

Siemens HiPath 4000 Rel 3.0 to Cisco Unified Communications Manager 7.1(3) using Cisco Unified Communications Manager-Session Management Edition 7.1(3) using SIP

November 24, 2009 - Initial Version

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Introduction

- This application note describes the necessary steps and configurations for connectivity between Siemens HiPath 4000 release 3.0, and a Cisco Unified Communications Manager (Cisco UCM) version 7.1(3) with Cisco Unified Communications Manager-Session Management Edition (Cisco UCM-SME) Version 7.1.3.
- The network topology diagram (Figures 1) shows the test setup for end-to-end interoperability between Cisco Unified Communications Manager (Cisco UCM) Release 7.1 connected to the Siemens HiPath 4000 PBX via a Cisco Session Management Edition (SME) using SIP trunks. Features tested are basic call, 3-way (ad-hoc) conference, call transfer (attended and unattended), call forward (all, busy and no answer), hold/resume, fax transmission, and DTMF interworking. This test setup also includes a connection to a PSTN simulator, using SIP trunks. Cisco Unified Border Element (Cisco UBE) is used as a session border controller (SBC), providing demarcation, security, and interworking services between the customer's private network and the service provider's SIP network.
- During testing, a Cisco 3825 voice gateway was used to run the Cisco Unified Border Element features set. However other Cisco voice gateways can be used. The decision to choose the Cisco gateway model is left to the customer. The customer should choose a Cisco IOS gateway model based on the capabilities and the capacity that will be required based on the planned network deployment. Here is a list of Cisco IOS products capable of running Cisco UBE.

<http://www.cisco.com/en/US/products/ps10536/index.html>[Cisco 2800 Series Integrated Services Routers](#)
[Cisco 3800 Series Integrated Services Routers](#)
[Cisco 3900 Series Integrated Services Routers](#)
[Cisco 7200VXR Routers](#)
[Cisco 7301 Routers](#)
[Cisco AS5350XM Universal Gateway](#)
[Cisco AS5400XM Universal Gateway](#)

Network Topology

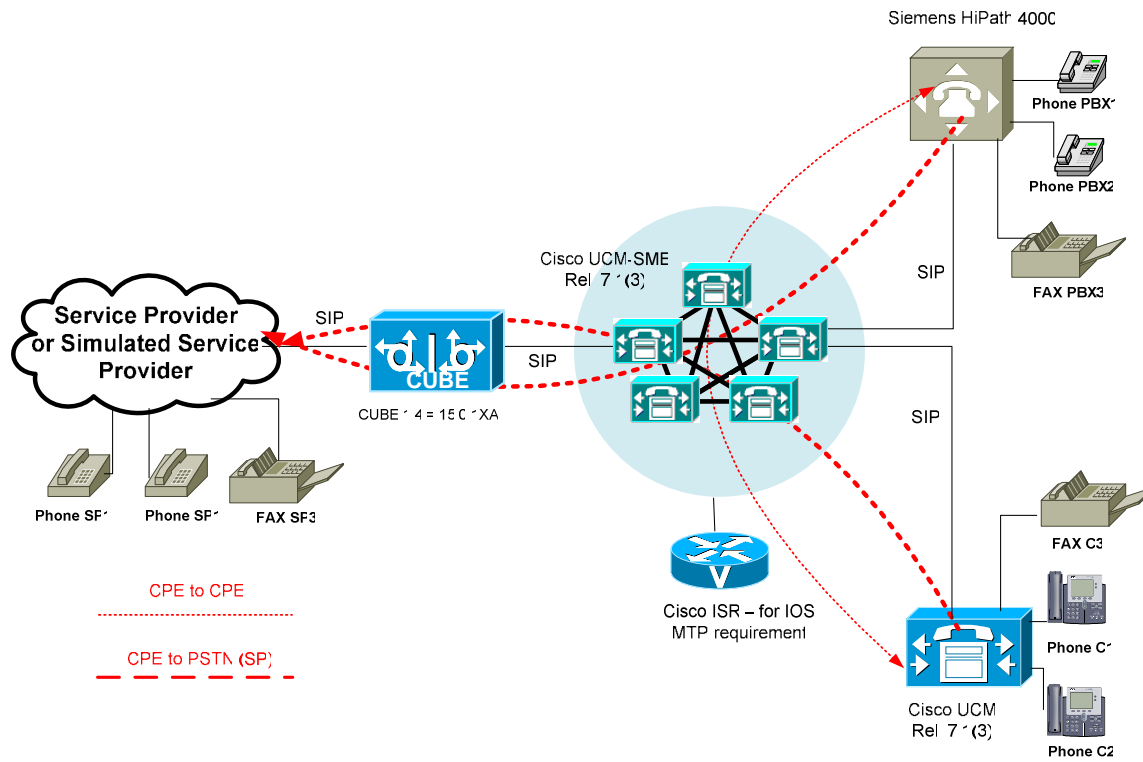


Figure 1. Basic Call Setup

Capabilities

- Voice/fax calls including supplementary services can be successfully established between endpoints controlled by the Siemens PBX and endpoints controlled by the Cisco Unified Communications Manager.
- Voice/fax calls including supplementary services can be successfully established between endpoints controlled by the Siemens PBX and the PSTN, using Cisco UBE as a session border controller.

LimitationsPBX

- Siemens PBX does not send calling name in the header P-Asserted identity, it only sends number.
- Siemens PBX does not send updated connected party information once call transfers are completed.
- Calls scenarios involving call hold/resume (i.e. call transfers, call conferences) from Cisco UCM endpoints fail to reconnect, unless MTP's are used. Siemens HiPath 4000 does not properly support SIP Call Hold: Cisco UCM sends mid-call INVITE with null IP address and "inactive" Media Attribute. Upon receiving mid-call INVITE placing call on hold, the Siemens PBX responds with a 200 OK with no SDP, causing Cisco UCM to fail to properly place call on hold. SIP trunks connecting SME to the Siemens HiPath 4000 must be configured with "Media Termination Point Required".
- Siemens HiPath 4000 does not support network/external call transfers (early attended) and network/external call forwards (unconditional, busy or no reply) over SIP trunks. When attempting early attended call transfers over SIP trunk, Siemens' phone display shows "Not Possible". The same behavior is also seen whenever trying to configure call forwarding to an external number reachable over SIP trunk. Also, testing indicates that Diversion headers sent inbound to the Siemens PBX are ignored, as calls forwarding to Siemens stations over SIP trunks do not display call forwarding information. These limitations prevent centralized voicemail services across SIP trunks.
- Siemens Hipath 4000 does not support T.38 fax relay. Upon receiving INVITE with SDP negotiating T.38 fax-relay, Siemens PBX responds with 488 Not Acceptable Media. On inbound fax calls, the Siemens PBX negotiates fax pass-through using G.711Alaw.

CUBE

- During call forward local (Unconditional, Busy or No reply) Originator do not update final destination number because CUBE generates the PA-ID information from the contact header and not from the information sent in PA-ID by CUCM or PBX.
- During call forward Network/External (Unconditional, Busy or No reply) Originator do not update final destination number because CUBE generates the PA-ID information from the contact header and not from the information sent in PA-ID by CUCM or PBX.

System Components**Hardware Components**

- Cisco MCS 7800 Unified Communications Manager Appliance
- Cisco 3825 voice gateway
- 2 Cisco Unified IP phone 7960 configured as SCCP phones
- 2 Cisco Unified IP phone 7970 configured as SIP phones
- Siemens HiPath 4000 PBX
- 2 Siemens E Optiset advanced plus digital phones (Euro/US model)

Software Requirements

The following software is required:

- Cisco Unified Communications Manager Release 7.1
- Cisco IOS Release 15.0.1XA
- Siemens HiPath 4000 PBX Release 3.0

Features

This section lists supported and unsupported features.

Features Supported

- Basic calls (See Limitations section for details.)
- CLIP-Calling line (Number) identification presentation
- CLIR-Calling line (Number) identification restriction
- COLP-Connected line (Number) identification presentation
- COLR- Connected line (Number) identification restriction (See Limitations section for details.)
- Consultation transfer – Local and Network/External (See Limitations section for details.)
- Early Attended transfer – Local and Network/External (See Limitations section for details.)
- Call forward Local – Unconditional, Busy and No reply (See Limitations section for details.)
- Call forward Network/External – Unconditional, Busy and No reply (See Limitations section for details.)
- DTMF interworking
- Fax transmissions using G.711 pass-through

Features Not Supported (See limitations)

- CNIP-Calling name identification presentation
- CNIR-Calling name identification restriction



- CONP-Connected name identification presentation
- CONR- Connected name identification restriction
- Centralized Message center voicemail integration
- T.38 Fax Relay

Application Note



This section contains configuration menus and commands and describes configuration sequences and tasks.

Configuring the Siemens HiPath 4000 PBX

1. Add the new access code to Dialing Plans using WABE + LDPLN.
2. Add the new trunk group access code using BUEND.
3. Configure trunk using TDCSU.
4. Configure Class of Trunk using COT.
5. Configure Class of Parameter for device handler using COP.
6. Add the new trunk board using BCSU.
7. Configure Class of Service using COSSU.
8. Configure Gateway Board
9. Configure Gateway
10. Configure RICHT
11. Configure LCR Out-dial Rules using LODR.
12. Configure station for Name and Number restrictions
13. Enable In-Band DTMF signaling for the Digital Stations using SBSCU.
14. Configure Digital Station for MWI application.
15. Configure Message Center's Service Access Number for MWI application
16. Software release information
17. Configuring Siemens HiPath Assistant (Screen shots)
 - Siemens HiPath 4000 Assistant Version 3.0
 - Gateway configuration
 - SIP parameters
 - CODEC parameter configuration



Configuration Menus and Commands for Hipath 4000

DPLN

DIS-WABE:GEN;

H500: AMO WABE STARTED

DIGIT INTERPRETATION		VALID FOR ALL DIAL PLANS			
CODE		CALL PROGRESS STATE			
		1	11111	11112	22
		0	12345	67890	12345 67890 12
					DIGIT ANALYSIS RESULT
					RESERVED/CONVERT DNI/ADD-INFO
					*=OWN NODE
0		. *****	. ***	** *
001	- 010	* *
111		. *****	*****	** *
1150	- 1159	. *****	*****	** *
					CO
					NETRTE
					TIE
					STN
					DESTNO 111
					DNNO 0- 0-111
					PDNNO 0- 0-111
12	- 15	. *****	*****	** *
21		. *****	*****	** *
22		. *****	*****	** *
222		. *****	*****	** *
23		. *****	*****	** *
24		. *****	*****	** *
25		. *****	*****	** *
26		. *****	*****	** *
26		. *****	*****	** *
27		. *****	*****	** *
					TIE
					KNOVRKY
					DNDKY
					TIE
					FWDKY
					MBKY
					MSGRKY
					DAKY
					DFWVDVCE
					DSSKY
DIGIT INTERPRETATION		VALID FOR ALL DIAL PLANS			
CODE		CALL PROGRESS STATE			
		1	11111	11112	22
		0	12345	67890	12345 67890 12
					DIGIT ANALYSIS RESULT
					RESERVED/CONVERT DNI/ADD-INFO
					*=OWN NODE
27		. *****	*****	** *
28		. *****	*****	** *
28		. *****	*****	** *
29		. *****	*****	** *
30		. *****	*****	** *
3000	- 3010	. *****	*****	** *
					AFWVDVCE
					VCRKY
					DFWVDVCE
					VCKY
					CONFKY
					STN
					DESTNO 30
					DNNO 0- 0- 30
					PDNNO 0- 0-222
3011	- 3020	. *****	*****	** *
					STN
					DESTNO 31
					DNNO 0- 0- 31
					PDNNO 0- 0- 31
3021	- 3030	. *****	*****	** *
					STN
					DESTNO 32
					DNNO 0- 0- 32
					PDNNO 0- 0- 32
DIGIT INTERPRETATION		VALID FOR ALL DIAL PLANS			
CODE		CALL PROGRESS STATE			
		1	11111	11112	22
		0	12345	67890	12345 67890 12
					DIGIT ANALYSIS RESULT
					RESERVED/CONVERT DNI/ADD-INFO
					*=OWN NODE
3031	- 3040	. *****	*****	** *
					STN
					DESTNO 33
					DNNO 0- 0- 33
					PDNNO 0- 0- 33
3041	- 3050	. *****	*****	** *
					STN
					DESTNO 35
					DNNO 0- 0- 35
					PDNNO 0- 0- 35



31		*	NAMEKY		
32		*	PARKKY		
33		*	CCKY		
34		*	HTKY		
35		*	STKY		
36	- 37	**** **	CO		
38		*	TIMEKY		
39		**** **	TIE		

Application Note

DIGIT INTERPRETATION		VALID FOR ALL DIAL PLANS			
CODE	CALL PROGRESS STATE	DIGIT ANALYSIS	RESERVED/CONVERT		
	1 1111 1112 22	RESULT	DNI/ADD-INFO		
	0 12345 67890 12345 67890 12		*=OWN NODE		
4000 - 4050	. **** **	STN	DESTNO 111		
			DNNO 0- 0-111		
			PDNNO 0- 0-111		
4051 - 4566	. **** **	STN	DESTNO 222		
			DNNO 0- 0-222		
			PDNNO 0- 0-222		
4567	. **** **	STN	DESTNO 34		
			DNNO 0- 0- 34		
			PDNNO 0- 0-200		
4568 - 4599	. **** **	STN	DESTNO 222		
			DNNO 0- 0-222		
			PDNNO 0- 0-222		

DIGIT INTERPRETATION		VALID FOR ALL DIAL PLANS			
CODE	CALL PROGRESS STATE	DIGIT ANALYSIS	RESERVED/CONVERT		
	1 1111 1112 22	RESULT	DNI/ADD-INFO		
	0 12345 67890 12345 67890 12		*=OWN NODE		
4600 - 4650	. **** **	STN	DESTNO 80		
			DNNO 0- 0- 80		
			PDNNO 0- 0- 80		
4651 - 4999	. **** **	STN	DESTNO 222		
			DNNO 0- 0-222		
			PDNNO 0- 0-222		
5000 - 5009	. **** **	STN	DESTNO 0		
			DNNO 0- 0-555*		
			R		
5010 - 5020	. **** **	STN	DESTNO 0		
			DNNO 0- 0-555*		
5021 - 5040	. **** **	STN	DESTNO 0		
			DNNO 0- 0-555*		

DIGIT INTERPRETATION			VALID FOR ALL DIAL PLANS				
CODE		CALL PROGRESS STATE				DIGIT ANALYSIS	RESERVED/CONVERT
		1 1111 1112 22					DNI /ADD-INFO
		0 12345 67890 12345 67890 12			RESULT		*=OWN NODE
5100	- 5109	. ***** **...		*	STN	DESTNO 0
5500	- 5501	. ***** **...		*	STN	DNNO 0- 0-555*
							DESTNO 56
							DNNO 0- 0-560
						PDNNO 0- 0-560	
555		. ***** **...		*	OWNNODE	
560		. ***** **...		*	TIE	
570		. ***** **...		*	TIE	

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578	*****	TIE	Application Note
59	*****	TIE	
6000 - 6009	*****	STN	
		DESTNO 111	
		DNNO 0- 0-111	
		PDNNO 0- 0-111	
DIGIT INTERPRETATION VALID FOR ALL DIAL PLANS			
CODE	CALL PROGRESS STATE 1 11111 11112 22 0 12345 67890 12345 67890 12	DIGIT ANALYSIS RESULT	RESERVED/CONVERT DNI/ADD-INFO *=OWN NODE
6123	*****	STN	R
			DESTNO 0
			DNNO 0- 0-555*
62	*****	AFFWDVCE	
7000 - 7002	*****	STN	
			DESTNO 56
			DNNO 0- 0-560
			PDNNO 0- 0-560
8000 - 8050	*****	STN	
			DESTNO 222
			DNNO 0- 0-222
			PDNNO 0- 0-222
8060	*****	TIE	
8070	*****	TIE	
8080	*****	TIE	
8088	*****	TIE	
DIGIT INTERPRETATION VALID FOR ALL DIAL PLANS			
CODE	CALL PROGRESS STATE 1 11111 11112 22 0 12345 67890 12345 67890 12	DIGIT ANALYSIS RESULT	RESERVED/CONVERT DNI/ADD-INFO *=OWN NODE
8100 - 8109	*****	STN	
			DESTNO 32
			DNNO 0- 0- 32
			PDNNO 0- 0- 32
8200 - 8209	*****	PARK	
83	*****	SPDC1	
84	*****	SPDC2	
88	*****	SCONSI	R
89	*****	SCONSCO	R
9	*****	TIE	
*13	*****	AHTVCE	
*15	*****	SPLIT	
*16	*****	AREM	
*17	*****	TRACE	
*18	*****	ACOSX	
*19	*****	KNOVR	
*20	*****	ADND	
DIGIT INTERPRETATION VALID FOR ALL DIAL PLANS			
CODE	CALL PROGRESS STATE 1 11111 11112 22 0 12345 67890 12345 67890 12	DIGIT ANALYSIS RESULT	RESERVED/CONVERT DNI/ADD-INFO *=OWN NODE
*25	*****	FWDTERM	
*26	*****	DFFWDVCE	
*27	*****	AFWDVCE	
*28	*****	DFWDVCE	
*29	*****	AFFWDVCE	
*91	*****	MBOFF	
#31	*****	AFFWDVCE	
#91	*****	MBON	
##27	*****	MWACT	
##28	*****	MWANS	
##29	*****	MWCAN	
##30	*****	MWCANORI	



