Fusion Connect: Connecting Cisco Unified Communications Manager Express (CME) 11.0 using SIP

August 2016
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Introduction

Service Providers today, such as Fusion Connect, 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.

- This application note describes how to configure a Cisco Unified Communications Manager Express (CUCME) 11.0, Cisco Unity Connection (CUC) 11.0.1 with connectivity to Fusion Connect SIP trunk service. The application note also covers support and configuration examples for Cisco Unity Connection (CUC) messaging integrated into the Cisco Unified Communications Manager Express. The deployment model covered in this application note is Customer Premises Equipment (Cisco Unified CME/CUC) to PSTN (Fusion Connect). Fusion Connect provides inbound and outbound call service.

- Testing was performed in accordance to Fusion connect 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 Connection), Cisco auto-attendant (BACD), fax using T.38 and G.711 (G3 and SG3 speeds).

- 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 CME.
Network Topology

Hardware Components
- This solution was tested with Cisco ISR4431
- Cisco IP Phones - This solution was tested with 7961 and 7975 and 8961 phones, but any Cisco IP Phone model supporting RFC2833 can be used
- Cisco ISR4431/K9 (1RU) with 1653212K/6147K bytes of memory - Processor board ID FOC18261KJL
- 4 Gigabit Ethernet interfaces and 2 Voice FXS interfaces
- Cisco Unity Connection - VMware - 1 vCPU: Intel(R) Xeon(R) CPU X5675 @ 3.07GHz
- HDD 160 GB, Memory 4096 Mbytes RAM

Software Requirements
- Cisco IOS gateway running Cisco Unified CME 11.0 Version 15.6(1)S, RELEASE SOFTWARE (fc4). This solution was tested with Cisco IOS image: "flash:/isr4400-universalk9.03.17.00.S.156-1.S-std.SPA.bin"
- This solution was tested with Cisco Unity Connection version (Version 11.0.1.10000-10)
Features

Features - Supported

- Basic Call using G.729 & G711
- Calling Party Number Presentation and Restriction
- Calling Name
- Call Transfer
- Conference
- Call Hold and Resume (See Caveat section for details)
- Call Forward All, Busy and No Answer
- Fax using T.38 (See Caveat section for details)
- Fax over G.711 (See Caveat section for details)
- Incoming DID Translation and Routing
- Outbound calls and Inbound calls
- Voicemail
- Auto-attendant (See Caveat section for details)

Features Not Supported

- Transfer - Blind
- Dual codecs (G711 & G729) are not supported by Fusion Connect at the moment
- 0 and 0+10 digit dial plan - operator assisted calls are not supported by Fusion Connect at the moment
- Rel100 feature is not supported by Fusion Connect
- Privacy ID feature is not supported by Fusion Connect

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 Cisco Unified CME Basic Automatic Call Distribution (BACD) was employed to enable the auto-attendant feature. The test was performed using the default codec G711u for auto attendant prompts. G.729 prompts can be used; however it was not tested here.

Hold & Resume:
- Cisco Unified 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.
Configuration Considerations

- 10 Digit BTN (Billing Telephone Number) is used to register Cisco Unified CME to Fusion connect SIP trunk (See configuration section for details)

- Fusion Connect 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 4-digit extensions configured on Cisco Unified CME 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 Cisco Unified CME (See configuration section for details)

- Fusion requires 10 digit BTN number in From and Contact header to process the basic calls successfully (See configuration section for details)
Configuration

Cisco IOS Version

Fusion# show version
Cisco IOS XE Software, Version 03.17.00.S - Standard Support Release
Cisco IOS Software, ISR Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 15.6(1)S, RELEASE SOFTWARE (fc4)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2015 by Cisco Systems, Inc.
Compiled Wed 25-Nov-15 14:33 by mcpree

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ROM: IOS-XE ROMMON

Fusion uptime is 1 week, 5 days, 4 hours, 24 minutes
Uptime for this control processor is 1 week, 5 days, 4 hours, 25 minutes
System returned to ROM by reload at 10:25:10 UTC Fri Jul 8 2016
System restarted at 10:29:57 UTC Fri Jul 8 2016
System image file is "bootflash:/isr4400-universalk9.03.17.00.S.156-1.S-std.SPA.bin"
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:

If you require further assistance please contact us by sending email to export@cisco.com.
Suite License Information for Module: 'esg'

