



Cisco Unified CallManager Release 5.0 - PBX Interoperability: Nortel CS1000M Release 4.0 Using SIP Trunk.

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Introduction

The purpose of this document is to detail the steps and configurations necessary for Cisco Unified CallManager 5.0 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 CallManager
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)

Integration highlights:

1. For Nortel CS1000M and Cisco Unified CallManager to interoperate with each other (Basic Call and/or Supplementary features), it required MTP resources. Therefore, the "Media Termination Point Required" box must be checked under the SIP Trunk configuration.
2. The method used by each system to pass the phone name and number information across the SIP trunk is different. Cisco Unified CallManager 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
3. 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 CallManager use "Remote-Party-Id" and "Privacy" whereas Nortel use "P-Asserted-Id" and "Privacy".
4. 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.
5. Both Nortel phones and Cisco Unified CallManager TNP phones (7961, 7970, 7971 and 7911) phones do not support SIP Blind Call Transfer.
6. Both systems support Call Forwarding (CFU, CFB, and CFNA) features. However, they are not able to update the phone displays properly after the call is forwarded. This is due to the differences between the two systems method of passing the name and number information.
7. Both systems support call conferencing using their local media resources. However, there is an intermittent issue with one way audio when the Nortel phone is the conferencing phone of two other CUCM phones.
8. Call Completion (Callback) Feature is not supported on either systems using standard SIP protocol.
9. Voice Messaging doesn't work across SIP Trunk between Cisco Unified CallManager and Nortel CS1000M PBX. Cisco Unified CallManager uses the SIP Diversion header to pass the redirect information across the SIP Trunk to the system which hosted the VM system. Nortel uses the SIP History-Info field instead. Due to the lack of support for one another method and differences, the redirect information is not passed across the SIP Trunk and therefore the voice mail system will treat the call as a direct access call and not a forwarded 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) can not have subscribers belonging to the one another. For example, Cisco Unity can be a voice mail system for Cisco Unified CallManager subscribers but not Nortel CS1000M subscribers using SIP Trunk.



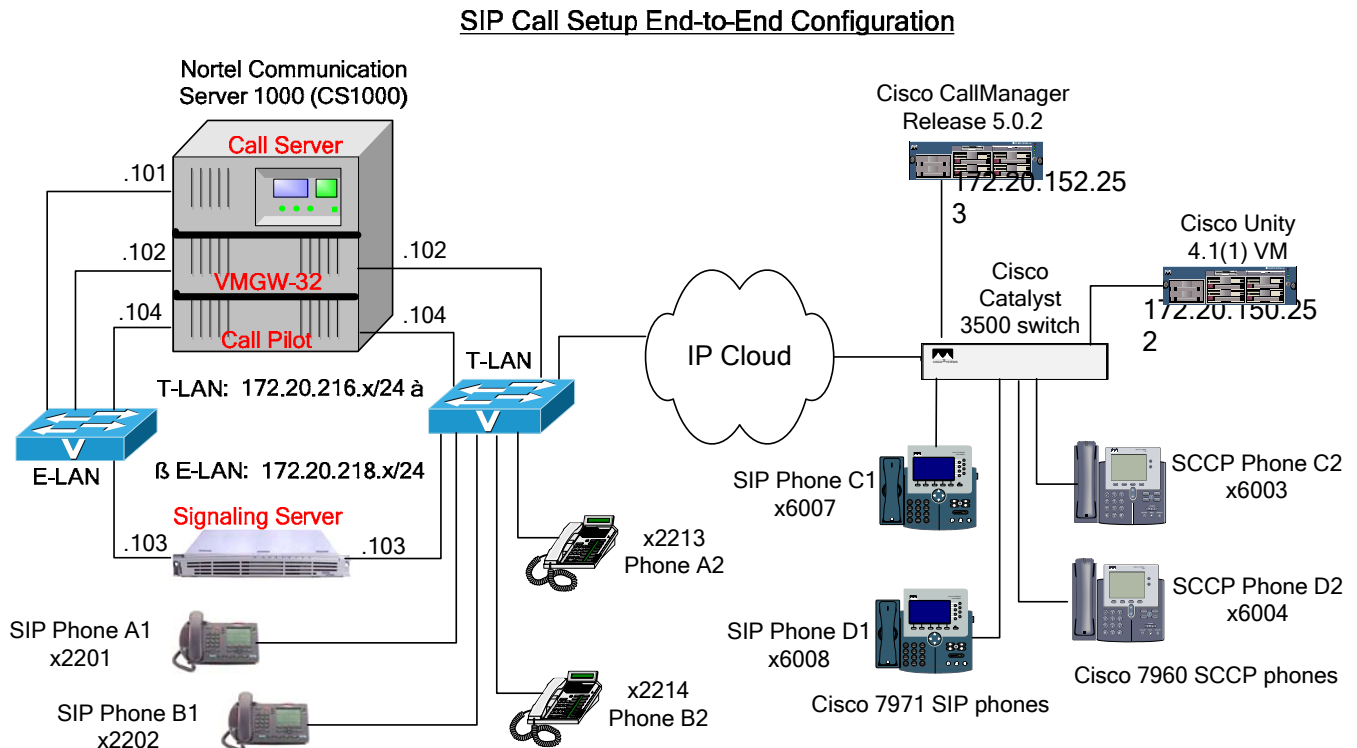
10. MWI Activation and De-activation message do not work across SIP Trunk. Cisco Unified CallManager use SIP Notify message for MWI notification. However, Nortel does not support this method.
11. End-to-end DTMF relay signaling doesn't work between the two systems and are incompatible with one another. Cisco Unified CallManager support both RFC2833 and KPML methods of DTMF-relay. However, on the other hand, use SIP INFO message instead.
12. For the Cisco Unified CallManager SIP Trunk configuration, "Media Termination Point Required", box must be checked in order for the two systems to communicate

Key Results:

1. Cisco Unified CallManager and Nortel CS1000M used different method of passing the name and number information. Cisco Unified CallManager used the "Remote-Party-Id" field while Nortel CS1000M used the "P-Asserted-Id" field
2. "Media Termination Point Required" check box must be enabled on the Cisco Unified CallManager SIP Trunk in order for the two systems to communicate with each other.
3. Basic call, Call Transfer, Call Forwarding, Conference Call, Hold and Resume all work fine with exception of the phone name and number display not being updated correctly.
4. MWI and DTMF-relay do not interoperate between the two systems.

Network Topology

Figure 1. Network Topology or Test Setup





Limitations

No support for MWI- Message Waiting Indication (lamp ON, lamp OFF) across the SIP Trunk

No end-to-end DTMF relay support across the SIP Trunk

No support for Callback across SIP trunk

CLIP/CLIR/CNIP/CNIR features – Please read the Integration Highlight section

COLP/CONP/COLR/CONR features - Please read the Integration Highlight section

System Components

Hardware Requirements

Cisco Unified CallManager MCS -7835H server,

Unity server MCS-7835H

Catalyst switch 3560 PoE-48

Cisco 7971 and 7960 IP phones

Nortel CS1000M

Nortel digital (2616) and IP (i2004) phones

Software Requirements

Cisco Unified CallManager Release 5.0.2

Cisco Unity Release 4.1(1)

CS1000M Release 4.0

Catalyst 3560 with IOS release: c3560-i5-mz.122-20.EX.bin

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

Configuration Sequence and Tasks

Nortel Communication Server 1000 PBX Configuration Sequence and Tasks

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

Cisco Unified CallManager:

1. Cisco Unified CallManager Software Version
2. Cisco Unified CallManager Group Configuration
3. Cisco Unified CallManager Default Device Pool Configuration
4. Cisco Unified CallManager Enterprise Parameters (Organization Top Level Domain) Configuration



5. Cisco Unified CallManager Media Resource Group Configuration
6. Cisco Unified CallManager Media Resource Group List Configuration
7. Cisco Unified CallManager SIP Phone Security Profile Configuration
8. Cisco Unified CallManager SIP Trunk Security Profile Configuration
9. Cisco Unified CallManager SCCP Security Profile Configuration
10. Cisco Unified CallManager SIP Profile for Nortel CS1000M PBX Configuration
11. Cisco Unified CallManager SIP Trunk Configuration to Nortel CS1000M PBX Configuration
12. Cisco Unified CallManager Phone Configuration List Configuration
13. Cisco Unified CallManager SCCP Phone Level Configuration
14. Cisco Unified CallManager SCCP Phone Directory Number (Ext 6003) Level Configuration
15. Cisco Unified CallManager SCCP Phone Directory Number (Ext 6004) Level Configuration
16. Cisco Unified CallManager SIP Phone Level Configuration
17. Cisco Unified CallManager SIP Phone Directory Number (Ext 6007) Level Configuration
18. Cisco Unified CallManager SIP Phone Directory Number (Ext 6008) Level Configuration
19. Cisco Unified CallManager Line Group for Unity Voice Mail Configuration
20. Cisco Unified CallManager Hunt List for Unity Voice Mail Configuration
21. Cisco Unified CallManager Hunt Pilot for Unity Voice Mail Configuration
22. Cisco Unified CallManager Voice Mail Profile Configuration
23. Cisco Unified CallManager Voice Mail Pilot Configuration
24. Cisco Unified CallManager Voice Mail Ports List Configuration (Lakers-VI)
25. Cisco Unified CallManager Voice Mail Ports Configuration (4 VM Ports)
26. Cisco Unified CallManager Voice Mail MWI ON and OFF Configuration
27. Cisco Unified CallManager Route Pattern (22XX) to Nortel PBX extensions Configuration
28. Cisco Unified CallManager Route Pattern (2500) to Nortel Call Pilot VM Pilot Number Configuration

