



# Avaya S8500 Version 4.0 to Cisco IOS Voice Gateway using SIP with E1-CAS to PSTN

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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 (E1-CAS) connectivity.
- The SIP protocol is used between Cisco IOS Voice gateway and Avaya S8500. The connection between Cisco IOS gateway and PSTN uses E1-CAS (E&M, immediate start).
- 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 E1-CAS 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 E1-CAS.
- 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](#)

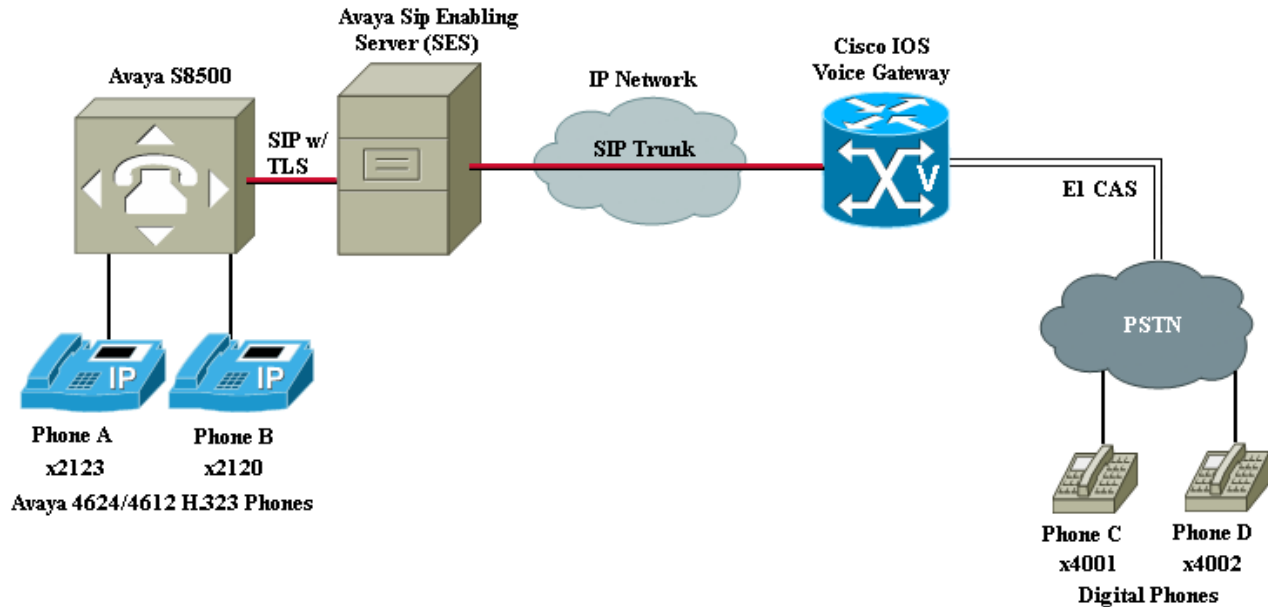
[Cisco AS5350XM Universal Gateway](#)

[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(18): c3845-ipvoice-mz.124-18.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
- 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)
- 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" log showing several "Info" messages from 1/15/2008 regarding "change ip-codec-set 3".



Figure 3. Signaling group (RFC 2833)

DEFINTY Site Administration - [SIP2 GEDI]

File Edit View System Action Tools Window Help

change signaling-group 1 send (return) help (f5) cancel (esc) enter (f3) schedule (f9) next (f7) previous (f8) next form (f6)

1

**SIGNALING GROUP**

**Group Number: 1**

Group Type: sip  
Transport Method: t1s

Near-end Node Name: c1an1 Far-end Node Name: avayasip2  
Near-end Listen Port: 5061 Far-end Listen Port: 5061  
Far-end Network Region: 1  
Far-end Domain:

Bypass If IP Threshold Exceeded?

DTMF over IP:  rtp-payload

Direct IP-IP Audio Connections?   
IP Audio Hairpinning?