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<thead>
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<th>Suite</th>
<th>Suite Current</th>
<th>Type</th>
<th>Suite Next reboot</th>
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Technology Package License Information:

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<tr>
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<td>ipbasek9</td>
</tr>
</tbody>
</table>

cisco ISR4431/K9 (1RU) processor with 1653212K/6147K bytes of memory
Processor board ID FTX1845AJ9S
4 Gigabit Ethernet interfaces
2 Voice FXS interfaces
32768K bytes of non-volatile configuration memory
4194304K bytes of physical memory
7057407K bytes of flash memory at bootflash:

Configuration register is 0x2102
Cisco Unified CME

Fusion#show running-config

!
version 15.6
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core

!
hostname Fusion

!
boot-start-marker
boot system flash isr4400-universalk9.03.17.01.S.156-1.156-1.S1-std.SPA.bin
boot-end-marker

!
vrf definition Mgmt-intf

!
address-family ipv4
exit-address-family

!
address-family ipv6
exit-address-family

!
no aaa new-model

!
ip name-server 10.85.0.232
ip domain name lab.tekvizion.com

!
subscriber templating
multilink bundle-name authenticated

voice service voip
ip address trusted list
  ipv4 216.86.XX.XX
  ipv4 10.71.1.135
  ipv4 10.70.40.2
no ip address trusted authenticate
callmonitor
address-hiding ¹
allow-connections sip to sip²
no supplementary-service sip moved-temporarily
no supplementary-service sip refer
redirect ip2ip
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none ³
sip
  session refresh⁴
  registrar server

voice class codec ¹⁵
codec preference 1 g729r8
codec preference 2 g711ulaw

¹ Enables IP address hiding between the private network (Cisco Unified CME side) and the public network (Fusion Connect Network)
² This command enables the Cisco Unified CME to perform basic IP to IP voice communication
³ 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
⁴ Use the SIP session refresh command to send the session refresh request
⁵ This command enables multiple codec support and performs codec filtering required for correct interoperability between Fusion Connect SIP network and Cisco Unified CME
voice class sip-profiles 1
request INVITE sip-header From modify "<sip:(.*)@(.*)" "<sip:9733397580@\2"
request INVITE sip-header Contact modify "<sip:(.*)@(.*)" "<sip:9733397580@\2"
request INVITE sip-header Diversion modify "<sip:(.*)@(.*)" "<sip:973339\1@\2"
request INVITE sip-header Privacy add "Privacy: id"

voice register global
mode cme
source-address 10.70.40.2 port 5060
max-dn 10
max-pool 10
load 7945 SIP45.9-4-2SR1-1S
load 7961 SIP41.9-2-1S
load 8961 sip8961.9-2-2SR1-9
authenticate register
authenticate realm cisco
timezone 8
tftp-path flash:
file text
create profile sync 0003244205203966
voice register dn 1
number 7580
call-forward b2bua noan 7777 timeout 9

6 SIP Profiles can be used to manipulate SIP header attributes
7 FROM header should contain 10 digit BTN number for successful registration
8 Contact header should contain 10 digit BTN number for successful registration
9 SIP profile is used to convert 4 digit extension number to 10 digit DID for call forward
10 "request INVITE sip-header Privacy add "privacy id" is added 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
name CPE-1
!
voice register dn  2
number 3067
name CPE-4
!
voice register pool  1
registration-timer max 600 min 60
busy-trigger-per-button 1
id mac 001D.A21A.2791
type 7961
number 1 dn 1
dtmf-relay rtp-nce
voice-class codec 1
username 7580 password
!
voice register pool  2
id mac 081F.F362.5DD1
type 8961
number 1 dn 2
voice-class codec 1
username 3067 password
!
!
voice translation-rule 1
rule 1 /97339\(\(\(\(\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)/\)
rule 2  /\('3067\)\)/ /408470\1/  
rule 3  /\('7...\)\)/ /973339\1/  

