Nortel CS1000M Release 4.0 using E1 ISO-QSIG to Cisco Unified CallManager Express 4.0(2)

November 1, 2007 Version 2

Table of Contents

Introduction ...................................................................................................................1
Network Topology ........................................................................................................2
Limitations ....................................................................................................................3
System Components .....................................................................................................5
  Hardware Requirements ..............................................................................................5
  Software Requirements ..............................................................................................5
Features .......................................................................................................................6
  Features Supported ...................................................................................................6
  Features Not Supported ............................................................................................7
Configuration ...............................................................................................................8
  Configuration sequence for the Nortel CS1000M PBX ..............................................8
  Configuring the Nortel CS1000M PBX .................................................................9
  Configuring the Local Cisco Unified CallManager Express (Cisco 3845) ...............18
  Configuring the Cisco Unified CallManager Express 2 (Cisco 2801) ....................26
Acronyms ..................................................................................................................32

Introduction

This is an Application Note for connectivity between a Nortel CS1000M Release 4.0 PBX and Cisco Unified CallManager Express Release 4.0(2) using a Cisco 3845 voice gateway with QSIG protocol.

The network topology diagram (Figure 1) shows the test setup for end-to-end interoperability with Cisco Unified CallManager Express Release 4.0(2) connected to the PBX via the 3845 E1 QSIG link. The 3845 IOS voice gateway was connected via H.323 to a Cisco 2801 IOS voice gateway. The two gateways were running Cisco Unified CallManager Express 4.0(2). Cisco Unified IP phones (models 7960 and 7961G) were connected to the 2 Cisco Unified CallManager Express gateways via SIP and SCCP, as per the figure. A NM-HDV and VWIC-2MFT-E1 were used for the E1 QSIG interface. Calls were made to test basic call, caller ID, conference, transfer, forward, call back, reroute, and MWI features.

This Application Note uses the 3845 voice gateway. However, the use of other Cisco voice gateways is also an option since Cisco Unified Call Manager Express does not depend on platform. The listed gateway families, below, can run Cisco Unified CallManager Express, but each have different IP phone support capability. Please check the product specifications to ensure you are obtaining the proper device to support your IP phone deployment.

Cisco IAD 2430 Series Integrated Access Devices
Cisco 2801 Integrated Services Router, 1760-V and 1751-V Access Routers
Cisco 2811 Integrated Services Router, 261xXM and 262xXM Series Access Routers
Cisco 2821 Integrated Services Router, 265xXM Access Router
Cisco 2691 Multiservice Access Router
Cisco 2851 Integrated Services Router
Cisco 3725 Multiservice Access Router
Cisco 3745 Multiservice Access Router
Cisco 3825 Integrated Services Router
Cisco 3845 Integrated Services Router
The inclusion of Cisco SIP phones in this application note is for reference only. Cisco Unified Communications Manager Express 4.0(3) supports SIP end-points with limited number of features.

**Network Topology**

Figure 1. Network Topology or Test Setup – basic calls configuration.
Limitations

Basic Calls

Cisco Unified CallManager Express (CME) does not support overlap sending. It supports overlap receiving.

Connected Name and Alerting Name are not supported on calls between PBX and Cisco Unified IP Phone running SIP. This is a CME SIP limitation

Calling Name Restriction is not supported for calls originated from Cisco Unified CallManager Express 4.0(3). This is a CME SIP limitation

Connected Number/Name Restriction is not supported from Cisco Unified CallManager Express 4.0(3). This is a CME SIP limitation

Call Transfers

A Consulted transfer or Early-attended transfer originated from a call placed from a phone on the remote Cisco Unified CallManager Express (CME2) to a SIP phone on the local Cisco Unified CallManager Express (CME1), and then transferred to a PBX phone (e.g., G1 calls C2, and C2 transfers to A) does not complete. This is a CME SIP limitation.

For local consulted and early-attended call transfers between SCCP phones and SIP phones, call originates from an external PBX phone, the Calling name and number updates are not supported. This is CME SIP to SCCP interworking limitation.

For local consulted, early attended and blind call transfers connected name and number are not supported. CME does not support Facility IE for call update information.

For external consulted and early attended call transfers with call flow, (CME IP phone calls PBX phone, PBX phone transfers back to different IP phone on CME (trombone)) the called (connected) name and number are not updated on the original phone after the transfer is complete. (e.g. Phone C1 calls Phone A, Phone A transfers to Phone D1). CME does not support Facility IE for call update information.