AMO-WABE -111 DIALLING PLANS, FEATURE ACCESS CODES
DISPLAY COMPLETED;

Dial Plan, DPLN

DISPLAY-LDPLN:TYPE=LDP,LDP="570"-"XXXXX";

H500: AMO LDPLN STARTED

DIPLNUM: 0		
LDPNO : 21	LDP : 570-XXXXX	
	SPC : 22	
	FDSFIELD : 0 SDSFIELD : 0 PINDP : N	
DPLN	LROUTE	LAUTH
0	570	1
1	570	1
2	570	1
3	570	1
4	570	1
5	570	1
6	570	1
7	570	1
8	570	1
9	570	1
10	570	1
11	570	1
12	570	1
13	570	1
14	570	1
15	570	1

AMO-LDPLN-111 ADMINISTRATION LCR DIALPLAN
DISPLAY COMPLETED;

Trunk Group Access Code, BUEND

DIS-BUEND;

H500: AMO BUEND STARTED

TRUNK GROUPS (FORMAT=S)

NO.	NAME	CHARCON
1	BRI ST1	(NEUTRAL)
2	BRI ST2	(NEUTRAL)
3	BRI ST3	(NEUTRAL)
4	BRI ST4	(NEUTRAL)
10	ANALOG TML2P	(NEUTRAL)
20	PRI PSSV1	(NEUTRAL)
21	PRI 2 PSSV1	(NEUTRAL)
22	ECMA 1	(NEUTRAL)
23	ECMA 2	(NEUTRAL)
24	PRI ETSI	(NEUTRAL)
25	PRI 2 ETSI	(NEUTRAL)
26	PRI ECMA 3	(NEUTRAL)
27	PRI ECMA 4	(NEUTRAL)
30	CAS 1	(NEUTRAL)
31	CAS 2	(NEUTRAL)
50	E&M 1	(NEUTRAL)
51	E&M1	(NEUTRAL)
53	E&M 3	(NEUTRAL)
60	DPNSS 1	(NEUTRAL)
61	DPNSS2	(NEUTRAL)
80	IP TRUNK GW1 SIP	(NEUTRAL)



AMO-BUEND-111 TRUNK GROUP
DISPLAY COMPLETED;

Trunk Configuration, TDCSU

DISPLAY-TDCSU: PEN1=1-1-103-0;

H500: AMO TDCSU STARTED

DIGITAL TRUNK (FORMAT=L)					
DEV	= HG3550IP	PEN	= 1-01-103-0	TGRP	= 80
PROTVAR	= PSS1V2	INS	= Y	SRCHMODE	= DSC
COTNO	= 80	COPNO	= 80	DPLN	= 0
ITR	= 0	COS	= 66	LCOSV	= 1
LCOSD	= 1	CCT	=	DESTNO	= 111
SEGMENT	= 8	DEDSVC	=	DEDSVC	= NONE
FACILITY	=	DITIDX	=	SRTIDX	=
TRTBL	= GDTR	SIDANI	= N	ATNTYP	= TIE
CBMATR	= NONE	NWMUXTIM	= 10	TCHARG	= N
SUPPRESS	= 0	DGTPR	=	CHIMAP	= N
ISDNIP	=	ISDNNP	=		
PNPL2P	=	PNPL1P	=	PNPAC	=
TRACOUNT	= 31	SATCOUNT	= MANY	NNO	= 111
ALARMNO	= 0	FIDX	= 1	CARRIER	= 1
ZONE	= EMPTY	COTX	= 80	FWDX	= 10
DOMTYPE	=	DOMAINNO	=	TPROFNO	=
INIGHT	=			CCHDL	= SIDEA
UUSCCX	= 16	UUSCCY	= 8	FNIDX	= 1
CLASSMRK	= EC & G711			SRCGRP	=
TCCID	= IP TR1 SIP				
BCNEG	= N	BCGR	= 1	LWPAR	= 0
LWPP	= 0	LWLT	= 0	LWPS	= 0
LWR1	= 0	LWR2	= 0		
DMCALLWD	= N	DMCSEC	= N	VNNO	=
SVCDOM	=				
BCHAN	= 1 && 30				

AMOUNT OF B-CHANNELS IN THIS DISPLAY-OUTPUT: 30

AMO-TDCSU-111 DIGITAL TRUNKS
DISPLAY COMPLETED;

Class of Trunk, COT

DIS-COT: 80;

H500: AMO COT STARTED

COT: 80 INFO:

DEVICE: INDEP SOURCE: DB

PARAMETER:

PRIORITY FOR AC WILL BE DETERMINED FROM MESSAGE
RECALL IF USER HANGS UP IN CONSULTATION CALL
TRUNK CALL TRANSFER
CHANGEOVER FROM HOLD TO RING TONE
KNOCKING OVERRIDE POSSIBLE
CALL EXTEND FOR BUSY, RING OR CALL STATE
NETWORKWIDE AUTOMATIC CALLBACK ON BUSY
NETWORKWIDE AUTOMATIC CALLBACK ON FREE
NETWORKWIDE CALL FORWARDING PERMITTED
NETWORKWIDE FORWARDING NO-ANSWER
REGISTRATION OF IMPLAUSIBLE EVENTS
DON'T RELEASE CALL TO BUSY HUNT GROUP
CAMP-ON ON DID CALLS
CAMP-ON AFTER EXTENSION OF INCOMING TRUNK CALLS

PRI
RCL
XFER
CHRT
KNOR
CEBC
CBBN
CBFN
FWDN
FNAN
IEVT
BSHT
KNDI
KNEX



```
END-OF-DIAL FOR BLOCK IS SET
EMERGENCY OVERRIDE/DISCONNECT VIA S0/S2 LINE
ACTIVATE TRANSIT COUNTER ADMINISTRATION FOR S0/S2 LINE
CONNECTION TO ROUTE OPTIMIZATION NODE
TSC-SIGNALING FOR NETWORKWIDE FEATURES (MANDATORY)
INCOMING CDR ACTIVATE PER TRUNK
TRUNK SENDS CALL CHARGES TO ORIGINATING NODE NUMBER
CALL FORWARDING PROGRAMING FOR OTHER SUBSCRIBERS
CALL FORWARDING VALIDATION PROCEDURES POSSIBLE
PIN NETWORKWIDE POSSIBLE
AOC PER CALL (AUTOMATICAL OR ON REQUEST), MAND. CORNET-NQ
CDR FOR BREAK OUT TO CARRIER
AUTOM.DTMF CONVERSION ON INCOM.CALL WHILE IN TALK STATE
SEND DIGITS VIA IN.BAND DTMF BEFORE ANSWER
NO TONE

BLOC
PROV
ATRS
ROPT
TSCS
ICZO
TRSC
CFOS
CFVA
PINR
AOCC
CDBO
AMFC
IBBA
NTON
```

```
AMO-COT -111 CLASS OF TRUNK FOR CALL PROCESSING
DISPLAY COMPLETED;
```

Class of Parameter for Device Handler, COP

DIS-COP:80;

H500: AMO COP STARTED

```
COP: 80 INFO:
DEVICE: INDEP SOURCE: DB
PARAMETER:
REGISTRATION OF LAYER 3 ADVISORIES
REFLECT RESTART INDICATOR AND B-CHANNEL BY RESTART
```

L3AR
RRST

```
AMO-COP -111 CLASS OF PARAMETER FOR DEVICE HANDLER
DISPLAY COMPLETED;
```

Class of Service, COSSU

DISPLAY-COSSU:TYPE=COS,COS=66;

H500: AMO COSSU STARTED

COS	VOICE	FAX	DTE
66	> TA TNOTCR COSXCD VCE FWDNWK TTT FWDECA	NOCO NOTIE	NOCO NOTIE

```
AMO-COSSU-111 CLASSES OF SERVICE
DISPLAY COMPLETED;
```

DISPLAY-COSSU:TYPE=LCOSV,LCOSV=1;

H500: AMO COSSU STARTED

LCOS	1	2	3	4	5	6	COPIN
V	123456789012345678901234567890123456789012345678901234						
	>SERVICE INFORMATION						
1	X.....						0
	>LCR ATTENDANT FOR VOICE						

```
AMO-COSSU-111 CLASSES OF SERVICE
DISPLAY COMPLETED;
```



```
DISPLAY COSSU-TYPE LCOSD,LCOSD-1;
```

```
H500: AMO COSSU STARTED
```

LCOS		LAUTH						COPIN
D		1	2	3	4	5	6	
		123456789012345678901234567890123456789012345678901234						
		>SERVICE INFORMATION						
1		XX						
		>LCR ATTENDANT FOR DATA						

```
AMO-COSSU-111          CLASSES OF SERVICE
DISPLAY COMPLETED;
```

Gateway Board configuration, STMIB

```
DISPLAY-STMIB;
```

```
H500: AMO STMIB STARTED
```

```
+-----+
| STM12IGW BOARD DATA |
+-----+

+-----+
| LTU = 1      SLOT = 103 |
+-----+
```

ETHERNET INTERFACE

```
-----
CUSIP      = 172.20.188.253
SNETMASK   = 255.255.255.0
DGWIP      = 172.20.188.1
VLAN       = NO
VLANID     = 0
```

GLOBAL DATA

```
-----
IDLE CODE PATTERN = 213
```

CONSTANT VALUES:

```
OPMODE = 1  DATA_VALID = YES
TRKPROT = SIP
```

PRIMARY AND SECONDARY GATEKEEPER

```
-----
PRIGKIP    =
PRIGKPN    = 1719
PRIGKID1   = PRIMARYRASMANAGERID
PRIGKID2   =
SECGKIP    =
SECGKPN    = 1719
SECGKID1   = SECONDARYRASMANAGERID
SECGKID2   =
TIMTOLIVE  = 120
```

MANAGEMENT STATION AND BACK-UP SERVER

```
-----
MGNTIP     =
MGNTPN     =
BUSIP      =
BUSPN      =
```

DMC DATA

```
-----
DMCCONN    = 0
```

WBM LOGIN DATA

```
-----
LOGINWBM   = HP4K-DEVEL  ROLE = ENGR
```



```
LOGINWBM = HP4K-SU      ROLE = SU
LOGINWBM = HP4K-ADMIN   ROLE = ADMIN
LOGINWBM = HP4K-READER  ROLE = READONLY
```

Application Note

```
GATEWAY DATA
-----
GWID1      = Gateway2_1_SIP
GWID2      =

H.235 SECURITY DATA
-----
GLOBID1    = siemensGateway2003
GLOBID2    =
TIMEWIN    = 90
GLOBPW     =
242-191-30-119-188-83-173-161-43-0-70-36-218-74-169-221-78-102-174-170
H235SEC    = NO

LEGK DATA
-----
GWNO       = 1
GWDIRNO    = 8080
REGEXTGK   = NO

SIP SUBSCRIBER
-----

SIP TRUNKING DATA FOR ERH
-----
GWAUTREQ   = NO
GWSECRET   = *****
GWUSERID   =
GWREALM    =

SIP TRUNKING DATA FOR SSA
-----
SIPREG     = NO
REGIP1     = 0.0.0.0
REGPORT1   = 5060
REGIP2     = 0.0.0.0
REGPORT2   = 5060
REGTIME    = 120

SNMP CONFIGURATION DATA
-----
CS1        = "public"
CS2        = ""
```

```
AMO-STMIB-111      CONFIGURATION OF HG35XX AND HG35XX-2 BOARDS
DISPLAY COMPLETED;
```

Gateway configuration, GKREG

DIS-GKREG;

H500: AMO GKREG STARTED

GWNO	1	GWATTR	INTGW	REGGW	HG3550V2 SIP
GWIPADDR	172.20 .188.253		GWDIRNO	8080	
DIPLNUM	0	DPLN	0		
LAUTH	1				
GATEWAY REGISTERED: YES					
IP GATEWAY IS CONFIGURED BY GKREG					
INFO:					
GWNO	2	GWATTR	INTGW	REGGW	HG3550V2
GWIPADDR	172.20 .188.254		GWDIRNO	8088	
DIPLNUM	0	DPLN	0		
LAUTH	1				
GATEWAY REGISTERED: NO					
IP GATEWAY IS CONFIGURED BY GKREG					

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EDCS#833326 Rev # Initial Revision



INFO:

```
GWNO      3          GWATTR  EXTGW  HG3550V2 SIP
GWIPADDR 172.20 .109.203      GWDIRNO 5060
DIPLNUM  0          DPLN  0
LAUTH     1
GATEWAY REGISTERED: NO
IP GATEWAY IS CONFIGURED BY GKREG
INFO:
```

Application Note

AMO-GKREG-111 GATEKEEPER REGISTRY
DISPLAY COMPLETED;

RICHT

DISPLAY-RICHT:MODE=LRTE,LRTE=570;
H500: AMO RICHT STARTED

```
+-----+
| LRTE = 570   NAME = IP TO GW3 SIP   (NEUTRAL)  LSVC = ALL
| DNNO =1 -1 -80 PDNNO = 0
| ROUTOPT = NO REROUT = YES  PLB = YES  FWDBL = NO
| DTMFCNV = FIX DTMFDSP = WITHOUT DTMFTEXT =
| DTMFPULS = PP80 BUGS = LIN  ROUTATT = NO  MAINGRP = 39
| EMCYRTT = NO  CONFTONE = NO  RERINGRP = NO  RTENO = 39
| INFO =
| NOPRCFWD = NO
| NITO = NO
| CLNAMEDL = NO
| FWDSWTCH = NO
| LINFEMER = NO
+-----+
| TGRP = 80  LDAT  IP TRUNK GW1 SIP   (NEUTRAL)  SUBGROUP = 23
+-----+
```

AMO-RICHT-111 TRUNK ROUTING
DISPLAY COMPLETED;

LCR Out-dial Rules, LODR

DISPLAY-LODR:ODR=1;
H500: AMO LODR STARTED

```
+-----+
| ODR   POSITION  CMD      PARAMETER
+-----+
| 1     1        ECHO      1
|      2        END
+-----+
```

H03: THE NEXT FREE ODR IS 3

AMO-LODR -111 ADMINISTRATION OF LCR OUTDIAL RULES
DISPLAY COMPLETED;

DISPLAY-LDAT:LROUTE=570;
DISPLAY-LDAT:LROUTE=570;
H500: AMO LDAT STARTED

```
+-----+
| LROUTE = 570  LDPLN      NAME = IP TO GW3 SIP      SERVICE = ALL
| TYPE = LCR                                DNNO OF ROUTE = 1 -1 -80
| SERVICE INFO =
+-----+
| LRTEL | LVAL | TGRP | ODR | LAUTH | SCHEDULE | CARRIER | LATR | LDSRT | COTIDX |
|-----|-----|-----|-----|-----|-----|-----|-----|-----|-----|
| ABCDEFGH | ZONE |
```



Application Note

```
AMO-LDAT -111          LCR-DIRECTIONS
DISPLAY COMPLETED;
```

In-Band DTMF signaling:

In order to enable In-band DTMF signaling on digital stations for Voicemail applications, the station configuration has to be changed so that the parameter DTMFCTRD=Y.

Name and Number Restrictions:

To use Name and Number Restrictions, the station configuration should be changed so that the parameter SSTNO=Y, (Secret Station Number must be set to Yes).

