SIP Trunking Configuration Guide for Cisco Unified Communications 500 Series

PAETEC

Version 1.0

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1 Overview

This document describes the configuration procedures required for the Cisco Unified Communications 500 Series to make full use of the capabilities of legacy McLeod VOIP platform, the versions of UC500 and network equipment used in the certification tests, compatibilities issues if any.

The UC500 is one of the many access devices that are interoperable with legacy McLeod network. It uses the Session Initiation Protocol (SIP) to communicate with legacy McLeod for call control signaling. It also translates voice to audio packets for transmission across a packet network.

This guide describes the specific configuration items for UC500 that is important for use with the legacy McLeod VOIP platform. It does not describe the purpose and use of all configuration options on the UC500. For those details, see the *Cisco Unified Communications Manager Express System Administrator Guide,* available from Cisco CCO.

2 PAETEC SIP Trunking Service Features

SIP Trunking from PAETEC delivers local and long distance voice service, secure data networking, and broadband Internet access on one performance-guaranteed connection to an office site with Cisco UC500 Series for Small Business.

Customers looking for a turnkey access solution can select the PAETEC equipment rental option, in which PAETEC provides a router that interfaces with your IP phone system equipment.

SIP Trunking can help your business if you need

- High-performance IP-based voice, MPLS VPN, and Internet service.
- Efficient, dynamically-managed bandwidth.
- Robust control of network call handling and other service features.
- Packaged pricing and the convenience of one provider.

Solution Features

- Real-time Web-based control of your network services from any Internet connected PC.
  - Re-route inbound calls from one location to another on the PAETEC IP based network, without long distance charges.
  - Balance the number of incoming calls to available staff and send overflow to voicemail, an answering service, or to another location.
  - Restrict specific types of calls to or from your business by area code, number, or call type.

- High-speed data connections; you choose your bandwidth.
  - 1.5 to 6.0 Mbps (with PAETEC equipment rental options).
  - 1.5 to 12.0 Mbps (with service terminating to your router or IP phone system).

- Dynamic bandwidth allocation optimizes performance.
- Free unlimited site-to-site calling to other SIP Trunking and Dynamic Integrated Access locations.
- Gateway access to the Public Switched Telephone Network.
- Ample scalability and flexibility.

You'll also appreciate

- Contractually guaranteed network performance.
  - Convenient abbreviated dialing between locations.
  - Optional components include:
  - MPLS VPN for the multi-site enterprise and VPN remote access.
  - Managed Firewall in the PAETEC network.
  - Web Hosting.
  - Additional telephone numbers in blocks of 20 or singles.
  - Additional PSTN call paths in increments of 6 (up to 48 call paths per
  - 1.5 Mbps).
  - Static IP addresses are available for advanced server applications or
  - VPN needs.
  - Branded e-mail addresses for sites with a large number of employees.

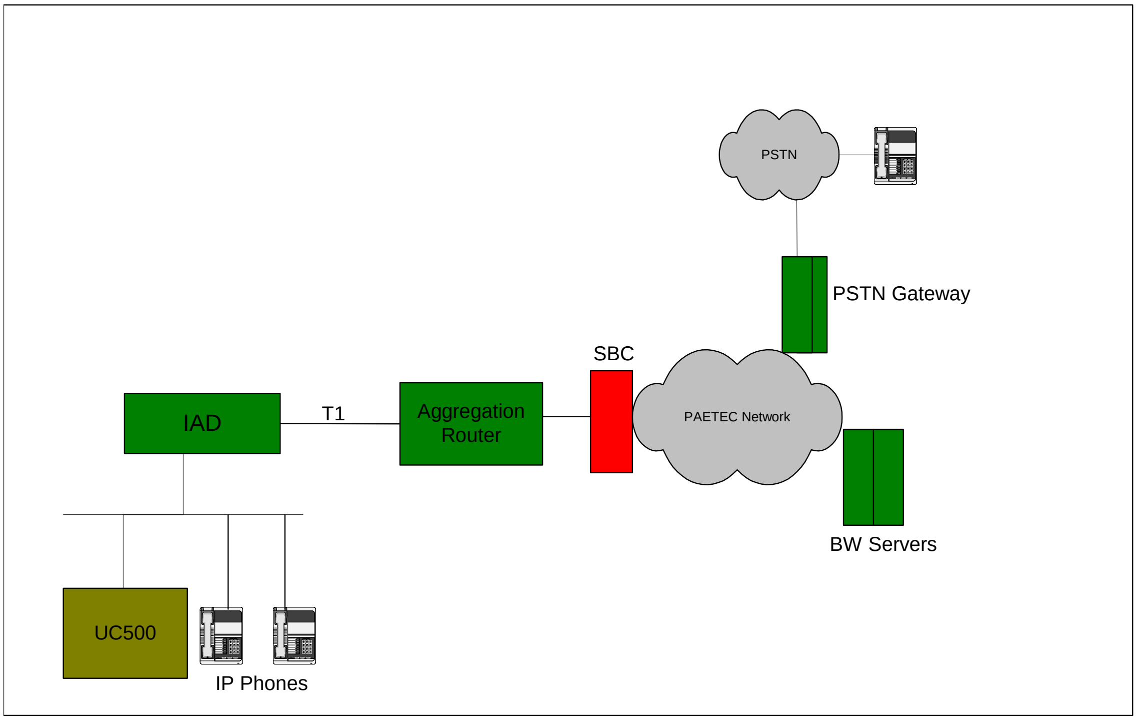
- Equipment rental, when provided by PAETEC, includes:
  - Ability to enable NAT (Network Address Translation) protection of DHCP services to a Local Area Network.

- Affordable battery back-up options to protect critical applications.
- Professional onsite installation, maintenance, and repair at no additional charge and without annual maintenance fees.

- Service Level Agreements guarantee the reliability of your critical voice and Internet services including 24 x 7 x 365 network monitoring and support via the PAETEC Network Management Center.

3 Sample Customer Premise Network Overview

The following diagram shows a typical network setup with our SIP trunk service offering. The UC500 connects to the network via an access router such as a Cisco or Adtran IAD. The IAD connects to the PAETEC network via either a single T1 or multiple T1 connections. The IP phones are on the same LAN behind the IAD as UC500.



4 Equipment and Software Validated

Cisco UC500 IP PBX Components	
Cisco UC500	12.4(11r)XW
Cisco IP Phone 7960	P00308000500
PAETEC SIP Trunking Service Components	
PACTEC BW	R14
PACTEC PSTN Gateway	V3.10.1.7.xp19
PACTEC SBC	V3.3.2(32042)

5 Device Capabilities and Known Interoperability Issues

This section describes some of the key interoperability features supported by the UC500, as well as interoperability issues and impact. The following table describes capabilities.

5.1 Capabilities

Device Type	Generic SIP Smart (Proxy Add)
SIP Proxy FQDN DNS Lookup (A, SRV, NAPTR)	A, SRV
Outbound Proxy Configurable	Yes
Outbound Proxy FQDN DNS Lookup (A, SRV, NAPTR)	A, SRV
SIP Connect (bulk registration)	Yes
Voicemail Support (PBX, BroadWorks, Both)	Both
Device Services	Supports attended Call Transfer, unattended call transfer, Call Conferencing, call hold, caller ID, call waiting and Call Forwarding.
Codec's	G.729A and G.711ulaw were tested
RFC 2833 DTMF	Yes
Fax	G.711ualw
Auto Attendant (PBX)	Yes

5.2 Interoperability Issues

This section lists the known interoperability issues between BroadWorks R14 and UC500.

BroadWorks	Cisco UC500			
Releases 14 SP1	**12.4(11r)XW**			
Anonymous Call Rejection The function is not supported Workaround: enable the feature on BW	X			
DND This function is broken. Workaround: enable DND on BroadWorks	X			
Phone Protocol Cisco SCCP (Skinny Call Control Protocol) should be used on Cisco IP Phones. Some functions may not be supported if SIP protocol is used on the phones.	X			

6 PAETEC BroadWorks Device Identity/Profile

Cisco UC500 was tested using Cisco UC500 device profile with following attributes, authentication also needs to be enabled on the trunk group.

