

**AT&T IP Flexible Reach service on MIS, PNT or AVPN transport: Connecting Cisco Unified Communications Manager Express 10.0 to AT&T IP Flexible Reach with Enhanced Features using SIP**

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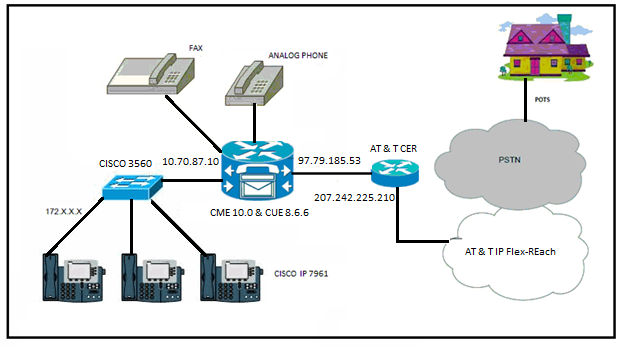
# **Introduction**

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 Flexible Reach 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.

* This application note describes how to configure a Cisco Unified Communications Manager Express (Cisco Unified CME) 10.0 ((IOS 15.3(3) M1)), Cisco Unity Express 8.6.6 with connectivity to AT&T’s IP Flexible 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 Unified CME/Cisco Unity Express) to PSTN (AT&T IP Flexible Reach SIP). AT&T IP Flexible Reach provides inbound and outbound call service.
* Testing was performed in accordance to AT&T’s IP Flexible 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, CODEC negotiation, 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)and teleconferencing.
* 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.

# **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.

## Hardware Components

* Cisco Integrated Service Router G2. This solution was tested with CISCO2921 but this application note applies to any ISR G2 platform.
* Cisco IP Phones. This solution was tested with 7961 and 7975 phones, but any Cisco IP Phone model supporting RFC2833 can be used).
* Cisco CISCO2921/K9 (revision 1.0) with 999424K/49152K bytes of memory. Processor board ID FTX1719AHKS.
* Internal Services Module (ISM) with Services Ready Engine (SRE), Cisco Unity Express 8.6.6 in slot/sub-slot 0/0.
* Service Module (SM) – SM-SRE-700-K9 or SM-SRE-900-K9.
* 4 Gigabit Ethernet interfaces, 2 terminal lines and 2 Voice FXS interfaces.

## Software Requirements

* Cisco IOS gateway running Cisco Unified Communication Manager Express (CUCME) 10.0 Version 15.3(3) M1, RELEASE SOFTWARE (fc1). This solution was tested with Cisco IOS image: "flash:c2900-universalk9-mz.SPA.153-3.M1.bin"
* 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 (8.6.6)

# **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
* CME: 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 ATT SIP network)

## Network Based Features – Supported

* Call forward (Unconditional, Busy, No Answer, Not reachable)
* Transfer (Blind, Consultative)
* Sequential Ringing
* Simultaneous Ringing

NOTE: Using the AT&T IP Flexible Reach Portal, provision TN(s) on the CPE with the Sequential Ring and simultaneous feature. Provisioning is self-explanatory. Please contact your AT&T representative, if you need help with the provisioning Network based feature.

## Features Not Supported

* Cisco Unified CME 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 14400 kbps.
* For T38 test related scenario achieved using “fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none” on dial peer
* For G711Passthrough test achieved using “fax protocol pass-through g711ulaw”

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.

Cisco Unity Express (CUE):

* CME CUE only supports G711.

Hold & Resume:

* CME consumes the re-invites for hold and resume scenarios. Re-invites for hold/resume from the network would potentially depend on the carrier/network the call is traversing.

PBX Based Call Forward Unconditional:

* PBX Based Unconditional Call Forwarding test is temporarily blocked due to AT&T Flexible Reach network issue.

