AT&T IP FlexReach AVPN/MIS/PNT transports: Connecting Cisco Unified Communications Manager Express 8.5 using SIP

March 18, 2011

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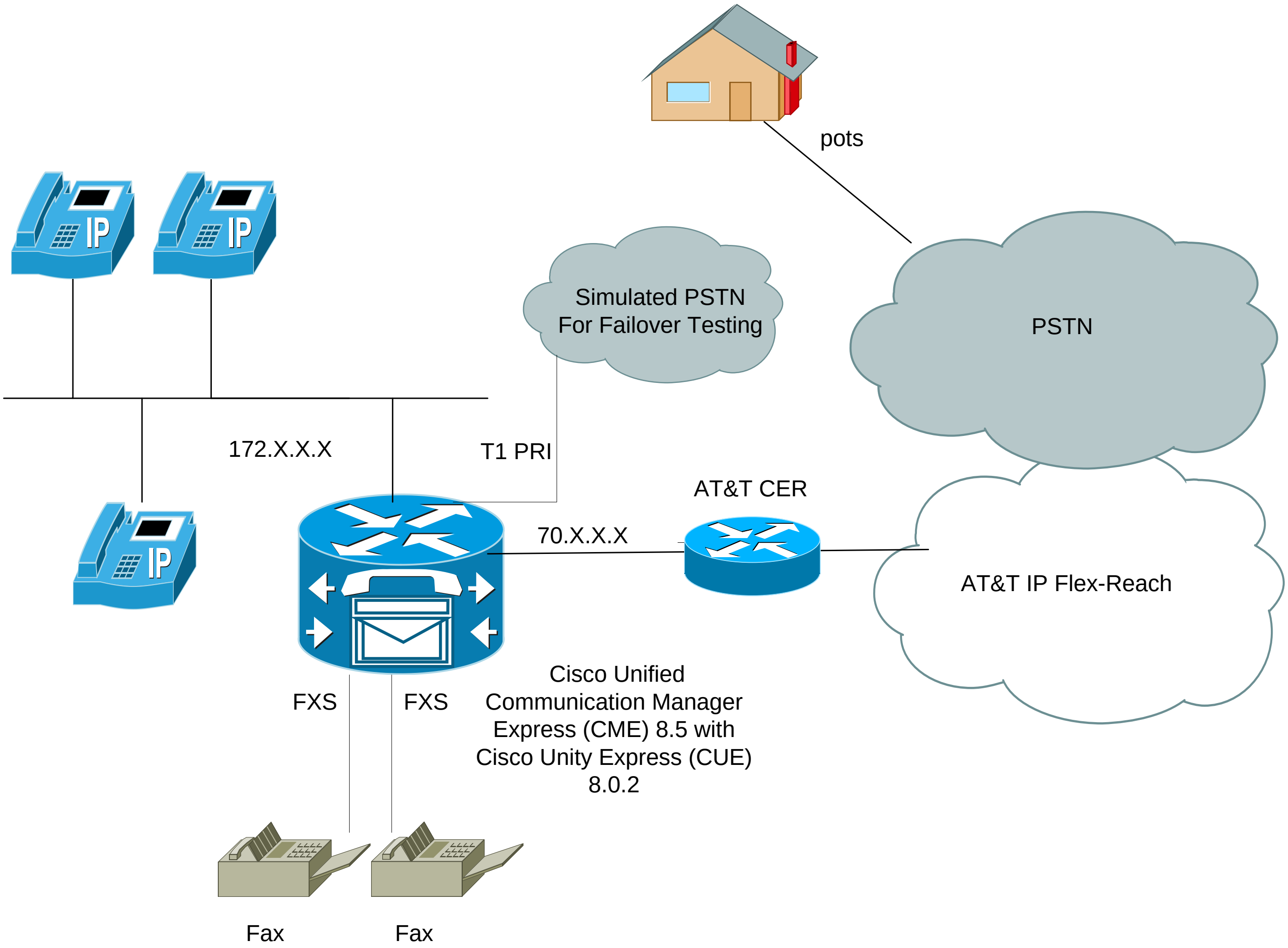
Introduction

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 FlexReach is a service provider 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 provides demarcation, security, interworking and session management services.

- This application note describes how to configure a Cisco Unified Communications Manager Express (CISCO UCME) 8.5 with Cisco Unity Express and Cisco Unified Border Element (CISCO UBE) function for connectivity to AT&T's IP Flex-Reach SIP trunk service. The application note also covers support and configuration example Cisco Unity Express (CUE) messaging integrated into the Cisco Unified Communications Manager Express. The deployment model covered in this application note is Customer Premises Equipment (CISCO UCME/CUE) to PSTN (AT&T IP Flex-Reach SIP). AT&T IP Flex-Reach provides inbound and outbound call service.
- Testing was performed in accordance to AT&T's IP Flex-Reach test plan and all features were verified. Key features verified are: inbound and outbound basic call (including international calls), calling name delivery, calling number and name restriction, DNIS translations, CODEC negotiation, advanced 8YY call prompter, intra-site transfers, intra-site conferencing, call hold and resume, call forward (forward all, busy and no answer), leaving and retrieving voicemail (Cisco Unity Express), CISCO auto-attendant (BACD), fax using T.38 and G.711 (G3 and SG3 speeds), teleconferencing, failover of unresponsive SIP network to PSTN and outbound/inbound calls to/from TDM networks.
- The Cisco Unified Border Element function 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 Express. The configuration described in this document details the important commands for successful interoperability. Care must be taken by the network administrator deploying CISCO UBE to ensure these commands are set per each dial-peer required, to interoperate to AT&T SIP network.
- For IP Flexible Reach service with AVPN access, the CUCME LAN IP address (facing the AT&T CER) can be an AT&T assigned signaling IP address. For IP Flexible Reach service with MIS/PNT (and optionally with AVPN) access, the CUCME LAN IP address (facing the CER) can be private IP address. This will be NATed by the AT&T managed CER (or customer managed/MRS managed CER for AVPN). Consult with AT&T provisioning engineer to resolve any IP addressing issues.
- Please refer to the Emergency 911/E911 Limitations and Restrictions section of this document for more information on Emergency 911/E911 services.
- Consult your Cisco representative for the correct IOS image and for the specific application and Device Unit License and Feature License requirements for all your Cisco Unified Communications Express with Cisco Unified Border Element components.