Cisco UC500 Identify/Device Profile	
Signaling Address Type	Intelligent Proxy Addressing
Number of Lines	Unlimited
Registration Capable	Enabled
Static Registration Capable	
E.164 Capable	
Trusted	
Authentication Override	
Video Capable	
RFC 3264 Hold	Enabled
Route Advance	
Wireless Integration	
PBX Integration	Enabled
Use Business Trunking Contact	Enabled
Forwarding Override	
Conference Device	
Music On Hold Device	
Web Based Configuration URL	
Auto Configuration Type	
Reset Event	
Enable Monitoring	
CPE System File Name	
Device File Format	

7 PAETEC BroadWorks Configuration

Reference PAETEC BroadWorks Provisioning MOP for Release 14 SP1 to build SIP Trunk services with following additions. (See References)

1. Device Profiles

   Use Template 6 device type template when creating Device Profile under Group. Authentication is required.

2. User Services Assignment

   Additional user services may need to be assigned due to workaround that BroadWorks has to provide, see section 4.2 Interoperability Issues for detail.

   Anonymous Call Rejection

   DND

8 Cisco Unified Communications 500 Series Configuration

UC500 comes with a default configuration; load the default configuration if it's not already loaded. Cisco Configuration Assistant was used to configure the UC500 under test initially with some customization to meet the specific PAETEC requirements, such as outbound proxy. The following describes the basic configuration of UC500 necessary for interoperating with PAETEC BroadWorks based SIP Trunking services.

1. Configure Network Settings

   Configure Ethernet port and POE ports, the IP addresses used are for illustration purpose only, the actual IP addresses can vary.

```
interface FastEthernet0/0
 description $FW_OUTSIDE$
 ip address 10.0.1.11 255.255.255.0
 ip nat outside
 ip virtual-reassembly
 duplex auto
 speed auto
```

```
interface FastEthernet0/1/0
 switchport voice vlan 100
 macro description cisco-phone
.
.
interface FastEthernet0/1/7
 switchport voice vlan 100
 macro description cisco-phone
```

2. Enable VLAN and IP Routing on POE Ports, the IP addresses used are for illustration purpose only, the actual IP addresses can vary

```
interface Vlan1
 no ip address
 bridge-group 1
 bridge-group 1 spanning-disabled
!
interface Vlan100
 no ip address
 bridge-group 100
 bridge-group 100 spanning-disabled
!
interface BVI1
 description $FW_INSIDE$
 ip address 192.168.10.1 255.255.255.0
 ip access-group 102 in
 ip nat inside
 ip virtual-reassembly
!
interface BVI100
 description $FW_INSIDE$
 ip address 10.1.1.1 255.255.255.0
 ip access-group 103 in
 ip nat inside
 ip virtual-reassembly

bridge irb
bridge 1 route ip
bridge 100 route ip
```

3. Configure DHCP, IP address pool may vary depending on the IP addressing scheme each company deploys.

```
ip dhcp relay information trust-all
ip dhcp use vrf connected
ip dhcp excluded-address 10.1.1.1 10.1.1.10
ip dhcp excluded-address 192.168.10.1 192.168.10.10
!
ip dhcp pool phone
  network 10.1.1.0 255.255.255.0
  default-router 10.1.1.1
  option 150 ip 10.1.1.1
!
ip dhcp pool data
  network 192.168.10.0 255.255.255.0
  default-router 192.168.10.1
  dns-server 205.152.0.20
```

4. Configure Quality of Service

   This is to ensure the voice traffic gets top priority over any other traffic for the benefit of voice quality.

```
class-map match-any voip
          match  dscp ef
          match precedence 5
          match protocol sip
          match protocol mgcp
          match protocol skinny

     policy-map voip
          class voip
          priority percent 90
          set  dscp 46
          class class-default

     interface fastethernet0/1 or interface serial1/0:0
          service-policy output voip
```

5. Configure DNS servers

```
ip name-server 209.253.113.2
ip name-server 209.253.113.10
```

6. Configure SIP Proxy, Registrar, MWI server, and Authentication

Set SIP proxy, registrar servers and MWI server, refer to the IPAC for actual server names or IP addresses that should be used for your accounts. Configure MWI server if voice mail service is ordered from PAETEC. Authentication needs to be configured under sip-ua.

```
sip-ua
 authentication username "USERNAME" password "PASSWORD"
 registrar dns: BWAS.global.voip.mcleodusa
 sip-server dns: BWAS.global.voip.mcleodusa
 mwi-server ipv4:64.199.64.220 expires 3600 port 5060 transport
     udp unsolicited
 host-registrar
  g729-annexb override
```

7. Configure Outbound Proxy and Global Voice Service Settings

Set outbound proxy; refer to the IPAC for the actual IP address of outbound proxy assigned to your accounts.

```
voice service voip
 allow-connections h323 to h323
 allow-connections h323 to sip
 allow-connections sip to h323
 allow-connections sip to sip
 no supplementary-service sip moved-temporarily
 no supplementary-service sip refer
 sip
  registrar server expires max 3600 min 3600
   localhost dns:englab1.mcleodusa.net
   outbound-proxy ipv4:64.199.64.21
```

8. Configure Codec's

This is to set g729r8 as the default codec for voice calls.

```
voice class codec 1
 codec preference 1 g729r8
 codec preference 2 g711ulaw
```

9. Configure Trunk Registration

This is to enable bulk registration by registering trunk group ID, it's not necessary to register all DID's Refer to the IPAC for actual trunk group ID number.

```
ephone-dn  53
 number 7133435377
 description SIP Trunk Registration
 preference 10
```

10. Configure Translation Rules and Profiles

Most of the following translations were added in by Cisco Configuration Assistant, they may need to be altered according to the dial plans deployed in your companies.

```
voice translation-rule 6
 rule 1 /7133434377/ /201/
 rule 2 /7133434378/ /202/
!
voice translation-rule 410
 rule 1 /^9\(.......)\)$/ /713\1/
 rule 2 /204/ /7133434380/
 rule 3 /203/ /7133434379/
 rule 4 /^9\(.*\)/ /\1/
 rule 5 /^...$/ /7133434376/
!
voice translation-rule 1111
 rule 1 /201/ /7133434377/
 rule 2 /202/ /7133434378/
 rule 3 /^...$/ /7133434376/
!
voice translation-rule 1112
 rule 1 /^9/ //
!
voice translation-rule 2000
 rule 1 /7133434380/ /204/
!
voice translation-rule 2001
 rule 1 /7133434379/ /203/
!
voice translation-rule 2222
!
!
voice translation-profile AA_Profile
 translate called 2001
!
voice translation-profile
CALLER_ID_TRANSLATION_PROFILE
 translate calling 1111
!
voice translation-profile CallBlocking
 translate called 2222
!
voice translation-profile
OUTGOING_TRANSLATION_PROFILE
 translate calling 1111
 translate called 1112
!
voice translation-profile PAETEC_Called_6
 translate called 6
!
voice translation-profile PSTN_CallForwarding
 translate redirect-target 410
 translate redirect-called 410
!
```

```
voice translation-profile PSTN_Outgoing
 translate calling 1111
 translate called 1112
 translate redirect-target 410
 translate redirect-called 410
!
voice translation-profile VM_Profile
 translate called 2000
```

11. Configure SIP Trunk dial peers

```
dial-peer voice 1000 voip
 description ** Incoming call from SIP trunk **
 voice-class codec 1
 voice-class sip dtmf-relay force rtp-nte
 session protocol sipv2
 session target sip-server
 incoming called-number .%
 dtmf-relay rtp-nte
 ip qos dscp cs5 media
 ip qos dscp cs4 signaling
 no vad
!
dial-peer voice 1001 voip
 description ** Outgoing call to SIP trunk (Generic SIP
Trunk Provider) **
 translation-profile outgoing PSTN_Outgoing
 destination-pattern 9T
 voice-class codec 1
 voice-class sip dtmf-relay force rtp-nte
 session protocol sipv2
 session target sip-server
 dtmf-relay rtp-nte
 ip qos dscp cs5 media
 ip qos dscp cs4 signaling
 no vad
!
dial-peer voice 1002 voip
 corlist outgoing call-local
 description ** star code to SIP trunk **
 destination-pattern *..
 voice-class codec 1
 voice-class sip dtmf-relay force rtp-nte
 session protocol sipv2
 session target sip-server
 dtmf-relay rtp-nte
 ip qos dscp cs5 media
 ip qos dscp cs4 signaling
 no vad
```

```
dial-peer voice 3002 voip
 description PAETEC
 translation-profile incoming PAETEC_Called_6
 voice-class codec 1
 voice-class sip dtmf-relay force rtp-nte
 session protocol sipv2
 session target sip-server
 incoming called-number 713343437[7-8]
 dtmf-relay rtp-nte
 ip qos dscp cs5 media
 ip qos dscp cs4 signaling
 no vad
```