# **Configuration considerations**

* When using G.729 between AT&T IP Flexible Reach and Cisco Unified Communication Manager Express (Cisco Unified CME) 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 Flexible 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 Unified CME 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 CME (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 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.
* While transferring from auto attend to station from PSTN caller needs converting 4 digit number to 10 digit on SIP profile intended for PSTN understandable configuration. For Example - request REFER sip-header Refer-To modify "sip:2712@97.x.x.x" "sip:7322162712@97.x.x.x”
* SIP Profiles may also be employed to advertise desired RTP payload packet size.
* “voice-class sip privacy id” is need to configure to make call From a CPE Phone to some PSTN phone; Pass Calling Party Number (CPN), marked private and Verify display at called party phone.

# **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>. 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.
* Emergency and 411 calls were terminated to a voicemail platform within AT&T.

# **Configuration**

## Cisco IOS Version

ATTcme#show version

Cisco IOS Software, C2900 Software (C2900-UNIVERSALK9-M), Version 15.3(3)M1, RELEASE SOFTWARE (fc1)

Technical Support: http://www.cisco.com/techsupport

Copyright (c) 1986-2013 by Cisco Systems, Inc.

Compiled Tue 22-Oct-13 01:08 by prod\_rel\_team

ROM: System Bootstrap, Version 15.0(1r)M16, RELEASE SOFTWARE (fc1)

ATTcme uptime is 5 days, 20 hours, 27 minutes

System returned to ROM by reload at 13:32:58 CST Thu Jan 23 2014

System restarted at 13:33:50 CST Thu Jan 23 2014

System image file is "flash:c2900-universalk9-mz.SPA.153-3.M1.bin"

Last reload type: Normal Reload

Last reload reason: Reload Command

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 CISCO2921/K9 (revision 1.0) with 999424K/49152K bytes of memory.

Processor board ID FTX1719AHKS

4 Gigabit Ethernet interfaces

2 terminal lines

2 Voice FXS interfaces

1 Internal Services Module (ISM) with Services Ready Engine (SRE)

Cisco Unity Express 8.6.6 in slot/sub-slot 0/0

DRAM configuration is 64 bits wide with parity enabled.

255K bytes of non-volatile configuration memory.

2048256K bytes of ATA System CompactFlash 0 (Read/Write)

License Info:

License UDI:

-------------------------------------------------

Device# PID SN

-------------------------------------------------

\*0 CISCO2921/K9 FTX1719AHKS

Technology Package License Information for Module:'c2900'

------------------------------------------------------------------------

Technology Technology-package Technology-package

Current Type Next reboot

------------------------------------------------------------------------

ipbase ipbasek9 Permanent ipbasek9

security None None None

uc uck9 Permanent uck9

data None None None

appx None None None

NtwkEss None None None

CollabPro None None None

Configuration register is 0x2102

## Cisco Unified Communication Manager Express (CUCME)

ATTcme#show running-configuration

Building configuration...

Current configuration : 14886 bytes

!

! Last configuration change at 09:47:18 CST Tue Dec 17 2013 by cisco

version 15.3

service timestamps debug datetime msec

service timestamps log datetime msec

no service password-encryption

!

hostname ATTcme

!

boot-start-marker

boot system flash:c2900-universalk9-mz.SPA.153-3.M1.bin

boot-end-marker

!

aqm-register-fnf

!

logging queue-limit 10000

logging buffered 10000000

logging rate-limit 10000

enable secret 5 $1$R/..$STLJKPXMGBWR.jAsKL6uH.

!

no aaa new-model

clock timezone CST -6 0

clock summer-time CDT recurring

!

!

!

!

!

!

!

ip domain name lab.tekvizion.com

ip name-server 10.64.1.3

ip cef

no ipv6 cef

multilink bundle-name authenticated

!

!

!

!

!

!

crypto pki trustpoint TP-self-signed-2928850252

enrollment selfsigned

subject-name cn=IOS-Self-Signed-Certificate-2928850252

revocation-check none

rsakeypair TP-self-signed-2928850252

!