Enable Layer 3 Test?   
Session Establishment Timer(min): 120

Right-click in a field to see a list of valid entries or help text

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 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 interface includes a menu bar (File, Edit, View, System, Action, Tools, Window, Help), a toolbar with various icons, and 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 log window showing several 'Info' messages from SIP2 regarding 'change ip-codec-set 3'.



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 'display 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 includes a menu bar (File, Edit, View, System, Action, Tools, Window, Help), a toolbar with various icons, and 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 6. Trunk group – p3 of 3

The screenshot shows the DEFINITY Site Administration interface for SIP2 GEDJ. The main configuration area displays the following settings for Trunk Group 1:

- 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 bottom pane shows a history 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 7. Node-names IP

The screenshot shows the Cisco DEFINTY 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 use the 'list node-names' and 'change node-names ip xxx' commands.

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 displays the DEFINITY Site Administration interface for SIP2 GEDI. The main window shows the configuration for 'IP NETWORK REGION' with the following parameters:

```

IP NETWORK REGION
Region: 1
Location: 1      Authoritative Domain: lab2.com
Name: CiscoLAB2
MEDIA PARAMETERS      Intra-region IP-IP Direct Audio: yes
Codecs Set: 3         Inter-region IP-IP Direct Audio: yes
UDP Port Min: 2048    IP Audio Hairpinning? y
UDP Port Max: 3029
DIFFSERV/TOS PARAMETERS      RTCP Reporting Enabled? y
Call Control PHB Value: 34    RTCP MONITOR SERVER PARAMETERS
Audio PHB Value: 46           Use Default Server Parameters? y
Video PHB Value: 26
802.1P/Q PARAMETERS
Call Control 802.1p Priority: 7
Audio 802.1p Priority: 6
Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS          RSUP Enabled? n
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, a 'History' tab is active, showing a log of recent changes to the IP-codec-set 3 configuration.

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 9. IP codec set

The screenshot shows the DEFINITY Site Administration interface for SIP2 GEDI. The main window displays the configuration for 'IP Codec Set' for 'Codec Set: 3'. The interface includes a menu bar, a toolbar, and a left-hand navigation pane with options like 'Start GEDI', 'Add User', and 'Change User Name'. The main content area contains a table for audio codecs and a section for media encryption.

**IP Codec Set**

Codec Set: 3

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:	

At the bottom of the window, 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 10. Uniform dialing plan

The screenshot shows the Cisco DEFINTY Site Administration interface for SIP2 GEDI. The main window displays the 'UNIFORM DIAL PLAN TABLE' with a 'Percent Full: 0' indicator. The table lists various matching patterns and their corresponding network configurations. The row for pattern '30' is highlighted with a red circle.

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	

At the bottom of the interface, there is a log window showing several 'Info' messages from 1/15/2008, all related to 'change ip-codec-set 3' for the SIP2 system.



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 GEDJ]

File Edit View System Action Tools Window Help

SIP2

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  
 Dgts

Grp No	FRL	NPA	Pfx	Hop	Toll	No.	Inserted	DCS/ 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
<a href="#">Edit</a> <a href="#">Delete</a>	Avaya_SIP1		
<a href="#">Edit</a> <a href="#">Delete</a>	Avaya_SIP1_b	<a href="#">Edit</a> <a href="#">Delete</a>	sip:\$(user)@172.20.212.254:5060;transport=tcp
<a href="#">Add Another Map</a>		<a href="#">Add Another Contact</a> <a href="#">Delete Group</a>	
<a href="#">Edit</a> <a href="#">Delete</a>	CM_Mercury_42	<a href="#">Edit</a> <a href="#">Delete</a>	sip:\$(user)@172.20.215.254:5060;transport=tcp
