



Siemens HiPath 4000 Rel 5.0 using SIP via Cisco Unified Communications Manager – Session Manager Edition 8.5 to Cisco Unified Communications Manager 8.5 and Cisco Unified Border Element (ASR1000 Rel. 3.3.1S) to Service Provider

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Introduction

- This application note describes the necessary steps and configurations for connectivity between Siemens Hipath 4000 Release 5.0 and Cisco Unified Communications Manager (Cisco UCM) Release 8.5 via Cisco Unified Communications Manager-Session Manager Edition (Cisco UCM-SME) Release 8.5. The Cisco UCM-SME also manages a SIP connection to a Cisco Unified Border Element (Cisco UBE) ASR1000 running software version 3.3.1S. The network topology diagram (Figure 1) shows the test setup for end-to-end interoperability between the different leaf nodes (Siemens Hipath 4000, Cisco UCM and Cisco UBE/SP) via the Cisco UCM-SME.
- The network topology diagram (Figures 1) shows the test setup for end-to-end interoperability between Cisco Unified Communications Manager (Cisco UCM) 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 Service Provider, 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 ASR1002 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.
- [Cisco 3900 Series Integrated Services Routers](#)
- [Cisco 2900 Series Integrated Services Routers](#)
- [Cisco 2800 Series Integrated Services Routers](#)
- [Cisco 3800 Series Integrated Services Routers](#)
- [Cisco AS5350XM Universal Gateway](#)
- [Cisco AS5400XM Universal Gateway](#)
- [Cisco ASR 1000 Series Aggregation Services Routers](#)
- If additional guidance on the Cisco UBE is needed, please refer to the Cisco UBE section on the Cisco Interoperability Portal (www.cisco.com/go/interoperability).
- Results may vary based on Service Provider being used.

Network Topology

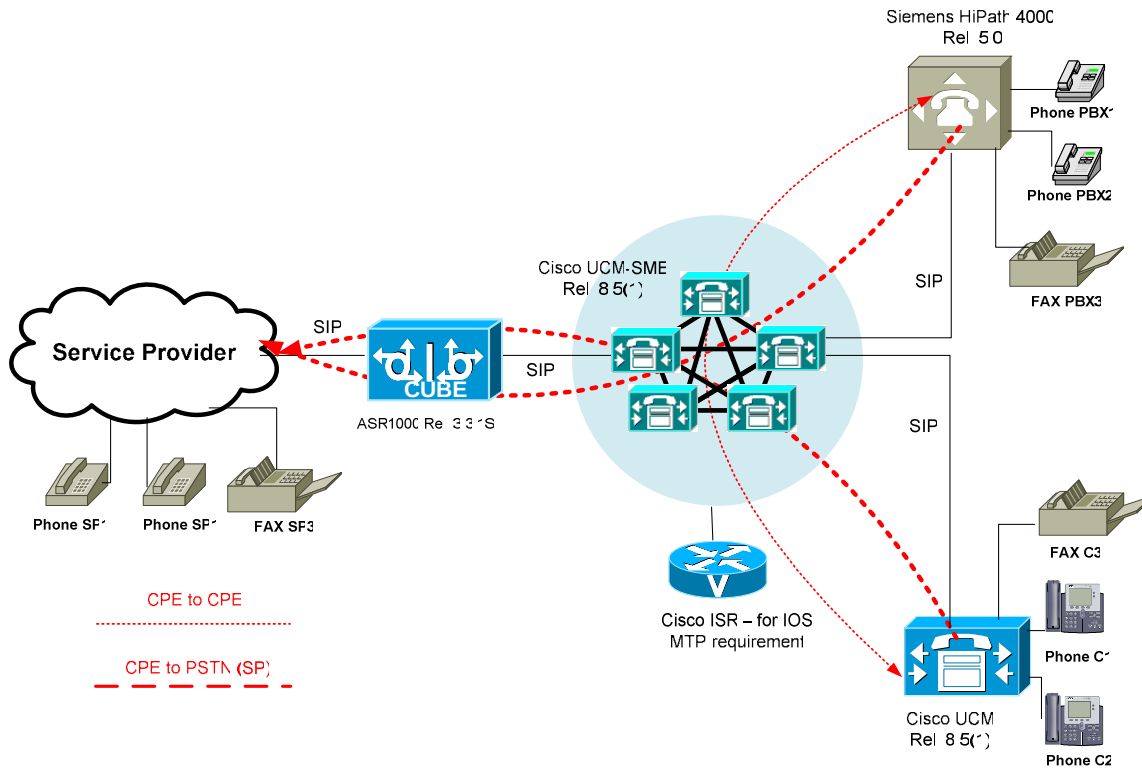


Figure 1. Basic Call Setup



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.

Limitations

PBX

- Siemens PBX does not update phones' displays during call alerting. Once call is completed, however, phone displays are updated with connected party information.
- Siemens PBX does not update phones displays after calls to Cisco UCM and/or Service provider are transferred/conferenced. Cisco UCM/Cisco UBE send UPDATE message with updated connected party information, but it is not passed to the Siemens end point.
- Siemens PBX does not provide Diversion Header and/or History-Info header on forwarded calls over SIP trunk. This limitation prevents Cisco UCM end points from providing call forward information when used as call forwarding targets. This also prevents centralized voicemail deployments using Cisco voicemail platforms integrated to Cisco UCM-SME and/or Cisco UCM. Inbound SIP NOTIFY messages are also rejected with 404 Not Found.
- Siemens SIP phones do not support Early Attended call transfers.
- Although T.38 fax relay is supported with this software release, whenever calls are established using G.729 and fax tone is detected by the Siemens PBX, it first tries to re-establish the call using G.711. If the remote end rejects the re-negotiation to the higher bandwidth codec, the Siemens PBX will not attempt T.38. If call is first established using G.711, however, then the Siemens PBX will attempt switching to T.38 fax relay.

Cisco UBE

- ASR 1000 software currently does not support codec selection preference (Voice Class Codec command) on Early Offer-to-Early Offer call scenarios. This can potentially cause issues whenever calls require switching codecs mid-call (e.g. calls to fax machines that trigger G.711 pass-through). Because of this, it is recommended creating a different dial-peer to be used for G.711-only calls. An example of a dial-peer to be used for fax/modem calls is provided later in this document.



System Components

Hardware Components

The following hardware is required:

- Cisco MCS 7800 Unified Communications Manager Appliance
- Cisco ASR 1000 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 8.5
- Cisco IOS Release ASR1000 RP1 3.3.1S
- Siemens HiPath 4000 PBX Release 5.0

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
- Consultation transfer – Local and Network/External
- Early Attended transfer – Local and Network/External
- 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 and T.38 fax relay (See Limitations section for details)

Features Not Supported (See limitations)

- Centralized Message center voicemail integration
- Message Waiting via SIP NOTIFY



Configuration

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

Configuring the Siemens HiPath 4000 PBX

1. Software release information
2. Add the new access code to Dialing Plans using WABE + LDPLN
3. Add the new trunk group access code using BUEND
4. Configure trunk using TDCSU
5. Configure Class of Trunk using COT
6. Configure Class of Parameter for device handler using COP
7. Configure Class of Service using COSSU
8. Configure Gateway using GKREG
9. Configure HG3500 board using CGWB
10. Configure Trunk Routing using RICHT
11. Configure LCR Out-dial Rules using LODR
12. Configure LCR Directions using LDAT
13. Configure stations using SBCSU
14. Enable In-Band DTMF signaling for the Digital Stations using SBCSU.
15. Web-Based Management for HG3500 V5 (Screen Captures)
 - o Basic Settings – Gateway Properties
 - o Voice Gateway – SIP Parameters
 - o Voice Gateway – Codec Parameters
 - o Voice Gateway – SIP Trunk Profile Parameters
 - o Voice Gateway – SIP Trunk Profile



Configuration Menus and Commands for Hipath 4000

Siemens HiPath 4000 Software Release

DISP-DBC:N;

H500: AMO DBC STARTED

```
+-----+
| SYSTEM CLASSIFICATION : SYSTEM 600          (H600  )
| HARDWARE ASSEMBLY    : COMPACT PCI          (CPCI  )
| OPERATING MODE       : SIMPLEX
| RESTART TYPE         : SYM
| HW-ARCHITECTURE      : 4000
| HW-ARCHITECTURE TYPE : 3
|
| 'NO OF' HW VALUES
|   LTG'S : 1   LTU'S : 15   LOG.LINES : 32000   MTS BD /GSN: 1
|   SIUP'S/LTU: 4   TMD24'S PER LTU: 4   PHYS.PORTS: 16000   HWY /MTS BD: 128
|   HDLC /DCL : 16   PBC /DCL : 6   PBC'S : 17
| LOG. SIU LINES : 81
| LOG. CONF LINES : 90
| LOG. DCL LINES : 91
| DB DIMENSIONING-NAME : LARGE          CONF-TABLE VERSION: 26
| DB SUSY'S:
|   SWITCH NUMBER : L31910Z0291U00001
| LOCATION : CUSTOMER
| BAPPL : BSMONO
| DBAPPL : DBLARGE
| 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
+-----+
```

AMO-DBC -111 DATABASE CONFIGURATION
DISPLAY COMPLETED;

DISP-VEGAS:LIST=LONG;

H500: AMO VEGAS STARTED

SYSTEM NO.	AMO	APS NO.	START	USER	STATUS
SWU: L31910Z0291U00001	REGEN	P30252B4700B00101	10.02.11	13:26 MIKO	FINISHED
AMO FINISHED : 10.02.2011 13:27:30					
SWU RES CODE APS: P30252B4700S00101 (DIR FILE: :PDS:APSI/PS/S0-E20SC)					
SWU AMO CODE APS: P30252B4700B00101 (DIR FILE: :PDS:APSI/PS/B0-E20BC)					
SWU AMO TEXT APS: P30252B4700B00101 (DIR FILE: :PDS:APSI/PS/B0-E20BC)					
BREAK MARK : NO					
RESERVATION : NO					
ADS: NO-ENTRY-IN-DBASE	REGEN	P30252B4700A00101	10.02.11	13:27 MIKO	FINISHED
AMO FINISHED : 10.02.2011 13:27:50					
ADS RES CODE APS: P30252B4700D00101 (DIR FILE: :PDS:APSI/PS/D0-E20DC)					
ADS AMO CODE APS: P30252B4700A00101 (DIR FILE: :PDS:APSI/PS/A0-E20AC)					
ADS AMO TEXT APS: P30252B4700A00101 (DIR FILE: :PDS:APSI/PS/A0-E20AC)					
BREAK MARK : NO					
RESERVATION : NO					

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

DISP-APS:Y,PSALL,"Y0-E20YC",;

H500: AMO APS STARTED

ADINIT STARTED
PROGRAM SYSTEM : Y0-E20YC
VERSION NUMBER : 50
CORRECTION VERSION NUMBER : 001
PART NUMBER : P30252N4701BH2502|V5 R1.2.25
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;

Dial Plans configuration, WABE

DISP-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
					ANALYSIS RESULT
					RESERVED/CONVERT DNI/ADD-INFO
					*=OWN NODE
101		*			NETRTE
102		*			NETRTE
103		*			NETRTE
150		.	*****	*****	TIE
199901 - 199902		.	*****	*****	TIE
200		.	*****	*****	TIE
201 - 202		.	*****	*****	TIE
26 - 27		.	*****	*****	TIE
4300 - 4310		.	*****	*****	STN
4311 - 4320		.	*****	*****	STN
					DESTNO 103
					DNNO 0- 0-103
					PDNNO 0- 0-203
					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
					ANALYSIS RESULT
					RESERVED/CONVERT DNI/ADD-INFO
					*=OWN NODE
5000 - 5015		.	*****	*****	STN
5297		.	*****	*****	STN
6999		.	*****	*****	TIE
7020 - 7029		.	*****	*****	STN
7030 - 7039		.	*****	*****	STN
7040 - 7049		.	*****	*****	STN
					DESTNO 103
					DNNO 0- 0-103
					PDNNO 0- 0-203
					DESTNO 0
					DNNO 0- 0-200*
					DESTNO 0
					DNNO 0- 0-200*
					R
					DESTNO 0
					DNNO 0- 0-200*
					DESTNO 0
					DNNO 0- 0-200*