!

crypto pki certificate chain TP-self-signed-2928850252

certificate self-signed 01

3082022B 30820194 A0030201 02020101 300D0609 2A864886 F70D0101 05050030

31312F30 2D060355 04031326 494F532D 53656C66 2D536967 6E65642D 43657274

69666963 6174652D 32393238 38353032 3532301E 170D3133 31303231 31373432

34305A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649

4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D32 39323838

35303235 3230819F 300D0609 2A864886 F70D0101 01050003 818D0030 81890281

81009226 CFC83B24 9EBDF5F2 FE40A30E 2076617A 3B513827 42CC51E6 2D3F058A

BF332FC2 1FC52ECB 8416DB72 86B90AF7 8E9340C9 D12855E2 BCBD739C 19BBCFFF

741827E4 5854F71A 1ADD3107 594E6DE8 15B9C832 3AEB959C C89531BE 804D156B

18449BFC 8E2E280F 9FE237EC 23065CAE A493463E 56517123 D597E662 2092EB91

39F50203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF 301F0603

551D2304 18301680 14B921D8 04919022 6C5E16A1 B34CBF9F 2651DB07 0B301D06

03551D0E 04160414 B921D804 9190226C 5E16A1B3 4CBF9F26 51DB070B 300D0609

2A864886 F70D0101 05050003 8181007A 746D01F4 DC78EFF6 58B8296B 5A754512

2C2D99C0 BC845B85 59C5F197 5887D34C 498AC6B2 530D829F 53D9D013 16D0C37F

67C0D27A F25978EB DB7C529F 9EFECB7A 2F2F5FB3 D7C9870E 9AAE981B 9AFA5E25

E72D1308 CFD52D96 7DB4E663 2B954327 89D4B76D 203FD733 D50B0724 D441A323

5704A4CC 2BDB8BB8 9C5936FD 34064D

quit

voice-card 0

dspfarm

dsp services dspfarm **1**

!

!

!

voice service pots

fax rate disable

!

voice service voip

no ip address trusted authenticate

callmonitor

address-hiding **2**

allow-connections h323 to h323 **3**

allow-connections h323 to sip **3**

allow-connections sip to h323 **3**

allow-connections sip to sip **3**

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

registrar server expires max 3600 min 180

early-offer forced **5**

midcall-signaling passthru **6**

privacy-policy passthru

g729 annexb-all **7**

!

voice class codec 4 **8**

codec preference 1 g729br8

codec preference 2 g729r8

codec preference 3 g711ulaw

!

voice class sip-profiles 2 **9**

request REFER sip-header Refer-To modify "sip:2712@97.79.185.53"

"sip:7322162712@97.79.185.53" **10**

request INVITE sip-header Diversion modify

"<sip:(.\*)@(.\*)>" "sip:732216\1@\2>" **11**

response ANY sdp-header Audio-Attribute add "a=ptime:30" **12**

request REINVITE sdp-header Attribute modify "a=T38FaxFillBitRemoval:0" "" **13**

!

voice class custom-cptone CONF

dualtone conference

frequency 600 900

cadence 300 150 300 100 300 50

!

!

voice register global

mode cme

source-address 10.70.87.10 port 5060

max-dn 100

max-pool 110

load 7975 SIP75.9-2-1S

mwi stutter

mwi reg-e164

dialplan-pattern 1 .... extension-length 4

voicemail 5000

tftp-path flash:

file text

create profile sync 0069050054835115

!

!

!

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 10DigitTo4

translate called 2

!

voice translation-profile LEGACY

translate called 3

!

voice translation-profile NPA

translate calling 1

!

!

!

!

Application **14**

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

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 call-retry-timer 15

param welcome\_prompt en\_bacd\_welcome.au

param max-time-call-retry 60

paramspace english prefix en

param voice-mail 5000

param service-name queue

!

service queue flash:app-b-acd-2.1.2.3.tcl

param queue-len 10

param aa-hunt1 3210

param aa-hunt2 3211

param number-of-hunt-grps 2

param queue-manager-debugs 1

!

!

license udi pid CISCO2921/K9 sn FTX1719AHKS

hw-module ism 0

!

hw-module pvdm 0/0

!

