



# Siemens HiPath 4000 Release 5.0 using SIP trunk to Cisco Unified Communications Manager 8.5

18<sup>th</sup> February, 2011, Initial Version

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## Introduction

- This application notes describes the necessary steps and configurations for connectivity between Siemens HiPath 4000 Release 5.0 and Cisco Unified Communications Manager (Cisco UCM) 8.5 over SIP trunk.
- The network topology diagram (Figure 1) shows the test setup for end-to-end interoperability between Cisco Unified Communications Manager 8.5 connected to Siemens HiPath 4000 Release 5.0 using SIP trunks via IP connectivity. Features tested are basic calls, 3-way (ad-hoc) conference, call transfer (attended and unattended), call forward (all, busy and no answer), hold/resume, DTMF interworking and MWI on/off.
- During testing, a Cisco 3845 voice gateway was used as the DSP farm point (MTP, Conference bridge), however other Cisco voice gateways can be used and the decision to choose what Cisco gateway model to use 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.

[Cisco 2800 Series Integrated Services Routers](#)

[Cisco 3800 Series Integrated Services Routers](#)

[Cisco AS5350XM Universal Gateway](#)

[Cisco AS5400XM Universal Gateway](#)

**Network Topology**

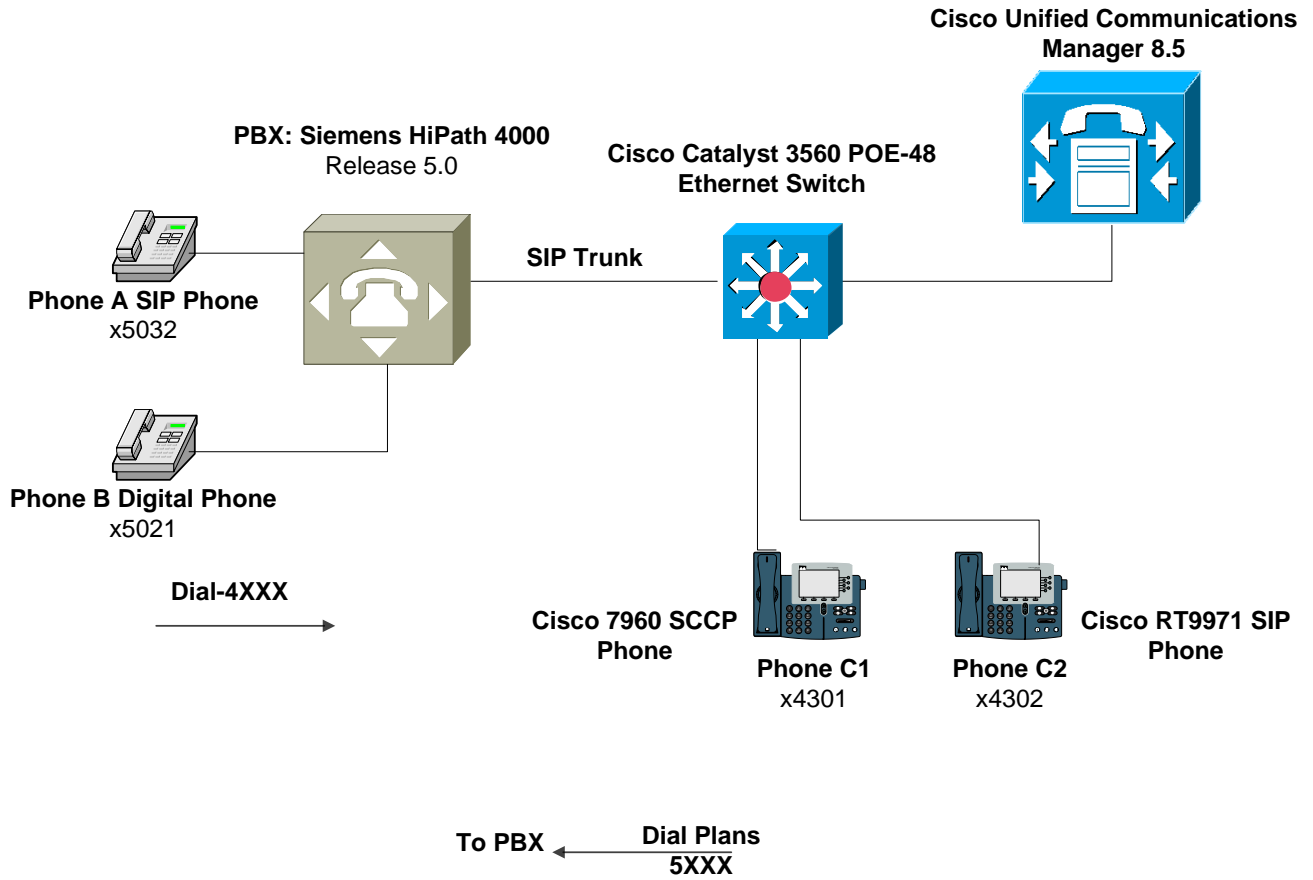


Figure 1. Basic Call Setup



## **System Components**

### **Hardware Requirements**

The following hardware is tested:

- Cisco MCS 7825H3 Unified Communications Manager Appliance
- Cisco Unified IP Phone 7960 (SCCP)
- Cisco Round Table Phone 9971 (SIP)
- Siemens HiPath 4000
- Siemens Optipoint 420 (SIP Phone)
- Siemens Optiset E advance plus (Digital Phone)

### **Software Requirements**

The following software is tested:

- Cisco Unified Communications Manager Release 8.5.1.10000-26
- Siemens HiPath 4000 Release 5.0



## Features

This section lists supported and unsupported features.

### Features Supported

- Basic calls
- CLIP- Calling line(number) identification presentation
- CLIR-Calling line (Number) identification restriction
- CNIR-Calling name identification restriction
- COLP-Connected line (Number) identification presentation
- COLR- Connected line (Number) identification restriction
- CONR- Connected name identification restriction
- Alerting name (See Limitations section for details.)
- Consultation transfer – Local and Network/External (See Limitations section for details.)
- Early Attended transfer – Local and Network/External (See Limitations section for details.)
- Call forward Local – Unconditional, Busy and No reply (See Limitations section for details.)
- Call forward Network/External – Unconditional, Busy and No reply (See Limitations section for details.)
- Hold and resume
- 3-way conference (ad-hoc) (See Limitations section for details)
- DTMF interworking

### Features Not Supported

- CNIP-Calling name identification presentation
- CONP-Connected name identification presentation
- Centralized Message center voicemail integration



## Limitations

These are the known limitations, caveats, or integration issues.

