

Avaya S8500 CM 3.0 using E1 ISO-QSIG to Cisco Unified CallManager Express 4.0(2)

November 1, 2007 Version 2

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Introduction

This is an Application Note for connectivity between an Avaya S8500 Communications Manager Release 3.0 PBX and Cisco Unified CallManager Express Release 4.0(2) using a Cisco 3845 voice gateway with QSIG protocol.

Voice mail testing was performed with an Octel 200 (S.4.1) using QSIG integration (E1-DTIC).

The network topology diagram (Figure 1) shows the test setup for end-to-end interoperability with Cisco Unified CallManager Express Release 4.0(2) connected to the PBX via the 3845 E1 QSIG link. The 3845 IOS voice gateway was connected via H.323 to a Cisco 2801 Cisco IOS voice gateway. The two gateways were running Cisco Unified CallManager Express 4.0(2). Cisco Unified IP phones (models 7960 and 7961G) were connected to the 2 Cisco Unified CallManager Express gateways via SIP and SCCP, as per the figure. A NM-HDV and VWIC-2MFT-E1 was used for the E1 QSIG interface. Calls were made to test basic call, caller ID, conference, transfer, forward, call back, reroute, and MWI features.

This Application Note uses the Cisco 3845 voice gateway. However, the use of other Cisco voice gateways is also an option since Cisco Unified Call Manager Express does not depend on platform. The listed gateway families, below, can run Cisco Unified CallManager Express, but each have different IP phone support capability. Please check the product specifications to ensure you are obtaining the proper device to support your IP phone deployment.

Cisco IAD 2430 Series Integrated Access Devices

Cisco 2801 Integrated Services Router, 1760-V and 1751-V Access Routers

Cisco 2811 Integrated Services Router, 261xXM and 262xXM Series Access Routers

Cisco 2821 Integrated Services Router, 265xXM Access Router

Cisco 2691 Multiservice Access Router

Cisco 2851 Integrated Services Router

Cisco 3725 Multiservice Access Router



Cisco 3745 Multiservice Access Router

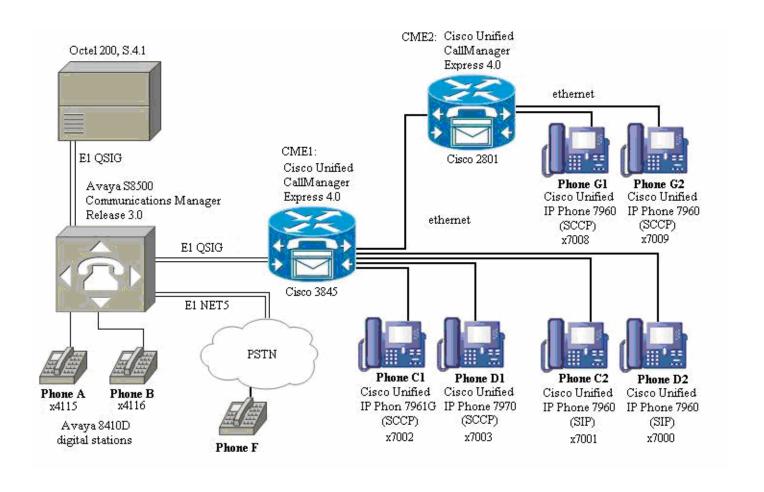
Cisco 3825 Integrated Services Router

Cisco 3845 Integrated Services Router

The inclusion of Cisco SIP phones in this application note is for reference only. Cisco Unified Communications Manager Express 4.0(3) supports SIP end-points with limited number of features.

Network Topology

Figure 1. Network Topology or Test Setup – basic calls configuration.





Limitations

Basic Calls

Cisco Unified CallManager Express does not support overlap sending. It supports overlap receiving.

Connected Name and Alerting Name are not supported on calls between PBX and Cisco Unified IP Phone running SIP. This is a CME SIP limitation

Calling Name Restriction is not supported for calls originated from Cisco Unified CallManager Express 4.0(2).

Connected Number/Name Restriction is not supported from Cisco Unified CallManager Express 4.0(2).

Call Transfers

A call-transfer (consult, early-attended or blind) originating from a call placed from a phone on the remote Cisco Unified CallManager Express (CME2) to a SIP phone on the local Cisco Unified CallManager Express (CME1), and then transferred to a PBX phone (e.g., G1 calls C2, and C2 transfers to A) does not complete. This is a CME SIP limitation.

For local consulted and early-attended call transfers between SCCP phones and SIP phones, call originates from an external PBX phone, the Calling name and number updates are not supported. This is CME SIP to SCCP interworking limitation.

For local consulted, early attended and blind call transfers connected name and number are not supported. CME dos not support Facility IE for call update information.

For external consulted and early attended call transfers with call flow, (CME IP phone calls PBX phone, PBX phone transfers back to different IP phone on CME (trombone)) the called (connected) name and number are not updated on the original phone after the transfer is complete. (e.g. Phone C1 calls Phone A, Phone A transfers to Phone D1). CME dos not support Facility IE for call update information.

Call Forwards

For local call forward calls involving SIP phones the forwarding name/number display is not supported. This is a CME SIP limitation.

For external Call forward calls the forwarding number is not supported on CME. CME does not support RedirectingName.

For external call forward calls, the called (connected) number is not updated on the original phone. This is a CME limitation.

For call forward calls, the called (connected) name is not updated on the SIP phone. This is a CME SIP limitation.