Cisco Unity:

1. Cisco Unity Software Version
2. Cisco Unity Integration (CM-LAKERS)
3. Cisco Unity Subscriber (SCCP 6003) Configuration
4. Cisco Unity Subscriber (SIP 6007) Configurations
5. Cisco Unity Subscriber (Nortel ext. 2201) Configurations
6. Cisco Unity Subscriber (Nortel ext. 2213) Configurations

Configuration Menus and Commands

Nortel Communication Server 1000 (CS1000) Configuration

Call Server Setup:

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

```
>ld 22
PT2000

REQ prt
TYPE adan dch 3

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



MBGA NO
H323
OVLN NO
OVLS NO

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

```
>ld 97
SCSYS000
MEM AVAIL: (U/P): 2854769  USED U P: 182454 59352  TOT: 3096575
DISK RECS AVAIL: 1152
REQ prt
TYPE supl
SUPL
```

SUPL SUPT SLOT XPEC0 XPEC1

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

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
DATE
PAGE
DES
```

➔ SIP Virtual trunk to Signaling Server

```
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 2500
TGAR 1
LDN NO
NCOS 0
SGRP 0
RNPG 0
SCI 0
SSU
LNRS 16
XLST

CLS CTD **FBA** WTA LPR MTD **FNA HTA** TDD HFD CRPD
MWA LMPN RMMD SMWD AAD IMD XHD IRA NID OLD VCE DRG1
POD DSX VMD CMSD SLKD CCSD SWD LNA **CNDA**
CFTA SFD MRD DDV **CNIA** CDCA MSID DAPA BFED 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 2500

HUNT 2500

EHT 2500

LHK 0

LPK 1

PLEV 02

CSDN

AST

IAPG 0

AACS NO

ITNA NO

DGRP

MLWU_LANG 0

DNDR 0

KEY 00 SCR 2201 0 MARP

CPND

NAME ZEUS_2201

XPLN 8

DISPLAY_FMT FIRST, LAST

01



02
03 MIK
04 MCK
05
06
07
08
09
10
11
12
13
14
15
16 MWK 2500
17 TRN
18 AO6
19 CFW 16 2500
20 RGA
21 PRK
22 RNP
23
24 PRS
25 CHG
26 CPN
27
28
29
30
31
DATE 24 MAY 2006
NACT

Phone A2 (2616):

REQ: prt
TYPE: 2616

TN 10
DATE
PAGE
DES

DES CS101A
TN 001 0 00 00
TYPE 2616
CDEN 8D
CUST 0
AOM 0
FDN 6100
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 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 CDMR
CPND_LANG ENG
RCO 0
EFD 6100
HUNT 6100
EHT 6100
LHK 0
PLEV 02
CSDN
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
DNDR 0
KEY 00 SCR 2213 0 MARP
CPND
NAME **ZEUS_2213**
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 61001
DATE
PAGE
DES

DES I2002



TN 06100001 VIRTUAL
TYPE I2002
CDEN 8D
CUST 0
ZONE 000
FDN 6100
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 6100
HUNT 6100
EHT 6100
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 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 01
TYPE 2616
CDEN 8D
CUST 0
AOM 0
FDN 2500
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 2500
LHK 0



PLEV 02
CSDN
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
DNDR 0
KEY 00 SCR 2214 0 MARP
CPND
NAME ZEUS_2214
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: 172.20.218.101 > Configuration > Call Server Configuration > Zone List > Zone 0 >

Zone Basic Property and Bandwidth Management

Input Description	Input Value
Zone Number (ZONE):	<input type="text" value="0"/>
Intrazone Bandwith (INTRA_BW):	<input type="text" value="10000"/>
Intrazone Strategy (INTRA_STGY):	Best Quality (BQ) <input type="button" value="v"/>
Interzone Bandwith (INTER_BW):	<input type="text" value="10000"/>
Interzone Strategy (INTER_STGY):	Best Quality (BQ) <input type="button" value="v"/>
Resource Type (RES_TYPE):	Shared (SHARED) <input type="button" value="v"/>
Branch Office Support (ZBRN):	<input type="checkbox"/>
Description (ZDES):	<input type="text"/>

11. Configure a new IP Telephony Node summary



Site: 172.20.216.101 > Configuration > IP Telephony Configuration >

Node Summary

New Node

Node: 101	Node IP: 172.20.216.100	<input type="button" value="Edit"/>	<input type="button" value="Transfer / Status"/>	<input type="button" value="Delete"/>
Voice LAN (TLAN) IP address	TN			
Signaling Server	172.20.216.103			
Pentium Card				
Succession Media Card	172.20.216.102	3 0	<input type="button" value="VGW Channels"/>	



12. Configure the Node section

The screenshot shows the Cisco configuration interface for a Node. The breadcrumb path is: Site: 172.20.218.101 > Configuration > IP Telephony Configuration > Node Summary > IP Telephony: Node ID 101 >. The main area is titled "Edit" and contains a "Save and Transfer" button and a "Cancel" button. Below these are several configuration fields:

Field	Value
Node ID	101
Voice LAN (TLAN) Node IP address	172.20.216.100
Management LAN (ELAN) gateway IP address	172.20.218.1
Management LAN (ELAN) subnet mask	255.255.255.0
Voice LAN (TLAN) subnet mask	255.255.255.0

Below the fields are several expandable sections:

- SNMP (with an "Add" button)
- VGW and IP phone codec profile
- QoS
- LAN configuration
- SNTP
- H323 GW Settings

The left sidebar contains a navigation menu with the following items:

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

13. Configure the VGW and IP phone codec profile section



VGW and IP phone codec profile

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

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"/>

QoS
LAN configuration
SNTP

Codec G711 Select

Codec Name **G711**

Voice payload size (ms/frame)

Voice payout (jitter buffer) nominal delay

Modifications may cause changes to dependent settings

Voice payout (jitter buffer) maximum delay

Modifications may cause changes to dependent settings

VAD

Codec G729A Select

Codec Name **G729A**

Voice payload size (ms/frame)

Voice payout (jitter buffer) nominal delay

Modifications may cause changes to dependent settings

Voice payout (jitter buffer) maximum delay

Modifications may cause changes to dependent settings

VAD

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

Codec Name **T38 FAX**

14. Configure the QoS section



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	G729A	Select	<input checked="" type="checkbox"/>
Codec	G723.1	Select	<input type="checkbox"/>
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			
SNTP			
H323 GW Settings			
Firmware			
SIP GW Settings			
SIP URI Map			
SIP CD Services			
Cards			Add
Signaling Servers			Add

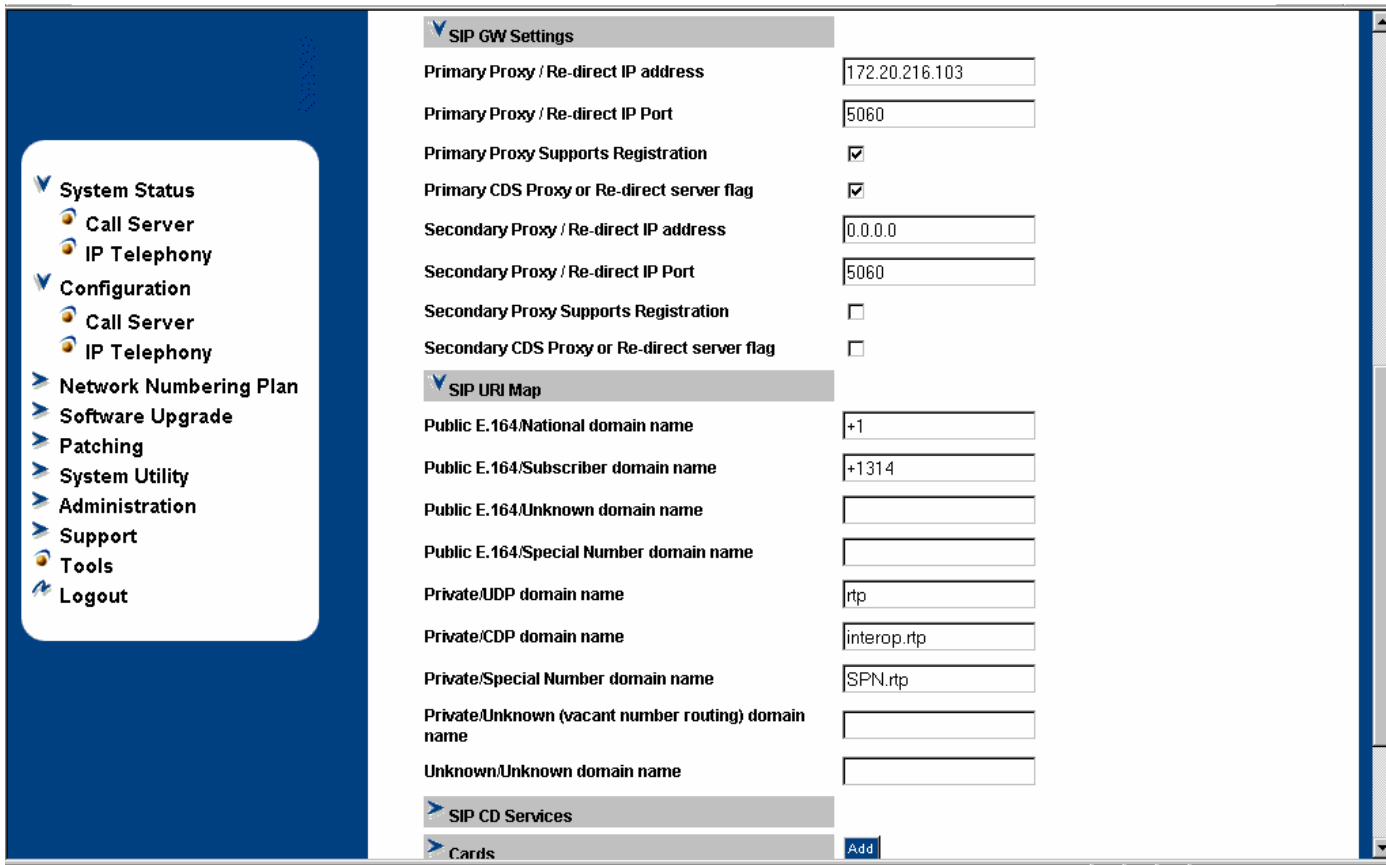


15. Configure LAN Configuration section

The screenshot displays the Cisco configuration interface for LAN settings. 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 area is titled 'LAN configuration' and is divided into 'Management LAN (ELAN) configuration' and 'Voice LAN (TLAN) configuration'. Under Management LAN, fields include Call server IP address (172.20.218.101), Survivable Succession Media Gateway IP address (0.0.0.0), Signaling port (15000), and Broadcast port (15001). Under Voice LAN, fields include Signaling port (5000) and Voice port (5200). A 'Routes' section at the bottom shows a table with IP address and Subnet mask columns, containing the entry 172.20.216.1 with subnet mask 255.255.255.0. A 'Remove' button is next to this entry.

IP address	Subnet mask	
172.20.216.1	255.255.255.0	Remove

16. Configure the SIP GW Setting section



The screenshot shows a configuration page with a left-hand navigation menu and a main configuration area. The navigation menu includes sections for System Status, Configuration, Network Numbering Plan, Software Upgrade, Patching, System Utility, Administration, Support, Tools, and Logout. The main configuration area is divided into several sections:

- SIP GW Settings:**
 - Primary Proxy / Re-direct IP address: 172.20.216.103
 - Primary Proxy / Re-direct IP Port: 5060
 - Primary Proxy Supports Registration:
 - Primary CDS Proxy or Re-direct server flag:
 - Secondary Proxy / Re-direct IP address: 0.0.0.0
 - Secondary Proxy / Re-direct IP Port: 5060
 - Secondary Proxy Supports Registration:
 - Secondary CDS Proxy or Re-direct server flag:
- SIP URI Map:**
 - Public E.164/National domain name: +1
 - Public E.164/Subscriber domain name: +1314
 - Public E.164/Unknown domain name: (empty)
 - Public E.164/Special Number domain name: (empty)
 - 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: (empty)
 - Unknown/Unknown domain name: (empty)
- SIP CD Services:** (Section header)
- Cards:** (Section header with an 'Add' button)

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



The screenshot displays the Cisco SIP CD Services configuration interface. On the left is a navigation menu with the following items: System Status, Call Server, IP Telephony, Configuration, Call Server, IP Telephony, Network Numbering Plan, Software Upgrade, Patching, System Utility, Administration, Support, Tools, and Logout. The main configuration area is titled 'SIP CD Services' and contains two sections: 'Cards' and 'Signaling Servers'. The 'Cards' section is expanded to show 'Card 172.20.218.102 Properties' with a 'Remove' button. The 'Signaling Servers' section is expanded to show 'Signaling Server 172.20.218.103 Properties' with a 'Remove' button. Below these sections are 'Save and Transfer' and 'Cancel' buttons. A note at the bottom states '* Mandatory fields of current configuration'. The configuration details for the selected card are as follows:

Field	Value	Requirement
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	<input checked="" type="checkbox"/>	
System name	SS_Node101	
System location		
System contact		

18. Configure the Signaling Server section



Signaling Server 172.20.218.103 Properties Remove

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

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

SIP Server Settings

NRS Overview
System Wide Settings
=> NRS Server Settings

Mode: Redirect

UDP transport enabled:

UDP port: 5060

UDP maximum transmission unit (MTU): 1500

TCP transport enabled:

TCP port: 5060

TCP maximum transmission unit (MTU): 1500

Network Connection Server (NCS) Settings

Primary NCS port: 16500

Alternate NCS port: 16500

Primary NCS timeout [Seconds]: 10

Save

*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 >

=> Service Domains

L1 Domains (UDP)

L0 Domains (CDP)

Gateway Endpoints

User Endpoints

Routing Entries

Default Routes

Collaborative Servers

View Service Domain Property

Domain name *

Domain description

* Mandatory field indicator



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: interop *

Domain description: CDP

Special number label:

Unqualified number label:

Endpoint authentication enabled: Not configured

Authentication password:

E.164 country code: 1

E.164 area code: 919

International dialing access code: 011

L1 domain dialing access code: 9

National dialing access code: 9

Local dialing access code: 9

24. Configure a SIP gateway



Network Routing Service

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

Location: [Configuration](#) > [Gateway Endpoints](#) > [View Gateway Endpoint Property](#) >

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

Service Domains

[L1 Domains \(UDP\)](#)[L0 Domains \(CDP\)](#)[=> Gateway Endpoints](#)[User Endpoints](#)[Routing Entries](#)[Default Routes](#)[Collaborative Servers](#)Endpoint name *Endpoint description
Tandem endpoint name [Look up](#)Endpoint authentication enabled Authentication password E.164 country code E.164 area code International dialing access code L1 domain dialing access code National dialing access code Local dialing access code Special number 1



Network Routing Service

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

Service Domains

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

International dialing access code

L1 domain dialing access code

National dialing access code

Local dialing access code

Special number 1

Special number 2

Static endpoint address type

Static endpoint address

H.323 Support

SIP support

SIP transport

SIP port

Network Connection Server enabled

** Mandatory field indicator*



25. Configure the Routing Entries

Network Routing Service

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

Location: Configuration > Routing Entries >

Routing Entries

Show Routing Entries for (Service Domain / L1 Domain / L0 Domain / Endpoint):

pbxlab.org / rtp / interop / CM_LAKERS [Look up](#)

Showing 1 - 4 of 4 < Previous Next >

#	DN Prefix	DN Type	Route Cost	SIP URI Phone Context
1	600	Level0 regional	1	interop.rtp
2	610	Level0 regional	1	interop.rtp
3	661	Level1 regional	1	rtp
4	662	Level1 regional	1	rtp



Cisco Unified CallManager Configuration

Cisco Unified CallManager Software Version

Cisco Unified CallManager Administration For Cisco IP Telecommunication Solutions Logged in as: CCM

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



Cisco Unified CallManager Administration

System version: 5.0.2.1000-3
Administration version: 1.1.0.0-1

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



Cisco Unified CallManager Group Configuration

Navigation Cisco Unified CallManager Admini

Cisco Unified CallManager Administration

For Cisco IP Telecommunication Solutions Logged in as: CCM

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

Cisco Unified CallManager Group Configuration

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

Status
Status: Ready

Cisco Unified CallManager Group Information
Cisco Unified CallManager Group: **Default (used by 31 devices)**

Cisco Unified CallManager Group Settings
Name*
 Auto-registration Cisco Unified CallManager Group

Cisco Unified CallManager Group Members
Available Cisco Unified CallManagers

▼ ▲

Selected Cisco Unified CallManagers * ▼
(Ordered by highest priority)



Cisco Unified CallManager Default Device Pool Configuration

Status	
Status: Ready	
Device Pool: Default (27 members**)	
Device Pool Settings	
Device Pool Name*	Default
Cisco Unified CallManager Group*	Default
Date/Time Group*	CMLocal
Region*	Default
Softkey Template*	Standard User
SRST Reference*	Disable
Calling Search Space for Auto-registration	< None >
Media Resource Group List	< None >
Network Hold MOH Audio Source	< None >
User Hold MOH Audio Source	< None >
Network Locale	< None >
User Locale	< None >
Connection Monitor Duration	
Multilevel Precedence and Preemption (MLPP) Information	
MLPP Indication*	Default
MLPP Preemption*	Default
MLPP Domain	< None >
Save Delete Copy Reset Add New	



Cisco Unified CallManager Enterprise Parameters (Organization Top Level Domain) Configuration

Allowed CDRonDemand get_file_list Queries Per Minute *	<input type="text" value="20"/>	20
Trace Parameters		
File Close Thread Flag *	<input type="text" value="False"/>	True
FileCloseThreadQueueWatermark *	<input type="text" value="100"/>	100
User Management Parameters		
Effective Access Privileges For Overlapping User Groups and roles *	<input type="text" value="Maximum"/>	Maximum
Service Manager TCP ports parameters		
Service Manager TCP Server communication port number *	<input type="text" value="8888"/>	8888
Service Manager TCP Client communication port number *	<input type="text" value="8889"/>	8889
CRS Application Parameters		
Auto Attendant Installed *	<input type="text" value="false"/>	false
IPCC Express Installed *	<input type="text" value="false"/>	false
Clusterwide Domain Configuration		
Organization Top Level Domain	<input type="text" value="pbxlab.org"/>	
Cluster Fully Qualified Domain Name	<input type="text"/>	
Cisco Support Use		
Cisco Support Use 1	<input type="text"/>	
<input type="button" value="Save"/> <input type="button" value="Set to Default"/> <input type="button" value="Reset"/>		
- indicates required item.		



Cisco Unified CallManager Media Resource Group Configuration

Status Status: Ready	
Media Resource Group Status Media Resource Group: SW_MRG (used by 4 devices)	
Media Resource Group Information	
Name*	SW_MRG
Description	SW_MRG
Devices for this Group	
Available Media Resources**	
▼ ▲	
Selected Media Resources *	ANN_2 (ANN) CFB_2 (CFB) MOH_2 (MOH) MTP_2 (MTP)
<input type="checkbox"/> Use Multicast for MOH Audio (If at least one multicast MOH resource is available)	
Save Delete Copy Reset Add New	



Cisco Unified CallManager Media Resource Group List Configuration

Media Resource Group List Status SW_MRGL (used by 12 devices)	
Media Resource Group List Information Name* SW_MRGL	
Media Resource Groups for this List	
Available Media Resource Groups	
	▼ ▲
Selected Media Resource Groups	SW_MRG ▼ ▲
<input type="button" value="Save"/> <input type="button" value="Delete"/> <input type="button" value="Copy"/> <input type="button" value="Reset"/> <input type="button" value="Add New"/>	
*- indicates required item.	



Cisco Unified CallManager SIP Phone Security Profile Configuration

Status	
Status: Ready	
SIP Phone Security Profile Information	
Name*	Standard SIP Profile for Auto Registration
Description	Standard SIP Profile for Auto Registration
Nonce Validity Time*	600
Device Security Mode	Non Secure
Transport Type*	TCP+UDP
<input type="checkbox"/> Enable Digest Authentication	
SIP Phone Security Profile CAPF Information	
Authentication Mode*	By Null String
Key Size (Bits)*	1024
Parameters used in Phone	
SIP Phone Port*	5060
<input type="button" value="Copy"/> <input type="button" value="Reset"/> <input type="button" value="Add New"/>	
*- indicates required item.	



Cisco Unified CallManager SIP Trunk Security Profile Configuration

Status
Status: Ready

SIP Trunk Security Profile Information

Name* Nortel101_5060

Description CS1K Node 101

Device Security Mode Non Secure

Incoming Transport Type* TCP+UDP

Outgoing Transport Type TCP

Enable Digest Authentication

Nonce Validity Time (mins)* 600

X.509 Subject Name

Incoming Port* 5060

Enable Application Level Authorization

Accept Presence Subscription

Accept Out-of-Dialog REFER

Accept Unsolicited Notification