- Call direction: From Siemens to Cisco UCM. Siemens does not update phone with the PAI information sent by Cisco UCM in 180 during alerting, but forwards PAI information it received in 200 OK and the Siemens end point is updated after connect.
- Call direction: From Cisco UCM to Siemens. Siemens does not send PAI information during alerting, but sends PAI information only in 200 OK and so the Cisco UCM endpoint is updated only after the call is established.
- When a call from Siemens to Cisco UCM is transferred locally, Cisco UCM sends a request UPDATE to Siemens with the information of new name and number. Siemens ignores this request and so the end point does not update its display.
- Siemens SIP phones do not support “Early attended all transfers”.
- Call flow: From Cisco UCM to Siemens PhoneA Early attended transfer to PhoneB. No ringback on Originating Phone during transfer. Siemens SIP trunk does not forward 180 ringing sent by Siemens Phone to Cisco UCM.
- Siemens phones do not support blind transfers. (Siemens Optiset E advance plus and Siemens Optipoint 420)
- When a call is originated from Cisco Unified Communications Manager to Siemens and Siemens transfers it locally, Siemens do not send information about the transferred to extension to CUCM.
- Call direction: Siemens to Cisco UCM, Early attended transfer back to Siemens. Upon Early attended call transfer by Cisco UCM, call is initially completed without voice path and after 5 seconds call is disconnected. The workaround for this limitation is to configure “SIP Rel1XX options : send PRACK if 1XX contains SDP” in SIP profile on Cisco UCM. (see configuration section for details)
- Siemens does not include diversion-header or history-info for call flows including “Call forward unconditional”, “Call forward Busy” and “Call forward No Reply”.
- RT Phone 9971 does not support display of forwarding information.
- Siemens SIP Phone (Optipoint 420) does not support display of forwarding information.
- When a user on the Cisco Unified Communications Manager side invokes call hold feature, MOH was not heard on the Siemens PBX phone. The workaround to get MOH streaming is to set CUCM service parameter **Clusterwide Parameters (Service)** -> **Duplex Streaming Enabled** to True.
- Siemens SIP Phone (Optipoint 420) require conferencing server to support 3 way conference. (Configuring External conferencing server is not part of this document).
- Centralized voicemail with Unity Connection is not possible since Siemens PBX does not support Diversion-header or History-info messages.



## Configuration

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

### Configuring the Siemens HiPath 4000 Release 5.0

WABE

```
dis-wabe:type=gen;
DIS-WABE:TYPE=GEN;
H500: AMO WABE STARTED
```

DIGIT INTERPRETATION		VALID FOR ALL DIAL PLANS			
CODE	CALL PROGRESS STATE	NODE/DIGIT	RESERVED/CONVERT		
	1 1111 1112 22	ANALYSIS	DNI/ADD-INFO		
	0 12345 67890 12345 67890 12	RESULT	*=OWN NODE		
101	* . . . . .	NETRTE			
102	* . . . . .	NETRTE	R		
103	* . . . . .	NETRTE			
150	. . . . .	TIE			
199901 - 199902	. . . . .	TIE			
201 - 202	. . . . .	TIE			
26 - 27	. . . . .	TIE	R		
4300 - 4310	. . . . .	STN			
			DESTNO 103		
			DNNO 0- 0-103		
			PDNNO 0- 0-203		
5020 - 5029	. . . . .	STN			
			DESTNO 0		
			DNNO 0- 0-200*		
5031 - 5056	. . . . .	STN			
			DESTNO 0		
			DNNO 0- 0-200*		

DIGIT INTERPRETATION		VALID FOR ALL DIAL PLANS			
CODE	CALL PROGRESS STATE	NODE/DIGIT	RESERVED/CONVERT		
	1 1111 1112 22	ANALYSIS	DNI/ADD-INFO		
	0 12345 67890 12345 67890 12	RESULT	*=OWN NODE		
5058 - 5059	. . . . .	STN			
			DESTNO 0		
			DNNO 0- 0-200*		
5067	. . . . .	STN			
			DESTNO 0		
			DNNO 0- 0-200*		
5200 - 5219	. . . . .	STN			
			DESTNO 0		
			DNNO 0- 0-200*		
5297	. . . . .	STN			
			DESTNO 0		
			DNNO 0- 0-200*		
9	. . . . .	TIE			
*0	* . . . *	ACDWORK			
*2	* . . . *	CDRACC			
*3	. . . . .	PUDIR			
*4	* . . . *	CONF			

DIGIT INTERPRETATION		VALID FOR ALL DIAL PLANS			
CODE	CALL PROGRESS STATE	NODE/DIGIT	RESERVED/CONVERT		
	1 1111 1112 22	ANALYSIS	DNI/ADD-INFO		
	0 12345 67890 12345 67890 12	RESULT	*=OWN NODE		







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
#3	.	***	*	..**	.....	..*	SPDI	
#4	.	***	*	..**	.....	..*	SNR	
#5	.	....	*	.....	.....	..	ADND	
#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

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
#*1	.	*****	..**	.....	.....	..	MWACT	
#*2	.	*..*	*	**	.....	..	BUZZ	
#*329	.	***	*	****	**	.....	FAX	
#*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	.	....	*	.....	.....	..*	SPDI PROG	
##4	.	*..*	*	**	.....	..*	LNR	

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
##5	.	....	*	.....	.....	..	DDND	
##7	.	*..*	*	**	.....	..*	KNOVR	
##90	.	....	*	.....	.....	..*	DSYSFWD	
##91	.	....	*	.....	.....	..	DFWDVCE	
##92	.	....	*	.....	.....	..*	DFWDEXT	
##93	.	....	*	.....	.....	..*	DFWDINT	
##99	.	****	*	*****	**	.....	DFWDREM	
								CFREMVAR CFU
								CFREMSE VOICE
###1	.	*..*	*	**	.....	..*	TRACE	
###20	.	....	*	**	.....	..*	MILLWAT	
###21	.	....	*	**	.....	..*	LOOPBACK	
###22	.	....	*	**	.....	..*	SILENCE	
###23	.	....	*	**	.....	..*	COMBO	



| ###6 | . ....\* ..... .. | MONTONE |

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

Trunk Group Access Code, BUEND

dis-buend;
DIS-BUEND;
H500: AMO BUEND STARTED

Table with 3 columns: NO., NAME, CHARCON. Rows include 1 1-PRI (NEUTRAL), 26 PRI ECMA 1 (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;
DIS-TDCSU:1-1-1-0;
H500: AMO TDCSU STARTED

Table with 4 columns: DEV, PEN, TGRP, and various configuration parameters like PROTVAR, COTNO, ITR, LCOSD, SEGMENT, FACILITY, TRTBL, CBMATTR, SUPPRESS, ISDNIP, PNPL2P, TRACOUNT, ALARMNO, ZONE, DOMTYPE, INIGHT, UUSCCX, CLASSMRK, TCCID, BCNEG, LWPP, LWR1, DMCALLWD, SVCDOM, BCHAN, INS, COPNO, COS, CCT, DEDSCC, DITIDX, SIDANI, NWMUXTIM, DGTPR, ISDNNP, PNPL1P, SATCOUNT, FIDX, COTX, DOMAINNO, UUSCCY, BCGR, LWLT, LWR2, VNNO, SRCHMODE, DPLN, LCOSV, DESTNO, DEDSVC, SRTIDX, ATNTYP, TCHARG, CHIMAP, NNO, CARRIER, FWDX, TPROFNO, CCHDL, FNIDX, SRCGRP, SECLEVEL, LWPAR, LWPS.



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

AMO-TDCSU-111 DIGITAL TRUNKS  
DISPLAY COMPLETED;

### Class of Trunk, COT

DISPLAY-COT:COTNO=59;  
DISPLAY-COT:COTNO=59;  
H500: AMO COT STARTED

COT: 59	INFO: SIP TRUNK	
DEVICE: INDEP	SOURCE: DB	
PARAMETER:		
TRUNK SIGNALING ANSWER		ANS
CALL EXTEND FOR BUSY, RING OR CALL STATE		CEBC
DON'T RELEASE CALL TO BUSY HUNT GROUP		BSHT
END-OF-DIAL FOR BLOCK IS SET		BLOC
SEND NO NODE NUMBER TO PARTNER		LWNC
INCOMING CIRCUIT FROM SYSTEM WITHOUT LCR		NLCR
TSC-SIGNALING FOR NETWORKWIDE FEATURES (MANDATORY)		TSCS
USE DEFAULT NODE NUMBER OF LINE		DFNN
INCOMING CIRCUIT FROM SYSTEM WITHOUT LCR (DATA)		NLRD
LINE WITH IMPLICIT NUMBERS FOR CARRIER		LINC
NO FLAG TRACE		NOFT
NO TONE		NTON

