

AT&T IP Toll-Free: Connecting Cisco Unified Communications Manager 6.1(1a) via the Cisco Unified Border Element using SIP

July 18, 2008

Table of Contents

Introduction	2
Network Topology	
System Components	
Hardware Components	3
Hardware Components Software Requirements	3
Features	
Features Supported	
Features Not Supported	
Caveats	
Configuration	
Configuring Cisco Unified Border Element (CUBE)	
Configuring the Cisco Unified Communications Manager	
(Optional) Configuring the Cisco IOS Gateway or MGCP (fax T.38 and fax G.711ulaw)	27
Acronyms	



Introduction

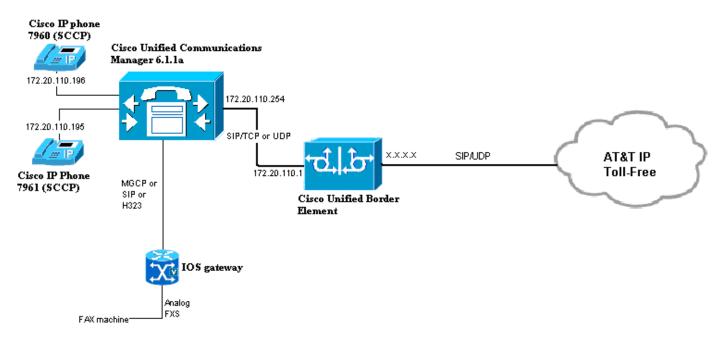
Service Providers today, such as AT&T, are offering alternative methods to connect to the PSTN via their IP network. Most of these services utilize SIP as the primary signaling method and a centralized IP to TDM gateway to provide on-net and off-net services. AT&T IP TollFree is a SP offering that allows connection to the PSTN and may offer the end customer a viable alternative to traditional PSTN connectivity via either Analog or T1 lines. A demarcation device between these services and customer owned services is recommended. The Cisco Unified Border Element providers demarcation, security, interworking and session management services.

- This application note describes how to configure a Cisco Unified Communications Manager (CUCM) 6.1.1a with a Cisco Unified Border Element (CUBE) for connectivity to AT&T's IP Toll-Free SIP trunk service. The deployment model covered in this application note is CPE (CUCM6.1.1a/CUBE) to PSTN (AT&T IP toll-free SIP). AT&T's IP Toll-Free provides inbound call service only (PSTN to CPE). Outbound calls (CPE to PSTN), including emergency 911 calls, are not supported.
- Testing was performed in accordance to AT&T's IP Toll-Free test plan and all features were verified. Key features verified are: Basic Call, DNIS translations, Codec Negotiation, Transfer Connect (8YY xfer), Intra-site Transfers, Intra-site Conferencing, Fax using G711ulaw (pass-through) and Fax using T.38 (G3 and SG3 speeds)
- The Cisco Unified Border Element configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between AT&T SIP network and Cisco Unified Communications. The configuration described in this document details the important commands to have enabled for interoperability to be successful and care must be taken, by the network administrator deploying CUBE in a production environment, to ensure these commands are set per each dial-peer requiring to interoperate to AT&T SIP network.



Network Topology

Figure 1. Basic Call Setup



System Components

Hardware Components

- Cisco IOS gateway running CUBE 1.2 (IOS image version 12.4(15)XZ or later)
 - Cisco Unified Border Element is an integrated Cisco IOS Software application that runs on various IOS platforms, follow the link for more details: http://www.cisco.com/go/cube
- Cisco MCS 7800 Series server (Cisco Unified Communications Manager)
- Cisco IP Phones (The topology diagram shows 7960 and 7961, but any Cisco IP phone model can be used)
- Cisco IOS Gateway (only needed if Fax, analog phones or TDM systems are to interconnect). This component may be a H323, SIP or MGCP gateway, the protocol is optional and the choice is left up to the customers network design.

Software Requirements

- Cisco Unified CM 6.1.1.2000-3 (6.1(1a))
- CUBE version 1.2 (IOS version 12.4(15)XZ or later, IOS version 12.4(20)T or later)
- Cisco GW IOS Release: 12.4 or later



Features

Features Supported

- Basic Call using G.729 or G.711ulaw
- Calling Party Number Presentation and Restriction
- AT&T Transfer Connect (8YY)
- Intra-site Call Transfer
- Intra-site Conference, see caveat section for details.
- Fax using G711ulaw and/or T.38
- Incoming DNIS Translation and Routing
- CUBE: performs Delay-Offer-to-Early-Offer transcoding of an initial SIP INIVTE without SDP

Features Not Supported

- CUCM/CUBE Codec negotiation of G.726
- AT&T does not support SIP Session Timer (Session-Expires and Min-SE headers)

Caveats

- When using G.729 between AT&T IP Toll-Free and Cisco Unified Border Element/Cisco Unified Communications Manager SIP trunk it is required to configure a Conference Bridge (CFB) resource on CUBE in order for Cisco Unified Communications Manager IP phone to initiate a three-way conference between G729 media end-points. See configuration section for details.
- For DTMF digit passing using RFC2833 you must set a payload-type value of 96 for "nte" (network telephone-events DTMF) on CUBE dial-peer pointing towards Cisco Unified Communications Manager. See configuration section for details.
- AT&T SIP trunk offering does not support the SIP Session Timer. Cisco CUBE version 1.2 (IOS 12.4(15)XZ or 12.4(20)T have been enabled to address this limmitation by default. The IOS CLI administrator can also utilize SIP profile configuration in order to add, modify or strip SIP headers from SIP messages as the administrator sees fit . . See configuration section for detailed example.
- Cisco Unified Communications Manager will only handle one codec type per SIP trunk. Eventhough AT&T SIP INVITE proposes a list
 of codecs (e.g. G729, G711, G726) the UCM SIP trunk will only allow the codec value it has been configured for. See configuration
 section for details



Configuration

Configuring Cisco Unified Border Element (CUBE)

Critical commands are marked bold with footnote and description at bottom of the page

```
c3825_CUBE#sh runn
Building configuration...
Current configuration: 4299 bytes
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
hostname c3825_CUBE
boot-start-marker
boot-end-marker
logging message-counter syslog
logging buffered 10000000
no logging console
no logging monitor
enable password cisco
no aaa new-model
ip cef
no ip domain lookup
multilink bundle-name authenticated
voice-card 0
dspfarm 1
dsp services dspfarm1
voice service voip
allow-connections sip to sip<sup>2</sup>
redirect ip2ip
fax protocol t38 ls-redundancy 0 hs-redundancy 0 fallback cisco<sup>3</sup>
sip
 early-offer forced4
 midcall-signaling passthru
 sip-profiles 15
```

¹ Command enables dsp farming to allow for conference bridge resources using G729 media on CUCM.

