



Avaya S8500 Communications Manager 4.0 to Cisco IOS Voice Gateway using H.323 with E1 NET5

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Introduction

- This Application note provides basic call interoperability and documented steps and configurations necessary for H.323 integration between Avaya S8500 Communications Manager (CM) 4.0 to Cisco IOS Voice Gateway providing E1 NET5 PSTN connectivity.
- The H323 protocol is used between Cisco IOS Voice gateway and Avaya S8500 CM 4.0. The connection between Cisco IOS Voice Gateway and PSTN uses E1-PRI with switch-type NET5 protocol.
- Features tested include Basic call, Call Transfer supervised, Call Transfer blind, Call Forward (All, Busy and No Answer), Three-way Conference, DTMF tones, Caller ID functionality between Avaya S8500 CM 4.0 users and the PSTN.
- The Cisco IOS Voice Gateway offers the advantage of providing connectivity between Avaya S8500 CM 4.0 and PSTN by offering H323 to ISDN 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 CM 4.0 and connected to the PSTN via E1 NET5 ISDN.
- This Application Notes uses the Cisco IAD2432 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 IAD2430 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

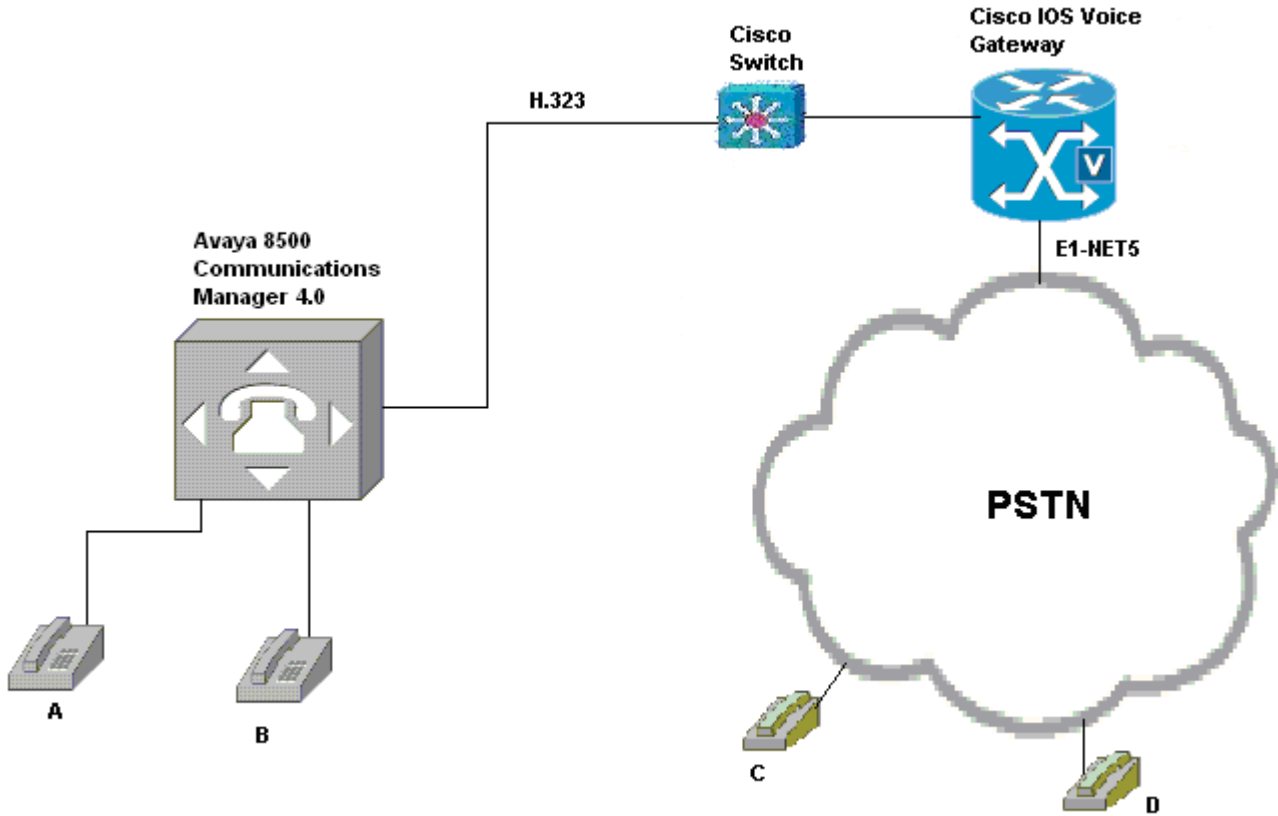


Figure 1. Network Topology



Limitations

- Basic call using G.726A – 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.