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

### Class of Parameter for Device Handler, COP

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

COP: 59	INFO: SIP TRUNK	
DEVICE: INDEP	SOURCE: DB	
PARAMETER:		
REGISTRATION OF LAYER 3 ADVISORIES		L3AR
CO TRUNK ACCESS:		
TRUNK ACCESS		TA
TOLL ACCESS:		
TRUNK ACCESS		TA

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

### Class of Service, COSSU

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

COS	VOICE	FAX	DTE
-----	-------	-----	-----





```

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

```

AMO-RICHT-111 TRUNK ROUTING  
 DISPLAY COMPLETED;

**LODR**

DISPLAY-LODR:ODR=103;  
 DISPLAY-LODR:ODR=103;  
 H500: AMO LODR STARTED

```

+-----+
| ODR POSITION CMD PARAMETER |
+-----+
| 103 * | 1 NPI ISDN |
| | 2 TON UNKNOWN |
| | 3 ECHOALL |
| | 4 END |
+-----+
| * = READY FOR USE BY AMO LDAT |
+-----+
|INFO:SIP TRUNK|
+-----+

```

H03: THE NEXT FREE ODR IS 3

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



LDAT

DISPLAY-LDAT:TYPE=NWLCR,LROUTE=103;
DISPLAY-LDAT:TYPE=NWLCR,LROUTE=103;
H500: AMO LDAT STARTED

Table with columns: LROUTE, NAME, SERVICE, TYPE, DNNO OF ROUTE, SERVICE INFO, LRTEL, LVAL, TGRP, ODR, LAUTH, SCHEDULE, CARRIER, ZONE, LATTR, LDSRT, COTIDX. Includes values for LROUTE=103, NAME=SIP TRUNK, SERVICE=ALL, DNNO=103, and various schedule and carrier settings.

AMO-LDAT -111 LCR-DIRECTIONS
DISPLAY COMPLETED;

In-Band DTMF signaling:

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

MOH configuration:

Make sure the values of MOH and Hold is 22.

<DISPLAY-ZAND:TYPE=TONES;
DISPLAY-ZAND:TYPE=TONES;
H500: AMO ZAND STARTED
CP-TONETABLE

Table with columns: CP, SIU. Lists various tones and their durations, including INTDTN (1), EXTDTN (1), SPEC (3), RRGBK (8), FRRGBK (8), BUSY (5), OVRTN (10), CAMP (13), DATA (4), HOLDLINE (0), NOTREACH (9), NOPO (9), NOTALWD (14), IMPOSSFM (9), MOH (22), NACK (9), PACK (1), VOICEMAL (0), MWI (3), CONG (5), NU (9), OFFHKTRQ (7), RTCHKNTF (1).



```
LCRET | 10
RECDIALT | 1
CAMPRBK | 13
BRKMSGWT | 0
CONFTRN | 10
TEST | 15
EXTDTN2 | 1
BONHK | 5
NODIAL | 5
HOLD | 22
ALERT | 0
CBKALERT | 10
OVR | 0
OVRWARN | 0
REORDER | 5
LOCANN | 0
KNCONF | 0
DND | 3
PACKWD | 1
ZIPTONE | 2
ANNICPT | 0
RTOTON1 | 10
RTOTON2 | 5
FCAMP | 7
DISCTONE | 9
CMALSTONE | 1
HOLDEXT | 0
HOLDINT | 0
HOLDHUNT | 8
HOLDCCS | 0
HOLDH1 | 0
HOLDH2 | 0
HUNTGRUP | 1
POSTDIAL | 1
CCMABSEN | 0
HOWLTONE | 0
ROUTTONE | 0
REQDIALT | 1
SECDIALT | 0
CONFDISC | 0
```

```
AMO-ZAND -111 SYSTEM DATA
DISPLAY COMPLETED;
```



## SIP trunk configuration

```
dis-gkreg;
```

```
DIS-GKREG;
```

```
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
+-----+
| GWNO      2          GWATTR INTGW  REGGW  HG3550V2 SIPQ
| GWIPADDR  17 .20 .188.14      GWDIRNO 199902
| DIPLNUM   0          DPLN 0
| LAUTH     1
| GATEWAY REGISTERED: YES
| IP GATEWAY IS CONFIGURED BY GKREG
| INFO:     2-SIP Q
| SECLEVEL: TRADITIO
+-----+
| GWNO      3          GWATTR EXTGW  HG3550V2 SIP
| GWIPADDR  172.20 .109.254      GWDIRNO 5060
| DIPLNUM   0          DPLN 0
| LAUTH     1
| GATEWAY REGISTERED: NO
| IP GATEWAY IS CONFIGURED BY GKREG
| INFO:
| SECLEVEL: TRADITIO
+-----+
```

```
AMO-GKREG-111      GATEKEEPER REGISTRY
DISPLAY COMPLETED;
```





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- [Payload](#)
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Network Interfaces

- LAN1 (LAN1)
- LAN2 (Redundancy for LAN1)

### LAN1/Customer LAN

Interface Name:	LAN1
Interface Is Active:	Yes
IP Address:	172.20.188.13
Subnet Mask:	255.255.255.0
MAC Address :	00:1a:e8:36:e2:d3
Ethernet Link Mode:	100FDX
Max. Data Packet Size (bytes):	1500
IEEE802.1p/q Tagging:	No

				SSL on	HP4K-DEVEL	hg3500	01/18/2011 17:03:02
				1-1-1	HG 3500 V5		5d 4h 20m



- Explorers**
- [Basic Settings](#)
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- Routing
  - IP Routing
    - Static Routes
      - Default Router**
      - DNS Server
      - Address Resolution Protocol
      - Routing Table
      - ICMP Request
    - PSTN
    - Dialing Parameters

### Default Router

Default Routing via: LAN  
IP Address of Default Router: 172.20.188.1

				SSL on	HP4K-DEVEL	hg3500	01/18/2011 17:04:42
				1-1-1	HG 3500 V5		5d 4h 21m



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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

<b>SIP User Agent</b>	
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
<b>SIP Server (Registrar / Redirect)</b>	
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
<b>RFC 3261 Timer Values</b>	
Transaction Timeout (msec):	32000
<b>SIP Transport Protocol</b>	
SIP via TCP:	Yes
SIP via UDP:	Yes
SIP via TLS:	Yes
<b>SIP Session Timer</b>	
RFC 4028 support:	Yes
Session Expires (sec):	1800

				SSL on	HP4K-DEVEL	hg3500	01/18/2011 17:05:32
				1-1-1	HG 3500 V5		5d 4h 22m



- 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 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
DNS-SRV Records	
Blocking time for unreachable destination(sec):	60
Outgoing Call Supervision	
MakeCallReq Timeout (sec):	3

				SSL on	HP4K-DEVEL	hg3500	01/18/2011 17:06:22
				1-1-1	HG 3500 V5		5d 4h 23m



- 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

### Codec Parameters

Codec	Priority	Voice Activity Detection	Frame Size
G.711 A-law	Priority 2	Off	30 msec
G.711 μ-law	Priority 1	Off	30 msec
G.723	not used	Off	30 msec
G.729	not used	Off	20 msec
G.729A	Priority 3	Off	20 msec
G.729B	not used	On	20 msec
G.729AB	Priority 7	On	60 msec

**T.38 Fax**

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

**Misc.**

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: 98  
 Redundant Transmission of RFC2833 Tones according to RFC2198: On

SSL on	HP4K-DEVEL	hg3500	01/28/2011 19:08:56
1-1-1	HG 3500 V5		15d 6h 25m

Notes:



- 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

				SSL on 1-1-1	HP4K-DEVEL HG 3500 V5	hg3500	01/18/2011 17:10:57 5d 4h 27m
--	--	--	--	-----------------	--------------------------	--------	----------------------------------

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

- Codec Parameters
- IP Networking Mode
- SIP Trunk Profile Parameter
- SIP Trunk Profiles
  - ⊕ AT&T FlexReach
  - ⊕ AT&T VoEVPN
  - ⊕ Belgacom
  - ⊕ Broadsoft
  - ⊕ Cisco UCM
  - ⊕ COLT
  - ⊕ DS-COM
  - ⊕ DS-COM\_Pilot
  - ⊕ Elisa
  - ⊕ Entel NGN
  - ⊕ HiPath MobileConnect
  - ⊕ MediatrixGateway
  - ⊕ NatTrkWithoutRegistration
  - ⊕ NatTrkWithRegistration
  - ⊕ NeoTel Austria
  - ⊕ T-Systems
  - ⊕ Verizon
  - ⊕ Vodafone
  - ⊕ Vodafone-CLIP-NoScreening
  - ⊕ Voiceflex

### SIP Trunk Profile

Provider Name: Cisco UCM Account/Authentication Required: No Domain Name: SIP Transport Protocol: UDP
<b>Registrar</b> Use Registrar: No IP Address / Host name: 172.20.188.13 Port: 5060 Reregistration Interval (sec): 120
<b>Proxy</b> IP Address / Host name: 172.20.109.254 Port: 5060
<b>Outbound Proxy</b> Use Outbound Proxy: No IP Address / Host name: 0.0.0.0 Port: 0
<b>Inbound Proxy</b> Use Inbound Proxy: No IP Address / Host name: 0.0.0.0 Port: 0



**Display Name of the station:**

```
DISPLAY-PERSI:TYPE=NAME,STNO=5032;
DISPLAY-PERSI:TYPE=NAME,STNO=5032;
H500: AMO PERSI STARTED
```

STNO	CHRISTIAN AND SURNAME	CHARCON	ORGANIZATIONAL UNIT
5032	HIPATH IP 3*		

```
AMO-PERSI-111          PERSONAL IDENTIFICATION DATA
DISPLAY COMPLETED;
```

**To change name "Change-Persi" should be used.**

**Name and Number Restrictions:**

```
DISPLAY-SBCSU:STNO=5032;
DISPLAY-SBCSU:STNO=5032;
H500: AMO SBCSU STARTED
```

```
----- USER DATA -----
STNO      =5032          OPT       =FPP          COS1      =5           DPLN      =0
MAINO     =5032          CONN     =SIP          COS2      =5           ITR       =0
PEN       =1- 1- 1- 3   LCOSV1   =15           COSX      =0
INS       =Y            ASYNCT   =           LCOSV2   =15
SSTNO    =Y            PERMACT  =N           LCOSD1    =15          CBKBMAX   =5
TRACE    =N            EXTBUS   =           LCOSD2    =15          RCBKB     =N
ALARMNO  =0            DFSVCANA =           SPDI      =30          RCBKNA    =N
HMUSIC   =1            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  =""
DTMFCTR  =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) -----
KN       VC          DMCALLWD MBCHL
-----
```

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





## Siemens HiPath 4000 Software Release

```
DISPLAY-DBC:VERBOSE=N;  
DISPLAY-DBC:VERBOSE=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;
```



## Configuring the Cisco Unified Communications Manager

### Cisco UCM Version

**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 ▾

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**Cisco Unified CM Administration**  
System version: 8.5.1.10000-26



Last Successful Logon: Jan 14, 2011 4:03:43 PM

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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.



## Configuring 7960 SCCP Phone (Page 1 of 4)

Navigation Path: Cisco UCM Administration → Device → Phone

**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

### Phone Configuration

Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

**Status**  
Status: Ready

#### Association Information

Modify Button Items

1	7960 Line [1] - 4301 (no partition)
2	7960 Line [2] - Add a new DN
3	Add a new SD
4	Add a new SD
5	Add a new SD
6	Add a new SD
----- Unassigned Associated Items -----	
7	Add a new SD
8	Add a new SURL
9	Add a new BLF SD
10	7960 Add a new BLF Directed Call Park
11	Privacy
12	None

#### Phone Type

Product Type: Cisco 7960  
Device Protocol: SCCP

#### Device Information

Registered with Cisco Unified Communications Manager CM-Telugu

Registration	Registered with Cisco Unified Communications Manager CM-Telugu
IP Address	172.20.109.41
Active Load ID	Unknown
<input checked="" type="checkbox"/> Device is Active	
<input checked="" type="checkbox"/> Device is trusted	
MAC Address*	00146A3C1BB9
Description	Phone Type:7960; Extn:4301
Device Pool*	Default <a href="#">View Details</a>
Common Device Configuration	< None > <a href="#">View Details</a>
Phone Button Template*	Standard 7960 SCCP
Softkey Template	Mobility User
Common Phone Profile*	Standard Common Phone Profile
Calling Search Space	< None >
AAR Calling Search Space	< None >
Media Resource Group List	< None >
User Hold MOH Audio Source	1-SampleAudioSource
Network Hold MOH Audio Source	1-SampleAudioSource
Location*	Hub_None
AAR Group	< None >
User Locale	English, United States
Network Locale	United States
Built In Bridge*	Default
Privacy*	Default
Device Mobility Mode*	Default <a href="#">View Current Device Mobility Settings</a>
Owner User ID	Wks03
Phone Load Name	
Join Across Lines	Default
Use Trusted Relay Point*	Default
BLF Audible Alert Setting (Phone Idle)*	Default
BLF Audible Alert Setting (Phone Busy)*	Default
Always Use Prime Line*	Default
Always Use Prime Line for Voice Message*	Default
Calling Party Transformation CSS	< None >
Geolocation	< None >

Use Device Pool Calling Party Transformation CSS  
 Retry Video Call as Audio  
 Ignore Presentation Indicators (internal calls only)  
 Allow Control of Device from CTI  
 Logged Into Hunt Group  
 Remote Device  
 Hot line Device\*\*\*\*\*



<b>Protocol Specific Information</b>	
Packet Capture Mode*	None
Packet Capture Duration	0
Presence Group*	Standard Presence group
Device Security Profile*	Cisco 7960 - Standard SCCP Non-Secure Profile
SUBSCRIBE Calling Search Space	< None >
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> Require DTMF Reception	
<input type="checkbox"/> RFC2833 Disabled	

<b>Certification Authority Proxy Function (CAPF) Information</b>	
Certificate Operation*	No Pending Operation
Authentication Mode*	By Null String
Authentication String	
<input type="button" value="Generate String"/>	
Key Size (Bits)*	1024
Operation Completes By	2011 1 29 12 (YYYY:MM:DD:HH)
Certificate Operation Status: None	
Note: Security Profile Contains Addition CAPF Settings.	

