



Avaya S8500 Version 4.0 to Cisco IOS Voice Gateway using SIP with T1-PRI to PSTN

February 20, 2008

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Introduction

- This Application note provides basic call interoperability and documented steps and configurations necessary for SIP integration between Avaya S8500 to Cisco IOS Voice Gateway providing PSTN (T1-PRI) connectivity.
- The SIP protocol is used between Cisco IOS Voice gateway and Avaya S8500. The connection between Cisco IOS gateway and PSTN uses T1-PRI (DMS-100 switch type).
- Features tested include Basic call, Call Transfer supervised, Call Transfer blind, Call Forward (Unconditional, Busy and No Answer), Three-way Conference, DTMF tones (in-band and relay via RFC2833), Digit Translation, and Call Hold.
- The Cisco IOS Voice Gateway offers the advantage of providing connectivity between Avaya S8500 and PSTN by offering SIP to T1-PRI inter-working functionality.
- The network topology diagram (Figure 1) shows the test setup for end-to-end interoperability with the Cisco IOS Voice Gateway connected to the Avaya S8500 (10/100baseT) and connected to the PSTN via T1-PRI.
- This Application Notes uses the Cisco 3845 IOS-voice-gateway, however other Cisco voice gateways are also an option to use since the voice gateway implementation does not depend on the platform. Below is a list of Cisco platforms capable of voice gateway functionality: Care must be taken when selecting a voice gateway platform depending on the capacity and capability required for the intended deployment.

[Cisco 1861 Integrated Services Router](#)

[Cisco IAD2400 Series Integrated Access Device](#)

[Cisco 2800 Series Integrated Services Routers](#)

[Cisco 3700 Series Routers](#)

[Cisco 3800 Series Integrated Services Routers](#)

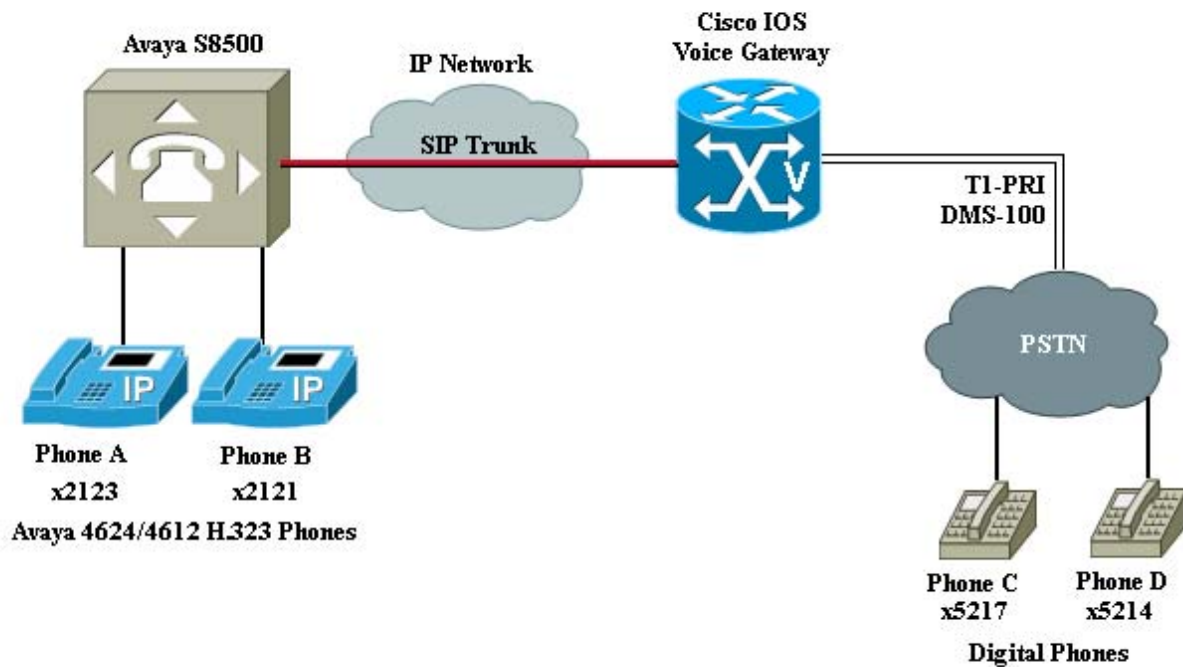
[Cisco AS5350XM Universal Gateway](#)