Network Topology

Figure 1. Basic Call Setup



The Cisco Unified Communication Manager Express depicted in Figure 1 also functions as a border element. It is not an AT&T managed device.

System Components

Hardware Components

- Cisco Integrated Service Router G2. This solution was tested with C3925 but this application note applies to any ISR G2 platform. Refer to the following link for more information on ISR G2 platforms:

  http://www.cisco.com/en/US/docs/routers/access/1900/roadmap/ISRG2_roadmap.html

  - Cisco IP Phones. This solution was tested with 7961 and 7975 phones, but any Cisco IP Phone model supporting RFC2833 can be used)
  - Cisco SRE (Service Ready Engine) compatible with ISR G2 platform to run Cisco Unity Express application. This solution was tested with ISM-SRE-300-K9 but the following modules may be used.
    - Internal Service Module (ISM) – ISM-SRE-300-K9
    - Service Module (SM) – SM-SRE-700-K9 or SM-SRE-900-K9
    - Enhanced Network Module (NME) - NME-CUE (will need SM-NM-ADPTR adapter)

    Please refer to the following CUE Compatibility Matrix link for information on hardware, platforms, minimum IOS software and service/network module requirements.

    http://www.cisco.com/en/US/docs/voice_ip_comm/unity_exp/compatibility/cuecomp.htm#HardwareISM-SRE-300-k9
  - Packet Voice Data Module (PVDM). You will need to install DSP modules (PVDM) on the ISR G2 platform if you require MTP, Transcoding or Conference Bridge resources for codecs other than G.711. DSPs are not required for basic calls. Follow the following link for system required DSP calculator.

    http://www.cisco.com/cgi-bin/Support/DSP/cisco_dsp_calc.pl

Software Requirements

- Cisco IOS gateway running Cisco Unified Communication Manager Express (CUCME) 8.5 (IOS version **15.1.3T** release). This solution was tested with Cisco IOS image: C3900-universalk9-mz.SPA.151.3.T.bin
  - Cisco Unified Border Element (CUBE) is an integrated Cisco IOS software application that requires a separate CUBE feature license to connect to AT&T SIP trunk.
  - Cisco Unity Express is an integrated Cisco IOS software application that requires a separate UNITY feature license to implement voicemail and other messaging features. This solution was tested with Cisco Unity Express version (8.0)

Consult your Cisco representative for the correct IOS image and for the specific application and Device Unit License and Feature License requirements for all your Cisco Unified Communications components. For reference, please follow this link:

http://www.cisco.com/en/US/prod/collateral/voicesw/ps6790/gatecont/ps5640/order_guide_c07_462222.html

Features

Features Supported

- Basic Call using G.729
- Calling Party Number Presentation and Restriction
- Calling Name
- AT&T Advanced 8YY Call Prompter (8YY)
- Intra-site Call Transfer
- Intra-site Conference (See Caveat section for details)
- Call Hold and Resume
- Call Forward All, Busy and No Answer
- AT&T IP Teleconferencing
- Fax using T.38 (See Caveat section for details)
- Fax over G.711 (See Caveat section for details)
- Incoming DNIS Translation and Routing
- CISCO UBE: performs Delayed-Offer-to-Early-Offer conversion of an initial SIP INVITE without SDP
- Outbound calls to AT&T's IP and TDM networks
- CPE voicemail managed service, leave and retrieve voice messages via incoming AT&T SIP trunk (Cisco Unity Express)
- Auto-attendant transfer-to service (See Caveat section for details)
- Failover (From non-responsive SIP network to legacy PSTN circuit)

Features Not Supported

- CISCO UCME/CISCO UBE Codec negotiation of G.722.1
- Real-Time Transport Control Protocol (RTCP) (See Caveat section for details)

Caveats

Fax:

- The maximum fax rate achieved using T.38 (G3 or SG3) fax protocol is only 9600 kbps. This is related to Cisco defect id CSCtk06444. where max bit rate is not updated and fails to advertise V.17 capability.
- An issue was found when receiving T.38 Fax at SG3 speed over certain local PSTN NPAs served by the legacy SBC platform. This is related to Cisco defect id: CSCto45871 where the CUCME fails to send a T.38 re-invite.