System Components

Hardware requirements

Cisco equipment

- Cisco 2432 (Cisco IOS Voice Gateway)
- Cisco 3560 powered Ethernet switch

Avaya equipment

- Avaya S8500
- TN2312BP IPSI
- TN799DP C-LAN
- TN2302AP IP Media Processor
- TN746B Analog
- TN2224B 2-wire Digital
- 2 - Digital stations 8410D

Software Requirements

- IOS Gateway: Cisco IOS Release – Cisco IAD2432: c2430-ik9o3s-mz.124-15.T3.bin
- Avaya S8500: Communications Manager 4.0

Features

Features Supported

- Basic Calls with G.711alaw, G.711ulaw, G.723 (5.3K and 6.3K), G.729(A and B)
- Calling name and Calling number
- Call Transfer Blind and Call Transfer Supervised
- Call Conference
- Caller ID restriction
- Call Forward All
- Call Forward No Reply
- Call Forward Busy
- DTMF using RFC2833
- DTMF in-band
- DTMF using H245 Alphanumeric or Signal
- Digit Translation



Features Not Supported

- G.726 codec
- G.722 codec



Configuration

Configuring Avaya S8500 CM 4.0

Signaling group (In band)

change signaling-group 16 send (return) help (F5) cancel (esc) enter (F3) schedule (F9) next (F7) previous (F8) next form (F6)

1 2 3 4 5

SIGNALING GROUP

Group Number: 16 **Group Type:** h.323

Remote Office? n **Max number of NCA TSC:** 10

SBS? n **Max number of CA TSC:** 10

IP Video? n **Trunk Group for NCA TSC:** 16

Trunk Group for Channel Selection: 16

TSC Supplementary Service Protocol: a

T303 Timer(sec): 10

Near-end Node Name: clan1 **Far-end Node Name:** CecilyGW

Near-end Listen Port: 1720 **Far-end Listen Port:** 1720

Far-end Network Region: 1

LRQ Required? n **Calls Share IP Signaling Connection?** n

RRQ Required? n **H245 Control Addr On FACility?** n

Media Encryption? n **Bypass If IP Threshold Exceeded?** n

DTMF over IP: in-band-q711 **H.235 Annex H Required?** n

Link Loss Delay Timer(sec): 90 **Direct IP-IP Audio Connections?** n

Enable Layer 3 Test? n **IP Audio Hairpinning?** n

Interworking Message: PROGRESS

DCP/Analog Bearer Capability: 3.1kHz

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

Select an entry from the list



Signaling group (RFC 2833)

change signaling-group 16 send (return) help (F5) cancel (esc) enter (F3) schedule (F9) next (F7) previous (F8) next form (F6)

1 | 2 | 3 | 4 | 5

SIGNALING GROUP

Group Number: 16 **Group Type:** h.323

Remote Office? n **Max number of NCA TSC:** 10

SBS? n **Max number of CA TSC:** 10

IP Video? n **Trunk Group for NCA TSC:** 16

Trunk Group for Channel Selection: 16

TSC Supplementary Service Protocol: a

T303 Timer(sec): 10

Near-end Node Name: clan1 **Far-end Node Name:** CecilyGW

Near-end Listen Port: 1720 **Far-end Listen Port:** 1720

Far-end Network Region: 1

LRQ Required? n **Calls Share IP Signaling Connection?** n

RRQ Required? n **H245 Control Addr On FACility?** n

Media Encryption? n **Bypass If IP Threshold Exceeded?** n

DTMF over IP: rtp-payload **H.235 Annex H Required?** n

Link Loss Delay Timer(sec): 90 **Direct IP-IP Audio Connections?** n

Enable Layer 3 Test? n **IP Audio Hairpinning?** n

Interworking Message: PROGRESS

DCP/Analog Bearer Capability: 3.1kHz

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

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



Signaling group (H245 Alphanumeric)

change signaling-group 16 send (return) help (f5) cancel (esc) enter (f3) schedule (f9) next (f7) previous (f8) next form (f6)

