



# Avaya S8500 Release CM 4.0 Using SIP to Cisco Unified Communications Manager Express 7.1

June 25, 2009

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## Introduction

- This application note demonstrates connectivity between an Avaya S8500 Communications Manager Release CM4.0 and Cisco Unified Communications Manager Express (CUCME) Release 7.1 (IOS 12.4[22]YB1) using a Cisco 3825 ISR with SIP protocol.
- Voice mail testing was not performed because Avaya does not support SIP MWI messaging across external trunks. Furthermore, CUCME does not relay SIP MWI to endpoints not registered to it. Also, differences in the way Avaya and Cisco pass call diversion information over SIP trunks prevents voicemail platforms integrated to either CUCME and/or Avaya PBX to support centralized voicemail services.
- The network topology diagram (Figure 1) shows the test setup for end-to-end interoperability with CUCME Release 7.1 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. On-board DSP resources (PVDM's) were 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.
- Testing was performed using Avaya 8400-series 2-wire digital phones, as well as 9600-series IP Phones configured for H.323 protocol. No Avaya SIP phones were tested, as the test setup did not include Avaya's SIP Enablement Services server.
- This application note uses the 3825 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.

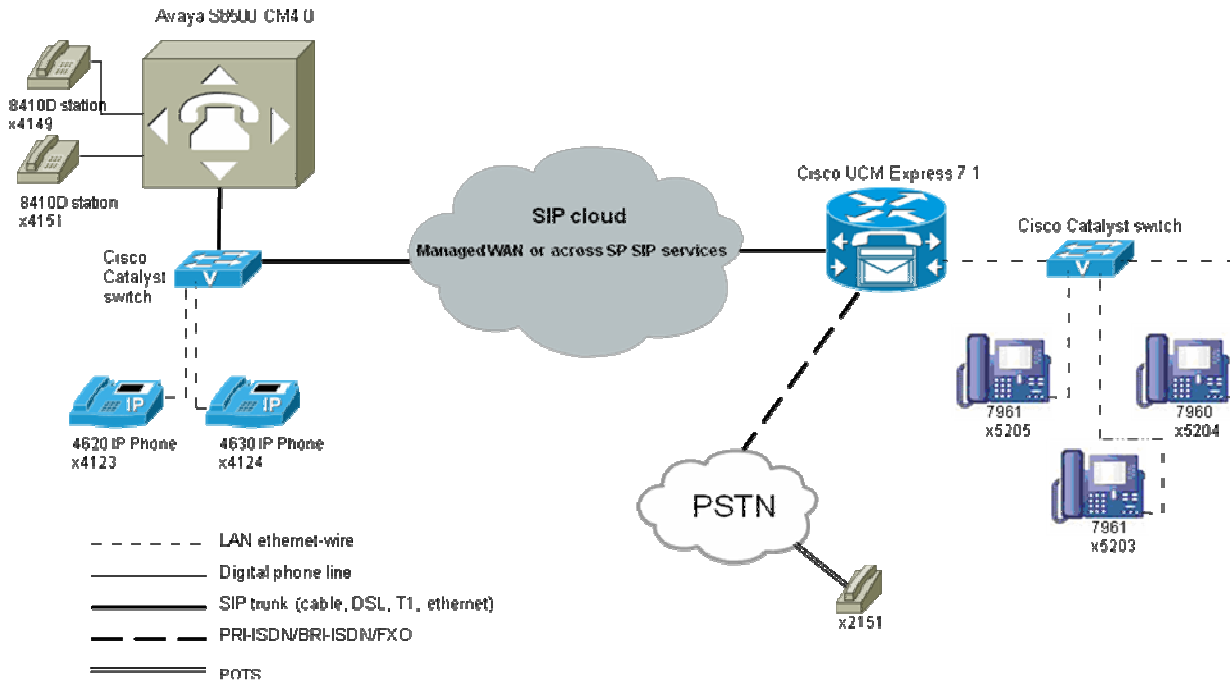
[Cisco 2800 series Integrated Services Router](#)

[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.

