Cisco Unified Communications Manager Express – SIP Implementation Guide

Version 3.0

October 15, 2007

Revision History

Revision	Date	Author	Comments
1.0	October 15, 2007	Tony Huynh	Initial Draft
2.0	October 23, 2007	Tony Huynh	Added mwi-server config
3.0	October 30, 2007	Tony Huynh	General formatting

Table of Contents

1. Introduction 3
1.1 Scope 3
1.2 Audience 3
2. Overview 3
2.1 Hardware 3
2.2 IP Phone Firmware 3
2.3 Network Topology, IP Addressing, and Extension Number Scheme 4
3. Configure System Configurations 5
3.1 Configure DHCP 5
3.2 Configure Fasthernet interface and switchports 6
3.3 Configure TFTP statements 8
4. Configure Basic Telephony Features 8
4.1 Configure Voice Service Parameters 8
4.2 Configure Voice Register Global Parameters 8
4.3 Configure Connection to CUE 9
4.4 Configure Extension and Parameters 10
4.5 Configure SIP Phone 11
4.6 Configure Advanced Parameters 11
4.6.1 Enable Presence for internal lines 11
4.6.2 Enable a Directory Number to be watched 12
4.6.3 Enabling a SIP Phone to Monitor BLF Status for Speed-Dials and Call Lists 12
4.7 Configure Parallel Hunt Group 13
5. Troubleshooting 14
5.1 SIP IP Phone does not get dial tone 14
5.1.1 Steps to remedy 14
5.2 IP Phone can’t upgrade to latest firmware 14
5.2.1 Steps to remedy 14
5.2.2 Debugs to collect 14
5.3 Can’t provision phone 14
5.3.1 Steps to remedy 14
5.4 SIP Phone gets DHCP but can’t get communicate with CME 15
5.4.1 Steps to remedy 15
6. Sample Config 15
7. Reference 22
7.1 Information on 3911 SIP IP Phones 22
7.2 CME configuration guide 22
7.3 CME Documentation 22

1. Introduction

1.1 Scope

This document provides step-by-step instructions for setting up a standalone CME using SIP phones. The document outlines a Cisco Unified Communications Manager Express (CME) system with 4 SIP phones, with configurations for setting up the CME system and SIP phones. Though the document will cover configuration steps to allow CME to interoperate with CUE, CUE configuration is outside of the scope of this paper.

1.2 Audience

This document is provided for Cisco Systems Engineers (SEs) and resellers who wish to deploy a CME system using SIP phones.

2. Overview

2.1 Hardware

The information in this document is based on the following hardware:

- Cisco 2801 running CME 4.2 with IOS 12.4(11)XW2
- 9-port double-wide 10/100BASE-T Ethernet switch HWIC
- Cisco 7970 SIP Phones
- Cisco 3911 SIP Phone

2.2 IP Phone Firmware

Please refer to the following link to determine the appropriate SIP firmware to use for each CME version.

http://www.cisco.com/en/US/products/sw/voicesw/ps4625/prod_installation_guide09186a00805acf50.html

Since we will be using CME 4.2, refer to the CME 4.2 Specifications link. In the following deployment, the following phoneloads were used.

- 7970 IP Phone (SIP) – SIP70.8-2-1S
- 3911 IP Phone (SIP) - SIP3951.8-0-2-9

You will need to have your CCO username/password in order to download SIP phoneloads. The CME zip file does not have SIP phone loads, so you will have to procure the firmware at the following site.

http://www.cisco.com/cgi-bin/tablebuild.pl/ip-7900ser
http://www.cisco.com/cgi-bin/tablebuild.pl/ip-3900ser

For 7970 phone
Download SIP phone load cmterm-7970_7971-sip.8-2-1.zip file and unzip the file into your TFTP folder.

For 3911 phone
Download the cmterm-3951-sip.8-0-2.zip file and unzip the file into your TFTP folder. NOTE: 3911 and 3951 phone use the same firmware, so don't be concerned that the filename reflects a 3951 phone.