Forwarded calls originated from a PBX extension to a remote Cisco Unified CallManager Express SCCP extension, and forwarded to a local Cisco Unified CallManager Express extension (e.g., A calls G1, and G1 forwards to C2), Cisco Unified CallManager Express performs a QSIG reroute, even though a QSIG reroute is not in order (i.e., there is no QSIG "hairpin" or "trombone"). This is a CME limitation

Forwarded calls hairpinned at a SIP extension (PBX phone calls Cisco Unified CallManager Express 4.0(2) SIP phone that forwards back to another PBX phone), the call completes, but Cisco Unified CallManager Express 4.0(3) does not perform a reroute, even if reroute is enabled. CME SIP limitation

Forwarded calls originated from a PBX extension to a local Cisco Unified CallManager Express SCCP extension, and forwarded to another local Cisco Unified CallManager Express extension (e.g., A calls C1, and C1 forwards to D1 or D2), Cisco Unified CallManager Express performs a reroute, and even though a reroute is not in order (i.e., there is no "hairpin" or "trombone").

For calls that are hairpinned at a SIP extension (PBX phone calls Cisco Unified CallManager Express 4.0(2) SIP phone that forwards unconditionally back to another PBX phone) when a CFNR number was set up resulted in a 3rd SETUP message from CME. The timeout is set under the CFNR command. If enough time passes before the final destination (B) answers, the CFNR is invoked, and the 3rd SETUP is sent from CME. A new (3rd) B-chan is set up. The 2nd one is then torn down.

Forwarded "trombone" (or "hairpin") calls originated from a PBX extension to a Cisco Unified CallManager Express 4.0(2) extension, and forwarded back to another PBX extension (e.g, A calls C1, C2, or G1, which forwards to B), "joined" calls (i.e., no Reroute or Path Replacement) could not be performed, because the PBX initiates Path Replacement after the call is joined. This feature can not be turned off. The only exception is when the forwarding is unconditional (CFU) and the forwarding phone is a SIP phone (e.g., C2). Then, there is not enough information in the 2nd SETUP message for the PBX to recognize it as a forwarded call, so there is no Path Replacement proposal, and the call is "joined". There are 2 B-channels in use. However, if CFNR is configured and enough time passes before the final destination answers for CFNR to be invoked, Cisco Unified CallManager Express 4.0(2) sends an additional (3rd)



SETUP message. A new (3rd) B-chan is set up, and the 2nd one is then torn down, following the scenario in the previous bullet. This 3rd SETUP message does have the call fwd diverting leg info. and Path Replacement does occur.

Forwarded calls that are initiated by overlap dialing from a PBX extension to a Cisco Unified CallManager Express 4.0(2) extension, the call completes, but Cisco Unified CallManager Express does not perform a reroute, even if reroute is enabled and the call is eligible for a reroute.

MWI

Cisco Unified Communications Manager Express 4.0(2) supports Cisco Unity integration with QSIG. However, in this instance, no testing was performed with Cisco Unified Communications Manager Express 4.0(3) as the message center PINX.

MWI was not tested for SIP extensions on Cisco Unified CallManager Express 4.0(3) with the PBX as the message center PINX. It was tested for SCCP extensions only.



System Components

Hardware Requirements

Cisco 3845 IOS voice gateway

NM-HDV

VWIC-2MFT-E1

Cisco 2801 IOS voice gateway

- (4) Cisco Unified IP phone 7960s
- (2) Cisco Unified IP phone 7961G
- (1) Avaya S8500 PBX
- (2) Avaya 8410D digital station phones
- (1) TN464F E1 trunk card (for PSTN link)
- (1) TN464GP E1 trunk card (for QSIG trunk)
- (1) Octel 200 voice mail system
 - (2) E1-DTIC

Software Requirements

Cisco Unified CallManager Express Release 4.0(2)

Cisco IOS Software, 3800 Software (C3845-IPVOICE-M), Version 12.4(11)T

Cisco IOS Software, 2801 Software (C2801-IPVOICE-M), Version 12.4(11)T

Avaya Communications Manager Release 3.0

Octel S.4.1 voice mail

G1, G2 - 7960 - SCCP

Cisco7960 IP phone version 7.2(T0.23)

Cisco 7960 IP phone app load P00308000400

Cisco 7960 IP phone boot load PC0303010001

C2, D2 - 7960 - SIP

Cisco7960 DSP load ID 4.0(2.0)[A0]

Cisco 7960 IP phone app load P0S3-08-4-00

Cisco 7960 IP phone boot load PC030301

C1 - 7961G - SCCP

Cisco7961G IP phone load file: SCCP41.8-0-3S

Cisco 7961G IP phone app load ID: Jar41sccp.8-0-2-25.sbn

Cisco 7961G IP phone boot load ID: 7961G_64-020704128Amd64meg.bin

Cisco 7970 IP phone app load ID: jar70sccp.8-0-2.25.sbn

Cisco 7970 IP phone boot load ID: 7970_64060118.bin



Features

Features Supported

Basic Call, ENBLOC

Basic Call, Overlap (From PBX to Cisco Unified CallManager Express only)

CLIP-Calling Line (Number) Identification Presentation on Basic Calls

CLIR-Calling Line (Number) Identification Restriction on Basic Calls

CNIP-Calling Name Identification Presentation on Basic Calls

CNIR-Calling Name Identification Restriction on Basic Calls (From PBX to Cisco Unified CallManager Express only)

COLP-Connected Line (Number) Identification Presentation on Basic Calls

CONP-Connected Name Identification Presentation (for calls between PBX and Cisco Unified IP Phones running SCCP)

Alerting Name (for calls between PBX and Cisco Unified IP Phones running SCCP)

Tandem PSTN call

Consultation Transfer - Local

Consultation Transfer – Network/External (See Limitations Section)

Early Attended Transfer - Local

Early Attended Transfer - Network/External (See Limitations Section)

Blind Transfer - Local (See Limitations Section)

Blind Transfer - Network/External (See Limitations Section)

 $Call\ Forward\ Unconditional\ by\ Join-Local\ (See\ Limitations\ Section)$

Call Forward Unconditional by Join - Network/External (See Limitations Section)

Call Forward Busy by Join – Local (See Limitations Section)

Call Forward Busy by Join - Network/External (See Limitations Section)