- 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 fully-attended and semi-attended 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 S8500 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 S8500 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 S8500 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 S8500 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 forwarded call.
- Centralized voicemail integration, on either side, is not supported. Issues with passing SIP MWI across SIP trunks, and differences in passing call diversion information do not allow for centralized voicemail services.
- Interoperability testing for this application note was conducted with Avaya's Media Shuffling enabled. Media Shuffling is a mechanism whereby media streams between endpoints are routed directly between the endpoints if possible, so as to optimize utilization of DSP resources. With Media Shuffling enabled, call conferences initiated by SIP phones registered on CUCME fail to properly reconnect the conference participants after the SIP phone drops off the conference call.



## **System Components**

### **Hardware Components**

The following hardware was tested:

- Cisco 3825 Integrated Services Router (ISR)
- Cisco Unified IP phones 7900 series (C7961 SCCP, C7961 SIP)
- Cisco Catalyst 6500 series switch
- Avaya S8500 Server
  - TN799 CLAN card
  - TN2312B IPSI
  - TN2302 Media Processor card
- Avaya 8410D digital phones
- Avaya 9600 series IP phones (H.323)

### **Software Requirements**

The following software is required:

- Cisco Unified Communications Manager Express: IOS 12.4(22)YB1 with IPVOICE feature set
- Cisco Unified IP phone 7961: SCCP41.8-4-2S load
- Cisco Unified IP phone 7961: SIP SIP 41.8-4-2S load
- Avaya S8500: Communications Manager 4.0.4.739.0
- TN799 CLAN firmware 26
- TN2302AP Media Processor firmware 118
- TN2312BP IPSI firmware 40
- Avaya 9600-series IP phone H.323 firmware version 3.0 (ha96xxua3\_00)



## 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 voicemail service over SIP trunks



## 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.00.4.739.0
  Disk Resident: R014x.00.4.739.0

TRANSLATION DATE
  Memory Resident: 1:10 am WED JUN 10, 2009
  Disk Resident: 1:10 am WED JUN 10, 2009
  Disk Second Copy: good
```

#### Circuit Packs Firmware Version

```
list configuration carrier |a Page 1

                                SYSTEM CONFIGURATION

Board Number   Board Type           Code      Vintage      Assigned Ports
                                u=unassigned t=tti p=psa

01A00          POWER SUPPLY         655A
01A01          IP SERVER INTFC      TN2312BP  HW06 FW040  01 02 03 04 05 06 07 08
01A02          CONTROL-LAN          TN799DP   HW01 FW026  u  u  u  u  u  u  u  u
                                17
01A03          IP MEDIA PROCESSOR   TN2302AP  HW20 FW118  01      03      05      07
01A04          DIGITAL LINE         TN2224B   000012     01 02 03 04 05 06 07 08
                                09 10 11 12 13 u  u  u
                                u  u  u  u  u  u  u  u
01A07          ANALOG LINE          TN746B    000002     01 02 03 04 05 06 07 u
                                u  u  u  u  u  u  u  u
```



Trunk Group

```
display trunk-group 5                                     Page 1 of 21
TRUNK GROUP
Group Number: 5          Group Type: sip          CDR Reports: y
Group Name: SIP Direct to CME Bench 8  CDR: 1      TN: 1      TAC: 805
Direction: two-way      Outgoing Display? n
Dial Access? n          Night Service:
Queue Length: 0
Service Type: tie       Auth Code? n
                               Signaling Group: 5
                               Number of Members: 4
```

```
display trunk-group 5                                     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 5 Page 3 of 21
TRUNK FEATURES
  ACA Assignment? n          Measured: none
                             Maintenance Tests? y