!

!

username cisco privilege 15 secret 5 $1$nLFW$WRszOoW9rJOVk52mlOTeP1

!

redundancy

!

!

ip ftp username cisco

ip ftp password cisco

!

translation-rule 101

Rule 1 0 8772888362

!

!

!

!

!

interface Embedded-Service-Engine0/0

no ip address

shutdown

!

interface GigabitEthernet0/0

description Wan Interface to AT&T trunk one

ip address 97.79.185.53 255.255.255.224

duplex auto

speed auto

!

interface ISM0/0 **15**

description unity express module

ip unnumbered GigabitEthernet0/1

service-module ip address 10.70.87.11 255.255.255.0

!Application: CUE Running on ISM

service-module ip default-gateway 10.70.87.1

!

interface GigabitEthernet0/1

description Lan Interface to Test network

ip address 10.70.87.10 255.255.255.0

duplex auto

speed auto

media-type rj45

!

interface GigabitEthernet0/2

no ip address

duplex auto

speed auto

!

interface ISM0/1

description Internal switch interface connected to Internal Service Module

no ip address

shutdown

!

interface Vlan1

no ip address

!

ip default-gateway 97.79.185.62

ip forward-protocol nd

!

ip http server

ip http authentication local

ip http secure-server

ip http secure-port 8443

ip http timeout-policy idle 60 life 86400 requests 10000

ip http path flash:

!

ip route 0.0.0.0 0.0.0.0 97.79.185.62

ip route 10.0.0.0 255.0.0.0 10.70.87.1

ip route 10.70.87.11 255.255.255.255 ISM0/0

ip route 172.16.0.0 255.255.0.0 10.70.87.1

!

!

!

tftp-server flash:apps75.9-2-1TH1-13.sbn

tftp-server flash:cnu75.9-2-1TH1-13.sbn

tftp-server flash:cvm75sip.9-2-1TH1-13.sbn

tftp-server flash:dsp75.9-2-1TH1-13.sbn

tftp-server flash:jar75sip.9-2-1TH1-13.sbn

tftp-server flash:SIP75.9-2-1S.loads

tftp-server flash:term75.default.loads

tftp-server flash:apps41.9-2-1TH1-13.sbn

tftp-server flash:cnu41.9-2-1TH1-13.sbn

tftp-server flash:cvm41sccp.9-2-1TH1-13.sbn

tftp-server flash:dsp41.9-2-1TH1-13.sbn

tftp-server flash:jar41sccp.9-2-1TH1-13.sbn

tftp-server flash:SCCP41.9-2-1S.loads

tftp-server flash:term41.default.loads

tftp-server flash:term61.default.loads

!

control-plane

!

!

voice-port 0/1/0

cptone IN

station-id number 2712

caller-id enable

!

voice-port 0/1/1

cptone IN

station-id number 2778

caller-id enable

!

!

!

!

!

!

mgcp behavior rsip-range tgcp-only

mgcp behavior comedia-role none

mgcp behavior comedia-check-media-src disable

mgcp behavior comedia-sdp-force disable

!

mgcp profile default

!

sccp local GigabitEthernet0/1 **16**

sccp ccm 10.70.87.10 identifier 1 version 7.0

sccp

!

sccp ccm group 100

bind interface GigabitEthernet0/1

associate ccm 1 priority 1

associate profile 2 register conference

associate profile 1 register transcode

!