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

<b>External Data Locations Information (Leave blank to use default)</b>	
Information	
Directory	
Messages	
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	

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

<b>MLPP Information</b>	
MLPP Domain	< None >
MLPP Indication*	Default
MLPP Preemption*	Default

<b>Do Not Disturb</b>	
<input type="checkbox"/> Do Not Disturb	
DND Option*	Ringer Off
DND Incoming Call Alert	< None >

<b>Product Specific Configuration Layout</b>	
<input type="checkbox"/> Disable Speakerphone	
<input type="checkbox"/> Disable Speakerphone and Headset	
PC Port *	Enabled
Settings Access*	Enabled
Gratuitous ARP*	Enabled
PC Voice VLAN Access*	Enabled
Video Capabilities*	Disabled
Auto Line Select*	Disabled
Web Access*	Enabled



Configuring 7960 SCCP Phone (Page 3 of 4)

Navigation Path: Cisco UCM Administration → Device → Phone

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

Navigation: Cisco Unified CM Administration

CMAAdministrator | Search Documentation | About | Log out

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

**Directory Number Configuration** Related Links: Configure Device (SEP00146A3C1BB9)

Save Delete Reset Apply Config Add New

**Status**  
 Status: Ready

**Directory Number Information**

Directory Number\* 4301  
 Route Partition < None >  
 Description  
 Alerting Name CUCM-03[ALERT]  
 ASCII Alerting Name CUCM-03[ALERT]  
 Allow Control of Device from CTI  
 Associated Devices SEP00146A3C1BB9  
 Edit Device Edit Line Appearance  
 Dissociate Devices

**Directory Number Settings**

Voice Mail Profile < None > (Choose <None> to use system default)  
 Calling Search Space < None >  
 Presence Group\* Standard Presence group  
 User Hold MOH Audio Source 1-SampleAudioSource  
 Network Hold MOH Audio Source 1-SampleAudioSource  
 Auto Answer\* Auto Answer Off

**AAR Settings**

AAR  or  AAR Destination Mask < None > AAR Group < None >  
 Retain this destination in the call forwarding history

**Call Forward and Call Pickup Settings**

	Voice Mail	Destination	Calling Search Space
Calling Search Space Activation Policy			Use System Default
Forward All	<input type="checkbox"/> or		< None >
Secondary Calling Search Space for Forward All			< None >
Forward Busy Internal	<input type="checkbox"/> or		< None >
Forward Busy External	<input type="checkbox"/> or		< None >
Forward No Answer Internal	<input type="checkbox"/> or		< None >
Forward No Answer External	<input type="checkbox"/> or		< None >
Forward No Coverage Internal	<input type="checkbox"/> or		< None >
Forward No Coverage External	<input type="checkbox"/> or		< None >
Forward on CTI Failure	<input type="checkbox"/> or		< None >
Forward Unregistered Internal	<input type="checkbox"/> or		< None >
Forward Unregistered External	<input type="checkbox"/> or		< None >
No Answer Ring Duration (seconds)			
Call Pickup Group			< None >



Configuring 7960 SCCP Phone (Page 4 of 4)

Navigation Path: Cisco UCM Administration → Device → Phone

---

**— Park Monitoring**

	Voice Mail	Destination	Calling Search Space
Park Monitoring Forward No Retrieve Destination External	<input type="checkbox"/> or	<input type="text"/>	< None > A blank value means to call the parker's line.
Park Monitoring Forward No Retrieve Destination Internal	<input type="checkbox"/> or	<input type="text"/>	< None > A blank value means to call the parker's line.
Park Monitoring Reversion Timer	<input type="text" value="0"/>	A blank value will use value set in Park Monitoring Reversion Timer service parameter	

---

**— MLPP Alternate Party Settings**

Target (Destination)

MLPP Calling Search Space < None >

MLPP No Answer Ring Duration (seconds)

---

**— Line Settings for All Devices**

Hold Reversion Ring Duration (seconds)  Setting the Hold Reversion Ring Duration to zero will disable the feature

Hold Reversion Notification Interval (seconds)  Setting the Hold Reversion Notification Interval to zero will disable the feature

Party Entrance Tone\* Default

---

**— Line 1 on Device SEP00146A3C1BB9**

Display (Internal Caller ID) CUCM-03 Display text for a line appearance is intended for displaying text such as a name instead of a directory number for internal calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.

ASCII Display (Internal Caller ID) CUCM-03

Line Text Label CUCM-03

ASCII Line Text Label CUCM-03

External Phone Number Mask

Visual Message Waiting Indicator Policy\* Use System Policy

Ring Setting (Phone Idle)\* Use System Default

Ring Setting (Phone Active) Use System Default Applies to this line when any line on the phone has a call in progress.

Call Pickup Group Audio Alert Setting(Phone Idle) Use System Default

Call Pickup Group Audio Alert Setting(Phone Active) Use System Default

Monitoring Calling Search Space < None >

---

**— Multiple Call/Call Waiting Settings on Device SEP00146A3C1BB9**

Note:The range to select the Max Number of calls is: 1-200

Maximum Number of Calls\*

Busy Trigger\*  (Less than or equal to Max. Calls)

---

**— Forwarded Call Information Display on Device SEP00146A3C1BB9**

Caller Name

Caller Number

Redirected Number

Dialed Number

---

**— Users Associated with Line**

---



## Configuring RT9971 SIP Phone (Page 1 of 5)

Navigation Path: Cisco UCM Administration → Device → Phone

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

Navigation: Cisco Unified CM Administration | Go

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CCMAdministrator | Search Documentation | About | Logout

### Phone Configuration

Related Links: Back To Find/List | Go

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

**Status:** Ready

---

**Association Information**

Modify Button Items

1	Line [1] - 4302 (no partition)
2	Line [2] - Add a new DN
3	Mobility
4	Add a new SD
5	Add a new SD
6	Add a new SD
----- Add On Module(s) -----	
7	None
8	None
9	None
10	None
11	None
12	None
13	None
14	None
15	None
16	None
17	None
18	None
19	None
20	None
21	None
22	None
23	None
24	NORÄ
25	None
26	None
27	None
28	None
29	None
30	None
31	None
32	None
33	None
34	None
35	None
36	None
37	None
38	None
39	None
40	None
41	None
42	None
43	None
44	None
45	None
46	None
47	None
48	None
49	None
50	None
51	None
52	None
53	None
54	None
55	None

**Phone Type**  
Product Type: Cisco 9971  
Device Protocol: SIP

**Device Information**

Registration: Registered with Cisco Unified Communications Manager CM-Telugu  
IP Address: 172.20.109.38  
Active Load ID: sip9971-9-1-1SR1  
Inactive Load ID: sip9971-9-1-0-12  
Download Status: Unknown

Device is Active  
 Device is trusted  
MAC Address\*: 1C17D337D3FD

Description: Phone Type: 9971, Extn: 4302

Device Pool\*: Default [View Details](#)

Common Device Configuration: < None > [View Details](#)

Phone Button Template\*: SEP1C17D337D3FD-SIP-Individual Template

Common Phone Profile\*: Standard Common Phone Profile

Calling Search Space: < None >

AAR Calling Search Space: < None >

Media Resource Group List: < None >

User Hold MOH Audio Source: 1-SampleAudioSource

Network Hold MOH Audio Source: 1-SampleAudioSource

Location\*: Hub\_None

AAR Group: < None >

User Locale: English, United States

Network Locale: United States

Built In Bridge\*: Default

Privacy\*: Default

Device Mobility Mode\*: Default [View Current Device Mobility Settings](#)

Owner User ID: Wks02

Phone Personalization\*: Default

Services Provisioning\*: Default

Phone Load Name:

Use Trusted Relay Point\*: Default

BLF Audible Alert Setting (Phone Idle)\*: Default

BLF Audible Alert Setting (Phone Busy)\*: Default

Always Use Prime Line\*: Default

Always Use Prime Line for Voice Message\*: Default

Calling Party Transformation CSS: < None >

Geolocation: < None >

Feature Control Policy: < None >

Use Device Pool Calling Party Transformation CSS  
 Ignore Presentation Indicators (internal calls only)  
 Allow Control of Device from CTI  
 Logged Into Hunt Group  
 Remote Device  
 Protected Device\*\*\*\*