12. Configure SIP Signaling Parameter

```
sip-ua
  no remote-party-id
  max-forwards 15
  retry invite 2
  retry response 3
  retry bye 3
  retry prack 6
  timers expires 300000
```

13. Configure Telephony Services

The actual phone firmware version, IP addresses and phone numbers may be different than what are in the example below, refer to the IPAC and Cisco document for the actual settings.

```
telephony-service
 video
 load 7960-7940 P00308000500
 load 7914 S00105000200
 load 7902 CP7902080002SCCP060817A
 load 7921 CP7921G-1.0.1
 load 7931 SCCP31.8-2-2SR2S
 load 7941GE SCCP41.8-2-2SR2S
 load 7941 SCCP41.8-2-2SR2S
 load 7961GE SCCP41.8-2-2SR2S
 load 7961 SCCP41.8-2-2SR2S
 load 7975 SCCP75.8-3-2S
 load 7965 SCCP45.8-3-2S
 load 7945 SCCP45.8-3-2S
 load 7942 SCCP42.8-3-2S
 load 7962 SCCP42.8-3-2S
 ..
..
```

```
max-ephones 14
 max-dn 56
 ip source-address 10.1.10.2 port 2000
 auto assign 10 to 19
 auto assign 5 to 8 type anl
 calling-number initiator
 url services http://10.1.10.1/voiceview/common/login.do
 url authentication http://10.1.10.1/voiceview/authentication/authenticate.do
 time-zone 5
 mwi relay
 max-conferences 8 gain -6
 call-forward pattern .T
 call-forward system redirecting-expanded
 moh music-on-hold.au
 multicast moh 239.10.16.16 port 2000
 web admin system name cisco secret XXXXX
 dn-webedit
 time-webedit
 transfer-system full-consult dss
 transfer-pattern 9.T
 transfer-pattern .T
 secondary-dialtone 9
 create cnf-files
```

14. Configure SIP Phone Users

    Add ephone-dn’s and ephones for IP phones.

```
ephone-dn  10  dual-line
 number 201 secondary 7133434377 no-reg both
 label 201
 call-forward busy 204
 call-forward noan 204 timeout 10
!
ephone-dn  11  dual-line
 number 202 secondary 7133434378 no-reg both
 label 202
 call-forward busy 204
 call-forward noan 204 timeout 10

ephone  5
 device-security-mode none
 mac-address 000B.BEF9.E718
 username "IPPhone1"
 type 7960
 keep-conference
 button  1:10
!
!ephone  6
 device-security-mode none
 mac-address 000B.BEF9.E678
 username "IPPhone2"
 type 7960
 button  1:11
```

15. Configure Voice Mail and Auto Attendant

```
interface FastEthernet0/0
 description $FW_OUTSIDE$
 ip address 10.0.1.11 255.255.255.0
 ip nat outside
 ip inspect SDM_LOW out
 ip virtual-reassembly
 duplex auto
 speed auto
!
interface Integrated-Service-Engine0/0
 description cue is initialized with default IMAP group$FW_INSIDE$
 ip unnumbered Loopback0
 ip nat inside
 ip virtual-reassembly
 service-module ip address 10.1.10.1 255.255.255.252
 service-module ip default-gateway 10.1.10.2

dial-peer voice 2000 voip
 description ** cue voicemail pilot number **
 destination-pattern 204
 b2bua
 voice-class sip outbound-proxy  ipv4:10.1.10.1
 session protocol sipv2
 session target ipv4:10.1.10.1
 dtmf-relay sip-notify
 codec g711ulaw
 no vad

dial-peer voice 2001 voip
 description ** cue auto attendant number **
 translation-profile outgoing PSTN_CallForwarding
 destination-pattern 203
 b2bua
 voice-class sip outbound-proxy  ipv4:10.1.10.1
 session protocol sipv2
 session target ipv4:10.1.10.1
 dtmf-relay sip-notify
 codec g711ulaw
 no vad
```

16. Configure CUE Service Module for Voice Mail and Auto Attendant

In addition to the configuration above on the UC500, the following needs to be configured on the CUE service module. The IP address and phone numbers are just the examples.

```
ntp server 10.1.10.2 prefer

software download server url "ftp://127.0.0.1/ftp" credentials hidden
"6u/dKTN/hsEuSAEfw40XlF2eFHnZfyUTSd8ZZNgd+Y9J3xlk2B35j0nfGW
TYHfmPSd8ZZNgd+Y9J3xlk2B35j0nfGWTYHfmPSd8ZZNgd+Y9J3xlk2B
35j0nfGWTYHfmP"

groupname Administrators create
groupname Broadcasters create
groupname IMAPgrp create

username cisco create
username u3 create
username u4 create
username u5 create
username u6 create
username FXS1 create
username FXS2 create
username FXS3 create
username FXS4 create
username IPPhone1 create
username IPPhone2 create
username u1 create
username u2 create
username IPPhone1 phonenumberE164 "7133434377"
username IPPhone2 phonenumberE164 "7133434378"
username FXS1 phonenumber "301"
username FXS2 phonenumber "302"
username FXS3 phonenumber "303"
username FXS4 phonenumber "304"
username IPPhone1 phonenumber "201"
username IPPhone2 phonenumber "202"
username IPPhone1 phonenumber "201" phonenumberE164
"7133434377"
username IPPhone2 phonenumber "202" phonenumberE164
"7133434378"
groupname Administrators member cisco
groupname IMAPgrp member u3
groupname IMAPgrp member u4
groupname IMAPgrp member u5
groupname IMAPgrp member u6
groupname IMAPgrp member FXS1
groupname IMAPgrp member FXS2
groupname IMAPgrp member FXS3
groupname IMAPgrp member FXS4
groupname IMAPgrp member IPPhone1
groupname IMAPgrp member IPPhone2
groupname IMAPgrp member u1
groupname IMAPgrp member u2
```

```
groupname Administrators privilege ManagePrompts
groupname Administrators privilege broadcast
groupname Administrators privilege local-broadcast
groupname Administrators privilege ManagePublicList
groupname Administrators privilege ViewPrivateList
groupname Administrators privilege vm-imap
groupname Administrators privilege superuser
groupname Broadcasters privilege broadcast
groupname IMAPgrp privilege vm-imap

restriction msg-notification min-digits 1
restriction msg-notification max-digits 30
restriction msg-notification dial-string preference 1 pattern * allowed

calendar biz-schedule systemschedule
 open day 1 from 00:00 to 24:00
 open day 2 from 00:00 to 24:00
 open day 3 from 00:00 to 24:00
 open day 4 from 00:00 to 24:00
 open day 5 from 00:00 to 24:00
 open day 6 from 00:00 to 24:00
 open day 7 from 00:00 to 24:00
 end schedule

ccn application autoattendant
 description "autoattendant"
 enabled
 maxsessions 6
 script "aa.aef"
 parameter "busClosedPrompt" "AABusinessClosed.wav"
 parameter "holidayPrompt" "AAHolidayPrompt.wav"
 parameter "welcomePrompt" "AAWelcome.wav"
 parameter "disconnectAfterMenu" "false"
 parameter "allowExternalTransfers" "false"
 parameter "MaxRetry" "3"
 parameter "busOpenPrompt" "AABusinessOpen.wav"
 parameter "businessSchedule" "systemschedule"
 parameter "operExtn" ""
 end application

ccn application ciscomwiapplication
 description "ciscomwiapplication"
 enabled
 maxsessions 6
 script "setmwi.aef"
 parameter "CallControlGroupID" "0"
 parameter "strMWI_OFF_DN" "A801"
 parameter "strMWI_ON_DN" "A800"
 end application
```

```
ccn application msgnotification
 description "msgnotification"
 enabled
 maxsessions 6
 script "msgnotify.aef"
 parameter "logoutUri"
"http://localhost/voicemail/vxmlscripts/mbxLogout.jsp"
 parameter "DelayBeforeSendDTMF" "1"
 end application

ccn application promptmgmt
 description "promptmgmt"
 enabled
 maxsessions 1
 script "promptmgmt.aef"
 end application

ccn application voicemail
 description "voicemail"
 enabled
 maxsessions 6
 script "voicebrowser.aef"
 parameter "logoutUri"
"http://localhost/voicemail/vxmlscripts/mbxLogout.jsp"
 parameter "uri" "http://localhost/voicemail/vxmlscripts/login.vxml"
 end application

ccn engine
 end engine

ccn subsystem jtapi
 ccm-manager address 0.0.0.0
 end subsystem

ccn subsystem sip
 gateway address "10.1.10.2"
 dtmf-relay sip-notify
 end subsystem

ccn trigger sip phonenumber 203
 application "autoattendant"
 enabled
 maxsessions 6
 end trigger

ccn trigger sip phonenumber 204
 application "voicemail"
 enabled
 maxsessions 6
 end trigger
```

```
voicemail mailbox owner "FXS2" size 775
 end mailbox

voicemail mailbox owner "FXS3" size 775
 end mailbox

voicemail mailbox owner "FXS4" size 775
 end mailbox

voicemail mailbox owner "IPPhone1" size 775
 expiration time 32767
 end mailbox

voicemail mailbox owner "IPPhone2" size 775
 expiration time 32767
 end mailbox
```