  Numbering Format: public   UUI Treatment: service-provider

                             Replace Unavailable Numbers? n

Show ANSWERED BY on Display? n
```

```
display trunk-group 5 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 5                                     Page 5 of 21
TRUNK GROUP
Administered Members (min/max): 1/4
Total Administered Members: 4

GROUP MEMBER ASSIGNMENTS

  Port      Name
  ---      -
1: T00009   SIP Direct
2: T00010   SIP Direct
3: T00005   SIP Direct
4: T00006   SIP Direct
5:
6:
7:
8:
9:
10:
11:
12:
13:
14:
15:
```

Signaling Group

```
display signaling-group 5
SIGNALING GROUP

Group Number: 5          Group Type: sip
                        Transport Method: tcp

Near-end Node Name: clan1      Far-end Node Name: CME_bench_8
Near-end Listen Port: 5060     Far-end Listen Port: 5060
Far-end Network Region: 1
Far-end Domain:

Bypass If IP Threshold Exceeded? n

DTMF over IP: rtp-payload     Direct IP-IP Audio Connections? y
                               IP Audio Hairpinning? n

Enable Layer 3 Test? n
Session Establishment Timer(min): 3
```



Locations Form

```
display locations Page 1 of 16
LOCATIONS
ARS Prefix 1 Required For 10-Digit NANP Calls? y
Loc No Name Timezone Rule NPA ARS Atd Disp Prefix Proxy Sel
      Offset
1: Main + 00:00 0 9 0 1 5
2:      :
3:      :
4:      :
5:      :
6:      :
7:      :
8:      :
9:      :
10:     :
11:     :
12:     :
13:     :
14:     :
```

**Note:** Parameter “Proxy Sel Rte Pat” must contain the Route Pattern used by the SIP trunk.



## Node Names IP

```
display node-names ip
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
CME_bench_8         172.20.236.249
CUCMEexpress        172.20.228.254
CecilyGW            172.20.174.20
Nortel_CS104        172.30.11.100
TACIPIPGW           172.16.243.218
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
( 16 of 17 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 “CME\_bench\_8” is the node name assigned to CUCME.

## IP Network Region

```
display ip-network-region 1
IP NETWORK REGION
Region: 1
Location: 1      Authoritative Domain: lab.com
Name: CiscoLAB
MEDIA PARAMETERS
  Codec Set: 3
  UDP Port Min: 2048
  UDP Port Max: 3029
  Intra-region IP-IP Direct Audio: yes
  Inter-region IP-IP Direct Audio: yes
  IP Audio Hairpinning? y
DIFFSERV/TOS PARAMETERS
  Call Control PHB Value: 34
  Audio PHB Value: 46
  Video PHB Value: 26
  RTCP Reporting Enabled? y
  RTCP MONITOR SERVER PARAMETERS
  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
  AUDIO RESOURCE RESERVATION PARAMETERS
  RSVP Enabled? n
H.323 IP ENDPOINTS
  H.323 Link Bounce Recovery? y
  Idle Traffic Interval (sec): 20
  Keep-Alive Interval (sec): 5
  Keep-Alive Count: 5
```

**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.729AB n           2           20
3: G.729A  n           2           20
4: G.711MU n           2           20
5:
6:
7:

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



Uniform Dialing

```
display uniform-dialplan 5 Page 1 of 2
```

UNIFORM DIAL PLAN TABLE Percent Full: 0

Matching Pattern	Len	Del	Insert Digits	Net	Conv	Node Num
5000	4	0	204	aar	n	
5001	4	0	201	aar	n	
5002	4	0	201	aar	n	
5003	4	0	204	aar	n	
5004	4	0	204	aar	n	
5005	4	0	204	aar	n	
5006	4	0	211	aar	n	
5010	4	0	224	aar	n	
5012	4	0	211	aar	n	
5015	4	0	211	aar	n	
5050	4	0	213	aar	n	
5051	4	0	226	aar	n	
5099	4	0	224	aar	n	
51	4	0	204	aar	n	
52	4	0	205	aar	n	
53	4	0	207	aar	n	