Call Forward No Reply by Join – Local (See Limitations Section)

Call Forward No Reply by Join - Network/External (See Limitations Section)

Call Forward Unconditional by Reroute – Network/External (See Limitations Section)

Call Forward Busy by Reroute – Network/External (See Limitations Section)

Call Forward No Reply by Reroute – Network/External (See Limitations Section)

MWI (See Limitations Section)



Features Not Supported

Overlap dialing from Cisco Unified CallManager Express 4.0(3) to PBX

CNIR-Calling Name Identification Restriction from Cisco Unified CallManager Express 4.0(3) to PBX

COLR- Connected Line (Number) Identification Restriction

CONR- Connected Name Identification Restriction

CONP-Connected Name Identification Presentation (for calls between PBX and Cisco Unified IP Phones running SIP)

Alerting Name (for calls between PBX and Cisco Unified IP Phones running SIP)

Blind Transfers initiated from PBX

H323/QSIG tandem transfers via SIP phone

CLIP-Calling Line (Number) Identification Presentation on Transferred Calls

CNIP-Calling Name Identification Presentation on Transferred Calls

COLP-Connected Line (Number) Identification Presentation on Transferred Calls

CONP-Connected Name Identification Presentation on Transferred Calls

CLIP-Calling Line (Number) Identification Presentation on Forwarded Calls to a PBX station.

COLP-Connected Line (Number) Identification Presentation on Forwarded Calls

CONP-Connected Name Identification Presentation on Forwarded Calls

Call Forward by Reroute for QSIG "trombone" from a Cisco Unified CallManager Express SIP extension

Call Forward by Reroute with overlap dialing

Cisco Unity integration with QSIG.

MWI with QSIG/SIP interworking

Call Completion to Busy Subscriber (Call Back when Free)

Call Completion on No Reply (Call Back Next Used)

Path Replacement for Call Transfer by Join

Path Replacement for Trombone Connection

Path Replacement for Call Diversion by Forward Switch



Configuration

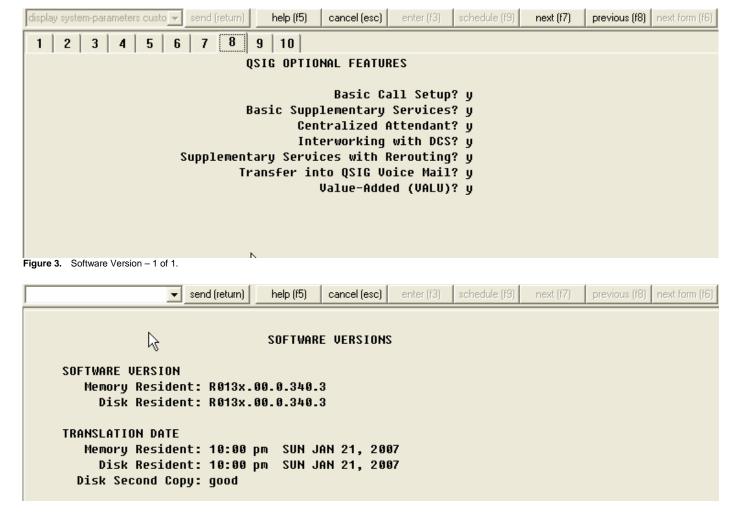
Configuration sequence for the Avaya S8500 Communications Manager 3.0 PBX

- 1. Check the system-parameter customer-option screen to insure the proper QSIG optional features are installed
- 2. Configure DS1 circuit pack.
- 3. Configure Signaling Group
- 4. Configure Trunk Group
- 5. Configure Route Pattern
- 6. Configure ISDN Public-Unknown numbering screen
- 7. Configure Uniform-Dialplan screen
- 8. Configure AAR analysis screen

Configuring the Avaya \$8500 Communications Manager 3.0

GLOBAL PARAMETERS

Figure 2. QSIG Options – 1 of 1.





CONFIGURATION FOR TRUNKS

Figure 4. Circuit Pack for E1-QSIG trunk to Cisco Unified CallManager Express – 1 of 1.

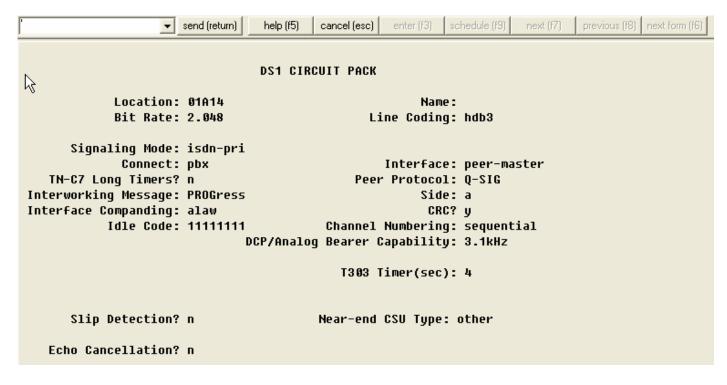




Figure 5. Trunk Group for E1-QSIG trunk to Cisco Unified CallManager Express – 1 of 3.

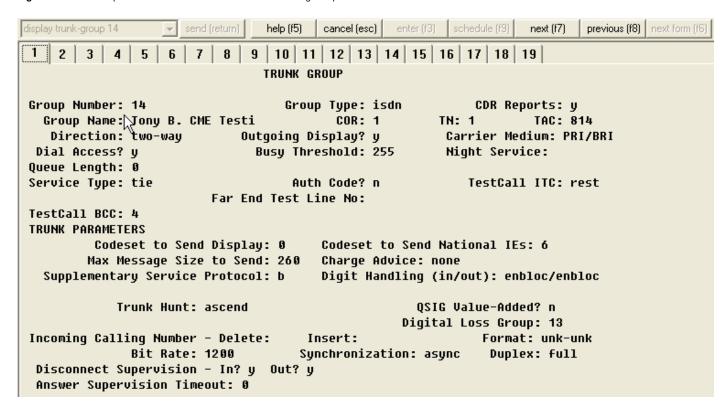




Figure 6. Trunk Group for E1-QSIG trunk to Cisco Unified CallManager Express – 2 of 3.

