwrite device level settings when device is roaming and in the same device mobility group.



Cisco Unified Communications Manager Enterprise Parameters (Organization Top Level Domain) Configuration

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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Enterprise Parameters Configuration

Save Set to Default Reset

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](#) * 8888

[Service Manager TCP Client communication port number](#) * 8889

CRS Application Parameters

[Auto Attendant Installed](#) * false

[IPCC Express Installed](#) * false

Clusterwide Domain Configuration

[Organization Top Level Domain](#)

[Cluster Fully Qualified Domain Name](#)

Denial-of-Service Protection

[Denial-of-Service Protection Flag](#) * True

Cisco Support Use

[Cisco Support Use 1](#)



Cisco Unified Communications Manager Media Termination Point Configuration

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Media Termination Point Configuration Related Links: [Back To Find/List](#)

Save Reset

Status

Status: Ready

Media Termination Point Information

Registration	Registered with Cisco Unified Communications Manager CM-Aries
IP Address	172.20.66.254
Media Termination Point Type*	Cisco Media Termination Point Software
Host Server*	CM-Aries
Media Termination Point Name*	<input type="text" value="MTP_2"/>
Description	<input type="text" value="MTP_CM-Aries"/>
Device Pool*	<input type="text" value="Default"/>

*- indicates required item.



Cisco Unified Communications Manager Media Resource Group Configuration

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Media Resource Group Configuration Related Links: [Back To Find/List](#)

Save Delete Copy Reset Add New

Media Resource Group Status

Media Resource Group: Aries_MRG (used by 20 devices)

Media Resource Group Information

Name*

Description

Devices for this Group

Available Media Resources**

▼ ▲

Selected Media Resources*

ANN_2 (ANN)

CFB_2 (CFB)

MOH_2 (MOH)

MTP_2 (MTP)


▼ ▲

Use Multicast for MOH Audio (If at least one multicast MOH resource is available)

*- indicates required item.








Cisco Unified Communications Manager Media Resource Group List Configuration


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Media Resource Group List Configuration Related Links: [Back To Find/List](#)

 Save  Delete  Copy  Reset  Add New

Status
 Status: Ready


Media Resource Group List Status
Media Resource Group List: Aries_MRGL (used by 20 devices)

Media Resource Group List Information
Name*

Media Resource Groups for this List
Available Media Resource Groups
Selected Media Resource Groups






Cisco Unified Communications Manager SIP Phone Security Profile Configuration

 **Cisco Unified CM Administration**
For Cisco Unified Communications Solutions

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Phone Security Profile Configuration Related Links: [Back To Find/List](#)

 Copy  Reset  Add New

Phone Security Profile Information

Product Type:	Cisco 7961
Device Protocol:	SIP
Name*	<input type="text" value="Cisco 7961 - Standard SIP Non-Secure Profile"/>
Description	<input type="text" value="Cisco 7961 - Standard SIP Non-Secure Profile"/>
Nonce Validity Time*	<input type="text" value="600"/>
Device Security Mode	<input type="text" value="Non Secure"/>
Transport Type*	<input type="text" value="TCP+UDP"/>
<input type="checkbox"/> Enable Digest Authentication	
<input type="checkbox"/> TFTP Encrypted Config	
<input type="checkbox"/> Exclude Digest Credentials in Configuration File	




Phone Security Profile CAPF Information

Authentication Mode*	<input type="text" value="By Null String"/>
Key Size (Bits)*	<input type="text" value="1024"/>

Note: These fields are related to the CAPF Information settings on the Phone Configuration page.

Parameters used in Phone

SIP Phone Port*	<input type="text" value="5060"/>
-----------------	-----------------------------------



Cisco Unified Communications Manager SIP Trunk Security Profile Configuration

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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SIP Trunk Security Profile Configuration Related Links: [Back To Find/List](#)

Save Delete Copy Reset Add New

SIP Trunk Security Profile Information

Name*	<input type="text" value="CS101"/>
Description	<input type="text" value="CS1000 Node 1"/>
Device Security Mode	<input type="text" value="Non Secure"/>
Incoming Transport Type*	<input type="text" value="TCP+UDP"/>
Outgoing Transport Type	<input type="text" value="TCP"/>
<input type="checkbox"/> Enable Digest Authentication	
Nonce Validity Time (mins)*	<input type="text" value="600"/>
X.509 Subject Name	<input type="text"/>
Incoming Port*	<input type="text" value="5060"/>
<input type="checkbox"/> Enable Application Level Authorization	
<input type="checkbox"/> Accept Presence Subscription	
<input type="checkbox"/> Accept Out-of-Dialog REFER	
<input type="checkbox"/> Accept Unsolicited Notification	
<input type="checkbox"/> Accept Replaces Header	

*- indicates required item.



Cisco Unified Communications Manager SCCP Security Profile Configuration

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Phone Security Profile Configuration Related Links: [Back To Find/List](#)

Copy Reset Add New

Status

Status: Ready

Phone Security Profile Information

Product Type: Cisco 7960
Device Protocol: SCCP

Name*
Description
Device Security Mode

Phone Security Profile CAPF Information

Authentication Mode*
Key Size (Bits)*

Note: These fields are related to the CAPF Information settings on the Phone Configuration page.

*- indicates required item.



Cisco Unified Communications Manager SIP Profile for Nortel CS1000M PBX Configuration

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SIP Profile Configuration Related Links: [Back To Find/List](#)

Save Delete Copy Reset Add New

Status

Status: Ready

SIP Profile Information

Name*

Description

Default MTP Telephony Event Payload Type*

Redirect by Application

Disable Early Media on 180

Parameters used in Phone

Timer Invite Expires (seconds)*

Timer Register Delta (seconds)*

Timer Register Expires (seconds)*

Timer T1 (msec)*

Timer T2 (msec)*

Retry INVITE*

Retry Non-INVITE*

Start Media Port*



SIP Profile Configuration

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

Save Delete Copy Reset Add New

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
Timer Keep Alive Expires (seconds)*	120
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
Abbreviated Dial URI*	



SIP Profile Configuration

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

Save Delete Copy Reset Add New

Telnet Level for 7940 and 7960*	<input type="text" value="Disabled"/>
Timer Keep Alive Expires (seconds)*	<input type="text" value="120"/>
Timer Subscribe Expires (seconds)*	<input type="text" value="120"/>
Timer Subscribe Delta (seconds)*	<input type="text" value="5"/>
Maximum Redirections*	<input type="text" value="70"/>
Off Hook To First Digit Timer (milliseconds)*	<input type="text" value="15000"/>
Call Forward URI*	<input type="text" value="x-cisco-serviceuri-cfwdall"/>
Abbreviated Dial URI*	<input type="text" value="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"/> Call Stats	

*- indicates required item.



Cisco Unified Communications Manager SIP Trunk Configuration to Nortel CS1000M PBX Configuration

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

Save Delete Reset Add New

Status

Status: Ready

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Device Name*	<input type="text" value="CS101"/>
Description	<input type="text" value="SIP Trunk to Nortel CS101"/>
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="Aries_MRGL"/>
Location*	<input type="text" value="Hub_None"/>
AAR Group	<input type="text" value="< None >"/>
Packet Capture Mode*	<input type="text" value="None"/>
Packet Capture Duration	<input type="text" value="0"/>

Media Termination Point Required
 Retry Video Call as Audio
 Transmit UTF-8 for Calling Party Name



Trunk Configuration

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

Save Delete Reset Add New

Unattended Port

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain

Call Routing Information

Inbound Calls

Significant Digits*

Connected Line ID Presentation*

Connected Name Presentation*

Calling Search Space

AAR Calling Search Space

Prefix DN

Redirecting Diversion Header Delivery - Inbound

Outbound Calls

Calling Party Selection*

Calling Line ID Presentation*

Calling Name Presentation*

Caller ID DN



Trunk Configuration

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

Save Delete Reset Add New

Caller Name

Redirecting Diversion Header Delivery - Outbound

SIP Information

Destination Address*	<input type="text" value="172.20.216.100"/>
<input type="checkbox"/> Destination Address is an SRV	
Destination Port*	<input type="text" value="5060"/>
MTP Preferred Originating Codec*	<input type="text" value="711ulaw"/>
Presence Group*	<input type="text" value="Standard Presence group"/>
SIP Trunk Security Profile*	<input type="text" value="CS101"/>
Rerouting Calling Search Space	<input type="text" value="< None >"/>
Out-Of-Dialog Refer Calling Search Space	<input type="text" value="< None >"/>
SUBSCRIBE Calling Search Space	<input type="text" value="< None >"/>
SIP Profile*	<input type="text" value="CUCM Profile"/>
DTMF Signaling Method*	<input type="text" value="No Preference"/>

*- indicates required item.