Accept Replaces Header

Save Delete Copy Reset Add New



Cisco Unified CallManager SCCP Security Profile Configuration

Status	
Status: Ready	
SCCP Security Profile Information	
Name*	Standard SCCP Profile for Auto Registration
Description	Standard SCCP Profile for Auto Registration
Device Security Mode	Non Secure
SCCP Security Profile CAPF Information	
Authentication Mode*	By Null String
Key Size (Bits)*	1024
Copy Reset Add New	
* - indicates required item.	



Cisco Unified CallManager SIP Profile for Nortel CS1000M PBX Configuration

Status	
Status: Ready	
SIP Profile Information	
Name*	CS1K_Profile
Description	Nortel CS1K SIP Profile
Default MTP Telephony Event Payload Type*	101
<input type="checkbox"/> Redirect by Application	
<input type="checkbox"/> Disable Early Media on 180	
Parameters used in Phone	
Timer Invite Expires (seconds)*	180
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10
Start Media Port*	16384
Stop Media Port*	32766



Setup Media Profile	327bb
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 (microseconds)*	15000
Call Forward URI*	x-cisco-serviceuri-cfwdall
Abbreviated Dial URI*	x-cisco-serviceuri-abbrdial
<input checked="" type="checkbox"/> Conference Join Enabled	
<input type="checkbox"/> RFC 2543 Hold	
<input checked="" type="checkbox"/> Semi Attended Transfer	
<input type="checkbox"/> Enable VAD	
<input type="checkbox"/> Stutter Message Waiting	
<input type="checkbox"/> Call Stats	

Cisco Unified CallManager SIP Trunk Configuration to Nortel CS1000M PBX Configuration



Status
Status: Ready

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Device Name*	CS1K_Node101
Description	SIP Trunk To Nortel CS1K Node 101
Device Pool*	Default
Call Classification*	Use System Default
Media Resource Group List	SW_MRGL
Location*	Hub_None
AAR Group	< None >
Packet Capture Mode*	None
Packet Capture Duration	0

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

Multilevel Precedence and Preemption (MLPP) Information



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

Caller Name

Redirecting Diversion Header Delivery - Outbound

SIP Information

Destination Address*

Destination Address is an SRV

Destination Port* Note: 0 indicates destination is SRV

MTP Preferred Originating Codec*

Presence Group*



Outbound Calls	
Calling Party Selection*	Originator
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Caller ID DN	
Caller Name	
<input checked="" type="checkbox"/> Redirecting Diversion Header Delivery - Outbound	

SIP Information	
Destination Address*	172.20.216.100
<input type="checkbox"/> Destination Address is an SRV	
Destination Port *	5060 <small>Note: 0 indicates destination is SRV</small>
MTP Preferred Originating Codec*	711ulaw
Presence Group*	Standard Presence group
SIP Trunk Security Profile*	Nortel101_5060
Rerouting Calling Search Space	< None >
Out-Of-Dialog Refer Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	CS1K_Profile
DTMF Signaling Method*	No Preference

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



Cisco Unified CallManager Phone Configuration List Configuration

Navigation Cisco Unified CallManager Administration

Cisco Unified CallManager Administration For Cisco IP Telecommunication Solutions Logged in as: CCM




























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

Find and List Phones Related Links: CAPF Report i

Status
9 records found

Search Options
Find Phone where Device Name begins with Find Search Within Results
Select item or enter search text
(device.name begins with any)

Search Results

	Device Name(Line)	Description	Device Pool	Device Protocol	Status	IP Address	Copy	Cop
<input type="checkbox"/>	 SEP000A8A48FA59	Auto 6006	Default	SCCP	Unknown	Unknown		
<input type="checkbox"/>	 SEP00115C0E55D4	7970 SIP_6008	Default	SIP	Registered with CM-LAKERS	172.20.152.12		
<input type="checkbox"/>	 SEP0013C3E3A62F	7971 SIP_6007	Default	SIP	Registered with CM-LAKERS	172.20.152.11		
<input type="checkbox"/>	 SEP00146A4D3C4D	7960 SIP_6001	Default	SIP	Registered with CM-LAKERS	172.20.152.14		
<input type="checkbox"/>	 SEP00146A4D3D96	SCCP_6004	Default	SCCP	Registered with CM-LAKERS	172.20.152.13		
<input type="checkbox"/>	 SEP0015632CE07B	7960 SIP_6002	Default	SIP	Registered with CM-LAKERS	172.20.152.19		
<input type="checkbox"/>	 SEP0015C63E817E	SCCP_6003	Default	SCCP	Registered with CM-LAKERS	172.20.152.17		
<input type="checkbox"/>	 SEP00170EE67E8E	Auto 6009	Default	SCCP	Registered with CM-LAKERS	172.20.152.16		
<input type="checkbox"/>	 SEP00170EEE2FBD	Auto 6010	Default	SCCP	Registered with CM-LAKERS	172.20.152.15		

Add New Select All Clear All Delete Selected Reset Selected Rows per Page 50

Cisco Unified CallManager SCCP Phone Level Configuration



Navigation Cisco Unified CallManager Admin

Cisco Unified CallManager Administration

For Cisco IP Telecommunication Solutions

Logged in as: CCM

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

Phone Configuration

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

Status
Status: Ready

Association Information

[Modify Button Items](#)

- 1 [Line \[1\] - 6003 \(no partition\)](#)
- 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 SURL](#)
- 9 [Add a new BLF SD](#)
- 10 Privacy
- 11 None

Phone Type

Product Type: Cisco 7960
Device Protocol: SCCP

Device Information

Registration	Registered with Cisco Unified CallManager CM-LAKERS
IP Address	172.20.152.17
MAC Address*	<input type="text" value="0015C63E817E"/>
Description	<input type="text" value="SCCP_6003"/>
Device Pool*	<input type="text" value="Default"/>
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="SW_MRGL"/>
User Hold Audio Source	<input type="text" value="1-SampleAudioSource"/>
Network Hold Audio Source	<input type="text" value="1-SampleAudioSource"/>
Location*	<input type="text" value=""/>



Location*	Hub_None
User Locale	< None >
Network Locale	< None >
Built In Bridge*	Default
Privacy*	Default
Owner User ID	< None >
Phone Load Name	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Ignore Presentation Indicators (internal calls only)	
<input checked="" type="checkbox"/> Allow Control of Device from CTI	
Protocol Specific Information	
Packet Capture Mode*	None
Packet Capture Duration	0
Presence Group*	Standard Presence group
SCCP Phone Security Profile*	Standard SCCP Profile for Auto Registration
SUBSCRIBE Calling Search Space	< None >
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> Require DTMF Reception	
<input type="checkbox"/> RFC2833 Disabled	
Expansion Module Information	
Module 1	< None >
Module 1 Load Name	
Module 2	< None >
Module 2 Load Name	



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	
<input type="checkbox"/>	Enable Extension Mobility
Log Out Profile	-- Not Selected --
Login in User ID	< None >
Log in Time	< None >
Log out Time	< None >
Certification Authority Proxy Function (CAPF) Information	
Certificate Operation*	No Pending Operation
Authentication String	<input type="text"/>
<input type="button" value="Generate String"/>	
Operation Completes By	2006 : 6 : 15 : 12 (YYYY:MM:DD:HH)
Certificate Operation Status: None	
MLPP Information	
MLPP Domain	< None >
MLPP Indication*	Default
MLPP Preemption*	Default



Certificate Operation Status: None

MLPP Information

MLPP Domain: < None >

MLPP Indication*: Default

MLPP Preemption*: Default

Secure Shell Information

Secure Shell User:

Secure Shell Password:

Product Specific Configuration Layout ?

Disable Speakerphone

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

i *- indicates required item.

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

Cisco Unified CallManager SCCP Phone Directory Number (Ext 6003) Level Configuration



Navigation Cisco Unified CallManager Admini

Cisco Unified CallManager Administration

For Cisco IP Telecommunication Solutions

Logged in as: CCM

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

Directory Number Configuration

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

Status
Status: Ready

Note: Changes to Line or Directory Number settings require restart.

Directory Number Information

Directory Number*	6003
Route Partition	< None >
Description	SCCP_6003
Alerting Name	SCCP_6003A
ASCII Alerting Name	SCCP_6003A

Allow Control of Device from CTI

Associated Devices

SEP0015C63E817E	Edit Device
-----------------	-----------------------------

[Edit Line Appearance](#)

▼ ▲

Dissociate Devices

--



Directory Number Settings		
Voice Mail Profile	Default	(Choose <None> to use system default)
Calling Search Space	< None >	
Presence Group*	Standard Presence group	
AAR Group	< None >	
User Hold Audio Source	1-SampleAudioSource	
Network Hold Audio Source	1-SampleAudioSource	
Auto Answer*	Auto Answer Off	

Call Forward and Call Pickup Settings		