² Command enables CUBE SIP-to-SIP functionality

³ Command enables fax-relay using T.38 globally on CUBE, to enable fax G711ulaw simply remove this feature by using the "no" command

⁴ Command enables delay-offer to early-offer translation of initial SIP INVITE w/o SDP.



```
voice class codec 16
codec preference 1 g729br8
codec preference 2 g729r8
codec preference 3 g711ulaw
voice class sip-profiles 15
request INVITE sip-header Supported remove
request INVITE sip-header Min-SE remove
archive
log config
 hidekeys
interface GigabitEthernet0/0
description "interface connecting to WAN via SP(AT&T SIP)"
ip address 99.136.XX.XX 255.255.255.0
duplex auto
speed auto
media-type rj45
interface GigabitEthernet0/1
description "interface connecting to Cisco Unified CM LAN"
ip address 172.20.110.XX 255.255.255.0
duplex auto
speed auto
media-type rj45
ip default-gateway 99.136.XX.XX
ip forward-protocol nd
ip route 0.0.0.0 0.0.0.0 99.136.XX.XX
```

⁵ Command enables a SIP profile to strip min-SE and session-timer headers from the SIP INVITE going towards AT&T network. This command is only necessary if your CUBE is running an IOS version earlier than 12.4(15)XZ.

⁶ Command provides codec filtering of G726 for incoming SIP INVITE from AT&T



```
ip route 172.20.110.0 255.255.255.0 GigabitEthernet0/1
ip http server
control-plane
sccp local GigabitEthernet0/17
sccp ccm 172.20.110.254 identifier 1 version 6.0
sccp
sccp ccm group 1
associate ccm 1 priority 1
associate profile 1 register cfb0018185bb7a18
dspfarm profile 1 conference
codec q729r8
codec g729abr8
codec g729ar8
codec g729br8
codec g711ulaw
maximum sessions 6
associate application SCCP
dial-peer voice 1041 voip
decription "default incoming call dial-peer from AT&T"
rtp payload-type cisco-codec-fax-ind 989
rtp payload-type nte 9610
voice-class codec 11
session protocol sipv212
incoming called-number 00000104.13
dtmf-relay rtp-nte14
fax-relay sg3-to-g315
fax protocol t38 nse ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw<sup>16</sup>
```

⁷ Command enables SCCP registration of dsp resources for conferencing on Cisco Unified Communications Manager

⁸ This id number will be created using the letters cfb as the leading characters and obtaining the complete mac address from the ethernet interface connecting to the Cisco Unified Communications Manager LAN. (example: "show interface GigabitEthernet 0/1")

⁹ Command removes the mentioned codec type from Dynamic Payload-Type value 96 in order to re-assign value 96 for RFC2833 DTMF relay ¹⁰ Command assigns Payload-Type value 96 to DTMF RFC2833 (nte events) for the particular dial-peer, 'other DTMF PT values (96-127) can be configured'

¹¹ Command applies the codec filtering rule to the particular dial-peer

¹² Command enables SIP trunk communication towards Cisco Unified Communications Manager Call Leg (outbound dial-peer)

¹³ Command sets the dial-peer configuration values to be applied to all incoming calls with a called number matching pattern. This number shall be set to match the "called" number pattern from AT&T's incoming INVITE message.

¹⁴ Command enables DTMF relay using RFC2833 for that particular dial-peer

¹⁵ Command enables interworking between SG3 and G3 faxes



```
dial-peer voice 1042 voip
description "outgoing dial-peer towards Cisco Unified CM"
destination-pattern 00000104.
rtp payload-type cisco-codec-fax-ind 9817
rtp payload-type nte 9618
voice-class codec 1
session protocol sipv2
session target ipv4:172.20.110.XX
dtmf-relay rtp-nte
fax-relay sg3-to-q319
fax protocol t38 nse Is-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw<sup>20</sup>
sip-ua
no remote-party-id
gatekeeper
shutdown
line con 0
password cisco
login
line aux 0
line vty 0 4
exec-timeout 0 0
password cisco
login
exception data-corruption buffer truncate
scheduler allocate 20000 1000
end
```

be configured'

¹⁶ Command enables fax T.38 for the particular dial-peer, you must remove this command (and the global command if set) in order for fax G711ulaw functionality.

17 Command removes the mentioned codec type from Dynamic Payload-Type value 96 in order to re-assign value 96 for RFC2833 DTMF relay

¹⁸ Command assigns Payload-Type value 96 to DTMF RFC2833 (nte events) for the particular dial-peer, 'other DTMF PT values (96-127) can

¹⁹ Command enables interworking between SG3 and G3 faxes

²⁰ Command enables fax T.38 for the particular dial-peer, you must remove this command (and the global command if set) in order for fax G711ulaw functionality.