AAR Analysis

```
display aar analysis 205 Page 1 of 2
```

AAR DIGIT ANALYSIS TABLE Percent Full: 2

Dialled String	Total		Route Pattern	Call Type	Node Num	ANI Req'd
	Min	Max				
205	7	7	5	aar		n
207	7	7	99	aar		n
208	7	7	2	aar		n
210	7	7	10	aar	2	n
211	7	7	11	aar		n
213	7	7	13	aar		n
214	7	7	14	aar	3	n
216	7	7	16	aar		n
217	7	7	17	aar		n
221	7	7	11	aar		n
222	7	7	21	aar		n
224	7	7	14	aar		n
225	4	4	13	aar		n
226	7	7	13	aar		n
227	7	7	21	aar		n



Route Pattern

```

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

```



System-Parameters IP Options

```
display system-parameters ip-options Page 1 of 3
IP-OPTIONS SYSTEM PARAMETERS

IP MEDIA PACKET PERFORMANCE THRESHOLDS
  Roundtrip Propagation Delay (ms)   High: 800   Low: 400
  Packet Loss (%)                   High: 40    Low: 15
  Ping Test Interval (sec): 20
  Number of Pings Per Measurement Interval: 10

RTCP MONITOR SERVER
  Default Server IP Address: . . .
  Default Server Port: 5005
  Default RTCP Report Period(secs): 5

AUTOMATIC TRACE ROUTE ON
  Link Failure? y

H.248 MEDIA GATEWAY
  Link Loss Delay Timer (min): 5

H.323 IP ENDPOINT
  Link Loss Delay Timer (min): 5
  Primary Search Time (sec): 75
  Periodic Registration Timer (min): 20
```

```
display system-parameters ip-options Page 2 of 3
IP-OPTIONS SYSTEM PARAMETERS

Always use G.711 (30ms, no SS) for intra-switch Music-On-Hold? n

IP DTMF TRANSMISSION MODE
  Intra-System IP DTMF Transmission Mode: rtp-payload
  Inter-System IP DTMF: See Signaling Group Forms

HYPERACTIVE MEDIA GATEWAY REGISTRATIONS
  Enable Detection and Alarms? n
```



Digital Station

```
display station 4149                                     Page 1 of 5
STATION
Extension: 4149                                         Lock Messages? n      BCC: 0
Type: 84100                                           Security Code: *      TN: 1
Port: 01A0407                                         Coverage Path 1:      COR: 1
Name: SIP1 - 4149                                     Coverage Path 2:      COS: 1
                                                       Hunt-to Station:

STATION OPTIONS
    Loss Group: 2
    Data Module? n
    Speakerphone: 2-way
    Display Language: english

    Time of Day Lock Table:
    Personalized Ringing Pattern: 1
    Message Lamp Ext: 4149
    Mute Button Enabled? y

    Survivable COR: internal
    Survivable Trunk Dest? y

    Media Complex Ext:
    IP SoftPhone? y
    Remote Office Phone? n
    IP Video Softphone? n
```

```
display station 4149                                     Page 2 of 5
STATION
FEATURE OPTIONS
    LWC Reception: spe
    LWC Activation? y
    LWC Log External Calls? n
    CDR Privacy? n
    Redirect Notification? y
    Per Button Ring Control? n
    Bridged Call Alerting? n
    Active Station Ringing: single

    H.320 Conversion? n
    Service Link Mode: as-needed
    Multimedia Mode: enhanced
    MWI Served User Type:
    AUDIX Name:

    Auto Select Any Idle Appearance? n
    Coverage Msg Retrieval? y
    Auto Answer: none
    Data Restriction? n
    Idle Appearance Preference? n
    Bridged Idle Line Preference? n
    Restrict Last Appearance? y

    Per Station CPN - Send Calling Number?

    Audible Message Waiting? n
    Display Client Redirection? n
    Select Last Used Appearance? n
    Coverage After Forwarding? n
    Multimedia Early Answer? n
    Direct IP-IP Audio Connections? y
    Always Use? n IP Audio Hairpinning? y
    Remote Softphone Emergency Calls: as-on-local
    Emergency Location Ext: 4149
```