	Voice Mail Destination	Calling Search Space
Forward All	<input type="checkbox"/> or	< None >
Secondary Calling Search Space for Forward All		< None > Find
Forward Busy Internal	<input type="checkbox"/> or 2214	< None >
Forward Busy External	<input type="checkbox"/> or 2214	< None >
Forward No Answer Internal	<input type="checkbox"/> or 2214	< None >
Forward No Answer External	<input type="checkbox"/> or 2214	< None >
Forward No Coverage Internal	<input type="checkbox"/> or 2202	< None >
Forward No Coverage External	<input type="checkbox"/> or 2202	< None >
Forward on CTI Failure	<input type="checkbox"/> or 2202	< None >
No Answer Ring Duration (seconds)	5	
Call Pickup Group	< None >	

MLPP Alternate Party Settings	
Target (Destination)	
MLPP Calling Search Space	< None >
MLPP No Answer Ring Duration (seconds)	



MLPP Calling Search Space

MLPP No Answer Ring Duration (seconds)

Line 1 on Device SEP0015C63E817E

Display (Internal Caller ID) 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.

ASCII Display (Internal Caller ID)

Line Text Label

ASCII Line Text Label

External Phone Number Mask

Message Waiting Lamp Policy*

Ring Setting (Phone Idle)*

Ring Setting (Phone Active) Applies to this line when any line on the phone has a call in progress.

Multiple Call/Call Waiting Settings on Device SEP0015C63E817E

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 SEP0015C63E817E

Caller Name

Caller Number

Redirected Number

Dialed Number



Cisco Unified CallManager SCCP Phone Directory Number (Ext 6004) Level Configuration

Navigation Cisco Unified CallManager Admini

Cisco Unified CallManager Administration For Cisco IP Telecommunication Solutions Logged in as: CCM

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

Directory Number Configuration Related Links: Back To Find/List

Status
Status: Ready

Note: Changes to Line or Directory Number settings require restart.

Directory Number Information

Directory Number*	6004
Route Partition	< None >
Description	SCCP_6004
Alerting Name	SCCP_6004A
ASCII Alerting Name	SCCP_6004A

Allow Control of Device from CTI

Associated Devices

SEP00146A4D3D96	Edit Device
-----------------	--------------------

Edit Line Appearance

▼ ▲

Dissociate Devices

--



Directory Number Settings		
Voice Mail Profile	Default	(Choose <None> to use system default)
Calling Search Space	< None >	
Presence Group*	Standard Presence group	
AAR Group	< None >	
User Hold Audio Source	1-SampleAudioSource	
Network Hold Audio Source	1-SampleAudioSource	
Auto Answer*	Auto Answer Off	

Call Forward and Call Pickup Settings		
Forward All	<input type="checkbox"/> or	Calling Search Space
Secondary Calling Search Space for Forward All		< None >
Forward Busy Internal	<input checked="" type="checkbox"/> or	< None >
Forward Busy External	<input checked="" type="checkbox"/> or	< None >
Forward No Answer Internal	<input checked="" type="checkbox"/> or	< None >
Forward No Answer External	<input checked="" type="checkbox"/> or	< None >
Forward No Coverage Internal	<input checked="" type="checkbox"/> or	< None >
Forward No Coverage External	<input checked="" type="checkbox"/> or	< None >
Forward on CTI Failure	<input checked="" type="checkbox"/> or	< None >
No Answer Ring Duration (seconds)		
Call Pickup Group		< None >

MLPP Alternate Party Settings	
Target (Destination)	
MLPP Calling Search Space	< None >
MLPP No Answer Ring Duration (seconds)	



Target (Destination)	<input type="text"/>
MLPP Calling Search Space	< None >
MLPP No Answer Ring Duration (seconds)	<input type="text"/>
Line 1 on Device SEP00146A4D3D96	
Display (Internal Caller ID)	<input type="text" value="SCCP_6004"/> Display text for a line appearance is intended for displaying text such as a name inst directory number for internal calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
ASCII Display (Internal Caller ID)	<input type="text" value="SCCP_6004"/>
Line Text Label	<input type="text" value="SCCP_6004"/>
ASCII Line Text Label	<input type="text" value="SCCP_6004"/>
External Phone Number Mask	<input type="text"/>
Message Waiting Lamp Policy*	Use System Policy
Ring Setting (Phone Idle)*	Ring
Ring Setting (Phone Active)	Use System Default Applies to this line when any line on the phone has a call in progress.
Multiple Call/Call Waiting Settings on Device SEP00146A4D3D96	
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="1"/> (Less than or equal to Max. Calls)
Forwarded Call Information Display on Device SEP00146A4D3D96	
<input checked="" type="checkbox"/> Caller Name	
<input checked="" type="checkbox"/> Caller Number	
<input checked="" type="checkbox"/> Redirected Number	
<input checked="" type="checkbox"/> Dialed Number	
<input type="button" value="Save"/> <input type="button" value="Delete"/> <input type="button" value="Copy"/> <input type="button" value="Reset"/> <input type="button" value="Add New"/>	



Cisco Unified CallManager SIP Phone Level Configuration

Navigation Cisco Unified CallManager Admini

Cisco Unified CallManager Administration For Cisco IP Telecommunication Solutions Logged in as: CCM

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

Phone Configuration Related Links: Back To Find/List

Status
Status: Ready

Association Information
Modify Button Items

- 7975 Line [1] - 6007 (no partition)
- 7975 Line [2] - Add a new DN
- Add a new SD
- Add a new SD
- 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
- Privacy
- None

Phone Type
Product Type: Cisco 7971
Device Protocol: SIP

Device Information

Registration	Registered with Cisco Unified CallManager CM-LAKERS
IP Address	172.20.152.11
MAC Address*	0013C3E3A62F
Description	7971 SIP_6007
Device Pool*	Default
Phone Button Template*	Standard 7971 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	SW_MRGL
User Hold Audio Source	1-SampleAudioSource
Network Hold Audio Source	1-SampleAudioSource
Location*	



Location*	Hub_None
User Locale	< None >
Network Locale	< None >
Built In Bridge*	Default
Privacy*	Default
Owner User ID	< None >
Phone Load Name	
<input type="checkbox"/> Ignore Presentation Indicators (internal calls only)	
<input checked="" type="checkbox"/> Allow Control of Device from CTI	
Protocol Specific Information	
Packet Capture Mode*	None
Packet Capture Duration	0
Presence Group*	Standard Presence group
SIP Dial Rules	< None >
MTP Preferred Originating Codec*	711ulaw
SIP Phone Security Profile*	Standard SIP Profile for Auto Registration
Rerouting Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	Standard SIP Profile
Digest User	< None >
<input checked="" type="checkbox"/> Media Termination Point Required	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> Require DTMF Reception	
External Data Locations Information (Leave blank to use default)	
Information	



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	
<input type="checkbox"/>	Enable Extension Mobility
Log Out Profile	-- Not Selected --
Login in User ID	< None >
Log in Time	< None >
Log out Time	< None >
Certification Authority Proxy Function (CAPF) Information	
Certificate Operation*	No Pending Operation
Authentication String	<input type="text"/>
<input type="button" value="Generate String"/>	
Operation Completes By	2006 : 6 : 15 : 12 (YYYY:MM:DD:HH)
Certificate Operation Status: None	
MLPP Information	
MLPP Domain	000000
Secure Shell Information	
Secure Shell User	<input type="text"/>



Secure Shell Information	
Secure Shell User	<input type="text"/>
Secure Shell Password	<input type="password"/>

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
Days Display Not Active	Sunday Monday Tuesday
Display On Time	07:30
Display On Duration	10:30
Display Idle Timeout	01:00
Span to PC Port*	Disabled
Logging Display*	PC Controlled
Load Server	<input type="text"/>

Save Delete Copy Reset Add New



Cisco Unified CallManager SIP Phone Directory Number (Ext 6007) Level Configuration

Navigation Cisco Unified CallManager Admini

Cisco Unified CallManager Administration For Cisco IP Telecommunication Solutions Logged in as: CCM

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

Directory Number Configuration Related Links: Back To Find/List

Status
Status: Ready

Note: Changes to Line or Directory Number settings require restart.

Directory Number Information

Directory Number*	6007
Route Partition	< None >
Description	7971 SIP_6007
Alerting Name	SIP_6007A
ASCII Alerting Name	SIP_6007A

Allow Control of Device from CTI

Associated Devices

SEP0013C3E3A62F	Edit Device
-----------------	-----------------------------

[Edit Line Appearance](#)

Dissociate Devices



Directory Number Settings		
Voice Mail Profile	Default	(Choose <None> to use system default)
Calling Search Space	< None >	
Presence Group*	Standard Presence group	
AAR Group	< None >	