Cisco Unified Communications Manager SCCP Phone Level Configuration

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For Cisco Unified Communications Solutions

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

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Status

Status: Ready

Association Information

[Modify Button Items](#)

- [Line \[1\] - 3501 \(no partition\)](#)
- [Line \[2\] - Add a new DN](#)
- [Add a new SD](#)
- [Add a new SD](#)
- [Add a new SD](#)
- [Add a new SD](#)
- Unassigned Associated Items -----
- [Add a new SD](#)
- [Add a new SURL](#)
- [Add a new BLF SD](#)
- [Add a new BLF Directed Call Park](#)
- Privacy

Phone Type

Product Type: Cisco 7960
Device Protocol: SCCP

Device Information

Registration	Registered with Cisco Unified Communications Manager CM-Aries
IP Address	172.20.66.16
MAC Address*	<input type="text" value="00192F07EE90"/>
Description	<input type="text" value="3501 7960-SCCP"/>
Device Pool*	<input type="text" value="Default"/> View Details
Common Device Configuration	<input type="text" value="< None >"/> View Details
Phone Button Template*	<input type="text" value="Standard 7960 SCCP"/>
Softkey Template	<input type="text" value="Standard User with CallBack"/>
Common Phone Profile*	<input type="text" value="Standard Common Phone Profile"/>
Calling Search Space	<input type="text" value="< None >"/>
AAR Calling Search Space	<input type="text" value="< None >"/>
Media Resource Group List	<input type="text" value="Aries_MRGL"/>



Phone Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.

File Edit View Favorites Tools Help

Address <https://172.25.67.151:8443/ccmadmin/deviceEdit.do?key=cfbff241-4763-4fd6-981a-9bcf48b5645e> Go Links >>

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Phone Configuration Related Links: Back To Find/List Go

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12 None	User Hold MOH Audio Source < None >
	Network Hold MOH Audio Source < None >
	Location* Hub_None
	AAR Group < None >
	User Locale < None >
	Network Locale < None >
	Built In Bridge* Default
	Privacy* Default
	Device Mobility Mode* Default View Current Device
	Owner User ID < None >
	Phone Load Name

Retry Video Call as Audio
 Ignore Presentation Indicators (internal calls only)
 Allow Control of Device from CTI
 Logged Into Hunt Group
 Remote Device

Protocol Specific Information

Done Internet



Protocol Specific Information

Packet Capture Mode*	None
Packet Capture Duration	0
Presence Group*	Standard Presence group
Device Security Profile*	Cisco 7960 - Standard SCCP Non-Secure Profile
SUBSCRIBE Calling Search Space	< None >

Unattended Port
 Require DTMF Reception
 RFC2833 Disabled

Certification Authority Proxy Function (CAPF) Information

Certificate Operation*	No Pending Operation
Authentication Mode*	By Null String
Authentication String	

Key Size (Bits)*	1024
Operation Completes By	2007 9 6 12 (YYYY:MM:DD:HH)

Certificate Operation Status: None
Note: Security Profile Contains Addition CAPF Settings.



Phone Configuration

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

Save Delete Copy Reset Add New

Expansion Module Information

Module 1	<input type="text" value=" < None >"/>
Module 1 Load Name	<input type="text"/>
Module 2	<input type="text" value=" < None >"/>
Module 2 Load Name	<input type="text"/>

External Data Locations Information (Leave blank to use default)

Information	<input type="text"/>
Directory	<input type="text"/>
Messages	<input type="text"/>
Services	<input type="text"/>
Authentication Server	<input type="text"/>
Proxy Server	<input type="text"/>
Idle	<input type="text"/>
Idle Timer (seconds)	<input type="text"/>

Extension Information

Enable Extension Mobility

Log Out Profile



Extension Information

Enable Extension Mobility

Log Out Profile -- Use Current Device Settings --

Log in Time < None >

Log out Time < None >

MLPP Information

MLPP Domain < None >

MLPP Indication* Default

MLPP Preemption* Default

Do Not Disturb

Do Not Disturb

DND Option* Ringer Off

DND Incoming Call Alert < None >

Product Specific Configuration Layout

Disable Speakerphone

Disable Speakerphone and Headset

CC Party*

?



Phone Configuration

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

Save Delete Copy Reset Add New

Product Specific Configuration Layout



<input type="checkbox"/>	Disable Speakerphone
<input type="checkbox"/>	Disable Speakerphone and Headset
PC Port *	Enabled
Settings Access*	Enabled
Gratuitous ARP*	Enabled
PC Voice VLAN Access*	Enabled
Video Capabilities*	Disabled
Auto Line Select*	Disabled
Web Access*	Enabled

Save Delete Copy Reset Add New

- *- indicates required item.
- **- Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.
- ***Note: Security Profile Contains Addition CAPF Settings.



Cisco Unified Communications Manager SCCP Phone Directory Number (Ext 3501) Level Configuration

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Phone Configuration Related Links: Back To Find/List

Save Delete Copy Reset Add New

Status
Status: Ready

Association Information		Phone Type																																																	
<p>Modify Button Items</p> <table border="1"><tr><td>1</td><td>Line [1] - 3501 (no partition)</td></tr><tr><td>2</td><td>Line [2] - Add a new DN</td></tr><tr><td>3</td><td>Add a new SD</td></tr><tr><td>4</td><td>Add a new SD</td></tr><tr><td>5</td><td>Add a new SD</td></tr><tr><td>6</td><td>Add a new SD</td></tr><tr><td colspan="2">----- Unassigned Associated Items -----</td></tr><tr><td>7</td><td>Add a new SD</td></tr><tr><td>8</td><td>Add a new SURF</td></tr><tr><td>9</td><td>Add a new BLF SD</td></tr><tr><td>10</td><td>Add a new BLF Directed Call Park</td></tr><tr><td>11</td><td>Privacy</td></tr></table>		1	Line [1] - 3501 (no partition)	2	Line [2] - Add a new DN	3	Add a new SD	4	Add a new SD	5	Add a new SD	6	Add a new SD	----- Unassigned Associated Items -----		7	Add a new SD	8	Add a new SURF	9	Add a new BLF SD	10	Add a new BLF Directed Call Park	11	Privacy	<p>Product Type: Cisco 7960 Device Protocol: SCCP</p> <hr/> <p>Device Information</p> <table border="1"><tr><td>Registration</td><td>Registered with Cisco Unified Communications Manager CM-Aries</td></tr><tr><td>IP Address</td><td>172.20.66.16</td></tr><tr><td>MAC Address*</td><td>00192F07EE90</td></tr><tr><td>Description</td><td>3501 7960-SCCP</td></tr><tr><td>Device Pool*</td><td>Default View Details</td></tr><tr><td>Common Device Configuration</td><td>< None > View Details</td></tr><tr><td>Phone Button Template*</td><td>Standard 7960 SCCP</td></tr><tr><td>Softkey Template</td><td>Standard User with CallBack</td></tr><tr><td>Common Phone Profile*</td><td>Standard Common Phone Profile</td></tr><tr><td>Calling Search Space</td><td>< None ></td></tr><tr><td>AAR Calling Search Space</td><td>< None ></td></tr><tr><td>Media Resource Group List</td><td>Aries_MRGL</td></tr></table>		Registration	Registered with Cisco Unified Communications Manager CM-Aries	IP Address	172.20.66.16	MAC Address*	00192F07EE90	Description	3501 7960-SCCP	Device Pool*	Default View Details	Common Device Configuration	< None > View Details	Phone Button Template*	Standard 7960 SCCP	Softkey Template	Standard User with CallBack	Common Phone Profile*	Standard Common Phone Profile	Calling Search Space	< None >	AAR Calling Search Space	< None >	Media Resource Group List	Aries_MRGL
1	Line [1] - 3501 (no partition)																																																		
2	Line [2] - Add a new DN																																																		
3	Add a new SD																																																		
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6	Add a new SD																																																		
----- Unassigned Associated Items -----																																																			
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Device Pool*	Default View Details																																																		
Common Device Configuration	< None > View Details																																																		
Phone Button Template*	Standard 7960 SCCP																																																		
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12	None
----	------

User Hold MOH Audio Source < None >

Network Hold MOH Audio Source < None >

Location* Hub_None

AAR Group < None >

User Locale < None >

Network Locale < None >

Built In Bridge* Default

Privacy* Default

Device Mobility Mode* Default [View Current Device](#)

[Mobility Settings](#)

Owner User ID < None >

Phone Load Name

Retry Video Call as Audio

Ignore Presentation Indicators (internal calls only)

Allow Control of Device from CTI

Logged Into Hunt Group

Remote Device

Protocol Specific Information

Done Internet



Protocol Specific Information

Packet Capture Mode*	None
Packet Capture Duration	0
Presence Group*	Standard Presence group
Device Security Profile*	Cisco 7960 - Standard SCCP Non-Secure Profile
SUBSCRIBE Calling Search Space	< None >

Unattended Port
 Require DTMF Reception
 RFC2833 Disabled

Certification Authority Proxy Function (CAPF) Information

Certificate Operation*	No Pending Operation
Authentication Mode*	By Null String
Authentication String	

Key Size (Bits)*	1024
Operation Completes By	2007 9 6 12 (YYYY:MM:DD:HH)

Certificate Operation Status: None
Note: Security Profile Contains Addition CAPF Settings.