1 | 2 | 3 | 4 | 5

SIGNALING GROUP

Group Number: 16 **Group Type:** h.323

Remote Office? n **Max number of NCA TSC:** 10

SBS? n **Max number of CA TSC:** 10

IP Video? n **Trunk Group for NCA TSC:** 16

Trunk Group for Channel Selection: 16

TSC Supplementary Service Protocol: a

T303 Timer(sec): 10

Near-end Node Name: c1an1 **Far-end Node Name:** CecilyGW

Near-end Listen Port: 1720 **Far-end Listen Port:** 1720

Far-end Network Region: 1

LRQ Required? n **Calls Share IP Signaling Connection?** n

RRQ Required? n **H245 Control Addr On FACility?** n

Media Encryption? n **Bypass If IP Threshold Exceeded?** n

DTMF over IP: out-of-band **H.235 Annex H Required?** n

Link Loss Delay Timer(sec): 90 **Direct IP-IP Audio Connections?** n

Enable Layer 3 Test? n **IP Audio Hairpinning?** n

Interworking Message: PROGRESS

DCP/Analog Bearer Capability: 3.1kHz

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

Select an entry from the list



Trunk group – p1 of 3

The screenshot displays the Cisco Unified Communications Manager (CUCM) configuration page for Trunk Group 16. The interface includes a top navigation bar with various icons and a search field containing 'SIP1'. Below this is a toolbar with function keys: 'send (return)', 'help (F5)', 'cancel (esc)', 'enter (F3)', 'schedule (F9)', 'next (F7)', 'previous (F8)', and 'next form (F6)'. A numeric keypad (1-21) is visible at the top of the main content area.

TRUNK GROUP

Group Number: 16 **Group Type: isdn** **CDR Reports: y**
Group Name: H.323 trunk to CecilyGW **COR: 1** **TN: 1** **TAC: 816**
Direction: two-way **Outgoing Display? n** **Carrier Medium: H.323**
Dial Access? y **Busy Threshold: 255** **Night Service:**
Queue Length: 0
Service Type: tie **Auth Code? n**
Member Assignment Method: auto
Signaling Group: 16
Number of Members: 3

The left sidebar contains a 'General' section with the following options: 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 of the sidebar are 'Advanced', 'Fault & Performance', 'Tasks', and 'Tree' buttons.



Trunk group – p3 of 3

display trunk-group 16 send (return) help (f5) cancel (esc) enter (f3) schedule (f9) next (f7) previous (f8) next form (f6)

1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21
---	---	---	---	---	---	---	---	---	----	----	----	----	----	----	----	----	----	----	----	----

TRUNK FEATURES

ACA Assignment? n	Measured: none
Internal Alert? n	Maintenance Tests? y
Data Restriction? n	NCA-TSC Trunk Member: 1
Send Name: y	Send Calling Number: y
Send EMU Visitor CPN? n	
Used for DCS? n	Format: unknown
Suppress # Outpulsing? n	UII IE Treatment: service-provider
	Replace Restricted Numbers? y
	Replace Unavailable Numbers? y
	Send Connected Number: y
	Hold/Unhold Notifications? n
	Modify Tandem Calling Number? n
Send UII IE? y	
Send UCID? n	
Send Codeset 6/7 LAI IE? y	

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



Node-names IP

The screenshot shows a Cisco configuration window titled "Node-names IP". The window has a menu bar with "SIP1" and a toolbar with various icons. Below the toolbar is a navigation bar with buttons: "send (return)", "help (F5)", "cancel (esc)", "enter (F3)", "schedule (F9)", "next (F7)", "previous (F8)", and "next form (F6)".

The main content area is divided into two sections. The left section is a blue sidebar with a "General" tab selected. It contains a list of tasks: "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". Below this list are tabs for "Advanced" and "Fault & Performance". At the bottom of the sidebar are "Tasks" and "Tree" buttons.