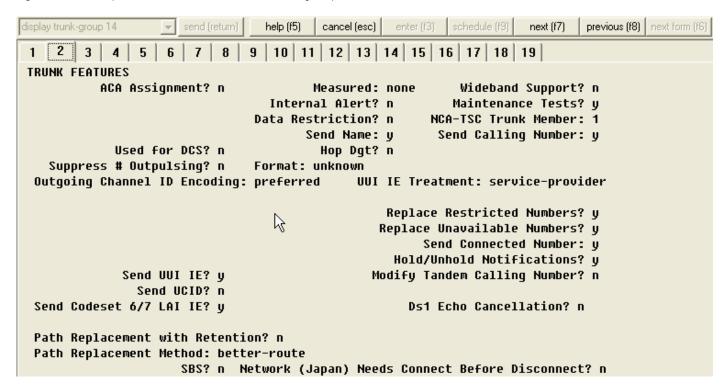


Figure 7. Trunk Group for E1-QSIG trunk to Cisco Unified CallManager Express – 3 of 3.

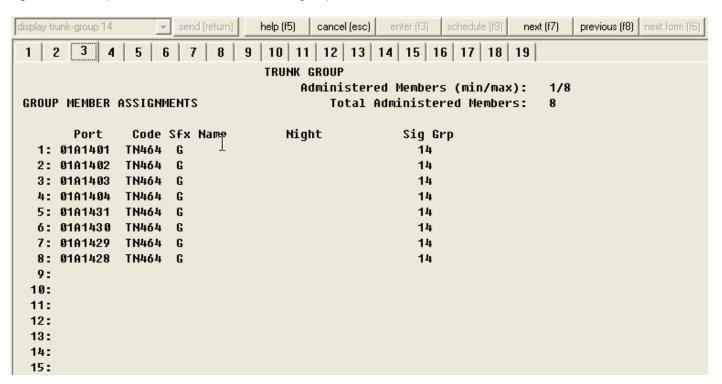




Figure 8. Signalling Group for E1-QSIG trunk to Cisco Unified CallManager Express – 1 of 1.

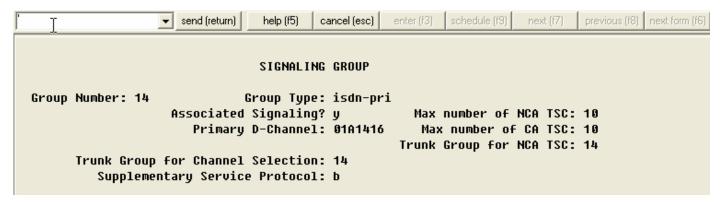


Figure 9. Circuit Pack for E1-NET5 trunk to PSTN – 1 of 1.

change ds1 1a13	send (return)	help (f5)	cancel (esc)	enter (f3)	schedule (f9)	next (f7)	previous (f8)	next form (f6)
1								
'		DS1 CIR	CUIT PACK					
Location: Bit Rate:			Li	Nam Ine Codin]	
Signaling Mode: Connect: TN-C7 Long Timers? Interworking Message: Interface Companding:	pbx n PROGress alaw		_	Interfac Protoco Ol Versio CR	1: 1			
Idle Code:		P/Analog		apabilit imer(sec	y: 3.1kHz): 4			
Slip Detection?	n		Near-end	CSU Type	: other			



Figure 10. Trunk Group for E1-NET5 trunk to PSTN – 1 of 3.

change trunk-group 13 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
TRUNK GROUP
Group Number: 1 Group Type: isdn CDR Reports: y Group Name: E1-ISDN COR: 1 TN: 1 TAC: 813 Direction: two-way Outgoing Display? y Carrier Medium: PRI/BRI Dial Access? y Busy Threshold: 255 Night Service:
Queue Length: 0 Service Type: tie Far End Test Line No: TestCall BCC: 4
TRUNK PARAMETERS Codeset to Send Display: 6 Codeset to Send National IEs: 6 Max Message Size to Send: 260 Charge Advice: none
Supplementary Service Protocol: C Digit Handling (in/out): enbloc/enbloc
Trunk Hunt: ascend QSIG Value-Added? n Digital Loss Group: 13
Incoming Calling Number - Delete: Insert: Format: unk-unk Bit Rate: 1200 Synchronization: async Duplex: Full Disconnect Supervision - In? y Out? y Answer Supervision Timeout: 0



Figure 11. Trunk Group for E1-NET5 trunk to PSTN – 2 of 3.

change trunk-group 13 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
TRUNK FEATURES
ACA Assignment? 🛮 Measured: none Wideband Support? 🗖
Internal Alert? n Maintenance Tests? y
Data Restriction? n NCA-TSC Trunk Member: 1
Send Name: [y] Send Calling Number: [y]
Used for DCS? n
Suppress # Outpulsing? n Format: unknown
Outgoing Channel ID Encoding: preferred UUI IE Treatment: service-provider
Desless Bestudeted Numbers D
Replace Restricted Numbers? U
Replace Unavailable Numbers? y Send Connected Number: y
Hold/Unhold Notifications? n
Send UUI IE? [v] Modify Tandem Calling Number? [v]
Send UCID? n
Send Codeset 6/7 LAI IE? U Ds1 Echo Cancellation?
SBS? n Network (Japan) Needs Connect Before Disconnect? n



Figure 12. Trunk Group for E1-NET5 trunk to PSTN – 3 of 3.