IP Station (H.323)

```
display station 4123                                     Page 1 of 5
STATION
Extension: 4123                                         Lock Messages? n      BCC: 0
Type: 9620                                             Security Code: *      TN: 1
Port: 500028                                          Coverage Path 1:      COR: 1
Name: 9620 test phone                                Coverage Path 2:      COS: 1
                                                       Hunt-to Station:
STATION OPTIONS
Loss Group: 19                                         Time of Day Lock Table:
                                                       Personalized Ringing Pattern: 1
                                                       Message Lamp Ext: 4123
                                                       Mute Button Enabled? y
Speakerphone: 2-way
Display Language: english
Survivable GK Node Name:
Survivable COR: internal                               Media Complex Ext:
Survivable Trunk Dest? y                               IP SoftPhone? n
                                                       Customizable Labels? y
```

```
display station 4123                                     Page 2 of 5
STATION
FEATURE OPTIONS
LWC Reception: spe                                     Auto Select Any Idle Appearance? n
LWC Activation? y                                     Coverage Msg Retrieval? y
LWC Log External Calls? n                             Auto Answer: none
COR Privacy? n                                       Data Restriction? n
Redirect Notification? y                               Idle Appearance Preference? n
Per Button Ring Control? n                           Bridged Idle Line Preference? n
Bridged Call Alerting? n                             Restrict Last Appearance? y
Active Station Ringing: single
                                                       EMU Login Allowed? n
H.320 Conversion? n                                  Per Station CPN - Send Calling Number?
Service Link Mode: as-needed
Multimedia Mode: enhanced
MWI Served User Type:
AUDIX Name:
                                                       Audible Message Waiting? n
                                                       Display Client Redirection? n
                                                       Select Last Used Appearance? n
                                                       Coverage After Forwarding? s
                                                       Multimedia Early Answer? n
                                                       Direct IP-IP Audio Connections? y
Emergency Location Ext: 4123                          Always Use? n IP Audio Hairpinning? y
```

### Configuring the Cisco Unified Communications Manager Express

show version



3825\_ISR#sho ver

Cisco IOS Software, 3800 Software (C3825-IPVOICEK9-M), Version 12.4(22)YB1, RELE

ASE SOFTWARE (fc1)

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

Copyright (c) 1986-2009 by Cisco Systems, Inc.

Compiled Wed 18-Mar-09 21:33 by prod\_rel\_team

ROM: System Bootstrap, Version 12.4(13r)T10, RELEASE SOFTWARE (fc1)

3825\_ISR uptime is 2 weeks, 5 days, 22 hours, 36 minutes

System returned to ROM by power-on

System restarted at 22:17:18 UTC Thu May 21 2009

System image file is "flash:c3825-ipvoicek9-mz.124-22.YB1.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](mailto:export@cisco.com).



Cisco 3825 (revision 1.2) with 225280K/36864K bytes of memory.

Processor board ID FHK1240F1YA

2 Gigabit Ethernet interfaces

24 Serial interfaces

1 terminal line

2 Channelized/Clear T1/PRI ports

1 cisco service engine(s)

DRAM configuration is 64 bits wide with parity enabled.

479K bytes of NVRAM.

125440K bytes of ATA System CompactFlash (Read/Write)

Configuration register is 0x2102

**show running-config**

3825\_ISR# sho run

Building configuration...

Current configuration : 8596 bytes

!