Phone Configuration

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

Save Delete Copy Reset Add New

Expansion Module Information

Module 1	<input type="text" value=" < None >"/>
Module 1 Load Name	<input type="text"/>
Module 2	<input type="text" value=" < None >"/>
Module 2 Load Name	<input type="text"/>

External Data Locations Information (Leave blank to use default)

Information	<input type="text"/>
Directory	<input type="text"/>
Messages	<input type="text"/>
Services	<input type="text"/>
Authentication Server	<input type="text"/>
Proxy Server	<input type="text"/>
Idle	<input type="text"/>
Idle Timer (seconds)	<input type="text"/>

Extension Information

Enable Extension Mobility

Log Out Profile



Extension Information

Enable Extension Mobility

Log Out Profile

Log in Time

Log out Time

MLPP Information

MLPP Domain

MLPP Indication*

MLPP Preemption*

Do Not Disturb

Do Not Disturb

DND Option*

DND Incoming Call Alert

Product Specific Configuration Layout

Disable Speakerphone

Disable Speakerphone and Headset

CC-Part *



Phone Configuration

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

Save Delete Copy Reset Add New

Product Specific Configuration Layout



<input type="checkbox"/> Disable Speakerphone	
<input type="checkbox"/> Disable Speakerphone and Headset	
PC Port *	Enabled
Settings Access*	Enabled
Gratuitous ARP*	Enabled
PC Voice VLAN Access*	Enabled
Video Capabilities*	Disabled
Auto Line Select*	Disabled
Web Access*	Enabled

Save Delete Copy Reset Add New



*- indicates required item.



** - Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.



***Note: Security Profile Contains Addition CAPF Settings.



Cisco Unified Communications Manager SCCP Phone Directory Number (Ext 3501) Level Configuration

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Directory Number Configuration Related Links: [Configure Device \(SEP00192F07EE\)](#)

Save Delete Reset Add New

Directory Number Information

Directory Number*

Route Partition

Description

Alerting Name

ASCII Alerting Name

Allow Control of Device from CTI

Associated Devices

▼ ▲

Dissociate Devices

Directory Number Settings

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

Calling Search Space

Presence Group*

User Hold MOH Audio Source



Directory Number Configuration

Related Links: [Configure Device \(SEP00192F07EE\)](#)

Save
 Delete
 Reset
 Add New

User Hold MOH Audio Source

Network Hold MOH Audio Source

Auto Answer*

AAR Settings

	Voice Mail	AAR Destination Mask	AAR Group
AAR	<input type="checkbox"/> or	<input type="text"/>	<input type="text" value=" < None >"/>
<input checked="" type="checkbox"/> Retain this destination in the call forwarding history			

Call Forward and Call Pickup Settings

	Voice Mail	Destination	Calling Search Space
Calling Search Space Activation Policy			<input type="text" value=" Use System Default"/>
Forward All	<input type="checkbox"/> or	<input type="text"/>	<input type="text" value=" < None >"/>
Secondary Calling Search Space for Forward All			<input type="text" value=" < None >"/>
Forward Busy Internal	<input type="checkbox"/> or	<input type="text"/>	<input type="text" value=" < None >"/>
Forward Busy External	<input type="checkbox"/> or	<input type="text"/>	<input type="text" value=" < None >"/>
Forward No Answer Internal	<input type="checkbox"/> or	<input type="text"/>	<input type="text" value=" < None >"/>
Forward No Answer External	<input type="checkbox"/> or	<input type="text"/>	<input type="text" value=" < None >"/>



Directory Number Configuration

Related Links: [Configure Device \(SEP00192F07EE90\)](#)

Save Delete Reset Add New

Forward No Coverage External	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward on CTI Failure	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward Unregistered Internal	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward Unregistered External	<input type="checkbox"/> or	<input type="text"/>	< None >
No Answer Ring Duration (seconds)	<input type="text"/>		
Call Pickup Group	< None > <input type="button" value="v"/>		

MLPP Alternate Party Settings

Target (Destination)	<input type="text"/>
MLPP Calling Search Space	< None > <input type="button" value="v"/>
MLPP No Answer Ring Duration (seconds)	<input type="text"/>

Line Settings for All Devices

Hold Reversion Ring Duration (seconds)	<input type="text"/>	Setting the Hold Reversion Ring Duration to zero will disable the
Hold Reversion Notification Interval (seconds)	<input type="text"/>	Setting the Hold Reversion Notification Interval to zero will disable the

Line 1 on Device SEP00192F07EE90

Display (Internal Caller ID)	<input type="text" value="One Aries"/>	Display text for a line appearance is intended for displaying text such as a name instead of a directory number for internal calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
------------------------------	--	---



Line 1 on Device SEP00192F07EE90

Display (Internal Caller ID)	<input type="text" value="One Aries"/>	Display text for a line appearance is intended for displaying text such as a name instead of a directory number for internal calls. If you specify a number, the person receiving a call may not see the proper identity of caller.
ASCII Display (Internal Caller ID)	<input type="text" value="One Aries"/>	
Line Text Label	<input type="text" value="One Aries"/>	
ASCII Line Text Label	<input type="text" value="One Aries"/>	
External Phone Number Mask	<input type="text"/>	
Visual Message Waiting Indicator Policy*	<input type="button" value="Use System Policy"/>	
Ring Setting (Phone Idle)*	<input type="button" value="Ring"/>	
Ring Setting (Phone Active)	<input type="button" value="Use System Default"/>	Applies to this line when any line on the phone has a call in progress.
Call Pickup Group Audio Alert Setting(Phone Idle)	<input type="button" value="Use System Default"/>	
Call Pickup Group Audio Alert Setting(Phone Active)	<input type="button" value="Use System Default"/>	
Monitoring Calling Search Space	<input type="button" value="< None >"/>	



Directory Number Configuration

Related Links: [Configure Device \(SEP00192F07EE90\)](#)

Save Delete Reset Add New

Monitoring Calling Search Space

Multiple Call/Call Waiting Settings on Device SEP00192F07EE90

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

Maximum Number of Calls*

Busy Trigger* (Less than or equal to Max. Calls)

Forwarded Call Information Display on Device SEP00192F07EE90

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

Users Associated with Line

[Associate End Users](#)

*- indicates required item.



Cisco Unified Communications Manager SIP Phone Level Configuration

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Status
Status: Ready

Association Information

Modify Button Items

1	Line [1] - 3502 (no partition)
2	Line [2] - Add a new DN
3	Add a new SD
4	Add a new SD
5	Add a new SD
6	Add a new SD
----- Unassigned Associated Items -----	
7	Add a new SD
8	Add a new SURF
9	Add a new BLF SD
10	Add a new BLF Directed Call Park
11	Intercom [1] - Add a new Intercom

Phone Type
Product Type: Cisco 7961G-GE
Device Protocol: SIP

Device Information

Registration	Registered with Cisco Unified Communications Manager CM-Aries
IP Address	172.20.66.11
MAC Address*	001955FA0E5E
Description	3502 7961 SIP Phone
Device Pool*	Default View Details
Common Device Configuration	< None > View Details
Phone Button Template*	Standard 7961G-GE SIP
Softkey Template	Standard User with CallBack
Common Phone Profile*	Standard Common Phone Profile
Calling Search Space	< None >
AAR Calling Search Space	< None >
Media Resource Group List	Aries_MRGL



Save Delete Copy Reset Add New

11	Intercom [1] - Add a new Intercom	Media Resource Group List	Aries_MRGL
12	Do Not Disturb	User Hold MOH Audio Source	< None >
13	Privacy	Network Hold MOH Audio Source	< None >
14	None	Location*	Hub_None
		AAR Group	< None >
		User Locale	< None >
		Network Locale	< None >
		Built In Bridge*	Default
		Privacy*	Default
		Device Mobility Mode*	Default View Current Device Mobility Settings
		Owner User ID	< None >
		Phone Personalization*	Default
		Phone Load Name	

Ignore Presentation Indicators (internal calls only)
 Allow Control of Device from CTI
 Logged Into Hunt Group
 Remote Device



Phone Configuration

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

Save Delete Copy Reset Add New

Protocol Specific Information

Packet Capture Mode*	<input type="text" value="None"/>
Packet Capture Duration	<input type="text" value="0"/>
Presence Group*	<input type="text" value="Standard Presence group"/>
SIP Dial Rules	<input type="text" value="< None >"/>
MTP Preferred Originating Codec*	<input type="text" value="711ulaw"/>
Device Security Profile*	<input type="text" value="Cisco 7961G-GE - Standard SIP Non-Secure Profile"/>
Rerouting Calling Search Space	<input type="text" value="< None >"/>
SUBSCRIBE Calling Search Space	<input type="text" value="< None >"/>
SIP Profile*	<input type="text" value="Standard SIP Profile"/>
Digest User	<input type="text" value="< None >"/>
<input type="checkbox"/> Media Termination Point Required	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> Require DTMF Reception	

Certification Authority Proxy Function (CAPF) Information

Certificate Operation*	<input type="text" value="No Pending Operation"/>
Authentication Mode*	<input type="text" value="By Null String"/>
Authentication String	<input type="text"/>



Phone Configuration

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

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Key Size (Bits)*

Operation Completes By (YYYY:MM:DD:HH)

Certificate Operation Status: None

Note: Security Profile Contains Addition CAPF Settings.