Call Forwards

For local call forward calls involving SIP phones the forwarding name/number display is not supported. This is a CME SIP limitation.

For external Call forward calls the forwarding number is not supported on CME. CME does not support RedirectingName.

For external call forward calls, the called (connected) number is not updated on the original phone. This is a CME limitation.

For call forward calls, the called (connected) name is not updated on the SIP phone. This is a CME SIP limitation.

Forwarded calls originated from a PBX extension to a remote Cisco Unified CallManager Express SCCP extension, and forwarded to a local Cisco Unified CallManager Express extension (e.g., A calls G1, and G1 forwards to C2), Cisco Unified CallManager Express performs a QSIG reroute, even though a QSIG reroute is not in order (i.e., there is no QSIG “hairpin” or “trombone”).

The Nortel PBX does not support reroute on forwarded calls resulting in a hairpin (i.e., Cisco Unified CallManager Express 4.0(2) phone calls a PBX phone that forwards back to another Cisco Unified CallManager Express 4.0(2) phone).

Forwarded calls that are hairpinned at a SIP extension (PBX phone calls Cisco Unified CallManager Express 4.0(2) SIP phone that forwards back to another PBX phone), the call completes, but Cisco Unified CallManager Express 4.0(2) does not perform a reroute, even if reroute is enabled.

Forwarded calls originated from a PBX extension to a local Cisco Unified CallManager Express SCCP extension, and forwarded to another local Cisco Unified CallManager Express extension (e.g., A calls C1, and C1 forwards to D1 or D2), Cisco Unified CallManager Express performs a reroute, and even though a reroute is not in order (i.e., there is no “hairpin” or “trombone”).

For calls that are hairpinned at a SIP extension (PBX phone calls Cisco Unified CallManager Express 4.0(2) SIP phone that forwards unconditionally back to another PBX phone) when a CFNR number was set up resulted in a 3rd SETUP message from CME. The timeout is set under the CFNR command. If enough time passes before the final destination (B) answers, the CFNR is invoked, and the 3rd SETUP is sent from CME. A new (3rd) B-channel is set up. The 2nd one is then torn down.

Forwarded calls that are initiated by overlap dialing from a PBX extension to a Cisco Unified CallManager Express extension, the call completes, but Cisco Unified CallManager Express does not perform a reroute, even if reroute is enabled and the call is eligible for a reroute.
MWI

Cisco Unified Communications Manager Express 4.0(3) supports Cisco Unity integration with QSIG. However, in this instance, no testing was performed with Cisco Unified Communications Manager Express 4.0(3) as the message center PINX.

MWI was not tested for SIP extensions on Cisco Unified CallManager Express 4.0(3) with the PBX as the message center PINX. It was tested for SCCP extensions only.
System Components

Hardware Requirements

Cisco 3845 IOS voice gateway
   NM-HDV
   VWIC-2MFT-E1
Cisco 2801 IOS voice gateway
(4) Cisco Unified IP phone 7960s
(2) Cisco Unified IP phone 7961G
(1) Nortel Communication Server 1000
(2) Nortel 2616 digital station phones
(1) NTBK50AA, (Release 06) E1 trunk cards
(1) NTRH30AA, (Release 12) voice mail card
Nortel CallPilot 201i voice mail system

Software Requirements

Cisco Unified CallManager Express Release 4.0(2)
Cisco IOS Software, 3800 Software (C3845-IPVOICE-M), Version 12.4(11)T
Cisco IOS Software, 2801 Software (C2801-IPVOICE-M), Version 12.4(11)T
Nortel CS1000M Release 4.0
Nortel CallPilot 201i Release 2.02

G1, G2 – 7960 – SCCP

Cisco 7960 IP phone version 7.2(T0.23)
Cisco 7960 IP phone app load P00308000400
Cisco 7960 IP phone boot load PC0303010001

C2, D2 – 7960 – SIP

Cisco 7960 DSP load ID 4.0(2.0)[A0]
Cisco 7960 IP phone app load P0S3-08-4-00
Cisco 7960 IP phone boot load PC030301