The right section displays a table titled "IP NODE NAMES". The table has two columns: "Name" and "IP Address".

Name	IP Address
CCM4.1	172.20.231.254
CCM4.1.2	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
avayasip1	172.20.212.254
clan1	172.20.212.253
clan1serverb	172.20.213.253
default	0.0.0.0
ipigw	172.20.192.102
medpro1	172.20.212.252
nortelcs1000	172.20.216.100
procr	172.20.212.200

Below the table, there is a message: "(14 of 14 administered node-names were displayed)". Below this message are two lines of text: "Use 'list node-names' command to see all the administered node-names" and "Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name".



IP Network Region

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

1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17 18 19

IP NETWORK REGION

Region: 1
Location: 1 Authoritative Domain: lab.com
Name: CiscoLAB

MEDIA PARAMETERS Intra-region IP-IP Direct Audio: no
Codec Set: 1 Inter-region IP-IP Direct Audio: no
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
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

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



IP codec set

change ip-codec-set 1 send (return) help (F5) cancel (esc) enter (F3) schedule (F9) next (F7) previous (F8) next form (F6)

1 | 2

IP Codec Set

Codec Set: 1

Audio Codec	Silence Suppression	Frames Per Pkt	Packet Size(ms)
1: G.711MU	<input checked="" type="checkbox"/>	<input type="text" value="2"/>	<input type="text" value="20"/>
2: G.711MU	<input type="checkbox"/>	<input type="text"/>	<input type="text"/>
3: G.722-64K	<input type="checkbox"/>	<input type="text"/>	<input type="text"/>
4: G.722.1-24K	<input type="checkbox"/>	<input type="text"/>	<input type="text"/>
5: G.722.1-32K	<input type="checkbox"/>	<input type="text"/>	<input type="text"/>
6: G.723-5.3K	<input type="checkbox"/>	<input type="text"/>	<input type="text"/>
7: G.723-6.3K	<input type="checkbox"/>	<input type="text"/>	<input type="text"/>
G.726A-32K	<input type="checkbox"/>	<input type="text"/>	<input type="text"/>
G.729	<input type="checkbox"/>	<input type="text"/>	<input type="text"/>
G.729A	<input type="checkbox"/>	<input type="text"/>	<input type="text"/>
G.729B	<input type="checkbox"/>	<input type="text"/>	<input type="text"/>

1: none
2:
3:



Uniform dialing plan

The screenshot shows the Cisco SIP1 configuration interface for a Uniform Dialing Plan. The main window displays a table titled "UNIFORM DIAL PLAN TABLE" with a "Percent Full: 0" indicator. The table lists various matching patterns and their associated configurations. The interface includes a left-hand menu with options like "Start GEDI", "Add User", and "Browse Dial Ranges". At the top, there are navigation buttons such as "send (return)", "help (f5)", and "cancel (esc)".

Matching Pattern	Len	Del	Insert Digits	Net	Conv	Node Num
23	4	0	216	aar	n	
30	4	0	207	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	
3090	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	
5000	4	0	224	aar	n	
5001	4	0	224	aar	n	



Route Pattern

display route-pattern 16 send (return) help (f5) cancel (esc) enter (f3) schedule (f9) next (f7) previous (f8) next form (f6)

1 | 2 | 3

Pattern Number: 16 Pattern Name:
 SCCAN? n Secure SIP? n

Grp No	FRL	NPA	Pfx	Hop	Toll	No.	Inserted	DCS/	IXC
			Mrk	Lnt	List	De1	Digits	QSIG	
							Dgts	Intw	
1:	16	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	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

Advanced
 Fault & Performance

Tasks Tree



Cisco IOS configuration

Cisco 2432 voice gateway configuration

```
E1-NET5#sh ver
Cisco IOS Software, 2400 Software (C2430-IK9O3S-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 Thu 24-Jan-08 14:04 by prod_rel_team
```