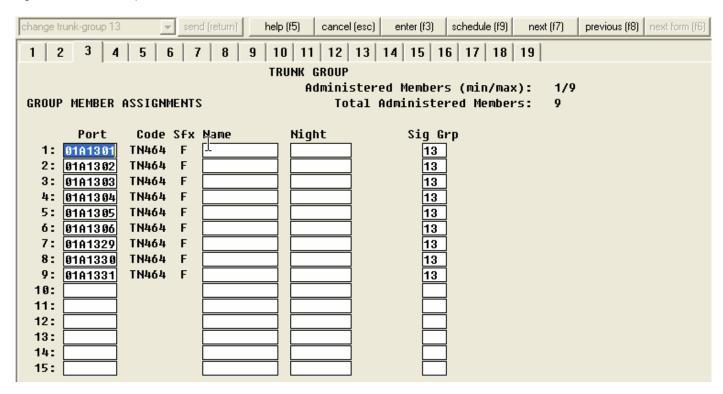
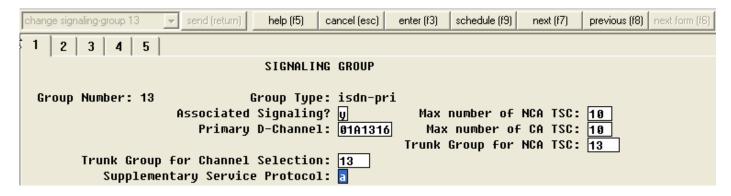


Figure 13. Signalling Group for E1-NET5 trunk to PSTN – 1 of 1.





DIAL PLANS AND ROUTE PATTERNS

Figure 14. Uniform Dial Plan – 1 of 1.

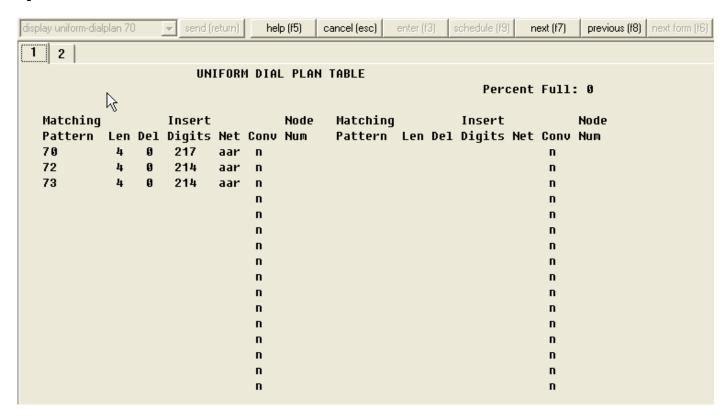




Figure 15. AAR Analysis – 1 of 1.

display aar analysis 217	send (return)	help (f5)	cancel (esc)	enter (f3)	schedule (f9)	next (f7)	previous (f8) next form (f6)
1 2							
	AAI	R DIGIT (ANALYSIS T	ABLE			
					Percent	Full:	2
Dialed	Total	L Ro	ute Cal	1 Node	ANI		
String	Min h	Max Pati	tern Typ	e Num	Reqd		
217	7 7	7 17	aar	•	n		
221	7 7	7 11	aar	4	n		
222	7 7	7 21	aar	•	n		
224	7 7	7 99	aar	•	n		
225	14 1	4 13	aar	•	n		
226	7 7	7 13	aar	•	n		
227	7 7	7 21	aar	•	n		
228	7 7	7 44	aar	•	n		
3	7 7	7 999	9 aar	•	n		
4	4 1	4 39	aar	•	n		
5	7 7	7 999	9 aar	•	n		
5 05 0	7 7	7 13	aar	•	n		
5554050	7 7	7 11	aar	•	n		
ó	7 7	7 999	9 aar	•	n		
7	7 7	7 999	9 aar	•	n		



Figure 16. Route Pattern for E1-QSIG trunk to Cisco Unified CallManager Express – 1 of 1.

display	route-pa	attern 1	7	V	send (return)	help (f5)	cancel (esc)	enter (f3)	sched	ule (f9)	next (f7)	previous (f8)	next form (f6)
1	2	3											
					Pattern N	lumber: 17	Pattern I	Name:					
		Ν				SCCAN? n		SIP? n					
	Grp	PISL I			•	No. Inser					DCS/		
	No			Mrk	Lmt List	_	:5				QSIG		
						Dgts					Intw		
	14	0				3					n	user	
2:											n	user	
3:											n	user	
4:											n	user	
5:											n	user	
6:											n	user	
	BCC	VALI	JE	TSC	CA-TSC	ITC BCIE	Service/Fo	eature B	BAND N	o. Nur	nbering	LAR	
	0 1	2 3 4	4 W		Request					ts For	_		
					•				Subad				
1:	y y	y y y	, n	y	as-needed	bothept				unk	r-unk	none	
2:	y y	y y i	, n	n		rest						none	
3:	yy	yy	, n	n		rest						none	
4:	y y	y y i	, n	n		rest						none	
5:	y y	yy	, n	n		rest						none	
6:	уy.	уý	, n	n		rest						none	



Figure 17. Route Pattern for E1-NET5 trunk to PSTN – 1 of 1.

ie i/.	Nout	Сгаш	5111 101 1	L I-INL I	5 HUIK TO PSTIN -	- 1 01 1.						
displa	y route-p	attern	13	7	send (return)	help (f5)	cancel (esc)	enter (f3)	schedule (f	9) next (f7)	previous (f8)	next form (f6)
1	2	3										
					Pattern No	umber: 13	Pattern I	Name:				
	Α.				;	SCCAN? n	Secure	SIP? n				
	Grß	FRL	NPA	Pfx	Hop Toll I	No. Insei	rted			DCS/	IXC	
	No			Mrk	Lmt List I	Del Diqit	ts			QS10		
					ı	Dgts				Intu	,	
1:	13	0				3				n	user	
2:										n	user	
3:										n	user	
4:										n	user	
5:										n	user	
6:										n	user	
"											450.	
	BCC	. VAI	LUE	TSC	CA-TSC	ITC BCIE	Service/Fo	eature BA	ND No.	Numbering	LAR	
			4 W		Request					Format		
									Subaddre			
1 1:	. y y		II N		as-needed	rest					none	
	99		_	_	us meeded	rest					none	
			_			rest					none	
			-			rest					none	
	. y y		-									
	уу		_			rest					none	
0:	уу	y y	y n	n		rest					none	



CONFIGURATIONS FOR PHONES

Figure 18. Digital Station Configuration – 1 of 2.