User Hold Audio Source	1-SampleAudioSource	
Network Hold Audio Source	1-SampleAudioSource	
Auto Answer*	Auto Answer Off	

Call Forward and Call Pickup Settings		
Forward All	<input type="checkbox"/> or	Calling Search Space
		< None >
Secondary Calling Search Space for Forward All		< None > Find
Forward Busy Internal	<input type="checkbox"/> or	2214 < None >
Forward Busy External	<input type="checkbox"/> or	2214 < None >
Forward No Answer Internal	<input type="checkbox"/> or	2214 < None >
Forward No Answer External	<input type="checkbox"/> or	2214 < None >
Forward No Coverage Internal	<input type="checkbox"/> or	2202 < None >
Forward No Coverage External	<input type="checkbox"/> or	2202 < None >
Forward on CTI Failure	<input type="checkbox"/> or	< None >
No Answer Ring Duration (seconds)	5	
Call Pickup Group		< None >

MLPP Alternate Party Settings	
Target (Destination)	
MLPP Calling Search Space	< None >
MLPP No Answer Ring Duration (seconds)	



Target (Destination)	<input type="text"/>
MLPP Calling Search Space	< None >
MLPP No Answer Ring Duration (seconds)	<input type="text"/>
Line 1 on Device SEP0013C3E3A62F	
Display (Internal Caller ID)	<input type="text" value="SIP_6007"/> 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.
ASCII Display (Internal Caller ID)	<input type="text" value="SIP_6007"/>
Line Text Label	<input type="text" value="SIP_6007"/>
ASCII Line Text Label	<input type="text" value="SIP_6007"/>
External Phone Number Mask	<input type="text"/>
Message Waiting Lamp Policy*	Use System Policy
Ring Setting (Phone Idle)*	Ring
Ring Setting (Phone Active)	Use System Default Applies to this line when any line on the phone has a call in progress.
Multiple Call/Call Waiting Settings on Device SEP0013C3E3A62F	
Note: The range to select the Max Number of calls is: 1-50	
Maximum Number of Calls*	<input type="text" value="4"/>
Busy Trigger*	<input type="text" value="1"/> (Less than or equal to Max. Calls)
Forwarded Call Information Display on Device SEP0013C3E3A62F	
<input checked="" type="checkbox"/> Caller Name	
<input checked="" type="checkbox"/> Caller Number	
<input checked="" type="checkbox"/> Redirected Number	
<input checked="" type="checkbox"/> Dialed Number	
<input type="button" value="Save"/> <input type="button" value="Delete"/> <input type="button" value="Copy"/> <input type="button" value="Reset"/> <input type="button" value="Add New"/>	



Cisco Unified CallManager SIP Phone Directory Number (Ext 6008) Level Configuration

Navigation Cisco Unified CallManager Admini

Cisco Unified CallManager Administration For Cisco IP Telecommunication Solutions Logged in as: CCM

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

Directory Number Configuration Related Links: Back To Find/List

Status
Status: Ready

Note: Changes to Line or Directory Number settings require restart.

Directory Number Information

Directory Number*	6008
Route Partition	< None >
Description	7970 SIP_6008
Alerting Name	SIP_6008A
ASCII Alerting Name	SIP_6008A

Allow Control of Device from CTI

Associated Devices

SEP00115C0E55D4

Edit Device

Edit Line Appearance

▼ ▲

Dissociate Devices



Directory Number Settings	
Voice Mail Profile	Default (Choose <None> to use system default)
Calling Search Space	< None >
Presence Group*	Standard Presence group
AAR Group	< None >
User Hold Audio Source	1-SampleAudioSource
Network Hold Audio Source	1-SampleAudioSource
Auto Answer*	Auto Answer Off

Call Forward and Call Pickup Settings	
Forward All	<input type="checkbox"/> or <input type="text"/> Calling Search Space: < None >
Secondary Calling Search Space for Forward All	< None > Find
Forward Busy Internal	<input type="checkbox"/> or <input type="text"/> < None >
Forward Busy External	<input type="checkbox"/> or <input type="text"/> < None >
Forward No Answer Internal	<input type="checkbox"/> or <input type="text"/> < None >
Forward No Answer External	<input type="checkbox"/> or <input type="text"/> < None >
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 >
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"/>



Target (Destination)	<input type="text"/>
MLPP Calling Search Space	< None >
MLPP No Answer Ring Duration (seconds)	<input type="text"/>
Line 1 on Device SEP00115C0E55D4	
Display (Internal Caller ID)	<input type="text" value="SIP_6008"/> 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.
ASCII Display (Internal Caller ID)	<input type="text" value="SIP_6008"/>
Line Text Label	<input type="text" value="SIP_6008"/>
ASCII Line Text Label	<input type="text" value="SIP_6008"/>
External Phone Number Mask	<input type="text"/>
Message Waiting Lamp Policy*	Use System Policy
Ring Setting (Phone Idle)*	Ring
Ring Setting (Phone Active)	Use System Default Applies to this line when any line on the phone has a call in progress.
Multiple Call/Call Waiting Settings on Device SEP00115C0E55D4	
Note: The range to select the Max Number of calls is: 1-50	
Maximum Number of Calls*	<input type="text" value="4"/>
Busy Trigger*	<input type="text" value="1"/> (Less than or equal to Max. Calls)
Forwarded Call Information Display on Device SEP00115C0E55D4	
<input checked="" type="checkbox"/> Caller Name	
<input checked="" type="checkbox"/> Caller Number	
<input checked="" type="checkbox"/> Redirected Number	
<input checked="" type="checkbox"/> Dialed Number	
<input type="button" value="Save"/> <input type="button" value="Delete"/> <input type="button" value="Copy"/> <input type="button" value="Reset"/> <input type="button" value="Add New"/>	



Cisco Unified CallManager Line Group for Unity Voice Mail Configuration

Navigation Cisco Unified CallManager Admini

Cisco Unified CallManager Administration For Cisco IP Telecommunication Solutions Logged in as: CCM

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

Line Group Configuration

Related Links: [Back To Fin](#)

Line Group Information

Line Group Name*

RNA Reversion Timeout*

Distribution Algorithm*

Hunt Options

No Answer*

Busy**

Not Available**

Line Group Member Information

Find Directory Numbers to Add to Line Group

Partition

Directory Number Contains

Available DN/Route Partition

6001
6002
6003
6004
6005

Current Line Group Members



6003
6004
6005

Add to Line Group

Current Line Group Members

Reverse Order of Selected DN/Route Partitions

Selected DN/Route Partition

6101
6102
6103
6104

Removed DN/Route Partition
(to be removed from Line Group when you click Save)

Directory Numbers

775 6101 (no partition)
775 6102 (no partition)
775 6103 (no partition)
775 6104 (no partition)

Save **Delete** **Add New**

i *- indicates required item.

i Fields marked with a ** are required when the Distribution Algorithm is set to Top Down or Circular, and are not used when the Distribution Algorithm is set to Idle or Broadcast. The No Answer setting is used for Longest Idle and Broadcast.



Cisco Unified CallManager Hunt List for Unity Voice Mail Configuration

Hunt List Configuration

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

Status
Status: Ready

Hunt List Information

Name*

Description

Cisco Unified CallManager Group*

Enable this Hunt List (change effective on Save; no reset required)

Hunt List Member Information

Add Line Group

Selected Groups * (ordered by highest priority)

Removed Groups (to be removed from Hunt List when you click Save)

Hunt List Details

[Lakers_LG](#)



Cisco Unified CallManager Hunt Pilot for Unity Voice Mail Configuration

Navigation Cisco Unified CallManager Admini

Cisco Unified CallManager Administration For Cisco IP Telecommunication Solutions Logged in as: CCM

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

Hunt Pilot Configuration

Related Links: [Back To Fin](#)

Status
Status: Ready

Pattern Definition

Hunt Pilot *

Partition

Description

Numbering Plan

Route Filter

MLPP Precedence

Hunt List * (Edit)

Route Option
 Route this pattern
 Block this pattern

Provide Outside Dial Tone Urgent Priority

Hunt Forward Settings

Use Personal Preferences Destination Calling Search Space

Forward Hunt No Answer

Forward Hunt Busy

Maximum Hunt Timer



Use Personal Preferences Destination		Calling Search Space
Forward Hunt No Answer	<input type="checkbox"/>	<input type="text"/> < None >
Forward Hunt Busy	<input type="checkbox"/>	<input type="text"/> < None >
Maximum Hunt Timer	<input type="text"/>	
Calling Party Transformations		
<input type="checkbox"/> Use Calling Party's External Phone Number Mask		
Calling Party Transform Mask	<input type="text"/>	
Prefix Digits (Outgoing Calls)	<input type="text"/>	
Calling Line ID Presentation	Default	
Calling Name Presentation	Default	
Connected Party Transformations		
Connected Line ID Presentation	Default	
Connected Name Presentation	Default	
Called Party Transformations		
Discard Digits	< None >	