**dspfarm profile 1 transcode**

**codec g729abr8**

**codec g729ar8**

**codec g711alaw**

**codec g711ulaw**

**codec g729r8**

**codec g729br8**

**codec g722-64**

**maximum sessions 3**

**associate application SCCP**

**!**

**dspfarm profile 2 conference**

**codec g729br8**

**codec g729r8**

**codec g729abr8**

**codec g729ar8**

**codec g711alaw**

**codec g711ulaw**

**codec g722-64**

**maximum sessions 3**

**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

preference 2

destination-pattern 1T

session protocol sipv2

session target ipv4:207.242.225.210 **17**

incoming called-number 732.......

voice-class codec 4 **18**

voice-class sip asserted-id pai **19**

no voice-class sip privacy-policy passthru **20**

voice-class sip early-offer forced **21**

voice-class sip profiles 2 **22**

dtmf-relay rtp-nte

fax rate 14400

fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none **23**

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

rtp payload-type nse 99

rtp payload-type nte 100

modem passthrough nse codec g711ulaw

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

playout-delay nominal 80

playout-delay mode fixed

no fax-relay sg3-to-g3

fax rate disable

no vad

!

dial-peer voice 1141 voip

description OUTGOING INTL CALL TO AT&T FACING ATT NW

translation-profile outgoing NPA

destination-pattern 01141583330158

session protocol sipv2

session target ipv4:207.242.225.210

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

no vad

!

dial-peer voice 109 voip

description Unity Express - ATTEND

destination-pattern 2300

session protocol sipv2

session target ipv4:10.70.87.11

dtmf-relay rtp-nte

codec g711ulaw

!

dial-peer voice 102 voip

description \*\*Star Code to SIP Trunk\*\*

translation-profile outgoing NPA

destination-pattern \*..

session protocol sipv2

session target ipv4:207.242.225.210

voice-class codec 4

voice-class sip asserted-id pai

voice-class sip privacy-policy passthru

voice-class sip dtmf-relay force rtp-nte

voice-class sip early-offer forced

voice-class sip profiles 2

dtmf-relay rtp-nte

no vad

!

dial-peer voice 1001 pots

preference 1

service session

destination-pattern 2712

no digit-strip

port 0/1/0

!

dial-peer voice 1002 pots

preference 1

service session

destination-pattern 2778

no digit-strip

port 0/1/1

!

!

dial-peer voice 101 voip

description Unity Express - VoiceMail

destination-pattern 50..

session protocol sipv2

session target ipv4:10.70.87.11

dtmf-relay rtp-nte

codec g711ulaw

!

dial-peer voice 1444 voip

description Operator call to 0 test

translation-profile outgoing NPA

preference 2

destination-pattern 0

translate-outgoing called 101

session protocol sipv2

session target ipv4:207.242.225.210

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

no vad

!

!

sip-ua

no remote-party-id

retry invite 2

mwi-server ipv4:10.70.87.11 expires 3600 port 5060 transport tcp unsolicited

!

!

!

gatekeeper

shutdown

!

!

telephony-service

sdspfarm units 5

sdspfarm transcode sessions 128

sdspfarm tag 1 transcode

sdspfarm tag 2 conference

no privacy

conference hardware

no auto-reg-ephone

authentication credential admin admin

em logout 0:0 0:0 0:0

max-ephones 5

max-dn 192

ip source-address 10.70.87.10 port 2000

caller-id block code \*123

timeouts interdigit 2

cnf-file location flash:

load 7961 SCCP41.9-2-1S.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 7960 Dec 16 2013 10:40:49

!

!

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 2 dual-line

number 2710

translate calling 1

!

!

ephone-dn 3 dual-line

number 2711

call-forward noan 5005 timeout 10

translate calling 1

!

!

ephone-dn 4 dual-line

caller-id block

!

!

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 55 dual-line

number 2755

label test one

name test one

!

!

ephone-dn 80

number 8000....

mwi on

!

!

ephone-dn 81

number 8001....

mwi off

!

!

!

ephone 2

mac-address 0022.905A.BA0D

ephone-template 1

username "phone C" password 1234

type 7961

button 1:2

!

!

!

ephone 3

mac-address 001D.A21A.26CA

ephone-template 1

username "Phone C" password 1234

codec g729r8

type 7961

button 1:3

!

!

!

ephone 4

mac-address 001D.A21A.291D

ephone-template 1

username "Phone D" password 1234

codec g729r8

type 7961

button 1:4

!

!

!

!