**Protocol Specific Information**

Packet Capture Mode\*: None

Packet Capture Duration: 0

Presence Group\*: Standard Presence group

SIP Dial Rules: < None >

MTP Preferred Originating Codec\*: 711ulaw

Device Security Profile\*: Cisco 9971 - Standard SIP Non-Secure Profile

Rerouting Calling Search Space: < None >



Configuring RT9971 SIP Phone (Page 2 of 5)

Navigation Path: Cisco UCM Administration → Device → Phone

56	None	SUBSCRIBE Calling Search Space	< None >
57	None	SIP Profile*	Standard SIP Profile
58	None	Digest User	< None >
59	None	<input type="checkbox"/> Media Termination Point Required	
60	None	<input type="checkbox"/> Unattended Port	
61	None	<input type="checkbox"/> Require DTMF Reception	
62	None	<b>Certification Authority Proxy Function (CAPF) Information</b>	
63	None	Certificate Operation*	No Pending Operation
64	None	Authentication Mode*	By Null String
65	None	Authentication String	
66	None	<input type="button" value="Generate String"/>	
67	None	Key Size (Bits)*	1024
68	None	Operation Completes By	2011 1 29 12 (YYYY:MM:DD:HH)
69	None	Certificate Operation Status:	None
70	None	Note:	Security Profile Contains Addition CAPF Settings.
71	None	<b>Expansion Module Information</b>	
72	None	Module 1	< None >
73	None	Module 1 Load Name	
74	None	Module 2	< None >
75	None	Module 2 Load Name	
76	None	Module 3	< None >
77	None	Module 3 Load Name	
78	None	<b>External Data Locations Information (Leave blank to use default)</b>	
79	None	Information	
80	None	Directory	
81	None	Messages	
82	None	Services	
83	None	Authentication Server	
84	None	Proxy Server	
85	None	Idle	
86	None	Idle Timer (seconds)	
87	None	Secure Authentication URL	
88	None	Secure Directory URL	
89	None	Secure Idle URL	
90	None	Secure Information URL	
91	None	Secure Messages URL	
92	None	Secure Services URL	
93	None	<b>Extension Information</b>	
94	None	<input type="checkbox"/> Enable Extension Mobility	
95	None	Log Out Profile	-- Use Current Device Settings --
96	None	Log in Time	< None >
97	None	Log out Time	< None >
98	None	<b>MLPP Information</b>	
99	None	MLPP Domain	< None >
100	None	<b>Do Not Disturb</b>	
101	None	<input type="checkbox"/> Do Not Disturb	
102	None	DND Option*	Use Common Phone Profile Setting
103	None	DND Incoming Call Alert	< None >
104	None	<b>Secure Shell Information</b>	
105	None	Secure Shell User	
106	None	Secure Shell Password	
107	None	----- Unassigned Associated Items -----	
108	None	<a href="#">Add a new SD</a>	
109	None	All Calls	
110	None	<a href="#">Add a new BLF Directed Call Park</a>	
111	None		
112	None		
113	None		
114	None		
115	None		
116	None		
117	None		





Configuring RT9971 SIP Phone (Page 3 of 5)

Navigation Path: Cisco UCM Administration → Device → Phone

<ul style="list-style-type: none"> <li>118 Call Park</li> <li>119 Call Pickup</li> <li>120 CallBack</li> <li>121 Group Call Pickup</li> <li>122 Hunt Group Logout</li> <li>123 <a href="#">Intercom [1] - Add a new Intercom</a></li> <li>124 Malicious Call Identification</li> <li>125 Meet Me Conference</li> <li>126 Other Pickup</li> <li>127 Quality Reporting Tool</li> <li>128 Redial</li> <li>129 <a href="#">Add a new SURL</a></li> <li>130 <a href="#">Add a new BLF SD</a></li> <li>131 Answer Oldest</li> <li>132 Do Not Disturb</li> <li>133 Services</li> <li>134 Record</li> <li>135 Privacy</li> <li>136 None</li> </ul>	<p><b>Product Specific Configuration Layout</b></p> <p><input type="checkbox"/> Disable Speakerphone</p> <p><input type="checkbox"/> Disable Speakerphone and Headset</p> <p>PC Port * <span style="float: right;">Enabled</span></p> <p>Back USB Port* <span style="float: right;">Enabled</span></p> <p>Side USB Port* <span style="float: right;">Enabled</span></p> <p>Cisco Camera* <span style="float: right;">Disabled</span></p> <p>Video Capabilities* <span style="float: right;">Disabled</span></p> <p>Enable/Disable USB Classes <span style="float: right;">Mass Storage Human Interface Device Audio Class</span></p> <p>SDIO * <span style="float: right;">Disabled</span></p> <p>Bluetooth * <span style="float: right;">Enabled</span></p> <p>Wifi * <span style="float: right;">Enabled</span></p> <p>Bluetooth Profiles* <span style="float: right;">Handsfree Human Interface Device</span></p> <p>Settings Access* <span style="float: right;">Enabled</span></p> <p>Gratuitous ARP* <span style="float: right;">Disabled</span></p> <p>PC Voice VLAN Access* <span style="float: right;">Enabled</span></p> <p>Web Access* <span style="float: right;">Disabled</span></p> <p>Days Display Not Active <span style="float: right;">Sunday Monday Tuesday</span></p> <p>Display On Time <span style="float: right;">07:30</span></p> <p>Display On Duration <span style="float: right;">10:30</span></p> <p>Display Idle Timeout <span style="float: right;">01:00</span></p> <p>HTTPS Server* <span style="float: right;">http and https Enabled</span></p> <p>Span to PC Port* <span style="float: right;">Disabled</span></p> <p>Logging Display* <span style="float: right;">Disabled</span></p> <p>Load Server <span style="float: right;"></span></p> <p>Recording Tone* <span style="float: right;">Disabled</span></p> <p>Recording Tone Local Volume* <span style="float: right;">100</span></p> <p>Recording Tone Remote Volume* <span style="float: right;">50</span></p> <p>Recording Tone Duration <span style="float: right;"></span></p> <p>Display On When Incoming Call* <span style="float: right;">Enabled</span></p> <p>RTCP* <span style="float: right;">Disabled</span></p> <p>Log Server <span style="float: right;"></span></p> <p>Advertise G.722 Codec* <span style="float: right;">Use System Default</span></p> <p>Wideband Headset UI Control* <span style="float: right;">Enabled</span></p> <p>Wideband Headset* <span style="float: right;">Enabled</span></p> <p>Peer Firmware Sharing* <span style="float: right;">Enabled</span></p> <p>Cisco Discovery Protocol (CDP): Switch Port* <span style="float: right;">Enabled</span></p> <p>Cisco Discovery Protocol (CDP): PC Port* <span style="float: right;">Enabled</span></p> <p>Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port* <span style="float: right;">Enabled</span></p> <p>Link Layer Discovery Protocol (LLDP): PC Port* <span style="float: right;">Enabled</span></p> <p>LLDP Asset ID <span style="float: right;"></span></p> <p>LLDP Power Priority* <span style="float: right;">Unknown</span></p> <p>802.1x Authentication* <span style="float: right;">User Controlled</span></p> <p>Detect Unified CM Connection Failure* <span style="float: right;">Normal</span></p> <p>Switch Port Remote Configuration* <span style="float: right;">Disabled</span></p> <p>PC Port Remote Configuration* <span style="float: right;">Disabled</span></p> <p>Automatic Port Synchronization* <span style="float: right;">Disabled</span></p> <p>Power Negotiation* <span style="float: right;">Enabled</span></p> <p>Restrict Data Rates* <span style="float: right;">Disabled</span></p>	<p style="text-align: center;">?</p> <p style="text-align: center;">Param</p> <p style="text-align: right;">Override Common Settings</p>
--	--	--