Auto-Attendant:

- The CUCME Basic Automatic Call Distribution (BACD) was employed to enable the auto-attendant feature. The test was performed using the default codec G711 for auto attendant prompts. G.729 prompts can be used; however it was not tested here.

Real-Time Transport Control Protocol (RTCP)

- CUCME does not support the periodic transmission of RTCP sender report to provide statistics of RTP flow. Certain AT&T networks such as AVPN require this feature.

Configuration considerations

- When using G.729 between AT&T IP Flex-Reach and Cisco Unified Communication Manager Express (CUCME)/ Cisco Unified Border Element (CUBE) SIP trunk, it is required to configure a conference bridge (CFB) resource to initiate a three-way conference between G729 media end-points. See configuration section for details.
- For forwarded calls from CUCME user to PSTN (out to AT&T's IP Flex-reach service) some AT&T serviced areas require that the SIP Diversion header contain the full 10-digit DID number of the forwarding party. In this application note the assumption was made that a typical customer will utilize extension numbers (4-digit assignments in this example) and map 10-digit DID number using CISCO UCME translation patterns. Because we use 4-digit extensions on our CUCME IP phones, it is necessary to expand the 4-digit extension included in the Diversion header of a forwarding INVITE message, to its full 10-digit DID number when the IP phone is set to call-forward. The requirement to expand the Diversion-Header has been achieved by the use of a SIP profile in CUBE (See configuration section for details.).
- Upon receiving inbound calls, AT&T SIP network will always select the first choice codec presented in the initial SIP INVITE (unless the end-device does not support the listed preferred codec), and processes calls accordingly. Customers wishing to place/receive G.711-only calls must configure separate dial-peer(s) on CUCME/CUBE with G.711 codec assigned. Typically, this solution is used for fax transmissions using G711.
- Some SIP components within AT&T core do not support the ":0" as the Boolean value within the "T38FaxFillBitRemoval" parameter within the SDP header of a fax Re-INVITE. Thus, a sip profile is used to remove this attribute to achieve fax T.38 interoperability across AT&T SIP core.
- SIP Profiles may also be employed to advertise desired RTP payload packet size.

Emergency 911/E911 Services Limitations and Restrictions

- Emergency 911/E911 Services Limitations and Restrictions - Although AT&T provides 911/E911 calling capabilities, AT&T does not warrant or represent that the equipment and software (e.g., IP PBX) reviewed in this customer configuration guide will properly operate with AT&T IP Flexible Reach to complete 911/E911 calls; therefore, it is Customer's responsibility to ensure proper operation with its equipment/software vendor.

  While AT&T IP Flexible Reach services support E911/911 calling capabilities under certain Calling Plans, there are circumstances when that E911/911 service may not be available, as stated in the Service Guide for AT&T IP Flexible Reach found at http://new.serviceguide.att.com <http://new.serviceguide.att.com/> . Such circumstances include, but are not limited to, relocation of the end user's CPE, use of a non-native or virtual telephone number, failure in the broadband connection, loss of electrical power, and delays that may occur in updating the Customer's location in the automatic location information database. Please review the AT&T IP Flexible Reach Service Guide in detail to understand the limitations and restrictions.

Configuration

Cisco IOS version

```
CME-G2-SP#sh ver
Cisco IOS Software, C3900 Software (C3900-UNIVERSALK9-M), Version 15.1(3)T, RELEASE SOFTWARE (fc1)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2010 by Cisco Systems, Inc.
Compiled Tue 16-Nov-10 01:48 by prod_rel_team

ROM: System Bootstrap, Version 15.0(1r)M6, RELEASE SOFTWARE (fc1)

CME-G2-SP uptime is 1 minute
System returned to ROM by power-on
System image file is "flash0:c3900-universalk9-mz.SPA.151-3.T.bin"
Last reload type: Normal Reload


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/wwl/export/crypto/tool/stqrg.html

If you require further assistance please contact us by sending email to
export@cisco.com.

Cisco CISCO3925-CHASSIS (revision 1.0) with C3900-SPE100/K9 with 989184K/59392K bytes of memory.
Processor board ID FTX1436A0X0
4 Gigabit Ethernet interfaces
1 terminal line
```

```
1 Virtual Private Network (VPN) Module
4 Voice FXS interfaces
1 Internal Services Module (ISM) with Services Ready Engine (SRE)
DRAM configuration is 72 bits wide with parity enabled.
255K bytes of non-volatile configuration memory.
500472K bytes of ATA System CompactFlash 0 (Read/Write)


License Info:

License UDI:

-------------------------------------------------
Device#   PID                   SN
-------------------------------------------------
*0        C3900-SPE100/K9       FOC14341EZJ



Technology Package License Information for Module:'c3900'

----------------------------------------------------------------
Technology    Technology-package          Technology-package
              Current       Type          Next reboot
-----------------------------------------------------------------
ipbase        ipbasek9      Permanent     ipbasek9
security      securityk9    Permanent     securityk9
uc            uck9          Permanent     uck9
data          datak9        Permanent     datak9

Configuration register is 0x2102
```