!

ephone 55

mac-address 4BEF.7097.5562

ephone-template 1

type CIPC

button 1:55

!

!

!

!

line con 0

exec-timeout 0 0

privilege level 15

line aux 0

line 2

no activation-character

no exec

transport preferred none

transport output pad telnet rlogin lapb-ta mop udptn v120 ssh

stopbits 1

line 131

no activation-character

no exec

transport preferred none

transport input all

transport output pad telnet rlogin lapb-ta mop udptn v120 ssh

stopbits 1

line vty 0 4

session-timeout 30

exec-timeout 0 0

privilege level 15

password cisco

login local

transport preferred none

transport input all

line vty 5 130

login

transport input all

line vty 131

login

no activation-character

no exec

transport preferred none

transport input all

transport output pad telnet rlogin lapb-ta ssh

!

scheduler allocate 20000 1000

ntp master

!

end

# **Configuration Notes**

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 Flexible Reach side).
3. This command enables the CUCME to perform basic IP to IP voice communication when used with a CUBE.
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. Need to configure Refer-To modify as 10 digit understandable PSTN number transfer scenario
11. 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.
12. This SIP Profile allows CUCME to advertise desired and supported payload packet size.
13. 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.
14. Enables the CUCME BACD auto attendant feature.
15. Cisco Unity Express Interface.
16. These sccp commands configure the shared DSP resources as conference bridge (CFB) and transcoder device for the CUCME.
17. This command sets the SIP server target for outgoing SIP calls.
18. This command assigns the voice class codec setting to this dial-peer.
19. 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.
20. 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.
21. This commands enables Early Offer SIP invite method.
22. This commands assigns the applicable SIP profile to use for this dial-peer.
23. Example of configuring T38 as fax protocol per dial peer.

# **Cisco Unity Express**

ATTcme-UnityExpress# sh run

Generating configuration:

clock timezone UTC

hostname ATTcme-UnityExpress

line console

system language preferred "en\_US"

ntp server 10.10.10.5 prefer

ntp server 10.70.80.10

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

site name local

phone-authentication credentials hidden "GixGRq8cUmFBn9D4jn38/3rHJv3LzLILSd8ZZNgd+Y9J3xlk2B35j0nfGWTYHfmPSd8ZZNgd+Y9J3xlk2B35j0nfGWTYHfmPSd8ZZNgd+Y9J3xlk2B35j0nfGWTYHfmP"

site-hostname 10.70.87.11

web credentials hidden "GixGRq8cUmFBn9D4jn38/3rHJv3LzLILSd8ZZNgd+Y9J3xlk2B35j0nfGWTYHfmPSd8ZZNgd+Y9J3xlk2B35j0nfGWTYHfmPSd8ZZNgd+Y9J3xlk2B35j0nfGWTYHfmP"

end site

license agent max-sessions 9

privilege ViewPrivateList create

privilege ViewHistoricalReports create

privilege local-broadcast create

privilege ManagePublicList create

privilege vm-imap create

privilege manage-users create

privilege ViewRealTimeReports create

privilege broadcast create

privilege ManagePrompts create

privilege manage-passwords create

groupname Broadcasters create

username PhoneD create

username PhoneA create

username administrator create

username PhoneC create

username PhoneB create

username Bruno create

username cisco create

privilege ViewPrivateList description "Privilege to view private list"

privilege ViewHistoricalReports description "Privilege to view historical reports"

privilege local-broadcast description "Privilege to send local broadcast messages"

privilege ManagePublicList description "Privilege to manage public lists"

privilege vm-imap description "Privilege to manage personal voicemail via IMAP client"

privilege manage-users description "Privilege to create, modify, and delete users and groups"

privilege ViewRealTimeReports description "Privilege to view realtime reports"

privilege broadcast description "Privilege to send local or remote broadcast messages"

privilege ManagePrompts description "Privilege to create, modify, or delete system prompts"

privilege manage-passwords description "Privilege to reset user passwords"