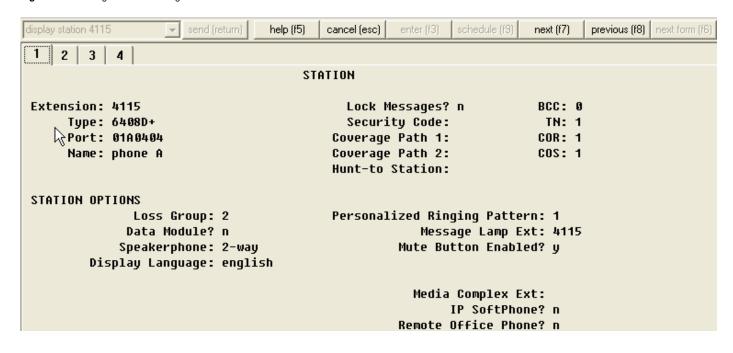
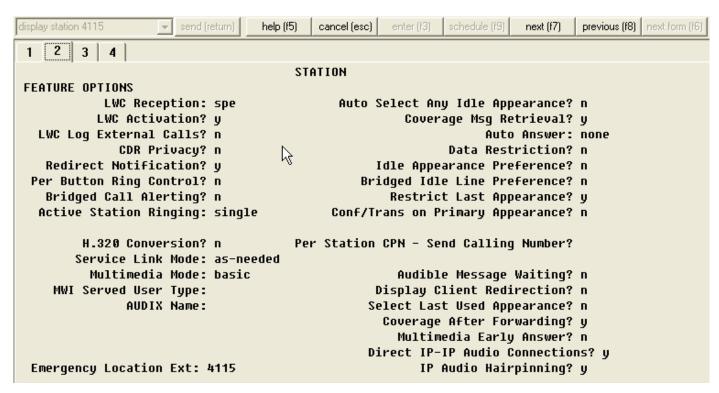




Figure 19. Digital Station Configuration – 2 of 2.

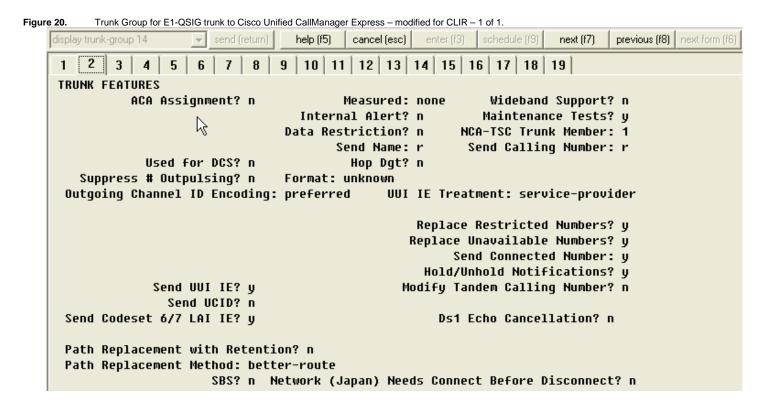




CLIR

For Calling Line ID Restriction (CLIR, CNIR) to be implemented, the associated trunk group must be modified.

On page 2 of the Trunk Group screen, "Send Name" field and "Send Calling Number" field must be changed to "r" for restricted.





CALL FORWARD BY JOIN

For diversion (CFU, CFB) to be accomplished by join instead of reroute, a coverage path must be assigned to the forwarding station.

On page 1 of the station form associated with the forwarding station, "Coverage Path 1" must be set to 1. See Figure 21.

On page 2 of the station form associated with the forwarding station, "Coverage after Forwarding" must be set to "y". See Figure 22.

Some system parameters also must be enabled:

On page 1 of the system parameters / coverage forwarding form, "QSIG VALU Coverage Overrides QSIG Diversion with Rerouting" must to be set to "y". See 0

On page 1 of the system parameters / coverage forwarding form, "Call Forward Override" must be set "y". See 0

On page 1 of the system parameters / coverage forwarding form, "Coverage After Forwarding" also must be set to "y". See 0

On page 2 of the system parameters / coverage forwarding form. "Coverage of Calls Redirected Off-net Enabled" needs to be set to "y". Figure 24.

Figure 21. Screen shot of station form for Call Forward by Join – 1 of 2.





Figure 22. Screen shot of station form for Call Forward by Join - 2 of 2.

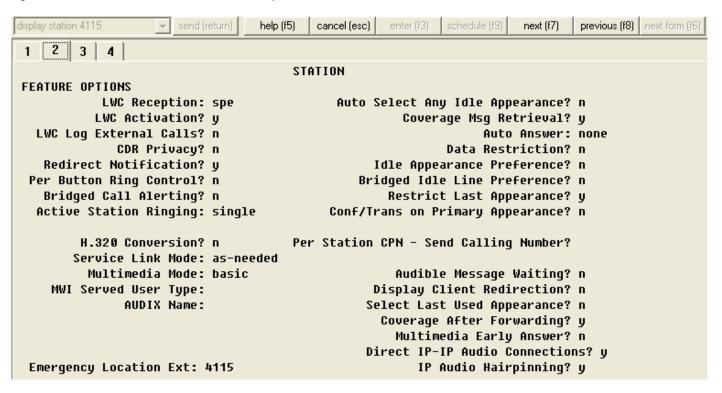


Figure 23. Screen shot of system parameters / coverage forwarding form for Call Forward by Join – 1 of 2.