DIGIT INTERPRETATION		VALID FOR ALL DIAL PLANS			
CODE		CALL PROGRESS STATE			
		1	11111	11112	22
		0	12345	67890	12345 67890 12
					ANALYSIS RESULT
					RESERVED/CONVERT DNI/ADD-INFO
					*=OWN NODE
7050 - 7059		.	*****	*****	STN
7060 - 7069		.	*****	*****	STN
					R
					DESTNO 0
					DNNO 0- 0-200*



7070	. ***** **...	STN	DESTNO 0 DNNO 0- 0-200* R DESTNO 0 DNNO 0- 0-200*
9	. ***** **...	TIE	
*0	. *...*	ACDWORK	
*2	. *...*	CDRACC	
*3	. ***** **...	PUDIR	
*4	. *...*	CONF	
*52	. *...*	MWCAN	
*530	. *...*	MBOFF	
*532	. *...*	MBON	

DIGIT INTERPRETATION VALID FOR ALL DIAL PLANS

CODE	CALL PROGRESS STATE						NODE/DIGIT	RESERVED/CONVERT
	1	11111	11112	22	ANALYSIS	DNI/ADD-INFO		
	0	12345	67890	12345	67890	12	RESULT	*=OWN NODE
*564	. *...*						ACDLOGOF	
*565	. *...*						ACDLOGON	
*570	. *...*						ACDPQS	
*571	. *...*						ACDPGS	
*580	. *...*						ACDSQS	
*581	. *...*						ACDSGS	
*590	. *...*						ACDEMMSG	
*591	. *...*						ACDSHMSG	
*61	. ***..						DCPA	
*62	. ***..						SELFPA	
*67	. *****						DISUON	
*68	. *****						DISUOFF	
*7	. *...*						CONSKY	
*80 - *89	. *****						PARK	
*9	. *...*						HOLD	
**1	. *...*						SPLIT	
**3	. *...*						PU	

DIGIT INTERPRETATION VALID FOR ALL DIAL PLANS

CODE	CALL PROGRESS STATE						NODE/DIGIT	RESERVED/CONVERT
	1	11111	11112	22	ANALYSIS	DNI/ADD-INFO		
	0	12345	67890	12345	67890	12	RESULT	*=OWN NODE
**41	. *...*						RMCONFP1	
**42	. *...*						RMCONFP2	
**43	. *...*						RMCONFP3	
**44	. *...*						RMCONFP4	
**45	. *...*						RMCONFP5	
**46	. *...*						RMCONFP6	
**47	. *...*						RMCONFP7	
**48	. *...*						RMCONFP8	
**50	. *...*						CAFGRV	
**51	. *...*						CAFGRNAV	
**6	. *...*						COMSPK	
**8	. *...*						MWANS	
200	. ***						OWNNODE	
**4	. *...*						RMCONFLP	
**5	. *...*						MONSLNT	
**#65	. *...*						CAFGR0FF	
**#274	. *...*						WSS	

DIGIT INTERPRETATION VALID FOR ALL DIAL PLANS

CODE	CALL PROGRESS STATE						NODE/DIGIT	RESERVED/CONVERT
	1	11111	11112	22	ANALYSIS	DNI/ADD-INFO		
	0	12345	67890	12345	67890	12	RESULT	*=OWN NODE
**#50	. *...*						CAFAV	
**#51	. *...*						CAFNAV	
**#55	. *...*						CAFAFWD	



*#56	. * . . . *			CAFD FWD	
*#58				DPIN	
*#590				DCOSX	
*#591				ACOSX	
*#65	. * . . . *			CAFLOGOF	
*#736				SIGNON	
*#739				SIGNOFF	
*#99				SAT	R
#0	. * . . . *			ACDNAV	
#1	. * . . . *			ACBK	
#2	. * . . . *			APRIV	
#3	. * . . . *			SPDI	
#4	. * . . . *			SNR	
#5				ADND	

DIGIT INTERPRETATION VALID FOR ALL DIAL PLANS

CODE	CALL PROGRESS STATE				NODE/DIGIT ANALYSIS	RESERVED/CONVERT DNI/ADD-INFO
	0	1	2	3		
	12345	67890	12345	67890	RESULT	*=OWN NODE
#61					SPDC1	
#62					SPDC2	
#80	. * . . . *				BROADCST	
#81	. * . . . *				SPKRCALL	
#8378					HWTEST	
#90					ASYSFWD	
#91					AFWDEXIN	
#92					AFWDEXT	
#93					AFWDINT	
#94					AFWDB	
#95					AFWDBNA	
#99					AFWDREM	
						CFREMVAR CFU
						CFREMSE VOICE
*#1					MWACT	
*#2					BUZZ	
*#329					FAX	

DIGIT INTERPRETATION VALID FOR ALL DIAL PLANS

CODE	CALL PROGRESS STATE				NODE/DIGIT ANALYSIS	RESERVED/CONVERT DNI/ADD-INFO
	0	1	2	3		
	12345	67890	12345	67890	RESULT	*=OWN NODE
*#75					DIGIDAT	
*#77					DTE	
*#8					MWCANORI	
*#92					AHTVCE	
*#93					DHTVCE	
*#94					AHTDTE	
*#95					DHTDTE	
*#96					AHTFAX	
*#97					DHTFAX	
##0					ACDAV	
##1					DCBK	
##2					DPRIV	
##3					SPDIPROG	
##4					LNR	
##5					DDND	
##7					KNOVR	
##90					DSYSFWD	

DIGIT INTERPRETATION VALID FOR ALL DIAL PLANS

CODE	CALL PROGRESS STATE				NODE/DIGIT ANALYSIS	RESERVED/CONVERT DNI/ADD-INFO
	0	1	2	3		
	12345	67890	12345	67890	RESULT	*=OWN NODE
##91					DFWDVCE	
##92					DFWDEXT	



##93	 *	 *		DFWDINT	
##99	. ****	. ****	. ** . *	DFWDREM	
###1	. *	 *		TRACE	CFREMVAR CFU
###20		. ** . *	. ** . .	MILLWAT	CFREMSE VOICE
###21		. ** . *	. ** . .	LOOPBACK	
###22		. ** . *	. ** . .	SILENCE	
###23		. ** . *	. ** . .	COMBO	
###6	 *			MONTONE	

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

LCR Dial Plan configuration, LDPLN

DISP-LDPLN: LDP,,, "91"- "XXX"- "XXX"- "XXXX",,,,,;
 H500: AMO LDPLN STARTED

DIPLNUM:	0				
LDPNO :	5	LDP :	91-XXX-XXX-XXXX		
		SPC :	22		
		FDSFIELD :	0	SDSFIELD :	0
				PINDP :	N
DPLN	LROUTE	LAUTH			
0	201	1			

AMO-LDPLN-111 ADMINISTRATION LCR DIALPLAN
 DISPLAY COMPLETED

Note: The LCR Dialplan entry above is used for outbound 10-digit calls to the PSTN, using G.729 codec.

DISP-LDPLN: LDP,,, "81"- "XXX"- "XXX"- "XXXX",,,,,;
 H500: AMO LDPLN STARTED

DIPLNUM:	0				
LDPNO :	5	LDP :	81-XXX-XXX-XXXX		
		SPC :	22		
		FDSFIELD :	0	SDSFIELD :	0
				PINDP :	N
DPLN	LROUTE	LAUTH			
0	201	1			

AMO-LDPLN-111 ADMINISTRATION LCR DIALPLAN
 DISPLAY COMPLETED

Note: The LCR Dialplan entry above is used for outbound calls requiring G.711 codec (e.g. fax and/or modem calls). This dialplan string allows the Siemens PBX to outpulse 81+10 digits, which will casue it to match a dial-peer configured to support G.711-only calls. The dial-peer in question will use a translation rule to strip out the leading “8” and send the 1+10 digit number to the Service Provider for processing.

Trunk Groups, BUEND

DISP-BUEND;
 H500: AMO BUEND STARTED
 TRUNK GROUPS (FORMAT=S)

NO.	NAME	CHARCON
1	T1 TGRP 1	(NEUTRAL)
2	T1 TGRP 2	(NEUTRAL)
3	E1 PRI TG 3	(NEUTRAL)
4	E1 PRI TG 4	(NEUTRAL)
5	E1 PRI TG 5	(NEUTRAL)
6	E1 PRI TG 6	(NEUTRAL)
150	150-SIP	(NEUTRAL)



151 151-SIP Q (NEUTRAL)

AMO-BUEND-111 TRUNK GROUP
DISPLAY COMPLETED;

Trunk configuration, TDCSU

DIS-TDCSU:1-1-1-0;

H500: AMO TDCSU STARTED

```
+-----+-----+-----+-----+
| DEV      = HG3550IP      PEN      = 1-01-001-0      TGRP      = 150      |
+-----+-----+-----+-----+
| PROTVAR  = PSS1V2        INS       = Y              SRCHMODE  = DSC      |
| COTNO    = 59            COPNO    = 59              DPLN       = 0        |
| ITR      = 0            COS       = 59              LCOSV      = 1        |
| LCOSD    = 1            CCT       = SIP TRUNK        DESTNO     = 111      |
| SEGMENT  = 8            DEDSCC    =                 DEDSVC     = NONE     |
| FACILITY  =              DITIDX    =                 SRTIDX     =          |
| TRTBLL   = GDTR         SIDANI    = N              ATNTYP     = TIE      |
| CBMATTR  = NONE        NWMUXTIM  = 10             TCHARG      = N        |
| SUPPRESS = 0            DGTPR     =                 CHIMAP     = N        |
| ISDNIP    =              ISDNNP   =                  PNPAC      =          |
| PNPL2P    =              PNPL1P   =                  NNO        = 111     |
| TRACOUNT  = 31          SATCOUNT = MANY            CARRIER   = 1        |
| ALARMNO   = 0           FIDX      = 1              FWDX       = 10       |
| ZONE      = EMPTY       COTX      = 59             TPROFNO    =          |
| DOMTYPE   =              DOMAINNO =                 CCHDL     = SIDEA    |
| INIGHT    =              UUSCCY   = 8              FNIDX      = 1        |
| UUSCCX    = 16          & G711    & G729A0PT        SRCGRP     = 1        |
| CLASSMRK  = EC          TCCID     =                 SECLEVEL   = TRADITIO  |
+-----+-----+-----+-----+
| BCNEG    = N            BCGR      = 1              LWPAR      = 10       |
| LWPP      = 0           LWLT      = 0              LWPS       = 0        |
| LWR1      = 0           LWR2      = 0              VVNO        =          |
| DMCALLWD  = Y           VNNO      =                 TSCS       =          |
| SVCDOM    =              TSCS     =                 TSCS       =          |
| BCHAN     = 1 && 10      TSCS     =                 TSCS       =          |
+-----+-----+-----+-----+
```

AMOUNT OF B-CHANNELS IN THIS DISPLAY-OUTPUT: 10

AMO-TDCSU-111 DIGITAL TRUNKS
DISPLAY COMPLETED;

Class of Trunk, COT

DISP-COT:COTNO=59;

H500: AMO COT STARTED

COT: 59 INFO: SIP TRUNK
DEVICE: INDEP SOURCE: DB
PARAMETER:

PRIORITY FOR AC WILL BE DETERMINED FROM MESSAGE	PRI
RECALL IF USER HANGS UP IN CONSULTATION CALL	RCL
TRUNK CALL TRANSFER	XFER
TRUNK SIGNALING ANSWER	ANS
KNOCKING OVERRIDE POSSIBLE	KNOR
CALL EXTEND FOR BUSY, RING OR CALL STATE	CEBC
NETWORKWIDE AUTOMATIC CALLBACK ON BUSY	CBBN
NETWORKWIDE AUTOMATIC CALLBACK ON FREE	CBFN
REGISTRATION OF IMPLAUSIBLE EVENTS	IEVT
DON'T RELEASE CALL TO BUSY HUNT GROUP	BSHT
END-OF-DIAL FOR BLOCK IS SET	BLOC
EMERGENCY OVERRIDE/DISCONNECT VIA S0/S2 LINE	PROV
SEND NO NODE NUMBER TO PARTNER	LWNC
ACTIVATE TRANSIT COUNTER ADMINISTRATION FOR S0/S2 LINE	ATRS
CONNECTION TO ROUTE OPTIMIZATION NODE	ROPT
TSC-SIGNALING FOR NETWORKWIDE FEATURES (MANDATORY)	TSCS



TRUNK SENDS CALL CHARGES TO ORIGINATING NODE NUMBER	TRSC
USE DEFAULT NODE NUMBER OF LINE	DFNN
CALL FORWARDING PROGRAMING FOR OTHER SUBSCRIBERS	CFOS
PIN NETWORKWIDE POSSIBLE	PINR
AOC PER CALL (AUTOMATICAL OR ON REQUEST), MAND. CORNET-NQ	AOCC
AUTOM.DTMF CONVERSION ON INCOM.CALL WHILE IN TALK STATE	AMFC
SEND DIGITS VIA IN.BAND DTMF BEFORE ANSWER	IBBA
NO TONE	NTON

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

Class of Parameter for Device Handler, COP

DISP-COP:COPNO=59;
H500: AMO COP STARTED

COP: 59	INFO: SIP TRUNK	
DEVICE: INDEP	SOURCE: DB	
PARAMETER:		
REGISTRATION OF LAYER 3 ADVISORIES		L3AR
REFLECT RESTART INDICATOR AND B-CHANNEL BY RESTART		RRST

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

Trunk Class of Service, COSSU

DISP-COSSU:TYPE=COS,COS=59;
H500: AMO COSSU STARTED

COS	VOICE	FAX	DTE
59	>SIP TRUNK TA TNOTCR COSXCD MB VCE FWDNWK TTT FWDECA	NOCO NOTIE	NOCO NOTIE

AMO-COSSU-111 CLASSES OF SERVICE
DISPLAY COMPLETED;

DISP-COSSU:TYPE=LCOSV,LCOSV=15;
H500: AMO COSSU STARTED

LCOS	1	2	3	4	5	6	COPIN
V	12345678901234567890123456789012345678901234						
	>SERVICE INFORMATION						
15	XXXXXXXXXXXXXXXXXXXX						0
	>15-STATE-WIDE						

AMO-COSSU-111 CLASSES OF SERVICE
DISPLAY COMPLETED;

DISP-COSSU:TYPE=LCOSD,LCOSD=15;
H500: AMO COSSU STARTED

LCOS	1	2	3	4	5	6	COPIN
D	12345678901234567890123456789012345678901234						



```
|>SERVICE INFORMATION|
+-----+-----+-----+-----+-----+-----+
| 15|XXXXXXXXXXXXXXXXX.....|.|
|>15-USA|
+-----+-----+-----+-----+-----+-----+
```

AMO-COSSU-111 CLASSES OF SERVICE
DISPLAY COMPLETED;

Gateway configuration, GKREG

DISP-GKREG:1;

H500: AMO GKREG STARTED

```
+-----+
| GWNO      1          GWATTR  INTGW  REGGW  HG3550V2 SIP
| GWIPADDR  172.20 .188.13      GWDIRNO  199901
| DIPLNUM   0          DPLN 0
| LAUTH     1
| GATEWAY REGISTERED: YES
| IP GATEWAY IS CONFIGURED BY GKREG
| INFO:     1-SIP
| SECLEVEL: TRADITIO
+-----+
```

AMO-GKREG-111 GATEKEEPER REGISTRY
DISPLAY COMPLETED;

HG3500 Board configuration, CGWB

DISP-CGWB:1,1,;

H500: AMO CGWB STARTED

```
+-----+
| CGW BOARD DATA
+-----+
| SIP      HG3550
+-----+
| LTU = 1      SLOT = 1      SMODE = NORMAL      POOLNO: 0
+-----+
```

GLOBAL DATA AND ETHERNET INTERFACE DATA - CONFIGURABLE VALUES:

```
-----
IPADR   = 172.20 .188.13      TCPD   = (4060)
NETMASK = 255.255.255.0      VLAN   = NO (NO)
DEFRT   = 172.20 .188.1      (0.0.0.0 = NOT CONFIGURED)
BITRATE = 100MBFD (AUTONEG)  VLANID = 0 (0)
PATTERN = 255 (213)
TRPR SIP = 20 (0)
TRPR SIPQ = 0 (0)
TRPRH323 = 0 (0)
TPRH323A = 0 (0)
DNSIPADR = 0.0.0.0      TLSP   = 4061 (4061)
```

GLOBAL DATA - CONSTANT VALUES:

```
-----
AMO INTERFACE VERSION: 0      OPMODE: 1      DATA_VALID: YES
```

SERVICE INTERFACE

```
-----
LOGINTRM = "HICOM" (TRM)
PASSW    = *****
```

ASC DATA - CONFIGURABLE VALUES:

```
-----
TOSPL   = 184 (184)      TOSSIGNL = 104 (104)
UDPPRTLO = 29100 (29100) UDPPRTHI = 30100 (30099)
T38FAX   = NO (YES)      REDRFCTN = YES (YES)
RFCFMOIP = YES (YES)      RFCDTMF  = YES (YES)
```

PRI01 : CODEC = G729A VAD = NO RTP-SIZE = 20



```
PRI02 : CODEC = G711U   VAD = NO    RTP-SIZE = 20
PRI03 : CODEC = G711A   VAD = NO    RTP-SIZE = 20
PRI04 : CODEC = NONE    VAD = NO    RTP-SIZE = 20
PRI05 : CODEC = NONE    VAD = NO    RTP-SIZE = 20
PRI06 : CODEC = NONE    VAD = NO    RTP-SIZE = 20
PRI07 : CODEC = NONE    VAD = YES   RTP-SIZE = 60

DSP CONFIGURATION DATA
-----
JITBUFD  = 60   (60)

PRIMARY AND SECONDARY GATEKEEPER
-----
PRIGKIP   = (0.0.0.0)
PRIGKPN   = 1719 (1719)
PRIGKID1  = PRIMARYRASMANNERID
            (PRIMARYRASMANNERID)
PRIGKID2  =
SECGKIP   = (0.0.0.0)
SECGKPN   = 1719 (1719)
SECGKID1  = SECONDARYRASMANNERID
            (SECONDARYRASMANNERID)
SECGKID2  =
TIMTOLIVE = 120(120)

MANAGEMENT STATION AND BACK-UP SERVER
-----
MGNTIP    = 172.20.188.15 (0.0.0.0)
MGNTPN    = 8000 (8000)
BUSIP     = 172.20.188.15 (0.0.0.0)
BUSPN     = 443 (443)

DMC DATA
-----
DMCCONN   = 0 (0)

WBM LOGIN DATA
-----
LOGINWBM  = HP4K-DEVEL   ROLE = ENGR (ADMIN)
LOGINWBM  = HP4K-SU      ROLE = SU (ADMIN)
LOGINWBM  = HP4K-ADMIN   ROLE = ADMIN (ADMIN)
LOGINWBM  = HP4K-READER  ROLE = READONLY (ADMIN)

GATEWAY DATA
-----
GWID1     = PRIMARYRASMANNERID
GWID2     =

H.235 SECURITY DATA
-----
GLOBID1   = siemensGateway2003
            (siemensGateway2003)
GLOBID2   =
TIMEWIN   = 10   (100)
SECSUBS   = NO   (NO)
SECTRNK   = NO   (NO)
GLOBPW    =
242-191-30-119-188-83-173-161-43-0-70-36-218-74-169-221-78-102-174-170

LEGK DATA
-----
GWNO      = 1 (0)
GWDIRNO   = 199901
REGEXTGK  = NO (NO)

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




SIP TRUNKING DATA FOR SSA

SIPREG = NO (NO)
REGIP1 = 0.0.0.0 (0.0.0.0)
PORTTCP1 = 5060 (5060)
PORTTLS1 = 5061 (5061)
REGIP2 = 0.0.0.0 (0.0.0.0)
PORTTCP2 = 5060 (5060)
PORTTLS2 = 5061 (5061)
REGTIME = 120 (120)

DLS DATA

DLSIPADR =
DLSPORT = 4061
DLSACPAS = NO

JB DATA - CONFIGURABLE VALUES:

JBMODE = 2
AVGDLYV = 40 (40)
MAXDLYV = 120 (120) MINDLYV = 20 (20)
PACKLOSS = 4 (4)
AVGDLYD = 60 (60) MAXDLYD = 200 (200)

AMO-CGWB -111 CONFIGURATION OF HG3500 BOARD
DISPLAY COMPLETED;

Trunk Routing configuration, RICHT

DISP-RICHT:CD,103;

H500: AMO RICHT STARTED

ROUTES FOR ALL DPLN										SVC = VCE
CODE	NAME, CQMAX, DESTNO AND CPS	TGRP CCNO	P L	DTMF			LRTE	CPAR	F	
	1 111112		B	CNV	DSP	TEXT	PULS			W
	12345 67890 123452						PAUSE			D
										B
103		150		F	W		PP300	103		
NEUTRAL	SIP TRUNK									
	DNNO: 103									
	PDNNO: 203									
	DESTNO :103									
	REROUT :YES									
ROUTES FOR ALL DPLN										SVC = FAX
CODE	NAME, CQMAX, DESTNO AND CPS	TGRP CCNO	P L	DTMF			LRTE	CPAR	F	
	1 111112		B	CNV	DSP	TEXT	PULS			W
	12345 67890 123452						PAUSE			D
										B
103		150						103		
NEUTRAL	SIP TRUNK									
	DNNO: 103									
	PDNNO: 203									
	DESTNO :103									
	REROUT :YES									
ROUTES FOR ALL DPLN										SVC = DTE
CODE	NAME, CQMAX, DESTNO AND CPS	TGRP CCNO	P L	DTMF			LRTE	CPAR	F	
	1 111112		B	CNV	DSP	TEXT	PULS			W
										D

	12345 67890 123452						PAUSE			B
103		150						103		
NEUTRAL	SIP TRUNK									
	DNNO: 103									
	PDNNO: 203									
	DESTNO :103									
	REROUT :YES									