[Cisco AS5400XM Universal Gateway](#)



Network Topology

Figure 1. Network Topology



Limitations

- Basic call using G.726-32K and G.722-64K failed. Avaya rejects G.726 and G.722 codec, even when the Avaya is set for G.726 and G.722.
- The Avaya H323 phones did not work when the trunk codec was set to G.723-5.3K or G.723-6.3K.
- For inband DTMF, the Avaya "DTMF over IP" setting in the signalling group had to be set to "in-band", not "in-band-g711".
- For basic calls with RFC2833 DTMF relay, it was noticed that the PBX phone (e.g., 'A') had to send DTMF to the PSTN digital phone (e.g., 'C') before it could play tones originating from the PSTN digital phone. This was due to the H.323/SIP interworking on the PBX itself, and was also noticed on calls involving only Avaya SIP and H.323 phones.
- On a call that originates from a PSTN station to a PBX station and that is then forwarded unconditionally or on busy (e.g., phone C calls phone A, and phone A forwards to phone D), the originating PSTN user does not get ringback. Cisco IOS Voice Gateway does not support SIP 181 message (call being forwarded) which causes SDP establishment failure during a call forward between PSTN phone and Avaya stations.



System Components

Hardware requirements

Cisco equipment

- Cisco 3845 (Cisco 3800 family routers)

Avaya equipment

- Avaya S8500
- Avaya SIP Enablement Server (SES)
- (2) Avaya 4624/4612 IP phones
- TN2312BP IPSI
- TN799DP C-LAN
- TN2302AP IP Media Processor

Software Requirements

- Cisco IOS Voice Gateway: Cisco IOS Release – Cisco 3845 Version 12.4(15)T3: c3845-ipvoice-mz.124-15.T3.bin
- Avaya Communications Manager Release 4.0
- Avaya SES software version 4.0
- Avaya 4612/4624 (H.323 phones) load: def24rl_8_3.bin

Features

Features Supported

- Basic call with G.729, G.729B, G.711ulaw and G.711Alaw
- Call Transfer Blind and Call Transfer Supervised
- Call Conference
- Call on Hold
- Call Forward All (See Limitations Section)
- Call Forward No Reply
- Call Forward Busy (See Limitations Section)
- Digit Translation
- DTMF in-band
- DTMF - RFC2833 (See Limitations Section)



Configuration

Avaya S8500 Configuration

Figure 2. Signaling group (In band)

The screenshot shows the DEFINITY Site Administration interface for SIP2 GEDI. The main configuration area is titled "SIGNALING GROUP" and displays the following settings:

- Group Number:** 1 (highlighted with a red box)
- Group Type:** sip
- Transport Method:** t1s
- Near-end Node Name:** clan1
- Far-end Node Name:** avayasip2
- Near-end Listen Port:** 5061
- Far-end Listen Port:** 5061
- Far-end Network Region:** 1
- Far-end Domain:** (empty field)
- DTMF over IP:** in-band (highlighted with a red box)
- Bypass If IP Threshold Exceeded?** n
- Direct IP-IP Audio Connections?** y
- IP Audio Hairpinning?** y
- Enable Layer 3 Test?** n
- Session Establishment Timer(min):** 120

The interface also includes a left-hand navigation menu with options like "Start GEDI", "Add User", "Change User Name", "Remove User", "Add Bridged Appearance", "Browse Dial Ranges", "Browse Stations", "Browse Unused Ports", "Find Unused Extension", and "Print Button Labels". At the bottom, there is a "History" tab showing a list of events:

Severity	Date/Time	System	Description
Info	1/15/2008 5:14:43 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:12:58 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:11:12 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:01:08 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:00:15 PM	SIP2	change ip-codec-set 3



Figure 3. Signaling group (RFC 2833)

The screenshot displays the DEFINITY Site Administration interface for configuring a signaling group. The main configuration area is titled "SIGNALING GROUP" and shows the following settings:

- Group Number:** 1 (highlighted with a red box)
- Group Type:** sip
- Transport Method:** t1s
- Near-end Node Name:** c1an1
- Near-end Listen Port:** 5061
- Far-end Node Name:** avayasip2
- Far-end Listen Port:** 5061
- Far-end Network Region:** 1
- Far-end Domain:** (empty field)
- Bypass If IP Threshold Exceeded?** n
- DTMF over IP:** rtp-payload (highlighted with a red box)
- Direct IP-IP Audio Connections?** y
- IP Audio Hairpinning?** y
- Enable Layer 3 Test?** n
- Session Establishment Timer(min):** 120

The interface also includes a left-hand navigation pane with options like "Start GEDI", "Add User", "Change User Name", "Remove User", "Add Bridged Appearance", "Browse Dial Ranges", "Browse Stations", "Browse Unused Ports", "Find Unused Extension", and "Print Button Labels". At the bottom, there is a "History" table with columns for Severity, Date/Time, System, and Description.

Severity	Date/Time	System	Description
Info	1/15/2008 5:14:43 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:12:58 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:11:12 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:01:08 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:00:15 PM	SIP2	change ip-codec-set 3



Figure 4. Trunk group – p1 of 3

The screenshot shows the Cisco DEFINITY Site Administration interface. The main window displays the configuration for trunk group 1. The configuration details are as follows:

```
TRUNK GROUP
Group Number: 1
Group Name: OUTSIDE CALL
Direction: two-way
Dial Access? n
Queue Length: 0
Service Type: tie
Group Type: sip
COR: 1
Outgoing Display? n
Auth Code? n
CDR Reports: y
TN: 1
TAC: 801
Signaling Group: 1
Number of Members: 6
```

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

- General
 - Start GEDI
 - Add User
 - Change User Name
 - Remove User
 - Add Bridged Appearance
 - Browse Dial Ranges
 - Browse Stations
 - Browse Unused Ports
 - Find Unused Extension
 - Print Button Labels
- Advanced
- Fault & Performance
- Tasks
- Tree

The bottom of the interface shows a log window with the following entries:

Severity	Date/Time	System	Description
Info	1/15/2008 5:14:43 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:12:58 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:11:12 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:01:08 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:00:15 PM	SIP2	change ip-codec-set 3

The status bar at the bottom indicates "Ready" and "NUM".



Figure 5. Trunk group – p2 of 3

The screenshot shows the DEFINITY Site Administration interface for a SIP2 trunk group. The main window displays the configuration for 'trunk-group 1'. The configuration parameters are as follows:

- Group Type: sip
- Unicode Name? y
- Redirect On OPTIM Failure: 5000
- SCCAN? n
- Digital Loss Group: 18
- Preferred Minimum Session Refresh Interval(sec): 900

The interface also includes a left-hand navigation menu with options such as 'Start GEDI', 'Add User', 'Change User Name', 'Remove User', 'Add Bridged Appearance', 'Browse Dial Ranges', 'Browse Stations', 'Browse Unused Ports', 'Find Unused Extension', and 'Print Button Labels'. At the bottom, there is a 'History' table with the following data:

Severity	Date/Time	System	Description
Info	1/15/2008 5:14:43 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:12:58 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:11:12 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:01:08 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:00:15 PM	SIP2	change ip-codec-set 3



Figure 6. Trunk group – p3 of 3

The screenshot shows the DEFINITY Site Administration interface for SIP2 GEDJ. The main window displays the configuration for trunk group 1, specifically the 'TRUNK FEATURES' section. The configuration includes:

- ACA Assignment? n
- Measured: none
- Maintenance Tests? y
- Numbering Format: private
- UUI Treatment: service-provider
- Replace Unavailable Numbers? n
- Show ANSWERED BY on Display? y

The interface also features a left sidebar with navigation options such as Start GEDI, Add User, Change User Name, Remove User, Add Bridged Appearance, Browse Dial Ranges, Browse Stations, Browse Unused Ports, Find Unused Extension, and Print Button Labels. A bottom status bar displays a log of recent events, including several 'Info' messages from SIP2 regarding 'change ip-codec-set 3'.