External Data Locations Information (Leave blank to use default)

Information	<input type="text"/>
Directory	<input type="text"/>
Messages	<input type="text"/>
Services	<input type="text"/>
Authentication Server	<input type="text"/>
Proxy Server	<input type="text"/>
Idle	<input type="text"/>
Idle Timer (seconds)	<input type="text"/>

Extension Information

Enable Extension Mobility

Log Out Profile

Log in Time < None >



Extension Information

Enable Extension Mobility

Log Out Profile -- Use Current Device Settings --

Log in Time < None >

Log out Time < None >

MLPP Information

MLPP Domain < None >

MLPP Indication* Default

MLPP Preemption* Default

Do Not Disturb

Do Not Disturb

DND Option* Ringer Off

DND Incoming Call Alert < None >

Product Specific Configuration Layout

Disable Speakerphone

Disable Speakerphone and Headset

CCP Part *



Phone Configuration

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

Save Delete Copy Reset Add New

PC Port *	Enabled
Settings Access*	Enabled
Gratuitous ARP*	Enabled
PC Voice VLAN Access*	Enabled
Video Capabilities*	Disabled
Auto Line Select*	Disabled
Web Access*	Enabled
Span to PC Port*	Disabled
Logging Display*	PC Controlled
Load Server	
Recording Tone*	Disabled
Recording Tone Local Volume*	100
Recording Tone Remote Volume*	50
Recording Tone Duration	
RTCP*	Disabled
"more" Soft Key Timer	5
Auto Call Select*	Enabled
Log Server	
Advertise G.722 Codec*	Use System Default



Phone Configuration

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

Save Delete Copy Reset Add New

RTCP*	Disabled
"more" Soft Key Timer	5
Auto Call Select*	Enabled
Log Server	
Advertise G.722 Codec*	Use System Default
Wideband Headset UI Control*	Enabled
Wideband Handset UI Control*	Enabled
Wideband Headset*	Enabled
Wideband Handset*	Use Phone Default
Peer Firmware Sharing*	Disabled
Cisco Discovery Protocol (CDP): Switch Port*	Enabled
Cisco Discovery Protocol (CDP): PC Port*	Enabled

Save Delete Copy Reset Add New



*- indicates required item.



** - Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.



***Note: Security Profile Contains Addition CAPF Settings.



Cisco Unified Communications Manager SIP Phone Directory Number (Ext 3502) Level Configuration

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Directory Number Configuration Related Links: [Configure Device \(SEP001955FA0E5E\)](#)

Save Delete Reset Add New

Directory Number Information

Directory Number*

Route Partition

Description

Alerting Name

ASCII Alerting Name

Allow Control of Device from CTI

Associated Devices

[Edit Device](#)
[Edit Line Appearance](#)

▼ ▲

Dissociate Devices

Directory Number Settings

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

Calling Search Space

Presence Group*



Directory Number Configuration

Related Links: Configure Device (SEP001955FA0E

Save Delete Reset Add New

Presence Group	Standard Presence group
User Hold MOH Audio Source	< None >
Network Hold MOH Audio Source	< None >
Auto Answer*	Auto Answer Off

AAR Settings

	Voice Mail	AAR Destination Mask	AAR Group
AAR	<input type="checkbox"/> or		< None >
<input checked="" type="checkbox"/> Retain this destination in the call forwarding history			

Call Forward and Call Pickup Settings

	Voice Mail	Destination	Calling Search Space
Calling Search Space Activation Policy			Use System Default
Forward All	<input type="checkbox"/> or		< None >
Secondary Calling Search Space for Forward All			< None >
Forward Busy Internal	<input type="checkbox"/> or		< None >
Forward Busy External	<input type="checkbox"/> or		< None >
Forward No Answer Internal	<input type="checkbox"/> or		< None >
Forward No Answer External	<input type="checkbox"/> or		< None >



Directory Number Configuration

Related Links: [Configure Device \(SEP001955FA0E\)](#)

Save Delete Reset Add New

Forward No Coverage Internal	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward No Coverage External	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward on CTI Failure	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward Unregistered Internal	<input type="checkbox"/> or	2215	< None >
Forward Unregistered External	<input type="checkbox"/> or	2215	< None >
No Answer Ring Duration (seconds)	<input type="text"/>		
Call Pickup Group	< None >		

MLPP Alternate Party Settings

Target (Destination)	<input type="text"/>
MLPP Calling Search Space	< None >
MLPP No Answer Ring Duration (seconds)	<input type="text"/>

Line Settings for All Devices

Hold Reversion Ring Duration (seconds)	<input type="text"/>	Setting the Hold Reversion Ring Duration to zero will disable the
Hold Reversion Notification Interval (seconds)	<input type="text"/>	Setting the Hold Reversion Notification Interval to zero will disat

feature

Line 1 on Device SEP001955FA0E5E



Save Delete Reset Add New

Line 1 on Device SEP001955FA0E5E

Display (Internal Caller ID)	<input type="text" value="Two Aries"/>	Display text for a line appearance is intended for displaying text such as a name instead of a directory number for internal calls. If you specify a number, the person receiving a call may not see the proper identity of caller.
ASCII Display (Internal Caller ID)	<input type="text" value="Two Aries"/>	
Line Text Label	<input type="text" value="Two Aries"/>	
ASCII Line Text Label	<input type="text" value="Two Aries"/>	
External Phone Number Mask	<input type="text"/>	
Visual Message Waiting Indicator Policy*	<input type="button" value="Use System Policy"/>	
Audible Message Waiting Indicator Policy*	<input type="button" value="Off"/>	
Ring Setting (Phone Idle)*	<input type="button" value="Use System Default"/>	
Ring Setting (Phone Active)	<input type="button" value="Use System Default"/>	Applies to this line when any line on the phone has a call in progress.
Call Pickup Group Audio Alert Setting(Phone Idle)	<input type="button" value="Use System Default"/>	
Call Pickup Group Audio Alert Setting(Phone Active)	<input type="button" value="Use System Default"/>	
Recording Option*	<input type="button" value="Call Recording Disabled"/>	



Directory Number Configuration Related Links: [Configure Device \(SEP001955FA0E\)](#)

Save Delete Reset Add New

Recording Option*	<input type="text" value="Call Recording Disabled"/>
Recording Profile	<input type="text" value="< None >"/>
Monitoring Calling Search Space	<input type="text" value="< None >"/>

Multiple Call/Call Waiting Settings on Device SEP001955FA0E5E

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

Maximum Number of Calls*	<input type="text" value="4"/>
Busy Trigger*	<input type="text" value="2"/> (Less than or equal to Max. Calls)

Forwarded Call Information Display on Device SEP001955FA0E5E

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

Users Associated with Line



Cisco Unified Communications Manager Route Pattern (22XX) to Nortel PBX extensions Configuration

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation | Cisco Unified CM Administration
CCMAdministrator | About

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Route Pattern Configuration Related Links: [Back To Find](#)

Save Delete Copy Add New

Pattern Definition

Route Pattern*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence*

Gateway/Route List* [\(Edit\)](#)

Route Option

Route this pattern

Block this pattern

Call Classification*

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

Require Forced Authorization Code

Authorization Level*

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask



Route Pattern Configuration

Related Links: [Back To Find](#)

Save Delete Copy Add New

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Calling Name Presentation*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

Network Service	Service Parameter Name	Service Parameter Value
-----------------	------------------------	-------------------------



Route Pattern Configuration

Related Links: [Back To Find/](#)

Save Delete Copy Add New

Calling Name Presentation*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

Network Service	Service Parameter Name	Service Parameter Value
<input type="text" value="-- Not Selected --"/>	<input type="text" value="< Not Exist >"/>	<input type="text"/>

*- indicates required item.



Acronyms

Acronym	Definitions
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
CUCM	Cisco Unified Communications Manager
DNS	Domain Name Server
FQDN	Fully Qualified Domain Name
MWI	Message Waiting Indicator
PSTN	Public Switched Telephone Network
SIP	Session Initiated Protocol

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Test Results

Testing was performed by Filli Cruz (August 2007)

Table 1. Basic Calls (Enbloc Sending)

Calls Made	Call Comp.?	Calling Number Displayed on Final Destination?	Calling Name Displayed on Final Destination?	Called Number Displayed on Orig. Side?	Called Name Displayed on Orig. Side?
Phone A1 to Phone C1	Yes	Yes ¹	Yes ¹	Yes ²	No
Phone A1 to Phone C2	Yes	Yes ¹	Yes ¹	Yes ²	No
Phone C1 to Phone A1	Yes	Yes ¹	Yes ¹	Yes ²	No
Phone C2 to Phone A1	Yes	Yes ¹	Yes ¹	Yes ²	No
Phone A2 to Phone C1	Yes	Yes ¹	Yes ¹	Yes ²	No
Phone A2 to Phone C2	Yes	Yes ¹	Yes ¹	Yes ²	No
Phone C1 to Phone A2	Yes	Yes ¹	Yes ¹	Yes ²	No
Phone C2 to Phone A2	Yes	Yes ¹	Yes ¹	Yes ²	No

¹ Nortel use P-Asserted-Id (PAI) field to pass the name and number information across SIP Trunk while Cisco Unified Communications Manager (CUCM) uses the Remote-Party-Id (RPID).