17. Configure Ad-hoc Conference and Transcoding

If ad-hoc conference and transcoding is needed, the following is the basic configuration to enable the service.

```
voice class custom-cptone CCAleavetone
 dualtone conference
  frequency 400 800
  cadence 400 50 200 50 200 50
!
voice class custom-cptone CCAjointone
 dualtone conference
  frequency 600 900
  cadence 300 150 300 100 300 50

  voice-card 0
   dspfarm
   dsp services dspfarm

  sccp local Loopback0
  sccp ccm 10.1.1.1 identifier 1 version 4.1
  sccp
  !
  sccp ccm group 1
   associate ccm 1 priority 1
   associate profile 1 register confprof1
   associate profile 2 register tranprof2
```

```
dspfarm profile 2 transcode
 codec g711ulaw
 codec g711alaw
 codec ilbc
 codec g723r63
 codec g723r53
 codec gsmamr-nb
 codec g729ar8
 codec g729abr8
 codec g729r8
 codec g729br8
 maximum sessions 1
 associate application SCCP
!
dspfarm profile 1 conference
 description DO NOT MODIFY, active CCA conference profile- Using
codec729
 codec g711ulaw
 codec g711alaw
 codec g729ar8
 codec g729abr8
 codec g729r8
 codec g729br8
 maximum sessions 2
 conference-join custom-cptone CCAjointone
 conference-leave custom-cptone CCAleavetone
 associate application SCCP

ephone-dn  41  dual-line
 number C002 no-reg primary
 conference ad-hoc
 preference 1
!
!
ephone-dn  42  dual-line
 number C002 no-reg primary
 conference ad-hoc
 no huntstop
!
!
ephone-dn  43  dual-line
 number C001 no-reg primary
 conference ad-hoc
 preference 1
 !
ephone-dn  44  dual-line
 number C001 no-reg primary
 conference ad-hoc
 no huntstop

telephony-service
 sdspfarm units 4
 sdspfarm transcode sessions 24
 sdspfarm tag 1 confprof1
 sdspfarm tag 2 tranprof2
```