Configuring the Cisco Unified Communications Manager

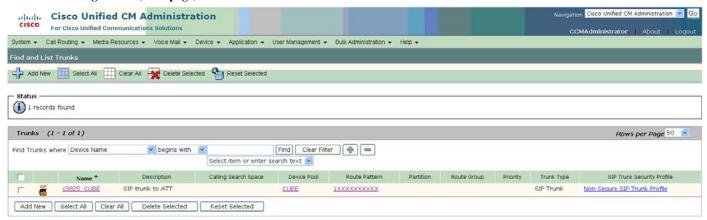
Cisco Unified Communications Manager version



This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco cryptographic products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with U.S. and local country laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

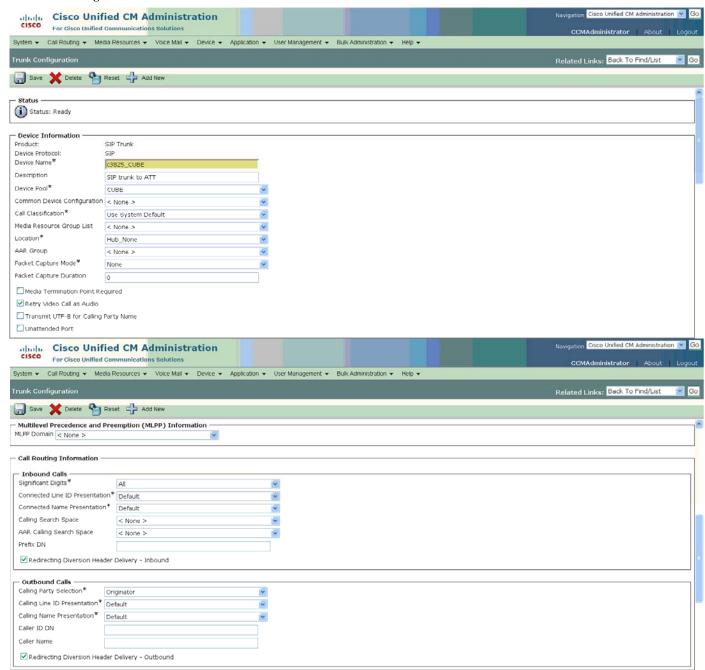
A summary of U.S. laws governing Cisco cryptographic products may be found at: http://www.cisco.com/wwi/export/crypto/tool/stqrg.html
If you require further assistance please contact us by sending email to export@cisco.com.

SIP Trunk configuration (Title page)

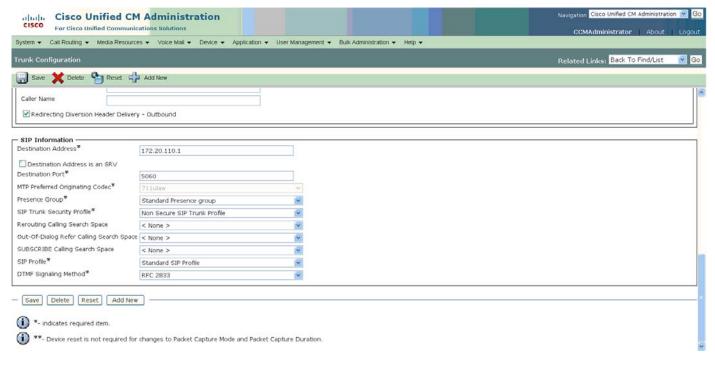




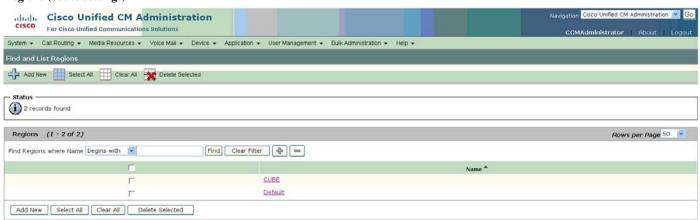
SIP Trunk configuration



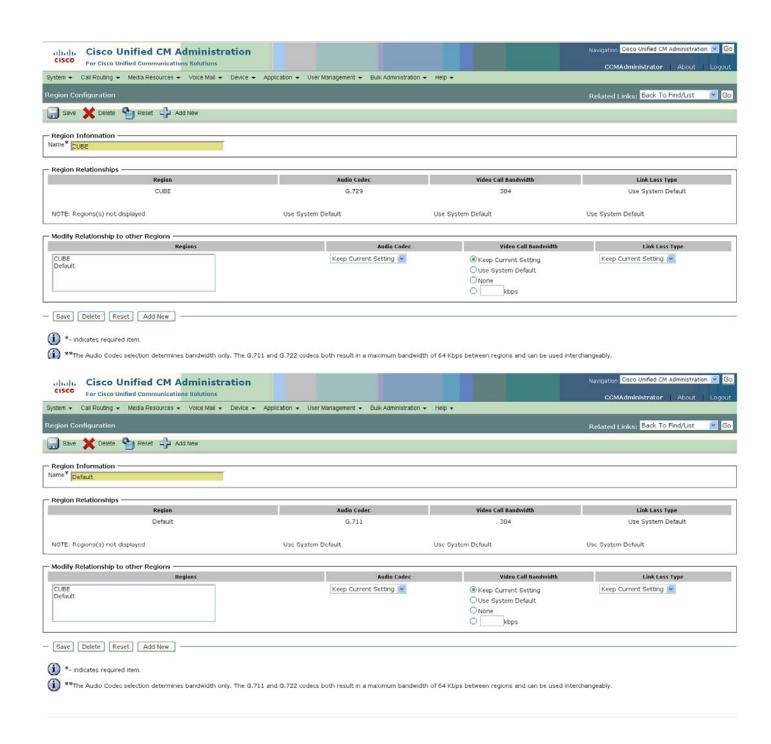




Regions (codec settings)