Cisco Unified Communication Manager Express (CUCME)

```
CME-G2-SP#sh run
Building configuration...


Current configuration : 16745 bytes
!
version 15.1
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname CME-G2-SP
!
boot-start-marker
boot system flash0:c3900-universalk9-mz.SPA.151-3.T.bin
boot-end-marker
!
!
card type t1 1 1
logging buffered 999999
logging rate-limit console 10000
no logging console
enable password cisco
!
no aaa new-model
!
!
crypto pki token default removal timeout 0
!
!
no ipv6 cef
ip source-route
ip cef
!
!
!
!
!
no ip domain lookup
!
multilink bundle-name authenticated
!
!
isdn switch-type primary-ni
!
!
voice-card 0
 dspfarm
```

```
 dsp services dspfarm[1]
!
!
!
voice service voip
 address-hiding[2]
 allow-connections h323 to h323 \
 allow connections h323 to sip      \
 allow-connections sip to h323      /  [3]
 allow-connections sip to sip        /
  no supplementary-service sip moved-temporarily
 redirect ip2ip
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none[4]
 h323
 sip
  header-passing
  error-passthru
  early-offer forced[5]
  midcall-signaling passthru[6]
  privacy-policy passthru
  g729 annexb-all[7]
!
!
voice class codec 4[8]
 codec preference 1 g729r8 bytes 30
 codec preference 2 g711ulaw
!
voice class sip-profiles 2[9]
 request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:732368\1@\2>"[10]
 response ANY sdp-header Audio-Attribute add "a=ptime:30"[11]
```

---

[1] This command enables DSP farming allowing DSP resources for Media Termination Point (MTP), Conference Bridge (CFB) or Transcoder.

[2] Enables IP address hiding between the private network (CUCME side) and the public network (AT&T FlexReach side)

[3] This command enables the CUCME with CUBE feature to perform basic IP to IP voice communication

[4] This command enables T.38 fax at global level, meaning all VoIP dial-peers not configured for specific fax protocol will use this setting. T.38 fax protocol may be configured under appropriate dial-peers.

[5] This command enables the use of Early Offer SIP INVITE method

[6] This command must be enabled at a global level to maintain integrity of SIP signaling between AT&T network and Cisco Unified Communications Manager Express (CUCME)

[7] This command allows the CUCME to negotiate all flavors of G729 codec and must be configured in order to interoperate seamlessly across AT&T's BVoIP services. The command can either be enabled globally, as in this example, or per dial-peer basis using the "voice-class sip g729 annexb-all" command

[8] This command enables multiple codec support and performs codec filtering required for correct interoperability between AT&T SIP network and Cisco UCME. Payload packet size can also be configured here.

[9] SIP Profiles can be used to manipulate SIP header attributes

[10] This SIP profile expands the Diversion header number from a 4-digit extension to a full 10-digit DID number in order to attain interoperability with AT&T's HIPCS (LEGACY) served users for forwarded calls.

[11] This SIP Profile allows CUCME to advertise desired and supported payload packet size

```
 request INVITE sdp-header Audio-Attribute add "a=ptime:30"
 request REINVITE sdp-header Attribute modify "a=T38FaxFillBitRemoval:0" ""[12]
!
!
voice class custom-cptone CONF
 dualtone conference
  frequency 600 900
  cadence 300 150 300 100 300 50
!
!
voice register global
 mode cme
 source-address 172.20.110.157 port 5060
 max-dn 100
 max-pool 185
 dialplan-pattern 1 .... extension-length 4
 tftp-path flash:
 create profile sync 0069050054835118
!
!!
voice translation-rule 1
 rule 1 /^.*\(....\)/ /732216\1/
!
voice translation-rule 2
 rule 2 /^.*\(....\)/ /\1/
!
voice translation-rule 3
 rule 3 /3143323714/ /2714/

voice translation-profile NPA
 translate calling 1
!
voice translation-profile 10DigitTo4
 translate called 2
!
voice translation-profile LEGACY
 translate called 3
!
application[13]
 service aa flash:app-b-acd-aa-2.1.2.3.tcl
  paramspace english index 0
  param menu-timeout 3
  param handoff-string aa
  param dial-by-extension-option 3
```

---

[12] This SIP profile removes the SDP attribute “T38FaxFillBitRemoval:0” from Cisco IOS gateway upspeed Re-INVITE (inbound call to CPE.) Some SIP components within AT&T’s SIP core do not support the “:0” as the Boolean value, instead some AT&T devices interpret the full attribute as the Boolean value (1=attribute present; 0=attribute not present). For this reason, we remove the attribute completely to achieve fax t.38 interoperability across AT&T’s entire SIP core.