Severity	Date/Time	System	Description
Info	1/15/2008 5:14:43 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:12:58 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:11:12 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:01:08 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:00:15 PM	SIP2	change ip-codec-set 3



Figure 7. Node-names IP

The screenshot shows the Cisco DEFINITY Site Administration interface for SIP2 GEDI. The main window displays a list of IP Node Names with columns for Name and IP Address. The list includes entries such as CCH3.3, CCM4.1, CCM4.1.2, CCM5.0-VENUS, CM-KLINGON, CM-POLARIS, CM-cluster1_s, IPIPGW, MAvantage, avayasip1, avayasip2, clan1, clan1server1, default, medpro1, and procr. Below the list, a message indicates that 16 of 16 administered node-names were displayed and provides instructions on how to view all administered node-names and how to change or add a node-name.

Name	IP Address
CCH3.3	172.20.31.254
CCM4.1	172.20.231.254
CCM4.1.2	172.20.236.2
CCM5.0-VENUS	172.20.214.254
CM-KLINGON	172.20.32.254
CM-POLARIS	172.20.236.50
CM-cluster1_s	172.20.241.253
IPIPGW	172.20.8.26
MAvantage	172.20.7.252
avayasip1	172.20.212.254
avayasip2	172.20.213.254
clan1	172.20.213.253
clan1server1	172.20.212.253
default	0.0.0.0
medpro1	172.20.213.252
procr	172.20.213.200

(16 of 16 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

Severity	Date/Time	System	Description
Info	1/15/2008 5:14:43 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:12:58 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:11:12 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:01:08 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:00:15 PM	SIP2	change ip-codec-set 3



Figure 8. IP Network Region

The screenshot shows the DEFINITY Site Administration interface for SIP2 GEDI. The main window displays the configuration for 'IP NETWORK REGION' (Region: 1, Location: 1, Authoritative Domain: lab2.com). The configuration is organized into several sections:

- MEDIA PARAMETERS:**
 - Intra-region IP-IP Direct Audio: yes
 - Inter-region IP-IP Direct Audio: yes
 - IP Audio Hairpinning? y
 - Codec Set: 3
 - UDP Port Min: 2048
 - UDP Port Max: 3029
- 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:**
 - RSUP 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

The interface also includes a left-hand navigation menu with options like 'Start GEDI', 'Add User', and 'Change User Name'. At the bottom, there is a 'History' log showing several 'Info' messages from SIP2 regarding 'change ip-codec-set 3'.



Figure 9. IP codec set (showing G.729, annex B).

The screenshot displays the 'DEFINITY Site Administration - [SIP2 GEDI]' window. The main configuration area is titled 'IP Codec Set' and shows 'Codec Set: 3'. Below this, there is a table with the following columns: Audio Codec, Silence Suppression, Frames Per Pkt, and Packet Size(ms). The first row is populated with 'G.729B', 'n', '2', and '20'. Below the table is a 'Media Encryption' section with three rows, the first of which is 'none'.