After you have unzipped both the ZIP files in your TFTP folder, copy all the firmware files onto the CME flash using your TFTP server. Make sure you copy all the following files onto flash.

```
SIP3951.8-0-2-9.loads
SIP3951.8-0-2-9.zz
DSP3951.0-0-0-1.zz
BOOT3951.0-0-0-9.zz
SIP70.8-2-1S.loads
term70.default.loads
term71.default.loads
apps70.8-0-2-55.sbn
cnu70.8-2-0-55.sbn
cvm70.sip.8-2-0-55.sbn
dsp70.8-2-0-55.sbn
jar70.sip.8-0-2-25.sbn
```

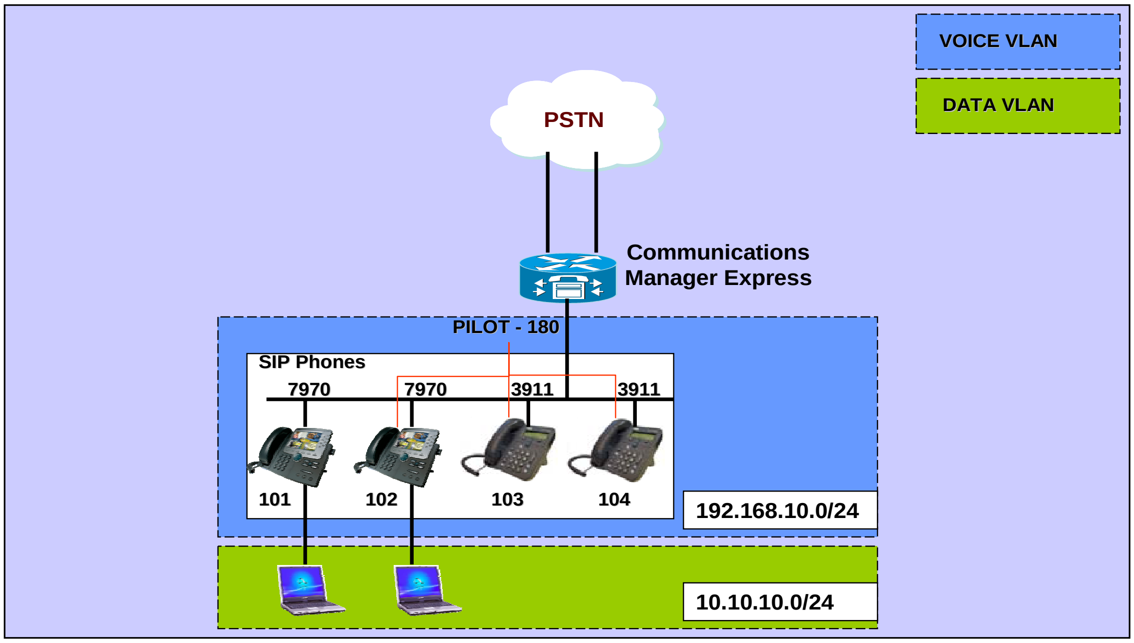
2.3 Network Topology, IP Addressing, and Extension Number Scheme

The following network topology will be used to deploy the CME system with SIP phones. The table below outlines the network addressing scheme that will be used in this setup.

Purpose	VLAN	Network	Interface	Interface Address
Voice	192	192.168.10.0/24	VLAN 192	192.168.10.1/24
Data	100	10.10.10.0/24	VLAN 100	10.10.10.1/24

Protocol	Phone Type	Extension Number	Phone Number External Mask
SIP	7970	101	4085251001
SIP	7970	102	4085251002
SIP	3911	103	4085251003
SIP	3911	104	4085251004
Voicemail Pilot	100	**AA Pilot**	110
MWI On	800	**MWI Off**	801

Cisco Unified Communications Manager Express – SIP Network Topology



3. Configure System Configurations

3.1 Configure DHCP

It will be necessary to configure two separate DHCP pools - IP Phones will use the Voice DHCP pool and PCs will use the Data DHCP pool. IP Phones will need to use DHCP option 150 to provide the IP address of the TFTP Server.

If there are any devices in either pool with static IP addresses, make sure these addresses are excluded from the DHCP pool to avoid addressing conflicts. You can use the **show ip dhcp binding** command to verify which addresses the IP Phones and PCs are receiving from the router.

<u>Sample config</u>:

```
ip dhcp excluded-address 10.10.10.1 10.10.10.10
ip dhcp excluded-address 192.168.10.1 192.168.10.10
!
ip dhcp pool data
  network 10.10.10.0 255.255.255.0
  default-router 10.10.10.1
!
ip dhcp pool voice
  network 192.168.10.0 255.255.255.0
  option 150 ip 192.168.10.1
  default-router 192.168.10.1
```

3.2 Configure Fasthernet interface and switchports

In the following section, you will configure the VLAN interfaces for both the DATA and VOICE VLAN and assign switchports into their respective VLANs.