[DISPLAY-SBCSU:5002;](#)

H500: AMO SBCSU STARTED

```
----- USER DATA -----
STNO    =5002      OPT    =OPTI  COS1    =2      DPLN    =1
MAIN0   =5002      CONN   =DIR    COS2    =2      ITR      =1
PEN     = 1- 3- 31- 2      LCOSV1   =6      COSX    =0
INS     =Y          ASYNCT =500    LCOSV2   =6
          PERMACT =          LCOSD1   =6
SSTNO   =Y          EXTBUS =          LCOSD2   =6      CBKBMAX =5
TRACE   =N          DFCVCANA=          SPDI     =0      RCBKB   =N
ALARMNO =0          FLASH  =          SPDC1    =        RCBKNA  =N
HLMUSIC =0          SPDC2  =          COMGRP   =0
PMIDX   =1

SECR    =N          DIGNODIS=N      DSSTNA   =N
STD     =55         CALLOG  =NONE    DSSTNB   =Y      TEXTSEL  =ENGLISH

REP     =0          OPTICOM =N       OPTIUSB  :      VPI      =
IDCR    =N          OPTICA  =1       OPTIS0A  :0     VCI      =
APPM    =           OPTIDA  =1       OPTISPA  :0     PATTERN  =
                                     OPTIABA  :0

DCFWBUSY=N          HEADSET =N       APMOBSR=        APICLASS=
DNIDSP   =N         HSKEY   =NORMAL  IPCODEC =        SECAPPL =
DTMFBLK  =N
DTMFCTRD=Y          BASICSVC=
DVCFIG   =OPTISET   TSI      =1       SPROT    =        SOPTIDX  =
                                     DPROT    =        DOPTIDX  =
                                     FPROT    =        FOPTIDX  =

----- ACTIVATION IDENTIFIERS FOR FEATURES -----
HTOS    :N         DND      :N
HTOD    :N         VCP      :Y      TWLOGIN  :N
HTOF    :N         CWT      :N

----- FEATURES AND GROUP MEMBERSHIPS -----
PUGR    :          ESSTN   :
KEYSYS  :N         NOPTNO  :
SRCGRP  :(1 )      TCLASS  : 0
HUNT CD :N

----- SUBSCRIBER ATTRIBUTES (AMO SDAT) -----
NONE
```

```
AMO-SBCSU-111          STATION AND S0-BUS CONFIGURATION OF SWITCHING UNIT
DISPLAY COMPLETED;
```



Configure Digital Station for MWI Application:

Application Note

To get the MWI light to work, the station configuration should be changed so that the parameter PMIDX (phoneMail Index parameter) needs to point to the IDX identifier that identifies the Message Center's Service Access Number.

[DIS-SBCSU:5002;](#)

H500: AMO SBCSU STARTED

```
----- USER DATA -----
STNO      =5002      OPT      =OPTI      COS1      =2          DPLN      =1
MAIN0     =5002      CONN     =DIR        COS2      =2          ITR       =1
PEN       = 1- 3- 31- 2      LCOSV1     =6          COSX      =0
INS       =Y          ASYNCT   =500      LCOSV2     =6
                        PERMACT  =          LCOSD1     =6
SSTNO     =N          EXTBUS   =          LCOSD2     =6          CBKBMAX   =5
TRACE     =N                        RCBKB      =N
ALARMNO   =0          DFSVCANA=          SPD1       =0          RCBKNA    =N
HMUSIC    =0          FLASH    =          SPDC1      =          CBKNAMB   =Y
PMIDX    =1                        SPDC2      =
SECR      =N          DIGNODIS=N      DSSTNA     =N          COMGRP    =0
STD       =55        CALLOG    =NONE    DSSTNB     =Y          TEXTSEL   =ENGLISH

REP       =0          OPTICOM  =N          OPTIUSB    :          VPI       =
IDCR      =N          OPTICA    =1          OPTIS0A    :0         VCI       =
APPM      =          OPTIDA    =1          OPTISPA    :0         PATTERN   =
                        OPTIABA  :0

DCFWBUSY=N          HEADSET  =N          APMOBUSR=          APICLASS=
DNIDSP    =N          HSKEY    =NORMAL  IPCODEC    =          SECAPPL   =
DTMFBLK   =N          BASICSVC=          IPPASSW    =
DTMFCTRD=Y          TSI       =1          SPROT      =          SOPTIDX   =
DVCFIG    =OPTISET   TSI       =1          DPROT      =          DOPTIDX   =
                        FPROT    =          FOPTIDX   =

----- ACTIVATION IDENTIFIERS FOR FEATURES -----
HTOS      :N          DND       :N
HTOD      :N          VCP       :Y          TWLOGIN   :N
HTOF      :N          CWT       :N

----- FEATURES AND GROUP MEMBERSHIPS -----
PUGR      :          ESSTN     :
KEYSYS    :N          NOPTNO    :
SRCGRP    :(1 )      TCLASS    : 0
HUNT CD   :N

----- SUBSCRIBER ATTRIBUTES (AMO SDAT) -----
NONE
-----
```

AMO-SBCSU-111 STATION AND S0-BUS CONFIGURATION OF SWITCHING UNIT
DISPLAY COMPLETED;

Configure Message Center's Service Access Number for MWI Application:

Since the MWI was tested with the Message Center integrated on Cisco Unified Communication Manager side, the PBX needs to configure an identifier for the Message Center's Service Access Number in order for the MWI light to work. Without this identifier the MWI light will not work.

[ADD-RICT:MODE=PM,IDX=1,SAN=6999;](#)

H500: AMO RICHT STARTED

AMO-RICT-111 TRUNK ROUTING

ADD COMPLETED;

[DISPLAY-RICT:MODE=PM;](#)

DISPLAY-RICT:MODE=PM;

H500: AMO RICHT STARTED

```
+-----+-----+-----+-----+
|  IDX  |      SAN      |      NAME      |  TYPE  |
+-----+-----+-----+-----+
|  1    | 6999          |                 | OTHER  |
+-----+-----+-----+-----+
```



AMO-RICHT-111 TRUNK ROUTING
DISPLAY COMPLETED;

Configure Digital Station's Class of Service for Mailbox MWI application

DISPLAY-COSSU:TYPE=COS,COS=2,FORMAT=L;
H500: AMO COSSU STARTED

COS	VOICE	FAX	DTE
2	> TA TSUID TNOTCR CDRS CDRC CDRIND CDRINT COSXCD MB DATA CFNR VCE SPKR FWDNWK RERING MSN CFB FWDDIR FWDBAS FWDECA FWDEXT CCBS CW GRPCAL SUTVA	NOCO NOTIE BASIC	TA TNOTCR BASIC

AMO-COSSU-111 CLASSES OF SERVICE
DISPLAY COMPLETED;

DISPLAY-PERSI:TYPE=NAME,STNO=5002;
H500: AMO PERSI STARTED

STNO	CHRISTIAN AND SURNAME	CHARCON	ORGANIZATIONAL UNIT
5002	HIPATH DT2*		

AMO-PERSI-111 PERSONAL IDENTIFICATION DATA
DISPLAY COMPLETED;

Siemens HiPath 4000 Software Release

DISPLAY-DBC:VERBOSE=N;
H500: AMO DBC STARTED

SYSTEM CLASSIFICATION	: SYSTEM 80	(H80)
HARDWARE ASSEMBLY	: EXTENDED COMPACT CXE	(CXE)
OPERATING MODE	: SIMPLEX	
RESTART TYPE	: SYM	



```
HW-ARCHITECTURE      : 4300
HW-ARCHITECTURE TYPE : 2

'NO OF' HW VALUES
  LTG'S      : 1  LTU'S      : 4  LOG.LINES : 12000  MTS BD /GSN: 1
  SIUP'S/LTU: 4  TMD24'S PER LTU: 4  PHYS.PORTS: 6000  HWY /MTS BD: 64
  HDLC /DCL : 5  PBC /DCL   : 1  PBC'S      : 17

LOG. SIU LINES      : 26
LOG. CONF LINES     : 35
LOG. DCL LINES      : 36
DB DIMENSIONING-NAME : SMALL          CONF-TABLE VERSION: 1
DB SUSY'S:
  SWITCH NUMBER : L31903Q1930A00001
LOCATION         : CUSTOMER
BAPPL          : BSMONO
DBAPPL         : DBSMALL
SYSTEM_ID      :

OVERLAY RESOURCES IN ADP:
  SLOTS        : 1000  MEMORY SPACE : 2000 KB
OVERLAY RESOURCES IN SWU:
  SLOTS        : 1000  MEMORY SPACE : 2000 KB
OVERLAY RESOURCES BEI MONO PROCESSING:
  SLOTS        : 400   MEMORY SPACE : 3000 KB
```

Application Note

```
AMO-DBC -111          DATABASE CONFIGURATION
DISPLAY COMPLETED;
```

DISPLAY-VEGAS:LIST=LONG;

```
H500: AMO VEGAS STARTED
      SYSTEM NO.      AMO  APS NO.      START      USER      STATUS
SWU: L31903Q1930A00001 REGEN P30252B4500B00108 07.06.09 22:01 CDBR    FREE
      SWU RES CODE APS: P30252B4500S00108 (DIR FILE: :PDS:APSI/PS/S0-EM0SC)
      SWU AMO CODE APS: P30252B4500B00108 (DIR FILE: :PDS:APSI/PS/B0-EM0BC)
      SWU AMO TEXT APS: P30252B4500B00108 (DIR FILE: :PDS:APSI/PS/B0-EM0BC)
      BREAK MARK      : NO
      RESERVATION     : NO
ADS: L31903Q1930A00001 REGEN P30252B4500A00108 07.06.09 22:02 CDBR    FREE
      ADS RES CODE APS: P30252B4500D00108 (DIR FILE: :PDS:APSI/PS/D0-EM0DC)
      ADS AMO CODE APS: P30252B4500A00108 (DIR FILE: :PDS:APSI/PS/A0-EM0AC)
      ADS AMO TEXT APS: P30252B4500A00108 (DIR FILE: :PDS:APSI/PS/A0-EM0AC)
      BREAK MARK      : NO
      RESERVATION     : NO
```

```
AMO-VEGAS-111        ADMIN. OF DATABASE GENERATION RUNS ON SUPPORT SYSTEM
DISPLAY COMPLETED;
```

DISPLAY-APS:ANS=Y,TYPE=PSALL,SP="Y0-EM0YC";

```
H500: AMO APS        STARTED
ADINIT STARTED
PROGRAM SYSTEM       : Y0-EM0YC
VERSION NUMBER       : 10
CORRECTION VERSION NUMBER : 001
PART NUMBER          : P30252N4508BH5402|V3.0 R8.2.54
PROGRAM SYSTEM WITH CODE SUBSYSTEMS
INTERFACE VERSION:
PROGRAM SYSTEM DOES NOT CONTAIN ANY INTERFACE VERSIONS
```

```
ADINIT COMPLETED
STATUS = H'0000
AMO-APS -111        SOFTWARE LOAD UPGRADE
DISPLAY COMPLETED;
```

HiPath 4000 Assistant Version 3.0

HiPath 4000 Assistant V3.0
Expert Mode

HG35xx Web Based Management

Board List			
PEN	Board Type	IP Address	Link
1-1-85	IGW	172.20.188.254	[connect]
1-1-103	IGW	172.20.188.253	[connect]
1-1-61	STMI-HFA	172.20.188.250	[connect]

The last update operation completed successfully
Last updated on: 2009-09-21 18:15:18

Update Board List

Connect: To connect to one of the listed boards, please click on the link in the far right column
Update: To start the 'Update board list' process, click on the 'Update' button above

Gateway Configuration



Front panel Wizard **Explorers** Maintenance Help Logoff

HG 3540/50 V2.0

Gateway Properties

General

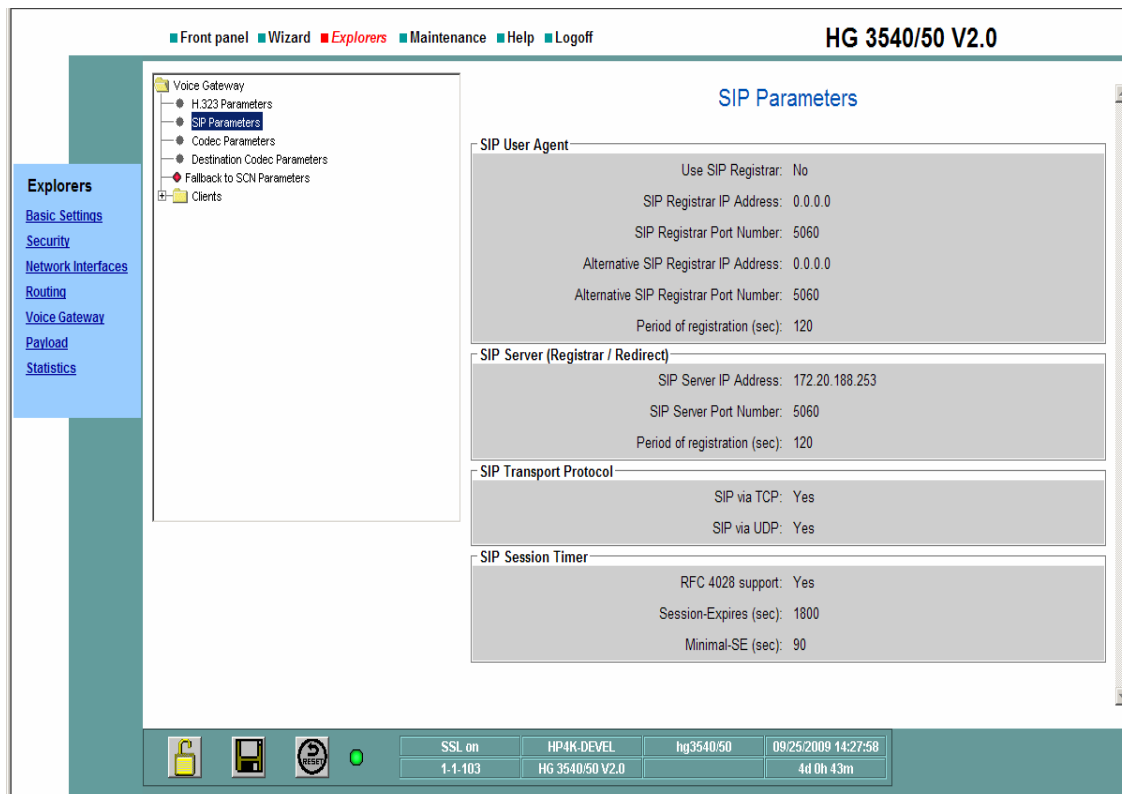
System Name: hg3540/50
 Gateway Location:
 Contact Address:
 System Country Code: 49 (Germany)
 Function Type: Trunking Gateway
 Gateway IP Address: 172.20.188.253
 Gateway Subnet Mask: 255.255.255.0

Additional Features

QoS - Fallback to SCN: No
 Conference Improvement: Yes
 Signaling Protocol for IP Networking: SIP
 SIP Protocolvariant for IP Networking: Native SIP
 Gatekeeper Type: default

SSL on	HP4K-DEVEL	hg3540/50	09/25/2009 14:25:28
1-1-103	HG 3540/50 V2.0		4d 0h 41m

SIP Parameters



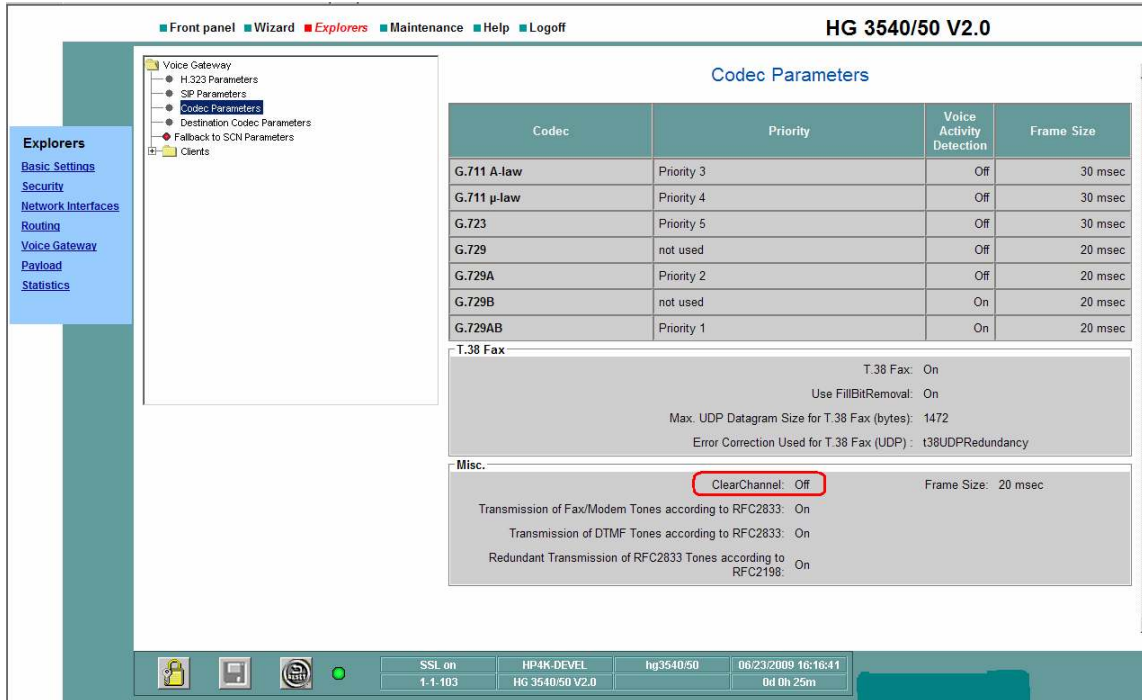
The screenshot displays the Cisco HG 3540/50 V2.0 web interface. The top navigation bar includes links for Front panel, Wizard, Explorers, Maintenance, Help, and Logoff. The main title is "HG 3540/50 V2.0". On the left, there is a sidebar with "Explorers" and a tree view showing the configuration hierarchy: Voice Gateway, H.323 Parameters, SIP Parameters (selected), Codec Parameters, Destination Codec Parameters, Fallback to SON Parameters, and Clients. The main content area is titled "SIP Parameters" and contains four sections:

- SIP User Agent**
 - Use SIP Registrar: No
 - SIP Registrar IP Address: 0.0.0.0
 - SIP Registrar Port Number: 5060
 - Alternative SIP Registrar IP Address: 0.0.0.0
 - Alternative SIP Registrar Port Number: 5060
 - Period of registration (sec): 120
- SIP Server (Registrar / Redirect)**
 - SIP Server IP Address: 172.20.188.253
 - SIP Server Port Number: 5060
 - Period of registration (sec): 120
- SIP Transport Protocol**
 - SIP via TCP: Yes
 - SIP via UDP: Yes
- SIP Session Timer**
 - RFC 4028 support: Yes
 - Session-Expires (sec): 1800
 - Minimal-SE (sec): 90

At the bottom of the interface, there is a status bar with icons for a lock, a floppy disk, and a reset button. To the right of these icons is a table with the following data:

SSL on	HP4K-DEVEL	hg3540/50	09/25/2009 14:27:58
1-1-103	HG 3540/50 V2.0		4d 0h 43m

CODEC parameter configuration



Front panel Wizard **Explorers** Maintenance Help Logoff HG 3540/50 V2.0

Explorers

- Basic Settings
- Security
- Network Interfaces
- Routing
- Voice Gateway
- Payload
- Statistics

Voice Gateway

- H.323 Parameters
- SIP Parameters
- CODEC Parameters**
- Destination Codec Parameters
- Fallback to SCN Parameters
- Clients

Codec Parameters

Codec	Priority	Voice Activity Detection	Frame Size
G.711 A-law	Priority 3	Off	30 msec
G.711 μ -law	Priority 4	Off	30 msec
G.723	Priority 5	Off	30 msec
G.729	not used	Off	20 msec
G.729A	Priority 2	Off	20 msec
G.729B	not used	On	20 msec
G.729AB	Priority 1	On	20 msec

T.38 Fax

T.38 Fax: On
 Use FillBitRemoval: On
 Max. UDP Datagram Size for T.38 Fax (bytes): 1472
 Error Correction Used for T.38 Fax (UDP): t38UDPRedundancy

Misc.

ClearChannel: **Off** Frame Size: 20 msec
 Transmission of Fax/Modem Tones according to RFC2833: On
 Transmission of DTMF Tones according to RFC2833: On
 Redundant Transmission of RFC2833 Tones according to RFC2198: On

SSL on HP4K-DEVEL hg3540/50 06/23/2009 16:16:41
 1-1-103 HG 3540/50 V2.0 0d 0h 25m

Configuring the Cisco Session Manager

1. Cisco Session Manager Version
2. Device pool and Region mapping configuration
3. Media Termination Point configuration
4. Media Resource Group configuration
5. Media Resource Group List configuration
6. Dual SIP Trunks configuration overview
7. G.711 SIP trunk configuration to Siemens
8. G.729 SIP trunk configuration to Siemens
9. G.711 SIP trunk configuration to CUCM
10. G.729 SIP trunk configuration to CUCM
11. G.711 SIP trunk configuration to SP
12. G.729 SIP trunk configuration to SP
13. Route Group configuration to Siemens
14. Route Group configuration to CUCM
15. Route Group configuration to SP
16. Route List configuration to Siemens
17. Route List configuration to CUCM
18. Route List configuration to SP
19. Route Pattern configuration to Siemens
20. Route Pattern configuration to CUCM
21. Route Pattern configuration to SP
22. Translation Pattern configuration (From Avaya towards SP)
23. Translation Pattern configuration (From SP towards Avaya/CUCM)



Application Note

Cisco Session Manager Version

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go

administrator | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Cisco Unified CM Administration

System version: 7.1.3.10000-11

Last Successful Logon: Nov 5, 2009 11:17:23 AM

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A summary of U.S. laws governing Cisco cryptographic products may be found at our [Export Compliance Product Report](#) web site.

For information about Cisco Unified Communications Manager please visit our [Unified Communications System Documentation](#) web site.

For Cisco Technical Support please visit our [Technical Support](#) web site.



Application Note

Configuration of Device Pool to Region mapping (Page 1 of 2)

Navigation Path: System → Region

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go

administrator | [About](#) | [Logout](#)

System ▾ | Call Routing ▾ | Media Resources ▾ | Voice Mail ▾ | Device ▾ | Application ▾ | User Management ▾ | Bulk Administration ▾ | Help ▾

Region Configuration Related Links: Back To Find/List Go

Save Delete Reset Apply Config Add New

Find and List Regions Information
Name*

Region Relationships

Region	Audio Codec	Video Call Bandwidth	Link Loss Type
Default_SME	G.711	384	Use System Default
Region_MGCP_G729	G.729	384	Use System Default
Region_SME_G729	G.711	384	Use System Default

NOTE: Regions(s) not displayed Use System Default Use System Default Use System Default

Modify Relationship to other Regions

Regions	Audio Codec	Video Call Bandwidth	Link Loss Type
<input type="text" value="Default_SME"/> <input type="text" value="Region_MGCP_G729"/> <input type="text" value="Region_SME_G729"/>	<input type="text" value="Keep Current Setting"/>	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="text" value=""/> kbps	<input type="text" value="Keep Current Setting"/>

Save Delete Reset Apply Config Add New



Application Note

Configuration of Device Pool to Region mapping (Page 2 of 2)

Navigation Path: System → Region

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go

administrator | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Region Configuration Related Links: Back To Find/List Go

Save Delete Reset Apply Config Add New

Find and List Regions Information
Name*

Region Relationships

Region	Audio Codec	Video Call Bandwidth	Link Loss Type
Default_SME	G.711	384	Use System Default
Region_MGCP_G729	G.729	384	Use System Default
Region_SME_G729	G.729	384	Use System Default

NOTE: Regions(s) not displayed Use System Default Use System Default Use System Default

Modify Relationship to other Regions

Regions	Audio Codec	Video Call Bandwidth	Link Loss Type
<input type="text" value="Default_SME"/> <input type="text" value="Region_MGCP_G729"/> <input type="text" value="Region_SME_G729"/>	Keep Current Setting	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="text" value=""/> kbps	Keep Current Setting

Save Delete Reset Apply Config Add New



Application Note

Configuration of the Media Termination Point

Navigation Path: Media Resources -> Media Termination Point

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and a navigation dropdown menu. Below this is a secondary navigation bar with links for System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled "Transcoder Configuration" and shows the configuration for a specific transcoder, MTP123456789012. The configuration is divided into two sections: "Transcoder Information" and "Media Termination Point Hardware Info". The "Transcoder Information" section shows the transcoder name, registration status, and IP addresses. The "Media Termination Point Hardware Info" section shows the transcoder type, description, MAC address, device pool, and common device configuration. At the bottom of the page, there are buttons for Save, Delete, Copy, Reset, Apply Config, and Add New.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go

administrator About Logout

System Call Routing Media Resources Voice Mail Device Application User Management Bulk Administration Help

Transcoder Configuration Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

Transcoder Information

Transcoder: MTP123456789012 (MTP123456789012)
Registration: Registered with Cisco Unified Communications Manager CM-Ferrari
IPv4 Address: 172.20.109.203
IPv6 Address: 0000:0000:0000:0000:0000:0000:0000:0000

Media Termination Point Hardware Info

Transcoder Type*: Cisco Media Termination Point Hardware
Description: MTP123456789012
MAC Address*: 123456789012
Device Pool*: Default View Details
Common Device Configuration: < None > View Details
Special Load Information: Leave blank to use default
☐ Trusted Relay Point

Save Delete Copy Reset Apply Config Add New

MTP configuration:

```
sccp local GigabitEthernet0/0
sccp ccm 172.20.109.252 identifier 1 version 7.0
sccp
!
sccp ccm group 1
bind interface GigabitEthernet0/0
associate ccm 1 priority 1
associate profile 50 register mtp123456789012
!
dspfarm profile 50 transcode
codec g711ulaw
codec g711alaw
codec g729ar8
codec g729abr8
codec ilbc
codec g723r63
codec g723r53
codec g729r8
codec g729br8
codec pass-through
maximum sessions 10
associate application SCCP
```



Application Note

Configuration of the Media Resource Group

Navigation Path: Media Resources -> Media Resource Group

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go

administrator | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Media Resource Group Configuration Related Links: Back To Find/List Go

Save Delete Copy Add New

Status
i Status: Ready

Media Resource Group Status
Media Resource Group: SME_MRG_HW (used by 6 devices)

Media Resource Group Information
Name*
Description

Devices for this Group
Available Media Resources**

ANN_3
CFB_3
MOH_3
MTP_2
MTP_3

Selected Media Resources*

ANN_2 (ANN)
CFB_2 (CFB)
MOH_2 (MOH)
MTP123456789012 (XCODE)

☐ Use Multi-cast for MOH Audio (If at least one multi-cast MOH resource is available)

Save Delete Copy Add New



Application Note

Configuration of the Media Resource Group List

Navigation Path: Media Resources -> Media Resource Group List

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation | Cisco Unified CM Administration | Go

administrator | About | Logout

System | Call Routing | Media Resources | Voice Mail | Device | Application | User Management | Bulk Administration | Help

Media Resource Group List Configuration | Related Links: Back To Find/List | Go

Save Delete Copy Add New

Status
 Status: Ready

Media Resource Group List Status
Media Resource Group List: SME_MRGL_HW (used by 6 devices)


Media Resource Group List Information
Name *

Media Resource Groups for this List
Available Media Resource Groups
Selected Media Resource Groups

Save Delete Copy Add New

Configuration of Dual SIP trunks - Overview






Navigation Path: Device -> Trunk


Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go
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Find and List Trunks

+ Add New |  Select All |  Clear All |  Delete Selected |  Reset Selected |  Apply Config to Selected

Status
i 6 records found

Trunks (1 - 6 of 6) Rows per Page: 50 ▾

Find Trunks where Device Name ▾ begins with ▾ SME Find Clear Filter + -
Select item or enter search text ▾

<input type="checkbox"/>	Name ^	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type	SIP Trunk Security Profile
<input type="checkbox"/>	 SME SIP CM-Polaris G711	SIP trunk to CM-Polaris using G.711		Default			RG_CM-Polaris	1	SIP Trunk	Non Secure SIP Trunk Profile
<input type="checkbox"/>	 SME SIP CM-Polaris G729	SIP trunk to CM-Polaris using G.729		DP_SME_G729			RG_CM-Polaris	2	SIP Trunk	Non Secure SIP Trunk Profile
<input type="checkbox"/>	 SME SIP CUBE G711	To PSTN using G711		Default			RG_External	1	SIP Trunk	Non Secure SIP Trunk Profile
<input type="checkbox"/>	 SME SIP CUBE G729	To PSTN using G729		DP_SME_G729			RG_External	2	SIP Trunk	Non Secure SIP Trunk Profile
<input type="checkbox"/>	 SME SIP Siemens Hipath G711	SIP trunk to Siemens Hipath using G.711		Default	1650123.502X				SIP Trunk	Non Secure SIP Trunk Profile
<input type="checkbox"/>	 SME SIP Siemens Hipath G729	SIP trunk to Siemens Hipath		DP_SME_G729					SIP Trunk	Non Secure SIP Trunk Profile

Add New | Select All | Clear All | Delete Selected | Reset Selected | Apply Config to Selected

Note: When SIP trunks are configured with “Media Termination Point Required”, CUCM performs SIP Early Media on outbound calls, advertising only the codec selected in parameter “MTP Preferred Originating Codec”. When Route Groups are configured with multiple SIP trunks performing SIP Early Media, outbound calls will advertise only the codec associated with the first choice trunk within the Route Group. If the first choice SIP trunk is the one configured for G.711 codec, all outbound calls will be placed using G.711 codec. Because of this, it is recommended that whenever “Media Termination Point Required” must be used (as with the Siemens HiPath 4000), and both G.711 and G.729 outbound calls must be supported, SIP trunks should not be configured in Route Groups. Instead, configure two different Route Patterns: one for G.711-only outbound calls (with the G.711-configured SIP trunk as its Gateway), and one for G.729-only calls (with the G.729-configured SIP trunk as its Gateway).



Application Note

Configuration of Dual SIP trunks to Siemens – G.711 (Page 1 of 3)

Navigation Path: Device -> Trunk

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Trunk Configuration Related Links: Back To Find/List Go

Save Delete Reset Apply Config Add New

Status

Status: Ready

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Device Name*	<input type="text" value="SME_SIP_Siemens_Hipath_G711"/>
Description	<input type="text" value="SIP trunk to Siemens Hipath using G.711"/>
Device Pool*	<input type="text" value="Default"/>
Common Device Configuration	<input type="text" value=" < None >"/>
Call Classification*	<input type="text" value="Use System Default"/>
Media Resource Group List	<input type="text" value="SME_MRGL_HW"/>
Location*	<input type="text" value="Hub_None"/>
AAR Group	<input type="text" value=" < None >"/>
Packet Capture Mode*	<input type="text" value="None"/>
Packet Capture Duration	<input type="text" value="0"/>
<input checked="" type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.	
Use Trusted Relay Point*	<input type="text" value="Default"/>

**Configuration of Dual SIP trunks to Siemens – G.711 (Page 2 of 3)****Navigation Path:** Device -> Trunk**Incoming Calling Party Settings**

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings**Default Prefix Settings**

Number Type	Prefix	Strip Digits	Use Device Pool CSS	Calling Search Space
Unknown Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input checked="" type="checkbox"/>	<input type="text" value=" < None >"/>

Multilevel Precedence and Preemption (MLPP) InformationMLPP Domain **Call Routing Information**☒ Remote-Party-Id☒ Asserted-IdentityAsserted-Type* SIP Privacy* **Inbound Calls**Significant Digits* Connected Line ID Presentation* Connected Name Presentation* Calling Search Space AAR Calling Search Space Prefix DN ☐ Redirecting Diversion Header Delivery - Inbound

**Configuration of Dual SIP trunks to Siemens – G.711 (Page 3 of 3)****Navigation Path:** Device -> Trunk


Outbound Calls	
Called Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Called Party Transformation CSS	
Calling Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS	
Calling Party Selection*	Originator
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Caller ID DN	
Caller Name	
<input type="checkbox"/> Redirecting Diversion Header Delivery - Outbound	

SIP Information	
Destination Address	172.20.188.253
Destination Address IPv6	
<input type="checkbox"/> Destination Address is an SRV	
Destination Port*	5060
MTP Preferred Originating Codec*	711ulaw
Presence Group*	Standard Presence group
SIP Trunk Security Profile*	Non Secure SIP Trunk Profile
Rerouting Calling Search Space	< None >
Out-Of-Dialog Refer Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	Standard SIP Profile
DTMF Signaling Method*	No Preference

Geolocation Configuration	
Geolocation	< None >
Geolocation Filter	< None >
<input type="checkbox"/> Send Geolocation Information	

Configuration of Dual SIP trunks to Siemens – G.729 (Page 1 of 3)






Navigation Path: Device -> Trunk



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Trunk Configuration Related Links: Back To Find/List Go