Configuring RT9971 SIP Phone (Page 4 of 5)

Navigation Path: Cisco UCM Administration → Device → Phone

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

Navigation: Cisco Unified CM Administration

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

CCMAdministrator | Search Documentation | About | Logout

Directory Number Configuration Related Links: Configure Device (SEP1C17D337D3FD)

Save Delete Reset Apply Config Add New

**Status**  
Status: Ready

**Directory Number Information**

Directory Number\* 4302  
Route Partition < None >  
Description  
Alerting Name CUCM-02[ALERT]  
ASCII Alerting Name CUCM-02[ALERT]  
 Allow Control of Device from CTI  
Associated Devices SEP1C17D337D3FD  
Dissociate Devices

Edit Device Edit Line Appearance

**Directory Number Settings**

Voice Mail Profile < None > (Choose <None> to use system default)  
Calling Search Space < None >  
Presence Group\* Standard Presence group  
User Hold MOH Audio Source 1-SampleAudioSource  
Network Hold MOH Audio Source 1-SampleAudioSource  
Auto Answer\* Auto Answer Off

**AAR Settings**

AAR	Voice Mail	AAR Destination Mask	AAR Group
<input type="checkbox"/>	<input type="checkbox"/> or		< None >

Retain this destination in the call forwarding history

**Call Forward and Call Pickup Settings**

	Voice Mail	Destination	Calling Search Space
Calling Search Space Activation Policy			Use System Default
Forward All	<input type="checkbox"/> or		< None >
Secondary Calling Search Space for Forward All			< None >
Forward Busy Internal	<input type="checkbox"/> or		< None >
Forward Busy External	<input type="checkbox"/> or		< None >
Forward No Answer Internal	<input type="checkbox"/> or		< None >
Forward No Answer External	<input type="checkbox"/> or		< None >
Forward No Coverage Internal	<input type="checkbox"/> or		< None >
Forward No Coverage External	<input type="checkbox"/> or		< None >
Forward on CTI Failure	<input type="checkbox"/> or		< None >
Forward Unregistered Internal	<input type="checkbox"/> or		< None >
Forward Unregistered External	<input type="checkbox"/> or		< None >
No Answer Ring Duration (seconds)			
Call Pickup Group			< None >



Configuring RT9971 SIP Phone (Page 5 of 5)

Navigation Path: Cisco UCM Administration → Device → Phone

**Park Monitoring**

	Voice Mail	Destination	Calling Search Space
Park Monitoring Forward No Retrieve Destination External	<input type="checkbox"/> or	<input type="text"/>	< None > A blank value means to call the parker's line.
Park Monitoring Forward No Retrieve Destination Internal	<input type="checkbox"/> or	<input type="text"/>	< None > A blank value means to call the parker's line.
Park Monitoring Reversion Timer	<input type="text" value="0"/>	A blank value will use value set in Park Monitoring Reversion Timer service parameter	

**MLPP Alternate Party Settings**

Target (Destination)

MLPP Calling Search Space < None >

MLPP No Answer Ring Duration (seconds)

**Line Settings for All Devices**

Hold Reversion Ring Duration (seconds)  Setting the Hold Reversion Ring Duration to zero will disable the feature

Hold Reversion Notification Interval (seconds)  Setting the Hold Reversion Notification Interval to zero will disable the feature

Party Entrance Tone\* Default

**Line 1 on Device SEP1C17D337D3FD**

Display (Internal Caller ID) CUCM-02 Display text for a line appearance is intended for displaying text such as a name instead of a directory number for internal calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.

ASCII Display (Internal Caller ID) CUCM-02

Line Text Label CUCM-02

ASCII Line Text Label CUCM-02

External Phone Number Mask

Visual Message Waiting Indicator Policy\* Use System Policy

Audible Message Waiting Indicator Policy\* Default

Ring Setting (Phone Idle)\* Use System Default

Ring Setting (Phone Active) Use System Default Applies to this line when any line on the phone has a call in progress.

Call Pickup Group Audio Alert Setting(Phone Idle) Use System Default

Call Pickup Group Audio Alert Setting(Phone Active) Use System Default

Recording Option\* Call Recording Disabled

Recording Profile < None >

Monitoring Calling Search Space < None >

Log Missed Calls

**Multiple Call/Call Waiting Settings on Device SEP1C17D337D3FD**

Note:The range to select the Max Number of calls is: 1-200

Maximum Number of Calls\*

Busy Trigger\*  (Less than or equal to Max. Calls)

**Forwarded Call Information Display on Device SEP1C17D337D3FD**

Caller Name

Caller Number

Redirected Number

Dialed Number

**Users Associated with Line**



## Configuring Media Resource Group

Navigation Path: Cisco UCM Administration → Media Resources → Media Resource Group

**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 ▾

### Media Resource Group Configuration

Related Links:

---

**Status**  
 Status: Ready

---

**Media Resource Group Status**  
Media Resource Group: SoftwareMTP (used by 4 devices)

---

**Media Resource Group Information**

Name\*

Description

---

**Devices for this Group**

Available Media Resources\*\*

cfb00233335da20
mtp002333335da20

Selected Media Resources\*

ANN_2 (ANN)
CFB_2 (CFB)
MOH_2 (MOH)
MTP_2 (MTP)

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

---



## Configuring Media Resource Group List

Navigation Path: Cisco UCM Administration → Media Resources → Media Resource Group List

The screenshot shows the Cisco Unified CM Administration interface for configuring a Media Resource Group List. The page title is "Media Resource Group List Configuration". The navigation path is: System > Call Routing > Media Resources > Advanced Features > Device > Application > User Management > Bulk Administration > Help. The user is logged in as CCMAdministrator. The page includes a "Related Links" section with "Back To Find/List".

**Media Resource Group List Configuration**

Save Delete Copy Add New

**Status**  
Status: Ready

**Media Resource Group List Status**  
Media Resource Group List: SoftwareMGL (used by 3 devices)

**Media Resource Group List Information**  
Name\* SoftwareMGL

**Media Resource Groups for this List**  
Available Media Resource Groups: Conf\_MRG, Ext\_MRG, SIP\_TRUNK\_MRG  
Selected Media Resource Groups: SoftwareMTP

Save Delete Copy Add New

\*- indicates required item.



## Configuring Device Pool

Navigation Path: Cisco UCM Administration → System → Device Pool

Cisco Unified CM Administration
Navigation Cisco Unified CM Administration

System
CCMAdministrator | Search Documentation | About | Logout

**Device Pool Configuration** Related Links: [Back To Find/List](#)

Save Delete Copy Reset Apply Config Add New

**Status**

- Update successful
- Click on the Reset button to have the changes take effect.