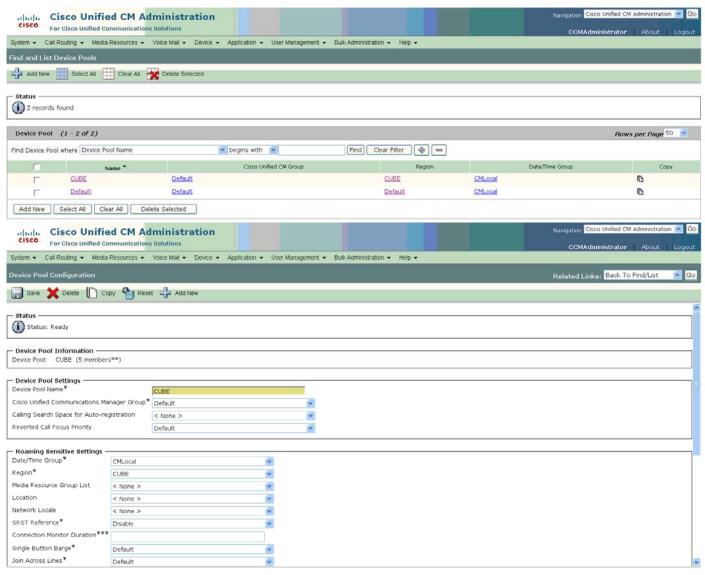




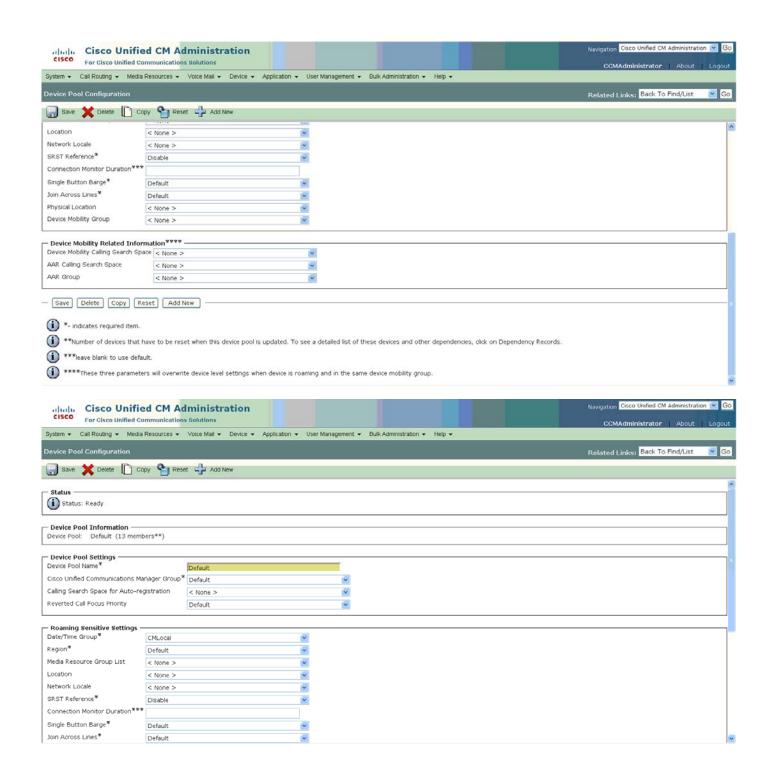




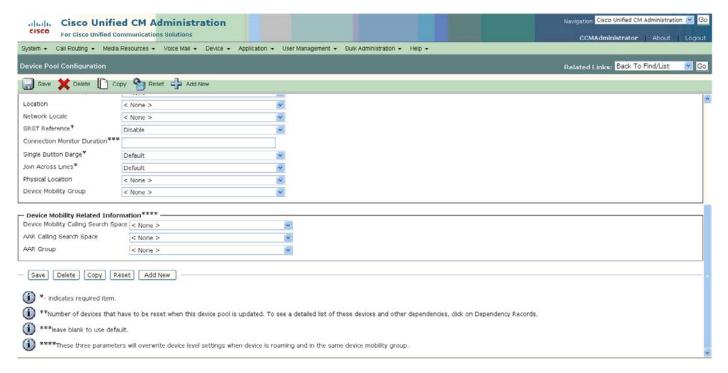
Device Pool



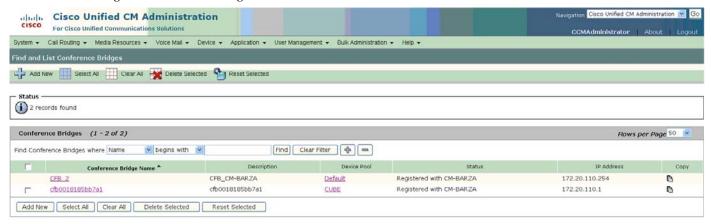




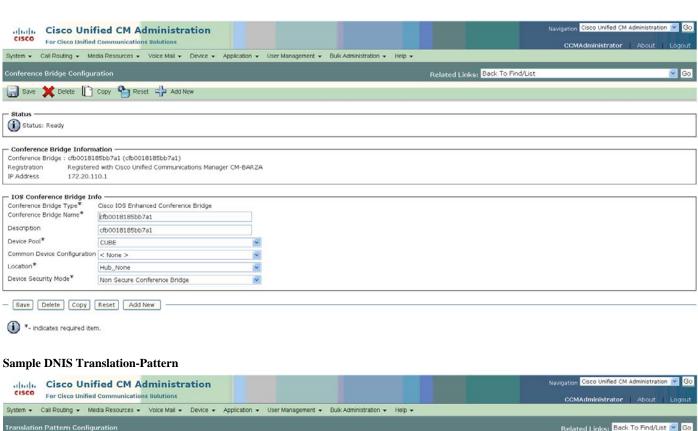


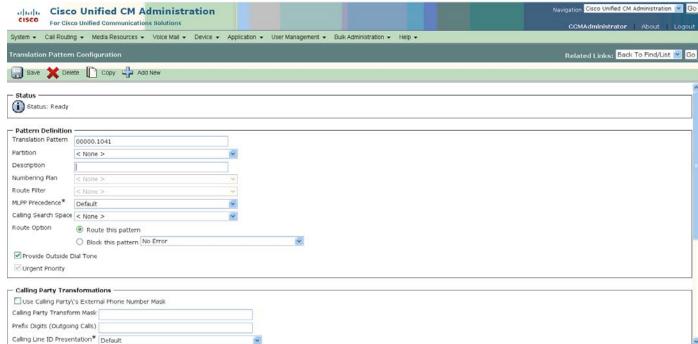


IOS conference bridge for G729 conferencing

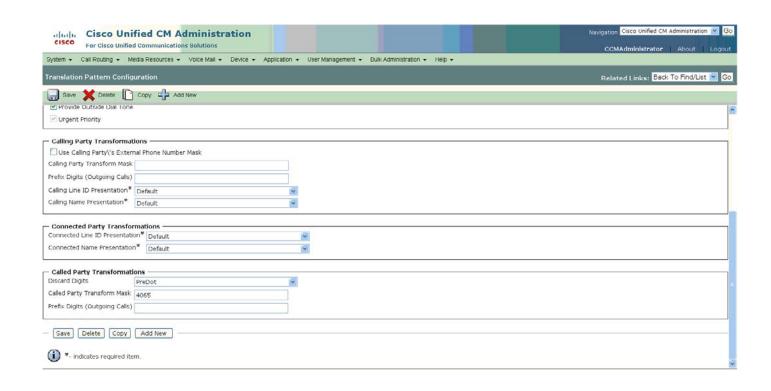






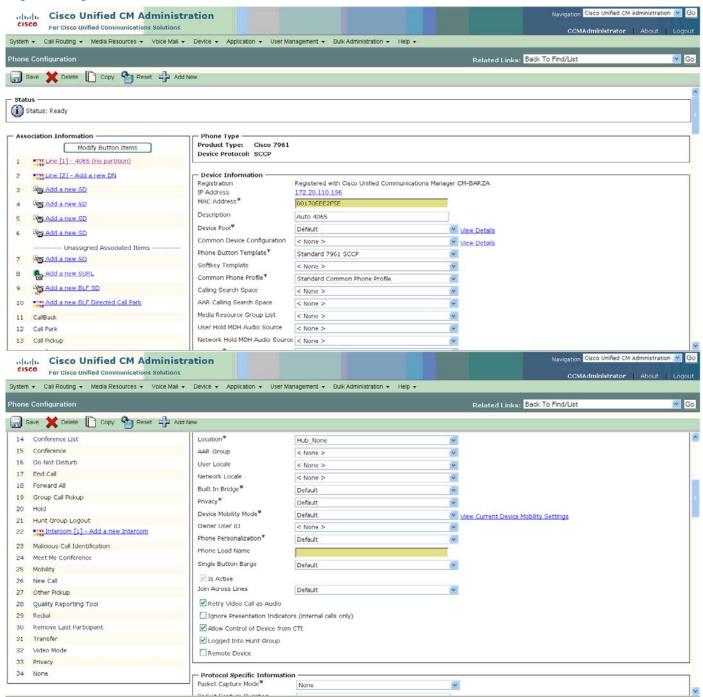




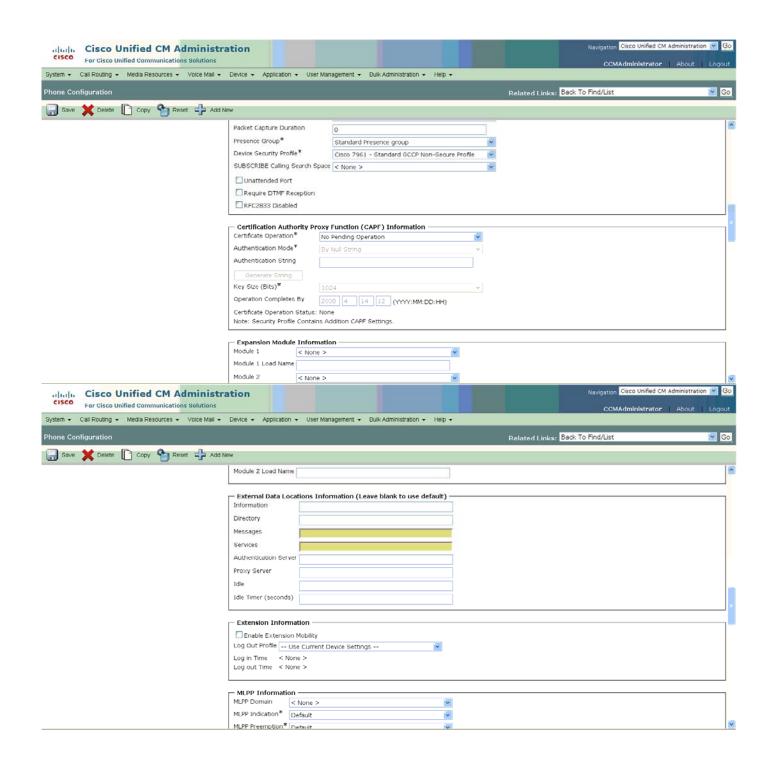




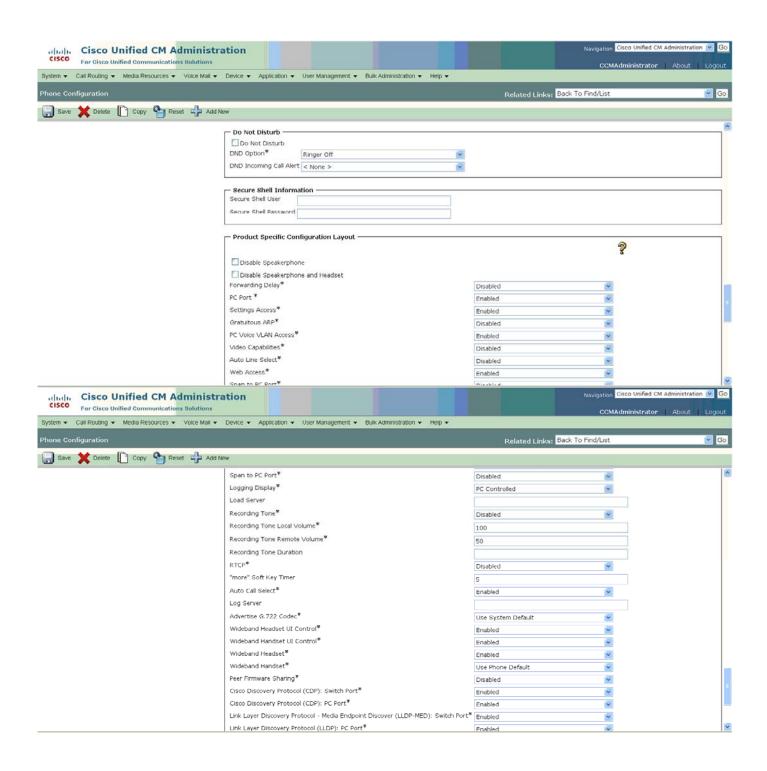
IP phone configuration



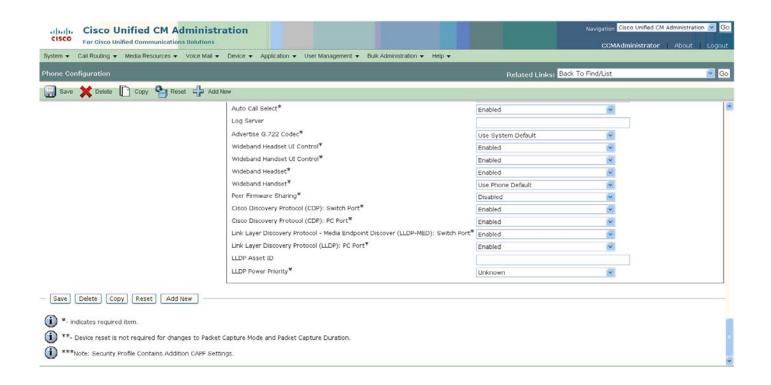






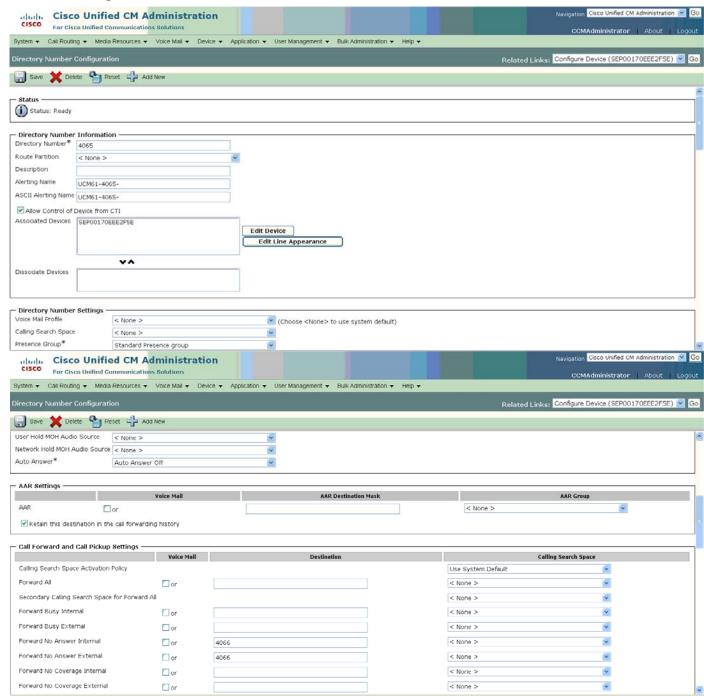




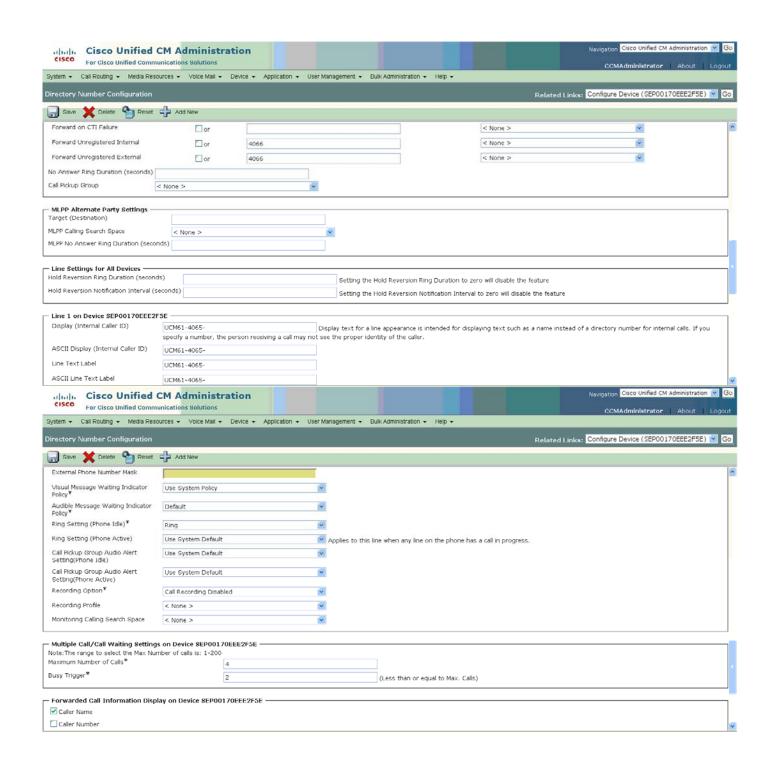




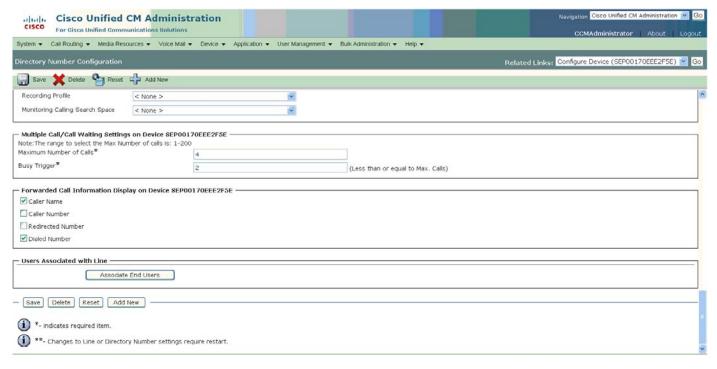
IP Phone DN configuration



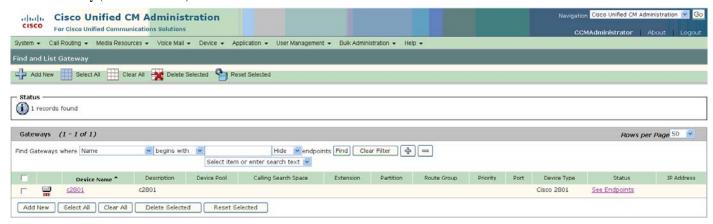