C1 – 7961G – SCCP

Cisco 7961G IP phone load file: SCCP41.8-0-3S
Cisco 7961G IP phone app load ID: Jar41scp.8-0-2-25.sbn
Cisco 7961G IP phone boot load ID: 7961G_64-020704128Amd64meg.bin
Features

Features Supported

Basic Call, ENBLOC
Basic Call, Overlap (from PBX to Cisco Unified CallManager Express only)
CLIP-Calling Line (Number) Identification Presentation
CLIR-Calling Line (Number) Identification Restriction
CNIP-Calling Name Identification Presentation
CNIR-Calling Name Identification Restriction (from PBX to Cisco Unified CallManager Express only)
COLP-Connected Line (Number) Identification Presentation on Basic Calls
CONP-Connected Name Identification Presentation (for calls between PBX and Cisco Unified IP Phones running SCCP)
Alerting Name (for calls between PBX and Cisco Unified IP Phones running SCCP)
Tandem PSTN call
Consultation Transfer – Local
Consultation Transfer – Network/External (See Limitations Section)
Early Attended Transfer – Local
Early Attended Transfer – Network/External (See Limitations Section)
Blind Transfer – Local (See Limitations Section)
Blind Transfer – Network/External (See Limitations Section)
Call Forward Unconditional by Join – Local (See Limitations Section)
Call Forward Unconditional by Join – Network/External (See Limitations Section)
Call Forward Busy by Join – Local (See Limitations Section)
Call Forward Busy by Join – Network/External (See Limitations Section)
Call Forward No Reply by Join – Local (See Limitations Section)
Call Forward No Reply by Join – Network/External (See Limitations Section)
Call Forward Unconditional by Reroute – Network/External (See Limitations Section)
Call Forward Busy by Reroute – Network/External (See Limitations Section)
Call Forward No Reply by Reroute – Network/External (See Limitations Section)
MWI (See Limitations Section)
Features Not Supported

Overlap dialing from Cisco Unified CallManager Express 4.0(3) to PBX
CNIR-Calling Name Identification Restriction from Cisco Unified CallManager Express 4.0(2) to PBX
COLR- Connected Line (Number) Identification Restriction
CONR- Connected Name Identification Restriction
CONP-Connected Name Identification Presentation (for calls between PBX and Cisco Unified IP Phones running SIP)
Alerting Name (for calls between PBX and Cisco Unified IP Phones running SIP)
H323/QSIG tandem transfers via SIP phone
CLIP-Calling Line (Number) Identification Presentation on Transferred Calls
CNIP-Calling Name Identification Presentation on Transferred Calls
COLP-Connected Line (Number) Identification Presentation on Transferred Calls
CONP-Connected Name Identification Presentation on Transferred Calls
CNIP-Calling Line (Name) Identification Presentation on Forwarded Calls to a PBX station
COLP-Connected Line (Number) Identification Presentation on Forwarded Calls
CONP-Connected Name Identification Presentation on Forwarded Calls
Call Forward by Reroute for QSIG "trombone" from a Cisco Unified CallManager Express SIP extension
Call Forward by Reroute with overlap dialing
Cisco Unity integration with QSIG.
MWI with QSIG/SIP interworking
Call Completion to Busy Subscriber (Call Back when Free)
Call Completion on No Reply (Call Back Next Used)
Path Replacement for Call Transfer by Join
Path Replacement for Trombone Connection
Path Replacement for Call Diversion by Forward Switch
**Configuration**

Configuration sequence for the Nortel CS1000M PBX

1. Configure T1-PRI-QSIG.
2. Configure Route List.
3. Configure Coordinated Dial Plan
4. Configure MSDL card
5. Configure Digital Station Phone
Configuring the Nortel CS1000M PBX

CONFIGURATION FOR TRUNKS

PRI config (PBX card slot 5)

>ld 22
PT2000

REQ prt
TYPE cequ

CEQU
MPED 8D
SUPL 000 004 008 012
  016 032 036 040
  044 048 064 068
  072 V096 V100
TD S 000
CONF 029 030 031 062
  094 095

DLOP NUM DCH FRM TMDI LCMT YALM T1TE TRSH
PRI 02 23 ESF NO B8S FDL - 00
  04 24 ESF YES B8S DG2 0 00
PRI2 05 06 07 11
DTI2 12 13 21
MISP