```
AMO-RICHT-111      TRUNK ROUTING
DISPLAY COMPLETED;
```

DISP-RICHT:LRTE,201;

H500: AMO RICHT STARTED

```

+-----+
| LRTE = 201      NAME = 201-OPEN # SIP      (NEUTRAL)  LSVC = ALL  

| DNNO =          201 PDNNO =                 301  

| ROUTOPT = NO      REROUT = YES    PLB = NO      FWDBL = NO  

| DTMFCNV = FIX     DTMFDSP = WITHOUT DTMFTEXT =  

| DTMFPULS = PP80   BUGS = LIN   ROUTATT = NO      MAINGRP =      7  

| EMCYRTT = NO      CONFTONE = NO    RERINGRP = NO    RTENO =      7  

| INFO =  

| NOPRCFWD = NO  

| NITO = NO  

| CLNAME DL = NO  

| FWDSWTC H = NO  

| LINFEMER = NO  

| NOINTRTE = NO  

+-----+
| TGRP = 150      LDAT  150-SIP                (NEUTRAL)  SUBGROUP =      3  

+-----+

```

```
AMO-RICHT-111      TRUNK ROUTING
DISPLAY COMPLETED;
```

LCR Out-dial Rules, LODR

```
DISP-LODR:104;
```

H500: AMO LODR STARTED

ODR	POSITION	CMD	PARAMETER
104	1	ECHO	1
	2	END	

H03: THE NEXT FREE ODR IS 3

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

DISP-L0DR:202;

H500: AMO LODR STARTED

ODR	POSITION	CMD	PARAMETER
202	1	ECHO	1
	2	ECHO	2
	3	ECHO	3
	4	ECHO	4
	5	END	

H03: THE NEXT FREE ODR IS 3

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

LCR Directions, LDAT

DISP-LDAT: , 103, ;



H500: AMO LDAT STARTED

+-----+-----+-----+-----+-----+-----+-----+-----+-----+-----+									
LROUTE = 103			NAME = SIP TRUNK					SERVICE = ALL	
TYPE = NWLCR			DNNO OF ROUTE =					103	
SERVICE INFO =									
+-----+-----+-----+-----+-----+-----+-----+-----+-----+-----+									
LRTTEL	LVAL	TGRP	ODR	LAUTH	SCHEDULE ABCDEFGH	CARRIER ZONE	LATTR	LDSRT	COTIDX
+-----+-----+-----+-----+-----+-----+-----+-----+-----+-----+									
1	1	150	104	1	*****	1	EMPTY	NONE	0
+-----+-----+-----+-----+-----+-----+-----+-----+-----+-----+									
DNNO = 103									
+-----+-----+-----+-----+-----+-----+-----+-----+-----+-----+									
GW1 = 3-0 GW2 = GW3 =									
GW4 = GW5 =									
+-----+-----+-----+-----+-----+-----+-----+-----+-----+-----+									

AMO-LDAT -111 LCR-DIRECTIONS
DISPLAY COMPLETED;

DISP-LDAT:LCR,201;

H500: AMO LDAT STARTED

+-----+-----+-----+-----+-----+-----+-----+-----+-----+-----+										
LROUTE = 201			LDPLN		NAME = 201-OPEN # SIP			SERVICE = ALL		
TYPE = LCR			DNNO OF ROUTE =				201			
SERVICE INFO =										
+-----+-----+-----+-----+-----+-----+-----+-----+-----+-----+										
LRTTEL	LVAL	TGRP	ODR	LAUTH	SCHEDULE	CARRIER	ZONE	LATTR	LDSRT	COTIDX
+-----+-----+-----+-----+-----+-----+-----+-----+-----+-----+										
1	1	150	202	1	*****	1	EMPTY	PUBNUM		150
+-----+-----+-----+-----+-----+-----+-----+-----+-----+-----+										
DNNO = 201										
+-----+-----+-----+-----+-----+-----+-----+-----+-----+-----+										
GW1 = 3-0 GW2 = GW3 =										
GW4 = GW5 =										
+-----+-----+-----+-----+-----+-----+-----+-----+-----+-----+										

AMO-LDAT -111 LCR-DIRECTIONS
DISPLAY COMPLETED;

Stations configuration

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).

DISP-SBCSU:STNO=7020;

H500: AMO SBCSU STARTED

----- USER DATA -----							
STNO	=7020	OPT	=OPTI	COS1	=5	DPLN	=0
MAINO	=7020	CONN	=DIR	COS2	=5	ITR	=0
PEN	=1- 2- 13- 0			LCOSV1	=15	COSX	=0
INS	=Y	ASYNCT	=500	LCOSV2	=15		
SSTNO	=N	PERMCT	=	LCOSD1	=15	CBKBMX	=5
TRACE	=N	EXTBUS	=	LCOSD2	=15	RCBKB	=N
ALARMNO	=0	DFSVCANA	=	SPDI	=30	RCBKNA	=N
HMUSIC	=0	FLASH	=	SPDC1	=1	CBKNAMB	=N
PMIDX	=1			SPDC2	=1		
SECR	=N	DIGNODIS	=N	DSSTNA	=N	COMGRP	=0
STD	=5	CALLOG	=ALL	DSSTNB	=Y	TEXTSEL	=AMERICAN



```
REP      =0          OPTICOM =N          OPTIUSB :          VPI      =
IDCR     =N          OPTICA  =1          OPTIS0A :0         VCI      =
APPM     =          OPTIDA  =0          OPTISPA :0         PATTERN =
                                OPTIUP0E:0          OPTIABA :0
```

```
DCFWBUSY=N          HEADSET =N          APMOBUSR=          APICLASS=
DNIDSP   =N          HSKEY   =NORMAL    IPCODEC =          SECAPPL =
DTMFBLK  =N          BASICSV=          IPPASSW =
DTMFCTR0=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   :Y          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;

DISP-SBCSU:STNO=7040;
H500: AMO SBCSU STARTED

```
----- USER DATA -----
STNO     =7040          OPT     =FPP          COS1      =5          DPLN      =0
MAIN0    =7040          CONN    =SIP          COS2      =5          ITR       =0
PEN       =1- 1- 1- 2  LCOV1    =15          COSX      =0
INS       =Y          ASYNCT   =          LCOV2    =15
SSTNO    =N          PERMACT  =Y          LCOVSD1  =15          CBKBMAX   =5
TRACE    =N          EXTBUS   =          LCOVSD2  =15          RCBKB     =N
ALARMNO  =0          DFSVCANA=          SPDI      =30          RCBKNA    =N
HMUSIC   =0          FLASH    =          SPDC1     =          CBKNAMB   =
PMIDX    =1          SPDC2     =
DVCFIG   =S0PP          LWPAR   =          PROT     =SBDSS1    OPTIDX   =10
SMGSUB   =N          IPCODEC  =G711P        USERID    =""
FIXEDIP  =N          AUTHREQ  =N          PASSWD    =
IPADDR   =0.0.0.0      SECZONE  =""
DTMFCTR0=Y
----- ACTIVATION IDENTIFIERS FOR FEATURES -----
HTOS     :N          DND      :N
HTOD     :N          VCP      :N          TWLOGIN :
HTOF     :N          CWT      :N
----- FEATURES AND GROUP MEMBERSHIPS -----
PUGR     :          ESSTN    :
KEYSYS   :          NOPTNO   :
SRCGRP   :(1 )      TCLASS   : 0
HUNT CD  :N
----- SUBSCRIBER ATTRIBUTES (AMO SDAT) -----
NONE
H62: FOR SIP SUBSCRIBERS PARAMETER MBCHL (MULTIPLE BCHANNEL) MUST BE
SET IN AMO-SDAT.
```

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



Stations COS configuration:

DISP-COSSU: TYPE=COS, COS=5;

H500: AMO COSSU STARTED

COS	VOICE	FAX	DTE
5	>5-DIGITAL TA TSUID TNOTCR CDRS CDRSTN CDRC CDRIND CDRINT COSXCD MB DATA CFNR VCE SPKR FWDNWK RERING MSN CFB FWDDIR FWDBAS FWDECA FWDEXT CCBS CW GRPCAL SUTVA	NOCO NOTIE	NOCO NOTIE

AMO-COSSU-111 CLASSES OF SERVICE
DISPLAY COMPLETED;

Stations Name configuration:

DISP-PERSI: NAME, 7020;

H500: AMO PERSI STARTED

STNO	CHRISTIAN AND SURNAME	CHARCON	ORGANIZATIONAL UNIT
7020	Hipath1 A1*		

AMO-PERSI-111 PERSONAL IDENTIFICATION DATA
DISPLAY COMPLETED;

DISP-PERSI: NAME, 7040;

H500: AMO PERSI STARTED

STNO	CHRISTIAN AND SURNAME	CHARCON	ORGANIZATIONAL UNIT
7040	Hipath1 A2*		

AMO-PERSI-111 PERSONAL IDENTIFICATION DATA
DISPLAY COMPLETED;



Web-Based Management for HG 3500 V5

Basic Settings – Gateway Properties

■ Front panel ■ Wizard ■ **Explorers** ■ Maintenance ■ Help ■ Logoff

Explorers
[Basic Settings](#)
[Security](#)
[Network Interfaces](#)
[Routing](#)
[Voice Gateway](#)
[Payload](#)
[Statistics](#)

Basic Settings

System

Gateway

Quality of Service

Timezone Settings

Port Management

Online Help Directory

Gateway Properties

General

System Name: hg3500
Gateway Location:
Contact Address:
System Country Code: 1 (United States of America)
Gateway IP Address: 172.20.188.13
Gateway Subnet Mask: 255.255.255.0

Additional Features

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



Voice Gateway - SIP Parameters

Front panel Wizard **Explorers** Maintenance Help Logoff

HG 3500 V5

Explorers

[Basic Settings](#)

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- Voice Gateway
 - H.323 Parameters
 - SIP Parameters**
 - Codec Parameters
 - IP Networking Mode
 - SIP Trunk Profile Parameter
 - SIP Trunk Profiles
 - Destination Codec Parameters
 - Fallback to SCN Parameters
 - Clients

SIP Parameters

SIP User Agent

Use SIP Registrar: No
SIP Registrar IP Address: 0.0.0.0
SIP Registrar TLS Port Number: 5061
SIP Registrar TCP/UDP Port Number: 5060
Alternative SIP Registrar IP Address: 0.0.0.0
Alternative SIP Registrar TLS Port Number: 5061
Alternative SIP Registrar TCP/UDP Port Number: 5060
Period of registration (sec): 120

SIP Server (Registrar / Redirect)

SIP Server IP Address: 172.20.188.13
SIP Server TCP/UDP Port Number: 5060
SIP Server TLS Port Number: 5061
Period of registration (sec): 120

RFC 3261 Timer Values

Transaction Timeout (msec): 32000

SIP Transport Protocol

SIP via TCP: Yes
SIP via UDP: Yes
SIP via TLS: Yes

SIP Session Timer

RFC 4028 support: Yes
Session Expires (sec): 1800
Minimal SE (sec): 90



Voice Gateway – Codec Parameters

Front panel Wizard **Explorers** Maintenance Help Logoff

HG 3500 V5

Explorers

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Voice Gateway

- H.323 Parameters
- SIP Parameters
- Codec Parameters**
- IP Networking Mode
- SIP Trunk Profile Parameter
- SIP Trunk Profiles
- Destination Codec Parameters
- Fallback to SCN Parameters
- Clients

Codec Parameters

Codec	Priority	Voice Activity Detection	Frame Size
G.711 A-law	Priority 3	Off	20 msec
G.711 μ -law	Priority 2	Off	20 msec
G.723	not used	Off	30 msec
G.729	not used	Off	20 msec
G.729A	Priority 1	Off	20 msec
G.729B	not used	On	20 msec
G.729AB	not used	On	60 msec

T.38 Fax

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

Misc.