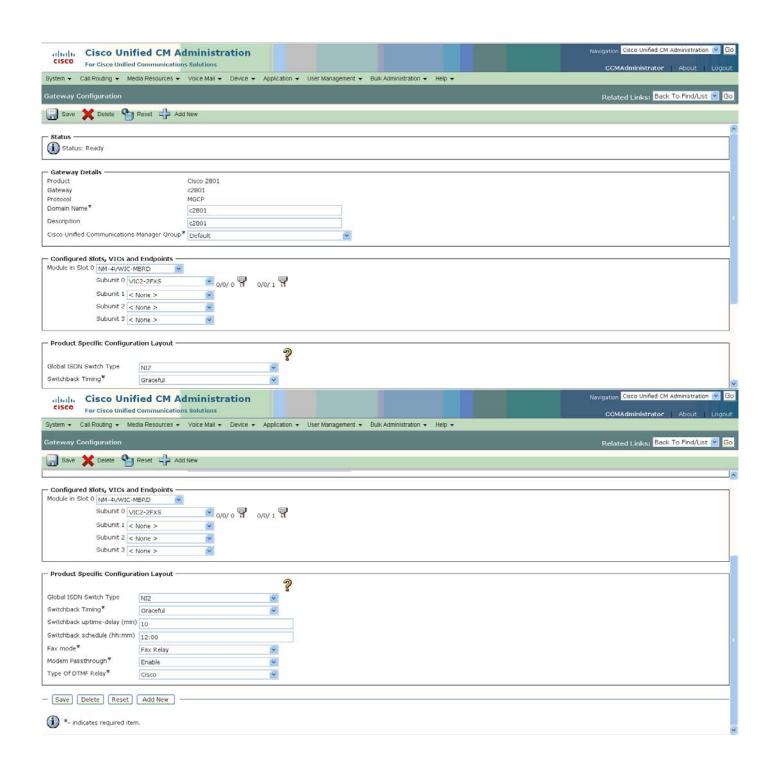




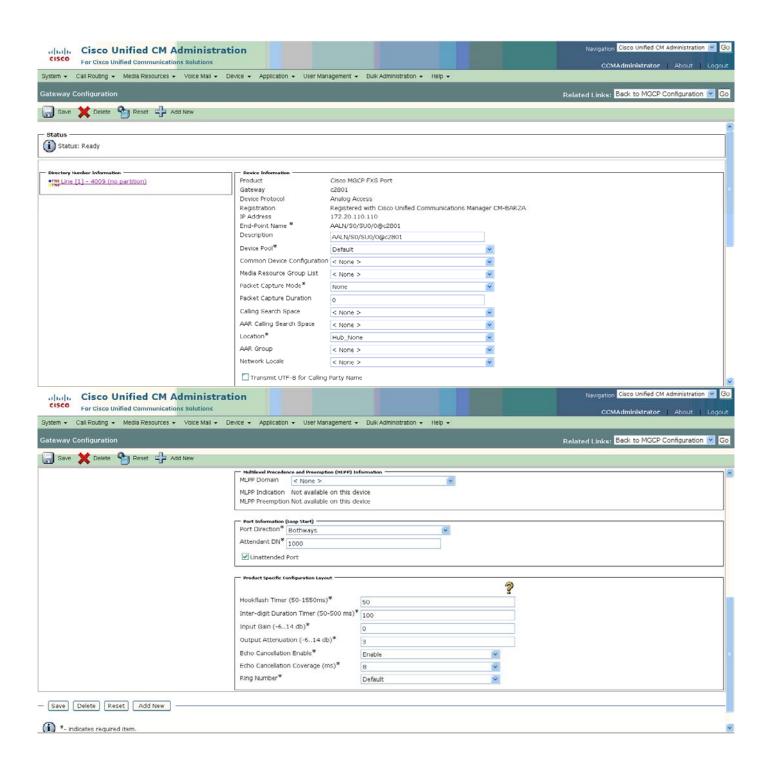
MGCP Gateway (for fax interface)













Critical commands have been bolded

(Optional) Configuring the Cisco IOS Gateway or MGCP (fax T.38 and fax G.711ulaw)

c2801# sh run Building configuration... Current configuration: 2945 bytes version 12.4 service timestamps debug datetime msec service timestamps log datetime msec no service password-encryption hostname c2801 boot-start-marker boot-end-marker logging buffered 100000000 no logging console enable password cisco ! no aaa new-model network-clock-participate wic 1 network-clock-participate wic 2 network-clock-select 1 T1 0/1/0 ip cef no ip domain lookup ip host CM-BARZA 172.20.110.254 multilink bundle-name authenticated isdn switch-type primary-ni voice-card 0 dsp services dspfarm ļ