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

Media Encryption		
1:	none	
2:		
3:		

Severity	Date/Time	System	Description
Info	1/15/2008 5:14:43 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:12:58 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:11:12 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:01:08 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:00:15 PM	SIP2	change ip-codec-set 3



Figure 10. Uniform dialing plan

DEFINTY® Site Administration - [SIP2 GEDI]

display uniform-dialplan 1 send (return) help (f5) cancel (esc) enter (f3) schedule (f9) next (f7) previous (f8) next form (f6)

UNIFORM DIAL PLAN TABLE Percent Full: 0

Matching Pattern	Len	Del	Insert Digits	Net	Conv	Node Num
16	4	0	201	aar	n	
1600	4	0	213	aar	n	
1612	4	0	216	aar	n	
1613	4	0	216	aar	n	
2200	4	0	204	aar	n	
2201	4	0	204	aar	n	
26	4	0	201	aar	n	
29	4	0	201	aar	n	
30	4	0	201	aar	n	
3503	4	0	201	aar	n	
3504	4	0	201	aar	n	
36	4	0	214	aar	n	
37	4	0	213	aar	n	
40	4	0	201	aar	n	
4131	4	0	201	aar	n	
4132	4	0	201	aar	n	

Severity | Date/Time | System | Description

- Info 1/15/2008 5:14:43 PM SIP2 change ip-codec-set 3
- Info 1/15/2008 5:12:58 PM SIP2 change ip-codec-set 3
- Info 1/15/2008 5:11:12 PM SIP2 change ip-codec-set 3
- Info 1/15/2008 5:01:08 PM SIP2 change ip-codec-set 3
- Info 1/15/2008 5:00:15 PM SIP2 change ip-codec-set 3

History | Schedule | Connection Status

NUM



Figure 11. AAR analysis

DEFINITY® Site Administration - [SIP2 GEDI]

File Edit View System Action Tools Window Help

SIP2

display aar analysis 201 send (return) help (f5) cancel (esc) enter (f3) schedule (f9) next (f7) previous (f8) next form (f6)

1 2

AAR DIGIT ANALYSIS TABLE Percent Full: 1

Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Reqd
201	7	7	1	aar		n
202	7	7	2	aar		n
204	7	7	4	aar		n
213	7	7	13	aar		n
214	7	7	14	aar		n
215	7	7	15	aar		n
216	7	7	6	aar	1	n
217	7	7	6	aar	6	n
224	7	7	224	aar		n
3	7	7	999	aar		n
4	7	7	999	aar		n
5	7	7	999	aar		n
6	7	7	999	aar		n
7	7	7	999	aar		n
8	7	7	999	aar		n

General

- Start GEDI
- Add User
- Change User Name
- Remove User
- Add Bridged Appearance
- Browse Dial Ranges
- Browse Stations
- Browse Unused Ports
- Find Unused Extension
- Print Button Labels

Advanced

Fault & Performance

Tasks Tree

Severity	Date/Time	System	Description
Info	1/15/2008 5:14:43 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:12:58 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:11:12 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:01:08 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:00:15 PM	SIP2	change ip-codec-set 3

History Schedule Connection Status

Ready NUM



Figure 12. Route Pattern

DEFINITY® Site Administration - [SIP2 GEDI]

File Edit View System Action Tools Window Help

display route-pattern 1 send (return) help (f5) cancel (esc) enter (f3) schedule (f9) next (f7) previous (f8) next form (f6)

1 2 3

Pattern Number: 1 Pattern Name: SES

Grp No FRL NPA Pfx Hop Toll No. Inserted DCS/ IXC
 Mrk Lmt List Del Digits Intw

Grp No	FRL	NPA	Pfx	Hop	Toll	No.	Inserted	DCS/ IXC	IXC
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

Grp No	BCC	VALUE	TSC	CA-TSC	ITC	BCIE	Service/Feature	PARM	No.	Numbering	LAR
1:	y	y	y	y	n	y	as-needed	rest			none
2:	y	y	y	y	n	n		rest			none
3:	y	y	y	y	n	n		rest			none
4:	y	y	y	y	n	n		rest			none
5:	y	y	y	y	n	n		rest			none
6:	y	y	y	y	n	n		rest			none

Severity	Date/Time	System	Description
Info	1/15/2008 5:14:43 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:12:58 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:11:12 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:01:08 PM	SIP2	change ip-codec-set 3
Info	1/15/2008 5:00:15 PM	SIP2	change ip-codec-set 3