² Original calling party phone display dialed number



Table 2. Basic Calls with Calling Name and Number Restrictions

Calls Made	Call Comp.?	Calling Number Restriction was honored at Final Destination?	Calling Name Restriction was honored at Final Destination?	Called Number Displayed on Orig. Side?	Called Name Displayed on Orig. Side?
Phone A1 to Phone C1 Phone A1 restricting Calling Name and Number	Yes	Yes ³	Yes ³	Yes ²	No
Phone A1 to Phone C2 Phone A1 restricting Calling Name and Number	Yes	Yes ³	Yes ³	Yes ²	No
Phone C1 to Phone A1 Phone C1 restricting Calling Name and Number	Yes	Yes ³	Yes ³	Yes ²	No
Phone C2 to Phone A1 Phone C2 restricting Calling Name and Number	Yes	Yes ³	Yes ³	Yes ²	No
Phone A2 to Phone C1 Phone A2 restricting Calling Name and Number	Yes	Yes ³	Yes ³	Yes ²	No
Phone A2 to Phone C2 Phone A2 restricting Calling Name and Number	Yes	Yes ³	Yes ³	Yes ²	No
Phone C1 to Phone A2 Phone C1 restricting Calling Name and Number	Yes	Yes ³	Yes ³	Yes ²	No
Phone C2 to Phone A2 Phone C2 restricting Calling Name and Number	Yes	Yes ³	Yes ³	Yes ²	No

³ Nortel use P-Asserted-Id (PAI) field to pass the name and number information across SIP Trunk while Cisco Unified Communications Manager (CUCM) uses the Remote-Party-Id (RPID). When restricting calling and name and number, the Nortel sets privacy parameter in PAI field as USER versus NONE when name and number is allowed. On the other hand, CUCM sets privacy parameter in its RPID field as FULL when restricting name and number versus OFF when name and number is allowed. Due to this difference, both systems extracted the calling party name and number information from the SIP From header of the incoming SIP INVITE message which has <sip:anonymous@anonymous.invalid> from Nortel and <sip:anonymous@anonymous.ip address> .from CUCM.

For Nortel PBX name and number restriction, change the phone CLS setting to DDGD and NAMD under LD11. For CUCM name and number restriction, this is done on the SIP trunk.



Table 3. Basic Calls with Connected Name and Number Restrictions

Calls Made	Call Comp.?	Calling Number Displayed on Final Destination?	Calling Name Displayed on Final Destination?	Connected Number Restriction was honored at Orig. Side?	Connected Name Restriction was honored at Orig. Side?	Notes
Phone A1 to Phone C1 Phone C1 restricting Connected Name and Number	Yes	Yes	Yes	No ⁴	No ⁴	
Phone A1 to Phone C2 Phone C2 restricting Connected Name and Number	Yes	Yes	Yes	No ⁴	No ⁴	
Phone C1 to Phone A1 Phone A1 restricting Connected Name and Number	Yes	Yes	Yes	No ⁴	No ⁴	
Phone C2 to Phone A1 Phone A1 restricting Connected Name and Number	Yes	Yes	Yes	No ⁴	No ⁴	
Phone A2 to Phone C1 Phone C1 restricting Connected Name and Number	Yes	Yes	Yes	No ⁴	No ⁴	
Phone A2 to Phone C2 Phone C2 restricting Connected Name and Number	Yes	Yes	Yes	No ⁴	No ⁴	
Phone C1 to Phone A2 Phone A2 restricting Connected Name and Number	Yes	Yes	Yes	No ⁴	No ⁴	
Phone C2 to Phone A2 Phone A2 restricting Connected Name and Number	Yes	Yes	Yes	No ⁴	No ⁴	

⁴ CUCM restriction is set under the SIP Trunk configuration. RPID privacy parameter is set to FULL. (Note: When name and number is allowed, privacy parameter is set to OFF). Nortel restriction is set under the phone CLS setting to DDGD and NAMD under LD11. PAI privacy parameter is set to USER. (Note: When name and number is allowed, privacy parameter is set to NONE). Due to the difference in the name and number delivery method, connected name and number restriction is not honored between the two systems.



Table 4. Alerting Name

Calls Made	Call Setup Comp.?	Alerting Name was sent by Final Destination during Alerting (ringing)?	Alerting Name was displayed on Orig. Side during Alerting (ringing)?
Phone A1 to Phone C1 Phone C1 does not answer	Yes	Yes	No ¹
Phone A1 to Phone C2 Phone C2 does not answer	Yes	Yes	No ¹
Phone C1 to Phone A1 Phone A1 does not answer	Yes	No ⁵	No ¹
Phone C2 to Phone A1 Phone A1 does not answer	Yes	No ⁵	No ¹
Phone A2 to Phone C1 Phone C1 does not answer	Yes	Yes	No ¹
Phone A2 to Phone C2 Phone C2 does not answer	Yes	Yes	No ¹
Phone C1 to Phone A2 Phone A2 does not answer	Yes	No ⁵	No ¹
Phone C2 to Phone A2 Phone A2 does not answer	Yes	No ⁵	No ¹

⁵ Nortel does NOT send PAI information in its 180 Ringing packet back to CUCM



Table 5. Call Transfers (Attended Local Transfers)

Calls Made	Call Comp?	Orig. Calling Number Displayed on Final Dest. Phone?	Orig. Calling Name Displayed on Final Dest. Phone?	Called Number Display on Orig. phone Updated After Transfer?	Called Name Display on Orig. Phone Updated After Transfer?
Phone C1 to Phone A1 Xfr to Phone B1	Yes	Yes	Yes	No	No
Phone C1 to Phone A1 Xfr to Phone B2	Yes	Yes	Yes	No	No
Phone C2 to Phone A1 Xfr to Phone B1	Yes	Yes	Yes	No	No
Phone C2 to Phone A1 Xfr to Phone B2	Yes	Yes	Yes	No	No
Phone C1 to Phone A2 Xfr to Phone B1	Yes	Yes	Yes	No	No
Phone C1 to Phone A2 Xfr to Phone B2	Yes	Yes	Yes	No	No
Phone C2 to Phone A2 Xfr to Phone B1	Yes	Yes	Yes	No	No
Phone C2 to Phone A2 Xfr to Phone B2	Yes	Yes	Yes	No	No
Phone A1 to Phone C1 Xfr to Phone D1	Yes	Yes	Yes	No	No
Phone A1 to Phone C1 Xfr to Phone D2	Yes	Yes	Yes	No	No
Phone A1 to Phone C2 Xfr to Phone D1	Yes	Yes	Yes	No	No
Phone A1 to Phone C2 Xfr to Phone D2	Yes	Yes	Yes	No	No
Phone A2 to Phone C1 Xfr to Phone D1	Yes	Yes	Yes	No	No
Phone A2 to Phone C1 Xfr to Phone D2	Yes	Yes	Yes	No	No
Phone A2 to Phone C2 Xfr to Phone D1	Yes	Yes	Yes	No	No
Phone A2 to Phone C2 Xfr to Phone D2	Yes	Yes	Yes	No	No



Table 6. Call Transfers (Early Attended Local Transfers)

Calls Made	Call Comp?	Orig. Calling Number Displayed on Final Dest. Phone?	Orig. Calling Name Displayed on Final Dest. Phone?	Called Number Display on Orig. phone Updated After Transfer?	Called Name Display on Orig. Phone Updated After Transfer?
Phone C1 to Phone A1 Xfr to Phone B1	Yes	Yes	Yes	No	No
Phone C1 to Phone A1 Xfr to Phone B2	Yes	Yes	Yes	No	No
Phone C2 to Phone A1 Xfr to Phone B1	Yes	Yes	Yes	No	No
Phone C2 to Phone A1 Xfr to Phone B2	Yes	Yes	Yes	No	No
Phone C1 to Phone A2 Xfr to Phone B1	Yes	Yes	Yes	No	No
Phone C1 to Phone A2 Xfr to Phone B2	Yes	Yes	Yes	No	No
Phone C2 to Phone A2 Xfr to Phone B1	Yes	Yes	Yes	No	No
Phone C2 to Phone A2 Xfr to Phone B2	Yes	Yes	Yes	No	No
Phone A1 to Phone C1 Xfr to Phone D1	Yes	Yes	Yes	No	No
Phone A1 to Phone C1 Xfr to Phone D2	Yes	Yes	Yes	No	No
Phone A1 to Phone C2 Xfr to Phone D1	Yes	Yes	Yes	No	No
Phone A1 to Phone C2 Xfr to Phone D2	Yes	Yes	Yes	No	No
Phone A2 to Phone C1 Xfr to Phone D1	Yes	Yes	Yes	No	No
Phone A2 to Phone C1 Xfr to Phone D2	Yes	Yes	Yes	No	No
Phone A2 to Phone C2 Xfr to Phone D1	Yes	Yes	Yes	No	No
Phone A2 to Phone C2 Xfr to Phone D2	Yes	Yes	Yes	No	No