[13] Enables the CUCME BACD auto attendant feature

```
  paramspace english language en
  param operator 6001
  param max-time-vm-retry 2
  param aa-pilot 6999
  param max-extension-length 4
  paramspace english location flash:
  param second-greeting-time 60
  param welcome-prompt _bacd_welcome.au
  param welcome_prompt en_bacd_welcome.au
  param call-retry-timer 15
  param voice-mail 5000
  paramspace english prefix en
  param max-time-call-retry 60
  param service-name queue
 !
 service queue flash:app-b-acd-2.1.2.3.tcl
  param queue-len 10
  param aa-hunt1 3210
  param queue-manager-debugs 1
  param number-of-hunt-grps 2
  param aa-hunt2 3211
 !
!
license udi pid C3900-SPE100/K9 sn FOC14341EZJ
hw-module ism 0
!
hw-module pvdm 0/0
!

!
username cisco privilege 15 secret 5 $1$cj/.$dU5cRW4lb02gUU.zgc1ME0
!
redundancy
!
!
controller t1 1/0
 pri-group timeslots 1-24

!
translation-rule 1
!

!
interface Loopback0
 ip address 172.1.1.1 255.255.255.0
!
interface GigabitEthernet0/0
 description $ETH-LAN$$ETH-SW-LAUNCH$$INTF-INFO-GE 0/0$
 ip address 172.20.110.157 255.255.255.0
 duplex auto
```

```
 speed auto
!
interface ISM0/0[14]
 ip unnumbered GigabitEthernet0/0
 service-module ip address 172.20.110.158 255.255.255.0
 !Application: CUE Running on ISM
 service-module ip default-gateway 172.20.110.1
!
interface GigabitEthernet0/1
 description connection to ATT Network
 ip address 70.132.127.108 255.255.255.0
 duplex auto
 speed auto
!
interface GigabitEthernet0/2
no ip address
duplex auto
 speed auto
!
interface Serial1/0:23
no ip address
encapsulation hdlc
isdn switch-type primary-ni
isdn incoming-voice voice
no cdp enable
!
interface ISM0/1
 description Internal switch interface connected to Internal Service Module
 shutdown
!
interface Vlan1
 no ip address
!
!
ip forward-protocol nd
!
no ip http server
no ip http secure-server
!
ip route 172.20.0.0 255.255.0.0 172.20.110.1
ip route 172.20.110.158 255.255.255.255 ISM0/0
ip route 207.242.225.200 255.255.255.255 70.132.127.110
ip route 207.242.225.210 255.255.255.255 70.132.127.110
!
logging esm config
access-list 101 permit ip host 207.242.225.200 host 70.132.127.108
access-list 101 permit ip host 207.242.225.210 host 70.132.127.108
access-list 101 permit ip host 70.132.127.108 host 207.242.225.200
```

---

[14] Cisco Unity Express Interface

```
access-list 101 permit ip host 70.132.127.108 host 207.242.225.210
!
!
!
!
nls resp-timeout 1
cpd cr-id 1
!
tftp-server flash0:dsp75.9-1-1TH1-16.sbn
tftp-server flash0:jar75sccp.9-1-1TH1-16.sbn
tftp-server flash0:SCCP75.9-1-1SR1S.loads
tftp-server flash0:term75.default.loads
tftp-server flash0:apps75.9-1-1TH1-16.sbn
tftp-server flash0:cnu75.9-1-1TH1-16.sbn
tftp-server flash0:cvm75sccp.9-1-1TH1-16.sbn
tftp-server flash0:jar41sccp.9-1-1TH1-16.sbn
tftp-server flash0:SCCP41.9-1-1SR1S.loads
tftp-server flash0:term41.default.loads
tftp-server flash0:term61.default.loads
tftp-server flash0:apps41.9-1-1TH1-16.sbn
tftp-server flash0:cnu41.9-1-1TH1-16.sbn
tftp-server flash0:cvm41sccp.9-1-1TH1-16.sbn
tftp-server flash0:dsp41.9-1-1TH1-16.sbn
!
control-plane
!
!
voice-port 0/1/0
!
voice-port 0/1/1
!
voice-port 0/1/2
!
voice-port 0/1/3
!
voice-port 1:0:23
!
!
sccp local GigabitEthernet0/0[15]
sccp ccm 172.20.110.157 identifier 1 version 7.0
sccp
!
sccp ccm group 100
 bind interface GigabitEthernet0/0
 associate ccm 1 priority 1
 associate profile 2 register con1cdf0f0afe00
 associate profile 1 register MTP1cdf0f0afe00
!
```

---

[15] These sccp commands configure the shared DSP resources as conference bridge (CFB) and transcoder device for the CUCME.

```
dspfarm profile 1 transcode
 codec g711ulaw
 codec g711alaw
 codec g729ar8
 codec g729abr8
 codec g729r8
 codec g729br8
 codec g722-64
 maximum sessions 3
 associate application SCCP
!
dspfarm profile 2 conference
 codec g711ulaw
 codec g711alaw
 codec g729ar8
 codec g729abr8
 codec g729r8
 codec g729br8
 codec g722-64
 maximum sessions 4
 conference-join custom-cptone CONF
 associate application SCCP
!
dial-peer voice 1999 voip
 description OUTGOING CALL TO AT&T FACING AT&T NETWORK
 translation-profile outgoing NPA
 destination-pattern 1T
session protocol sipv2
 session target ipv4:207.242.225.210[16]
 incoming called-number 732.......
 voice-class codec 4[17]
 voice-class sip asserted-id pai[18]
 voice-class sip privacy-policy passthru[19]
 voice-class sip early-offer forced[20]
 voice-class sip profiles 2[21]
 dtmf-relay rtp-nte
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none[22]
 no vad
!
```

---

[16] This command sets the SIP server target for outgoing SIP calls.

[17] This command assigns the voice class codec setting to this dial-peer

[18] This command enables the delivery of caller id information using P-asserted-ID method, across the SIP trunk. This command can either be issued globally or per dial peer.

[19] This command allows for privacy settings to be transparently passed between AT&T network and CUCME. This command can either be issued globally or per dial peer.

[20] This commands enables Early Offer SIP invite method.