ClearMode (ClearChannelData): Off Frame Size: 20 msec

RFC2833

Transmission of Fax/Modem Tones according to RFC2833: On
Transmission of DTMF Tones according to RFC2833: On
Payload Type for RFC2833: 101
Redundant Transmission of RFC2833 Tones according to RFC2198: On

Voice Gateway – SIP Trunk Profile Parameters

Front panel Wizard **Explorers** Maintenance Help Logoff

HG 3500 V5

Explorers

[Basic Settings](#)
[Security](#)
[Network Interfaces](#)
[Routing](#)
[Voice Gateway](#)
[Payload](#)
[Statistics](#)

Voice Gateway

- H.323 Parameters
- SIP Parameters
- Codec Parameters
- IP Networking Mode
- SIP Trunk Profile Parameter**
- SIP Trunk Profiles
- Destination Codec Parameters
- Fallback to SCN Parameters
- Clients

SIP Trunk Profile Parameter

SIP Protocol Variant for IP Networking: Native SIP
Use Profiles for Trunks via Native SIP: Yes



Voice Gateway – SIP Trunk Profile

The screenshot displays the Cisco Unified Communications Manager (HG 3500 V5) interface. The top navigation bar includes links for Front panel, Wizard, Explorers, Maintenance, Help, and Logoff. The left sidebar shows the Explorers menu with options: Basic Settings, Security, Network Interfaces, Routing, Voice Gateway, Payload, and Statistics. The main content area is titled "SIP Trunk Profile" and shows the configuration for the CM-TITAN profile. The configuration is organized into several sections:

- Provider Information:**
 - Provider Name: CM-TITAN
 - Account/Authentication Required: No
 - Domain Name:
 - SIP Transport Protocol: UDP
- Registrar:**
 - Use Registrar: No
 - IP Address / Host name: 0.0.0.0
 - Port: 0
 - Reregistration Interval (sec): 120
- Proxy:**
 - IP Address / Host name: 172.20.236.252
 - Port: 5060
- Outbound Proxy:**
 - Use Outbound Proxy: No
 - IP Address / Host name: 0.0.0.0
 - Port: 0
- Inbound Proxy:**
 - Use Inbound Proxy: No
 - IP Address / Host name: 0.0.0.0
 - Port: 0

Configuring the Cisco Unified Communications Manager – Session Manager Edition

1. Cisco Session Manager Version
2. Device Pool and Region mapping configuration
3. SIP Profile (used by SIP trunks) configuration
4. SIP Trunk Security Profile (used by SIP trunks) configuration
5. SIP trunk configuration to Siemens PBX
6. SIP Trunk configuration to Cisco UCM
7. SIP Trunk to Cisco UBE
8. Route Pattern configuration to Siemens PBX
9. Route Pattern configuration to Cisco UCM
10. Route Pattern configuration to PSTN



Cisco Unified Communications Manager – Session Manager Edition software version

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

Navigation Cisco Unified CM Administration Go

ccmadministrator | Search Documentation | About | Logout

System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help

Cisco Unified CM Administration

System version: 8.5.1.10000-26



Last Successful Login: Jan 26, 2011 8:48:02 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

Navigation Path: System → Region

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

Navigation Cisco Unified CM Administration Go

ccmadministrator | Search Documentation | About | Logout

System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help

Region Configuration Related Links: Back To Find/List Go

Save Delete Reset Apply Config Add New

Region Information
Name* Default

Region Relationships

Region	Max Audio Bit Rate	Max Video Call Bit Rate (Includes Audio)	Link Loss Type
Default	64 kbps (G.722, 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	Max Audio Bit Rate	Max Video Call Bit Rate (Includes Audio)	Link Loss Type
Default	Keep Current Setting	☒ Keep Current Setting ☐ Use System Default ☐ None ☐ kbps	Keep Current Setting

Save Delete Reset Apply Config Add New



Configuration of SIP Profile used by SIP trunks

Navigation Path: Device → Device Settings → SIP Profile

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 the subtitle "For Cisco Unified Communications Solutions". The user is logged in as "ccmadministrator". The main menu shows various configuration areas: System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The current page is "SIP Profile Configuration", which is part of the "Device" configuration path. The page title is "SIP Profile Configuration" and it includes a "Related Links" section with a "Back To Find/List" button. The configuration area is titled "SIP Profile Information" and contains the following fields and options:

- Name*: Early Media SIP Profile
- Description: SIP Profile with Early Media and OPTIONS Enabled
- Default MTP Telephony Event Payload Type*: 101
- Resource Priority Namespace List: < None >
- Early Offer for G.Clear Calls*: Disabled
- ☐ Redirect by Application
- ☐ Disable Early Media on 180
- ☐ Outgoing T.38 INVITE include audio mline
- ☐ Enable ANAT
- ☐ Require SDP Inactive Exchange for Mid-Call Media Change



Parameters used in Phone

Timer Invite Expires (seconds)*	180
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10
Start Media Port*	16384
Stop Media Port*	32766
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off
Caller ID Blocking*	Off
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70
Off Hook To First Digit Timer (milliseconds)*	15000
Call Forward URI*	x-cisco-serviceuri-cfwdall
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial
☒ Conference Join Enabled	
☐ RFC 2543 Hold	
☒ Semi Attended Transfer	
☐ Enable VAD	
☐ Stutter Message Waiting	

Trunk Specific Configuration

Reroute Incoming Request to new Trunk based on*	Never
RSVP Over SIP*	Local RSVP
☒ Fall back to local RSVP	
SIP RelXX Options*	Send PRACK if 1xx Contains SDP
☐ Deliver Conference Bridge Identifier	
☒ Early Offer support for voice and video calls (insert MTP if needed)	
☐ Send send-receive SDP in mid-call INVITE	

SIP OPTIONS Ping

☒ Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"

Ping Interval for In-service and Partially In-service Trunks (seconds)*	60
Ping Interval for Out-of-service Trunks (seconds)*	120
Ping Retry Timer (milliseconds)*	500
Ping Retry Count*	6



Configuration of SIP Trunk Security Profile used by SIP trunks

Navigation Path: System → Security → SIP Trunk Security Profile

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go

ccmadministrator | [Search Documentation](#) | [About](#) | [Logout](#)

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

SIP Trunk Security Profile Configuration Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

Status
Status: Ready

SIP Trunk Security Profile Information

Name*	Non Secure SIP Trunk Profile
Description	Non Secure SIP Trunk Profile authenticated by null Str
Device Security Mode	Non Secure
Incoming Transport Type*	TCP+UDP
Outgoing Transport Type	TCP
☐ Enable Digest Authentication	
Nonce Validity Time (mins)*	600
X.509 Subject Name	
Incoming Port*	5060
☐ Enable Application Level Authorization	
☐ Accept Presence Subscription	
☒ Accept Out-of-Dialog REFER**	
☒ Accept Unsolicited Notification	
☐ Accept Replaces Header	
☐ Transmit Security Status	



Configuration of SIP trunk to Siemens PBX

Navigation Path: Device → Trunk

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

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System ▾ | Call Routing ▾ | Media Resources ▾ | Advanced Features ▾ | Device ▾ | Application ▾ | User Management ▾ | Bulk Administration ▾ | Help ▾

Trunk Configuration

Related Links: Back To Find/List

 Save  Delete  Reset  Add New

Device Information

Product:

SIP Trunk

Device Protocol:

SIP

Trunk Service Type

None(Default)

Device Name*

Siemens_Hipath4000_rel5

Description

Siemens Hipath 4000 Release 5

Device Pool*

Default

Common Device Configuration

< None >

Call Classification*

Use System Default

Media Resource Group List

< None >

Location*

Hub_None

AAR Group

< None >

Tunneled Protocol*

None

QSIG Variant*

No Changes

ASN.1 ROSE OID Encoding*

No Changes

Packet Capture Mode*

None

Packet Capture Duration

0

☐ Media Termination Point Required

☒ Retry Video Call as Audio

☐ Path Replacement Support

☐ Transmit UTF-8 for Calling Party Name

☐ Transmit UTF-8 Names in QSIG APDU

☐ Unattended Port

☐ 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.

Consider Traffic on This Trunk Secure*

When using both sRTP and TLS

Route Class Signaling Enabled*

Default

Use Trusted Relay Point*

Default

☒ PSTN Access

☐ Run On All Active Unified CM Nodes

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Intercompany Media Engine (IME)				
E.164 Transformation Profile < 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				
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	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	☒
Connected Party Settings				
Connected Party Transformation CSS < None >				
☒ Use Device Pool Connected Party Transformation CSS				



Outbound Calls

Called Party Transformation CSS

☒ Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS

☒ Use Device Pool Calling Party Transformation CSS

Calling Party Selection*

Calling Line ID Presentation*

Calling Name Presentation*

Caller ID DN

Caller Name

☒ Redirecting Diversion Header Delivery - Outbound

SIP Information

Destination

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1*	172.20.188.13		5060 + -

MTP Preferred Originating Codec*

Presence Group*

SIP Trunk Security Profile*

Rerouting Calling Search Space

Out-Of-Dialog Refer Calling Search Space

SUBSCRIBE Calling Search Space

SIP Profile*

DTMF Signaling Method*

Normalization Script

Normalization Script

☐ Enable Trace

	Parameter Name	Parameter Value
1		+ -



Configuration of SIP trunk to Cisco UCM

Navigation Path: Device → Trunk

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration

ccmadministrator

Search Documentation

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Trunk Configuration

Related Links: Back To Find/List

Save Delete Reset Add New

Device Information

Product:

SIP Trunk

Device Protocol:

SIP

Trunk Service Type

None(Default)

Device Name*

CM_POLARIS_SIP

Description

SIP Trunk to CM-Polaris

Device Pool*

Default

Common Device Configuration

< None >

Call Classification*

Use System Default

Media Resource Group List

< None >

Location*

Hub_None

AAR Group

< None >

Tunneled Protocol*

None

QSIG Variant*

No Changes

ASN.1 ROSE OID Encoding*

No Changes

Packet Capture Mode*

None

Packet Capture Duration

0

☐ Media Termination Point Required

☒ Retry Video Call as Audio

☐ Path Replacement Support

☐ Transmit UTF-8 for Calling Party Name

☐ Transmit UTF-8 Names in QSIG APDU

☐ Unattended Port

☐ 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.

Consider Traffic on This Trunk Secure*

When using both sRTP and TLS

Route Class Signaling Enabled*

Default

Use Trusted Relay Point*

Default

☒ PSTN Access

☐ Run On All Active Unified CM Nodes



Intercompany Media Engine (IME)				
E.164 Transformation Profile < 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				
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	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	☒
Connected Party Settings				
Connected Party Transformation CSS < None >				
☒ Use Device Pool Connected Party Transformation CSS				



Outbound Calls		
Called Party Transformation CSS	< None >	
☒ Use Device Pool Called Party Transformation CSS		
Calling Party Transformation CSS	< None >	
☒ 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		
☒ Redirecting Diversion Header Delivery - Outbound		

SIP Information		
Destination		
☐ Destination Address is an SRV		
	Destination Address	Destination Address IPv6
1 *	172.20.236.50	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 *	Early Media SIP Profile	
DTMF Signaling Method *	RFC 2833	

Normalization Script		
Normalization Script	< None >	
☐ Enable Trace		
	Parameter Name	Parameter Value
1		



Configuration of SIP trunk to Cisco UBE

Navigation Path: Device → Trunk

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

Navigation Cisco Unified CM Administration

ccmadministrator | Search Documentation | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

Trunk Configuration

Related Links: Back To Find/List

Save Delete Reset Add New

Device Information

Product:

SIP Trunk

Device Protocol:

SIP

Trunk Service Type

None(Default)

Device Name*

Verizon_SIP_Trunk

Description

Verizon SIP trunk to PSTN

Device Pool*

Default

Common Device Configuration

< None >

Call Classification*

Use System Default

Media Resource Group List

< None >

Location*

Hub_None

AAR Group

< None >

Tunneled Protocol*

None

QSIG Variant*

No Changes

ASN.1 ROSE OID Encoding*

No Changes

Packet Capture Mode*

None

Packet Capture Duration

0

☐ Media Termination Point Required

☒ Retry Video Call as Audio

☐ Path Replacement Support

☐ Transmit UTF-8 for Calling Party Name

☐ Transmit UTF-8 Names in QSIG APDU

☐ Unattended Port

☐ 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.