Ready NUM



Avaya SIP Enablement Server (SES) Configuration

Figure 13. SES Server

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	172.20.213.254	home/edge

Force All
Migrate Home/Edge



Figure 14. SES Trusted Hosts

Help Exit

Top

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

List Trusted Hosts

<u>Commands</u>	<u>IP Address</u>	<u>Trusted by Host</u>	<u>Comment</u>
Edit Delete	172.20.15.123	172.20.213.254	Chris_GW
Edit Delete	172.20.150.251	172.20.213.254	Chinh_CCM
Edit Delete	172.20.212.253	172.20.213.254	Avaya CM1
Edit Delete	172.20.212.254	172.20.213.254	Avaya SIP1
Edit Delete	172.20.214.254	172.20.213.254	CCM-Venus
Edit Delete	172.20.215.254	172.20.213.254	CM-Mercury
Edit Delete	172.20.236.50	172.20.213.254	CM-Polaris

Add Another Trusted Host



Figure 15. SES DN 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 172.20.213.254

Commands	Name	Commands	Contact
Edit Delete	Avaya_SIP1		
Edit Delete	Avaya_SIP1_b	Edit Delete	sip:\$(user)@172.20.212.254:5060;transport=tcp
Add Another Map		Add Another Contact Delete Group	
Edit Delete	CM_Mercury_42	Edit Delete	sip:\$(user)@172.20.215.254:5060;transport=tcp
Add Another Map		Add Another Contact Delete Group	
Edit Delete	CCM-Venus	Edit Delete	sip:\$(user)@172.20.214.254:5060;transport=tcp
Add Another Map		Add Another Contact Delete Group	
Edit Delete	CCM-Venus_VM	Edit Delete	sip:\$(user)@172.20.214.254:5060;transport=tcp
Add Another Map		Add Another Contact Delete Group	
Edit Delete	Chinh_CCM	Edit Delete	sip:\$(user)@172.20.150.251:5060;transport=tcp
Add Another Map		Add Another Contact Delete Group	
Edit Delete	CM_Polaris_500X	Edit Delete	sip:\$(user)@172.20.236.50:5060;transport=tcp
Add Another Map		Add Another Contact Delete Group	
Edit Delete	Chris_GW	Edit Delete	sip:\$(user)@172.20.15.123:5060;transport=tcp
Add Another Map		Add Another Contact Delete Group	
Add Map In New Group			



Figure 16. SES 30XX Routing Entry

Integrated Management
SIP Server Management
Server: 172.20.213.254

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.



Cisco IOS Voice Gateway (c3845) configuration

show version

Cisco IOS Software, 3800 Software (C3845-IPVOICE-M), Version 12.4(15)T3, RELEASE SOFTWARE (fc1)
Technical Support: <http://www.cisco.com/techsupport>
Copyright (c) 1986-2008 by Cisco Systems, Inc.
Compiled Fri 25-Jan-08 00:11 by prod_rel_team

ROM: System Bootstrap, Version 12.3(11r)T1, RELEASE SOFTWARE (fc1)

3845_West uptime is 2 weeks, 43 minutes
System returned to ROM by reload at 18:17:14 UTC Wed Feb 6 2008
System image file is "flash:c3845-ipvoice-mz.124-15.T3.bin"

Cisco 3845 (revision 1.0) with 224256K/37888K bytes of memory.
Processor board ID FHK0847F136
2 Gigabit Ethernet interfaces
24 Serial interfaces
2 Channelized E1/PRI ports
2 Channelized T1/PRI ports
2 Voice FXS interfaces
DRAM configuration is 64 bits wide with parity enabled.
479K bytes of NVRAM.
62592K bytes of ATA System CompactFlash (Read/Write)