NOTE: Prior to configuring VLANs, be sure to add the VLANs above to the VLAN database with the following commands:

```
CME-SIP#vlan database
    % Warning: It is recommended to configure VLAN from config mode,
     as VLAN database mode is being deprecated. Please consult user
      documentation for configuring VTP/VLAN in config mode.

CME-SIP(vlan)#vlan 100
VLAN 100 modified:
CME-SIP(vlan)#vlan 192
VLAN 192 modified:
CME-SIP(vlan)#exit
APPLY completed.
Exiting....
CME-SIP#
```

In the following section, configure the switchports to be connected to both the Voice and Data VLANs. IP Phones will automatically be assigned into the Voice VLAN and PCs connected to either the switchport directly or connected to the switchport on the IP Phone will be assigned to the Data VLAN.

```
interface FastEthernet0/3/0
 description 7970 Phone
 switchport trunk native vlan 100
 switchport mode trunk
 switchport voice vlan 192
 spanning-tree portfast
!
interface FastEthernet0/3/1
 description 7970 Phone
 switchport trunk native vlan 100
 switchport mode trunk
 switchport voice vlan 192
 spanning-tree portfast
!
 interface FastEthernet0/3/2
 description 3911 Phone
 switchport trunk native vlan 100
 switchport mode trunk
 switchport voice vlan 192
 spanning-tree portfast
!
interface FastEthernet0/3/3
 description 3911 Phone
 switchport trunk native vlan 100
 switchport mode trunk
 switchport voice vlan 192
 spanning-tree portfast
! !
interface Vlan100
description Data VLAN
 ip address 10.10.10.1 255.255.255.0
!
interface Vlan192
description Voice VLAN
 ip address 192.168.10.1 255.255.255.0
```

3.3 Configure TFTP statements

The following configuration allows CME to serve the IP Phones their firmware. **[Required]**

```
tftp-server flash:SIP3951.8-0-2-9.loads
tftp-server flash:SIP3951.8-0-2-9.zz
tftp-server flash:DSP3951.0-0-0-1.zz
tftp-server flash:BOOT3951.0-0-0-9.zz
tftp-server flash:SIP70.8-2-1S.loads
tftp-server flash:term70.default.loads
tftp-server flash:term71.default.loads
tftp-server flash:apps70.8-0-2-55.sbn
tftp-server flash:cnu70.8-2-0-55.sbn
tftp-server flash:cvm70.sip.8-2-0-55.sbn
tftp-server flash:dsp70.8-2-0-55.sbn
tftp-server flash:jar70.sip.8-0-2-25.sbn
```

4. Configure Basic Telephony Features

4.1 Configure Voice Service Parameters

Configure system to allow calls from SIP to SIP endpoints and enable SIP registrar. **[Required]**

```
voice service voip
allow-connections sip to sip Enable sip to sip calls
sip
registrar server expires max 1200 min 300 Enable IOS SIP registrar
```

4.2 Configure Voice Register Global Parameters

In the following section, you will be configuring voice register global parameters. Voice Register Global configurations for SIP are similar to telephony-service configuration parameters for SCCP phones. **[Required]**

Sample configuration:

```
voice register global
 mode cme  Set IOS SIP registrar to CME mode
 source-address 192.168.10.1 port 5060 Set source address for phone registration
 max-dn 20 Set max extensions
 max-pool 10 Set max phones
 load 7970 SIP7 SIP70.8-2-1S Specify phone loads for each phone type
 load 3911 SIP3951.8-0-2-9 Specify phone loads for each phone type
 authenticate register Set authentication for phone registration
 authenticate realm cisco.com
 tftp-path flash:  Specify path for tftp files
 create profile Create config files for all phones
 dialplan-pattern 1 4085251... extension-length 3  Configure dial-plan pattern for system
```

4.3 Configure Connection to CUE

Configure a dial-peer and MWI server to interoperate with CUE. In order for CME to interoperate with CUE, it is necessary to configure SIP CME as a back to back user agent (B2BUA) - meaning all signaling and RTP stream goes through the CME. The following configuration required is to enable connectivity to CUE.

```
dial-peer voice 2 voip
   destination-pattern 1.0  Specify destination-pattern to reach CUE VM and AA
   session target ipv4:10.1.10.1  Configure IP address to reach CUE
   session protocol sipv2
   dtmf-relay sip-notify  Configure DTMF method to communicate with CUE
   b2bua  Enable B2BUA for CME for calls to CUE
   codec g711ulaw
   no vad
```