 Save  Delete  Reset  Apply Config  Add New

Status
 Status: Ready

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Device Name*	<input type="text" value="SME_SIP_Siemens_Hipath_G729"/>
Description	<input type="text" value="SIP trunk to Siemens Hipath"/>
Device Pool*	<input type="text" value="DP_SME_G729"/>
Common Device Configuration	<input type="text" value=" < None >"/>
Call Classification*	<input type="text" value=" Use System Default"/>
Media Resource Group List	<input type="text" value=" SME_MRGL_HW"/>
Location*	<input type="text" value=" Hub_None"/>
AAR Group	<input type="text" value=" < None >"/>
Packet Capture Mode*	<input type="text" value=" None"/>
Packet Capture Duration	<input type="text" value=" 0"/>
<input checked="" type="checkbox"/> Media Termination Point Required <input checked="" type="checkbox"/> Retry Video Call as Audio <input type="checkbox"/> Transmit UTF-8 for Calling Party Name <input type="checkbox"/> Unattended Port <input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information. Use Trusted Relay Point* <input type="text" value=" Default"/>	

Configuration of Dual SIP trunks to Siemens – G.729 (Page 2 of 3)

Navigation Path: Device -> Trunk

Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings
Default Prefix Settings

Number Type	Prefix	Strip Digits	Use Device Pool CSS	Calling Search Space
Unknown Number	Default	0	<input checked="" type="checkbox"/>	< None >

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain < None >

Call Routing Information

☒ Remote-Party-Id

☒ Asserted-Identity

Asserted-Type* Default

SIP Privacy* Default

Inbound Calls

Significant Digits*	All
Connected Line ID Presentation*	Default
Connected Name Presentation*	Default
Calling Search Space	< None >
AAR Calling Search Space	< None >
Prefix DN	

☒ Redirecting Diversion Header Delivery - Inbound

**Configuration of Dual SIP trunks to Siemens – G.729 (Page 3 of 3)****Navigation Path:** Device -> Trunk

Outbound Calls	
Called Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Called Party Transformation CSS	
Calling Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS	
Calling Party Selection*	Originator
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Caller ID DN	
Caller Name	
<input checked="" type="checkbox"/> Redirecting Diversion Header Delivery - Outbound	

SIP Information	
Destination Address	172.20.188.253
Destination Address IPv6	
<input type="checkbox"/> Destination Address is an SRV	
Destination Port*	5060
MTP Preferred Originating Codec*	G729/G729a
Presence Group*	Standard Presence group
SIP Trunk Security Profile*	Non Secure SIP Trunk Profile
Rerouting Calling Search Space	< None >
Out-Of-Dialog Refer Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	Standard SIP Profile
DTMF Signaling Method*	No Preference

Geolocation Configuration	
Geolocation	< None >
Geolocation Filter	< None >
<input type="checkbox"/> Send Geolocation Information	



Application Note

Configuration of Dual SIP trunks to CUCM – G.711 (Page 1 of 3)

Navigation Path: Device -> Trunk

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Trunk Configuration Related Links: Back To Find/List Go

Save Delete Reset Apply Config Add New

Status
 Status: Ready

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Device Name*	<input type="text" value="SME_SIP_CM-Polaris_G711"/>
Description	<input type="text" value="SIP trunk to CM-Polaris using G.711"/>
Device Pool*	Default
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	SME_MRGL_HW
Location*	Hub_None
AAR Group	< None >
Packet Capture Mode*	None
Packet Capture Duration	<input type="text" value="0"/>
<input type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.	
Use Trusted Relay Point*	Default

**Configuration of Dual SIP trunks to CUCM – G.711 (Page 2 of 3)****Navigation Path:** Device -> Trunk

Incoming Calling Party Settings				
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.				
Clear Prefix Settings		Default Prefix Settings		
Number Type	Prefix	Strip Digits	Use Device Pool CSS	Calling Search Space
Unknown Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input checked="" type="checkbox"/>	<input type="text" value="< None >"/>

Multilevel Precedence and Preemption (MLPP) Information	
MLPP Domain	<input type="text" value="< None >"/>

Call Routing Information	
<input checked="" type="checkbox"/> Remote-Party-Id	
<input checked="" type="checkbox"/> Asserted-Identity	
Asserted-Type*	<input type="text" value="Default"/>
SIP Privacy*	<input type="text" value="Default"/>

Inbound Calls	
Significant Digits*	<input type="text" value="All"/>
Connected Line ID Presentation*	<input type="text" value="Default"/>
Connected Name Presentation*	<input type="text" value="Default"/>
Calling Search Space	<input type="text" value="< None >"/>
AAR Calling Search Space	<input type="text" value="< None >"/>
Prefix DN	<input type="text"/>
<input type="checkbox"/> Redirecting Diversion Header Delivery - Inbound	

**Configuration of Dual SIP trunks to CUCM – G.711 (Page 3 of 3)****Navigation Path:** Device -> Trunk

Outbound Calls	
Called Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Called Party Transformation CSS	
Calling Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS	
Calling Party Selection*	Originator
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Caller ID DN	
Caller Name	
<input type="checkbox"/> Redirecting Diversion Header Delivery - Outbound	

SIP Information	
Destination Address	172.20.236.50
Destination Address IPv6	
<input type="checkbox"/> Destination Address is an SRV	
Destination Port*	5060
MTP Preferred Originating Codec*	711ulaw
Presence Group*	Standard Presence group
SIP Trunk Security Profile*	Non Secure SIP Trunk Profile
Rerouting Calling Search Space	< None >
Out-Of-Dialog Refer Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	Standard SIP Profile
DTMF Signaling Method*	No Preference

Geolocation Configuration	
Geolocation	< None >
Geolocation Filter	< None >
<input type="checkbox"/> Send Geolocation Information	



Application Note

Configuration of Dual SIP trunks to CUCM – G.729 (Page 1 of 3)

Navigation Path: Device -> Trunk

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Trunk Configuration Related Links: Back To Find/List Go

Save Delete Reset Apply Config Add New

Status
 Status: Ready

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Device Name*	<input type="text" value="SME_SIP_CM-Polaris_G729"/>
Description	<input type="text" value="SIP trunk to CM-Polaris using G.729"/>
Device Pool*	<input type="text" value="DP_SME_G729"/>
Common Device Configuration	<input type="text" value=" < None >"/>
Call Classification*	<input type="text" value="Use System Default"/>
Media Resource Group List	<input type="text" value="SME_MRGL_HW"/>
Location*	<input type="text" value="Hub_None"/>
AAR Group	<input type="text" value=" < None >"/>
Packet Capture Mode*	<input type="text" value="None"/>
Packet Capture Duration	<input type="text" value="0"/>
<input type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.	
Use Trusted Relay Point*	<input type="text" value="Default"/>

**Configuration of Dual SIP trunks to CUCM – G.729 (Page 2 of 3)****Navigation Path:** Device -> Trunk

Incoming Calling Party Settings				
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.				
Clear Prefix Settings		Default Prefix Settings		
Number Type	Prefix	Strip Digits	Use Device Pool CSS	Calling Search Space
Unknown Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input checked="" type="checkbox"/>	<input type="text" value="< None >"/>

Multilevel Precedence and Preemption (MLPP) Information	
MLPP Domain	<input type="text" value="< None >"/>

Call Routing Information	
<input checked="" type="checkbox"/> Remote-Party-Id	
<input checked="" type="checkbox"/> Asserted-Identity	
Asserted-Type*	<input type="text" value="Default"/>
SIP Privacy*	<input type="text" value="Default"/>

Inbound Calls	
Significant Digits*	<input type="text" value="All"/>
Connected Line ID Presentation*	<input type="text" value="Default"/>
Connected Name Presentation*	<input type="text" value="Default"/>
Calling Search Space	<input type="text" value="< None >"/>
AAR Calling Search Space	<input type="text" value="< None >"/>
Prefix DN	<input type="text"/>
<input checked="" type="checkbox"/> Redirecting Diversion Header Delivery - Inbound	

**Configuration of Dual SIP trunks to CUCM – G.729 (Page 3 of 3)****Navigation Path:** Device -> Trunk

Outbound Calls	
Called Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Called Party Transformation CSS	
Calling Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS	
Calling Party Selection*	Originator
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Caller ID DN	408853XXXX
Caller Name	
<input checked="" type="checkbox"/> Redirecting Diversion Header Delivery - Outbound	

SIP Information	
Destination Address	172.20.236.50
Destination Address IPv6	
<input type="checkbox"/> Destination Address is an SRV	
Destination Port*	5060
MTP Preferred Originating Codec*	G729/G729a
Presence Group*	Standard Presence group
SIP Trunk Security Profile*	Non Secure SIP Trunk Profile
Rerouting Calling Search Space	< None >
Out-Of-Dialog Refer Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	Standard SIP Profile
DTMF Signaling Method*	No Preference

Geolocation Configuration	
Geolocation	< None >
Geolocation Filter	< None >
<input type="checkbox"/> Send Geolocation Information	



Application Note

Configuration of Dual SIP trunks to PSTN – G.711 (Page 1 of 3)

Navigation Path: Device -> Trunk

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Trunk Configuration Related Links: Back To Find/List Go

Save Delete Reset Apply Config Add New

Status
 Status: Ready

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Device Name*	<input type="text" value="SME_SIP_CUBE_G711"/>
Description	<input type="text" value="To PSTN using G711"/>
Device Pool*	<input type="text" value="Default"/>
Common Device Configuration	<input type="text" value=" < None >"/>
Call Classification*	<input type="text" value="Use System Default"/>
Media Resource Group List	<input type="text" value="SME_MRGL_HW"/>
Location*	<input type="text" value="Hub_None"/>
AAR Group	<input type="text" value=" < None >"/>
Packet Capture Mode*	<input type="text" value="None"/>
Packet Capture Duration	<input type="text" value="0"/>
<input type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.	
Use Trusted Relay Point*	<input type="text" value="Default"/>

Configuration of Dual SIP trunks to PSTN – G.711 (Page 2 of 3)

Navigation Path: Device -> Trunk

Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings

Default Prefix Settings

Number Type	Prefix	Strip Digits	Use Device Pool CSS	Calling Search Space
Unknown Number	Default	0	<input checked="" type="checkbox"/>	< None >

Multilevel Precedence and Preemption (MLPP) Information
MLPP Domain < None >

Call Routing Information
☒ Remote-Party-Id
☒ Asserted-Identity
Asserted-Type* Default
SIP Privacy* Default

Inbound Calls
Significant Digits* All
Connected Line ID Presentation* Default
Connected Name Presentation* Default
Calling Search Space < None >
AAR Calling Search Space < None >
Prefix DN
☐ Redirecting Diversion Header Delivery - Inbound

**Configuration of Dual SIP trunks to PSTN – G.711 (Page 3 of 3)****Navigation Path:** Device -> Trunk

Outbound Calls	
Called Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Called Party Transformation CSS	
Calling Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS	
Calling Party Selection*	Originator
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Caller ID DN	
Caller Name	
<input type="checkbox"/> Redirecting Diversion Header Delivery - Outbound	

SIP Information	
Destination Address	172.20.109.203
Destination Address IPv6	
<input type="checkbox"/> Destination Address is an SRV	
Destination Port*	5060
MTP Preferred Originating Codec*	711ulaw
Presence Group*	Standard Presence group
SIP Trunk Security Profile*	Non Secure SIP Trunk Profile
Rerouting Calling Search Space	< None >
Out-Of-Dialog Refer Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	Standard SIP Profile
DTMF Signaling Method*	RFC 2833

Geolocation Configuration	
Geolocation	< None >
Geolocation Filter	< None >
<input type="checkbox"/> Send Geolocation Information	



Application Note

Configuration of Dual SIP trunks to PSTN – G.729 (Page 1 of 3)

Navigation Path: Device -> Trunk

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Trunk Configuration Related Links: Back To Find/List Go

Save Delete Reset Apply Config Add New

Status
 Status: Ready

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Device Name*	<input type="text" value="SME_SIP_CUBE_G729"/>
Description	<input type="text" value="To PSTN using G729"/>
Device Pool*	DP_SME_G729
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	< None >
Location*	Hub_None
AAR Group	< None >
Packet Capture Mode*	None
Packet Capture Duration	<input type="text" value="0"/>
<input type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.	
Use Trusted Relay Point*	Default

Configuration of Dual SIP trunks to PSTN – G.729 (Page 2 of 3)

Navigation Path: Device -> Trunk

Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings

Default Prefix Settings

Number Type	Prefix	Strip Digits	Use Device Pool CSS	Calling Search Space
Unknown Number	Default	0	<input checked="" type="checkbox"/>	< None >

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain

Call Routing Information

☒ Remote-Party-Id

☒ Asserted-Identity

Asserted-Type*

SIP Privacy*

Inbound Calls

Significant Digits*

Connected Line ID Presentation*

Connected Name Presentation*

Calling Search Space

AAR Calling Search Space

Prefix DN

☒ Redirecting Diversion Header Delivery - Inbound

**Configuration of Dual SIP trunks to PSTN – G.729 (Page 3 of 3)****Navigation Path:** Device -> Trunk

Outbound Calls	
Called Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Called Party Transformation CSS	
Calling Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS	
Calling Party Selection*	Originator
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Caller ID DN	
Caller Name	
<input checked="" type="checkbox"/> Redirecting Diversion Header Delivery - Outbound	

SIP Information	
Destination Address	172.20.109.203
Destination Address IPv6	
<input type="checkbox"/> Destination Address is an SRV	
Destination Port*	5060
MTP Preferred Originating Codec*	711ulaw
Presence Group*	Standard Presence group
SIP Trunk Security Profile*	Non Secure SIP Trunk Profile
Rerouting Calling Search Space	< None >
Out-Of-Dialog Refer Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	Standard SIP Profile
DTMF Signaling Method*	No Preference

Geolocation Configuration	
Geolocation	< None >
Geolocation Filter	< None >
<input type="checkbox"/> Send Geolocation Information	



Application Note

Configuration of Route Groups – To Siemens PBX

Navigation Path: Call Routing → Route/Hunt → Route Group

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Route Group Configuration Related Links: Back To Find/List Go

Save Delete Add New

Route Group Information

Route Group Name*

Distribution Algorithm* Top Down

Route Group Member Information

Find Devices to Add to Route Group

Device Name contains Find

Available Devices**

SME_SIP_CM-Polaris_G711

SME_SIP_CM-Polaris_G729

SME_SIP_CUBE_G711

SME_SIP_CUBE_G729

SME_SIP_Siemens_Hipath_G711

Port(s) All

Add to Route Group

Current Route Group Members

Selected Devices***

SME_SIP_Siemens_Hipath_G711 (All Ports)

SME_SIP_Siemens_Hipath_G729 (All Ports)

Reverse Order of Selected Devices

Removed Devices****

Route Group Members

[SME_SIP_Siemens_Hipath_G711](#)

[SME_SIP_Siemens_Hipath_G729](#)

Save Delete Add New



Application Note

Configuration of Route Groups – To CUCM

Navigation Path: Call Routing → Route/Hunt → Route Group

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and a navigation menu with options like System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The user is logged in as "administrator".

The main content area is titled "Route Group Configuration". It features a "Related Links" section with a "Back To Find/List" link. Below this are three main sections:

- Route Group Information:** Contains fields for "Route Group Name*" (set to "RG_CM-Polaris") and "Distribution Algorithm*" (set to "Top Down").
- Route Group Member Information:** Includes a "Find Devices to Add to Route Group" section with a "Device Name contains" field, a "Find" button, and a list of "Available Devices**". The list includes "SIP_Avaya_G711", "SIP_Avaya_G729", "SIP_CM-Telugu_G711", "SIP_CM-Telugu_G729", and "SME_SIP_CM-Polaris_G711". There is also a "Port(s)" dropdown set to "All" and an "Add to Route Group" button.
- Current Route Group Members:** Shows "Selected Devices***" with a list containing "SME_SIP_CM-Polaris_G711 (All Ports)" and "SME_SIP_CM-Polaris_G729 (All Ports)". A "Reverse Order of Selected Devices" button is present. Below this is a "Removed Devices****" section which is currently empty.

At the bottom, the "Route Group Members" section lists the current members: "SIP SME_SIP_CM-Polaris_G711" and "SIP SME_SIP_CM-Polaris_G729". At the very bottom of the interface are "Save", "Delete", and "Add New" buttons.