ROM: System Bootstrap, Version 12.3(7r)T2, RELEASE SOFTWARE (fc1)

```
E1-NET5 uptime is 1 day, 23 hours, 10 minutes
System returned to ROM by reload
System image file is "flash:c2430-ik9o3s-mz.124-15.T3.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 IAD2432 (R527x) processor (revision 4.1) with 119808K/11264K bytes of memo
ry.
Processor board ID FHK1013F23V
R527x CPU at 225MHz, Implementation 40, Rev 3.1
1 On-Board Twenty-Four FXS Analog Voice Module V1.3
2 FastEthernet interfaces
11 Serial interfaces
2 Channelized E1/PRI ports
1 Virtual Private Network (VPN) Module
DRAM configuration is 64 bits wide with parity disabled.
63K bytes of non-volatile configuration memory.
System fpga version is 250027
System readonly fpga version is 250025
Option for system fpga is 'system'.
62720K bytes of ATA System CompactFlash (Read/Write)
62720K bytes of ATA Slot0 CompactFlash (Read/Write)
```

Configuration register is 0x2102

```
=====
E1-NET5#sh run
Building configuration...
```

```
Current configuration : 2405 bytes
!
```



```
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname E1-NET5
!
boot-start-marker
boot system flash:c2430-ik9o3s-mz.124-15.T3.bin
boot-end-marker
!
card type e1 11
logging buffered 1000000
!
no aaa new-model
network-clock-participate E1 1/0
network-clock-participate E1 1/1
!
!
no ip domain lookup
!
!
ip auth-proxy max-nodata-conns 3
ip admission max-nodata-conns 3
!
multilink bundle-name authenticated
isdn switch-type primary-net5
isdn gateway-max-interworking
!
voice-card 0
!
!
voice rtp send-recv
!
voice service voip
h323
!
!
voice translation-rule 12
rule 1 /23/ /20\1/
!
!
voice translation-profile rule13
translate called 1
!
!
archive
log config
hidekeys
!
!
controller E1 1/0
pri-group timeslots 1-10,16
!
```

¹ This sets or changes the card type (eg E1/T1)

² This individual rule translate the number that starts with “23” and replaces the first two digits with “20”, then copy the rest of the dialed digits.

³ The voice translation rules are applied to voice translation profile, which can then applied to dial peers.



```
controller E1 1/1
!
!
interface FastEthernet0/0
no ip address
shutdown
duplex half
speed 100
!
interface FastEthernet0/1
ip address 172.20.174.40 255.255.255.0
duplex auto
speed auto
!
interface Serial1/0:15
no ip address
encapsulation hdlc
isdn switch-type primary-net54
isdn timer T310 10000
isdn incoming-voice voice
isdn supp-service name calling
isdn send-alerting
isdn bchan-number-order ascending
no cdp enable
!
ip default-gateway 172.20.174.1
ip http server
no ip http secure-server
!
ip forward-protocol nd
ip route 0.0.0.0 0.0.0.0 172.20.174.1
!
!
control-plane
!
!
!
voice-port 1/0:15
!
!
dial-peer voice 2000 pots
destination-pattern 20..
direct-inward-dial
port 1/0:15
forward-digits all
!
!
!
dial-peer voice 4100 voip
translation-profile incoming rule15
destination-pattern 41..
rtp payload-type nte 127
session target ipv4:172.20.212.253
incoming called-number 23..
```

⁴ Specify E1-NET5 interface.

⁵ This specifies a digit string that can be matched by an incoming call to associate the call with a dial peer.



```
dtmf-relay rtp-nte6
codec g729br87
!
!
gateway
timer receive-rtp 1200
!
sip-ua
no remote-party-id
!
!
line con 0
exec-timeout 0 0
password cisco
login
line aux 0
line vty 0 4
exec-timeout 0 0
password cisco
login
!
end
```

⁶ Specify DTMF parameter here. This command is for testing DTMF RFC2833; removing this command will do DTMF in-band. NOTE: Insert "dtmf-relay h245-alphanumeric h245-signal", and remove "dtmf-relay rtp-nte" to test DTMF for H245/Out-of-band signaling when using H323 trunk.