Consider Traffic on This Trunk Secure*

When using both sRTP and TLS

Route Class Signaling Enabled*

Default

Use Trusted Relay Point*

Default

☒ PSTN Access

☐ Run On All Active Unified CM Nodes



Intercompany Media Engine (IME)				
E.164 Transformation Profile < 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				
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	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	☒
Connected Party Settings				
Connected Party Transformation CSS < None >				
☒ Use Device Pool Connected Party Transformation CSS				



Outbound Calls		
Called Party Transformation CSS	< None >	
☒ Use Device Pool Called Party Transformation CSS		
Calling Party Transformation CSS	< None >	
☒ Use Device Pool Calling Party Transformation CSS		
Calling Party Selection *	Originator	
Calling Line ID Presentation *	Default	
Calling Name Presentation *	Default	
Caller ID DN	4089333321	
Caller Name	Cisco Systems, Inc.	
☒ Redirecting Diversion Header Delivery - Outbound		

SIP Information		
Destination		
☐ Destination Address is an SRV		
1 *	Destination Address	Destination Address IPv6 Destination Port
	172.20.110.151	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 *	Early Media SIP Profile	
DTMF Signaling Method *	RFC 2833	

Normalization Script		
Normalization Script < None >		
☐ Enable Trace		
1	Parameter Name	Parameter Value

Note: Service Providers require caller ID to match a valid DID number assigned to customer in order to process calls. Entering the billing phone number assigned to SIP trunk by Service Provider within the Caller ID DN field ensures that all outbound calls to Service Provider are processed.



Configuration of Route Patterns – To Siemens PBX

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration

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Route Pattern Configuration

Related Links: Back To Find/List

Save Delete Copy Add New

Pattern Definition

Route Pattern*70XX
Route Partition< None >
DescriptionTo Siemens Hipath 4000 Rel. 5
Numbering Plan-- Not Selected --
Route Filter< None >
MLPP Precedence*Default
Resource Priority Namespace Network Domain< None >
Route Class*Default
Gateway/Route List*Siemens_Hipath4000_rel5 (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 ServiceService Parameter NameService Parameter Value
-- Not Selected --< Not Exist >



Configuration of Route Patterns – To Cisco UCM

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration

ccmadministrator | Search Documentation | About | Logout

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Route Pattern Configuration Related Links: Back To Find/List

Save Delete Copy Add New

Pattern Definition

Route Pattern*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence*

Resource Priority Namespace Network Domain

Route Class*

Gateway/Route List*

Route Option

Call Classification*

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

☐ Require Forced Authorization Code

Authorization Level*

☐ Require Client Matter Code

50XX

< None >

Route to CM-Polaris

-- Not Selected --

< None >

Default

< None >

Default

CM_POLARIS_SIP

☒ Route this pattern
☐ Block this pattern

No Error

OnNet

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
-- Not Selected --	< Not Exist >	



Configuration of Route Patterns – To PSTN

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration

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Route Pattern Configuration Related Links: Back To Find/List

Save Delete Copy Add New

Pattern Definition

Route Pattern *	9.@
Route Partition	< None >
Description	
Numbering Plan *	NANP
Route Filter	< None >
MLPP Precedence *	Default
Resource Priority Namespace Network Domain	< None >
Route Class *	Default
Gateway/Route List *	Verizon_SIP_Trunk (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

Connected Party Transformations

Connected Line ID Presentation *	Default
Connected Name Presentation *	Default

Called Party Transformations

Discard Digits	PreDot
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 Translation Patterns – To Siemens PBX

Navigation Path: Call Routing → Translation Pattern

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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Translation Pattern Configuration Related Links: [Back To Find/List](#)

Save Delete Copy Add New

Pattern Definition

Translation Pattern

408933.7XXX

Partition

< None >

Description

Translation to Siemens 4-digit numbers

Numbering Plan

< None >

Route Filter

< None >

MLPP Precedence*

Default

Resource Priority Namespace Network Domain

< None >

Route Class*

Default

Calling Search Space

< None >

External Call Control Profile

< None >

Route Option

☒ Route this pattern

☐ Block this pattern

No Error

☐ Provide Outside Dial Tone

☒ Urgent Priority

☐ Route Next Hop By Calling Party Number

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

PreDot

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type*

Cisco CallManager

Called Party Numbering Plan*

Cisco CallManager

Note: This Translation Pattern is used to convert 10-digit DID numbers to 4-digit extension assigned to Siemens stations.



Configuration of Translation Patterns – To Cisco UCM

Navigation Path: Call Routing → Translation Pattern

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

Navigation Cisco Unified CM Administration G

ccmadministrator | Search Documentation | About | Logout

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Translation Pattern Configuration

Related Links: Back To Find/List G

Save X Delete Copy + Add New

Status
 Status: Ready

Pattern Definition

Translation Pattern

408933.5XXX

Partition

< None >

Description

Translation to Cisco UCM 4-digit numbers

Numbering Plan

< None >

Route Filter

< None >

MLPP Precedence*

Default

Resource Priority Namespace Network Domain

< None >

Route Class*

Default

Calling Search Space

< None >

External Call Control Profile

< None >

Route Option

☒ Route this pattern

☐ Block this pattern No Error

☐ Provide Outside Dial Tone

☒ Urgent Priority

☐ Route Next Hop By Calling Party Number

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 PreDot

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type* Cisco CallManager

Called Party Numbering Plan* Cisco CallManager

Note: This Translation Pattern is used to convert 10-digit DID numbers to 4-digit extension assigned to Cisco UCM stations.



Configuring the Cisco Unified Communications Manager

1. Cisco Unified Communications Manager Version
2. Service Parameter configuration – Duplex Streaming of MoH
3. Device pool and Region mapping configuration
4. Conference Bridge configuration
5. Media Resource Group configuration
6. Media Resource Group List configuration
7. SIP Profile configuration
8. SIP Trunk to SME configuration
9. Route Pattern configuration to Siemens PBX
10. Route Pattern configuration to PSTN
11. Cisco IP Phone 7965 SCCP configuration
12. Cisco IP Phone 7965 SIP configuration
13. MGCP Fax gateway configuration

Cisco Unified Communications Manager software version

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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Cisco Unified CM Administration
System version: 8.5.1.10000-26

Last Successful Logon: Jan 26, 2011 8:48:02 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 Service Parameters – Duplex Streaming for MoH

Navigation Path: System → Service Parameter

The screenshot shows the 'Service Parameter Configuration' page in Cisco Unified CM Administration. The 'Clusterwide Parameters (Service)' section is expanded, showing a list of parameters. The 'Duplex Streaming Enabled' parameter is highlighted, and its value is set to 'True'.

Parameter	Value	Default
Default Network Hold MOH Audio Source ID *	1	1
Default User Hold MOH Audio Source ID *	1	1
Duplex Streaming Enabled *	True	False
Media Exchange Interface Capability Timer *	8	8
Send Multicast MOH in H.245 OLC Message *	True	True
Media Exchange Timer *	12	12
Media Exchange Stop Streaming Timer *	8	8
Open Video Channel Response Timer for SIP Interop *	500	500
Port Received Timer After Call Connection *	500	500
Media Resource Allocation Timer *	12	12
MTP and Transcoder Resource Throttling Percentage *	95	95
Intercluster Capabilities Mismatch Timer *	1000	1000
Silence Suppression *	False	False
Silence Suppression for Gateways *	False	False
Strip G.729 Annex B (Silence Suppression) from Capabilities *	False	False

Note: In order for Music-on-hold to be heard by Siemens and/or PSTN end-points, service parameter “Duplex Streaming Enabled” must be set to “True”.

Configuration of Device Pool to Region mapping

Navigation Path: System → Region

The screenshot shows the 'Region Configuration' page in Cisco Unified CM Administration. The 'Region Information' section shows the 'Name' as 'Default'. The 'Region Relationships' table shows the mapping of regions to audio codecs, video call bandwidth, and link loss type.

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

The 'Modify Relationship to other Regions' section shows the 'Regions' list with 'Default' and 'G711_Region'. The 'Audio Codec' is set to 'Keep Current Setting', the 'Video Call Bandwidth' is set to 'Keep Current Setting', and the 'Link Loss Type' is set to 'Keep Current Setting'.



Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go

ccmadadministrator | 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

Region Information

Name* G711_Region

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
Default G711_Region	Keep Current Setting	☒ Keep Current Setting ☐ Use System Default ☐ None kbps	Keep Current Setting

Save Delete Reset Apply Config Add New

Configuration of Conference Bridge

Navigation Path: Media Resources → Conference Bridge

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Navigation Cisco Unified CM Administration Go

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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 IOS 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

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 set to "Cisco Unified CM Administration". Below this is a secondary navigation bar with links for "ccmadministrator", "Search Documentation", "About", and "Logout". The main content area is titled "Media Resource Group Configuration" and includes a "Related Links" section with a "Back To Find/List" link. The configuration page is divided into several sections: "Status" (showing "Status: Ready"), "Media Resource Group Status" (showing "Media Resource Group: MRG_Polaris (used by 125 devices)"), "Media Resource Group Information" (with fields for "Name*" and "Description", both containing "MRG_Polaris"), and "Devices for this Group". The "Devices for this Group" section contains two lists: "Available Media Resources**" (listing C0300115C28E8B C, CFB0001C9D93A99, CFB_2, and MTP_2) and "Selected Media Resources*" (listing ANN_2 (ANN), CFB112233445566 (CFB), MOH_2 (MOH), and MTP0015F90D0970 (XCODE)). At the bottom of this section is a checkbox labeled "Use Multi-cast for MOH Audio (If at least one multi-cast MOH resource is available)".



Configuration of 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

ccmadministrator | 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
i 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 SIP Profile

Navigation Path: Device → Device Settings → SIP Profile

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration

ccmadministrator | Search Documentation | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

SIP Profile Configuration Related Links: Back To Find/List

Save Delete Copy Reset Apply Config Add New

SIP Profile Information

Name*

Early Media and Options SIP Profile

Description

Default SIP Profile

Default MTP Telephony Event Payload Type*

101

Resource Priority Namespace List

< None >

Early Offer for G.Clear Calls*

Disabled

☐ Redirect by Application

☐ Disable Early Media on 180

☐ Outgoing T.38 INVITE include audio mline

☐ Enable ANAT

☐ Require SDP Inactive Exchange for Mid-Call Media Change

Parameters used in Phone

Timer Invite Expires (seconds)*

180

Timer Register Delta (seconds)*

5

Timer Register Expires (seconds)*

3600

Timer T1 (msec)*

500

Timer T2 (msec)*

4000

Retry INVITE*

6

Retry Non-INVITE*

10

Start Media Port*

16384

Stop Media Port*

32766

Call Pickup URI*

x-cisco-serviceuri-pickup

Call Pickup Group Other URI*

x-cisco-serviceuri-opickup

Call Pickup Group URI*

x-cisco-serviceuri-gpickup

Meet Me Service URI*

x-cisco-serviceuri-meetme

User Info*

None

DTMF DB Level*

Nominal

Call Hold Ring Back*

Off



Caller ID Blocking*	Off
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70
Off Hook To First Digit Timer (milliseconds)*	15000
Call Forward URI*	x-cisco-serviceuri-cfwdall
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial
☒ Conference Join Enabled	
☐ RFC 2543 Hold	
☒ Semi Attended Transfer	
☐ Enable VAD	
☐ Stutter Message Waiting	

Trunk Specific Configuration

Reroute Incoming Request to new Trunk based on*	Never
RSVP Over SIP*	Local RSVP
☒ Fall back to local RSVP	
SIP Rel1XX Options*	Send PRACK if 1xx Contains SDP
☐ Deliver Conference Bridge Identifier	
☒ Early Offer support for voice and video calls (insert MTP if needed)	
☐ Send send-receive SDP in mid-call INVITE	

SIP OPTIONS Ping

☒ Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"	
Ping Interval for In-service and Partially In-service Trunks (seconds)*	60
Ping Interval for Out-of-service Trunks (seconds)*	120
Ping Retry Timer (milliseconds)*	500
Ping Retry Count*	6