Application Note

Configuration of Route Groups – To PSTN

Navigation Path: Call Routing → Route/Hunt → Route Group

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go

administrator | About | Logout

System ▾ | Call Routing ▾ | Media Resources ▾ | Voice Mail ▾ | Device ▾ | Application ▾ | User Management ▾ | Bulk Administration ▾ | Help ▾

Route Group Configuration Related Links: Back To Find/List Go

Save Delete Add New

Route Group Information

Route Group Name *

Distribution Algorithm * Top Down

Route Group Member Information

Find Devices to Add to Route Group

Device Name contains Find

Available Devices **

SIP_Avaya_G711

SIP_Avaya_G729

SIP_CM-Telugu_G711

SIP_CM-Telugu_G729

SME_SIP_CM-Polaris_G711

Port(s) All

Add to Route Group

Current Route Group Members

Selected Devices ***

SME_SIP_CUBE_G711 (All Ports)

SME_SIP_CUBE_G729 (All Ports)

Reverse Order of Selected Devices

Removed Devices ****

Route Group Members

SME_SIP_CUBE_G711

SME_SIP_CUBE_G729

Save Delete Add New



Application Note

Configuration of Route Lists – To Siemens PBX

Navigation Path: Call Routing → Route/Hunt → Route List

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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Route List Configuration | Related Links: Back To Find/List | Go

Save | Delete | Copy | Reset | Apply Config | Add New

Status
Status: Ready

Route List Information
☒ Device is trusted
Name*
Description
Cisco Unified Communications Manager Group*
☒ Enable this Route List (change effective on Save; no reset required)

Route List Member Information
Selected Groups**
Removed Groups***

Route List Details
 [RG_Siemens](#)

Save | Delete | Copy | Reset | Apply Config | Add New



Application Note

Configuration of Route Lists – To CUCM

Navigation Path: Call Routing → Route/Hunt → Route List

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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Route List Configuration Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

Status
Status: Ready

Route List Information
☒ Device is trusted
Name* RL_CM-Polaris
Description Route List to CM-Polaris
Cisco Unified Communications Manager Group* Default
☒ Enable this Route List (change effective on Save; no reset required)

Route List Member Information
Selected Groups** RG_CM-Polaris
Add Route Group
Removed Groups***

Route List Details
 RG_CM-Polaris

Save Delete Copy Reset Apply Config Add New



Application Note

Configuration of Route Lists – To PSTN

Navigation Path: Call Routing → Route/Hunt → Route List

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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Route List Configuration Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

Status
Status: Ready

Route List Information
☒ Device is trusted
Name* RL_External
Description
Cisco Unified Communications Manager Group* Default
☒ Enable this Route List (change effective on Save; no reset required)

Route List Member Information
Selected Groups** RG_External
Add Route Group
Removed Groups***

Route List Details
RG_External

Save Delete Copy Reset Apply Config Add New



Configuration of Route Patterns – To Siemens PBX (Extensions and other endpoints using G.729 codec)

Navigation Path: Call Routing → Route/Hunt → Route Pattern

Application Note

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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Route Pattern Configuration

Related Links: Back To Find/List Go

Save Delete Copy Add New

Status

Status: Ready

Pattern Definition

Route Pattern*

500X

Route Partition

< None >

Description

Numbering Plan

-- Not Selected --

Route Filter

< None >

MLPP Precedence*

Default

Resource Priority Namespace Network Domain

< None >

Gateway/Route List*

SME_SIP_Siemens_Hipath_G729

(Edit)

Route Option

Route this pattern

Block this pattern

No Error

Call Classification*

OnNet

Allow Device Override

Provide Outside Dial Tone

Allow Overlap Sending

Urgent Priority

Require Forced Authorization Code

Authorization Level*

0

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Default

Calling Name Presentation*

Default

Calling Party Number Type*

Cisco CallManager

Calling Party Numbering Plan*

Cisco CallManager

Connected Party Transformations

Connected Line ID Presentation*

Default

Connected Name Presentation*

Default

Called Party Transformations

Discard Digits

< None >

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type*

Cisco CallManager

Called Party Numbering Plan*

Cisco CallManager

ISDN Network-Specific Facilities Information Element

Network Service Protocol

-- Not Selected --

Carrier Identification Code

Network Service

Service Parameter Name

Service Parameter Value

-- Not Selected --

< Not Exist >



Configuration of Route Patterns – To Siemens PBX (Fax/Modem lines using G.711 codec)

Navigation Path: Call Routing → Route/Hunt → Route Pattern

Application Note

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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Route Pattern Configuration Related Links: [Back To Find/List](#) | Go

Save Delete Copy Add New

Status
 Status: Ready

Pattern Definition
Route Pattern*
Route Partition
Description
Numbering Plan
Route Filter
MLPP Precedence*
Resource Priority Namespace Network Domain
Gateway/Route List* [\(Edit\)](#)
Route Option
☒ Route this pattern
☐ Block this pattern
Call Classification*
☐ Allow Device Override ☐ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority
☐ Require Forced Authorization Code
Authorization Level*
☐ Require Client Matter Code

Calling Party Transformations
☐ Use Calling Party's External Phone Number Mask
Calling Party Transform Mask
Prefix Digits (Outgoing Calls)
Calling Line ID Presentation*
Calling Name Presentation*
Calling Party Number Type*
Calling Party Numbering Plan*

Connected Party Transformations
Connected Line ID Presentation*
Connected Name Presentation*

Called Party Transformations
Discard Digits
Called Party Transform Mask
Prefix Digits (Outgoing Calls)
Called Party Number Type*
Called Party Numbering Plan*

ISDN Network-Specific Facilities Information Element
Network Service Protocol
Carrier Identification Code

Network Service	Service Parameter Name	Service Parameter Value
<input type="text" value="-- Not Selected --"/>	<input type="text" value=" < Not Exist >"/>	<input type="text"/>



Configuration of Route Patterns – To CUCM

Navigation Path: Call Routing → Route/Hunt → Route Pattern

Application Note

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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Route Pattern Configuration

Related Links: Back To Find/List Go

Save Delete Copy Add New

Status

Status: Ready

Pattern Definition

Route Pattern*

501X

Route Partition

< None >

Description

Numbering Plan

-- Not Selected --

Route Filter

< None >

MLPP Precedence*

Default

Resource Priority Namespace Network Domain

< None >

Gateway/Route List*

RL_CM-Polaris

(Edit)

Route Option

Route this pattern

Block this pattern No Error

Call Classification*

OnNet

Allow Device Override

Provide Outside Dial Tone

Allow Overlap Sending

Urgent Priority

Require Forced Authorization Code

Authorization Level*

0

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Default

Calling Name Presentation*

Default

Calling Party Number Type*

Cisco CallManager

Calling Party Numbering Plan*

Cisco CallManager

Connected Party Transformations

Connected Line ID Presentation*

Default

Connected Name Presentation*

Default

Called Party Transformations

Discard Digits

< None >

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type*

Cisco CallManager

Called Party Numbering Plan*

Cisco CallManager

ISDN Network-Specific Facilities Information Element

Network Service Protocol

-- Not Selected --

Carrier Identification Code

Network Service

Service Parameter Name

Service Parameter Value



Configuration of Route Patterns – To PSTN (G.729 and/or G.711 calls)

Navigation Path: Call Routing → Route/Hunt → Route Pattern

Application Note

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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Route Pattern Configuration Related Links: Back To Find/List Go

Save Delete Copy Add New

Status
 Status: Ready

Pattern Definition
Route Pattern*
Route Partition
Description
Numbering Plan
Route Filter
MLPP Precedence*
Resource Priority Namespace Network Domain
Gateway/Route List* [\(Edit\)](#)
Route Option
☒ Route this pattern
☐ Block this pattern
Call Classification*
☐ Allow Device Override ☒ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority
☐ Require Forced Authorization Code
Authorization Level*
☐ Require Client Matter Code

Calling Party Transformations
☐ Use Calling Party's External Phone Number Mask
Calling Party Transform Mask
Prefix Digits (Outgoing Calls)
Calling Line ID Presentation*
Calling Name Presentation*
Calling Party Number Type*
Calling Party Numbering Plan*

Connected Party Transformations
Connected Line ID Presentation*
Connected Name Presentation*

Called Party Transformations
Discard Digits
Called Party Transform Mask
Prefix Digits (Outgoing Calls)
Called Party Number Type*
Called Party Numbering Plan*

ISDN Network-Specific Facilities Information Element
Network Service Protocol
Carrier Identification Code
Network Service Service Parameter Name Service Parameter Value



Configuration of Route Patterns – To PSTN (G.711 only calls)

Navigation Path: Call Routing → Route/Hunt → Route Pattern

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Route Pattern Configuration Related Links: Back To Find/List Go

Save Delete Copy Add New

Status
 Status: Ready

Pattern Definition
Route Pattern* 81XXXXXXXXXX
Route Partition < None >
Description
Numbering Plan -- Not Selected --
Route Filter < None >
MLPP Precedence* Default
Resource Priority Namespace Network Domain < None >
Gateway/Route List* RL_External (Edit)
Route Option
☒ Route this pattern
☐ Block this pattern No Error
Call Classification* OffNet
☐ Allow Device Override ☒ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority
☐ Require Forced Authorization Code
Authorization Level* 0
☐ Require Client Matter Code

Calling Party Transformations
☐ Use Calling Party's External Phone Number Mask
Calling Party Transform Mask
Prefix Digits (Outgoing Calls)
Calling Line ID Presentation* Default
Calling Name Presentation* Default
Calling Party Number Type* Cisco CallManager
Calling Party Numbering Plan* Cisco CallManager

Called Party Transformations
Discard Digits < None >
Called Party Transform Mask
Prefix Digits (Outgoing Calls)
Called Party Number Type* Cisco CallManager
Called Party Numbering Plan* Cisco CallManager

ISDN Network-Specific Facilities Information Element
Network Service Protocol -- Not Selected --
Carrier Identification Code
Network Service Service Parameter Name Service Parameter Value
-- Not Selected -- < Not Exist >

Save Delete Copy Add New

Note: This example shows an optional Route Pattern configured for G.711-only outbound calls. Typically, this would be used for fax/modem transmissions, when G.711 codec is required for successful transmission. When fax/modem calls must be placed, using this example “8” must be dialed before the 10-digit telephone number, instead of the standard outside dial



access code “9”. Alternatively, a Route Pattern not requiring having to dial a different leading digit can be implemented.


This Route Pattern is configured using a unique Route Partition, with parameter “Called Party Transformation Prefix Digit (Outgoing Calls)” configured with a prefix digit (using the CUBE example configuration shown in the previous pages, this prefix would be “8”) . This partition is then assigned to a Calling Search Space that is assigned to fax machines/modems. When an outside telephone number is dialed using lines associated with this newly-created Calling Search Space, the Route Pattern assigned to this different partition is used in place of the standard outside dial access Route Pattern. Also, to ensure that inbound fax/modem calls are established using G.711, configure SIP trunks and/or MGCP/H.323 gateways supporting fax/modems into a Region using G.711 codec.

Application Note

Configuring the Cisco Unified Communications Manager

1. Cisco Unified Communications Manager Version
2. Device pool and Region mapping configuration
3. Conference Bridge configuration
4. Media Resource Group configuration
5. Media Resource Group List configuration
6. Cisco IP Phone 7960 SCCP Configuration
7. Cisco IP Phone 7960 SIP Configuration
8. SIP Trunks configuration overview
9. SIP Trunk configuration to IOS GW
10. G.711 SIP Trunk configuration to SME
11. G.729 SIP Trunk configuration to SME
12. Route Pattern configuration to IOS GW
13. Route Group configuration to SME
14. Route List configuration to SME
15. Route Pattern configuration to SP
16. Route Pattern configuration to Avaya
17. Route Pattern configuration to Avaya FAX
18. Calling Line ID restriction configuration



**Cisco Unified CM Administration**
For Cisco Unified Communications Solutions


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Cisco Unified CM Administration

System version: 7.1.3.10000-11



Last Successful Logon: Nov 10, 2009 10:12:20 AM

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A summary of U.S. laws governing Cisco cryptographic products may be found at our [Export Compliance Product Report](#) web site.

For information about Cisco Unified Communications Manager please visit our [Unified Communications System Documentation](#) web site.

For Cisco Technical Support please visit our [Technical Support](#) web site.



Configuration of Device Pool to Region mapping (Page 1 of 2)

Navigation Path: System → Region

Application Note

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Region Configuration Related Links: Back To Find/List Go

Save Delete Reset Apply Config Add New

Region Information

Name*

Region Relationships

Region	Audio Codec	Video Call Bandwidth	Link Loss Type
Default	G.729	800	Use System Default
G711_Region	G.711	384	Use System Default
NOTE: Region(s) not displayed Use System Default Use System Default Use System Default			

Modify Relationship to other Regions

Regions	Audio Codec	Video Call Bandwidth	Link Loss Type
<div><div>Default</div><div>G711_Region</div></div>	<div>Keep Current Setting ▾</div>	<div><input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="text" value=""/> kbps</div>	<div>Keep Current Setting ▾</div>

Save

Delete

Reset

Apply Config

Add New



Configuration of Device Pool to Region mapping (Page 2 of 2)

Navigation Path: System → Region

Application Note

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Region Configuration Related Links: Back To Find/List Go

Save Delete Reset Apply Config Add New

Region Information
Name*

Region Relationships

Region	Audio Codec	Video Call Bandwidth	Link Loss Type
Default	G.711	384	Use System Default
G711_Region	G.711	384	Use System Default
NOTE: Regions(s) not displayed Use System Default Use System Default Use System Default			

Modify Relationship to other Regions

Regions	Audio Codec	Video Call Bandwidth	Link Loss Type
<div>Default G711_Region</div>	<div>Keep Current Setting</div>	<div><input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="text" value=""/> kbps</div>	<div>Keep Current Setting</div>

Save Delete Reset Apply Config Add New



Configuration of Conference Bridge

Navigation Path: Media Resources → Conference Bridge

Application Note

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For Cisco Unified Communications Solutions

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Conference Bridge Configuration Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

Status
Status: Ready

Conference Bridge Information
Conference Bridge : CFB112233445566 (Conference Bridge on IOS DSP Farm)
Registration Registered with Cisco Unified Communications Manager CM-Polaris
IPv4 Address 172.20.236.101

IOS Conference Bridge Info
Conference Bridge Type* Cisco IOS Enhanced Conference Bridge
☒ Device is trusted
Conference Bridge Name* CFB112233445566
Description Conference Bridge on IOS DSP Farm
Device Pool* Default
Common Device Configuration < None >
Location* Hub_None
Device Security Mode* Non Secure Conference Bridge
Use Trusted Relay Point* Default

Save Delete Copy Reset Apply Config Add New

Conference Bridge configuration:

```
sccp local GigabitEthernet0/0
sccp ccm 172.20.236.50 identifier 1 version 7.0
sccp
!
sccp ccm group 1
bind interface GigabitEthernet0/0
associate ccm 1
priority 1
associate profile 98 register cfb112233445566
!
dspfarm profile 98 conference
codec g729r8
codec g711ulaw
maximum sessions 8
associate application SCCP
```



Configuration of Media Resource Group

Navigation Path: Media Resources → Media Resource Group

Application Note

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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Media Resource Group Configuration | Related Links: Back To Find/List | Go

Save Delete Copy Add New

Status
 Status: Ready

Media Resource Group Status
Media Resource Group: MRG_Polaris (used by 115 devices)

Media Resource Group Information
Name* MRG_Polaris
Description MRG_Polaris

Devices for this Group
Available Media Resources**
CFB0001C9D93A99 (CFB)
CFB_2 (CFB)
MTP0001C9D93A9E (XCODE)
MTP0015F90D0970 (XCODE)
MTP_2 (MTP)
▼ ▲
Selected Media Resources*
ANN_2 (ANN)
CFB112233445566 (CFB)
MOH_2 (MOH)
☐ Use Multi-cast for MOH Audio (If at least one multi-cast MOH resource is available)

Save Delete Copy Add New



Configuration of Media Resource Group List

Navigation Path: Media Resources → Media Resource Group List

Application Note

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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Media Resource Group List Configuration

Related Links: Back To Find/List ▾ | Go

Save Delete Copy Add New

Status

Status: Ready

Media Resource Group List Status

Media Resource Group List: MRGL_Polaris (used by 115 devices)

Media Resource Group List Information

Name *

Media Resource Groups for this List

Available Media Resource Groups

Selected Media Resource Groups

MRG_Polaris

Save Delete Copy Add New



Configuration of Cisco SCCP 7970 Phone (Page 1 of 5)

Navigation Path: Device → Phone

Application Note

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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Phone Configuration Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

Status
 Status: Ready

Association Information
[Modify Button Items](#)

1	Line [1] - 5015 in phones
2	Line [2] - Add a new DN
3	Add a new SD
4	Add a new SD
5	Add a new SD
6	Add a new SD
7	Add a new SD
8	Add a new SD
----- Unassigned Associated Items -----	
9	Add a new SD
10	Add a new SURL
11	Add a new BLF SD
12	Add a new BLF Directed Call Park
13	CallBack
14	Call Park
15	Call Pickup
16	Conference List
17	Conference
18	Do Not Disturb
19	End Call
20	Forward All
21	Group Call Pickup