```
archive
log config
 hidekeys
controller T1 0/1/0
framing esf
linecode b8zs
pri-group timeslots 1-24 service mgcp
controller T1 0/1/1
framing esf
linecode b8zs
pri-group timeslots 1-24
interface FastEthernet0/0
ip address 172.20.110.110 255.255.255.0
duplex auto
speed auto
interface FastEthernet0/1
no ip address
shutdown
duplex auto
speed auto
interface Serial0/1/0:23
no ip address
encapsulation hdlc
isdn switch-type primary-ni
isdn incoming-voice voice
isdn bind-I3 ccm-manager
no cdp enable
interface Serial0/1/1:23
no ip address
encapsulation hdlc
isdn switch-type primary-dms100
isdn protocol-emulate network
isdn incoming-voice voice
no cdp enable
interface BRI0/2/0
```



```
no ip address
isdn switch-type basic-ni
isdn point-to-point-setup
interface BRI0/2/1
no ip address
isdn switch-type basic-ni
isdn point-to-point-setup
ip route 0.0.0.0 0.0.0.0 172.20.110.1
ip http server
no ip http secure-server
disable-eadi
control-plane
voice-port 0/0/0
voice-port 0/0/1
voice-port 0/1/0:23
voice-port 0/1/1:23
voice-port 0/2/0
voice-port 0/2/1
ccm-manager mgcp
ccm-manager music-on-hold
ccm-manager config server CM-BARZA
ccm-manager config
mgcp
mgcp call-agent CM-BARZA 2427 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode out-of-band
mgcp rtp unreachable timeout 1000 action notify
mgcp modem passthrough voip mode nse
mgcp package-capability rtp-package
mgcp package-capability sst-package
mgcp package-capability pre-package
no mgcp fax t38 inhibit<sup>21</sup>
no mgcp package-capability res-package
```