**Device Pool Information**

Device Pool: Default (25 members)\*\*)

**Device Pool Settings**

Device Pool Name\*

Cisco Unified Communications Manager Group\*

Calling Search Space for Auto-registration

Adjunct CSS

Reverted Call Focus Priority

Local Route Group

Intercompany Media Services Enrolled Group

**Roaming Sensitive Settings**

Date/Time Group\*

Region\*

Media Resource Group List

Location

Network Locale

SRST Reference\*

Connection Monitor: Duration\*\*\*

Single Button Barge\*

Join Across Lines\*

Physical Location

Device Mobility Group

**Device Mobility Related Information\*\*\*\***

Device Mobility Calling Search Space

AAR Calling Search Space

AAR Group

Calling Party Transformation CSS

Called Party Transformation CSS

**Geolocation Configuration**

Geolocation

Geolocation Filter

**Call Routing Information**

**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.

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	<input type="text" value="Default"/>	<input type="text" value=""/>	<input type="text" value="&lt; None &gt;"/>
International Number	<input type="text" value="Default"/>	<input type="text" value=""/>	<input type="text" value="&lt; None &gt;"/>
Unknown Number	<input type="text" value="Default"/>	<input type="text" value=""/>	<input type="text" value="&lt; None &gt;"/>
Subscriber Number	<input type="text" value="Default"/>	<input type="text" value=""/>	<input type="text" value="&lt; None &gt;"/>

**Incoming Called 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.

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value="&lt; None &gt;"/>
International Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value="&lt; None &gt;"/>
Unknown Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value="&lt; None &gt;"/>
Subscriber Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value="&lt; None &gt;"/>

**Connected Party Settings**

Connected Party Transformation CSS



## Configuring Region

Navigation Path: Cisco UCM Administration → System → Region

**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 ▾

### Region Configuration

Related Links:

**Region Information**

Name\*

---

**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
<input type="text" value="Default"/> <input type="text" value="Region_SME_G711"/> <input type="text" value="Region_SME_G729"/>	<input type="button" value="Keep Current Setting"/>	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="text" value=""/> kbps	<input type="button" value="Keep Current Setting"/>

\*- indicates required item.



## Configuring SIP trunk to Siemens HiPath 4000 Rel 5.0 (Page 1 of 2)

Navigation Path: Cisco UCM Administration → Device → Trunk

**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

### Trunk Configuration

Related Links: Back To Find/List Go

Save Delete Reset Add New

---

**Status**  
Update successful

---

**Device Information**

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	SIEMENS_HIPATH_SIP_TRUNK
Description	SIP trunk to Siemens HiPath 4000 Rel 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

---

**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





Configuring SIP trunk to Siemens HiPath 4000 Rel 5.0 (Page 2 of 2)

Navigation Path: Cisco UCM Administration → Device → Trunk

**Inbound Calls**

Significant Digits\*

Connected Line ID Presentation\*

Connected Name Presentation\*

Calling Search Space

AAR Calling Search Space

Prefix DN

Redirecting Diversion Header Delivery - Inbound

**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.

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

**Connected Party Settings**

Connected Party Transformation CSS

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* <input type="text" value="172.20.188.13"/>	<input type="text"/>	<input type="text" value="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 <input type="text"/>	<input type="text"/>

**Geolocation Configuration**

Geolocation

Geolocation Filter

Send Geolocation Information



## Configuring SIP Trunk security profile

Navigation Path: Cisco UCM Administration → System → Security → SIP Trunk Security Profile

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

Navigation: Cisco Unified CM Administration

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### SIP Trunk Security Profile Configuration

Related Links:

---

**Status**  
 Status: Ready

---

**SIP Trunk Security Profile Information**

Name\*

Description

Device Security Mode

Incoming Transport Type\*

Outgoing Transport Type

Enable Digest Authentication

Nonce Validity Time (mins)\*

X.509 Subject Name

Incoming Port\*

Enable Application Level Authorization

Accept Presence Subscription

Accept Out-of-Dialog REFER\*\*

Accept Unsolicited Notification

Accept Replaces Header

Transmit Security Status

---



## Configuring SIP Profile for Trunk (Page 1 of 2)

Navigation Path: Cisco UCM Administration → Device → Device Settings → SIP Profile

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

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### SIP Profile Configuration

Related Links: Back To Find/List [Go]

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

---

**Status**

- Status: Ready
- All SIP devices using this profile must be restarted before any changes will take affect.

---

**SIP Profile Information**

Name*	Early Offer SIP Profile for Siemens
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



## Configuring SIP Profile for Trunk (Page 2 of 2)

Navigation Path: Cisco UCM Administration → Device → Device Settings → SIP Profile

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
<input checked="" type="checkbox"/> Conference Join Enabled	
<input type="checkbox"/> RFC 2543 Hold	
<input checked="" type="checkbox"/> Semi Attended Transfer	
<input type="checkbox"/> Enable VAD	
<input type="checkbox"/> Stutter Message Waiting	

---

**Trunk Specific Configuration**

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

---

**SIP OPTIONS Ping**

<input type="checkbox"/> 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

---

Save Delete Copy Reset Apply Config Add New

**Note:** Call direction: Siemens to Cisco UCM, Early attended transfer back to Siemens. Upon Early attended call transfer by Cisco UCM, call is initially completed without voice path and after 5 seconds call is disconnected. The workaround for this limitation is to configure "SIP Rel1XX options : send PRACK if 1XX contains SDP" in SIP profile on Cisco UCM.



## Configuring Service Parameters for MOH

Navigation Path: Cisco UCM Administration → System → Service Parameters

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

Navigation: Cisco Unified CM Administration

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**Service Parameter Configuration** Related Links:

**Clusterwide Parameters (Route Plan)**

<a href="#">Stop Routing on Out of Bandwidth Flag</a> *	False	False
<a href="#">Stop Routing on Unallocated Number Flag</a> *	True	True
<a href="#">Stop Routing on User Busy Flag</a> *	True	True

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

**Clusterwide Parameters (Route Class Signaling)**

<a href="#">Route Class Trunk Signaling Enabled</a> *	True	True
<a href="#">SIP Route Class Naming Authority</a> *	cisco.com	cisco.com

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

**Clusterwide Parameters (Hunt List)**

<a href="#">Stop Hunting on Out of Bandwidth Flag</a> *	False	False
<a href="#">Use Pickup Group Of Line Group Member DN</a> *	False	False

**Clusterwide Parameters (Service)**

<a href="#">Default Network Hold MOH Audio Source ID</a> *	1	1
<a href="#">Default User Hold MOH Audio Source ID</a> *	1	1
<a href="#">Duplex Streaming Enabled</a> *	True	False
<a href="#">Media Exchange Interface Capability Timer</a> *	8	8
<a href="#">Send Multicast MOH in H.245 OLC Message</a> *	True	True
<a href="#">Media Exchange Timer</a> *	12	12
<a href="#">Media Exchange Stop Streaming Timer</a> *	8	8
<a href="#">Open Video Channel Response Timer for SIP Interop</a> *	500	500
<a href="#">Port Received Timer After Call Connection</a> *	500	500
<a href="#">Media Resource Allocation Timer</a> *	12	12
<a href="#">MTP and Transcoder Resource Throttling Percentage</a> *	95	95
<a href="#">Intercluster Capabilities Mismatch Timer</a> *	1000	1000
<a href="#">Silence Suppression</a> *	False	False
<a href="#">Silence Suppression for Gateways</a> *	False	False



## Configuring Calling Name and Number Restriction

Navigation Path: Cisco UCM Administration → Device → Trunk

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

Navigation: Cisco Unified CM Administration [Go]  
CCMAdministrator | Search Documentation | About | Logout

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### Trunk Configuration

Related Links: Back To Find/List [Go]

Save [X] Delete [R] Reset [A] Add New [P]

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

#### 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\*: Restricted

Calling Name Presentation\*: Restricted

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\*: 711ulaw

Presence Group\*: Standard Presence group

SIP Trunk Security Profile\*: Non Secure SIP Trunk Profile for Siemens

Rerouting Calling Search Space: < None >



## Configuring Connected Name and Number Restriction

Navigation Path: Cisco UCM → Device → Trunk

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

Navigation: Cisco Unified CM Administration Go  
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### Trunk Configuration

Related Links: Back To Find/List Go

Save Delete Reset Add New

**Inbound Calls**

Significant Digits\* All

Connected Line ID Presentation\* **Restricted**

Connected Name Presentation\* **Restricted**

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 >	<input checked="" type="checkbox"/>

**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



## Acronyms

Acronym	Definition
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
PBX	Private Branch Exchange





### **Important Information**

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