!  
!  
voice translation-profile test  
!  
voice translation-profile to-10dig  
translate calling 2  
!  
voice translation-profile to-ext  
translate called 1  
!  
voice-card 0/1  
no watchdog  
!  
license udi pid ISR4431/K9 sn FOC18261KJL  
license accept end user agreement  
license boot level appxk9  
license boot level uck9  
license boot level securityk9  
!  
spanning-tree extend system-id  
!  
username cisco privilege 15 secret 5  
!  
redundancy  
mode none  
!  
vlan internal allocation policy ascending
interface GigabitEthernet0/0/0
  ip address 10.64.4.19 255.255.0.0
  shutdown
  media-type rj45
  negotiation auto

interface GigabitEthernet0/0/1
  no ip address
  shutdown
  media-type rj45
  negotiation auto

interface GigabitEthernet0/0/2
  description WAN
  ip address 192.65.XX.XXX 255.255.255.128
  media-type rj45
  negotiation auto

interface GigabitEthernet0/0/3
  description LAN
  ip address 10.70.40.2 255.255.255.0
  media-type rj45
  negotiation auto

interface Service-Engine0/1/0

interface GigabitEthernet0
  vrf forwarding Mgmt-intf
no ip address
negotiation auto
!
interface Vlan1
no ip address
shutdown
!
ip forward-protocol nd
no ip http server
no ip http secure-server
ip tftp source-interface GigabitEthernet0/0/3
ip route 0.0.0.0 0.0.0.0 192.65.XX.XX
ip route 10.64.0.0 255.255.0.0 10.70.40.1
ip route 10.70.40.0 255.255.0.0 10.70.40.1
ip route 10.85.0.232 255.255.255.255 GigabitEthernet0/0/3
ip route 172.16.24.0 255.255.255.255 GigabitEthernet0/0/3
!
tftp-server flash:RedHotChiliPeppers.wav
tftp-server flash:apps45.9-4-2ES9.sbn
tftp-server flash:cnu45.9-4-2ES9.sbn
tftp-server flash:cvm45sip.9-4-2ES9.sbn
tftp-server flash:dsp45.9-4-2ES9.sbn
tftp-server flash:jar45sip.9-4-2ES9.sbn
tftp-server flash:SIP45.9-4-2SR1-1S.loads
tftp-server flash:term45.default.loads
tftp-server flash:term65.default.loads
tftp-server flash:SampleAudioSource.g729.wav
tftp-server flash:SampleAudioSource.wav
tftp-server flash:ToneOnHold.ulaw.wav
tftp-server flash:apps41.9-2-1TH1-13.sbn

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

tftp-server flash:cvm41sip.9-2-1TH1-13.sbn

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

tftp-server flash:jar41sip.9-2-1TH1-13.sbn

tftp-server flash:SIP41.9-2-1S.loads

tftp-server flash:term61.default.loads

tftp-server flash:dkern8961.100609R2-9-2-2SR1-9.sebn

tftp-server flash:kern8961.9-2-2SR1-9.sebn

tftp-server flash:rootfs8961.9-2-2SR1-9.sebn

tftp-server flash:sboot8961.031610R1-9-2-2SR1-9.sebn

tftp-server flash:sip8961.9-2-2SR1-9.loads

tftp-server flash:skern8961.022809R2-9-2-2SR1-9.sebn

!

control-plane

!

voice-port 0/1/0

cptone IN

station-id number 973339XXXX

caller-id enable

!

voice-port 0/1/1

!

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
telephony-service
cconference transfer-pattern
no auto-reg-episode
max-episodes 20
max-dn 20
ip source-address 10.70.40.2 port 2000
voicemail 7777
max-conferences 8 gain -6
call-forward pattern .T
moh enable-g711 "flash:ToneOnHold.ulaw.wav"
multicast moh 239.0.2.1 port 2000
transfer-system full-consult
transfer-pattern .T
create cnf-files version-stamp 7960 Jul 21 2016 06:58:11
!

dial-peer voice 100 voip
description inbound from Fusion
translation-profile incoming to-ext
session protocol sipv2
session target sip-server
session transport udp
incoming called-number 973339....
voice-class codec 1
dtmf-relay rtp-nte
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none

11 7777 - Voicemail pilot number used in this example
! 
dial-peer voice 200 voip
description outbound to Fusion
translation-profile outgoing to-10dig
destination-pattern [2-9]........
session protocol sipv2
session target sip-server\[12\]
session transport udp
voice-class codec 1\[13\]
o no voice-class sip asserted-id
voice-class sip profiles 1\[14\]
dtmf-relay rtp-nte
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none\[15\]
no vad
!
dial-peer voice 101 voip
description inbound from Fusion
translation-profile incoming to-ext
shutdown
session protocol sipv2
session target sip-server
session transport udp
incoming called-number 408470....
voice-class codec 1

