



Avaya S8500 CM Release 4.0 with Avaya SIP Enablement Services Server Using SIP to Cisco Unified Communications Manager Express 7.0

April 1, 2008

Table of Contents

Introduction	2
Network Topology	3
Limitations	4
System Components	5
Hardware Components	5
Software Requirements	5
Features	6
Features Supported	6
Features Not Supported	6
Configuration	7
Configuring the Avaya S8500 Server	7
Configuring the Avaya SIP Enablement Services Server	17
Configuring the Cisco Unified Communications Manager Express (Delete if Toll Bypass)	24
Acronyms	35



Introduction

- This application note demonstrates connectivity between an Avaya S8500 Communications Manager Release 4.0 with Avaya SIP Enablement Services (SES) server and Cisco Unified Communications Manager Express (CUCME) Release 7.0 (IOS 12.4[20]T) using a Cisco 3845 ISR with SIP protocol.
- Voice mail testing was not performed because Avaya SES does not support SIP MWI messaging across external trunks.
- The network topology diagram (Figure 1) shows the test setup for end-to-end interoperability with CUCME Release 7.0 connected to the PBX via SIP trunk. Unified IP phones (models 7960 and 7961G) were registered to the CUCME via SIP and SCCP, as shown in Figure 1. An NM-HDV2 was used for transcoding resources. Validated call flows include basic call using G729 across SIP trunk, caller ID, conference using G729 across SIP trunk, call-transfer (hairpin and REFER method), call-forward (hairpin and Moved-Temporarily method), call-hold/resume.
- This application note uses the 3845 ISR router. However, the use of other Cisco voice gateways is also an option because the CUCME application is independent of the Cisco platform. The gateway families listed below can run CUCME, but each has different VoIP capacity or capability. Please check the product specifications to ensure that you are obtaining the proper device to support your VoIP deployment, including add-ons such as DSP resources.

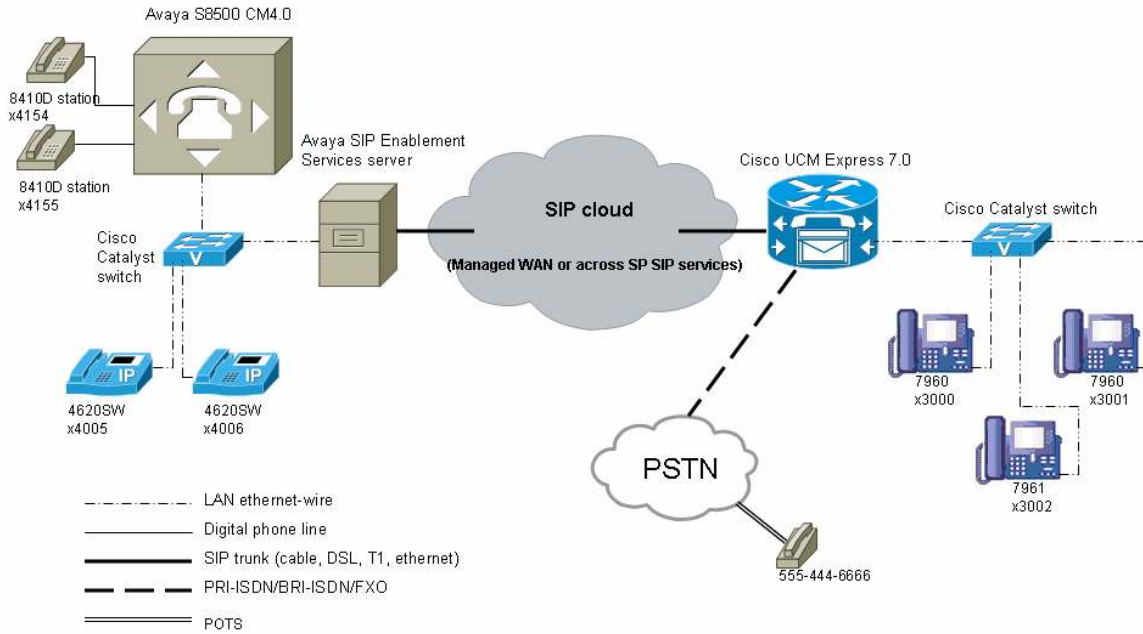
[Cisco 2800 series Integrated Services Router](#)

[Cisco 3800 series Integrated Services Router](#)

[Cisco AS5400 series Universal Gateways](#)

Network Topology

Figure 1. Basic Call Setup





Limitations

The following section lists known limitations, caveats, or integration issues.