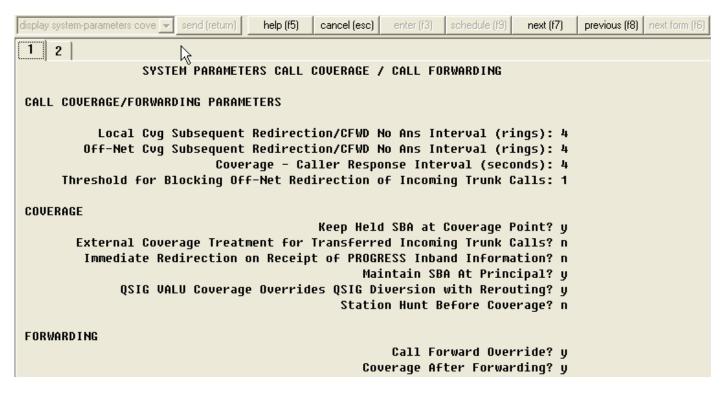
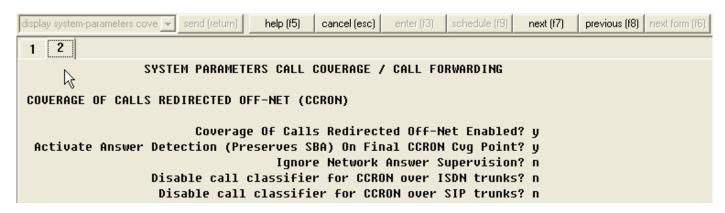




Figure 24. Screen shot of system parameters / coverage forwarding form for Call Forward by Join – 2 of 2.





Configuring the Local Cisco Unified CallManager Express (Cisco 3845)

Building configuration...

```
Current configuration: 4460 bytes
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
hostname c3845CME
boot-start-marker
boot system flash:c3845-ipvoice-mz.124-11.T.bin
boot-end-marker
logging buffered 10000000
no logging console
enable password cisco
no aaa new-model
network-clock-participate wic 0
network-clock-select 1 E1 0/0/1
ip cef
no ip dhcp use vrf connected
ip dhcp excluded-address 200.1.1.1
ip dhcp pool phone
 network 200.1.1.0 255.255.255.0
 option 150 ip 200.1.1.1
 default-router 200.1.1.1
no ip domain lookup
multilink bundle-name authenticated
isdn switch-type primary-qsig
voice-card 0
no dspfarm
voice call send-alert
voice service pots
supplementary-service qsig call-forward
voice service voip
qsig decode
allow-connections h323 to h323
allow-connections h323 to sip
allow-connections sip to h323
allow-connections sip to sip
h323
```



```
sip
 bind control source-interface GigabitEthernet0/1
 bind media source-interface GigabitEthernet0/1
 rel1xx disable
 min-se 100
 ds0-num
 header-passing
 registrar server
voice register global
mode cme
source-address 200.1.1.1 port 5060
max-dn 100
max-pool 192
load 7960-7940 POS3-07-5-00
tftp-path flash:
create profile sync 0594123930921449
voice register dn 1
number 7000
name Zidane
huntstop
voice register dn 2
number 7001
call-forward b2bua noan 5050 timeout 5
name Platini
huntstop
voice register pool 1
id mac 000F.9054.2FC2
type 7960
number 1 dn 1
max registrations 240
dtmf-relay rtp-nte
description Zidane
voice register pool 2
id mac 0012.4362.BF71
type 7960
number 1 dn 2
max registrations 240
dtmf-relay rtp-nte
description Platini
!
!
!
```



```
!
!
controller E1 0/0/0
controller E1 0/0/1
clock source line primary
pri-group timeslots 1-31
!
!
interface GigabitEthernet0/0
ip address 172.20.8.26 255.255.255.0
duplex auto
speed auto
media-type rj45
no keepalive
interface GigabitEthernet0/1
ip address 200.1.1.1 255.255.255.0
duplex auto
speed auto
media-type rj45
no keepalive
interface Serial0/0/1:15
no ip address
encapsulation hdlc
isdn switch-type primary-qsig
isdn overlap-receiving
isdn incoming-voice voice
isdn bchan-number-order ascending
no cdp enable
ip default-gateway 172.20.8.1
ip route 0.0.0.0 0.0.0.0 172.20.8.1
ip route 201.2.2.0 255.255.255.0 172.20.8.27
ip http server
ip http authentication local
ip http path flash:
tftp-server flash:P003-07-5-00.bin
tftp-server flash:P003-07-5-00.sbn
tftp-server flash:P0S3-07-5-00.bin
tftp-server flash:P0S3-07-5-00.sb2
tftp-server flash:P0S3-07-5-00.loads
tftp-server flash:TERM41.7-0-3-0S
tftp-server flash:P0030702T023
control-plane
!
!
voice-port 0/0/1:15
mwi
```



```
!
!
dial-peer voice 6000 voip
destination-pattern 700[89]
session target ipv4:201.2.2.1
no vad
dial-peer voice 95558000 pots
destination-pattern 3...
direct-inward-dial
port 0/0/1:15
forward-digits all
dial-peer voice 2200 pots
destination-pattern 41..
incoming called-number ....
direct-inward-dial
port 0/0/1:15
forward-digits all
dial-peer voice 5050 pots
destination-pattern 50..
direct-inward-dial
port 0/0/1:15
forward-digits all
gateway
timer receive-rtp 1200
sip-ua
retry options 0
mwi-server ipv4:200.1.1.1 expires 3600 port 5060 transport udp
telephony-service
load 7960-7940 P0030702T023
load 7961 TERM41.7-0-3-0S
max-ephones 96
max-dn 192
ip source-address 200.1.1.1 port 2000
system message ABC Corp
max-conferences 8 gain -6
call-forward pattern .T
moh music-on-hold.au
dn-webedit
time-webedit
transfer-system full-blind
transfer-pattern ....
secondary-dialtone 9
create cnf-files version-stamp Jan 01 2002 00:00:00
!
ephone-dn 3 dual-line
number 7002
label 7002
```



```
description Pele
name Pele
call-forward noan 5050 timeout 5
huntstop channel
!
ephone-dn 4 dual-line
number 7003
label 7003
description Beckenbauer
name Beckenbauer
huntstop channel
ephone-dn 5 dual-line
call-waiting ring
number 7004
label 7004
description H. Sanchez
name Sanchez
huntstop channel
ephone 3
mac-address 0017.0EEE.2F5E
type 7961
keep-conference
button 1:3
!
!
!
ephone 4
mac-address 0015.2B8F.351B
type 7961
keep-conference
button 1:4
ephone 5
mac-address 0014.1C48.DE7A
type 7960
keep-conference
button 1:5
!
!
line con 0
password cisco
login
stopbits 1
line aux 0
stopbits 1
line vty 0 4
exec-timeout 0 0
password cisco
login
scheduler allocate 20000 1000
```