Phone Type
Product Type: Cisco 7970
Device Protocol: SCCP

Device Information

Registration	Registered with Cisco Unified Communications Manager CM-Polaris
IP Address	172.20.236.32
Active Load ID	SCCP70.8-5-2S
<input checked="" type="checkbox"/> Is Active	
MAC Address*	000E839C1543
Description	Cisco 7970 SCCP - Bench 8 (5005)
Device Pool*	Default View Details
Common Device Configuration	MigratedCommonDeviceConfig1 View Details
Phone Button Template*	Standard 7970 SCCP
Softkey Template	Qsig Custom
Common Phone Profile*	Standard Common Phone Profile
Calling Search Space	tp_phones_rp
AAR Calling Search Space	tp_phones_rp
Media Resource Group List	MRGL_Polaris
User Hold MOH Audio Source	1-SampleAudioSource
Network Hold MOH Audio Source	1-SampleAudioSource
Location*	Hub_None
AAR Group	< None >
User Locale	< None >
Network Locale	< None >
Built In Bridge*	Default
Privacy*	Default



Configuration of Cisco SCCP 7970 Phone (Page 2 of 5)

Navigation Path: Device → Phone

Application Note

22	Hold	Device Mobility Mode*	Default	View Current Device
23	Hunt Group Logout			
24	Intercom [1] - Add a new Intercom	Owner User ID	tenacity	
25	Malicious Call Identification	Phone Personalization*	Default	
26	Meet Me Conference	Services Provisioning*	Both	
27	Mobility	Phone Load Name		
28	New Call	Single Button Barge	Default	
29	Other Pickup	Join Across Lines	Default	
30	Quality Reporting Tool	Use Trusted Relay Point*	Default	
31	Redial	BLF Audible Alert Setting (Phone Idle)*	Default	
32	Remove Last Participant	BLF Audible Alert Setting (Phone Busy)*	Default	
33	Transfer	Always Use Prime Line*	Default	
34	Video Mode	Always Use Prime Line for Voice Message*	Default	
35	Privacy	Calling Party Transformation CSS	< None >	
36	None	Geo Location	< None >	
		<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS		
		<input checked="" type="checkbox"/> Retry Video Call as Audio		
		<input type="checkbox"/> Ignore Presentation Indicators (internal calls only)		
		<input checked="" type="checkbox"/> Allow Control of Device from CTI		
		<input checked="" type="checkbox"/> Logged Into Hunt Group		
		<input type="checkbox"/> Remote Device		
		<input type="checkbox"/> Protected Device ****		



Configuration of Cisco SCCP 7970 Phone (Page 3 of 5)

Navigation Path: Device → Phone

Application Note

Protocol Specific Information	
Packet Capture Mode*	None
Packet Capture Duration	60
Presence Group*	Standard Presence group
Device Security Profile*	Cisco 7970 - Standard SCCP Secure Profile
SUBSCRIBE Calling Search Space	< None >
<input type="checkbox"/> Unattended Port	
<input checked="" type="checkbox"/> Require DTMF Reception	
<input type="checkbox"/> RFC2833 Disabled	

Certification Authority Proxy Function (CAPF) Information	
Certificate Operation*	No Pending Operation
Authentication Mode*	By Null String
Authentication String	
<input type="button" value="Generate String"/>	
Key Size (Bits)*	1024
Operation Completes By	2009 11 15 12 (YYYY:MM:DD:HH)
Certificate Operation Status: None	
Note: Security Profile Contains Addition CAPF Settings.	

Expansion Module Information	
Module 1	< None >
Module 1 Load Name	
Module 2	< None >
Module 2 Load Name	

External Data Locations Information (Leave blank to use default)	
Information	
Directory	
Messages	
Services	
Authentication Server	



Configuration of Cisco SCCP 7970 Phone (Page 4 of 5)

Navigation Path: Device → Phone

Application Note

Proxy Server	<input type="text"/>
Idle	<input type="text"/>
Idle Timer (seconds)	<input type="text"/>

Extension Information

☐ Enable Extension Mobility

Log Out Profile

Log in Time

Log out Time

MLPP Information

MLPP Domain

MLPP Indication*

MLPP Preemption*

Do Not Disturb

☐ Do Not Disturb

DND Option*

DND Incoming Call Alert

Secure Shell Information

Secure Shell User

Secure Shell Password

Product Specific Configuration Layout



☐ Disable Speakerphone

☐ Disable Speakerphone and Headset

Forwarding Delay*

PC Port*

Settings Access*

Gratuitous ARP*

PC Voice VLAN Access*

Video Capabilities*

Auto Line Select*

Web Access*

Days Display Not Active

Display On Time

Display On Duration

Display Idle Timeout

Span to PC Port*

Logging Display*

Load Server

Recording Tone*

Recording Tone Local Volume*

Recording Tone Remote Volume*

Recording Tone Duration

Display On When Incoming Call*

RTCP*



Configuration of Cisco SCCP 7970 Phone (Page 5 of 5)

Navigation Path: Device → Phone

Application Note

"more" Soft Key Timer	5
Auto Call Select*	Enabled
Log Server	
Advertise G.722 Codec*	Disabled
Wideband Headset UI Control*	Enabled
Wideband Handset UI Control*	Enabled
Wideband Headset*	Enabled
Wideband Handset*	Use Phone Default
Peer Firmware Sharing*	Disabled
Cisco Discovery Protocol (CDP): Switch Port*	Enabled
Cisco Discovery Protocol (CDP): PC Port*	Enabled
Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port*	Enabled
Link Layer Discovery Protocol (LLDP): PC Port*	Enabled
LLDP Asset ID	
LLDP Power Priority*	Unknown
IPv6 Load Server	
IPv6 Log Server	
802.1x Authentication*	User Controlled
Detect Unified CM Connection Failure*	Normal
Minimum Ring Volume*	0-Silent
Headset Sidetone Level*	Use Phone Default

Save Delete Copy Reset Apply Config Add New

Configuration of Cisco SIP 7961 Phone (Page 1 of 4)

Navigation Path: Device → Phone

Application Note

Cisco Unified CM Administration
 For Cisco Unified Communications Solutions

Navigation
 Cisco Unified CM Administration
Go

ccmadministrator
About
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System
Call Routing
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Phone Configuration
Related Links:
Back To Find/List

Save
Delete
Copy
Reset
Apply Config
Add New

Status
 Status: Ready

Association Information
Modify Button Items

1	Line [1] - 5013 in phones
2	Line [2] - Add a new DN
3	Add a new SD
4	Add a new SD
5	Add a new SD
6	Add a new SD
----- Unassigned Associated Items -----	
7	Add a new SD
8	Add a new SURL
9	Add a new BLF SD
10	Add a new BLF Directed Call Park
11	Intercom [1] - Add a new Intercom
12	Do Not Disturb
13	Call Park
14	Call Pickup
15	CallBack
16	Conference List
17	Conference
18	End Call
19	Forward All
20	Group Call Pickup
21	Hold
22	Hunt Group Logout
23	Malicious Call Identification
24	Meet Me Conference

Phone Type
Product Type: Cisco 7961
Device Protocol: SIP

Device Information

Registration	Unregistered
IPv4 Address	172.20.236.24
Active Load ID	Unknown
<input checked="" type="checkbox"/> Device is Active	
<input checked="" type="checkbox"/> Device is trusted	
MAC Address*	00152B340DC4
Description	SIP Phone - 5003
Device Pool*	G711_Pool View Details
Common Device Configuration	< None > View Details
Phone Button Template*	Standard 7961 SIP
Softkey Template	Standard Manager
Common Phone Profile*	Standard Common Phone Profile
Calling Search Space	tp_phones_rp
AAR Calling Search Space	< None >
Media Resource Group List	MRGL_Polaris
User Hold MOH Audio Source	< None >
Network Hold MOH Audio Source	< None >
Location*	Hub_None
AAR Group	< None >
User Locale	< None >
Network Locale	< None >
Built In Bridge*	Default
Privacy*	Default
Device Mobility Mode*	Default View Current Device



Configuration of Cisco SIP 7961 Phone (Page 2 of 4)

Navigation Path: Device → Phone

Application Note

25	Mobility	Owner User ID	< None >
26	New Call	Phone Suite*	Default
27	Other Pickup	Services Provisioning*	Default
28	Quality Reporting Tool	Phone Load Name	SIP41.8-5-2S
29	Redial	Single Button Barge	Default
30	Remove Last Participant	Join Across Lines	Default
31	Transfer	Use Trusted Relay Point*	Default
32	Privacy	BLF Audible Alert Setting (Phone Idle)*	Default
33	None	BLF Audible Alert Setting (Phone Busy)*	Default
		Always Use Prime Line*	Default
		Always Use Prime Line for Voice Message*	Default
		Calling Party Transformation CSS	< None >
		Geolocation	< None >
		<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS	
		<input type="checkbox"/> Ignore Presentation Indicators (internal calls only)	
		<input checked="" type="checkbox"/> Allow Control of Device from CTI	
		<input checked="" type="checkbox"/> Logged Into Hunt Group	
		<input type="checkbox"/> Remote Device	
		<input type="checkbox"/> Protected Device ****	

Protocol Specific Information	
Packet Capture Mode*	None
Packet Capture Duration	0
Presence Group*	Standard Presence group
SIP Dial Rules	< None >
MTP Preferred Originating Codec*	711ulaw
Device Security Profile*	Cisco 7961 - Standard SIP Non-Secure Profile
Rerouting Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	Standard SIP Profile
Digest User	< None >
<input type="checkbox"/> Media Termination Point Required	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> Require DTMF Reception	

Certification Authority Proxy Function (CAPF) Information	
Certificate Operation*	No Pending Operation
Authentication Mode*	By Null String
Authentication String	
<input type="button" value="Generate String"/>	
Key Size (Bits)*	1024
Operation Completes By	2009 11 20 12 (YYYY:MM:DD:HH)
Certificate Operation Status: None	
Note: Security Profile Contains Addition CAPF Settings.	

Expansion Module Information	
Module 1	< None >
Module 1 Load Name	
Module 2	< None >
Module 2 Load Name	



Configuration of Cisco SIP 7961 Phone (Page 3 of 4)

Navigation Path: Device → Phone

Application Note

External Data Locations Information (Leave blank to use default)	
Information	<input type="text"/>
Directory	<input type="text"/>
Messages	<input type="text"/>
Services	<input type="text"/>
Authentication Server	<input type="text"/>
Proxy Server	<input type="text"/>
Idle	<input type="text"/>
Idle Timer (seconds)	<input type="text"/>

Extension Information	
<input type="checkbox"/> Enable Extension Mobility	
Log Out Profile	-- Use Current Device Settings --
Log in Time	< None >
Log out Time	< None >

MLPP Information	
MLPP Domain	< None >

Do Not Disturb	
<input type="checkbox"/> Do Not Disturb	
DND Option*	Use Common Phone Profile Setting
DND Incoming Call Alert	< None >

Secure Shell Information	
Secure Shell User	phoneadmin
Secure Shell Password	*****



Configuration of Cisco SIP 7961 Phone (Page 4 of 4)

Navigation Path: Device → Phone

Application Note

Product Specific Configuration Layout



<input type="checkbox"/> Disable Speakerphone	
<input type="checkbox"/> Disable Speakerphone and Headset	
PC Port *	Enabled
Settings Access*	Enabled
Gratuitous ARP*	Disabled
PC Voice VLAN Access*	Enabled
Auto Line Select*	Disabled
Web Access*	Enabled
Span to PC Port*	Disabled
Logging Display *	PC Controlled
Load Server	
Recording Tone*	Disabled
Recording Tone Local Volume*	100
Recording Tone Remote Volume*	50
Recording Tone Duration	
RTCP*	Disabled
"more" Soft Key Timer	5
Auto Call Select*	Enabled
Log Server	
Advertise G.722 Codec*	Use System Default
Wideband Headset UI Control*	Enabled
Wideband Handset UI Control*	Enabled
Wideband Headset*	Enabled
Wideband Handset*	Use Phone Default
Peer Firmware Sharing*	Disabled
Cisco Discovery Protocol (CDP): Switch Port*	Enabled
Cisco Discovery Protocol (CDP): PC Port*	Enabled
Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port*	Enabled
Link Layer Discovery Protocol (LLDP): PC Port*	Enabled
LLDP Asset ID	
LLDP Power Priority*	Unknown
Display Refresh Rate*	Normal
IPv6 Load Server	
IPv6 Log Server	
802.1x Authentication*	User Controlled
Detect Unified CM Connection Failure*	Normal
Minimum Ring Volume*	0-Silent
Headset Sidetone Level*	Use Phone Default
Enbloc Dialing*	Enabled



Configuration of Dual SIP trunks to SME - Overview

Navigation Path: Device → Trunk

Application Note

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go

ccmadministrator | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Find and List Trunks

Add New Select All Clear All Delete Selected Reset Selected Apply Config to Selected

Status
 2 records found

Trunks (1 - 2 of 2) Rows per Page 50 ▾

Find Trunks where Device Name ▾ begins with ▾ Find Clear Filter

Select item or enter search text ▾

<input type="checkbox"/>	Name ▲	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type	SIP Trunk Security Profile
<input type="checkbox"/>	Sess Mgr SIP_trunk_G711	SIP Trunk to Session Manager G.711	tp_phones_rp	G711_Pool			SME_Route_Group	1	SIP Trunk	SIP_Trunks
<input type="checkbox"/>	Sess Mgr SIP_trunk_G729	SIP trunk to Session Manager	tp_phones_rp	Default			SME_Route_Group	2	SIP Trunk	SIP_Trunks

Add New Select All Clear All Delete Selected Reset Selected Apply Config to Selected



Configuration of FAX Gateway (Page 1 of 2)

Navigation Path: Device → Gateway

Application Note

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go

ccmadministrator | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Gateway Configuration Related Links: Back To Find/List Go

Save Delete Reset Apply Config Add New

Status
Status: Ready

Gateway Details
Product Cisco 3825
Gateway 3825DSPfarm.pbxmlab.org
Protocol MGCP
Domain Name* 3825DSPfarm.pbxmlab.org
Description 3825 - Bench 8 - 172.20.236.101
Cisco Unified Communications Manager Group* Default

Configured Slots, VICs and Endpoints
Module in Slot 0 NM-4VWIC-MBRD
Subunit 0 VIC2-2FXO 0/0/ 0 0/0/ 1
Subunit 1 VIC2-2FXS 0/1/ 0 0/1/ 1
Subunit 2 < None >
Subunit 3 < None >
Module in Slot 1 < None >
Module in Slot 2 NM-HDV
Subunit 0 VWIC-2MFT-T1 2/0/ 0 2/0/ 1

Product Specific Configuration Layout
Global ISDN Switch Type 4ESS
Switchback Timing* Graceful
Switchback uptime-delay (min) 10
Switchback schedule (hh:mm) 12:00
Type Of DTMF Relay* Current GW Config
Modem Passthrough* Enable
Cisco Fax Relay* Disable
T38 Fax Relay* Enable
RTP Package Capability* Enable
MT Package Capability* Disable
RES Package Capability* Disable
PRE Package Capability* Enable
SST Package Capability* Enable
RTP Unreachable OnOff* Enable
RTP Unreachable timeout (ms)* 1000
RTCP Report Interval (secs)* 0
Simple SDP* Enable

Save Delete Reset Apply Config Add New



Configuration of FAX Gateway Analog Endpoint (Page 2 of 2)

Navigation Path: Device → Gateway

Application Note

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation **Cisco Unified CM Administration** Go

ccmadministrator | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Gateway Configuration Related Links: **Back to MGCP Configuration** Go

Save Delete Reset Apply Config Add New

Status
Status: Ready

Directory Number Information
[Line \[1\] - 5014 in phones](#)

Device Information

Product	Cisco MGCP FXS Port
Gateway	3825DSPfarm.pbxmlab.org
Device Protocol	Analog Access
Registration	Registered with Cisco Unified Communications Manager CM-Polaris
IP Address	172.20.236.101
End-Point Name *	AALN/S0/SU1/0@3825DSPfarm.pbxmlab.org
Description	AALN/S0/SU1/0@3825DSPfarm.pbxmlab.org
Device Pool*	G711_Pool
Common Device Configuration	< None >
Media Resource Group List	MRGL_Polaris
Packet Capture Mode*	None
Packet Capture Duration	0
Calling Search Space	tp_phones_rp
AAR Calling Search Space	< None >
Location*	Hub_None
AAR Group	< None >
Network Locale	< None >
Use Trusted Relay Point*	Default
Geo Location	< None >
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
Calling Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS	

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain	< None >
MLPP Indication	Not available on this device
MLPP Preemption	Not available on this device

Port Information (Loop Start)

Port Direction*	Bothways
Attendant DN*	5015
<input checked="" type="checkbox"/> Unattended Port	

Product Specific Configuration Layout

?