- The Avaya 4600 series IP SIP phone drops all conference participants if the conference initiator is an Avaya 4600 phone user and drops from the conference. Later Avaya phones (9600 series) can perform the "conference remain" function, although this has not been tested by Cisco.
- Although some calling name/number and connected name/number updates are achieved during call transfers and call forwards, name/number updates are not fully interoperable because Cisco and Avaya use different CLID presentation forms. The CUCME uses SIP remote-party-ID headers to present calling and connected name/number updates, while Avaya uses SIP P-asserted-identity headers to present these updates
- A Cisco Unified IP phone registered to CUCME using the SIP protocol cannot be configured to use g711 codec with local CUCME phones and g729 across an SIP trunk. A Cisco Unified SIP phone will always set g711 as the priority codec, forcing g711 across the SIP trunk. The workaround is to set the Cisco Unified SIP phone to g729 codec only.
- A Cisco Unified IP phone registered to CUCME using an SIP protocol does not support "blind" transfer. It performs only full-attendant and semi-attendant transfer.
- CUCME does not honor caller ID restriction from Avaya due to the "From" header format that Avaya uses for caller ID restriction. Avaya includes the "anonymous" statement in the "From" header, but also includes the calling number causing the CUCME to present the calling number to the called party.
- Mid-call DTMF will not interoperate using in-band form. Only RFC2833 is supported for this integration.
- The Avaya SES server honors the SIP REFER message during a call transfer initiated by a Cisco Unified CME phone, allowing for the optimization of bandwidth for the SIP trunk. However, the Avaya SES server will not send SIP REFER during a call transfer initiated by an Avaya phone and will perform an RTP stream hairpin to connect the transferred call.
- The Avaya SES server honors the SIP "Moved-Temporarily" message during a call forward initiated by a CUCME phone, allowing for the optimization of bandwidth for the SIP trunk. However, the Avaya SES server will not send the SIP "Moved-Temporarily" message during a call forward initiated by an Avaya phone and will perform an RTP stream hairpin to connect the call forwarded call.
- Centralized voice mail integration, on either side, was not tested. SIP MWI is not interoperable.



System Components

Hardware Components

The following hardware was tested:

- Cisco 3845 Integrated Services Router (ISR)
- Cisco Unified IP phones 7900 series (C7960 skinny, C7961 SIP)
- Cisco Catalyst 3560 series switch
- Avaya S8500 Server
 - TN799 CLAN card
 - TN2312B IP server interface
 - TN2302 media processor card
- Avaya SIP Enablement Services server

Software Requirements

The following software is required:

- Cisco Unified Communications Manager Express: IOS 12.4(20)T with IPVOICE feature set
- Cisco Unified IP phone 7960: Skinny P00308000500 load
- Cisco Unified IP phone 7961: SIP SIP 41.8-3-3S load
- Avaya S8500: Communications Manager 4.0
- Avaya SIP Enablement Services server: SES04.0-04.0.033.6



Features

Features Supported

- Basic call with G729 across SIP trunk
- Calling Name and Calling Number presentation
- Calling Name and Calling Number restriction (see [Limitations](#) section for details)
- DTMF (RFC2833 only; see [Limitations](#) section for details)
- Toll avoidance and toll bypass
- Call Transfer using SIP REFER or using RTP stream hairpin (full-attendant, semi-attendant, and blind; see [Limitations](#) section for details)
- Call Forward using Moved-Temporarily or using RTP stream hairpin (all, busy, and no answer; see [Limitations](#) section for details)
- Hold/ Resume
- Three-way conference (see [Limitations](#) section for details)

Features Not Supported

- Centralized voice mail (MWI) over SIP trunk



Configuration

This section contains configuration menus and commands, and describes configuration sequences and tasks.

Configuring the Avaya S8500 Server

Software Version

```
list configuration software-versions

                                SOFTWARE VERSIONS

SOFTWARE VERSION
  Memory Resident: R014x.00.1.731.2
  Disk Resident: R014x.00.1.731.2

TRANSLATION DATE
  Memory Resident: 1:10 am WED AUG 27, 2008
  Disk Resident: 1:10 am WED AUG 27, 2008
  Disk Second Copy: good

Command successfully completed
Command: |
```



IP Server Interface

```
display ipserver-interface 1
IP SERVER INTERFACE (IPSI) ADMINISTRATION - PORT NETWORK 1

IP Control? y          Socket Encryption? y
Ignore Connectivity in Server Arbitration? n      Enable QoS? n

Primary IPSI
-----
Location: 1A01
Host: 172.20.212.2
DHCP ID: ipsi-A01a

Command:
```

Trunk Group

```
display trunk-group 1                                     Page 1 of 21
TRUNK GROUP

Group Number: 1          Group Type: sip          CDR Reports: y
Group Name: SIP TRUNK GROUP      COR: 1          TN: 1          TAC: 801
Direction: two-way          Outgoing Display? n
Dial Access? n          Night Service:
Queue Length: 0
Service Type: tie          Auth Code? n

                          Signaling Group: 1
                          Number of Members: 6
```



```
display trunk-group 1                                     Page 2 of 21
  Group Type: sip
TRUNK PARAMETERS
  Unicode Name? y
                                     Redirect On OPTIM Failure: 5000
  SCCAN? n                                     Digital Loss Group: 18
                                     Preferred Minimum Session Refresh Interval(sec): 600
```

```
display trunk-group 1                                     Page 3 of 21
TRUNK FEATURES
  ACA Assignment? n                                     Measured: none
                                                         Maintenance Tests? y
  Numbering Format: private
                                                         UUI Treatment: service-provider
                                                         Replace Unavailable Numbers? n
  Show ANSWERED BY on Display? n
```



display trunk-group 1 Page 4 of 21

PROTOCOL VARIATIONS

Mark Users as Phone? n
Prepend '+' to Calling Number? n
Send Transferring Party Information? n