Appendix A: Sample Configuration Files

The following are example files and should be used for reference only.

```
parser config cache interface
parser config interface
no service pad
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
service internal
service compress-config
!
hostname UC520
!
boot-start-marker
boot system flash uc500-advipservicesk9-mz.124-11.XW6
boot-end-marker
!
logging buffered 256000
enable secret 5 $1$NNyJ$xPw59qrOgmXzHCFHbmTtj0
!
aaa new-model
!
!
aaa authentication login default local
aaa authorization exec default local
!
!
aaa session-id common
clock timezone CST -6
clock summer-time CDT recurring
!
ip cef
!
!
ip dhcp relay information trust-all
ip dhcp use vrf connected
ip dhcp excluded-address 10.1.1.1 10.1.1.10
ip dhcp excluded-address 192.168.10.1 192.168.10.10
!
ip dhcp pool phone
   network 10.1.1.0 255.255.255.0
   default-router 10.1.1.1
   option 150 ip 10.1.1.1
!
ip dhcp pool data
   network 192.168.10.0 255.255.255.0
   default-router 192.168.10.1
   dns-server 205.152.0.20
!
!
ip domain name englab1.mcleodusa.net
ip name-server 209.253.113.10
ip inspect name SDM_LOW cuseeme
ip inspect name SDM_LOW dns
ip inspect name SDM_LOW ftp
ip inspect name SDM_LOW h323
ip inspect name SDM_LOW https
ip inspect name SDM_LOW icmp
ip inspect name SDM_LOW imap
```

```
ip inspect name SDM_LOW pop3
ip inspect name SDM_LOW netshow
ip inspect name SDM_LOW rcmd
ip inspect name SDM_LOW realaudio
ip inspect name SDM_LOW rtsp
ip inspect name SDM_LOW esmtp
ip inspect name SDM_LOW sqlnet
ip inspect name SDM_LOW streamworks
ip inspect name SDM_LOW tftp
ip inspect name SDM_LOW tcp
ip inspect name SDM_LOW udp router-traffic timeout 300
ip inspect name SDM_LOW vdolive
!
stcapp ccm-group 1
stcapp
!
stcapp feature access-code
!
multilink bundle-name authenticated
!
voice service voip
 allow-connections h323 to h323
 allow-connections h323 to sip
 allow-connections sip to h323
 allow-connections sip to sip
 no supplementary-service sip moved-temporarily
 no supplementary-service sip refer
 sip
  registrar server expires max 3600 min 3600
   localhost dns:englab1.mcleodusa.net
   outbound-proxy ipv4:64.199.64.21
  no update-callerid
!
!
voice class codec 1
 codec preference 1 g729r8
 codec preference 2 g711ulaw
!
voice class custom-cptone CCAleavetone
 dualtone conference
  frequency 400 800
  cadence 400 50 200 50 200 50
!
!
voice class custom-cptone CCAjointone
 dualtone conference
  frequency 600 900
  cadence 300 150 300 100 300 50
!
voice translation-rule 6
 rule 1 /7133434377/ /201/
 rule 2 /7133434378/ /202/
!
voice translation-rule 410
 rule 1 /^9\(.......)\)$/ /713\1/
 rule 2 /204/ /7133434380/
 rule 3 /203/ /7133434379/
 rule 4 /^9\(.*\)/ /\1/
 rule 5 /^...$/ /7133434376/
!
voice translation-rule 1111
 rule 1 /201/ /7133434377/
 rule 2 /202/ /7133434378/
```

```
 rule 3 /^...$/ /7133434376/
!
voice translation-rule 1112
 rule 1 /^9/ //
!
voice translation-rule 2000
 rule 1 /7133434380/ /204/
!
voice translation-rule 2001
 rule 1 /7133434379/ /203/
!
voice translation-rule 2222
!
!
voice translation-profile AA_Profile
 translate called 2001
!
voice translation-profile CALLER_ID_TRANSLATION_PROFILE
 translate calling 1111
!
voice translation-profile CallBlocking
 translate called 2222
!
voice translation-profile OUTGOING_TRANSLATION_PROFILE
 translate calling 1111
 translate called 1112
!
voice translation-profile PAETEC_Called_6
 translate called 6
!
voice translation-profile PSTN_CallForwarding
 translate redirect-target 410
 translate redirect-called 410
!
voice translation-profile PSTN_Outgoing
 translate calling 1111
 translate called 1112
 translate redirect-target 410
 translate redirect-called 410
!
voice translation-profile VM_Profile
 translate called 2000
!
!
voice-card 0
 dspfarm
 dsp services dspfarm
!
username cisco privilege 15 secret 5 $1$PSlT$BuHoSkLLbIf8VH21ajH801
archive
 log config
  hidekeys
!
bridge irb
!
interface Loopback0
 description $FW_INSIDE$
 ip address 10.1.10.2 255.255.255.252
 ip access-group 101 in
 ip nat inside
 ip virtual-reassembly
!
interface FastEthernet0/0
```

```
 description $FW_OUTSIDE$
 ip address 10.0.1.11 255.255.255.0
 ip nat outside
 ip inspect SDM_LOW out
 ip virtual-reassembly
 duplex auto
 speed auto
!
interface Integrated-Service-Engine0/0
 description cue is initialized with default IMAP group$FW_INSIDE$
 ip unnumbered Loopback0
 ip nat inside
 ip virtual-reassembly
 service-module ip address 10.1.10.1 255.255.255.252
 service-module ip default-gateway 10.1.10.2
!
interface FastEthernet0/1/0
 switchport voice vlan 100
 macro description cisco-phone
!
interface FastEthernet0/1/1
 switchport voice vlan 100
 macro description cisco-phone
!
interface FastEthernet0/1/2
 switchport voice vlan 100
 macro description cisco-phone
!
interface FastEthernet0/1/3
 switchport voice vlan 100
 macro description cisco-phone
!
interface FastEthernet0/1/4
 switchport voice vlan 100
 macro description cisco-phone
!
interface FastEthernet0/1/5
 switchport voice vlan 100
 macro description cisco-phone
!
interface FastEthernet0/1/6
 switchport voice vlan 100
 macro description cisco-phone
!
interface FastEthernet0/1/7
 switchport voice vlan 100
 macro description cisco-phone
!
interface FastEthernet0/1/8
 switchport mode trunk
 macro description cisco-switch
!
interface Vlan1
 no ip address
 bridge-group 1
 bridge-group 1 spanning-disabled
!
interface Vlan100
 no ip address
 bridge-group 100
 bridge-group 100 spanning-disabled
!
interface BVI1
```

```
 description $FW_INSIDE$
 ip address 192.168.10.1 255.255.255.0
 ip access-group 102 in
 ip nat inside
 ip virtual-reassembly
!
interface BVI100
 description $FW_INSIDE$
 ip address 10.1.1.1 255.255.255.0
 ip access-group 103 in
 ip nat inside
 ip virtual-reassembly
!
ip route 0.0.0.0 0.0.0.0 10.0.1.1
ip route 10.1.10.1 255.255.255.255 Integrated-Service-Engine0/0
!
ip http server
ip http authentication local
ip http secure-server
ip http path flash:
ip nat inside source list 1 interface FastEthernet0/0 overload
!
access-list 1 remark SDM_ACL Category=2
access-list 1 permit 10.1.1.0 0.0.0.255
access-list 1 permit 192.168.10.0 0.0.0.255
access-list 1 permit 10.1.10.0 0.0.0.3
access-list 100 remark auto generated by SDM firewall configuration
access-list 100 remark SDM_ACL Category=1
access-list 100 deny   ip 10.1.1.0 0.0.0.255 any
access-list 100 deny   ip 192.168.10.0 0.0.0.255 any
access-list 100 deny   ip 10.0.1.0 0.0.0.255 any
access-list 100 deny   ip host 255.255.255.255 any
access-list 100 deny   ip 127.0.0.0 0.255.255.255 any
access-list 100 permit ip any any
access-list 101 remark auto generated by SDM firewall configuration
access-list 101 remark SDM_ACL Category=1
access-list 101 permit tcp 10.1.1.0 0.0.0.255 eq 2000 any
access-list 101 permit udp 10.1.1.0 0.0.0.255 eq 2000 any
access-list 101 deny   ip 10.1.1.0 0.0.0.255 any
access-list 101 deny   ip 192.168.10.0 0.0.0.255 any
access-list 101 deny   ip 10.0.1.0 0.0.0.255 any
access-list 101 deny   ip host 255.255.255.255 any
access-list 101 deny   ip 127.0.0.0 0.255.255.255 any
access-list 101 permit ip any any
access-list 102 remark auto generated by SDM firewall configuration
access-list 102 remark SDM_ACL Category=1
access-list 102 deny   ip 10.1.10.0 0.0.0.3 any
access-list 102 deny   ip 10.1.1.0 0.0.0.255 any
access-list 102 deny   ip 10.0.1.0 0.0.0.255 any
access-list 102 deny   ip host 255.255.255.255 any
access-list 102 deny   ip 127.0.0.0 0.255.255.255 any
access-list 102 permit ip any any
access-list 103 remark auto generated by SDM firewall configuration
access-list 103 remark SDM_ACL Category=1
access-list 103 permit tcp 10.1.10.0 0.0.0.3 any eq 2000
access-list 103 permit udp 10.1.10.0 0.0.0.3 any eq 2000
access-list 103 deny   ip 10.1.10.0 0.0.0.3 any
access-list 103 deny   ip 192.168.10.0 0.0.0.255 any
access-list 103 deny   ip 10.0.1.0 0.0.0.255 any
access-list 103 deny   ip host 255.255.255.255 any
access-list 103 deny   ip 127.0.0.0 0.255.255.255 any
access-list 103 permit ip any any
access-list 104 remark auto generated by SDM firewall configuration
```

```
access-list 104 remark SDM_ACL Category=1
access-list 104 deny   ip 10.1.10.0 0.0.0.3 any
access-list 104 deny   ip 10.1.1.0 0.0.0.255 any
access-list 104 deny   ip 192.168.10.0 0.0.0.255 any
access-list 104 permit icmp any host 10.0.1.11 echo-reply
access-list 104 permit icmp any host 10.0.1.11 time-exceeded
access-list 104 permit icmp any host 10.0.1.11 unreachable
access-list 104 permit udp any any eq 5060
access-list 104 permit udp any eq 5060 any
access-list 104 permit udp any any range 16384 32767
access-list 104 permit udp host 205.152.0.20 eq domain any
access-list 104 permit udp host 209.253.113.10 eq domain any
access-list 104 deny   ip 10.0.0.0 0.255.255.255 any
access-list 104 deny   ip 172.16.0.0 0.15.255.255 any
access-list 104 deny   ip 192.168.0.0 0.0.255.255 any
access-list 104 deny   ip 127.0.0.0 0.255.255.255 any
access-list 104 deny   ip host 255.255.255.255 any
access-list 104 deny   ip host 0.0.0.0 any
access-list 104 deny   ip any any log
snmp-server community public RO
!
!
tftp-server flash:cmterm-7941-7961-sccp.7.0.3.tar
tftp-server flash:APPS-1.0.1.SBN
tftp-server flash:CP7921G-1.0.1.LOADS
tftp-server flash:GUI-1.0.1.SBN
tftp-server flash:SYS-1.0.1.SBN
tftp-server flash:dsp11.8-2-2TR2.sbn
tftp-server flash:cvm11sccp.8-2-2TR2.sbn
tftp-server flash:apps11.8-2-2TR2.sbn
tftp-server flash:jar11sccp.8-2-2TR2.sbn
tftp-server flash:cnu11.8-2-2TR2.sbn
tftp-server flash:SCCP11.8-2-2SR2S.loads
tftp-server flash:term06.default.loads
tftp-server flash:term11.default.loads
tftp-server flash:SCCP70.8-2-2SR2S.loads
tftp-server flash:apps70.8-2-2TR2.sbn
tftp-server flash:term71.default.loads
tftp-server flash:dsp70.8-2-2TR2.sbn
tftp-server flash:term70.default.loads
tftp-server flash:cvm70sccp.8-2-2TR2.sbn
tftp-server flash:cnu70.8-2-2TR2.sbn
tftp-server flash:jar70sccp.8-2-2TR2.sbn
tftp-server flash:dsp41.8-2-2TR2.sbn
tftp-server flash:SCCP41.8-2-2SR2S.loads
tftp-server flash:term41.default.loads
tftp-server flash:cvm41sccp.8-2-2TR2.sbn
tftp-server flash:cnu41.8-2-2TR2.sbn
tftp-server flash:term61.default.loads