Hookflash Timer (50-1550ms)*	50
Inter-digit Duration Timer (50-500 ms)*	100
Input Gain (-6..14 db)*	0
Output Attenuation (-6..14 db)*	3
Echo Cancellation Enable*	Enable
Echo Cancellation Coverage (ms)*	64
Ring Number*	Default
Impedance*	Default GW config



Configuration of SIP trunk to SME – G.711 (Page 1 of 3)

Navigation Path: Device → Trunk

Application Note

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go

ccmadministrator | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Trunk Configuration Related Links: Back To Find/List Go

Save Delete Reset Apply Config Add New

Status
 Status: Ready

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Device Name*	<input type="text" value="Sess_Mgr_SIP_trunk_G711"/>
Description	<input type="text" value="SIP Trunk to Session Manager G.711"/>
Device Pool*	<input type="text" value="G711_Pool"/>
Common Device Configuration	<input "="" type="text" value=" < None > "/>
Call Classification*	<input type="text" value="Use System Default"/>
Media Resource Group List	<input type="text" value="MRGL_Polaris"/>
Location*	<input type="text" value="Hub_None"/>
AAR Group	<input "="" type="text" value=" < None > "/>
Packet Capture Mode*	<input type="text" value="None"/>
Packet Capture Duration	<input type="text" value="0"/>
<input type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.	
Use Trusted Relay Point*	<input type="text" value="Default"/>



Configuration of SIP trunk to SME – G.711 (Page 2 of 3)

Navigation Path: Device → Trunk

Application Note

Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings

Default Prefix Settings

Number Type	Prefix	Strip Digits	Use Device Pool CSS	Calling Search Space
Unknown Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input checked="" type="checkbox"/>	<input type="text" value="< None >"/>

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain

Call Routing Information

☒ Remote-Party-Id

☒ Asserted-Identity

Asserted-Type*

SIP Privacy*

Inbound Calls

Significant Digits*

Connected Line ID Presentation*

Connected Name Presentation*

Calling Search Space

AAR Calling Search Space

Prefix DN

☐ Redirecting Diversion Header Delivery - Inbound



Configuration of SIP trunk to SME – G.711 (Page 3 of 3)

Navigation Path: Device → Trunk

Application Note

Outbound Calls	
Called Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Called Party Transformation CSS	
Calling Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS	
Calling Party Selection*	Originator
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Caller ID DN	
Caller Name	
<input type="checkbox"/> Redirecting Diversion Header Delivery - Outbound	

SIP Information	
Destination Address	172.20.109.252
Destination Address IPv6	
<input type="checkbox"/> Destination Address is an SRV	
Destination Port*	5060
MTP Preferred Originating Codec*	711ulaw
Presence Group*	Standard Presence group
SIP Trunk Security Profile*	SIP Trunks
Rerouting Calling Search Space	< None >
Out-Of-Dialog Refer Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	Standard SIP Profile
DTMF Signaling Method*	No Preference

Geo Location Configuration	
Geo Location	-- Not Selected --
Geo Location Filter	< None >
<input type="checkbox"/> Send GeoLocation Information	



Configuration of SIP trunk to SME – G.729 (Page 1 of 3)

Navigation Path: Device → Trunk

Application Note

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go

ccmadministrator | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Trunk Configuration Related Links: Back To Find/List Go

Save Delete Reset Apply Config Add New

Status
 Status: Ready

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Device Name*	<input type="text" value="Sess_Mgr_SIP_trunk_G729"/>
Description	<input type="text" value="SIP trunk to Session Manager"/>
Device Pool*	<input type="text" value="Default"/>
Common Device Configuration	<input type="text" value="< None >"/>
Call Classification*	<input type="text" value="Use System Default"/>
Media Resource Group List	<input type="text" value="MRGL_Polaris"/>
Location*	<input type="text" value="Hub_None"/>
AAR Group	<input type="text" value="< None >"/>
Packet Capture Mode*	<input type="text" value="None"/>
Packet Capture Duration	<input type="text" value="0"/>
<input type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.	
Use Trusted Relay Point*	<input type="text" value="Default"/>



Configuration of SIP trunk to SME – G.729 (Page 2 of 3)

Navigation Path: Device → Trunk

Application Note

Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings

Default Prefix Settings

Number Type	Prefix	Strip Digits	Use Device Pool CSS	Calling Search Space
Unknown Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input checked="" type="checkbox"/>	<input type="text" value="< None >"/>

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain

Call Routing Information

☒ Remote-Party-Id

☒ Asserted-Identity

Asserted-Type*

SIP Privacy*

Inbound Calls

Significant Digits*

Connected Line ID Presentation*

Connected Name Presentation*

Calling Search Space

AAR Calling Search Space

Prefix DN

☒ Redirecting Diversion Header Delivery - Inbound



Configuration of SIP trunk to SME – G.729 (Page 3 of 3)

Navigation Path: Device → Trunk

Application Note

Outbound Calls	
Called Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Called Party Transformation CSS	
Calling Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS	
Calling Party Selection*	Originator
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Caller ID DN	
Caller Name	
<input checked="" type="checkbox"/> Redirecting Diversion Header Delivery - Outbound	

SIP Information	
Destination Address	172.20.109.252
Destination Address IPv6	
<input type="checkbox"/> Destination Address is an SRV	
Destination Port*	5060
MTP Preferred Originating Codec*	711ulaw
Presence Group*	Standard Presence group
SIP Trunk Security Profile*	SIP Trunks
Rerouting Calling Search Space	tp_phones_rp
Out-Of-Dialog Refer Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	Standard SIP Profile
DTMF Signaling Method*	No Preference

Geo Location Configuration	
Geo Location	-- Not Selected --
Geo Location Filter	< None >
<input type="checkbox"/> Send GeoLocation Information	



Configuration of Route Group to SME

Navigation Path: Call Routing → Route/Hunt → Route Group

Application Note

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go

ccmadministrator | About | Logout

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Route Group Configuration Related Links: Back To Find/List Go

Save Delete Add New

Route Group Information

Route Group Name*

Distribution Algorithm* Top Down ▾

Route Group Member Information

Find Devices to Add to Route Group

Device Name contains Find

Available Devices**

172.20.212.253

172.20.8.26

AALN/S0/SU0/0@3825DSPfarm.pbxlab.org

CM-Mercury_SIP

CM-Neptune

Port(s) All ▾

Add to Route Group

Current Route Group Members

Selected Devices***

Sess_Mgr_SIP_trunk_G711 (All Ports)

Sess_Mgr_SIP_trunk_G729 (All Ports)

Reverse Order of Selected Devices

Removed Devices****

Route Group Members

[Sess_Mgr_SIP_trunk_G711](#)

[Sess_Mgr_SIP_trunk_G729](#)

Save Delete Add New



Configuration of Route List to SME

Navigation Path: Call Routing → Route/Hunt → Route List

Application Note

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go

ccmadministrator | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Route List Configuration Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

Status
 Status: Ready

Route List Information
Name*
Description
Cisco Unified Communications Manager Group* Default ▾
☒ Enable this Route List (change effective on Save; no reset required)

Route List Member Information
Selected Groups**

SME_Route_Group

▼ ▲

Add Route Group

Removed Groups***

▼ ▲

Route List Details
 [SME_Route_Group](#)

Save Delete Copy Reset Apply Config Add New



Configuration of Route Pattern to PSTN through SME

Navigation Path: Call Routing → Route/Hunt → Route Pattern

Application Note

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation | Cisco Unified CM Administration | Go

ccmadministrator | About | Logout

System ▾ | Call Routing ▾ | Media Resources ▾ | Voice Mail ▾ | Device ▾ | Application ▾ | User Management ▾ | Bulk Administration ▾ | Help ▾

Route Pattern Configuration | Related Links: [Back To Find/List](#) | Go

Save Delete Copy Add New

Status
 Status: Ready

Pattern Definition
Route Pattern*
Route Partition
Description
Numbering Plan
Route Filter
MLPP Precedence*
Resource Priority Namespace Network Domain
Gateway/Route List* [\(Edit\)](#)
Route Option
☒ Route this pattern
☐ Block this pattern
Call Classification*
☐ Allow Device Override ☒ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority
☐ Require Forced Authorization Code
Authorization Level*
☐ Require Client Matter Code

Calling Party Transformations
☐ Use Calling Party's External Phone Number Mask
Calling Party Transform Mask
Prefix Digits (Outgoing Calls)
Calling Line ID Presentation*
Calling Name Presentation*
Calling Party Number Type*
Calling Party Numbering Plan*

Connected Party Transformations
Connected Line ID Presentation*
Connected Name Presentation*

Called Party Transformations
Discard Digits
Called Party Transform Mask
Prefix Digits (Outgoing Calls)
Called Party Number Type*
Called Party Numbering Plan*


ISDN Network-Specific Facilities Information Element
Network Service Protocol
Carrier Identification Code
Network Service Service Parameter Name



Configuration of Route Patterns to Siemens PBX through SME

Navigation Path: Call Routing → Route/Hunt → Route Pattern

Application Note





**Cisco Unified CM Administration**
For Cisco Unified Communications Solutions


Navigation | Cisco Unified CM Administration | Go

ccmadministrator | About | Logout

System ▾ | Call Routing ▾ | Media Resources ▾ | Voice Mail ▾ | Device ▾ | Application ▾ | User Management ▾ | Bulk Administration ▾ | Help ▾

Route Pattern Configuration Related Links: [Back To Find/List](#) | Go

 Save  Delete  Copy  Add New

Status
 Status: Ready

Pattern Definition

Route Pattern*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence*

Resource Priority Namespace Network Domain

Gateway/Route List* [\(Edit\)](#)

Route Option
☒ Route this pattern
☐ Block this pattern

Call Classification*

☐ Allow Device Override ☐ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority

☐ Require Forced Authorization Code

Authorization Level*

☐ Require Client Matter Code

Calling Party Transformations

☐ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Calling Name Presentation*

Calling Party Number Type*

Calling Party Numbering Plan*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type*

Called Party Numbering Plan*

ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

Network Service	Service Parameter Name	Service Parameter Value
<input type="text" value="-- Not Selected --"/>	<input type="text" value="< Not Exist >"/>	<input type="text"/>



Configuring CUBE

CUBE#show version

Cisco IOS Software, 3800 Software (C3845-ADVENTERPRISEK9-M), Version 15.0(1)XA, RELEASE SOFTWARE (fc2)

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

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Compiled Thu 22-Oct-09 03:08 by prod_rel_team

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

IOSGW_SM uptime is 2 days, 23 hours, 19 minutes

System returned to ROM by power-on

System image file is "flash:c3845-adventerprisek9-mz.150-1.XA.bin"

Cisco 3845 (revision 1.0) with 484351K/39936K bytes of memory.

Processor board ID FHK1240F25Z

2 Gigabit Ethernet interfaces

1 Virtual Private Network (VPN) Module

2 Voice FXS interfaces

DRAM configuration is 64 bits wide with parity enabled.

479K bytes of NVRAM.

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

License Info:

License UDI:

```
-----  
Device# PID          SN  
-----  
*0      CISCO3845-MB    FOC12391507
```

Configuration register is 0x2102

CUBE#show run

Building configuration...

Current configuration : 5346 bytes

!

! Last configuration change at 19:28:46 UTC Thu Oct 22 2009

!

version 12.4

service timestamps debug datetime msec

service timestamps log datetime msec

no service password-encryption

!

hostname CUBE

```

!
boot-start-marker
boot system flash c3845-ipvoice_ivs-mz.124-24.6.13.PIA12
boot-end-marker
!
!card type command needed for slot 1
logging buffered 100000
enable secret 5 $1$v.Z3$YVkreNYDyhm388NGF3f1u0
!
no aaa new-model
network-clock-participate slot 1
!
ip source-route
ip cef
!
no ip domain lookup
no ipv6 cef
multilink bundle-name authenticated
!
voice-card 0
!
voice-card 1
dspfarm
dsp services dspfarm
!
voice service voip
address-hiding
allow-connections sip to sip
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw1
sip
bind control source-interface GigabitEthernet0/0
bind media source-interface GigabitEthernet0/0
header-passing error-passthru
asserted-id pai2
no update-callerid
midcall-signaling passthru
privacy-policy passthru3
g729 annexb-all
!
voice class codec 1
codec preference 1 g729r8
codec preference 2 g711ulaw
!
voice translation-rule 14

```

¹ This command enables router to up speed to t38. To pass-through G711, the command has to be changed to “fax protocol pass-through g711ulaw”

² This command enables router to send P-Asserted ID within the SIP Message Header. Alternatively, this command can also be applied to individual dial-peers (voice-class sip asserted-id pai)

³ This command enables router to transparently pass through all received Privacy values. Alternatively, this command can also be applied to individual dial-peers (voice-class sip privacy-policy passthru)

```

!
voice translation-profile outbound_g7115
translate called 1
!
license udi pid CISCO3845-MB sn FOC12391507
archive
log config
hidekeys
!
interface GigabitEthernet0/0
ip address 172.20.109.203 255.255.255.0
duplex auto
speed auto
media-type rj45
!
interface GigabitEthernet0/1
no ip address
duplex auto
speed auto
media-type rj45
!
ip forward-protocol nd
!
no ip http server
ip route 0.0.0.0 0.0.0.0 172.20.109.1
!
control-plane
!
voice-port 0/0/0
timeouts ringing infinity
station-id number 14081238004
caller-id enable
!
voice-port 0/0/1
!
mgcp fax t38 ecm
mgcp behavior g729-variants static-pt
!
sccp local GigabitEthernet0/0
!
dial-peer voice 1408 voip
description Towards PSTN
destination-pattern 14081238...
session protocol sipv2
session target ipv4:172.20.8.46
incoming called-number 1408T
voice-class codec 1
voice-class sip early-offer forced
dtmf-relay rtp-nte

```

⁴ This translation rule is used on dial-peer 9001 (outgoing G.711-only calls) to strip the prefix “8” (sent by CUCM to match a different dial-peer whenever G.711-only calls are placed) from the telephone number

⁵ This translation profile, containing the previously-defined translation rule, is assigned to the dial-peer used to place outbound G.711-only calls (dial-peer 9001 in this configuration example)



```
fax-relay ecm disable
fax-relay sg3-to-g3
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw1
!
dial-peer voice 1510 voip
description Towards SME
destination-pattern 15101234...
session protocol sipv2
session target ipv4:172.20.109.252
incoming called-number 1510T
dtmf-relay rtp-nte
fax-relay ecm disable
fax-relay sg3-to-g3
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw1
!
dial-peer voice 9001 voip6
translation-profile outgoing outbound_g711
destination-pattern 81.....
codec g711ulaw
voice-class sip early-offer forced
session protocol sipv2
session target sip-server
dtmf-relay rtp-nte
fax-relay ecm disable
fax-relay sg3-to-g3
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw1
!
gateway
timer receive-rtp 600
!
sip-ua
!
gatekeeper
shutdown
!
telephony-service
sdspfarm units 1
sdspfarm transcode sessions 4
sdspfarm tag 1 mtp001122334455
max-ephones 10
max-dn 10
ip source-address 172.20.109.203 port 2000
max-conferences 12 gain -6
transfer-system full-consult
transfer-pattern ....
create cnf-files version-stamp Jan 01 2002 00:00:00
!
line con 0
line aux 0
line vty 0 4
```

⁶ This dial-peer (optional) is used for outbound G.711-only calls. Typically used for fax/modem transmissions. It matches a Route Pattern configured in CUCM/SME, which sends a prefix (in this example “8”) along with the telephone number string



```
password cisco
```

```
login
```

```
!
```

```
exception data-corruption buffer truncate
```

```
scheduler allocate 20000 1000
```

```
end
```

Application Note

Acronyms

Acronym	Definitions
ANF-PR	Additional Network Feature Path Replacement
AOC	Advice-of-charge. Information element is sent with the connection setup information for incoming Euro-ISDN connections. The AOC IE is used for call charge calculation.
CUCM	Cisco Unified Communications Manager
CCBS	Call Completion to Busy Subscriber
CCNR	Call Completion on No Reply
CFB	Call Forwarding on Busy
CFNR	Call Forwarding No Reply
CFU	Call Forwarding Unconditional
CLIP	Calling Line (Number) Identification Presentation
CLIR	Calling Line (Number) Identification Restriction
CNIP	Calling Name Identification Presentation
CNIR	Calling Name Identification Restriction
COLP	Connected Line (Number) Identification Presentation
COLR	Connected Line (Number) Identification Restriction
CONP	Connected Name Identification Presentation
CONR	Connected Name Identification Restriction
CT	Call Transfer
MWI	Message Waiting Indicator
PSTN	Public Switched Telephone Network



Important Information

Application Note

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