Telephone Event Payload Type: 101

display trunk-group 1 Page 5 of 21

TRUNK GROUP

Administered Members (min/max): 1/6
Total Administered Members: 6

GROUP MEMBER ASSIGNMENTS

Port	Name
1: T00001	SIP TRUNK
2: T00002	SIP TRUNK
3: T00003	SIP TRUNK
4: T00004	SIP TRUNK
5: T00005	SIP TRUNK
6: T00006	SIP TRUNK
7:	
8:	
9:	
10:	
11:	
12:	
13:	
14:	
15:	



Signaling Group

```
display signaling-group 1
SIGNALING GROUP
Group Number: 1          Group Type: sip
                        Transport Method: tls

Near-end Node Name: clan1      Far-end Node Name: avayasip1
Near-end Listen Port: 5061     Far-end Listen Port: 5061
Far-end Network Region: 1
Far-end Domain:

Bypass If IP Threshold Exceeded? n
DTMF over IP: rtp-payload     Direct IP-IP Audio Connections? n
                               IP Audio Hairpinning? n
Enable Layer 3 Test? n
Session Establishment Timer(min): 120
Command:
```



Node Names IP

```
display node-names ip                                     Page 1 of 2
IP NODE NAMES
Name                IP Address
CCM4_1              172.20.231.254
CM-EUROPA           172.20.236.254
CM-NEPTUNE          172.20.236.2
CM-POLARIS          172.20.236.50
CUCMExpress         172.20.228.254
CecilyGW            172.20.174.40
TFTP                172.20.2.181
TonyBGW             172.20.8.26
avayasip1          172.20.212.254
clan1               172.20.212.253
clan1serverb       172.20.213.253
default             0.0.0.0
medpro1             172.20.212.252
nortelcs1000       172.20.216.100
procr               172.20.212.200

( 15 of 15 administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

Note: The name “avayasip1” is the SES IP address.

IP Network Region

```
display ip-network-region 1                             Page 1 of 19
IP NETWORK REGION
Region: 1
Location: 1      Authoritative Domain: lab.com
Name: CiscoLAB
MEDIA PARAMETERS
  Codec Set: 3      Intra-region IP-IP Direct Audio: yes
                   Inter-region IP-IP Direct Audio: no
  UDP Port Min: 2048  IP Audio Hairpinning? y
  UDP Port Max: 3029
DIFFSERV/TOS PARAMETERS
  Call Control PHB Value: 34  RTCP Reporting Enabled? y
  Audio PHB Value: 46        RTCP MONITOR SERVER PARAMETERS
  Video PHB Value: 26        Use Default Server Parameters? y
802.1P/Q PARAMETERS
  Call Control 802.1p Priority: 7
  Audio 802.1p Priority: 6
  Video 802.1p Priority: 5
H.323 IP ENDPOINTS
  H.323 Link Bounce Recovery? y
  Idle Traffic Interval (sec): 20
  Keep-Alive Interval (sec): 5
  Keep-Alive Count: 5
AUDIO RESOURCE RESERVATION PARAMETERS
  RSUP Enabled? n
```

Note: The Codec Set parameter determines what codec values are set to be used.



```
display ip-network-region 1 Page 2 of 19
IP NETWORK REGION

INTER-GATEWAY ALTERNATE ROUTING / DIAL PLAN TRANSPARENCY
Incoming LDN Extension:
Conversion To Full Public Number - Delete:      Insert:
Maximum Number of Trunks to Use for IGAR:
Dial Plan Transparency in Survivable Mode? n

BACKUP SERVERS(IN PRIORITY ORDER)      H.323 SECURITY PROFILES
1                                       1 challenge
2                                       2
3                                       3
4                                       4
5
6                                       Allow SIP URI Conversion? y

TCP SIGNALING LINK ESTABLISHMENT FOR AVAYA H.323 ENDPOINTS
Near End Establishes TCP Signaling Socket? y
Near End TCP Port Min: 61440
Near End TCP Port Max: 61444
```

IP Codec Set

```
display ip-codec-set 3 Page 1 of 2
IP Codec Set

Codec Set: 3

Audio      Silence      Frames      Packet
Codec      Suppression  Per Pkt     Size(ms)
1: G.729B  n           2           20
2: G.729   n           2           20
3: G.729AB n           2           20
4: G.729A  n           2           20
5: G.711MU n           2           20
6:
7:

Media Encryption
1: none
2:
3:
```



Uniform Dialing

```
display uniform-dialplan 3
```

Page 1 of 2

UNIFORM DIAL PLAN TABLE

Percent Full: 0

Matching Pattern	Len	Del	Insert Digits	Net	Conv	Node Num
30	4	0	224	aar	n	
3005	4	0	210	aar	n	
3006	4	0	210	aar	n	
3011	4	0	210	aar	n	
3012	4	0	210	aar	n	
3013	4	0	210	aar	n	
3014	4	0	210	aar	n	
3020	4	0	210	aar	n	
360	4	0	214	aar	n	
37	4	0	214	aar	n	
40	4	0	204	aar	n	
4050	4	0	211	aar	n	
4104	4	0	226	aar	n	
42	4	0	224	aar	n	
43	4	0	224	aar	n	
5000	4	0	204	aar	n	

AAR Analysis

```
display aar analysis 224
```

Page 1 of 2

AAR DIGIT ANALYSIS TABLE

Percent Full: 2

Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Reqd
224	7	7	99	aar		n
225	4	4	13	aar		n
226	7	7	13	aar		n
227	7	7	21	aar		n
228	7	7	44	aar		n
229	7	7	7	aar		n
3	7	7	999	aar		n
414357	10	10	10	aar		n
5	7	7	999	aar		n
5050	7	7	13	aar		n
5554050	7	7	11	aar		n
7	7	7	999	aar		n
8	7	7	999	aar		n



Route Pattern

```

display route-pattern 99                                     Page 1 of 3
Pattern Number: 99 Pattern Name: CCS Server A
SCCAN? n Secure SIP? n
Grp FRL NPA Pfx Hop Toll No. Inserted DCS/ IXC
No Mrk Lmt List Del Digits Intw
1: 1 0 3 n user
2: n user
3: n user
4: n user
5: n user
6: n user

BCC VALUE TSC CA-TSC ITC BCIE Service/Feature PARM No. Numbering LAR
0 1 2 M 4 W Request Dgts Format Subaddress
1: y y y y y n n rest none
2: y y y y y n n rest none
3: y y y y y n n rest none
4: y y y y y n n rest none
5: y y y y y n n rest none
6: y y y y y n n rest none

```

Customer options (IP)

```

display system-parameters customer-options                 Page 2 of 10
OPTIONAL FEATURES
IP PORT CAPACITIES USED
Maximum Administered H.323 Trunks: 800 13
Maximum Concurrently Registered IP Stations: 2400 5
Maximum Administered Remote Office Trunks: 800 0
Maximum Concurrently Registered Remote Office Stations: 2400 0
Maximum Concurrently Registered IP eCons: 0 0
Max Concur Registered Unauthenticated H.323 Stations: 0 0
Maximum Video Capable H.323 Stations: 0 0
Maximum Video Capable IP Softphones: 0 0
Maximum Administered SIP Trunks: 10 8
Maximum Number of DS1 Boards with Echo Cancellation: 80 0
Maximum TN2501 UAL Boards: 10 0
Maximum Media Gateway UAL Sources: 250 0
Maximum TN2602 Boards with 80 VoIP Channels: 128 0
Maximum TN2602 Boards with 320 VoIP Channels: 128 0
Maximum Number of Expanded Meet-me Conference Ports: 0 0
(NOTE: You must logoff & login to effect the permission changes.)

```



```
display system-parameters customer-options Page 4 of 10
OPTIONAL FEATURES

Emergency Access to Attendant? y IP Stations? y
  Enable 'dadmin' Login? y
  Enhanced Conferencing? y ISDN Feature Plus? y
  Enhanced EC500? y ISDN Network Call Redirection? n
Enterprise Survivable Server? n ISDN-BRI Trunks? y
  Enterprise Wide Licensing? n ISDN-PRI? y
  ESS Administration? n Local Survivable Processor? n
  Extended Cvg/Fwd Admin? y Malicious Call Trace? y
External Device Alarm Admin? y Media Encryption Over IP? y
Five Port Networks Max Per MCC? n Mode Code for Centralized Voice Mail? y
  Flexible Billing? n
Forced Entry of Account Codes? y Multifrequency Signaling? y
  Global Call Classification? y Multimedia Call Handling (Basic)? y
  Hospitality (Basic)? y Multimedia Call Handling (Enhanced)? y
Hospitality (G3V3 Enhancements)? y
  IP Trunks? y

IP Attendant Consoles? y
(NOTE: You must logoff & login to effect the permission changes.)
```



Configuring the Avaya SIP Enablement Services Server

Software Version

Help Exit

Software Version

The Software Version Web page displays the software version of the active server.