⁷ Specify CODEC here when testing CODEC. Also change CODEC settings at the PBX end to match the specified codec at the IOS Media Gateway



Acronyms

Acronym	Definitions
Cisco IOS	Cisco Internetwork Operating System
PSTN	Public switched Telephone Network

Important Information

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Test results

	Test Case	Description	Test Result
1	Phone A calls Phone C. Note: If IP PBX supports more than one codec (e.g. G729, G723..etc) test each supported codec and make separate results for each codec under "Test Results"	Phone A hears ring-back Ensure voice path, in both directions, is achieved when Phone C is answered Verify Calling Name and number are presented to called phone Verify call is disconnected at both ends when phone is hung-up	G.711alaw: PASS G.711ulaw: PASS G.722-64K: FAIL G.723-5.3K: PASS G.723-6.3K: PASS G.726-32K: FAIL G.729: PASS G.729B: PASS
2	Phone D calls Phone B. Note: If IP PBX supports more than one codec (e.g. G729, G723..etc) test each supported codec and make separate results for each codec under "Test Results"	Phone D hears ring-back Ensure voice path, in both directions, is achieved when Phone B is answered Verify Calling Name and number are presented to called phone Verify call is disconnected at both ends when phone is hung-up	G.711alaw: PASS G.711ulaw: PASS G.722-64K: FAIL G.723-5.3K: PASS G.723-6.3K: PASS G.726-32K: FAIL G.729: PASS G.729B: PASS
3	Phone A calls Phone C, Phone A hangs-up before Phone C answers	Ensure call is released and Phone C stops ringing	PASS
4	Phone C calls Phone A, Phone C hangs-up before Phone A answers	Ensure call is released and Phone A stops ringing	PASS
5	Call Transfer – Blind Phone D calls Phone B, Phone B transfers to Phone C before phone C answers the call.	Ensure call is transferred to Phone C and Calling number/name are updated correctly.	PASS – Calling name/number are not updated correctly. Phone C display Phone B name and number.
6	Call Transfer – Blind Phone D calls Phone B, Phone B transfers to Phone A before phone A answers the call.	Ensure call is transferred to Phone A and Calling number/name are updated correctly.	PASS – Calling name/number are updated correctly.
7	Call Transfer – Blind Phone A calls Phone B, Phone B transfers to Phone C before phone C answers the call.	Ensure call is transferred to Phone C and Calling number/name are updated correctly.	PASS - Calling name/number are not updated correctly. Phone C display Phone B name and number.
8	Call Transfer – Supervised Phone D calls Phone B, Phone B transfers to Phone C after phone C answers the call.	Ensure call is transferred to Phone C and Calling number/name are updated correctly.	PASS - Calling name/number are not updated correctly. Phone C display Phone B name and number.
9	Call Transfer – Supervised Phone D calls Phone B; Phone B transfers to Phone A after phone A answers the call.	Ensure call is transferred to Phone A and Calling number/name are updated correctly.	PASS – Calling name/number are updated correctly.
10	Call Transfer – Supervised Phone A calls Phone B, Phone B transfers to Phone C after	Ensure call is transferred to Phone C and Calling number/name are updated correctly.	PASS – Calling name/name are not updated correctly. Phone C



	Test Case	Description	Test Result
	phone C answers the call.		display Phone B name and number.
11	Call Conference – Using phone conference button. Phone C calls phone A, phone A calls phone D. When phone D answers, activate “conference” on phone A	Ensure all three phones (Phone A, C and D) are in a conference call. Voice path is established all three ways at each phone.	PASS
12	Call Conference – Phone D leaves call	Ensure Phones C and A remain on call	PASS
13	Call Conference – Phone C leaves call	Ensure Phone A and D remain on call	PASS
14	Call Conference – Phone A leaves call	Ensure Phone C and D remain on call	PASS
15	Call Conference – Using phone conference button Phone C calls phone A, phone A calls phone B. When phone B answers, activate “conference” on phone A	Ensure all three phones (Phone A, B and C) are in a conference call. Voice path is established all three ways at each phone.	PASS
16	Call Conference – Phone B leaves call	Ensure Phones C and A remain on call	PASS
17	Call Conference – Phone C leaves call	Ensure Phone A and B remain on call	PASS
18	Call Conference – Phone A leaves call	Ensure Phone B and C remain on call	PASS
19	Call on Hold Phone A calls Phone C, press the “Hold” button on phone A.	Ensure established call between phone A and phone C is not dropped during the time the call is placed on hold, and call resumes normally after removing call on hold.	PASS
20	Caller ID restricted Restrict Calling number and name on phone A. Place a call from phone A to phone C, answer phone C	Ensure the display on phone C does not display phone A’s number. Must be restricted.	PASS⁸
21	Caller ID restricted Restrict Calling number and name on phone C. Place a call from phone C to phone A, answer phone A	Ensure the display on phone A does not display phone C’s number. Must be restricted	PASS
22	Call Forward – All	Ensure the call is forwarded	PASS – Calling name/number are