! end

c3845CME#



Configuring the Cisco Unified CallManager Express 2 (Cisco 2801)

```
c2801CME#sh run
Building configuration...
Current configuration: 3121 bytes
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
hostname c2801CME
boot-start-marker
boot system flash:c2801-ipvoice-mz.124-11.T.bin
boot-end-marker
logging buffered 100000000
no logging console
enable password cisco
no aaa new-model
network-clock-participate wic 1
network-clock-select 1 E1 0/1/1
ip cef
no ip dhcp use vrf connected
ip dhcp excluded-address 201.2.2.1
ip dhcp pool phone
 network 201.2.2.0 255.255.255.0
 option 150 ip 201.2.2.1
 default-router 201.2.2.1
1
no ip domain lookup
multilink bundle-name authenticated
isdn switch-type primary-qsig
voice-card 0
voice service pots
supplementary-service qsig call-forward
voice service voip
qsig decode
allow-connections h323 to h323
allow-connections h323 to sip
allow-connections sip to h323
allow-connections sip to sip
h323
sip
```



```
registrar server expires max 600 min 60
!
!
!
!
!
controller E1 0/1/0
controller E1 0/1/1
pri-group timeslots 1-31
interface FastEthernet0/0
ip address 172.20.8.27 255.255.255.0
duplex auto
speed auto
interface FastEthernet0/1
ip address 201.2.2.1 255.255.255.0
duplex auto
speed auto
interface Serial0/1/1:15
no ip address
encapsulation hdlc
isdn switch-type primary-qsig
isdn incoming-voice voice
no cdp enable
ip default-gateway 172.20.8.1
ip route 0.0.0.0 0.0.0.0 172.20.8.1
ip route 200.1.1.0 255.255.255.0 172.20.8.26
ip http server
ip http authentication local
ip http path flash:
disable-eadi
tftp-server flash:P003-07-5-00.bin
tftp-server flash:P003-07-5-00.sbn
tftp-server flash:P0S3-07-5-00.bin
tftp-server flash:P0S3-07-5-00.sb2
```



```
tftp-server flash:P0S3-07-5-00.loads
tftp-server flash:TERM41.7-0-3-0S
tftp-server flash:P0030702T023
control-plane
!
!
!
voice-port 0/1/1:15
dial-peer voice 4000 voip
destination-pattern 7..[01234]
session target ipv4:200.1.1.1
no vad
dial-peer voice 9 voip
destination-pattern 4...
session target ipv4:200.1.1.1
no vad
dial-peer voice 2200 voip
destination-pattern 2...
session target ipv4:200.1.1.1
dtmf-relay rtp-nte
no vad
dial-peer voice 5000 voip
destination-pattern 50..
session target ipv4:200.1.1.1
dtmf-relay rtp-nte
no vad
telephony-service
load 7960-7940 P0030702T023
load 7961 TERM41.7-0-3-0S
max-ephones 30
max-dn 150
ip source-address 201.2.2.1 port 2000
system message CBA Corp
max-conferences 8 gain -6
call-forward pattern .T
moh music-on-hold.au
dn-webedit
time-webedit
transfer-system full-blind
transfer-pattern ....
secondary-dialtone 9
create cnf-files version-stamp Jan 01 2002 00:00:00
!
ephone-dn 1 dual-line
number 7008
label 7008
```



```
description Ronaldinho
name Ronaldinho
call-forward noan 5050 timeout 5
huntstop channel
!
ephone-dn 4 dual-line
number 7009
label 7009
description Tevez
name Tevez
huntstop channel
ephone 1
mac-address 000F.9069.DB2C
type 7960
keep-conference
button 1:1
ephone 4
mac-address 0030.94C3.31AD
type 7960
keep-conference
button 1:4
!
line con 0
password cisco
login
line aux 0
line vty 04
exec-timeout 0 0
password cisco
login
scheduler allocate 20000 1000
end
```

c2801CME#



Acronyms

Acronym	Definitions
BRI	Basic Rate ISDN
CAMA	Centralized Automatic Message Accounting
CAS	Channel Associated Signaling
CFB	Call Forward when Busy
CFNR	Call Forward when No Reply
CFU	Call Forward Unconditional
СО	Central Office
FGD	Feature Group "D"
FXO	Foreign Exchange – Office
FXS	Foreign Exchange – Station
IOS	Internetworking Operating System
MCID	Malicious Caller ID
MGCP	Media Gateway Control Protocol
МоН	Music on Hold
MWI	Message Waiting Indication
PBX	Private Branch Exchange
PRI	Primary Rate ISDN
PSAP	Public Service Access Point
SIP	Session Initiation Protocol
ToH	Tone on Hold



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