[21] This commands assigns the applicable SIP profile to use for this dial-peer

[22] Example of configuring T38 as fax protocol per dial peer

```
dial-peer voice 19991 voip
 description ALTERNATE ROUTE FOR OUTGOING CALL TO AT&T FACING AT&T NETWQRK
 preference 1
 translation-profile incoming NPA
 translation-profile outgoing NPA
 destination-pattern 1T
session protocol sipv2
 session target ipv4:1.1.1.1
 incoming called-number 732.......
 voice-class codec 4
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
 voice-class sip profiles 2
 dtmf-relay rtp-nte
 fax rate 14400
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 19992 voip
 description ALTERNATE ROUTE TO PSTN
 preference 2
 destination-pattern 1T
 direct-inward-dial
 translate-outgoing calling 1
 port 1/0:23
 forward-digits all
!
dial-peer voice 1998 voip
 description OUTGOING CALL TO BENCH PHONE VIA AT&T LEGACY NETWORK
 destination-pattern 3143463905
translation-profile outgoing NPA
 session protocol sipv2
 session target ipv4:207.242.225.210
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
 voice-class sip profiles 2
 dtmf-relay rtp-nte
 fax rate 14400
no vad
!
dial-peer voice 1997 voip
 description OUTGOING CALL TO BENCH PHONE VIA AT&T CISCO GW
 destination-pattern 9142223902
 translation-profile outgoing NPA
 session protocol sipv2
 session target ipv4:207.242.225.210
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
```

```
 voice-class sip early-offer forced
 voice-class sip profiles 2
 dtmf-relay rtp-nte
 fax rate 14400
 no vad
!
dial-peer voice 1141 voip
 description OUTGOING INTERNATIONAL CALL TO AT&T FACING AT&T NETWORK
 translation-profile outgoing NPA
 destination-pattern 01141583330158
 session protocol sipv2
 session target ipv4:207.242.225.210
 incoming called-number 314.......
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
 dtmf-relay rtp-nte
 fax rate 14400
 no vad
!
dial-peer voice 511 voip
 description OUTGOINT N11 CALL TO AT&T FACING AT&T NETWORK
translation-profile outgoing NPA
destination-pattern 911
 session protocol sipv2
 session target ipv4:207.242.225.210
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
 voice-class sip profiles 2
 dtmf-relay rtp-nte
 fax rate 14400
 no vad
!
dial-peer voice 1000 voip
 description INCOMING DIAL PEER FACING AT&T NETWORK
 translation-profile incoming 10DigitTo4
 session protocol sipv2
 incoming called-number 732216....
 voice-class codec 4
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
 voice-class sip profiles 2
 dtmf-relay rtp-nte
 fax rate 14400
 no vad
!
dial-peer voice 1996 voip
 description INCOMING DIAL PEER FROM AT&T LEGACY NETWORK
```

```
 translation-profile incoming LEGACY
 destination-pattern 1T
 session protocol sipv2
 session target ipv4:207.242.225.210
 incoming called-number 3143323714
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
 voice-class sip profiles 2
 dtmf-relay rtp-nte
 codec g722-64
 fax rate 14400
!
dial-peer voice 4715 pots
 description ONLY USED FOR FAX TEST CALLS
 shutdown
 destination-pattern 2715
 port 0/1/2
!
dial-peer voice 4717 pots
description ONLY USED FOR FAX TEST CALLS
 shutdown
 destination-pattern 2717
 port 0/1/3
!
dial-peer voice 8715 voip
 description OUTGOING AND INCOMING FAX OVER G711 CALL TO/FROM AT&T FACING AT&T NETWORK
 translation-profile outgoing NPA
 destination-pattern 1T
 session protocol sipv2
 session target ipv4:207.242.225.210
 incoming called-number 7322162715
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
 voice-class sip profiles 2
 dtmf-relay rtp-nte
 playout-delay nominal 80
 playout-delay mode fixed
 codec g711ulaw
 no fax-relay sg3-to-g3
 fax rate 14400
 no vad
!
dial-peer voice 8717 voip
 description INCOMING FAX OVER G711 CALL FROM AT&T LEGACY NETWORK
 translation-profile incoming LEGACY
 destination-pattern 1T
 session protocol sipv2
 session target ipv4:207.242.225.210
```

```