```
Operating System: Linux 2.6.11-AV18 i686 i686
SES Release String: SES-4.0.0.0-033.6
Software Load: SES04.0-04.0.033.6
Server BIOS Build ID: PLJH61AUS
RSA Version ID: PLEH08B
```

Help

- Alarms
 - Current Alarms
 - SNMP Traps
- Diagnostics
 - System Logs
 - Temperature/Voltage
 - Ping
 - Traceroute
 - Netstat
 - Modem Test
- Server
 - Status Summary
 - Process Status
 - Shutdown Server
 - Server Date/Time
 - Software Version
- Server Configuration
 - Configure Server
 - Eject CD-ROM
- Server Upgrades
 - Manage Software
 - Make Upgrade Permanent
 - Boot Partition
- Data Backup/Restore
 - Backup Now
 - Backup History
 - Schedule Backup
 - Backup Logs
 - View/Restore Data
 - Restore History
 - Format PC Card
- Security
 - Modem
 - FTP
 - Authentication File
 - Firewall
 - WebLM Software
 - WebLM License Admin
 - Tripwire
 - Tripwire Commands
 - Trusted Certificates
 - SSH Keys
- Miscellaneous
 - Download Files



List Users (SIP Phones)

Help Exit

Top

- Users
 - List
 - Add
 - Search
 - Edit
 - Delete
 - Password
 - Default Profile
 - Registered Users
- Conferences
- Media Server Extensions
 - Emergency Contacts
- Hosts
- Media Servers
 - Address Map Priorities
- Adjunct Systems
- Trusted Hosts
- Services
- Server Configuration
- Certificate Management
 - IM logs
- Trace Logger
- Export/Import to ProVision

List Users

Showing users 1 to 10 out of 10 users.

User ID	Host	Name
<input type="checkbox"/> 4001	172.20.212.254	Cisco1 Lab1
<input type="checkbox"/> 4002	172.20.212.254	4002 4002
<input type="checkbox"/> 4003	172.20.212.254	Cisco3 LAB1
<input type="checkbox"/> 4004	172.20.212.254	Cisco4 LAB1
<input checked="" type="checkbox"/> 4005	172.20.212.254	cisco lab5
<input checked="" type="checkbox"/> 4006	172.20.212.254	Cisco Remote_1
<input type="checkbox"/> 4007	172.20.212.254	Joe SIP
<input type="checkbox"/> 4008	172.20.212.254	Tony SIP
<input type="checkbox"/> 4009	172.20.212.254	cisco lab4009
<input type="checkbox"/> 4010	172.20.212.254	home cisco

Task:



List Host (SIP Trunk End Points)

Help Exit

Top

- Users
- Conferences
- Media Server Extensions
 - Emergency Contacts
- Hosts
 - List
 - Migrate Home/Edge
- Media Servers
 - Address Map Priorities
- Adjunct Systems
- Trusted Hosts
 - Services
- Server Configuration
- Certificate Management
 - IM logs
- Trace Logger
- Export/Import to ProVision

List Hosts

Status	Commands	Host	Type
up to date	Edit Map Go-To Test-Link Delete	172.20.212.254	home/edge

[Force All](#)
[Migrate Home/Edge](#)



List Host Map

Help Exit

Top

- Users
- Conferences
- Media Server Extensions
 - Emergency Contacts
- Hosts
 - List
 - Migrate Home/Edge
- Media Servers
 - Address Map Priorities
- Adjunct Systems
- Trusted Hosts
 - Services
- Server Configuration
- Certificate Management
 - IM logs
- Trace Logger
- Export/Import to ProVision

List Host Address Map

Host 172.20.212.254

Commands	Name	Commands	Contact
Edit Delete	IP_IP_GW_233		
Edit Delete	IP_IP_GW_5000		
Edit Delete	IP_IP_GW_5001		
Edit Delete	IP_IP_GW_5010		
		Edit Delete	sip:\$(user)@172.20.174.40:5060;transport=udp
Add Another Map		Add Another Contact	Delete Group
Edit Delete	TonyB_CME		
Edit Delete	TonyB_CME_2		
		Edit Delete	sip:\$(user)@172.20.8.26:5060;transport=udp
Add Another Map		Add Another Contact	Delete Group
Edit Delete	CM-Polaris		
Edit Delete	CM-Polaris_615X		
Edit Delete	CM-Polaris_715X		
		Edit Delete	sip:\$(user)@172.20.236.50:5060;transport=tcp
Add Another Map		Add Another Contact	Delete Group
Edit Delete	CCM-Mercury		
		Edit Delete	sip:\$(user)@172.20.215.254:5060;transport=tcp
Add Another Map		Add Another Contact	Delete Group
Edit Delete	CCMS.0-Venus		
		Edit Delete	sip:\$(user)@172.20.214.254:5060;transport=tcp
Add Another Map		Add Another Contact	Delete Group

Help Exit

Top