tftp-server flash:apps41.8-2-2TR2.sbn
tftp-server flash:jar41sccp.8-2-2TR2.sbn
tftp-server flash:CP7902080002SCCP060817A.sbin
tftp-server flash:S00105000200.sbn
tftp-server flash:cmterm_7936.3-3-13-0.bin
tftp-server flash:term31.default.loads
tftp-server flash:dsp31.8-2-2TR2.sbn
tftp-server flash:cvm31sccp.8-2-2TR2.sbn
tftp-server flash:apps31.8-2-2TR2.sbn
tftp-server flash:cnu31.8-2-2TR2.sbn
tftp-server flash:jar31sccp.8-2-2TR2.sbn
tftp-server flash:SCCP31.8-2-2SR2S.loads
tftp-server flash:P00308000500.sbn
tftp-server flash:P00308000500.bin
```

```
tftp-server flash:P00308000500.sb2
tftp-server flash:P00308000500.loads
tftp-server flash:term62.default.loads
tftp-server flash:term42.default.loads
tftp-server flash:cnu42.8-3-1-22.sbn
tftp-server flash:cvm42sccp.8-3-1-22.sbn
tftp-server flash:dsp42.8-3-1-22.sbn
tftp-server flash:jar42sccp.8-3-1-22.sbn
tftp-server flash:SCCP42.8-3-2S.loads
tftp-server flash:apps42.8-3-1-22.sbn
tftp-server flash:cvm45sccp.8-3-1-22.sbn
tftp-server flash:apps45.8-3-1-22.sbn
tftp-server flash:dsp45.8-3-1-22.sbn
tftp-server flash:term45.default.loads
tftp-server flash:cnu45.8-3-1-22.sbn
tftp-server flash:term65.default.loads
tftp-server flash:jar45sccp.8-3-1-22.sbn
tftp-server flash:SCCP45.8-3-2S.loads
tftp-server flash:dsp75.8-3-1-22.sbn
tftp-server flash:term75.default.loads
tftp-server flash:jar75sccp.8-3-1-22.sbn
tftp-server flash:cnu75.8-3-1-22.sbn
tftp-server flash:apps75.8-3-1-22.sbn
tftp-server flash:SCCP75.8-3-2S.loads
tftp-server flash:cvm75sccp.8-3-1-22.sbn
!
control-plane
!
bridge 1 route ip
bridge 100 route ip
!
!
voice-port 0/0/0
 timeouts ringing infinity
!
voice-port 0/0/1
 timeouts ringing infinity
!
voice-port 0/0/2
 timeouts ringing infinity
!
voice-port 0/0/3
 timeouts ringing infinity
!
voice-port 0/1/0
!
voice-port 0/1/1
!
voice-port 0/1/2
!
voice-port 0/1/3
!
voice-port 0/4/0
 auto-cut-through
 signal immediate
 input gain auto-control -15
 description Music On Hold Port
!
sccp local Loopback0
sccp ccm 10.1.1.1 identifier 1 version 4.1
sccp
!
sccp ccm group 1
```

```
 associate ccm 1 priority 1
 associate profile 1 register confprof1
 associate profile 2 register tranprof2
!
dspfarm profile 2 transcode
 codec g711ulaw
 codec g711alaw
 codec ilbc
 codec g723r63
 codec g723r53
 codec gsmamr-nb
 codec g729ar8
 codec g729abr8
 codec g729r8
 codec g729br8
 maximum sessions 1
 associate application SCCP
!
dspfarm profile 1 conference
 description DO NOT MODIFY, active CCA conference profile- Using codec729
 codec g711ulaw
 codec g711alaw
 codec g729ar8
 codec g729abr8
 codec g729r8
 codec g729br8
 maximum sessions 2
 conference-join custom-cptone CCAjointone
 conference-leave custom-cptone CCAleavetone
 associate application SCCP
!
dial-peer cor custom
 name internal
 name local
 name domestic
 name international
!
!
dial-peer cor list call-internal
 member internal
!
dial-peer cor list call-local
 member local
!
dial-peer cor list call-domestic
 member domestic
!
dial-peer cor list call-international
 member international
!
dial-peer cor list user-internal
 member internal
!
dial-peer cor list user-local
 member internal
 member local
!
dial-peer cor list user-domestic
 member internal
 member local
 member domestic
!
dial-peer cor list user-international
```

```
 member internal
 member local
 member domestic
 member international
!
!
dial-peer voice 1 pots
 service stcapp
 port 0/0/0
!
dial-peer voice 2 pots
 service stcapp
 port 0/0/1
!
dial-peer voice 3 pots
 service stcapp
 port 0/0/2
!
dial-peer voice 4 pots
 service stcapp
 port 0/0/3
!
dial-peer voice 5 pots
 description ** MOH Port **
 destination-pattern ABC
 port 0/4/0
 no sip-register
!
dial-peer voice 2000 voip
 description ** cue voicemail pilot number **
 destination-pattern 204
 b2bua
 voice-class sip outbound-proxy  ipv4:10.1.10.1
 session protocol sipv2
 session target ipv4:10.1.10.1
 dtmf-relay sip-notify
 codec g711ulaw
 no vad
!
dial-peer voice 2001 voip
 description ** cue auto attendant number **
 translation-profile outgoing PSTN_CallForwarding
 destination-pattern 203
 b2bua
 voice-class sip outbound-proxy  ipv4:10.1.10.1
 session protocol sipv2
 session target ipv4:10.1.10.1
 dtmf-relay sip-notify
 codec g711ulaw
 no vad
!
dial-peer voice 2002 voip
 description ** cue voicemail PSTN number **
 translation-profile outgoing VM_Profile
 destination-pattern 7133434380$
 b2bua
 voice-class sip outbound-proxy  ipv4:10.1.10.1
 session protocol sipv2
 session target ipv4:10.1.10.1
 dtmf-relay sip-notify
 codec g711ulaw
 no vad
!
```

```
dial-peer voice 2003 voip
 description ** cue auto attendant PSTN number **
 translation-profile outgoing AA_Profile
 destination-pattern 7133434379$
 b2bua
 voice-class sip outbound-proxy  ipv4:10.1.10.1
 session protocol sipv2
 session target ipv4:10.1.10.1
 dtmf-relay sip-notify
 codec g711ulaw
 no vad
!
dial-peer voice 1000 voip
 description ** Incoming call from SIP trunk **
 voice-class codec 1
 voice-class sip dtmf-relay force rtp-nte
 session protocol sipv2
 session target sip-server
 incoming called-number .%
 dtmf-relay rtp-nte
 ip qos dscp cs5 media
 ip qos dscp cs4 signaling
 no vad
!
dial-peer voice 1001 voip
 description ** Outgoing call to SIP trunk (Generic SIP Trunk Provider)
**
 translation-profile outgoing PSTN_Outgoing
 destination-pattern 9T
 voice-class codec 1
 voice-class sip dtmf-relay force rtp-nte
 session protocol sipv2
 session target sip-server
 dtmf-relay rtp-nte
 ip qos dscp cs5 media
 ip qos dscp cs4 signaling
 no vad
!
dial-peer voice 1002 voip
 corlist outgoing call-local
 description ** star code to SIP trunk **
 destination-pattern *..
 voice-class codec 1
 voice-class sip dtmf-relay force rtp-nte
 session protocol sipv2
 session target sip-server
 dtmf-relay rtp-nte
 ip qos dscp cs5 media
 ip qos dscp cs4 signaling
 no vad
!
dial-peer voice 1003 voip
 description ** AA from SIP Trunk **
 translation-profile incoming AA_Profile
 voice-class codec 1
 voice-class sip dtmf-relay force rtp-nte
 session protocol sipv2
 session target sip-server
 incoming called-number 7133434379
 dtmf-relay rtp-nte
 ip qos dscp cs5 media
 ip qos dscp cs4 signaling
 no vad
```

```
!
dial-peer voice 1004 voip
 description ** VM from SIP Trunk **
 translation-profile incoming VM_Profile
 voice-class codec 1
 voice-class sip dtmf-relay force rtp-nte
 session protocol sipv2
 session target sip-server
 incoming called-number 7133434380
 dtmf-relay rtp-nte
 ip qos dscp cs5 media
 ip qos dscp cs4 signaling
 no vad
!
dial-peer voice 3002 voip
 description PAETEC
 translation-profile incoming PAETEC_Called_6
 voice-class codec 1
 voice-class sip dtmf-relay force rtp-nte
 session protocol sipv2
 session target sip-server
 incoming called-number 713343437[7-8]
 dtmf-relay rtp-nte
 ip qos dscp cs5 media
 ip qos dscp cs4 signaling
 no vad
!
!
sip-ua
 authentication username 7133434376 password 7 135246415858577E78737E
 no remote-party-id
 retry invite 2
 retry register 10
 timers connect 100
 mwi-server ipv4:64.199.64.220 expires 3600 port 5060 transport udp
unsolicited
 registrar dns:englab1.mcleodusa.net expires 3600
 sip-server dns:englab1.mcleodusa.net
  host-registrar
 g729-annexb override
!
!
!
telephony-service
 sdspfarm units 4
 sdspfarm transcode sessions 24
 sdspfarm tag 1 confprof1
 sdspfarm tag 2 tranprof2
 video
 load 7960-7940 P00308000500
 load 7914 S00105000200
 load 7902 CP7902080002SCCP060817A
 load 7921 CP7921G-1.0.1
 load 7931 SCCP31.8-2-2SR2S
 load 7941GE SCCP41.8-2-2SR2S
 load 7941 SCCP41.8-2-2SR2S
 load 7961GE SCCP41.8-2-2SR2S
 load 7961 SCCP41.8-2-2SR2S
 load 7975 SCCP75.8-3-2S
 load 7965 SCCP45.8-3-2S
 load 7945 SCCP45.8-3-2S
 load 7942 SCCP42.8-3-2S
 load 7962 SCCP42.8-3-2S
```

```
 load 7971 SCCP70.8-2-2SR2S
 load 7970 SCCP70.8-2-2SR2S
 load 7936 cmterm_7936.3-3-13-0
 load 7906 SCCP11.8-2-2SR2S
 load 7911 SCCP11.8-2-2SR2S
 max-ephones 14
 max-dn 56
 ip source-address 10.1.10.2 port 2000
 auto assign 10 to 19
 auto assign 5 to 8 type anl
 calling-number initiator
 url services http://10.1.10.1/voiceview/common/login.do
 url authentication
http://10.1.10.1/voiceview/authentication/authenticate.do
 time-zone 5
 mwi relay
 max-conferences 8 gain -6
 call-forward pattern .T
 call-forward system redirecting-expanded
 moh music-on-hold.au
 multicast moh 239.10.16.16 port 2000
 web admin system name cisco secret 5 $1$LueE$BYPOgzzgSJk4ep/WZBDMN0
 dn-webedit
 time-webedit
 transfer-system full-consult dss
 transfer-pattern 9.T
 transfer-pattern .T
 secondary-dialtone 9
 create cnf-files version-stamp 7960 Apr 07 2008 22:17:59
!
ephone-dn  5  dual-line
!
ephone-dn  9
 number BCD no-reg primary
 description MoH
 moh ip 239.10.16.8 port 2139 out-call ABC
!
!
ephone-dn  10  dual-line
 number 201 secondary 7133434377 no-reg both
 label 201
 description Usr UC
 name Usr UC
 call-forward busy 204
 call-forward noan 204 timeout 10

!
!
ephone-dn  11  dual-line
 number 202 secondary 7133434378 no-reg both
 label 202
 description Usr UC
 name Usr UC
 call-forward busy 204
 call-forward noan 204 timeout 10
!
ephone-dn  41  dual-line
 number C002 no-reg primary
 conference ad-hoc
 preference 1
!
ephone-dn  42  dual-line
 number C002 no-reg primary
```

```
 conference ad-hoc
  no huntstop
!
!
ephone-dn  43  dual-line
 number C001 no-reg primary
 conference ad-hoc
 preference 1
!
ephone-dn  44  dual-line
 number C001 no-reg primary
 conference ad-hoc
 no huntstop
!
!
ephone-dn  53
 number 7133434376
 description SIP Trunk Registration
 preference 10
!
ephone-dn  55
 number A801... no-reg primary
 mwi off
!
!
ephone-dn  56
 number A800... no-reg primary
 mwi on
!
!
ephone  1
 device-security-mode none
 video
 mac-address B8FA.95EE.0000
 username "FXS1"
 type anl
 button  1:5
!
ephone  5
 device-security-mode none
 mac-address 000B.BEF9.E718
 username "IPPhone1"
 type 7960
 keep-conference
 button  1:10
!
ephone  6
 device-security-mode none
 mac-address 000B.BEF9.E678
 username "IPPhone2"
 type 7960
 button  1:11
!
line con 0
 no modem enable
line aux 0
line 2
 no activation-character
 no exec
 transport preferred none
 transport input all
line vty 5 100
!
```

```
ntp master
```