! Last configuration change at 17:54:10 UTC Wed Jun 10 2009 by cisco

! NVRAM config last updated at 23:15:35 UTC Mon Jun 8 2009 by cisco

!

version 12.4

service timestamps debug datetime msec

service timestamps log datetime msec

no service password-encryption

!

hostname 3825\_ISR

!



```
boot-start-marker
boot system flash:c3825-ipvoicek9-mz.124-22.YB1.bin
boot-end-marker
!
logging message-counter syslog
logging buffered 10000000
enable secret 5 $1$o/C6$75SPaNY1/MBJVPV5XnMI5/
!
no aaa new-model
no network-clock-participate slot 1
!
voice-card 01
dspfarm
dsp services dspfarm
!
voice-card 1
dspfarm
!
dot11 syslog
ip source-route
ip cef
!
!
ip domain name yourdomain.com
no ipv6 cef
multilink bundle-name authenticated
!
```

---

<sup>1</sup> This command enables the PVDM to farm (share) DSP resources. In this config example, the DSP is used for the transcoding application.



```
!  
isdn switch-type primary-qsig  
!  
!  
voice service pots  
!  
!  
voice service voip  
no notify redirect ip2ip  
qsig decode  
allow-connections sip to sip2  
redirect ip2ip  
sip  
registrar server3  
g729 annexb-all4  
!  
!  
!  
voice class codec 15  
codec preference 1 g711ulaw  
codec preference 2 g729r8  
codec preference 3 g729br8  
!  
voice class codec 2  
codec preference 1 g729r8  
codec preference 2 g729br8
```

---

<sup>2</sup> This command enables SIP B2BUA feature on CUBE

<sup>3</sup> Enables the SIP registrar service for SIP end-points to register.

<sup>4</sup> Enables CUBE to accept all versions of G729 codec (annex A, annex B).

<sup>5</sup> This command provides codec selection and codec filtering.



```
codec preference 3 g711ulaw
```

```
!
```

```
!
```

```
!
```

```
!
```

```
!
```

```
!
```

```
!
```

```
!
```

```
voice register global6
```

```
mode cme
```

```
source-address 172.20.236.249 port 5060
```

```
max-dn 100
```

```
max-pool 185
```

```
load 7961 SIP41.8-4-2S
```

```
voicemail 5210
```

```
tftp-path flash:
```

```
create profile sync 0903144640020194
```

```
!
```

```
voice register dn 17
```

```
number 5200
```

```
call-forward b2bua busy 5210
```

```
call-forward b2bua mailbox 5200
```

```
call-forward b2bua noan 5210 timeout 12
```

```
name CME Test1
```

```
huntstop
```

```
mwi
```

---

<sup>6</sup> SIP global settings for SIP phone registration.

<sup>7</sup> SIP phone directory number settings.



```
!  
voice register dn 2  
number 5203  
call-forward b2bua busy 4123  
call-forward b2bua mailbox 5203  
call-forward b2bua noan 4124 timeout 20  
name CME Test2  
huntstop  
mwi  
!  
voice register template 1  
!  
voice register pool 18  
id mac 001E.13E6.3292  
type 7961  
number 1 dn 1  
template 1  
dtmf-relay rtp-nte  
voice-class codec 1  
no call-waiting  
description CME_test_1  
!  
voice register pool 2  
id mac 0015.2B34.0DC4  
type 7961  
number 1 dn 2  
template 1  
dtmf-relay rtp-nte
```

---

<sup>8</sup> SIP phone registration settings.



```
voice-class codec 2

no call-waiting

description CME_test_2

!

!

!

!

username cisco privilege 15 secret 5 $1$mE.S$p719vngMGTG4OTLhBbWg0/

archive

log config

hidekeys

!

!

controller T1 1/0/09

clock source internal

pri-group timeslots 1-24

!

controller T1 1/0/1

!

!

!

!

!

interface GigabitEthernet0/0

description $ETH-LAN$$ETH-SW-LAUNCH$$INTF-INFO-GE 0/0$

ip address 172.20.236.249 255.255.255.0

duplex full

speed 100
```