- Users
- Conferences
- Media Server Extensions
- Emergency Contacts
- Hosts
 - List
 - Migrate Home/Edge
- Media Servers
 - Address Map Priorities
- Adjunct Systems
- Trusted Hosts
- Services
- Server Configuration
- Certificate Management
 - IM logs
- Trace Logger
- Export/Import to Provision

Edit Host Map Entry

Name*

Pattern*

Replace URI

Fields marked * are required.

Update



Media Server

Help Exit

Top

- Users
- Conferences
- Media Server Extensions
 - Emergency Contacts
- Hosts
- Media Servers
 - List
 - Add
 - Address Map Priorities
- Adjunct Systems
- Trusted Hosts
 - Services
- Server Configuration
- Certificate Management
 - IM logs
- Trace Logger
- Export/Import to ProVision

List Media Servers

Commands		Interface	Host			
Edit	Extensions	Map	Test-Link	Delete	172.20.212.253	172.20.212.254

[Add Another Media Server Interface](#)



Help Exit

Top

- Users
- Conferences
- Media Server Extensions
- Emergency Contacts
- Hosts
- Media Servers
 - List
 - Add
 - Address Map Priorities
- Adjunct Systems
- Trusted Hosts
- Services
- Server Configuration
- Certificate Management
- IM logs
- Trace Logger
- Export/Import to ProVision

Edit Media Server Interface

Media Server Interface Name*

Host

SIP Trunk

SIP Trunk Link Type TCP TLS

SIP Trunk IP Address*

Media Server

Media Server Admin Address (see Help)

Media Server Admin Login

Media Server Admin Password

Media Server Admin Password Confirm

SMS Connection Type

SSH Telnet

Fields marked * are required.



Configuring the Cisco Unified Communications Manager Express (Delete if Toll Bypass)

show version

Router#sh ver

Cisco IOS Software, 3800 Software (C3845-IPVOICEK9-M), Version 12.4(20)T, RELEAS

E SOFTWARE (fc3)

Technical Support: <http://www.cisco.com/techsupport>

Copyright (c) 1986-2008 by Cisco Systems, Inc.

Compiled Fri 11-Jul-08 02:28 by prod_rel_team

ROM: System Bootstrap, Version 12.3(11r)T2, RELEASE SOFTWARE (fc1)

Router uptime is 1 day, 4 hours, 44 minutes

System returned to ROM by power-on

System restarted at 18:55:28 UTC Tue Aug 26 2008

System image file is "flash:c3845-ipvoicek9-mz.124-20.T.bin"

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

A summary of U.S. laws governing Cisco cryptographic products may be found at:

<http://www.cisco.com/wwl/export/crypto/tool/stqrg.html>

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

Cisco 3845 (revision 1.0) with 483328K/40960K bytes of memory.

Processor board ID FTX0933A1JA



2 Gigabit Ethernet interfaces

5 Serial interfaces

2 Channelized/Clear T1/PRI ports

2 Channelized (E1 or T1)/PRI ports

1 802.11 Radio

DRAM configuration is 64 bits wide with parity enabled.

479K bytes of NVRAM.

125184K bytes of ATA System CompactFlash (Read/Write)

Configuration register is 0x2102



show running-config

Router#sh run

Building configuration...

Current configuration : 5333 bytes

!

! Last configuration change at 17:14:16 UTC Wed Aug 27 2008

! NVRAM config last updated at 17:14:17 UTC Wed Aug 27 2008

!

version 12.4

service timestamps debug datetime msec

service timestamps log datetime msec

no service password-encryption

!

hostname Router

!

boot-start-marker

boot-end-marker

!

!

card type command needed for slot 3

logging message-counter syslog

logging buffered 10000000

no logging console

enable password cisco

!

no aaa new-model

clock calendar-valid

network-clock-participate slot 3¹

network-clock-participate wic 0

network-clock-select 1 T1 0/0/0²

!

¹ The NM-HDV2 module must be part of the network-clock scheme in order to share (farm) its DSP resources. These resources are used for the transcoding application in this example.

² The T1 card providing PSTN access should be part of the network-clock scheme to avoid T1 clock slips.