 incoming called-number 3143323714
 voice-class sip asserted-id pai
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
 voice-class sip profiles 2
 dtmf-relay rtp-nte
 playout-delay nominal 80
 playout-delay mode fixed
 codec g711ulaw
 no fax-relay sg3-to-g3
 no vad
!
!
sip-ua
 mwi-server ipv4:172.20.110.158 expires 3600 port 5060 transport tcp unsolicited
!
!
!
gatekeeper
 shutdown
!
!
telephony-service
 sdspfarm units 5
 sdspfarm transcode sessions 128
 sdspfarm tag 1 con1cdf0f0afe00
 sdspfarm tag 2 MTP1cdf0f0afe00
 conference hardware
 no auto-reg-ephone
 em logout 0:0 0:0 0:0
 max-ephones 96
 max-dn 192
 ip source-address 172.20.110.157 port 2000
 service phone g722CodecSupport 2
 load 7961 sccp41.9-1-1SR1S.loads
 load 7962 SCCP42.8-5-4S.loads
 load 7975 SCCP75.9-1-1SR1S.loads
 voicemail 5000
 mwi relay
 max-conferences 4 gain -6
 call-forward pattern .T
 moh music-on-hold.au
 web admin system name cisco password cisco
 dn-webedit
 time-webedit
 transfer-system full-consult
 transfer-pattern .T
 secondary-dialtone 9
 create cnf-files version-stamp Jan 01 2002 00:00:00
!
```

```
!
ephone-template  1
 softkeys idle  Cfwdall ConfList Dnd Gpickup HLog Join Login Newcall Pickup Redial RmLstC
 softkeys seized  Redial Pickup Gpickup HLog Meetme Endcall
 softkeys connected  Acct ConfList Confrn Endcall Flash HLog Hold Join Park RmLstC Select Trnsfer
!
!
ephone-dn  1  dual-line
 number 2714
 label Phone A
 name Phone A
 call-forward busy 5000
 call-forward noan 5000 timeout 10
 translate calling 1
!
!
ephone-dn  2  dual-line
 number 2715
 label Phone B
 name Phone B
!
!
ephone-dn  3  dual-line
 number 2716
 label Phone C
 name Phone C
!
!
ephone-dn  4  dual-line
 number 2717
 label Phone D
 name Phone D
!
!
ephone-dn  5  dual-line
 number 3143323714
!
!
ephone-dn  6  dual-line
 number 3143323715
 label Phone F
 name Phone F
!
!
ephone-dn  10  dual-line
 number 6245
 name Chadi Testing
!
!
ephone-dn  30  dual-line
```

```
 number 3333
 conference meetme
 preference 1
 no huntstop
!
!
ephone-dn  31  dual-line
 number 3333
 conference meetme
 preference 2
 no huntstop
!
!
ephone-dn  32  dual-line
 number 3333
 conference meetme
 preference 3
 no huntstop
!
!
ephone-dn  33  dual-line
 number 3333
 conference meetme
 preference 4
!
!
ephone-dn  34  dual-line
 number 4444
 name conference
 conference ad-hoc
 no huntstop
!
!
ephone-dn  35  dual-line
 number 4444
 name conference
 conference ad-hoc
 preference 1
 no huntstop
!
!
ephone-dn  36  dual-line
 number 4444
 name conference
 conference ad-hoc
 preference 2
 no huntstop
!
!
ephone-dn  37  dual-line
```

```
 number 4444
 name conference
 conference ad-hoc
 preference 3
!
!
ephone-dn  80
 number 8000....
 mwi on
!
!
ephone-dn  81
 number 8001....
 mwi off
!
!
ephone  1
 device-security-mode none
 mac-address 001D.705F.B383
 ephone-template 1
 codec g722-64
 type 7975
 button  1:1 2:5
!
!
!
ephone  2
 device-security-mode none
 mac-address 001B.0CDB.B399
 ephone-template 1
 codec g722-64
 type 7961
 button  1:2 2:6
!
!
!
ephone  3
 device-security-mode none
 mac-address 0017.0EEE.3279
 ephone-template 1
 codec g722-64
 type 7961
 button  1:3
!
!
!
ephone  4
 device-security-mode none
 mac-address 001B.0CDB.CB27
 ephone-template 1
```

```
 codec g722-64
 type 7961
 button  1:4
!
!
!
ephone  5
 device-security-mode none
 codec g722-64
!
!
!
ephone  6
 device-security-mode none
 codec g722-64
!
!
!
ephone  10
 device-security-mode none
 mac-address 0023.0433.CF07
 codec g722-64
 type 7962
 button  1:10
!
!
sip-ua
 retry invite 2[23]
 no remote-party id
!
!
line con 0
 password cisco
 login
 stopbits 1
line aux 0
 stopbits 1
line 195
 no activation-character
 no exec
 transport preferred none
 transport input all
 transport output lat pad telnet rlogin lapb-ta mop udptn v120 ssh
 stopbits 1
line vty 0 4
 access-class 23 in
 exec-timeout 15 0
 privilege level 15
```