4.4 Configure MWI Server for SIP Phones

Configure MWI Server for SIP phones

```
sip-ua
 mwi-server ipv4:10.1.10.1 expires 120 port 5060 transport udp

NOTE : 10.1.10.1 is the IP address of CUE
```

4.5 Configure Extension and Parameters

Configure "voice register dn" to create extension numbers for ephones. In the network topology above, there are 4 extensions, which we will create now. **[Required]**

voice register dn 1
name Phone1 ***Set display name***
label 4085251001 ***Set display label***
number 101 ***Set extension number***
mwi ***Enable MWI notification***
call-forward b2bua noan 100 timeout 20 ***Configure call forward noan to voicemail pilot***
call-forward b2bua busy 100 timeout 20 ***Configure call forward busy to voicemail pilot***
allow watch ***Allow this number to be watched (presence)***
!
voice register dn 2
name Phone2
label 4085251002
number 102
mwi ***Enable MWI notification***
call-forward b2bua noan 100 timeout 20 ***Configure call forward noan to voicemail pilot***
call-forward b2bua busy 100 timeout 20 ***Configure call forward busy to voicemail pilot***
allow watch
!
voice register dn 3
name Phone3
label 4085251003
number 103
mwi ***Enable MWI notification***
call-forward b2bua noan 100 timeout 20 ***Configure call forward noan to voicemail pilot***
call-forward b2bua busy 100 timeout 20 ***Configure call forward busy to voicemail pilot***
allow watch
!
voice register dn 4
name Phone4
label 4085251004
number 104
mwi ***Enable MWI notification***
call-forward b2bua noan 100 timeout 20 ***Configure call forward noan to voicemail pilot***
call-forward b2bua busy 100 timeout 20 ***Configure call forward busy to voicemail pilot***
allow watch

4.6 Configure SIP Phone

In the following configuration, you will be configuring voice register pool parameters for each sip phone. Voice register pool for SIP phones is identical to ephones for SCCP phones. **[Required]**

Sample configuration:

voice register pool 3
id mac 001A.A11B.500E ***Specify phone mac-address***
type 3911 ***Specify phone type***
number 1 dn 3 ***Assign button 1 dn tag 3***
dtmf-relay sip-notify ***Configure dtmf-relay sip-notify to work with CUE***
codec g711ulaw ***Specify codec***
username user1 password cisco ***Configure username and password for SIP registrar***
!

NOTE: Multiple methods for DTMF can be configured under voice register pool, but for each SIP phone that has a voicemail box on CUE, configure "**dtmf-relay sip-notify**".

4.7 Configure Advanced Parameters

In the following section, you will be configuring advanced parameters for SIP phones such as presence with BLF status. Presence with BLF allows either a SCCP phone or SIP phone to monitor the status of another SIP extension – enabling presence information between phones. **[Optional]**

The following phones support SIP presence service on CME.

Restrictions

BLF Call-List

- Supported only on Cisco Unified IP Phone 7941G, 7941GE, 7961G, 7961GE, 7970G, and 7971GE.

BLF Speed-Dial

- Supported only on Cisco Unified IP Phone 7941G, 7941GE, 7961G, 7961GE, 7970G, and 7971GE.

4.7.1 Enable Presence for internal lines

Perform the following steps to enable the router to accept incoming presence requests from internal watchers and SIP trunks.

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **presence enable**
5. **exit**
6. **presence**
7. **max-subscription number**
8. **presence call-list**

9. **end**

<u>Sample config</u>

```
Presence  Enable presence service
 presence call-list  Enable BLF monitoring of directory numbers
 max-subscription 120  Configure max number watched sessions
!
sip-ua
 presence enable  Enable router to accept incoming presence request
```

4.7.2 Enable a Directory Number to be watched

To enable a line associated with a directory number to be monitored by a phone registered to a Cisco Unified CME router, perform the following steps. The line is enabled as a presentity and phones can subscribe to its line status through BLF call-list and BLG speed-dial features. There is no restriction on the type of phone that can have its lines monitored; any line on any IP phone or on an analog phone on supported voice gateways can be a presentity.