D-Channel for T1-QSIG trunk to Cisco Unified CallManager Express (PBX card slot 5)

ld 22
PT2000

REQ prt
TYPE adan dch 5

ADAN DCH 5
CTYP MSDL
CARD 05
PORT 1
DES E1_QSIG_CME
USR PRI
DCHL 5
OTBF 32
PARM RS422 DTE
DRAT 64KC
CLOK EXT
IFC ISGF
P INX_CUST 0
ISDN_MCNT 300
CLID OPT0
CO_TYPE STD
SIDE NET
CNEG 1
RLS ID **
QCHID YES
B-Channels for T1-QSIG route to Cisco Unified CallManager Express (PBX card slot 5)

$id 21
PT1000

REQ: prt
TYPE: rdb
CUST 0
ROUT 105

TYPE RDB
CUST 00
DMOD
ROUT 105
DES E1_QSIG_CME
TKTP TIE
NPID_TBL_NUM 0
ESN NO
CNVT NO
SAT NO
RCLS EXT
VTRK NO
NODE
DTRK YES
BRIP NO
DGTP PRI2
ISDN YES
MODE PRA
IFC ISGF
SBN NO
PNI 00000
NCNA NO
NCRD NO
CTYP UKWN
INAC NO
ISAR NO
CPFXS YES
DACP NO
INTC NO
DSEL VOD
PTYP DTT
AUTO NO
DNIS NO
DCDR NO
ICOG IAO
SRCH LIN
TRMB YES
STEP
ACOD 205
TCP NO
TARG 01
CLEN 1
BILN NO
OABS
INST
ANTK
SIGO STD
ICIS YES
TIMR ICF 512
OGF 512
EOD 13952
NRD 10112
DDL 70
ODT 4096
RGV 640
GRD 896
SFB 3
NBS 2048
NBL 4096

IENB 5
TFD 0
VSS 0

PAGE 002

VGD 6
DRNG NO
CDR NO
VRAT NO
MUS 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
PLEV 2
ALRM NO
ART 0
SGRP 0
AACR NO
B-Channels for T1-QSIG trunk to Cisco Unified CallManager Express (PBX card slot 5)

Id 20

PT0000
REQ: prt
TYPE: tie
TN 5 1
DATE
PAGE

DES E1_QSIG_CME
TN 005 01
TYPE TIE
CDEN SD
CUST 0
TRK PRI2
PDCA 1
PCML A
NCOS 0
RTMB 105 1
B-CHANNEL SIGNALING
TGAR 1
AST NO
IAPG 0
CLS UNR DTN WTA LPR APN THFD
   P10 VNL
TKID
AACR NO
DATE 4 DEC 2006
ROUTE LIST

Route for card in slot 5 (T1-QSIG trunk to Cisco Unified CallManager Express)

>ld 86
ESN000

MEM AVAIL: (U/P): 2819884 USED U P: 205875 70816 TOT: 3096575
DISK RECS AVAIL: 1152
REQ   prt
CUST  0
FEAT  rlb
RLI  5

RLI  5
ENTR  0
LTER  NO
ROUT  105
TOD  0 ON  1 ON  2 ON  3 ON
  4 ON  5 ON  6 ON  7 ON
VNS  NO
CNV  NO
EXP  NO
FRL  0
DMI  0
FCI  0
FSNI 0
SBOC NRR
IDBB DBD
IOHQ NO
OHQ  NO
CBQ  NO

ISET  0
NALT  5
MFRL  0
OVLL  2
CDP - COORDINATED DIAL PLAN

CDP for 40XX (toward T1-QSIG trunk to Cisco Unified CallManager Express)

>ld 87
ESN000

MEM AVAIL: (U/P): 2819884 USED U P: 205875 70816 TOT: 3096575
DISK RECS AVAIL: 1152
REQ prt
CUST 0
FEAT cdp
TYPE dsc
DSC 40
DSC 40
FLEN 0
DSP LSC
RLI 5
NPA
NXX

Nortel E1 MSDL card configuration

>ld 73
DDB000
MEM AVAIL: (U/P): 2819884 USED U P: 205875 70816 TOT: 3096575
DISK RECS AVAIL: 1152
REQ prt
TYPE lpti
SCH0111
TYPE pri2
FEAT lpti
LOOP 5