```
!  
ip source-route  
ip cef  
!  
!  
!  
!  
multilink bundle-name authenticated  
!  
!  
isdn switch-type primary-ni2  
!  
voice-card 0  
no dspfarm  
!  
voice-card 33  
dspfarm  
dsp services dspfarm  
!  
!  
!  
voice service pots  
!  
voice service voip  
no notify redirect ip2ip  
allow-connections sip to sip4  
h323  
sip  
min-se 1200  
registrar server5  
g729 annexb-all6
```

³ This command enables the NM-HDV2 module to farm (share) DSP resources. In this config example, the DSP is used for the transcoding application.

⁴ Enables the SIP B2BUA feature on CUBE.

⁵ Enables the SIP registrar service for SIP end-points to register.



```
!  
!  
voice class codec 17  
  codec preference 1 g729br8 bytes 20  
  codec preference 2 g729r8 bytes 20  
  codec preference 3 g711ulaw bytes 160  
!  
!  
!  
!  
!  
!  
!  
!  
!  
!  
!  
!  
voice register global8  
  mode cme  
  source-address 172.20.8.26 port 5060  
  max-dn 100  
  max-pool 192  
  load 7961 SIP41.8-3-3S  
  tftp-path flash:  
  create profile sync 0070911722917364  
!  
voice register dn 19  
  number 3002  
  name H. Moreno  
  huntstop  
!  
!
```

⁶ Enables CUBE to accept all G729 codec flavors (annexB, annexA).

⁷ This command provides codec selection and codec filtering.

⁸ SIP global settings for SIP phone registration.

⁹ SIP phone directory number settings.



voice register template 1

!

voice register pool 1¹⁰

id mac 0015.2B8F.351B

type 7961

number 1 dn 1

template 1

dtmf-relay rtp-nte

no call-waiting

description Moreno

!

!

!

!

!

!

!

archive

log config

hidekeys

!

!

controller T1 0/0/0¹¹

framing esf

clock source line primary

linecode b8zs

pri-group timeslots 1-4,24

!

controller T1 0/0/1

framing esf

linecode b8zs

!

!

¹⁰ SIP phone registration settings.

¹¹ Configuration of T1 providing PRI PSTN access.



```
!  
!  
!  
interface GigabitEthernet0/0  
ip address 172.20.8.26 255.255.255.0  
duplex auto  
speed auto  
media-type rj45  
!  
interface GigabitEthernet0/1  
no ip address  
shutdown  
duplex auto  
speed auto  
media-type rj45  
!  
interface Serial0/0/0:2312  
no ip address  
encapsulation hdlc  
isdn switch-type primary-ni2  
isdn overlap-receiving  
isdn incoming-voice voice  
no cdp enable  
!  
!  
ip forward-protocol nd  
ip route 172.20.0.0 255.255.0.0 172.20.8.1  
!  
no ip http server  
no ip http secure-server  
!  
!  
!  
!
```

¹² T1 interface settings (PSTN access).



```
fttp-server flash:P00308000500.bin13
fttp-server flash:P00308000500.sb2
fttp-server flash:P00308000500.loads
fttp-server flash:P00308000500.sbn
fttp-server flash:cnu41.8-3-2-27.sbn
fttp-server flash:cvm41sip.8-3-2-27.sbn
fttp-server flash:dsp41.8-3-2-27.sbn
fttp-server flash:jar41sip.8-3-2-27.sbn
fttp-server flash:SIP41.8-3-3S.loads
fttp-server flash:term61.default.loads
fttp-server flash:apps41.8-3-2-27.sbn
!
control-plane
!
!
!
voice-port 0/0/0:23
!
!
!
sccp local GigabitEthernet0/014
sccp ccm 172.20.8.26 identifier 1
sccp
!
sccp ccm group 1
  associate ccm 1 priority 1
  associate profile 1 register mtp00146a7299e0
!
dspfarm profile 1 transcode
  codec g711ulaw
  codec g711alaw
  codec g729ar8
  codec g729abr8
```

¹³ Recommended skinny and SIP phone loads for CUCME 4.3/7.0.

¹⁴ Transcoder settings to register to CUCME for conference calls between g711 and g729 end-points.



```
codec g729r8
codec g729br8
maximum sessions 5
associate application SCCP
!
!
dial-peer voice 4100 voip15
description SIP to AvayaS8500
destination-pattern 4...
voice-class codec 1
voice-class sip g729 annexb-all
session protocol sipv2
session target ipv4:172.20.212.254
dtmf-relay rtp-nte
!
dial-peer voice 3000 voip16
description incoming dial-peer
voice-class codec 1
voice-class sip g729 annexb-all
session protocol sipv2
incoming called-number 3...
dtmf-relay rtp-nte
!
dial-peer voice 2000 pots17
destination-pattern 2...
direct-inward-dial
port 0/0/0:23
forward-digits all
!
!
!
sip-ua
```

¹⁵ Outgoing dial-peer to Avaya PBX.

¹⁶ Incoming dial-peer from Avaya PBX.

¹⁷ Outgoing dial-peer to PSTN.



```
!  
!  
telephony-service18  
sdsfarm units 1  
sdsfarm transcode sessions 10  
sdsfarm tag 1 mtp00146a7299e0  
load 7960-7940 P00308000500  
max-ephones 96  
max-dn 192  
ip source-address 172.20.8.26 port 2000  
system message CME InterOp  
max-conferences 8 gain -6  
call-forward pattern ....  
moh music-on-hold.au  
dn-webedit  
time-webedit  
transfer-system full-consult  
transfer-pattern ....  
secondary-dialtone 9  
create cnf-files version-stamp 7960 Aug 26 2008 18:56:36  
!  
!  
ephone-dn 1 dual-line  
call-waiting ring  
number 3000  
label 3000  
description Carlos Vela  
name Carlos Vela  
!  
!  
ephone-dn 2 dual-line  
number 3001  
label 3001  
description Salcido
```

¹⁸ CUCME and skinny phone settings.