Configuration register is 0x2102

show run

Building configuration...

```
Current configuration : 2936 bytes
!
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname 3845_West
!
boot-start-marker
boot-end-marker
!
card type t1 0 0
logging buffered 1000000
no logging console
enable secret 5 $1$MFhi$AqqpDsFeO4Sb/IkzkrcmO/
!
!
no aaa new-model
network-clock-participate wic 0
network-clock-select 1 T1 0/0/0
voice-card 0
no dspfarm
```



```
!  
ip cef  
!  
no ip domain lookup  
!  
!  
isdn switch-type primary-dms100  
isdn gateway-max-interworking  
!  
!  
voice call carrier capacity active  
!  
voice service pots  
!  
voice service voip  
!  
!  
!  
voice translation-rule 1 1  
rule 1 /30/ /52\1/  
!  
voice translation-rule 2  
rule 1 /11/ /21\1/  
!  
!  
voice translation-profile PSTN-to-SIP  
translate called 2  
!  
voice translation-profile SIP-to-PSTN 2  
translate called 1  
!  
!  
controller T1 0/0/0  
framing esf  
linecode b8zs  
pri-group timeslots 1-24  
!  
controller T1 0/0/1  
framing esf  
linecode b8zs  
!  
ip tcp synwait-time 13  
!  
!  
interface GigabitEthernet0/0  
description $ETH-LAN$$ETH-SW-LAUNCH$$INTF-INFO-GE 0/0$  
ip address 10.10.10.1 255.255.255.248  
shutdown  
duplex auto  
speed auto  
media-type rj45  
no keepalive  
!
```

¹ The above translation rule replaces the number “30xx” with “52xx”

² This defines the translation profile for the called number. In this particular example, the called number “30xx” is converted to “52xx”.



```
interface GigabitEthernet0/1
ip address 172.20.15.123 255.255.255.0
duplex auto
speed auto
media-type rj45
no keepalive
!
!
interface Serial0/0/0:23
no ip address
encapsulation hdlc
isdn switch-type primary-dms100
isdn protocol-emulate network
isdn incoming-voice voice
isdn channel-id invert extend-bit
no cdp enable
!
ip forward-protocol nd
ip route 0.0.0.0 0.0.0.0 172.20.15.1
!
ip http server
ip http authentication local
!
!
control-plane
!
voice-port 0/0/0:23
!
voice-port 0/1/0
!
voice-port 0/1/1
!
dial-peer voice 4000 pots
translation-profile incoming PSTN-to-SIP
destination-pattern 5...
incoming called-number 1...
direct-inward-dial
port 0/0/0:23
forward-digits all
!
dial-peer voice 519 voip
translation-profile incoming SIP-to-PSTN3
destination-pattern 2...
< rtp payload-type nte 127>4
session protocol sipv2
session target ipv4:172.20.213.254
incoming called-number 3...5
dtmf-relay rtp-nte6
codec g711ulaw7
```

³ This defines the call number translation profile for incoming calls. Here, a called number “30xx” is converted to “40xx” for an incoming call associated with this dial peer (i.e., via this SIP trunk).

⁴ This command is not necessary when using Avaya Communications Manager 4.0, and was not used for the creation of this application note. It is only needed with prior releases of Avaya Communications Manager.

⁵ This is to specify a digit string that can be matched by an incoming call to associate the call with a dial peer. For this example, user dials 3xxx will associate to this dial-peer.

⁶ This is for Dual-Tone Multifrequency (DTMF) tones to be sent RFC2833. It is omitted for inband DTMF.

⁷ Also changed to G.711A, G.729, and G.729B during testing.



```
!  
banner login ^C  
!  
line con 0  
  exec-timeout 600 0  
  password cisco  
  login  
line aux 0  
line vty 0 4  
  exec-timeout 600 0  
  privilege level 15  
  password cisco  
  login  
  transport input telnet  
line vty 5 15  
  privilege level 15  
  login local  
  transport input telnet  
!  
scheduler allocate 20000 1000  
!  
end
```



Acronyms

Acronym	Definitions
codec	compressor/decompressor
PBX	Private Branch Exchange
PSTN	Public Switched Telephone Network
IOS	Internetworking Operating System

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