\[12\] This command sets the SIP server target for outgoing SIP calls
\[13\] This command assigns the voice class codec setting to this dial-peer
\[14\] This command assigns the applicable SIP profile to use for this dial-peer
\[15\] Example of configuring T.38 as fax protocol per dial peer
dtmf-relay rtp-n-te
no vad
!
dial-peer voice 999 voip
  description "Emergency and Operator call"
  destination-pattern 91.
  session protocol sipv2
  session target sip-server
  session transport udp
  voice-class codec 1
  no voice-class sip asserted-id
  voice-class sip profiles 1
  dtmf-relay rtp-n-te
  no vad
!
dial-peer voice 7777 voip
  description voice mail pilot
  huntstop
  max-conn 8
  destination-pattern 7777
  session protocol sipv2
  session target ipv4:10.71.1.135:5060
  session transport udp
  voice-class sip bind control source-interface GigabitEthernet0/0/3
  voice-class sip bind media source-interface GigabitEthernet0/0/3
  dtmf-relay rtp-n-te
  codec g711ulaw

---

16 Dial peer used to route the calls to Cisco unity connection server for voicemail access
17 This command sets the Cisco unity connection server target for voicemail calls
no vad
!
dial-peer voice 1000 pots
  preference 2
  service session
  destination-pattern 7583
  no digit-strip
  port 0/1/0
!
gateway
  timer receive-rtp 1200
!
sip-ua
  credentials number 973339XXX username 973339XXXX password 7 realm asterisk
  authentication username 9733397580 password 7 realm asterisk
  mwi-server ipv4:10.71.1.135 expires 3600 port 5060 transport udp unsolicited
  registrar ipv4:216.86.41.69 expires 3600
  sip-server ipv4:216.86.41.69:5060
!
ephone-template 1
  softkeys hold  Resume Newcall Select Join
  softkeys idle  Redial Newcall ConfList RmLstC Cfwdall Join Pickup Login HLog Dnd Gpickup
  softkeys seized  Endcall Redial Cfwdall Meetme Pickup Callback
  softkeys alerting  Endcall Callback
  softkeys connected  Hold Endcall Confrn Trnsfer Select Join ConfList RmLstC Park Flash
!

ephone-dn 1 dual-line
  number 7593
label 7593
name CPE-3

ephone-dn 2 dual-line
number 7583
label 7583
name CPE-2

ephone 1
conference max-length 16
device-security-mode none
mac-address 0008.3031.F49B
ephone-template 1
max-calls-per-button 2
type 7975
button 1:1

ephone 2
device-security-mode none
mac-address 0C27.2431.5F6B
ephone-template 1
max-calls-per-button 2
type 7965
button 1:2

ephone 3
device-security-mode none


line con 0
exec-timeout 0 0
logging synchronous
stopbits 1
line aux 0
privilege level 15
stopbits 1
line vty 0
exec-timeout 0 0
no activation-character
logging synchronous
login local
transport preferred ssh
transport input telnet ssh
line vty 1 4
exec-timeout 0 0
logging synchronous
login local
transport input telnet ssh

ntp server 10.10.10.5

end
Cisco Unity Connection

Version Details

Cisco Unity Connection User Configuration

**Navigation:** Cisco Unity Connection → Users → Users

1. Set **Alias** = 7583 is used for this example
2. Set **First Name** = cisco is used to identify this User
3. Set **Last Name** = cisco is used for this example
4. Set **Display Name** = 7583 is used in this example
5. Set **SMTP Address** = 7583 is used in this example
6. Set **Extension** = 7583 is used in this example
7. Set **Phone System** = Fusion is used in this example
8. All other values are default
Cisco Unity Connection User Configuration (Continued)

9. All values are default

<table>
<thead>
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<th>Class of Service</th>
<th>Voice Mail User COS</th>
</tr>
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<tbody>
<tr>
<td>Active Schedule</td>
<td>All Hours</td>
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<td>View</td>
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<tr>
<td>Set for Self-enrollment at Next Sign-In</td>
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</tr>
<tr>
<td>List in Directory</td>
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<tr>
<td>Send Non-Delivery Receipts on Failed Message Delivery</td>
<td></td>
</tr>
<tr>
<td>Skip PIN When Calling From a Known Extension</td>
<td></td>
</tr>
<tr>
<td>Use Short Calendar Caching Poll Interval</td>
<td></td>
</tr>
<tr>
<td>Recorded Name</td>
<td>Play/Record</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Location</th>
</tr>
</thead>
<tbody>
<tr>
<td>Address</td>
</tr>
<tr>
<td>Building</td>
</tr>
<tr>
<td>City</td>
</tr>
<tr>
<td>State</td>
</tr>
<tr>
<td>Postal Code</td>
</tr>
<tr>
<td>Country</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Time Zone</th>
<th>(GMT-06:00) America/Chicago</th>
</tr>
</thead>
</table>