⁸ Change Avaya station display restricted to **blank or yes**, and change the trunk group number to restrict.



	Test Case	Description	Test Result
	Place phone A to forward all calls to phone D. Place a call from phone C to phone A.	immediately to phone D and the call is established successfully between phone C and phone D. Calling name/number shall be updated correctly	updated correctly.
23	Call Forward – All Place phone A to forward all calls to phone B. Place a call from phone D to phone A.	Ensure the call is forwarded immediately to phone B and the call is established successfully between phone D and phone B. Calling name/number shall be updated correctly	PASS – Calling name is updated correctly (No Calling number being display).
24	Call Forward – All Place phone A to forward all calls to phone D. Place a call from phone B to phone A.	Ensure the call is forwarded immediately to phone D and the call is established successfully between phone B and phone D. Calling name/number shall be updated correctly	PASS – Calling name/number are updated correctly.
25	Call Forward – No Reply Place phone A to forward calls on no reply to phone D. Place a call from phone C to phone A.	Ensure the call is forwarded to phone D, after phone A rings a predetermined number of times with no answer. The call shall be established successfully between phone C and phone D. Calling name/number shall be updated correctly	PASS – Calling name/number are updated correctly.
26	Call Forward – No Reply Place phone A to forward calls on no reply to phone B. Place a call from phone D to phone A.	Ensure the call is forwarded to phone B, after phone A rings a predetermined number of times with no answer. The call shall be established successfully between phone D and phone B. Calling name/number shall be updated correctly	PASS – Calling name is updated correctly. (No Calling number being display).
27	Call Forward – No Reply Place phone B to forward calls on no reply to phone D. Place a call from phone A to phone B.	Ensure the call is forwarded to phone D, after phone B rings a predetermined number of times with no answer. The call shall be established successfully between phone A and phone D. Calling name/number shall be updated correctly	PASS – Calling name/number are updated correctly.
28	Call Forward – Busy Place phone A to forward calls on busy to phone D. Place a call from phone B to phone A, then place a call from phone C to phone A.	Ensure the second call is forwarded to phone D, on busy. The call shall be established successfully between phone C and phone D. Calling name/number and connected name/number shall be updated correctly	PASS – Calling name/number are updated correctly.
29	Call Forward – Busy Place phone A to forward calls on busy to phone B. Place a call from phone D to phone A, then place a call from phone C to phone A.	Ensure the second call is forwarded to phone B, on busy. The call shall be established successfully between phone C and phone B. Calling name/number shall be updated correctly	PASS – Calling name displayed correctly on Phone B (No Calling number.)
30	Call Forward – Busy Place phone B to forward calls on busy to phone C. Place a call from phone D to phone B, then place a call from phone A to phone B.	Ensure the second call is forwarded to phone D, on busy. The call shall be established successfully between phone A and phone C. Calling name/number shall be updated correctly	PASS – Calling name/number are updated correctly.



	Test Case	Description	Test Result
31	DTMF Phone A calls Phone C. Phone C calls Phone A. RFC2833 and/or H245- (Alphanumeric or Signal) to G711 and vice-versa	Run independent tests from both ends to ensure DTMF tones arrive at both ends correctly.	In-band: PASS H245/Out-of-band signaling: PASS RFC2833: PASS
32	Digit translation – The voice gateway can manipulate the digits of the called 10-digit number sent by PSTN and 4 digit number sent by PBX	Ensure the voice gateway is capable of manipulating the called and calling digits to match configured dial-peers and route calls appropriately	PASS



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