Intercompany Media Engine (IME)				
E.164 Transformation Profile < 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		tp_phones_rp		
AAR Calling Search Space		< None >		
Prefix DN 				
☒ Redirecting Diversion Header Delivery - Inbound				
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	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	☒
Connected Party Settings				
Connected Party Transformation CSS < None >				
☒ Use Device Pool Connected Party Transformation CSS				



Outbound Calls

Called Party Transformation CSS

☒ Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS

☒ Use Device Pool Calling Party Transformation CSS

Calling Party Selection*

Calling Line ID Presentation*

Calling Name Presentation*

Caller ID DN

Caller Name

☒ Redirecting Diversion Header Delivery - Outbound

SIP Information

Destination

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port	
1 *	172.20.236.252		5060	+ -

MTP Preferred Originating Codec*

Presence Group*

SIP Trunk Security Profile*

Rerouting Calling Search Space

Out-Of-Dialog Refer Calling Search Space

SUBSCRIBE Calling Search Space

SIP Profile*

DTMF Signaling Method*

Normalization Script

Normalization Script

☐ Enable Trace

	Parameter Name	Parameter Value	
1			+ -



Configuration of Route Pattern to Siemens PBX via Cisco UCM – SME

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

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

Navigation Cisco Unified CM Administration

ccmadministrator | [Search Documentation](#) | [About](#) | [Logout](#)

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

 Save  Delete  Copy  Add New

Pattern Definition

Route Pattern*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence*

Resource Priority Namespace Network Domain

Route Class*

Gateway/Route List*

Route Option

7XXX

route_p

To Hipath 4000 Rel. 5.0

-- Not Selected --

< None >

Default

< None >

Default

CM_TITAN_SIP [\(Edit\)](#)

☒ Route this pattern
☐ Block this pattern No Error

Call Classification*

☐ 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*

Calling Name Presentation*

Calling Party Number Type*

Calling Party Numbering Plan*

Default

Default

Cisco CallManager

Cisco CallManager

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Default

Default

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type*

Called Party Numbering Plan*

< None >

Cisco CallManager

Cisco CallManager

ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

Network Service

Service Parameter Name

Service Parameter Value

-- Not Selected --

-- Not Selected --

< Not Exist >



Configuration of Route Pattern to PSTN via Cisco UCM – SME

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration

ccmadministrator | Search Documentation | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

Route Pattern Configuration Related Links: Back To Find/List

Save Delete Copy Add New

Status
Update successful

Pattern Definition
Route Pattern* 9@
Route Partition route_p
Description To PSTN via SME
Numbering Plan* NANP
Route Filter < None >
MLPP Precedence* Default
Resource Priority Namespace Network Domain < None >
Route Class* Default
Gateway/Route List* CM_TITAN_SIP (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

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 PSTN via Cisco UCM - SME (G.711 only calls)

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

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

Navigation Cisco Unified CM Administration

ccmadministrator | Search Documentation | About | Logout

System ▾ | Call Routing ▾ | Media Resources ▾ | Advanced Features ▾ | Device ▾ | Application ▾ | User Management ▾ | Bulk Administration ▾ | Help ▾

Route Pattern Configuration

Related Links: Back To Find/List

Save  Delete  Copy  Add New

Pattern Definition

Route Pattern*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence*

Resource Priority Namespace Network Domain

Route Class*

Gateway/Route List*

Route Option

81XXXXXXXXXX

route_p

Route to PSTN via G.711

-- Not Selected --

< None >

Default

< None >

Default

CM_TITAN_SIP [\(Edit\)](#)

☒ Route this pattern
☐ Block this pattern No Error

Call Classification*

☐ 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*

Calling Name Presentation*

Calling Party Number Type*

Calling Party Numbering Plan*

Default

Default

Cisco CallManager

Cisco CallManager

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Default

Default

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type*

Called Party Numbering Plan*

< None >

Cisco CallManager

Cisco CallManager

ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

Network Service

Service Parameter Name

Service Parameter Value

-- Not Selected --

-- Not Selected --

< Not Exist >

Note: The Route Pattern above is used for outbound calls requiring G.711 codec (e.g. fax and/or modem calls). This Route Pattern allows the outpulsing of 8+1+10 digits, which will match a dial-peer configured to support G.711-only calls. The dial-peer in question will use a translation rule to strip out the leading “8” and send the 1+10 digit number to the Service Provider for processing.



25	None	Device Mobility Mode*	Default	View Current Device
26	None		Mobility Settings	
27	None	Owner User ID	< None >	
28	None	Phone Personalization*	Default	
29	None	Services Provisioning*	Default	
30	None	Phone Load Name		
31	None	Single Button Barge	Default	
32	None	Join Across Lines	Default	
33	None	Use Trusted Relay Point*	Default	
34	None	BLF Audible Alert Setting (Phone Idle)*	Default	
35	None	BLF Audible Alert Setting (Phone Busy)*	Default	
36	None	Always Use Prime Line*	Default	
37	None	Always Use Prime Line for Voice Message*	Default	
38	None	Calling Party Transformation CSS	< None >	
39	None	Geolocation	< None >	
40	None	☒ Use Device Pool Calling Party Transformation CSS		
41	None	☒ Retry Video Call as Audio		
42	None	☐ Ignore Presentation Indicators (internal calls only)		
43	None	☒ Allow Control of Device from CTI		
44	None	☐ Logged Into Hunt Group		
45	None	☐ Remote Device		
46	None	☐ Protected Device ****		
47	None	☐ Hot line Device *****		
48	None			
49	None			
50	None			
51	None			
52	None			
53	None			
54	None			
----- Unassigned Associated Items -----				
55	Add a new SD	Protocol Specific Information	Packet Capture Mode*	None
56	Add a new SURF		Packet Capture Duration	0
			Presence Group*	Standard Presence group
			Device Security Profile*	Cisco 7965 - Standard SCCP Non-Secure Profile
			SUBSCRIBE Calling Search Space	< None >



57	Add a new BLF SD	☐ Unattended Port
58	Add a new BLF Directed Call Park	☐ Require DTMF Reception
59	CallBack	☐ RFC2833 Disabled
60	Call Park	
61	Call Pickup	
62	Conference List	Certification Authority Proxy Function (CAPF) Information
63	Conference	Certificate Operation* No Pending Operation
64	Do Not Disturb	Authentication Mode* By Null String
65	End Call	Authentication String
66	Forward All	Generate String
67	Group Call Pickup	Key Size (Bits)* 1024
68	Hold	Operation Completes By 2011 4 11 12 (YYYY:MM:DD:HH)
69	Hunt Group Logout	Certificate Operation Status: None
70	Intercom [1] - Add a new Intercom	Note: Security Profile Contains Addition CAPF Settings.
71	Malicious Call Identification	
72	Meet Me Conference	Expansion Module Information
73	Mobility	Module 1 < None >
74	New Call	Module 1 Load Name
75	Other Pickup	Module 2 < None >
76	Quality Reporting Tool	Module 2 Load Name
77	Redial	
78	Remove Last Participant	External Data Locations Information (Leave blank to use default)
79	Transfer	Information
80	Video Mode	Directory
81	Privacy	Messages
82	None	Services
		Authentication Server
		Proxy Server
		Idle
		Idle Timer (seconds)
		Secure Authentication URL
		Secure Directory URL



Secure Idle URL	
Secure Information URL	
Secure Messages URL	
Secure Services URL	

Extension Information

☐ Enable Extension Mobility	
Log Out Profile	-- Use Current Device Settings --
Log in Time	< None >
Log out Time	< None >

MLPP Information

MLPP Domain	< None >
MLPP Indication*	Default
MLPP Preemption*	Default

Do Not Disturb

☐ Do Not Disturb	
DND Option*	Use Common Phone Profile Setting
DND Incoming Call Alert	< None >

Secure Shell Information

Secure Shell User	
Secure Shell Password	

Product Specific Configuration Layout

		Param	Override Common Settings
☐ Disable Speakerphone			
☐ Disable Speakerphone and Headset			
Forwarding Delay*		Disabled	
PC Port *		Enabled	



Settings Access*	Enabled	☐
Gratuitous ARP*	Disabled	
PC Voice VLAN Access*	Enabled	
Video Capabilities*	Disabled	☐
Auto Line Select*	Disabled	
Web Access*	Disabled	☐
Days Display Not Active	Sunday Monday Tuesday	☐
Display On Time	07:30	☐
Display On Duration	10:30	☐
Display Idle Timeout	01:00	☐
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		
Display On When Incoming Call*	Disabled	
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 Headset*	Enabled	
Peer Firmware Sharing*	Enabled	☐
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	
Wireless Headset Hookswitch Control*	Disabled	
IPv6 Load Server		☐
IPv6 Log Server		
802.1x Authentication*	User Controlled	☐
Detect Unified CM Connection Failure*	Normal	
Minimum Ring Volume*	0-Silent	
HTTPS Server*	http and https Enabled	
Handset/Headset Monitor*	Enabled	
Enbloc Dialing*	Enabled	
Switch Port Remote Configuration*	Disabled	☐
PC Port Remote Configuration*	Disabled	☐
Automatic Port Synchronization*	Disabled	☐



24	None	Device Mobility Mode*	Default	View Current Device
25	None		Mobility Settings	
26	None	Owner User ID	< None >	
27	None	Phone Personalization*	Default	
28	None	Services Provisioning*	Default	
29	None	Phone Load Name		
30	None	Single Button Barge	Default	
31	None	Join Across Lines	Default	
32	None	Use Trusted Relay Point*	Default	
33	None	BLF Audible Alert Setting (Phone Idle)*	Default	
34	None	BLF Audible Alert Setting (Phone Busy)*	Default	
35	None	Always Use Prime Line*	Default	
36	None	Always Use Prime Line for Voice Message*	Default	
37	None	Calling Party Transformation CSS	< None >	
38	None	Geolocation	< None >	
39	None	☒ Use Device Pool Calling Party Transformation CSS		
40	None	☐ Ignore Presentation Indicators (internal calls only)		
41	None	☒ Allow Control of Device from CTI		
42	None	☒ Logged Into Hunt Group		
43	None	☐ Remote Device		
44	None	☐ Protected Device ****		
45	None	☐ Hot line Device *****		
46	None	Protocol Specific Information		
47	None	Packet Capture Mode*	None	
48	None	Packet Capture Duration	0	
49	None	Presence Group*	Standard Presence group	
50	None	SIP Dial Rules	< None >	
51	None	MTP Preferred Originating Codec*	711ulaw	
52	None	Device Security Profile*	Cisco 7965 - Standard SIP Non-Secure Profile	
53	None	----- Unassigned Associated Items -----		
54	None	Line [4] - Add a new DN		
55	None	Add a new SD		



57	Add a new SURL	Rerouting Calling Search Space	< None >
58	Add a new BLF SD	SUBSCRIBE Calling Search Space	< None >
59	Add a new BLF Directed Call Park	SIP Profile*	Standard SIP Profile
60	Do Not Disturb	Digest User	< None >
61	Intercom [1] - Add a new Intercom	☐ Media Termination Point Required	
62	Call Park	☐ Unattended Port	
63	Call Pickup	☐ Require DTMF Reception	
64	CallBack	Certification Authority Proxy Function (CAPF) Information	
65	Conference List	Certificate Operation*	No Pending Operation
66	Conference	Authentication Mode*	By Null String
67	End Call	Authentication String	
68	Forward All	Generate String	
69	Group Call Pickup	Key Size (Bits)*	1024
70	Hold	Operation Completes By	2011 4 11 12 (YYYY:MM:DD:HH)
71	Hunt Group Logout	Certificate Operation Status: None	
72	Malicious Call Identification	Note: Security Profile Contains Addition CAPF Settings.	
73	Meet Me Conference	Expansion Module Information	
74	Mobility	Module 1	< None >
75	New Call	Module 1 Load Name	
76	Other Pickup	Module 2	< None >
77	Quality Reporting Tool	Module 2 Load Name	
78	Redial	External Data Locations Information (Leave blank to use default)	
79	Remove Last Participant	Information	
80	Transfer	Directory	
81	Privacy	Messages	
82	None	Services	
		Authentication Server	
		Proxy Server	