1. **enable**
2. **configure terminal**
3. **voice register dn dn-tag**
4. **number number**
5. **allow watch**
6. **end**

<u>Sample Config</u>

```
voice register dn  1
 number 101
 allow watch  Allow this number to be watched
 mwi
 name Phone1
 label 4085251001

- Repeat this configuration for each extension number that needs to be watched
```

NOTE: This step was already done when we first configured "voice register dns".

4.7.3 Enabling a SIP Phone to Monitor BLF Status for Speed-Dials and Call Lists

A watcher can monitor the status of lines associated with internal and external directory numbers (presentities) through the BLF speed-dial and BLF call-list presence features. To enable the BLF notification features on a SIP phone, perform the following steps.

1. **enable**
2. **configure terminal**
3. **voice register pool pool-tag**
4. **number tag dn dn-tag**

5. **blf-speed-dial tag number label string**
6. **presence call-list**
7. **exit**
8. **voice register global**
9. **mode cme**
10. **create profile**
11. **restart**
12. **end**

Sample Config

```
voice register pool  1
 id mac 0016.47CD.9BD7
 type 7970
 number 1 dn 1
 presence call-list  Enable this phone to have presence call list
 dtmf-relay sip-notify
 username user1 password cisco
 codec g711ulaw
 blf-speed-dial 2 102 label "Phone2"  Enable this line to monitor extension 1002
 blf-speed-dial 3 103 label "3911-1"  Enable this line to monitor extension 1003
 blf-speed-dial 4 104 label "3911-2"  Enable this line to monitor extension 1004
```

Note: Be sure to perform “restart” every time you change a SIP phone configuration.

Reference for configuring SIP Presence Service:
http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_configuration_guide_chapter09186a008087793a.html#wp999282

4.8 Configure Parallel Hunt Group

In this section, will be assigning extensions 102, 103, and 104 into a parallel hunt group – a parallel hunt group is a hunt group that rings all members in the group simultaneously.

```
voice hunt-group 1
pilot 180  Configure Hunt group pilot number
list 102, 103, 104  Specify members in hunt-group
final 100  Specify final number as Voicemail Pilot
```

5. Troubleshooting

5.1 SIP IP Phone does not get dial tone

A common cause for SIP IP Phones not being able to get dial tone is that there is another phone with the same extension – as of CME 4.2, shared line is not supported on SIP Phones. Thus SIP phones can not share the same extension among multiple phones. Additionally, make sure that the SIP phone is provisioned with a proper extension.

5.1.1 Steps to remedy

- Make sure that SIP phone has extension configured and extension shows on SIP phone
- Make sure that there is not another SIP or SCCP phone configured with the same extension

5.2 IP Phone can't upgrade to latest firmware

The most likely causes for failure to be able to upgrade a phone is missing firmware files placed on the CME flash or missing "tftp-server" commands.

5.2.1 Steps to remedy

- First, check that the necessary firmware files are stored on the flash – perform "dir flash" to check flash for files
- Check to see if you have updated the OS79XX.TXT file to reflect the correct firmware – the 79XX phones check this file to load the appropriate firmware and change from SCCP to SIP
- Check to make sure that the correct tftp-server statements are added for each firmware file – see section 3.3
- Make sure the "load" command under voice register global is added for each type of SIP phone – section 4.2

5.2.2 Debugs to collect

In order to troubleshoot further, collect the following debugs in order to see if the phone is able to get the appropriate phone loads from the CME flash.

- Debug tftp events

5.3 Can't provision phone

The most like causes for not being able to provision is phone is that the phone doesn't have the proper IP address with TFTP server option.

5.3.1 Steps to remedy

- First, check to see that the phone receives an IP address and the proper TFTP server IP address
- Make sure that all proper "voice register global" commands are added – see section 4.2.

- Check that you are using the correct mac-address underneath each “voice register pool” configuration

5.4 SIP Phone gets DHCP but can’t get communicate with CME

The most likely cause for the SIP phone not being able to communicate with the CME after the phone gets DHCP (with the proper TFTP server address) is because the “username” and “password” credentials are not configured.

5.4.1 Steps to remedy

- Check and make sure that “username” and “password” are defined under the “voice register pool” configuration for each phone.
- If the “username” and “password” are configured, but the command “create profile” is not performed under “voice register global”, then the configuration file will not be updated and the phone will not get this new configuration. Be sure to reset of the phone after performing “create profile”.