<table>
<thead>
<tr>
<th>Language</th>
</tr>
</thead>
<tbody>
<tr>
<td>Use System Default Language</td>
</tr>
<tr>
<td>English (United States)</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Department</th>
</tr>
</thead>
<tbody>
<tr>
<td>Manager</td>
</tr>
</tbody>
</table>
1. Set **Phone System Name = Fusion** is used for this example
Port Group

**Navigation:** Telephony Integrations → Port Group

1. Set **Display Name** = *Fusion-1* is used for this example
2. Check **Register with SIP Server**

<table>
<thead>
<tr>
<th>Port Group</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Display Name</strong></td>
<td>Fusion-1</td>
</tr>
<tr>
<td><strong>Integration Method</strong></td>
<td>SIP</td>
</tr>
<tr>
<td><strong>Reset Status</strong></td>
<td>Reset Not Required</td>
</tr>
</tbody>
</table>

### Session Initiation Protocol (SIP) Settings

- **Register with SIP Server**
- **Authenticate with SIP Server**
- **Authentication Username**
- **Authentication Password**
- **Contact Line Name**
- **SIP Security Profile**
- **SIP Transport Protocol**

### Advertised Codec Settings

<table>
<thead>
<tr>
<th>Display Name</th>
<th>Packet Size</th>
</tr>
</thead>
<tbody>
<tr>
<td>G.711 mu-law</td>
<td>20</td>
</tr>
<tr>
<td>G.729</td>
<td>20</td>
</tr>
</tbody>
</table>

### Message Waiting Indicator Settings

- **Enable Message Waiting Indicators**
- **Delay between Requests**
- **Maximum Concurrent Requests**
- **Retries After Successful Attempt**
- **Retry Interval After Successful Attempt**

---

Note: Testing was conducted in tekVizion labs
Navigation: Telephony Integrations → Port Group → Edit → Servers

Edit Servers

SIP Servers
- Delete Selected
- Add

<table>
<thead>
<tr>
<th>Order</th>
<th>IPv4 Address or Host Name</th>
<th>IPv6 Address or Host Name</th>
<th>Port</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>10.70.40.2</td>
<td></td>
<td>5060</td>
</tr>
</tbody>
</table>

Delete Selected
Add

TFTP Servers
- Delete Selected
- Add

Delete Selected
Add

IPv6 Addressing Mode
- Preference for Signaling: IPv4
- Preference for Media: IPv4
Port
1. Set Port Name = Fusion-1-001 is used for this example
2. Phone System = Fusion
3. Port Group = Fusion-1
4. Server = clus13-cuc is used for this example
Acronyms

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP</td>
<td>Session Initiation Protocol</td>
</tr>
<tr>
<td>MGCP</td>
<td>Media Gateway Control Protocol</td>
</tr>
<tr>
<td>SCCP</td>
<td>Skinny Client Control Protocol</td>
</tr>
<tr>
<td>Cisco Unified CME</td>
<td>Cisco Unified Communications Manager Express</td>
</tr>
<tr>
<td>SP</td>
<td>Service Provider</td>
</tr>
<tr>
<td>PSTN</td>
<td>Public switched telephone network</td>
</tr>
<tr>
<td>DTMF</td>
<td>Dual Tone Multi-Frequency</td>
</tr>
<tr>
<td>CUC</td>
<td>Cisco Unity Connection</td>
</tr>
<tr>
<td>VOIP</td>
<td>Voice Over Internet Protocol</td>
</tr>
<tr>
<td>IP</td>
<td>Internet Protocol</td>
</tr>
<tr>
<td>TDM</td>
<td>Time-division multiplexing</td>
</tr>
<tr>
<td>BTN</td>
<td>Billed (or Billing) Telephone Number</td>
</tr>
<tr>
<td>CODEC</td>
<td>Coder-Decoder (in this document a device used to digitize and undigitize voice signals)</td>
</tr>
</tbody>
</table>

Important Information

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