Table 7. Call Transfers (Blind Local)

Calls Made	Call Comp?	Orig. Calling Number Displayed on Final Dest. Phone?	Orig. Calling Name Displayed on Final Dest. Phone?	Called Number Display on Orig. phone Updated After Transfer?	Called Name Display on Orig. Phone Updated After Transfer?
Phone C1 to Phone A1 Xfr to Phone B1	N/A ⁶				
Phone C1 to Phone A1 Xfr to Phone B2	N/A ⁶				
Phone C2 to Phone A1 Xfr to Phone B1	N/A ⁶				
Phone C2 to Phone A1 Xfr to Phone B2	N/A ⁶				
Phone C1 to Phone A2 Xfr to Phone B1	N/A ⁶				
Phone C1 to Phone A2 Xfr to Phone B2	N/A ⁶				
Phone C2 to Phone A2 Xfr to Phone B1	N/A ⁶				
Phone C2 to Phone A2 Xfr to Phone B2	N/A ⁶				
Phone A1 to Phone C1 Xfr to Phone D1	N/A ⁶				
Phone A1 to Phone C1 Xfr to Phone D2	N/A ⁶				
Phone A1 to Phone C2 Xfr to Phone D1	Yes	Yes	Yes	No	No
Phone A1 to Phone C2 Xfr to Phone D2	Yes	Yes	Yes	No	No
Phone A2 to Phone C1 Xfr to Phone D1	N/A ⁶				
Phone A2 to Phone C1 Xfr to Phone D2	N/A ⁶				
Phone A2 to Phone C2 Xfr to Phone D1	Yes	Yes	Yes	No	No
Phone A2 to Phone C2 Xfr to Phone D2	Yes	Yes	Yes	No	No

⁶ Nortel does not support Blind Call Transfer feature



Table 8. Call Transfers (Attended Network/External)

Calls Made	Call Comp.?	Orig. Calling Number Displayed on Final Dest. Phone?	Orig. Calling Name Displayed on Final Dest. Phone?	Called Number Display on Orig. Phone Updated After Transfer?	Called Name Display on Orig. Phone Updated After Transfer?
Phone C1 to Phone A1 Xfr to Phone D1	Yes	No	No	No	No
Phone C1 to Phone A1 Xfr to Phone D2	Yes	No	No	No	No
Phone C2 to Phone A1 Xfr to Phone D1	Yes	No	No	No	No
Phone C2 to Phone A1 Xfr to Phone D2	Yes	No	No	No	No
Phone C1 to Phone A2 Xfr to Phone D1	Yes	No	No	No	No
Phone C1 to Phone A2 Xfr to Phone D2	Yes	No	No	No	No
Phone C2 to Phone A2 Xfr to Phone D1	Yes	No	No	No	No
Phone C2 to Phone A2 Xfr to Phone D2	Yes	No	No	No	No
Phone A1 to Phone C1 Xfr to Phone B1	Yes	No	No	No	No
Phone A1 to Phone C1 Xfr to Phone B2	Yes	No	No	No	No
Phone A2 to Phone C1 Xfr to Phone B1	Yes	No	No	No	No
Phone A2 to Phone C1 Xfr to Phone B2	Yes	No	No	No	No
Phone A1 to Phone C2 Xfr to Phone B1	Yes	No	No	No	No
Phone A1 to Phone C2 Xfr to Phone B2	Yes	No	No	No	No
Phone A2 to Phone C2 Xfr to Phone B1	Yes	No	No	No	No
Phone A2 to Phone C2 Xfr to Phone B2	Yes	No	No	No	No



Table 9. Call Transfers (Early Attend Network/External)

Calls Made	Call Comp.?	Orig. Calling Number Displayed on Final Dest. Phone?	Orig. Calling Name Displayed on Final Dest. Phone?	Called Number Display on Orig. Phone Updated After Transfer?	Called Name Display on Orig. Phone Updated After Transfer?
Phone C1 to Phone A1 Xfr to Phone D1	Yes	No	No	No	No
Phone C1 to Phone A1 Xfr to Phone D2	Yes	No	No	No	No
Phone C2 to Phone A1 Xfr to Phone D1	Yes	No	No	No	No
Phone C2 to Phone A1 Xfr to Phone D2	Yes	No	No	No	No
Phone C1 to Phone A2 Xfr to Phone D1	Yes	No	No	No	No
Phone C1 to Phone A2 Xfr to Phone D2	Yes	No	No	No	No
Phone C2 to Phone A2 Xfr to Phone D1	Yes	No	No	No	No
Phone C2 to Phone A2 Xfr to Phone D2	Yes	No	No	No	No
Phone A1 to Phone C1 Xfr to Phone B1	Yes	No	No	No	No
Phone A1 to Phone C1 Xfr to Phone B2	Yes	No	No	No	No
Phone A2 to Phone C1 Xfr to Phone B1	Yes	No	No	No	No
Phone A2 to Phone C1 Xfr to Phone B2	Yes	No	No	No	No
Phone A1 to Phone C2 Xfr to Phone B1	Yes	No	No	No	No
Phone A1 to Phone C2 Xfr to Phone B2	Yes	No	No	No	No
Phone A2 to Phone C2 Xfr to Phone B1	Yes	No	No	No	No
Phone A2 to Phone C2 Xfr to Phone B2	Yes	No	No	No	No



Table 10. Call Transfers (Blind Network/Network)

Calls Made	Call Comp.?	Orig. Calling Number Displayed on Final Dest. Phone?	Orig. Calling Name Displayed on Final Dest. Phone?	Called Number Display on Orig. Phone Updated After Transfer?	Called Name Display on Orig. Phone Updated After Transfer?
Phone C1 to Phone A1 Xfr to Phone D1	N/A ⁶				
Phone C1 to Phone A1 Xfr to Phone D2	N/A ⁶				
Phone C2 to Phone A1 Xfr to Phone D1	N/A ⁶				
Phone C2 to Phone A1 Xfr to Phone D2	N/A ⁶				
Phone C1 to Phone A2 Xfr to Phone D1	N/A ⁶				
Phone C1 to Phone A2 Xfr to Phone D2	N/A ⁶				
Phone C2 to Phone A2 Xfr to Phone D1	N/A ⁶				
Phone C2 to Phone A2 Xfr to Phone D2	N/A ⁶				
Phone A1 to Phone C1 Xfr to Phone B1	N/A ⁶				
Phone A1 to Phone C1 Xfr to Phone B2	N/A ⁶				
Phone A2 to Phone C1 Xfr to Phone B1	N/A ⁶				
Phone A2 to Phone C1 Xfr to Phone B2	N/A ⁶				
Phone A1 to Phone C2 Xfr to Phone B1	Yes	Yes	Yes	No	No
Phone A1 to Phone C2 Xfr to Phone B2	Yes	Yes	Yes	No	No
Phone A2 to Phone C2 Xfr to Phone B1	Yes	Yes	Yes	No	No
Phone A2 to Phone C2 Xfr to Phone B2	Yes	Yes	Yes	No	No



Table 11. Call Forward Unconditional (Local)

Calls Made	Call Comp.?	Original Calling Number Displayed on Final Dest.?	Original Calling Name Displayed on Final Dest.?	Forwarding Called Number Displayed on Final Dest.?	Forwarding Called Name Displayed on Final Dest.?	Final Dest. Connected Number Updated at orig. Side?	Final Dest. "Connected Name" Updated at Orig. Side?
Phone C1 to Phone A1 -CFU to Phone B1	Yes	Yes	Yes	Yes	No	No	No
Phone C1 to Phone A1 -CFU to Phone B2	Yes	Yes	No	Yes	Yes	No	No
Phone C2 to Phone A1 -CFU to Phone B1	Yes	Yes	Yes	Yes	No	No	No
Phone C2 to Phone A1 -CFU to Phone B2	Yes	Yes	No	Yes	Yes	No	No
Phone C1 to Phone A2 -CFU to Phone B1	Yes	Yes	Yes	Yes	No	No	No
Phone C1 to Phone A2 -CFU to Phone B2	Yes	Yes	No	Yes	Yes	No	No
Phone C2 to Phone A2 -CFU to Phone B1	Yes	Yes	Yes	Yes	No	No	No
Phone C2 to Phone A2 -CFU to Phone B2	Yes	Yes	No	Yes	Yes	No	No
Phone A1 to Phone C1 -CFU to Phone D1	Yes	Yes	Yes	Yes	Yes	No	No
Phone A1 to Phone C1 -CFU to Phone D2	Yes	Yes	Yes	Yes	Yes	No	No
Phone A1 to Phone C2 -CFU to Phone D1	Yes	Yes	Yes	Yes	Yes	No	No
Phone A1 to Phone C2 -CFU to Phone D2	Yes	Yes	Yes	Yes	Yes	No	No
Phone A2 to Phone C1 -CFU to Phone D1	Yes	Yes	Yes	Yes	Yes	No	No
Phone A2 to Phone C1 -CFU to Phone D2	Yes	Yes	Yes	Yes	Yes	No	No
Phone A2 to Phone C2 -CFU to Phone D1	Yes	Yes	Yes	Yes	Yes	No	No
Phone A2 to Phone C2 -CFU to Phone D2	Yes	Yes	Yes	Yes	Yes	No	No