Idle Timer (seconds)	
Secure Authentication URL	
Secure Directory URL	
Secure Idle URL	
Secure Information URL	
Secure Messages URL	
Secure Services URL	

Extension Information

☐ Enable Extension Mobility

Log Out Profile

Log in Time

Log out Time

MLPP Information

MLPP Domain

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*

Param

Override Common Settings



PC Port *	Enabled	
Settings Access*	Enabled	☐
Gratuitous ARP*	Disabled	
PC Voice VLAN Access*	Enabled	
Auto Line Select*	Disabled	
Web Access*	Disabled	☐
Days Display Not Active	Sunday Monday Tuesday	☐
Display On Time	07:30	☐
Display On Duration	10:30	☐
Display Idle Timeout	01:00	☐
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		
Display On When Incoming Call*	Disabled	
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 Headset*	Enabled	
Peer Firmware Sharing*	Enabled	☐
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	
Wireless Headset Hookswitch Control*	Disabled	
IPv6 Load Server		☐
IPv6 Log Server		
802.1x Authentication*	User Controlled	☐
Detect Unified CM Connection Failure*	Normal	
Minimum Ring Volume*	0-Silent	
HTTPS Server*	http and https Enabled	
Handset/Headset Monitor*	Disabled	
Enbloc Dialing*	Enabled	
Switch Port Remote Configuration*	Disabled	☐
PC Port Remote Configuration*	Disabled	☐
Automatic Port Synchronization*	Disabled	☐



Configuration of MGCP FAX Gateway

Navigation Path: Device → Gateway

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration

ccmadministrator | Search Documentation | About | Log out

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

Gateway Configuration | Related Links: Back To Find/List

Save | Delete | Reset | Apply Config | Add New

Status
Status: Ready

Gateway Details
Product: Cisco 3825
Gateway: 3825DSPfarm.pbxmlab.org
Protocol: MGCP
⚠ Device is not trusted
Domain Name*: 3825DSPfarm.pbxmlab.org
Description: 3825 on 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: NM-HDV2-2PORT-T1
Subunit 0: <None> Begin Port 0
Subunit 1: <None> Begin Port 0
Module in Slot 2: NM-HDV
Subunit 0: <None>

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*: Disable
Cisco Fax Relay*: Disable
T38 Fax Relay*: Disable
RTP Package Capability*: Enable
MT Package Capability*: Enable
RES Package Capability*: Enable
PRE Package Capability*: Enable
SST Package Capability*: Enable
RTP Unreachable OnOff*: Enable
RTP Unreachable timeout (ms)*: 1000
RTCP Report Interval (secs)*: 0
Simple SDP*: Disable



Configuration of MGCP FAX Gateway Analog Endpoint

Navigation Path: Device → Gateway

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration

ccmadadministrator

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Gateway Configuration

Related Links: [Back to MGCP Configuration](#)

Save Delete Reset Apply Config Add New

Status
 Status: Ready

Directory Number Information
[7915 Line \[1\] - 5014 in phones](#)
[7915](#)

Device Information
Product Cisco MGCP FXS Port
Gateway 3825DSPfarm.pbxmlab.org
Device Protocol Analog Access
 Device is not trusted
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 rp_phones
AAR Calling Search Space rp_phones
Location* Hub_None
AAR Group < None >
Network Locale < None >
Use Trusted Relay Point* Default
Geolocation < None >
☐ Transmit UTF-8 for Calling Party Name
Calling Party Transformation CSS < None >
☒ Use Device Pool Calling Party Transformation CSS
☐ Hot line Device

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
Prefix DN
☐ Unattended Port

Product Specific Configuration Layout

Hookflash Timer (50-1550ms)* 50
Guard-Out Timer (300-3000ms)* 1000
Inter-digit Duration Timer (50-500 ms)* 100
Input Gain (-6..14 db)* 0
Output Attenuation (-6..14 db)* 0
Echo Cancellation Enable* Enable
Echo Cancellation Coverage (ms)* 64
Ring Number* Default
Impedance* Default GW config



Configuring Cisco UBE

CUBE-ASR1K_Vz_151#**show version**

Cisco IOS Software, IOS-XE Software (PPC_LINUX_IOSD-ADVENTERPRISEK9-M), Version 15.1(2)S1, RELEASE SOFTWARE (fc2)
Technical Support: <http://www.cisco.com/techsupport>
Copyright (c) 1986-2011 by Cisco Systems, Inc.
Compiled Wed 01-Jun-11 04:09 by mcpre

ROM: IOS-XE ROMMON

CUBE-ASR1K_Vz_151 uptime is 4 weeks, 3 days, 3 hours, 29 minutes
Uptime for this control processor is 4 weeks, 3 days, 3 hours, 32 minutes
System returned to ROM by reload
System image file is "bootflash:asr1000rp1-adventerprisek9.03.03.01.S.151-2.S1.bin"
Last reload reason: Reload Command

cisco ASR1002 (2RU) processor with 1703423K/6147K bytes of memory.
4 Gigabit Ethernet interfaces
32768K bytes of non-volatile configuration memory.
4194304K bytes of physical memory.
7798783K bytes of eUSB flash at bootflash:.

Configuration register is 0x2102

CUBE-ASR1K_Vz_151#**sho run**

Building configuration...

Current configuration : 5705 bytes
!
! Last configuration change at 12:52:18 UTC Mon Jul 11 2011
!
version 15.1
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname CUBE-ASR1K_Vz_151
!
boot-start-marker
boot system bootflash:asr1000rp1-adventerprisek9.03.03.01.S.151-2.S1.bin
boot-end-marker
!
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
logging buffered 2000000



```
logging console errors
enable password cisco
!
no aaa new-model
!
ipc zone default
association 1
no shutdown
!
ip source-route
!
!
ip domain name cicolab.globalipcom.com
ip name-server 172.30.218.36
!
!
multilink bundle-name authenticated
!
!
voice service voip
allow-connections sip to sip
no supplementary-service sip refer
redirect ip2ip
sip
header-passing error-passthru
asserted-id pai1
localhost dns:gw1.cicolab.globalipcom.com
no update-callerid
early-offer forced
midcall-signaling passthru
privacy-policy passthru2
g729 annexb-all
!
!
voice translation-rule 13
rule 1 /81/ /1\1/
!
!
voice translation-profile Outgoing-Fax-G7114
translate called 1
!
redundancy
mode none
!
```

¹ 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)

² 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)

³ This translation rule is used on dial-peer 9003 (outgoing G.711-only calls) to strip the prefix “8” (sent by CUCM/Siemens PBX to match a different dial-peer whenever G.711-only calls are placed) from the telephone number

⁴ 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 9003 in this configuration example)



```
interface GigabitEthernet0/0/0
ip address 172.20.110.151 255.255.255.0
negotiation auto
!
interface GigabitEthernet0/0/1
no ip address
standby delay minimum 30 reload 60
standby version 2
standby 1 priority 50
standby 1 track 1 decrement 10
shutdown
negotiation auto
bfd interval 500 min_rx 500 multiplier 3
!
interface GigabitEthernet0/0/2
no ip address
shutdown
negotiation auto
!
interface GigabitEthernet0/0/3
no ip address
shutdown
negotiation auto
!
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
shutdown
negotiation auto
!
ip default-gateway 172.20.110.1
ip forward-protocol nd
!
no ip http server
no ip http secure-server
ip route 172.20.0.0 255.255.0.0 172.20.110.1
ip route 172.30.218.0 255.255.255.0 172.20.110.150
!
logging esm config
!
tftp-server flash:
!
control-plane
!
dial-peer voice 9000 voip
description To Service Provider network – SP facing
destination-pattern .....
session protocol sipv2
session target sip-server
session transport udp
voice-class sip asserted-id pai
dtmf-relay rtp-nte
fax protocol pass-through g711ulaw
!
dial-peer voice 9900 voip
description To Service Provider network – SME facing
session protocol sipv2
```




```
session transport udp
incoming called-number .....
voice-class sip asserted-id pai
dtmf-relay rtp-nte
fax protocol pass-through g711ulaw
!
dial-peer voice 9001 voip
description Incoming DID's to SME – SME facing
destination-pattern 408933....
session protocol sipv2
session target ipv4:172.20.236.252
session transport tcp
dtmf-relay rtp-nte
fax protocol pass-through g711ulaw
!
dial-peer voice 9901 voip
description Incoming DID's to SME – SP facing
session protocol sipv2
session target sip-server
session transport tcp
incoming called-number 408933....
dtmf-relay rtp-nte
fax protocol pass-through g711ulaw
!
dial-peer voice 9004 voip
description To Service Provider International calls – SP facing
destination-pattern 011T
session protocol sipv2
session target sip-server
session transport udp
voice-class sip early-offer forced
dtmf-relay rtp-nte
fax protocol pass-through g711ulaw
!
dial-peer voice 9904 voip
description To Service Provider International calls – SME facing
session protocol sipv2
session target sip-server
session transport udp
incoming called-number 011T
dtmf-relay rtp-nte
fax protocol pass-through g711ulaw
!
dial-peer voice 408 voip
description To Service Provider using 7-digit calling – SP facing
destination-pattern .....
session protocol sipv2
session target sip-server
session transport udp
voice-class sip asserted-id pai
dtmf-relay rtp-nte
fax protocol pass-through g711ulaw
!
dial-peer voice 4408 voip
description To Service Provider using 7-digit dialing – SME facing
session protocol sipv2
session transport udp
```



```
voice-class sip asserted-id pai
incoming called-number .....
dtmf-relay rtp-nte
fax protocol pass-through g711ulaw
!
dial-peer voice 9003 voip5
description Fax calls using G.711 – SP facing
translation-profile outgoing Outgoing-Fax-G711
destination-pattern 81.....
session protocol sipv2
session target sip-server
voice-class sip early-offer forced
dtmf-relay rtp-nte
codec g711ulaw
fax protocol pass-through g711ulaw
!
dial-peer voice 9903 voip
description Fax calls using G.711 – SME facing
session protocol sipv2
incoming called-number 81.....
voice-class sip early-offer forced
dtmf-relay rtp-nte
codec g711ulaw
fax protocol pass-through g711ulaw
!
sip-ua
set pstn-cause 1 sip-status 503
set pstn-cause 102 sip-status 503
retry invite 2
retry bye 2
retry cancel 2
timers trying 1000
sip-server ipv4:xxx.xxx.xxx.xxx:xxxx6
g729-annexb override
!
!
line con 0
password cisco
login
stopbits 1
line aux 0
stopbits 1
line vty 0 4
password cisco
login
!
end
```

⁵ 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

⁶ This parameter defines the IP address of the Service Provider’s SIP Proxy



Acronyms

Acronym	Definitions
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
SIP	Session Initiation Protocol
SP	Service Provider
SME	Session Manager Edition

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Corporate Headquarters

Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-4000
800 553-NETS (6387)
Fax: 408 526-4100

European Headquarters

Cisco Systems International
BV
HArlerbergpark
HArlerbergweg 13-19
1101 CH Amsterdam
The Netherlands
www-europe.cisco.com
Tel: 31 0 20 357 1000
Fax: 31 0 20 357 1100

Americas Headquarters

Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-7660
Fax: 408 527-0883

Asia Pacific Headquarters

Cisco Systems, Inc.
Capital Tower
168 Robinson Road
#22-01 to #29-01
Singapore 068912
www.cisco.com
Tel: +65 317 7777
Fax: +65 317 7799

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