6. Sample Config

```
CME-SIP#sh ver
Cisco IOS Software, 2801 Software (C2801-IPVOICE-M), Version 12.4(11)XW2, RELEASE SOFTWARE (fc1)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2007 by Cisco Systems, Inc.
Compiled Mon 02-Jul-07 19:10 by prod_rel_team

ROM: System Bootstrap, Version 12.3(8r)T6, RELEASE SOFTWARE (fc1)

CME-SIP uptime is 18 hours, 55 minutes
System returned to ROM by reload at 17:01:34 UTC Wed Oct 3 2007
System image file is "flash:c2801-ipvoice-mz.124-11.XW2.bin"

Cisco 2801 (revision 4.1) with 235520K/26624K bytes of memory.
Processor board ID FHK084510HS
11 FastEthernet interfaces
1 terminal line
2 Voice FXO interfaces
3 DSPs, 48 Voice resources
1 cisco service engine(s)
DRAM configuration is 64 bits wide with parity disabled.
191K bytes of NVRAM.
62720K bytes of ATA CompactFlash (Read/Write)

Configuration register is 0x2102

CME-SIP#  sh run
Building configuration...

Current configuration : 6227 bytes
!
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
```

```
!
hostname CME-SIP
!
boot-start-marker
boot-end-marker
!
logging buffered 999999
no logging console
enable password cisco
!
no aaa new-model
ip cef
!
!
no ip dhcp use vrf connected
ip dhcp excluded-address 10.10.10.1 10.10.10.10
ip dhcp excluded-address 192.168.10.1 192.168.10.10
!
ip dhcp pool data
   network 10.10.10.0 255.255.255.0
   default-router 10.10.10.1
!
ip dhcp pool voice
   network 192.168.10.0 255.255.255.0
   option 150 ip 192.168.10.1
   default-router 192.168.10.1
!
!
no ip domain lookup
multilink bundle-name authenticated
!
!
!
voice service voip
 allow-connections sip to sip
 sip
  registrar server expires max 1200 min 300
!
!
!
!
!
voice register global
 mode cme
 source-address 192.168.10.1 port 5060
 max-dn 20
 max-pool 10
 load 7970 SIP70.8-2-1S
 load 3911 SIP3951.8-0-2-9
 authenticate register
 authenticate realm cisco.com
 voicemail 100
 tftp-path flash:
 create profile sync 0000589556325309
!
```

```
voice register dn  1
 number 101
 mwi
 call-forward b2bua noan 100 timeout 20
 allow watch
 name Phone1
 label 4085251001
!
voice register dn  2
 number 102
 mwi
 call-forward b2bua noan 100 timeout 20
 allow watch
 name Phone2
 label 4085251002
!
voice register dn  3
 number 103
 mwi
 call-forward b2bua noan 100 timeout 20
 allow watch
 name Phone3
 label 4085251003
!
voice register dn  4
 number 104
 mwi
 call-forward b2bua noan 100 timeout 20
 allow watch
 name Phone4
 label 4085251004
!
voice register pool  1
 id mac 0016.47CD.9BD7
 type 7970
 number 1 dn 1
 presence call-list
 dtmf-relay sip-notify
 username user1 password cisco
 codec g711ulaw
 blf-speed-dial 2 102 label "Phone2"
 blf-speed-dial 3 103 label "3911-1"
 blf-speed-dial 4 104 label "3911-2"
!
voice register pool  2
 id mac 0014.6948.1D52
 type 7970
 number 1 dn 2
 dtmf-relay sip-notify
 username user2 password cisco
 codec g711ulaw
!
voice register pool  3
 id mac 001A.A11B.4FCE
 type 3911
```

```
 number 1 dn 3
 dtmf-relay sip-notify
 username user3 password cisco
 codec g711ulaw
!
voice register pool  4
 id mac 001A.A11B.500E
 type 3911
 number 1 dn 4
 dtmf-relay sip-notify
 username user4 password cisco
 codec g711ulaw
!
voice hunt-group 1 parallel
 final 100
 list 102,103,104
 pilot 180
!
!
!
!
voice-card 0
!
!
!
archive
 log config
 hidekeys
!
!
!
interface Loopback0
 ip address 10.1.10.2 255.255.255.0
!
interface FastEthernet0/0
 no ip address
 shutdown
 duplex auto
 speed auto
!
interface Service-Engine0/0
 ip unnumbered Loopback0
 service-module ip address 10.1.10.1 255.255.255.0
 service-module ip default-gateway 10.1.10.2