²¹ This command "must" be entered in order for fax T.38 to be enabled, in order to enable fax G711ulaw the command must be entered to inhibit T.38



```
no mgcp package-capability fxr-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax rate 14400
mgcp rtp payload-type g726r16 static
mgcp profile default
dial-peer voice 999 pots
service mgcpapp
port 0/0/0
dial-peer voice 998 pots
service mgcpapp
port 0/0/1
line con 0
password cisco
login
line aux 0
line vty 04
exec-timeout 0 0
password cisco
login
scheduler allocate 20000 1000
end
c2801#
```



Acronyms

Acronym	Definitions
SIP	Session Initiation Protocol
MGCP	Media Gateway Control Protocol



Important Information

THE SPECIFICATIONS AND INFORMATION REGARDING THE PRODUCTS IN THIS MANUAL ARE SUBJECT TO CHANGE WITHOUT NOTICE. ALL STATEMENTS, INFORMATION, AND RECOMMENDATIONS IN THIS MANUAL ARE BELIEVED TO BE ACCURATE BUT ARE PRESENTED WITHOUT WARRANTY OF ANY KIND, EXPRESS OR IMPLIED. USERS MUST TAKE FULL RESPONSIBILITY FOR THEIR APPLICATION OF ANY PRODUCTS.

IN NO EVENT SHALL CISCO OR ITS SUPPLIERS BE LIABLE FOR ANY INDIRECT, SPECIAL, CONSEQUENTIAL, OR INCIDENTAL DAMAGES, INCLUDING, WITHOUT LIMITATION, LOST PROFITS OR LOSS OR DAMAGE TO DATA ARISING OUT OF THE USE OR INABILITY TO USE THIS MANUAL, EVEN IF CISCO OR ITS SUPPLIERS HAVE BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGES.



Corporate Headquarters

Cisco Systems, Inc. 170 West Tasman Drive San Jose, CA 95134-1706 USA www.cisco.com

Tel: 408 526-4000 800 553-NETS (6387)

Fax: 408 526-4100

© 2008 Cisco Systems, Inc. All rights reserved.

European Headquarters

Cisco Systems International BV

Haarlerbergpark Haarlerbergweg 13-19 1101 CH Amsterdam The Netherlands

www-europe.cisco.com Tel: 31 0 20 357 1000 Fax: 31 0 20 357 1100

Americas Headquarters

Cisco Systems, Inc. 170 West Tasman Drive San Jose, CA 95134-1706

www.cisco.com Tel: 408 526-7660 Fax: 408 527-0883

Asia Pacific Headquarters

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

Cisco Systems has more than 200 offices in the following countries and regions. Addresses, phone numbers, and fax numbers are listed on the Cisco Web site at www.cisco.com/go/offices.

Argentina • Australia • Austria • Belgium • Brazil • Bulgaria • Canada • Chile • China PRC • Colombia • Costa Rica • Croatia • Czech Republic • Denmark • Dubai, UAE • Finland • France • Germany • Greece • Hong Kong SAR • Hungary • India • Indonesia • Ireland • Israel • Italy • Japan • Korea • Luxembourg • Malaysia • Mexico • The Netherlands • New Zealand • Norway • Peru • Philippines • Poland • Portugal • Puerto Rico • Romania • Russia • Saudi Arabia • Scotland • Singapore • Slovakia • Slovenia • South Africa • Spain • Sweden • Switzerland • Taiwan • Thailand • Turkey Ukraine • United Kingdom • United States • Venezuela • Vietnam • Zimbabwe

CCDE, CCENT, Cisco Eos, Cisco Lumin, Cisco Nexus, Cisco StadiumVision, the Cisco logo, DCE, and Welcome to the Human Network are trademarks; Changing the Way We Work, Live, Play, and Learn is a service mark; and Access Registrar, Aironet, AsyncOS, Bringing the Meeting To You, Catalyst, CCDA, CCDP, CCIE, CCIP, CCNA, CCNP, CCSP, CCVP, Cisco, the Cisco Certified Internetwork Expert logo, Cisco IOS, Cisco Press, Cisco Systems, Cisco Systems Capital, the Cisco Systems logo, Cisco Unity, Collaboration Without Limitation, EtherFast, EtherSwitch, Event Center, Fast Step, Follow Me Browsing, FormShare, GigaDrive, HomeLink, Internet Quotient, IOS, iPhone, iQ Expertise, the iQ logo, iQ Net Readiness Scorecard, iQuick Study, IronPort, the IronPort logo, LightStream, Linksys, MediaTone, MeetingPlace, MGX, Networkers, Networking Academy, Network Registrar, PCNow, PIX, PowerPanels, ProConnect, ScriptShare, SenderBase, SMARTnet, Spectrum Expert, StackWise, The Fastest Way to Increase Your Internet Quotient, TransPath, WebEx, and the WebEx logo are registered trademarks of Cisco Systems, Inc. and/or its affiliates in the United States and certain other countries.

All other trademarks mentioned in this document or Website are the property of their respective owners. The use of the word partner does not imply a partnership relationship between Cisco and any other company. (0805R)