Cisco Unified Communications Manager Express

http://blog.i-1.nl

SCCP Phone configuration

4-step-configuration:

1) Set up TFTP server on router for phones to pick SCCP firmware
2) Define system parameters for SCCP phones
3) Create the phone lines (DN)
4) Create a phone and map it to a phone line (DN)

SIP Phone configuration

4-step-configuration:

1) Set up TFTP server on router for phones to pick SIP firmware
2) SIP configuration and setup of registrar server
3) Create the voice register dn's (phone lines)
4) Create the voice register pools (phone configuration)

Softkeys

Acct
Short for "account code." Provides access to configured accounts

Answer
Picks up incoming call

Barge
Allows a user to join (barge) a call on a SIP shared line (Cisco Unified CME 7.1 or a later version)

Callback
Requests callback notification when a busy called line becomes free

CBarge
Barges (joins) a call on a shared octo-line directory number (Cisco Unified CME 4.3 or a later version)

CFwdALL
Short for "call forward all." Forwards all calls

ConfList
Lists all parties in a conference (Cisco Unified CME 4.1 or a later version)

Confrn
Short for "conference." Connects callers to a conference call

DND
Short for "do not disturb." Enables the do-not-disturb features

EndCall
Ends the current call

GPickUp
Short for "group call pickup." Selectively picks up calls coming into a phone number that is a member of a pickup group

Flash
Short for "hookflash." Provides hookflash functionality for public switched telephone network (PSTN) services on calls connected to the PSTN via a foreign exchange office (FXO) port

HLog
Places the phone of an ephone-hunt group agent into the not-ready status or, if the phone is in the not-ready status, it places the phone into the ready status

Hold
Places an active call on hold and resumes the call

Join
Joins an established call to a conference (Cisco Unified CME 4.1 or a later version)

LiveRcd
Starts the recording of a call (Cisco Unified CME 4.3 or a later version)

Login
Provides personal identification number (PIN) access to restricted phone features

MeetMe
Initiates a meet-me conference (Cisco Unified CME 4.1 or a later version)

Mobility
Soft key that forwards a call to the PSTN number defined by the Single Number Reach (SNR) feature (Cisco Unified CME 7.1 or a later version)

NewCall
Opens a line on a speakerphone to place a new call

Park
Places an active call on hold so it can be retrieved from another phone in the system

PickUp
Selectively picks up calls coming into another extension

Redial
Redials the last number dialed

Resume
Connects to the call on hold

RmLstC
Removes the last party added to a conference. This soft key only works for the conference creator (Cisco Unified CME 4.1 or a later version)

Select
Selects a call or a conference on which to take action (Cisco Unified CME 4.1 or a later version)

Trnsfer
Short for "call transfer." Transfers an active call to another extension

TrnsfVM
Transfers a call to a voice-mail extension number (Cisco Unified CME 4.3 or a later version)

Terminology

CUCME (Cisco Unified Communications Manager Express)
Callmanager software that runs on a Cisco router.
Act is a signaling proxy for call events initiated over protocols as SCCP & SIP.

SCCP (Skinny Call Control Protocol)
Developed by Selsius Corporation
Owned by Cisco Systems
Messaging system towards the CUCME

SCCP Endpoint (client or dial peer)
Cisco 7900 series IP Phones, Cisco IP Communicator, Cisco Unity, IP Blue

SIP (Session Initiation Protocol)
Developed by by Henning Schulzrinne and Mark Handley
Signaling Protocol used for controlling multimedia communication sessions (voice) over IP

SIP Endpoint (client or dial peer)
Cisco 7900 series IP Phones, Cisco IP Communicator, IP Blue, X-Lite

TFTP (Trivial File Transfer Protocol)
A file transfer protocol that is used to transfer files between the CUCME and endpoints

Codec
The application of data compression of digital audio signals containing speech

DHCP (Dynamic Host Configuration Protocol)
Dynamically distributes IP addresses to destination hosts (Phones & Computers)

CME GUI
CME configuration trough a web interface

Softkey (Template)
Software buttons on the phone displays which can be configured based on different phone states

DN (Directory Number)
Phone number extension (phone line)

Call States (adjustable softkeys)