<a href="#">Add Another Map</a>		<a href="#">Add Another Contact</a> <a href="#">Delete Group</a>	
<a href="#">Edit</a> <a href="#">Delete</a>	CCM-Venus	<a href="#">Edit</a> <a href="#">Delete</a>	sip:\$(user)@172.20.214.254:5060;transport=tcp
<a href="#">Add Another Map</a>		<a href="#">Add Another Contact</a> <a href="#">Delete Group</a>	
<a href="#">Edit</a> <a href="#">Delete</a>	CCM-Venus_VM	<a href="#">Edit</a> <a href="#">Delete</a>	sip:\$(user)@172.20.214.254:5060;transport=tcp
<a href="#">Add Another Map</a>		<a href="#">Add Another Contact</a> <a href="#">Delete Group</a>	
<a href="#">Edit</a> <a href="#">Delete</a>	Chinh_CCM	<a href="#">Edit</a> <a href="#">Delete</a>	sip:\$(user)@172.20.150.251:5060;transport=tcp
<a href="#">Add Another Map</a>		<a href="#">Add Another Contact</a> <a href="#">Delete Group</a>	
<a href="#">Edit</a> <a href="#">Delete</a>	CM_Polaris_500X	<a href="#">Edit</a> <a href="#">Delete</a>	sip:\$(user)@172.20.236.50:5060;transport=tcp
<a href="#">Add Another Map</a>		<a href="#">Add Another Contact</a> <a href="#">Delete Group</a>	
<a href="#">Edit</a> <a href="#">Delete</a>	Chris_GW	<a href="#">Edit</a> <a href="#">Delete</a>	sip:\$(user)@172.20.15.123:5060;transport=tcp
<a href="#">Add Another Map</a>		<a href="#">Add Another Contact</a> <a href="#">Delete Group</a>	
<a href="#">Add Map In New Group</a>			



Figure 16. SES 30XX Routing Entry

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**



## Cisco IOS Voice Gateway (c3845) configuration

### show version

Cisco IOS Software, 3800 Software (C3845-IPVOICE-M), Version 12.4(18), RELEASE SOFTWARE (fc1)  
Technical Support: <http://www.cisco.com/techsupport>  
Copyright (c) 1986-2007 by Cisco Systems, Inc.  
Compiled Fri 30-Nov-07 23:49 by prod\_rel\_team

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

3845\_West uptime is 1 day, 5 minutes  
System returned to ROM by reload at 22:11:19 UTC Mon Jan 14 2008  
System image file is "flash:c3845-ipvoice-mz.124-18.bin"

Cisco 3845 (revision 1.0) with 224256K/37888K bytes of memory.  
Processor board ID FHK0847F136  
2 Gigabit Ethernet interfaces  
2 Channelized E1/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 system flash:c3845-ipvoice-mz.124-18.bin
boot-end-marker
!
logging buffered 1000000 debugging
enable secret 5 $1$MFhi$AqqpDsFeO4Sb/IkzkrcmO/
!
no aaa new-model
network-clock-participate slot 1
network-clock-select 1 E1 1/0/0
voice-card 0
no dspfarm
!
voice-card 1
dspfarm
!
ip cef
```



```
!  
no ip domain lookup  
!  
voice call carrier capacity active  
!  
voice service pots  
!  
voice service voip  
!  
!  
!  
voice translation-rule 1 1  
rule 1 /30/ /40\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 E1 1/0/0  
ds0-group 1 timeslots 1-5 type e&m-immediate-start  
!  
controller E1 1/0/1  
!  
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  
!  
interface GigabitEthernet0/1  
ip address 172.20.15.123 255.255.255.0  
duplex auto  
speed auto  
media-type rj45  
no keepalive  
!  
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
```

---

<sup>1</sup> The above translation rule replaces the number “30xx” with “40xx”

<sup>2</sup> This defines the translation profile for the called number. In this particular example, the called number “30xx” is converted to “40xx”.



```
control-plane
!
!
voice-port 0/1/0
!
voice-port 0/1/1
!
voice-port 1/0/0:1
!
!
dial-peer voice 4000 pots
translation-profile incoming PSTN-to-SIP
destination-pattern 4...
incoming called-number 1...
port 1/0/0:1
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
!
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
```

<sup>3</sup> 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).

<sup>4</sup> 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.

<sup>5</sup> 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.

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

<sup>7</sup> Also changed to G.711A, G.729, and G.729B during testing.



## Acronyms

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

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