! CUE Service Module Sample Configuration

```
clock timezone America/Los_Angeles

hostname se-10-1-10-1.unspecified

ip domain-name (none)

ntp server 10.1.10.2 prefer

software download server url "ftp://127.0.0.1/ftp" credentials hidden
"6u/dKTN/hsEuSAEfw40XlF2eFHnZfyUTSd8ZZNgd+Y9J3xlk2B35j0nfGWTYHfmPSd8ZZNgd
+Y9J3xlk2B35j0nfGWTYHfmPSd8ZZNgd+Y9J3xlk2B35j0nfGWTYHfmP"

groupname Administrators create
groupname Broadcasters create
groupname IMAPgrp create

username cisco create
username u3 create
username u4 create
username u5 create
username u6 create
username FXS1 create
username FXS2 create
username FXS3 create
username FXS4 create
username IPPhone1 create
username IPPhone2 create
username u1 create
username u2 create
username IPPhone1 phonenumberE164 "7133434377"
username IPPhone2 phonenumberE164 "7133434378"
username FXS1 phonenumber "301"
username FXS2 phonenumber "302"
username FXS3 phonenumber "303"
username FXS4 phonenumber "304"
username IPPhone1 phonenumber "201"
username IPPhone2 phonenumber "202"
username IPPhone1 phonenumber "201" phonenumberE164 "7133434377"
username IPPhone2 phonenumber "202" phonenumberE164 "7133434378"

groupname Administrators member cisco
groupname IMAPgrp member u3
groupname IMAPgrp member u4
groupname IMAPgrp member u5
groupname IMAPgrp member u6
groupname IMAPgrp member FXS1
groupname IMAPgrp member FXS2
groupname IMAPgrp member FXS3
groupname IMAPgrp member FXS4
groupname IMAPgrp member IPPhone1
groupname IMAPgrp member IPPhone2
groupname IMAPgrp member u1
groupname IMAPgrp member u2
groupname Administrators privilege ManagePrompts
groupname Administrators privilege broadcast
groupname Administrators privilege local-broadcast
groupname Administrators privilege ManagePublicList
```

```
groupname Administrators privilege ViewPrivateList
groupname Administrators privilege vm-imap
groupname Administrators privilege superuser
groupname Broadcasters privilege broadcast
groupname IMAPgrp privilege vm-imap

restriction msg-notification min-digits 1
restriction msg-notification max-digits 30
restriction msg-notification dial-string preference 1 pattern * allowed

backup server url "ftp://127.0.0.1/ftp" credentials hidden
"EWlTygcMhYmjazXhE/VNXHCkplVV4KjescbDaLa4fl4WLSPFvv1rWUnfGWTYHfmPSd8ZZNgd
+Y9J3xlk2B35j0nfGWTYHfmPSd8ZZNgd+Y9J3xlk2B35j0nfGWTYHfmP"

calendar biz-schedule systemschedule
 open day 1 from 00:00 to 24:00
 open day 2 from 00:00 to 24:00
 open day 3 from 00:00 to 24:00
 open day 4 from 00:00 to 24:00
 open day 5 from 00:00 to 24:00
 open day 6 from 00:00 to 24:00
 open day 7 from 00:00 to 24:00
 end schedule

ccn application autoattendant
 description "autoattendant"
 enabled
 maxsessions 6
 script "aa.aef"
 parameter "busClosedPrompt" "AABusinessClosed.wav"
 parameter "holidayPrompt" "AAHolidayPrompt.wav"
 parameter "welcomePrompt" "AAWelcome.wav"
 parameter "disconnectAfterMenu" "false"
 parameter "allowExternalTransfers" "false"
 parameter "MaxRetry" "3"
 parameter "busOpenPrompt" "AABusinessOpen.wav"
 parameter "businessSchedule" "systemschedule"
 parameter "operExtn" ""
 end application

ccn application ciscomwiapplication
 description "ciscomwiapplication"
 enabled
 maxsessions 6
 script "setmwi.aef"
 parameter "CallControlGroupID" "0"
 parameter "strMWI_OFF_DN" "A801"
 parameter "strMWI_ON_DN" "A800"
 end application

ccn application msgnotification
 description "msgnotification"
 enabled
 maxsessions 6
 script "msgnotify.aef"
 parameter "logoutUri"
"http://localhost/voicemail/vxmlscripts/mbxLogout.jsp"
 parameter "DelayBeforeSendDTMF" "1"
 end application

ccn application promptmgmt
 description "promptmgmt"
 enabled
```

```
 maxsessions 1
 script "promptmgmt.aef"
 end application

ccn application voicemail
 description "voicemail"
 enabled
 maxsessions 6
 script "voicebrowser.aef"
 parameter "logoutUri"
"http://localhost/voicemail/vxmlscripts/mbxLogout.jsp"
 parameter "uri" "http://localhost/voicemail/vxmlscripts/login.vxml"
 end application

ccn engine
 end engine

ccn subsystem jtapi
 ccm-manager address 0.0.0.0
 end subsystem

ccn subsystem sip
 gateway address "10.1.10.2"
 dtmf-relay sip-notify
 end subsystem

ccn trigger sip phonenumber 203
 application "autoattendant"
 enabled
 maxsessions 6
 end trigger

ccn trigger sip phonenumber 204
 application "voicemail"
 enabled
 maxsessions 6
 end trigger

ccn trigger sip phonenumber 7133434380
 application "voicemail"
 enabled
 maxsessions 6
 end trigger

service imap
 enable
 end imap

service phone-authentication
 end phone-authentication

service voiceview
 enable
 end voiceview

voicemail callerid
voicemail default mailboxsize 775
voicemail broadcast recording time 300
voicemail mailbox owner "FXS1" size 775
 end mailbox

voicemail mailbox owner "FXS2" size 775
 end mailbox
```

```
voicemail mailbox owner "FXS3" size 775
 end mailbox

voicemail mailbox owner "FXS4" size 775
 end mailbox

voicemail mailbox owner "IPPhone1" size 775
 expiration time 32767
 end mailbox

voicemail mailbox owner "IPPhone2" size 775
 expiration time 32767
 end mailbox

voicemail mailbox owner "u1" size 775
 end mailbox

voicemail mailbox owner "u2" size 775
 end mailbox

voicemail mailbox owner "u3" size 775
 end mailbox

voicemail mailbox owner "u4" size 775
 end mailbox

voicemail mailbox owner "u5" size 775
 end mailbox

voicemail mailbox owner "u6" size 775
 end mailbox

end
```