!
interface FastEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
!
interface FastEthernet0/3/0
 description 7970 Phone
 switchport trunk native vlan 100
 switchport mode trunk
```

```
 switchport voice vlan 192
 spanning-tree portfast
!
interface FastEthernet0/3/1
 description 7970 Phone
 switchport trunk native vlan 100
 switchport mode trunk
 switchport voice vlan 192
 spanning-tree portfast
!
interface FastEthernet0/3/2
 description 3911 Phone
 switchport trunk native vlan 100
 switchport mode trunk
 switchport voice vlan 192
 spanning-tree portfast
!
interface FastEthernet0/3/3
 description 3911 Phone
 switchport trunk native vlan 100
 switchport mode trunk
 switchport voice vlan 192
 spanning-tree portfast
!
interface FastEthernet0/3/4
 description Phone
 switchport trunk native vlan 100
 switchport mode trunk
 switchport voice vlan 192
 spanning-tree portfast
!
interface FastEthernet0/3/5
 description Phone
 switchport trunk native vlan 100
 switchport mode trunk
 switchport voice vlan 192
 spanning-tree portfast
!
interface FastEthernet0/3/6
 description Phone
 switchport access vlan 192
 switchport trunk native vlan 100
 switchport mode trunk
 switchport voice vlan 192
 spanning-tree portfast
!
interface FastEthernet0/3/7
 description Phone
 switchport access vlan 192
 switchport trunk native vlan 100
 switchport mode trunk
 switchport voice vlan 192
 spanning-tree portfast
!
interface FastEthernet0/3/8
```

```
 switchport access vlan 192
!
interface Vlan1
 no ip address
!
interface Vlan100
 ip address 10.10.10.1 255.255.255.0
!
interface Vlan192
 ip address 192.168.10.1 255.255.255.0
!
ip route 10.1.10.1 255.255.255.255 Service-Engine0/0
!
!
ip http server
!
!
!
tftp-server flash:BOOT3951.0-0-0-9.zz
tftp-server flash:SIP3951.8-0-2-9.zz
tftp-server flash:DSP3951.0-0-0-1.zz
tftp-server flash:SIP3951.8-0-2-9.loads
tftp-server flash:SIP70.8-2-1S.loads
tftp-server flash:term70.default.loads
tftp-server flash:term71.default.loads
tftp-server flash:apps70.8-0-2-55.sbn
tftp-server flash:cnu70.8-2-0-55.sbn
tftp-server flash:cvm70.sip.8-2-0-55.sbn
tftp-server flash:dsp70.8-2-0-55.sbn
tftp-server flash:jar70.sip.8-0-2-25.sbn
!
control-plane
!
!
!
voice-port 0/1/0
!
voice-port 0/1/1
!
!
!
!
!
dial-peer voice 2 voip
 description ** cue voicemail pilot number **
 translation-profile outgoing PSTN_CallForwarding
 destination-pattern 100
 b2bua
 session protocol sipv2
 session target ipv4:10.1.10.1
 dtmf-relay sip-notify
 codec g711ulaw
 no vad
!
dial-peer voice 3 voip
```

```
 description ** cue auto attendant number **
 translation-profile outgoing PSTN_CallForwarding
 destination-pattern 110
 b2bua
 session protocol sipv2
 session target ipv4:10.1.10.1
 dtmf-relay sip-notify
 codec g711ulaw
 no vad
!
!
presence
 presence call-list
 max-subscription 120
!
sip-ua
 mwi-server ipv4:10.1.10.1 expires 120 port 5060 transport udp
 presence enable
!
!
!
!
!
line con 0
line aux 0
line 66
 no activation-character
 no exec
 transport preferred none
 transport input all
 transport output pad telnet rlogin lapb-ta mop udptn v120
line vty 0 4
 password cisco
 login
!
scheduler allocate 20000 1000
end

CME-SIP#
```

7. Reference

7.1 Information on 3911 SIP IP Phones

http://www.cisco.com/en/US/products/ps7193/products_qanda_item0900aecd8069bb1a.shtml

7.2 CME configuration guide

http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_configuration_guide_book09186a00807c5776.html

7.3 CME Documentation

http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_documentation_roadmap09186a0080189132.html