---

[23] This command should be enabled when failover dial-peers are configured.

```
 password cisco
 login
 transport input telnet
!
exception data-corruption buffer truncate
scheduler allocate 20000 1000
end
```

Cisco Unity Express (CUE)

```
se-172-20-110-158# sh run

Generating configuration:


clock timezone UTC

hostname se-172-20-110-158

line console

system language preferred "en_US"

ntp server 172.20.110.157 prefer

software download server url "ftp://127.0.0.1/ftp" credentials hidden
"6u/dKTN/hsEuSAEfw40XlF2eFHnZfyUTSd8ZZNgd+Y9J3xlk2B35j
0nfGWTYHfmPSd8ZZNgd+Y9J3xlk2B35j0nfGWTYHfmPSd8ZZNgd+Y9J3xlk2B35j0nfGWTYHfmP"

license agent max-sessions 9

privilege ViewRealTimeReports create
privilege ViewPrivateList create
privilege manage-passwords create
privilege manage-users create
privilege ViewHistoricalReports create
privilege ManagePublicList create
privilege vm-imap create
privilege local-broadcast create
privilege ManagePrompts create
privilege broadcast create

groupname Broadcasters create

username PhoneA create
username cisco create
username Bruno create
```

```
privilege ViewRealTimeReports description "Privilege to view realtime reports"
privilege ViewPrivateList description "Privilege to view private list"
privilege manage-passwords description "Privilege to reset user passwords"
privilege manage-users description "Privilege to create, modify, and delete users and groups"
privilege ViewHistoricalReports description "Privilege to view historical reports"
privilege ManagePublicList description "Privilege to manage public lists"
privilege vm-imap description "Privilege to manage personal voicemail via IMAP client"
privilege local-broadcast description "Privilege to send local broadcast messages"
privilege ManagePrompts description "Privilege to create, modify, or delete system prompts"
privilege broadcast description "Privilege to send local or remote broadcast messages"
privilege ViewRealTimeReports operation report.realtime
privilege ViewPrivateList operation voicemail.lists.private.view
privilege manage-passwords operation user.pin
privilege manage-passwords operation user.password
privilege manage-passwords operation system.debug
privilege manage-users operation user.mailbox
privilege manage-users operation user.remote
privilege manage-users operation user.pin
privilege manage-users operation user.password
privilege manage-users operation user.notification
privilege manage-users operation group.configuration
privilege manage-users operation system.debug
privilege manage-users operation user.configuration
privilege ViewHistoricalReports operation report.historical.view
privilege ManagePublicList operation voicemail.lists.public
privilege ManagePublicList operation system.debug
privilege vm-imap operation voicemail.imap.user
privilege local-broadcast operation broadcast.local
privilege local-broadcast operation system.debug
privilege ManagePrompts operation prompt.modify
privilege ManagePrompts operation system.debug
privilege broadcast operation broadcast.remote
privilege broadcast operation broadcast.local
privilege broadcast operation system.debug

groupname Administrators member cisco
groupname Broadcasters privilege broadcast

username PhoneA phonenumber "2714"

restriction msg-notification create
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/VNXHCkplVV4KjescbDaLa4fl4WLSPFvv1rWUnfGWTYHfmP
Sd8ZZNgd+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 aa
 description "autoattendant"
 enabled
 maxsessions 10
 script "aa.aef"
 parameter "dialByExtnAnytime" "false"
 parameter "busOpenPrompt" "AABusinessOpen.wav"
 parameter "dialByExtnAnytimeInputLength" "4"
 parameter "operExtn" "1001"
 parameter "welcomePrompt" "AAWelcome.wav"
 parameter "disconnectAfterMenu" "false"
 parameter "dialByFirstName" "false"
 parameter "busClosedPrompt" "AABusinessClosed.wav"
 parameter "allowExternalTransfers" "false"
 parameter "holidayPrompt" "AAHolidayPrompt.wav"
 parameter "businessSchedule" "systemschedule"
 parameter "MaxRetry" "3"
 end application

ccn application ciscomwiapplication aa
 description "ciscomwiapplication"
 enabled
 maxsessions 10
 script "setmwi.aef"
 parameter "CallControlGroupID" "0"
 parameter "strMWI_OFF_DN" "8001"
 parameter "strMWI_ON_DN" "8000"
 end application

ccn application msgnotification aa
 description "msgnotification"
 enabled
 maxsessions 10
 script "msgnotify.aef"
 parameter "logoutUri" "http://localhost/voicemail/vxmlscripts/mbxLogout.jsp"
 parameter "DelayBeforeSendDTMF" "1"
 end application

ccn application promptmgmt aa
 description "promptmgmt"
```

```
 enabled
 maxsessions 10
 script "promptmgmt.aef"
 parameter "appManagementScript" ""
 end application

ccn application voicemail aa
 description "voicemail"
 enabled
 maxsessions 10
 script "voicebrowser.aef"
 parameter "uri" "http://localhost/voicemail/vxmlscripts/login.vxml"
 parameter "logoutUri" "http://localhost/voicemail/vxmlscripts/mbxLogout.jsp"
 end application

ccn engine
 end engine

ccn reporting historical
 database local
 description "se-172-20-110-158"
 end reporting

ccn subsystem jtapi
 ccm-manager address 0.0.0.0
 end subsystem

ccn subsystem sip
 gateway address "172.20.110.157"
 end subsystem

ccn trigger http urlname msgnotifytrg
 application "msgnotification"
 enabled
 maxsessions 2
 end trigger

ccn trigger http urlname mwiapp
 application "ciscomwiapplication"
 enabled
 maxsessions 1
 end trigger

ccn trigger sip phonenumber 5000
 application "voicemail"
 enabled
 maxsessions 2
 end trigger

service voiceview
```

```
 enable
 end voiceview

voicemail broadcast recording time 300
voicemail default messagesize 240
voicemail notification restriction msg-notification
voicemail mailbox owner "PhoneA" size 300
 expiration time 10
 messagesize 120
 end mailbox

end
se-172-20-110-158#
```

Acronyms

Acronym	Definitions
SIP	Session Initiation Protocol
MGCP	Media Gateway Control Protocol
SCCP	Skinny Client Control Protocol
CISCO UCM	Cisco Unified Communications Manager
CISCO UBE	Cisco Unified Border Element
SP	Service Provider
PSTN	Public switched telephone network
IP	Internet Protocol
TDM	Time-division multiplexing
CODEC	Coder-Decoder (in this document a device used to digitize and undigitize voice signals)

Important Information

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

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
USA
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

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