privilege ViewPrivateList operation voicemail.lists.private.view

privilege ViewHistoricalReports operation report.historical.view

privilege local-broadcast operation broadcast.local

privilege local-broadcast operation system.debug

privilege ManagePublicList operation voicemail.lists.public

privilege ManagePublicList operation system.debug

privilege vm-imap operation voicemail.imap.user

privilege manage-users operation user.mailbox

privilege manage-users operation user.configuration

privilege manage-users operation user.pin

privilege manage-users operation system.debug

privilege manage-users operation group.configuration

privilege manage-users operation user.remote

privilege manage-users operation user.password

privilege manage-users operation user.notification

privilege ViewRealTimeReports operation report.realtime

privilege broadcast operation broadcast.local

privilege broadcast operation broadcast.remote

privilege broadcast operation system.debug

privilege ManagePrompts operation prompt.modify

privilege ManagePrompts operation system.debug

privilege manage-passwords operation user.pin

privilege manage-passwords operation system.debug

privilege manage-passwords operation user.password

groupname Administrators member administrator

groupname Administrators member cisco

groupname Broadcasters privilege broadcast

username PhoneD phonenumber "2712"

username PhoneA phonenumber "2709"

username PhoneC phonenumber "2711"

username PhoneB phonenumber "2710"

restriction msg-notification create

restriction msg-notification min-digits 1

restriction msg-notification max-digits 30

restriction msg-notification dial-string preference 1 pattern 12142425967 allowed

restriction msg-notification dial-string preference 2 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 5005 aa

description "5005"

enabled

maxsessions 2

script "aa.aef"

parameter "dialByExtnAnytime" "false"

parameter "busOpenPrompt" "AABusinessOpen.wav"

parameter "dialByExtnAnytimeInputLength" "4"

parameter "operExtn" ""

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 autoattendant aa

description "autoattendant"

enabled

maxsessions 10

script "aa.aef"

parameter "dialByExtnAnytime" "false"

parameter "busOpenPrompt" "AABusinessOpen.wav"

parameter "dialByExtnAnytimeInputLength" "4"

parameter "operExtn" "2304"

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 1

script "promptmgmt.aef"

parameter "appManagementScript" ""

end application

ccn application voicemail aa

description "Cisco Voicemail"

enabled

maxsessions 4

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 "ATTcme-UnityExpress"

end reporting

ccn subsystem sip

gateway address "10.70.87.10"

mwi sip unsolicited

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 1000

application "promptmgmt"

enabled

locale "en\_US"

maxsessions 1

end trigger

ccn trigger sip phonenumber 2300

application "autoattendant"

enabled

maxsessions 2

end trigger

ccn trigger sip phonenumber 2301

application "autoattendant"

enabled

locale "en\_US"

maxsessions 2

end trigger

ccn trigger sip phonenumber 5000

application "voicemail"

enabled

maxsessions 2

end trigger

ccn trigger sip phonenumber 5005

application "5005"

enabled

maxsessions 2

end trigger

service phone-authentication

end phone-authentication

service voiceview

enable

end voiceview

voicemail notification enable

voicemail notification preference all

voicemail notification allow-login

voicemail broadcast mwi

voicemail broadcast recording time 300

voicemail default messagesize 240

voicemail notification restriction msg-notification

voicemail operator telephone 2305

voicemail mailbox owner "PhoneA" size 300

description "PhoneA's Mailbox"

expiration time 10

messagesize 120

end mailbox

voicemail mailbox owner "PhoneB" size 300

description "PhoneB's Mailbox"

expiration time 10

messagesize 120

end mailbox

voicemail mailbox owner "PhoneC" size 300

description "PhoneC's Mailbox"

expiration time 10

messagesize 120

end mailbox

voicemail mailbox owner "PhoneD" size 300

description "PhoneD's Mailbox"

expiration time 10

messagesize 120

end mailbox

end

# **Acronyms**

|  |  |
| --- | --- |
| 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) |
| AVPN | AT&T Virtual Private Network |
| MIS | Managed Internet Services |
| PNT | Private Network Transport |

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