Table 12. Call Forward Busy (Local)

Calls Made	Call Comp.?	Original Calling Number Displayed on Final Dest.?	Original Calling Name Displayed on Final Dest.?	Forwarding Called Number Displayed on Final Dest.?	Forwarding Called Name Displayed on Final Dest.?	Final Dest. Connected Number Updated at orig. Side?	Final Dest. "Connected Name" Updated at Orig. Side?
Phone C1 to Phone A1 –CFB to Phone B1	Yes	Yes	Yes	Yes	No	No	No
Phone C1 to Phone A1 –CFB to Phone B2	Yes	Yes	No	Yes	Yes	No	No
Phone C2 to Phone A1 –CFB to Phone B1	Yes	Yes	Yes	Yes	No	No	No
Phone C2 to Phone A1 –CFB to Phone B2	Yes	Yes	No	Yes	Yes	No	No
Phone C1 to Phone A2 –CFB to Phone B1	Yes	Yes	Yes	Yes	No	No	No
Phone C1 to Phone A2 –CFB to Phone B2	Yes	Yes	No	Yes	Yes	No	No
Phone C2 to Phone A2 –CFB to Phone B1	Yes	Yes	Yes	Yes	No	No	No
Phone C2 to Phone A2 –CFB to Phone B2	Yes	Yes	No	Yes	Yes	No	No
Phone A1 to Phone C1 –CFB to Phone D1	Yes	Yes	Yes	Yes	Yes	No	No
Phone A1 to Phone C1 –CFB to Phone D2	Yes	Yes	Yes	Yes	Yes	No	No
Phone A1 to Phone C2 –CFB to Phone D1	Yes	Yes	Yes	Yes	Yes	No	No
Phone A1 to Phone C2 –CFB to Phone D2	Yes	Yes	Yes	Yes	Yes	No	No
Phone A2 to Phone C1 –CFB to Phone D1	Yes	Yes	Yes	Yes	Yes	No	No
Phone A2 to Phone C1 –CFB to Phone D2	Yes	Yes	Yes	Yes	Yes	No	No
Phone A2 to Phone C2 –CFB to Phone D1	Yes	Yes	Yes	Yes	Yes	No	No
Phone A2 to Phone C2 –CFB to Phone D2	Yes	Yes	Yes	Yes	Yes	No	No



Table 13. Call Forward No Answer (Local)

Calls Made	Call Comp.?	Original Calling Number Displayed on Final Dest.?	Original Calling Name Displayed on Final Dest.?	Forwarding Called Number Displayed on Final Dest.?	Forwarding Called Name Displayed on Final Dest.?	Final Dest. Connected Number Updated at orig. Side?	Final Dest. "Connected Name" Updated at Orig. Side?
Phone C1 to Phone A1 -CFNA to Phone B1	Yes	Yes	Yes	Yes	No	No	No
Phone C1 to Phone A1 -CFNA to Phone B2	Yes	Yes	No	Yes	Yes	No	No
Phone C2 to Phone A1 -CFNA to Phone B1	Yes	Yes	Yes	Yes	No	No	No
Phone C2 to Phone A1 -CFNA to Phone B2	Yes	Yes	No	Yes	Yes	No	No
Phone C1 to Phone A2 -CFNA to Phone B1	Yes	Yes	Yes	Yes	No	No	No
Phone C1 to Phone A2 -CFNA to Phone B2	Yes	Yes	No	Yes	Yes	No	No
Phone C2 to Phone A2 -CFNA to Phone B1	Yes	Yes	Yes	Yes	No	No	No
Phone C2 to Phone A2 -CFNA to Phone B2	Yes	Yes	No	Yes	Yes	No	No
Phone A1 to Phone C1 -CFNA to Phone D1	Yes	Yes	Yes	Yes	Yes	No	No
Phone A1 to Phone C1 -CFNA to Phone D2	Yes	Yes	Yes	Yes	Yes	No	No
Phone A1 to Phone C2 -CFNA to Phone D1	Yes	Yes	Yes	Yes	Yes	No	No
Phone A1 to Phone C2 -CFNA to Phone D2	Yes	Yes	Yes	Yes	Yes	No	No
Phone A2 to Phone C1 -CFNA to Phone D1	Yes	Yes	Yes	Yes	Yes	No	No
Phone A2 to Phone C1 -CFNA to Phone D2	Yes	Yes	Yes	Yes	Yes	No	No
Phone A2 to Phone C2 -CFNA to Phone D1	Yes	Yes	Yes	Yes	Yes	No	No
Phone A2 to Phone C2 -CFNA to Phone D2	Yes	Yes	Yes	Yes	Yes	No	No



Table 14. Call Forward Unconditional (Network/External)

Calls Made	Call Comp.?	Original Calling Number Displayed on Final Dest.?	Original Calling Name Displayed on Final Dest.?	Forwarding Called Number Displayed on Final Dest.?	Forwarding Called Name Displayed on Final Dest.?	Final Dest. Connected Number Updated at Orig. Side?	Final Dest. Connected Name Updated at Orig. Side?
Phone C1 to Phone A1 -CFU to Phone D1	Yes	Yes	Yes	No	No	No	No
Phone C1 to Phone A1 -CFU to Phone D2	Yes	Yes	Yes	No	No	No	No
Phone C2 to Phone A1 -CFU to Phone D1	Yes	Yes	Yes	No	No	No	No
Phone C2 to Phone A1 -CFU to Phone D2	Yes	Yes	Yes	No	No	No	No
Phone C1 to Phone A2 -CFU to Phone D1	Yes	Yes	Yes	No	No	No	No
Phone C1 to Phone A2 -CFU to Phone D2	Yes	Yes	Yes	No	No	No	No
Phone C2 to Phone A2 -CFU to Phone D1	Yes	Yes	Yes	No	No	No	No
Phone C2 to Phone A2 -CFU to Phone D2	Yes	Yes	Yes	No	No	No	No
Phone A1 to Phone C1 -CFU to Phone B1	Yes	Yes	Yes	No	No	No	No
Phone A1 to Phone C1 -CFU to Phone B2	Yes	Yes	Yes	No	No	No	No
Phone A2 to Phone C1 -CFU to Phone B1	Yes	Yes	Yes	No	No	No	No
Phone A2 to Phone C1 -CFU to Phone B2	Yes	Yes	Yes	No	No	No	No
Phone A1 to Phone C2 -CFU to Phone B1	Yes	Yes	Yes	No	No	No	No
Phone A1 to Phone C2 -CFU to Phone B2	Yes	Yes	Yes	No	No	No	No
Phone A2 to Phone C2 -CFU to Phone B1	Yes	Yes	Yes	No	No	No	No
Phone A2 to Phone C2 -CFU to Phone B2	Yes	Yes	Yes	No	No	No	No



Table 15. Call Forward Busy (Network/External)

Calls Made	Call Comp.?	Original Calling Number Displayed on Final Dest.?	Original Calling Name Displayed on Final Dest.?	Forwarding Called Number Displayed on Final Dest.?	Forwarding Called Name Displayed on Final Dest.?	Final Dest. Connected Number Updated at Orig. Side?	Final Dest. Connected Name Updated at Orig. Side?
Phone C1 to Phone A1 -CFU to Phone D1	Yes	Yes	Yes	No	No	No	No
Phone C1 to Phone A1 -CFU to Phone D2	Yes	Yes	Yes	No	No	No	No
Phone C2 to Phone A1 -CFU to Phone D1	Yes	Yes	Yes	No	No	No	No
Phone C2 to Phone A1 -CFU to Phone D2	Yes	Yes	Yes	No	No	No	No
Phone C1 to Phone A2 -CFU to Phone D1	Yes	Yes	Yes	No	No	No	No
Phone C1 to Phone A2 -CFU to Phone D2	Yes	Yes	Yes	No	No	No	No
Phone C2 to Phone A2 -CFU to Phone D1	Yes	Yes	Yes	No	No	No	No
Phone C2 to Phone A2 -CFU to Phone D2	Yes	Yes	Yes	No	No	No	No
Phone A1 to Phone C1 -CFU to Phone B1	Yes	Yes	Yes	No	No	No	No
Phone A1 to Phone C1 -CFU to Phone B2	Yes	Yes	Yes	No	No	No	No
Phone A2 to Phone C1 -CFU to Phone B1	Yes	Yes	Yes	No	No	No	No
Phone A2 to Phone C1 -CFU to Phone B2	Yes	Yes	Yes	No	No	No	No
Phone A1 to Phone C2 -CFU to Phone B1	Yes	Yes	Yes	No	No	No	No
Phone A1 to Phone C2 -CFU to Phone B2	Yes	Yes	Yes	No	No	No	No
Phone A2 to Phone C2 -CFU to Phone B1	Yes	Yes	Yes	No	No	No	No
Phone A2 to Phone C2 -CFU to Phone B2	Yes	Yes	Yes	No	No	No	No