---

<sup>9</sup> Configuration of T1 ISDN PRI trunk providing PSTN access.



```
media-type rj45
!
interface Service-Engine0/0
ip unnumbered GigabitEthernet0/0
service-module ip address 172.20.236.248 255.255.255.0
service-module ip default-gateway 172.20.236.1
!
interface GigabitEthernet0/1
no ip address
shutdown
duplex auto
speed auto
media-type rj45
!
interface Serial1/0/0:2310
no ip address
encapsulation hdlc
isdn switch-type primary-qsig
isdn incoming-voice voice
no cdp enable
!
ip default-gateway 172.20.236.1
ip forward-protocol nd
ip route 0.0.0.0 0.0.0.0 GigabitEthernet0/0
ip route 172.20.236.248 255.255.255.255 Service-Engine0/0
!
ip http server
ip http access-class 23
```

---

<sup>10</sup> T1 ISDN PRI interface settings (PSTN access).



```
ip http authentication local
no ip http secure-server
ip http timeout-policy idle 60 life 86400 requests 10000
!
!
!
!
!
fttp-server flash:P00308000500.bin11
fttp-server flash:P00308000500.sb2
fttp-server flash:P00308000500.loads
fttp-server flash:P00308000500.sbn
fttp-server flash:cnu41.8-4-1-23.sbn
fttp-server flash:dsp41.8-4-1-23.sbn
fttp-server flash:jar41sip.8-4-1-23.sbn
fttp-server flash:SIP41.8-4-2S.loads
fttp-server flash:apps41.8-4-1-23.sbn
fttp-server flash:term61.default.loads
fttp-server flash:apps70.8-4-1-23.sbn
fttp-server flash:cnu70.8-4-1-23.sbn
fttp-server flash:cvm70sccp.8-4-1-23.sbn
fttp-server flash:dsp70.8-4-1-23.sbn
fttp-server flash:jar70sccp.8-4-1-23.sbn
fttp-server flash:SCCP70.8-4-2S.loads
fttp-server flash:term70.default.loads
fttp-server flash:term71.default.loads
fttp-server flash:cvm70sip.8-4-1-23.sbn
fttp-server flash:jar70sip.8-4-1-23.sbn
```

---

<sup>11</sup> Recommended SCCP and SIP phone loads for CUCME 7.1.



```
fttp-server flash:SIP70.8-4-2S.loads
fttp-server flash:cvm41sip.8-4-1-23.sbn
fttp-server flash:cvm41sccp.8-4-1-23.sbn
fttp-server flash:jar41sccp.8-4-1-23.sbn
fttp-server flash:SCCP41.8-4-2S.loads
fttp-server flash:term41.default.loads
fttp-server flash:apps41.8-5-2TH1-9.sbn
fttp-server flash:cnu41.8-5-2TH1-9.sbn
fttp-server flash:cvm41sip.8-5-2TH1-9.sbn
fttp-server flash:dsp41.8-5-2TH1-9.sbn
fttp-server flash:jar41sip.8-5-2TH1-9.sbn
fttp-server flash:SIP41.8-5-2S.loads
!
control-plane
!
!
!
voice-port 1/0/0:23
!
ccm-manager fax protocol cisco
!
mgcp fax t38 ecm
!
sccp local GigabitEthernet0/012
sccp ccm 172.20.236.249 identifier 1 version 7.0
sccp
!
sccp ccm group 1
```

---

<sup>12</sup> Transcoder settings to register to CUCME for conference calls between g711 and g729 end-points.



```
associate ccm 1 priority 1
associate profile 1 register mtp002333351560
!
dspfarm profile 1 transcode
codec g711ulaw
codec g711alaw
codec g729ar8
codec g729abr8
codec g729r8
codec g729br8
maximum sessions 10
associate application SCCP
!
!
dial-peer voice 4000 voip13
description Avaya SIP1 trunk
destination-pattern 4...
voice-class sip g729 annexb-all
session protocol sipv2
session target ipv4:172.20.212.253
session transport tcp
incoming called-number 5...
dtmf-relay rtp-nte
!
dial-peer voice 5210 voip14
description CUE Voicemail
destination-pattern 5210
```

---

<sup>13</sup> Dial-peer to Avaya PBX.