MFF CRC
ACRC NO
ALRM REG
RAIE NO
G1OS YES
SLP 5 24 H 30 1 H
BPV 128 122
CRC 201 97
FAP 28 1
RATS 10
GP2 20 100 S 12 S 12 S 4 S
MNG1 15 M
NCG1 15 M
OSG1 15 M
MNG2 15 S
NCG2 15 S
OSG2 15 S
PERS 50
CLRS 50
OOSC 5
CONFIGURATIONS FOR MERIDIAN PHONES x2213 AND x2214

LD 11

REQ PRT
TYPE:
TYPE 2616
TN 1 0
DATE
PAGE
DES

DES CS101A
TN 001 000 00
TYPE 2616
CDEN 8D
CUST 0
AOM 0
FDN 2500
TGAN 1
LDN NO
NCOS 0
SGRP 0
RNPG 0
SCI 0
SSU
XLS

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 CSDS SWD LND CNDA
  CFTA SFD MRD DDV CNIA CDCA MSID DAPA BFED RCBD
  ICDD CDMD LLCN MCTD CLBD AUTU
  GPUD DPUD DNDA CFXD ARHD CLTD ASCD
  CPFA CPTA ABDD CFHD FICD NAID BUZZ AGMD MOAD AHD
  DDGA NAMA
  DRDD EXR0
  USRD ULAD RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3 MCBN CDMR
CPND_LANG ENG
Rco 0
EFD 2500
HUNT 2500
EHT 2500
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 ZEUS13
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  DISPLAY_FMT FIRST, LAST
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Configuring the Local Cisco Unified CallManager Express (Cisco 3845)

c3845CME#sh run
Building configuration...

Current configuration : 4560 bytes

version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption

hostname c3845CME

boot-start-marker
boot system flash:c3845-ipvoice-mz.124-11.T.bin
boot-end-marker

logging buffered 10000000
no logging console
enable password cisco

no aaa new-model
network-clock-participate wic 0
network-clock-select 1 E1 0/0/1
ip cef

no ip dhcp use vrf connected
ip dhcp excluded-address 200.1.1.1

ip dhcp pool phone
  network 200.1.1.0 255.255.255.0
  option 150 ip 200.1.1.1
default-router 200.1.1.1

no ip domain lookup
multilink bundle-name authenticated
isdn switch-type primary-qsig
voice-card 0
no dspfarm
voice call send-alert
voice service pots
  supplementary-service qsig call-forward
voice service voip
  qsig decode
  allow-connections h323 to h323
  allow-connections h323 to sip
  allow-connections sip to h323
  allow-connections sip to sip
  h323
  sip
    bind control source-interface GigabitEthernet0/1
    bind media source-interface GigabitEthernet0/1
    rel1xx disable
    min-se 100
ds0-num
header-passing
registrar server

voice register global
mode cme
source-address 200.1.1.1 port 5060
max-dn 100
max-pool 192
load 7960-7940 POS3-07-5-00
tftp-path flash:
create profile sync 0012445613433369
!
voice register dn 1
number 4000
call-forward b2bua noan 2214 timeout 5
name Zidane
huntstop
!
voice register dn 2
number 4001
call-forward b2bua busy 2500
call-forward b2bua noan 2500 timeout 5
name Platini
huntstop
!
voice register pool 1
id mac 000F.9054.2FC2
type 7960
number 1 dn 1
max registrations 240
dtmf-relay rtp-nte
description Zidane
!
voice register pool 2
id mac 0012.4362.BF71
type 7960
number 1 dn 2
max registrations 240
dtmf-relay rtp-nte
description Platini
!
!
controller E1 0/0/0