Called Party Transform Mask	<input type="text"/>	
Prefix Digits (Outgoing Calls)	<input type="text"/>	
AAR Group Settings		
AAR Group	< None >	
External Number Mask	<input type="text"/>	
Save Delete Copy Add New		
*- indicates required item.		








Cisco Unified CallManager Voice Mail Profile Configuration

Navigation Cisco Unified CallManager Admini

Cisco Unified CallManager Administration For Cisco IP Telecommunication Solutions Logged in as: CCM

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

Voice Mail Profile Configuration Related Links: [Back To Find/Li](#)

Status
Status: Ready

Voice Mail Profile Information

Voice Mail Profile Default (used by 9 devices)


Voice Mail Profile Name*


Description

Voice Mail Pilot**

Voice Mail Box Mask

Make this the default Voice Mail Profile for the System

 *- indicates required item.

 **- The Voice Mail Pilot is comprised of the Voice Mail Pilot Number and it's corresponding Calling Search Space Name (< Voice Mail Pilot Number >/< Callin Space >).



Cisco Unified CallManager Voice Mail Pilot Configuration

Navigation Cisco Unified CallManager Admini

Cisco Unified CallManager Administration For Cisco IP Telecommunication Solutions Logged in as: CCM

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

Voice Mail Pilot Configuration Related Links: [Back To Find/Li](#)

Status
 Status: Ready

Voice Mail Pilot Information

Voice Mail Pilot Number

Calling Search Space

Description

Make this the default Voice Mail Pilot for the system

*- indicates required item.



Cisco Unified CallManager Voice Mail Ports List Configuration (Lakers-VI)

Navigation Cisco Unified CallManager Admini

Cisco Unified CallManager Administration For Cisco IP Telecommunication Solutions Logged in as: CCM

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

Find and List Voice Mail Ports

+ [Grid Icon] [Refresh Icon]

Status
8 records found

Search Options
Find Voice Mail Port where Device Name begins with Find Search Within Results
Select item or enter search text

(device.name begins with any)

Search Results

Device Name	Description	Device Pool	SCCP Security Profile	Status	IP Address
<input type="checkbox"/> CiscoUM1-VI1	VM ports for Unity Connection 1.1	Default	Standard SCCP Profile for Auto Registration	Registered with CM-LAKERS	172.20.23
<input type="checkbox"/> CiscoUM1-VI2	VM ports for Unity Connection 1.1	Default	Standard SCCP Profile for Auto Registration	Registered with CM-LAKERS	172.20.23
<input type="checkbox"/> CiscoUM1-VI3	VM ports for Unity Connection 1.1	Default	Standard SCCP Profile for Auto Registration	Registered with CM-LAKERS	172.20.23
<input type="checkbox"/> CiscoUM1-VI4	VM ports for Unity Connection 1.1	Default	Standard SCCP Profile for Auto Registration	Registered with CM-LAKERS	172.20.23
<input type="checkbox"/> Lakers-VI1	VM Ports to Cisco Unity	Default	Standard SCCP Profile for Auto Registration	Registered with CM-LAKERS	172.20.15
<input type="checkbox"/> Lakers-VI2	VM Ports to Cisco Unity	Default	Standard SCCP Profile for Auto Registration	Registered with CM-LAKERS	172.20.15
<input type="checkbox"/> Lakers-VI3	VM Ports to Cisco Unity	Default	Standard SCCP Profile for Auto Registration	Registered with CM-LAKERS	172.20.15
<input type="checkbox"/> Lakers-VI4	VM Ports to Cisco Unity	Default	Standard SCCP Profile for Auto Registration	Registered with CM-LAKERS	172.20.15

Add New Select All Clear All Delete Selected Reset Selected Rows per Page 50

Cisco Unified CallManager Voice Mail Ports Configuration (4 VM Ports)



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

Voice Mail Port Configuration

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

Status
Status: Ready

Device Information

Registration	Registered with Cisco Unified CallManager CM-LAKERS
IP Address	172.20.150.252
Port Name*	<input type="text" value="Lakers-V11"/>
Description	<input type="text" value="VM Ports to Cisco Unity"/>
Device Pool*	<input type="text" value="Default"/>
Calling Search Space	<input type="text" value="< None >"/>
AAR Calling Search Space	<input type="text" value="< None >"/>
Location*	<input type="text" value="Hub_None"/>
SCCP Phone Security Profile*	<input type="text" value="Standard SCCP Profile for Auto Registration"/>

Directory Number Information

Directory Number*	<input type="text" value="6101"/>
Partition	<input type="text" value="< None >"/>
Calling Search Space	<input type="text" value="< None >"/>
AAR Group	<input type="text" value="< None >"/>
Internal Caller ID Display	<input type="text" value="VoiceMail"/>
Internal Caller ID Display (ASCII format)	<input type="text" value="VoiceMail"/>
External Number Mask	<input type="text"/>



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

Voice Mail Port Configuration

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

Status
Status: Ready

Device Information

Registration	Registered with Cisco Unified CallManager CM-LAKERS
IP Address	172.20.150.252
Port Name*	<input type="text" value="Lakers-VI2"/>
Description	<input type="text" value="VM Ports to Cisco Unity"/>
Device Pool*	<input type="text" value="Default"/>
Calling Search Space	<input type="text" value="< None >"/>
AAR Calling Search Space	<input type="text" value="< None >"/>
Location*	<input type="text" value="Hub_None"/>
SCCP Phone Security Profile*	<input type="text" value="Standard SCCP Profile for Auto Registration"/>

Directory Number Information

Directory Number*	<input type="text" value="6102"/>
Partition	<input type="text" value="< None >"/>
Calling Search Space	<input type="text" value="< None >"/>
AAR Group	<input type="text" value="< None >"/>
Internal Caller ID Display	<input type="text" value="VoiceMail"/>
Internal Caller ID Display (ASCII format)	<input type="text" value="VoiceMail"/>
External Number Mask	<input type="text"/>



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

Voice Mail Port Configuration

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

Status
Status: Ready

Device Information

Registration	Registered with Cisco Unified CallManager CM-LAKERS
IP Address	172.20.150.252
Port Name*	<input type="text" value="Lakers-VI3"/>
Description	<input type="text" value="VM Ports to Cisco Unity"/>
Device Pool*	<input type="text" value="Default"/>
Calling Search Space	<input type="text" value="< None >"/>
AAR Calling Search Space	<input type="text" value="< None >"/>
Location*	<input type="text" value="Hub_None"/>
SCCP Phone Security Profile*	<input type="text" value="Standard SCCP Profile for Auto Registration"/>

Directory Number Information

Directory Number*	<input type="text" value="6103"/>
Partition	<input type="text" value="< None >"/>
Calling Search Space	<input type="text" value="< None >"/>
AAR Group	<input type="text" value="< None >"/>
Internal Caller ID Display	<input type="text" value="VoiceMail"/>
Internal Caller ID Display (ASCII format)	<input type="text" value="VoiceMail"/>
External Number Mask	<input type="text"/>



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

Voice Mail Port Configuration

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

Status
Status: Ready

Device Information

Registration	Registered with Cisco Unified CallManager CM-LAKERS
IP Address	172.20.150.252
Port Name*	Lakers-VI4
Description	VM Ports to Cisco Unity
Device Pool*	Default
Calling Search Space	< None >
AAR Calling Search Space	< None >
Location*	Hub_None
SCCP Phone Security Profile*	Standard SCCP Profile for Auto Registration

Directory Number Information

Directory Number*	6104
Partition	< None >
Calling Search Space	< None >
AAR Group	< None >
Internal Caller ID Display	VoiceMail
Internal Caller ID Display (ASCII format)	VoiceMail
External Number Mask	

Cisco Unified CallManager Voice Mail MWI ON and OFF Configuration



Navigation Cisco Unified CallManager Administration

Cisco Unified CallManager Administration For Cisco IP Telecommunication Solutions Logged in as: CCM

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

Message Waiting Configuration Related Links: Back To Fin

Status
Status: Ready

Message Waiting Information

Message Waiting Number* 198

Partition < None >

Description MWI ON for Cisco Unity

Message Waiting Indicator* On Off

Calling Search Space < None >

Save Delete Copy Add New

*- indicates required item.

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



Navigation Cisco Unified CallManager Administration

Cisco Unified CallManager Administration For Cisco IP Telecommunication Solutions Logged in as: CCM

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

Route Pattern Configuration

Related Links: [Back To Fin](#)

Status
Status: Ready

Pattern Definition

Route Pattern*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence*

Gateway/Route List*