```
name C. Salcido
!
!
ephone 1
mac-address 000A.416B.8539
type 7960
keep-conference endcall
button 1:1
!
!
!
ephone 2
mac-address 000F.9069.DB2C
type 7960
keep-conference endcall
button 1:2
!
!
!
line con 0
password cisco
login
line aux 0
line vty 0 4
exec-timeout 0 0
password cisco
login
!
scheduler allocate 20000 1000
end
```



Acronyms

Acronym

CM

CUCME

DSP

DTMF

ISR

MWI

RTP

SIP

Definitions

Communications Manager

Cisco Unified CallManager Express

digital signal processor

dual-tone multi-frequency

Integrated Services Routers

Message Waiting Indication

Real-Time Protocol

Session Initiation Protocol



Important Information

THE SPECIFICATIONS AND INFORMATION REGARDING THE PRODUCTS IN THIS MANUAL ARE SUBJECT TO CHANGE WITHOUT NOTICE. ALL STATEMENTS, INFORMATION, AND RECOMMENDATIONS IN THIS MANUAL ARE BELIEVED TO BE ACCURATE BUT ARE PRESENTED WITHOUT WARRANTY OF ANY KIND, EXPRESS OR IMPLIED. USERS MUST TAKE FULL RESPONSIBILITY FOR THEIR APPLICATION OF ANY PRODUCTS.

IN NO EVENT SHALL CISCO OR ITS SUPPLIERS BE LIABLE FOR ANY INDIRECT, SPECIAL, CONSEQUENTIAL, OR INCIDENTAL DAMAGES, INCLUDING, WITHOUT LIMITATION, LOST PROFITS OR LOSS OR DAMAGE TO DATA ARISING OUT OF THE USE OR INABILITY TO USE THIS MANUAL, EVEN IF CISCO OR ITS SUPPLIERS HAVE BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGES.



Corporate Headquarters

Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-4000
800 553-NETS (6387)
Fax: 408 526-4100

European Headquarters

Cisco Systems International
BV
Haarlerbergpark
Haarlerbergweg 13-19
1101 CH Amsterdam
The Netherlands
www-europe.cisco.com
Tel: 31 0 20 357 1000
Fax: 31 0 20 357 1100

Americas Headquarters

Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-7660
Fax: 408 527-0883

Asia Pacific Headquarters

Cisco Systems, Inc.
Capital Tower
168 Robinson Road
#22-01 to #29-01
Singapore 068912
www.cisco.com
Tel: +65 317 7777
Fax: +65 317 7799

Cisco Systems has more than 200 offices in the following countries and regions. Addresses, phone numbers, and fax numbers are listed on the Cisco Web site at www.cisco.com/go/offices.

Argentina • Australia • Austria • Belgium • Brazil • Bulgaria • Canada • Chile • China PRC • Colombia • Costa Rica • Croatia • Czech Republic • Denmark • Dubai, UAE • Finland • France • Germany • Greece • Hong Kong SAR • Hungary • India • Indonesia • Ireland • Israel • Italy • Japan • Korea • Luxembourg • Malaysia • Mexico • The Netherlands • New Zealand • Norway • Peru • Philippines • Poland • Portugal • Puerto Rico • Romania • Russia • Saudi Arabia • Scotland • Singapore • Slovakia • Slovenia • South Africa • Spain • Sweden • Switzerland • Taiwan • Thailand • Turkey • Ukraine • United Kingdom • United States • Venezuela • Vietnam • Zimbabwe

© 2007 Cisco Systems, Inc. All rights reserved.

CCVP, the Cisco Logo, and the Cisco Square Bridge logo are trademarks of Cisco Systems, Inc.; Changing the Way We Work, Live, Play, and Learn is a service mark of Cisco Systems, Inc.; and Access Registrar, Aironet, BPX, Catalyst, CCDA, CCDP, CCIE, CCIP, CCNA, CCNP, CCSP, Cisco, the Cisco Certified Internetwork Expert logo, Cisco IOS, Cisco Press, Cisco Systems, Cisco Systems Capital, the Cisco Systems logo, Cisco Unity, Enterprise/Solver, EtherChannel, EtherFast, EtherSwitch, Fast Step, Follow Me Browsing, FormShare, GigaDrive, GigaStack, HomeLink, Internet Quotient, IOS, iPhone, IP/TV, iQ Expertise, the iQ logo, iQ Net Readiness Scorecard, iQuick Study, LightStream, Linksys, MeetingPlace, MGX, Networking Academy, Network Registrar, *Packet*, PIX, ProConnect, RateMUX, ScriptShare, SlideCast, SMARTnet, StackWise, The Fastest Way to Increase Your Internet Quotient, and TransPath are registered trademarks of Cisco Systems, Inc. and/or its affiliates in the United States and certain other countries.

All other trademarks mentioned in this document or Website are the property of their respective owners. The use of the word partner does not imply a partnership relationship between Cisco and any other company. (0612R)

Printed in the USA