Alerting	Seized
Connected	Remote in Use
Hold	Ringing
Idle	

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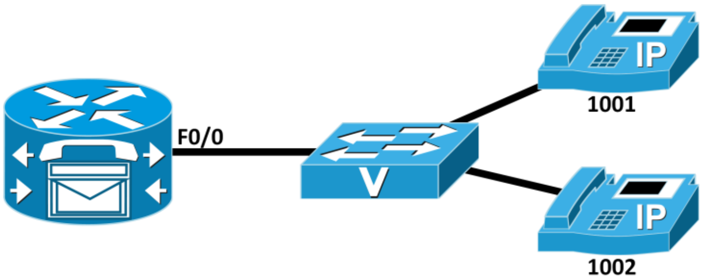
V1.0

SCCP Configuration Example

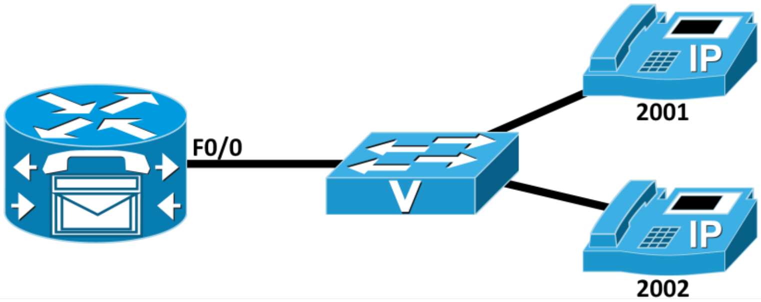



```
! 1) Set up TFTP server on router for phones to pick SCCP firmware
tftp-server flash:PHONE/7940-7960/P00308000500.bin alias P00308000500.bin
tftp-server flash:PHONE/7940-7960/P00308000500.loads alias P00308000500.loads
tftp-server flash:PHONE/7940-7960/P00308000500.sb2 alias P00308000500.sb2
tftp-server flash:PHONE/7940-7960/P00308000500.sbn alias P00308000500.sbn
!
! 2) Define system parameters for SCCP phones
telephony-service
 no auto-reg-ephone
 max-ephones 2
 max-dn 2
 ip source-address 10.10.110.3 port 2000
 load 7960-7940 P00308000500
 create cnf-files
!
! 3) Create the phone lines (DN)
ephone-dn  1  octo-line
 number 1001
 description 1111-1001
!
ephone-dn  2  octo-line
 number 1002
 description 1111-1002
!
! 4) Create a phone and map it to a phone line (DN)
ephone  1
 description 79XX PHONE 1001
 mac-address 0079.96FF.287F
 type 7960
 button  1:1
!
ephone  2
 description 79XX PHONE 1002
 mac-address 003C.D56A.9826
 type 7960
 button  1:2
```

CME GUI Configuration

```
ip http server
no ip http secure-server
ip http path flash:/GUI
!
web admin system name admin password cisco
dn-webedit
```

SIP Configuration Example




```
! 1) Set up TFTP server on router for phones to pick SIP firmware
tftp-server flash:PHONE/7940-7960/POS3-08-9-00.loads alias POS3-08-6-00.loads
tftp-server flash:PHONE/7940-7960/POS3-08-9-00.sb2 alias POS3-08-6-00.sb2
tftp-server flash:PHONE/7940-7960/P003-08-9-00.bin alias P003-08-6-00.bin
tftp-server flash:PHONE/7940-7960/P003-08-9-00.sbn alias P003-08-6-00.sbn
!
! 2) SIP configuation and setup of registrar server
voice service voip
 allow-connections sip to sip
 sip
  bind all source-interface Vlan100
  registrar server expires max 600 min 60
!
voice register global
 mode cme
 source-address 11.11.101.1 port 5060
 max-dn 2
 max-pool 2
 load 7960-7940 POS3-08-6-00
 authenticate register
 tftp-path flash:
 create profile
!
! 3) Create the voice register dn's (phone lines)
voice register dn  1
 number 2001
!
voice register dn  2
 number 2002
!
! 4) Create the voice register pools (phone configuration)
voice register pool  1
 description 79XX PHONE 2001
 id mac 003F.A3F4.AA54
 type 7960
 number 1 dn 1
 dtmf-relay sip-notify
 username 2001 password cisco
 description 2222-2001
!
voice register pool  2
 description 79XX PHONE 2002
 id mac 002D.456E.345A
 type 7960
 number 1 dn 2
 dtmf-relay sip-notify
 username 2002 password cisco
 description 2222-2002
```