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



Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Calling Name Presentation*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

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.



Cisco Unified CallManager Route Pattern (2500) to Nortel Call Pilot VM Pilot Number Configuration

Navigation Cisco Unified CallManager Admini

Cisco Unified CallManager Administration For Cisco IP Telecommunication Solutions Logged in as: CCM

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

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

Status
Status: Ready

Pattern Definition

Route Pattern*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence*

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

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



Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Calling Name Presentation*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

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.



Cisco Unity Configuration

Cisco Unity Software Version

The screenshot shows the Cisco Unity Configuration interface. On the left is a navigation menu with links for Settings, Software Versions, Recordings, Contacts, Phone Languages, and GUI Languages. The main content area is titled 'Configuration' and has a sub-section 'Software Versions'. It displays a list of software components and their versions. At the bottom left of the interface is the Cisco Unity logo and a 'Log off' link. At the bottom of the page is the copyright notice: © 1998-2005 Cisco Systems, Inc.

Component	Version
Cisco Unity Version	4.1
Build Number	4.1(1)
Windows Server Version	Microsoft Windows 2000 build 2195 (Service Pack 4)
System Administrator DLL	4.1.0.237
AVLOGMGRSVR	4.1.0.111
AVRESLOADERSVR	4.0.4.53
DOH	4.1.0.259
AvResMgr	4.0.3.46
AvMiuSvr	4.1.0.209
AVVIRTUALQUEUEESVR	4.0.3.21
AVSASCHEDULERSVR	4.0.4.39
AvRulerSvr	4.0.3.86
AVARBITERSVR	4.1.0.220
AVCONVENGSVR	4.1.0.146
AvPhraseServerSvr	4.1.0.89
AVPAGERCONVSVR	4.1.0.118
AVFAILURECONVSVR	4.0.3.34
AVCONVMGRSVR	4.1.0.146
AVDOHMMSVR	4.0.4.2
AvStatMonSvr	4.1.0.105
AVTrapSVR	4.1.0.84
AVRSASVR	4.0.4.21



Cisco Unity Integration (CM-LAKERS)

Integration
[Cisco CallManager](#)

Integrations

Cisco CallManager

Integration Type	Cisco CallManager
Switch File	cisco0002.ini

CM-KINGS

Primary Server	172.20.150.251:2000
Device Name Prefix	CiscoUM1-VI
MWI On Extension	3598
MWI Off Extension	3599
Reconnect After CallManager Failback	Yes

CCM41

Primary Server	172.20.150.253:2000
Device Name Prefix	CCM41-VI
MWI On Extension	3698
MWI Off Extension	3699
Reconnect After CallManager Failback	Yes

CM-MOON

Primary Server	172.20.201.254:2000
Device Name Prefix	MoonUM1-VI
MWI On Extension	4198
MWI Off Extension	4199

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Integration
Cisco CallManager

Integrations

MWI On Extension	3598
MWI Off Extension	3599
Reconnect After CallManager Failback	Yes

CCM41

Primary Server	172.20.150.253:2000
Device Name Prefix	CCM41-VI
MWI On Extension	3698
MWI Off Extension	3699
Reconnect After CallManager Failback	Yes

CM-MOON

Primary Server	172.20.201.254:2000
Device Name Prefix	MoonUM1-VI
MWI On Extension	4198
MWI Off Extension	4199
Reconnect After CallManager Failback	Yes

CM-LAKERS

Primary Server	172.20.152.253:2000
Device Name Prefix	Lakers-VI
MWI On Extension	6198
MWI Off Extension	6199
Reconnect After CallManager Failback	Yes

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Subscribers

- [Profile](#)
- [Account](#)
- [Phone Password](#)
- [Private Lists](#)
- [Conversation](#)
- [Call Transfer](#)
- [Greetings](#)
- [Caller Input](#)
- [Messages](#)
- [Message Notification](#)
- [Alternate Extensions](#)
- [Features](#)

Lakers 6003

Call Transfer

Transfer incoming calls to subscriber's phone?

No (send directly to subscriber's greeting)

Yes, ring subscriber's extension:

Yes, ring subscriber at this number:

While transferring, notify caller?

Do not play the "Wait while I transfer your call" prompt

Transfer type:

Release to switch

Supervise transfer

Rings to wait for:

If the call is busy

Always hold

No holding

Ask caller

Gather caller information:

Announce

Introduce (call for *name*)

Confirm (call can be accepted or refused)

Ask caller's name

Cisco **Unity** [Log off](#)

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Cisco Unity Subscriber (SIP 6007) Configurations

Subscribers

- [Profile](#)
- [Account](#)
- [Phone Password](#)
- [Private Lists](#)
- [Conversation](#)
- [Call Transfer](#)
- [Greetings](#)
- [Caller Input](#)
- [Messages](#)
- [Message Notification](#)
- [Alternate Extensions](#)
- [Features](#)

Cisco
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Lakers 6007

Profile

Subscriber Information

First name:

Last name:

Display name:

Class of service: [View](#)

Extension:

Fax ID:

Recorded voice: Volume

Active schedule: [View](#)

Time zone:

Set subscriber for self-enrollment at next login

List in phone directory

Show subscriber in e-mail server address book

Exchange Information

Alias:

Server:



Subscribers

- [Profile](#)
- [Account](#)
- [Phone Password](#)
- [Private Lists](#)
- [Conversation](#)
- [Call Transfer](#)
- [Greetings](#)
- [Caller Input](#)
- [Messages](#)
- [Message Notification](#)
- [Alternate Extensions](#)
- [Features](#)

Lakers 6007

Call Transfer

Transfer incoming calls to subscriber's phone?

No (send directly to subscriber's greeting)

Yes, ring subscriber's extension:

Yes, ring subscriber at this number:

While transferring, notify caller?

Do not play the "Wait while I transfer your call" prompt

Transfer type:

Release to switch

Supervise transfer

Rings to wait for:

If the call is busy

Always hold

No holding

Ask caller

Gather caller information:

Announce

Introduce (call for *name*)

Confirm (call can be accepted or refused)

Ask caller's name

Cisco **Unity** [Log off](#)

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Subscribers

- [Profile](#)
- [Account](#)
- [Phone Password](#)
- [Private Lists](#)
- [Conversation](#)
- [Call Transfer](#)
- [Greetings](#)
- [Caller Input](#)
- [Messages](#)
- [Message Notification](#)
- [Alternate Extensions](#)
- [Features](#)

CS101 SIP2201

Call Transfer

Transfer incoming calls to subscriber's phone?

- No (send directly to subscriber's greeting)
- Yes, ring subscriber's extension:
- Yes, ring subscriber at this number:

While transferring, notify caller?

- Do not play the "Wait while I transfer your call" prompt

Transfer type:

- Release to switch
- Supervise transfer

Rings to wait for:

If the call is busy

- Always hold
- No holding
- Ask caller

Gather caller information:

- Announce
- Introduce ("call for name")
- Confirm (call can be accepted or refused)
- Ask caller's name

Cisco **Unity** [Log off](#)

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Cisco Unity Subscriber (Nortel ext. 2213) Configurations

Subscribers

- [Profile](#)
- [Account](#)
- [Phone Password](#)
- [Private Lists](#)
- [Conversation](#)
- [Call Transfer](#)
- [Greetings](#)
- [Caller Input](#)
- [Messages](#)
- [Message Notification](#)
- [Alternate Extensions](#)
- [Features](#)

CS101 Zeus2213

Profile

Subscriber Information

First name: CS101

Last name: Zeus2213

Display name: CS101 Zeus2213

Class of service: (Default Subscriber) View

Extension: 2213

Fax ID:

Recorded voice: 0.0 0.0 Volume

Active schedule: Weekdays View

Time zone: Default

Set subscriber for self-enrollment at next login

List in phone directory

Show subscriber in e-mail server address book

Exchange Information

Alias: CZeus2213

Server: CHINHUNITY

Cisco Unity Log off

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Subscribers

- Profile
- Account
- Phone Password
- Private Lists
- Conversation
- Call Transfer
- Greetings
- Caller Input
- Messages
- Message Notification
- Alternate Extensions
- Features

CS101 Zeus2213

Call Transfer

Transfer incoming calls to subscriber's phone?

No (send directly to subscriber's greeting)

Yes, ring subscriber's extension:

Yes, ring subscriber at this number:

While transferring, notify caller?

Do not play the "Wait while I transfer your call" prompt

Transfer type:

Release to switch

Supervise transfer

Rings to wait for:

If the call is busy

Always hold

No holding

Ask caller

Gather caller information:

Announce

Introduce ('call for name')

Confirm (call can be accepted or refused)

Ask caller's name

Cisco
Unity [Log off](#)

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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



Acronym	Definitions
CUCM	Cisco Unified CallManager
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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