Table 16. Call Forward No Reply by join (Network/External)

Calls Made	Call Comp.?	Original Calling Number Displayed on Final Dest.?	Original Calling Name Displayed on Final Dest.?	Forwarding Called Number Displayed on Final Dest.?	Forwarding Called Name Displayed on Final Dest.?	Final Dest. Connected Number Updated at Orig. Side?	Final Dest. Connected Name Updated at Orig. Side?
Phone C1 to Phone A1 -CFU to Phone D1	Yes	Yes	Yes	No	No	No	No
Phone C1 to Phone A1 -CFU to Phone D2	Yes	Yes	Yes	No	No	No	No
Phone C2 to Phone A1 -CFU to Phone D1	Yes	Yes	Yes	No	No	No	No
Phone C2 to Phone A1 -CFU to Phone D2	Yes	Yes	Yes	No	No	No	No
Phone C1 to Phone A2 -CFU to Phone D1	Yes	Yes	Yes	No	No	No	No
Phone C1 to Phone A2 -CFU to Phone D2	Yes	Yes	Yes	No	No	No	No
Phone C2 to Phone A2 -CFU to Phone D1	Yes	Yes	Yes	No	No	No	No
Phone C2 to Phone A2 -CFU to Phone D2	Yes	Yes	Yes	No	No	No	No
Phone A1 to Phone C1 -CFU to Phone B1	Yes	Yes	Yes	No	No	No	No
Phone A1 to Phone C1 -CFU to Phone B2	Yes	Yes	Yes	No	No	No	No
Phone A2 to Phone C1 -CFU to Phone B1	Yes	Yes	Yes	No	No	No	No
Phone A2 to Phone C1 -CFU to Phone B2	Yes	Yes	Yes	No	No	No	No
Phone A1 to Phone C2 -CFU to Phone B1	Yes	Yes	Yes	No	No	No	No
Phone A1 to Phone C2 -CFU to Phone B2	Yes	Yes	Yes	No	No	No	No
Phone A2 to Phone C2 -CFU to Phone B1	Yes	Yes	Yes	No	No	No	No
Phone A2 to Phone C2 -CFU to Phone B2	Yes	Yes	Yes	No	No	No	No



Table 17. Hold and Resume with MoH

Calls Made	Hold Work ?	MoH Heard ?	Resume Work ?
Phone C1 to Phone A1 A1 perform Hold/Resume	Yes	No ⁷	Yes
Phone C1 to Phone A2 A2 perform Hold/Resume	Yes	No ⁷	Yes
Phone C2 to Phone A1 A1 perform Hold/Resume	Yes	No ⁷	Yes
Phone C2 to Phone A2 A2 perform Hold/Resume	Yes	No ⁷	Yes
Phone A1 to Phone C1 C1 perform Hold/Resume	Yes	Yes	Yes
Phone A1 to Phone C2 C2 perform Hold/Resume	Yes	Yes	Yes
Phone A2 to Phone C1 C1 perform Hold/Resume	Yes	Yes	Yes
Phone A2 to Phone C2 C2 perform Hold/Resume	Yes	Yes	Yes

⁷ Nortel PBX used is not equipped with MoH source



Table 18. Call Conferencing (Local)

Calls Made	Call Sustained?	Number Updated on the Remaining Parties When the Conferencing Phone Drops Out?	Name Updated on the Remaining Parties When the Conferencing Phone Drops Out?
Phone C1 call Phone A1, Phone A1 conf Phone B1	Yes	No ⁸	No ⁸
Phone C1 call Phone A2, Phone A2 conf Phone B2	Yes	No ⁸	No ⁸
Phone C2 call Phone A1, Phone A1 conf Phone B2	Yes	No ⁸	No ⁸
Phone C2 call Phone A2, Phone A2 conf Phone B1	Yes	No ⁸	No ⁸
Phone A1 call Phone C1, Phone C1 conf Phone D1	Yes	No ⁸	No ⁸
Phone A1 call Phone C2, Phone C2 conf Phone D2	Yes	No ⁸	No ⁸
Phone A2 call Phone C1, Phone C1 conf Phone D2	Yes	No ⁸	No ⁸
Phone A2 call Phone C2, Phone C2 conf Phone D1	Yes	No ⁸	No ⁸

⁸ Name and number display updated on the conferenced party but not on original party display



Table 19. Call Conferencing (Network/External)

Calls Made	Call Sustained?	Number Updated on the Remaining Parties When the Conferencing Phone Drops Out?	Name Updated on the Remaining Parties When the Conferencing Phone Drops Out?
Phone C1 call Phone A1, Phone A1 conf Phone D1	Yes	No	No
Phone C1 call Phone A2, Phone A2 conf Phone D2	Yes	No	No
Phone C2 call Phone A1, Phone A1 conf Phone D2	Yes	No	No
Phone C2 call Phone A2, Phone A2 conf Phone D1	Yes	No	No
Phone A1 call Phone C1, Phone C1 conf Phone B1	Yes	No	No
Phone A1 call Phone C2, Phone C2 conf Phone B2	Yes	No	No
Phone A2 call Phone C1, Phone C1 conf Phone B2	Yes	No	No
Phone A2 call Phone C2, Phone C2 conf Phone B1	Yes	No	No



Table 20. VoiceMail Access, leave a message and SIP MWI Activate

Calls Made	Able to leave VM ?	MWI Message Sent?	MWI lamp turned ON?
Phone A1 call Phone C1 – CFU to Unity Phone A1 leaves a message for C1 Unity send MWI activate message to user C1	Yes	Yes	Yes
Phone A1 call Phone C2 – CFU to Unity Phone A1 leaves a message for C2 Unity send MWI activate message to user C2	Yes	Yes	Yes
Phone C1 call Phone A1 – CFU to Unity Phone C1 leaves a message for A1 Unity send MWI activate message to user A1	No ^{9, 10}	No ^{9, 10}	No ^{9, 10}
Phone C1 call Phone A2 – CFU to Unity Phone C1 leaves a message for A2 Unity send MWI activate message to user A2	No ^{9, 10}	No ^{9, 10}	No ^{9, 10}
Phone A1 call Phone D1 – CFU to 3 rd party VM system Phone A leaves a message for D1 3 rd party VM send MWI activate message to user D1	No ^{9, 10}	No ^{9, 10}	No ^{9, 10}
Phone A1 call Phone D2 – CFU to 3 rd party VM system Phone A leaves a message for D2 3 rd party VM send MWI activate message to user D2	No ^{9, 10}	No ^{9, 10}	No ^{9, 10}
Phone C2 call Phone B1 – CFU to 3 rd party VM system Phone C2 leaves a message for B1 3 rd party VM send MWI activate message to user B1	Yes	Yes	Yes
Phone C2 call Phone B2 – CFU to 3 rd party VM system Phone C2 leaves a message for B2 3 rd party VM send MWI activate message to user B2	Yes	Yes	Yes

⁹ There is a difference in the method used by both systems in passing the REDIRECTING information across the SIP trunk. CUCM uses the SIP Diversion header whereas Nortel uses the History-Info field instead. Since the redirect information is not recognized, the caller will not get the called party personal greeting to leave a voice message. Due to this difference, Cisco Unity or Nortel Call Pilot will not work as a centralized voice mail system for both Cisco Unified CallManager and Nortel CS1000M subscribers. That is, each system’s phone users can not be subscribed to the other system’s voice mail system using the SIP Trunk. For example, Cisco Unity can be a message center for the Cisco Unified CallManager users but it can not be a message center for Nortel CS1000M users.

¹⁰ There is a difference in the method used by both systems to relay DTMF. CUCM uses RFC2833 or KPML to transmit digits while the Nortel uses SIP INFO message for DTMF relay.



Table 21. VoiceMail Access, retrieve the message and SIP MWI Deactivate

Calls Made	Able to retrieve VM ?	MWI Message Sent?	MWI lamp turned OFF?
Phone C1 retrieve voice message from Unity Unity send MWI de-activate message to user C1	Yes	Yes	Yes
Phone C2 retrieve voice message from Unity Unity send MWI de-activate message to user C2	Yes	Yes	Yes
Phone A1 retrieve voice message from Unity Unity send MWI de-activate message to user A1	No ^{9,10}	No ^{9,10}	No ^{9,10}
Phone A2 retrieve voice message from Unity Unity send MWI de-activate message to user A2	No ^{9,10}	No ^{9,10}	No ^{9,10}
Phone D1 retrieve the message from 3 rd party VM system 3 rd party VM send MWI de-activate message to user D1	No ^{9,10}	No ^{9,10}	No ^{9,10}
Phone D2 retrieve the message from 3 rd party VM system 3 rd party VM send MWI de-activate message to user D2	No ^{9,10}	No ^{9,10}	No ^{9,10}
Phone B1 retrieve the message from 3 rd party VM system 3 rd party VM send MWI de-activate message to user B1	Yes	Yes	Yes
Phone B2 retrieve the message from 3 rd party VM system 3 rd party VM send MWI de-activate message to user B2	Yes	Yes	Yes