Appendix B: Instructions for Enabling DND and Anonymous Rejection

There are two methods for enabling and disabling Do Not Disturb feature via the legacy McLeod SIP Trunking Service. The first method uses Feature Access Codes to enable and disable the features from the customer phone. The second method uses the legacy McLeod web portal to enable and disable the features.

The method to enable and disable Anonymous Call Reject is through legacy McLeod web portal.

1. Using Feature Access Codes (also known as * codes)

Requirement

- The user must have the ability to invoke the Call forwarding features in the legacy McLeod call controller. Permissions to use these features are controlled by the Enterprise and/or Group administrator.

- The user must dial the Feature Access Code (FAC) from their direct extension or IP phone. The customer IP PBX must be configured to pass the station level ANI in the SIP message to the PAETEC network. The PAETEC call controller uses the calling number to enable or disable the feature for the corresponding user.

Do Not Disturb Feature Access Codes

The table below displays the list of valid Feature Access Codes for the Call Forwarding options available as part of the standard service.

Feature	Enable Feature	Disable Feature
Do Not Disturb	*78	*79

- **Do Not Disturb** – This feature will automatically send calls to the user's voice mail if it's enabled

.

  - To enable this feature:
    1. Dial ***78**. Note: To eliminate post-dial delays, dial ***78#.**
    2. A message will be played as following to confirm the successful activation of the feature, "Your Do Not Disturb service has been activated successfully, thanks you"

  - To disable this feature:
    1. Dial ***79**. Note: To eliminate post-dial delays, dial ***79#**.
    2. The following announcement will play: "Your Do Not Disturb service has been deactivated successfully, thank you"

2. Using the Web Portal

a. Do Not Disturb Requirement

- The user must have the ability to invoke the Do Not Disturb features in the legacy McLeod call controller. Permissions to use these features are controlled by the Enterprise and/or Group administrator.

- The user must have the ability to log into the legacy McLeod web portal as a user. The Enterprise and/or Group administrator also controls the ability to access user information via the web portal. The web portal is accessible via the following link: http://www.mcleodusa.com.

After logging into the web portal as a user and entering the Manage Your Services and the Dynamic Integrated Access section of the account, the system will bring up the initial screen showing the profile screen for the user.

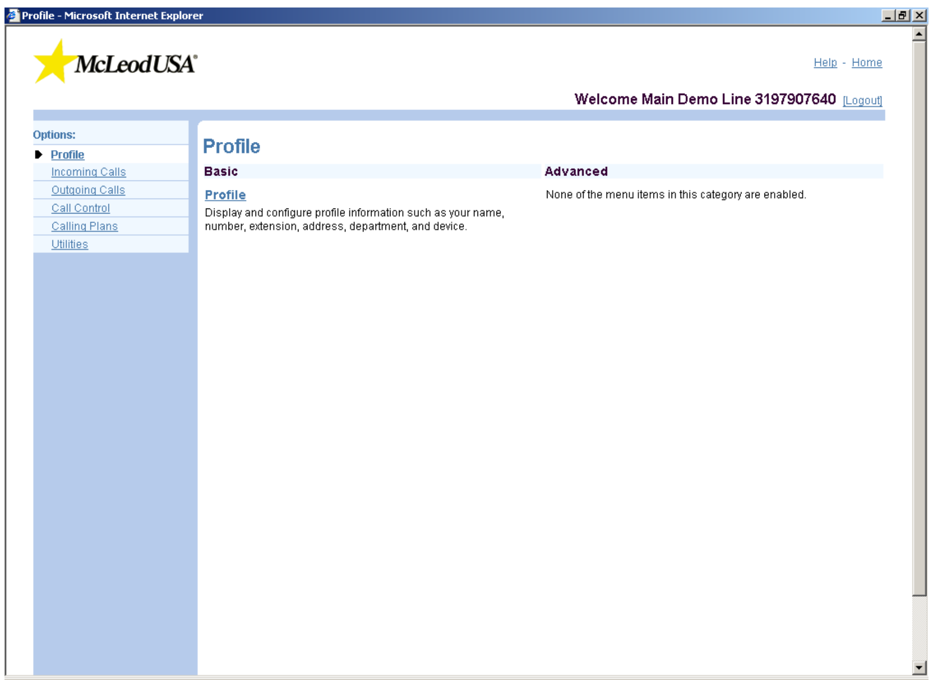


Figure 1: User Profile Page

To enable Do Not Disturb features, click on the Incoming Calls link in the Options section on the left of the screen. This will bring up a list of all the services that can be applied to incoming calls for the user. Select Do Not Disturb, and On to turn the feature on.

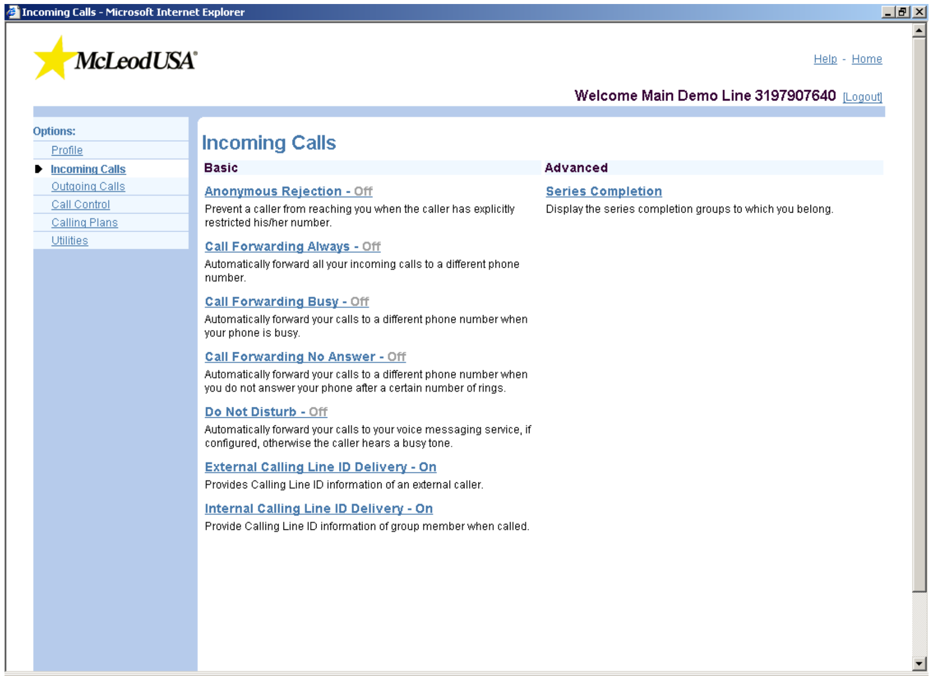


Figure 2: Incoming Calls Services

b. Anonymous Rejection

Requirement

- Same as in above section.

After logging into the web portal as a user and entering the Manage Your Services and the Dynamic Integrated Access section of the account, the system will bring up the initial screen showing the profile screen for the user as shown in Figure 1.

To enable Anonymous Rejection features, click on the Incoming Calls link in the Options section on the left of the screen. This will bring up a list of all the services that can be applied to incoming calls for the user. Select Anonymous Rejection, and On to turn the feature on.

Appendix C: Document Revision History

Document Revision Number	Date Last Revised	Revision Author(s)	Comments
0.1	4/18/2008	Elton Nie	First Draft

References

1. PAETEC BroadWorks Provisioning MOP for Release 14 SP1
2. Cisco. March 2007. *Cisco Unified Communications Manager Express System Administrator Guide.* Available from Cisco CCO.