<sup>14</sup> Dial-peer to Cisco Unity Express voicemail.



```
no voice-class sip g729 annexb-all
session protocol sipv2
session target ipv4:172.20.236.248
dtmf-relay rtp-nte
codec g711ulaw
!
dial-peer voice 2100 pots15
description To PSTN phone
destination-pattern 21..
direct-inward-dial
port 1/0/0:23
forward-digits all
!
dial-peer voice 5050 voip16
description To Octel 200
destination-pattern 5050
voice-class codec 2
voice-class sip g729 annexb-all
session protocol sipv2
session target ipv4:172.20.212.253
session transport tcp
dtmf-relay rtp-nte
!
!
sip-ua
mwi-server ipv4:172.20.236.248 expires 3600 port 5060 transport tcp unsolicited17
```

---

<sup>15</sup> Dial-peer to PSTN phone

<sup>16</sup> Dial-peer to Avaya voicemail system (Octel 200)

<sup>17</sup> Setting defining Cisco Unity Express as the SIP message waiting server



```
!  
!  
telephony-service18  
  sdsfarm units 1  
  sdsfarm transcode sessions 10  
  sdsfarm tag 1 mtp002333351560  
  em logout 0:0 0:0 0:0  
  max-ephones 96  
  max-dn 500  
  ip source-address 172.20.236.249 port 2000  
  system message CME Bench 8  
  load 7960-7940 P00308000500  
  load 7941 SCCP41.8-4-2S  
  load 7961 SCCP41.8-4-2S  
  voicemail 5210  
  mwi relay  
  max-conferences 8 gain -6  
  call-forward pattern ....  
  moh music-on-hold.au  
  transfer-system full-consult  
  transfer-pattern ....  
  secondary-dialtone 9  
  create cnf-files version-stamp 7960 Jun 08 2009 22:41:22  
!  
!  
ephone-dn 1 dual-line  
  call-waiting ring  
  number 5205
```

---

<sup>18</sup> CUCME and SCCP phone settings.



```
name CME Test3
call-forward busy 4124
call-forward noan 5210 timeout 12
mwi-type both
!
!
ephone-dn 2 dual-line
number 5209
name CME Test4
call-forward noan 4124 timeout 8
mwi-type both
!
!
ephone-dn 3 dual-line
number 5204
name CME Test5
call-forward busy 5210
call-forward noan 5210 timeout 12
mwi-type both
!
!
!
!
ephone 1
mac-address 0015.2BB5.C4C1
codec g729r8
type 7961
keep-conference endcall
button 1:1
```



```
!  
!  
!  
ephone 2  
  mac-address 001B.0CAD.6662  
  codec g729r8  
  type 7961  
  keep-conference endcall  
  button 1:2  
!  
!  
!  
ephone 3  
  mac-address 000B.BEAD.4C1C  
  codec g729r8  
  type 7960  
  keep-conference endcall  
  button 1:3  
!  
!  
!  
line con 0  
  login local  
line aux 0  
line 194  
  no activation-character  
  no exec  
  transport preferred none  
  transport input all
```



```
transport output pad telnet rlogin lapb-ta mop udptn v120
line vty 0 4
access-class 23 in
privilege level 15
login local
transport input telnet
line vty 5 15
access-class 23 in
privilege level 15
login local
transport input telnet
!
scheduler allocate 20000 1000
ntp server 172.20.2.181
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



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