controller E1 0/0/1
clock source line primary
pri-group timeslots 1-31
!
!
!
!
interface GigabitEthernet0/0
   ip address 172.20.8.26 255.255.255.0
duplex auto
speed auto
media-type rj45
no keepalive
!
interface GigabitEthernet0/1
   ip address 200.1.1.1 255.255.255.0
duplex auto
speed auto
media-type rj45
no keepalive
!
interface Serial0/0/1:15
no ip address
encapsulation hdlc
isdn switch-type primary-qsig
isdn overlap-receiving
isdn incoming-voice voice
isdn bchan-number-order ascending
no cdp enable
!
ip default-gateway 172.20.8.1
ip route 0.0.0.0 0.0.0.0 172.20.8.1
ip route 201.2.2.0 255.255.255.0 172.20.8.27
!  
!  
ip http server  
ip http authentication local  
ip http path flash:  
!  
!  
tftp-server flash:PO03-07-5-00.bin  
tftp-server flash:PO03-07-5-00.sbn  
tftp-server flash:POS3-07-5-00.bin  
tftp-server flash:POS3-07-5-00.sb2  
tftp-server flash:POS3-07-5-00.loads  
tftp-server flash:TERM41.7-0-3-0S  
tftp-server flash:PO030702T023  
!  
control-plane  
!  
!  
voice-port 0/0/1:15  
mwi  
!  
!  
!  
!  
!  
dial-peer voice 6000 voip  
destination-pattern 4..[89]  
session target ipv4:201.2.2.1  
no vad  
!  
dial-peer voice 95558000 pots  
destination-pattern 3...  
direct-inward-dial  
port 0/0/1:15  
forward-digits all  
!  
dial-peer voice 2200 pots
destination-pattern 2...
imcoming called-number ....
direct-inward-dial
port 0/0/1:15
forward-digits all
!
dial-peer voice 5200 voip
destination-pattern 5...
session target ipv4:201.2.2.1
no vad
!
!
gateway
timer receive-rtp 1200
!
sip-ua
retry options 0
mwi-server ipv4:200.1.1.1 expires 3600 port 5060 transport udp
!
!
telephony-service
load 7960-7940 P0030702T023
load 7961 TERM41.7-0-3-0S
max-ephones 96
max-dn 192
ip source-address 200.1.1.1 port 2000
system message ABC Corp
mwi relay
max-conferences 8 gain -6
call-forward pattern .T
moh music-on-hold.au
dn-webedit
time-webedit
transfer-system full-blind
transfer-pattern ....
secondary-dialtone 9
create cnf-files version-stamp Jan 01 2002 00:00:00
!
!
ephone-dn 3 dual-line
call-waiting ring
number 4002
label 4002
description Pele
name Pele
call-forward busy 2500
call-forward noan 2500 timeout 5
huntstop channel
!
!
ephone-dn 4 dual-line
call-waiting ring
number 4003
label 4003
description Beckenbauer
name Beckenbauer
huntstop channel
!
!
ephone-dn 5 dual-line
call-waiting ring
number 4004
label 4004
description H. Sanchez
name Sanchez
huntstop channel
!
!
ephone 3
mac-address 0017.0EEE.2F5E
type 7961
keep-conference
button 1:3
!
!
ephone 4
mac-address 0015.2B8F.351B
type 7961
keep-conference
button 1:4
!
!
!
ephone 5
mac-address 0014.1C48.DE7A
type 7960
keep-conference
button 1:5
!
!
!
!
line con 0
password cisco
login
stopbits 1
line aux 0
stopbits 1
line vty 0 4
exec-timeout 0 0
password cisco
login
!
scheduler allocate 20000 1000
!
end
c3845CME#
Configuring the Cisco Unified CallManager Express 2 (Cisco 2801)
c2801CME#sh run
Building configuration...

Current configuration : 3139 bytes
!
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname c2801CME
!
boot-start-marker
boot system flash:c2801-ipvoice-mz.124-11.T.bin
boot-end-marker
!
logging buffered 100000000
no logging console
enable password cisco
!
no aaa new-model
network-clock-participate wic 1
network-clock-select 1 E1 0/1/1
ip cef
!
!
no ip dhcp use vrf connected
ip dhcp excluded-address 201.2.2.1
!
ip dhcp pool phone
  network 201.2.2.0 255.255.255.0
  option 150 ip 201.2.2.1
  default-router 201.2.2.1
!
!
no ip domain lookup
multilink bundle-name authenticated
isdn switch-type primary-qsig

voice-card 0

voice service pots

voice service voip
  qsig decode
  allow-connections h323 to h323
  allow-connections h323 to sip
  allow-connections sip to h323
  allow-connections sip to sip

h323
  sip
  registrar server expires max 600 min 60

controller E1 0/1/0

controller E1 0/1/1
  pri-group timeslots 1-31
interface FastEthernet0/0
  ip address 172.20.8.27 255.255.255.0
duplex auto
  speed auto

interface FastEthernet0/1
  ip address 201.2.2.1 255.255.255.0
duplex auto
  speed auto

interface Serial0/1:15
  no ip address
  encapsulation hdlc
  isdn switch-type primary-qsig
  isdn incoming-voice voice
  no cdp enable

  ip default-gateway 172.20.8.1
  ip route 0.0.0.0 0.0.0.0 172.20.8.1
  ip route 200.1.1.0 255.255.255.0 172.20.8.26

  ip http server
  ip http authentication local
  ip http path flash:

  disable-eadi

  tftp-server flash:P003-07-5-00.bin
tftp-server flash:P003-07-5-00.sbn
tftp-server flash:P0S3-07-5-00.bin
tftp-server flash:P0S3-07-5-00.sb2
tftp-server flash:P0S3-07-5-00.loads
tftp-server flash:TERM41.7-0-3-0S
tftp-server flash:P0030702T023

! control-plane
!
!
! voice-port 0/1/1:15
!
!
!

dial-peer voice 4000 voip
   destination-pattern 4..[0123]
   session target ipv4:200.1.1.1
   no vad
!
dial-peer voice 9 voip
   destination-pattern 3...
   session target ipv4:200.1.1.1
   no vad
!
dial-peer voice 5200 pots
   destination-pattern 5...
   incoming called-number ....
   direct-inward-dial
   port 0/1/1:15
   forward-digits all
!
dial-peer voice 2200 voip
   destination-pattern 2...
   session target ipv4:200.1.1.1
   dtmf-relay rtp-nte
   no vad
!
!
!
telephony-service
load 7960-7940 P0030702T023
load 7961 TERM41.7-0-3-0S

max-ephones 30
max-dn 150
ip source-address 201.2.2.1 port 2000
system message CBA Corp
max-conferences 8 gain -6
call-forward pattern .T
moh music-on-hold.au
dn-webedit
time-webedit
transfer-system full-blind
transfer-pattern ....
secondary-dialtone 9
create cnf-files version-stamp 7960 Oct 12 2006 11:41:08
!
!
ephone-dn 1 dual-line
number 4008
label 4008
description Ronaldinho
name Ronaldinho
call-forward busy 2213
call-forward noan 2500 timeout 5
huntstop channel
!
!
ephone-dn 4 dual-line
number 4009
label 4009
description Tevez
name Tevez
call-forward noan 5002 timeout 10
huntstop channel
!
!
ephone 1
mac-address 000F.9069.DB2C
type 7960
<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definitions</th>
</tr>
</thead>
<tbody>
<tr>
<td>BRI</td>
<td>Basic Rate ISDN</td>
</tr>
<tr>
<td>CAMA</td>
<td>Centralized Automatic Message Accounting</td>
</tr>
<tr>
<td>CAS</td>
<td>Channel Associated Signaling</td>
</tr>
<tr>
<td>CFB</td>
<td>Call Forward when Busy</td>
</tr>
<tr>
<td>CFNR</td>
<td>Call Forward when No Reply</td>
</tr>
<tr>
<td>CFU</td>
<td>Call Forward Unconditional</td>
</tr>
<tr>
<td>CO</td>
<td>Central Office</td>
</tr>
<tr>
<td>FGD</td>
<td>Feature Group “D”</td>
</tr>
<tr>
<td>FXO</td>
<td>Foreign Exchange – Office</td>
</tr>
<tr>
<td>FXS</td>
<td>Foreign Exchange – Station</td>
</tr>
<tr>
<td>IOS</td>
<td>Internetworking Operating System</td>
</tr>
<tr>
<td>MCID</td>
<td>Malicious Caller ID</td>
</tr>
<tr>
<td>MGCP</td>
<td>Media Gateway Control Protocol</td>
</tr>
<tr>
<td>MoH</td>
<td>Music on Hold</td>
</tr>
<tr>
<td>MWI</td>
<td>Message Waiting Indication</td>
</tr>
<tr>
<td>PBX</td>
<td>Private Branch Exchange</td>
</tr>
<tr>
<td>PRI</td>
<td>Primary Rate ISDN</td>
</tr>
<tr>
<td>PSAP</td>
<td>Public Service Access Point</td>
</tr>
<tr>
<td>SIP</td>
<td>Session Initiation Protocol</td>
</tr>
<tr>
<td>ToH</td>
<td>Tone on Hold</td>
</tr>
</tbody>
</table>
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