



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

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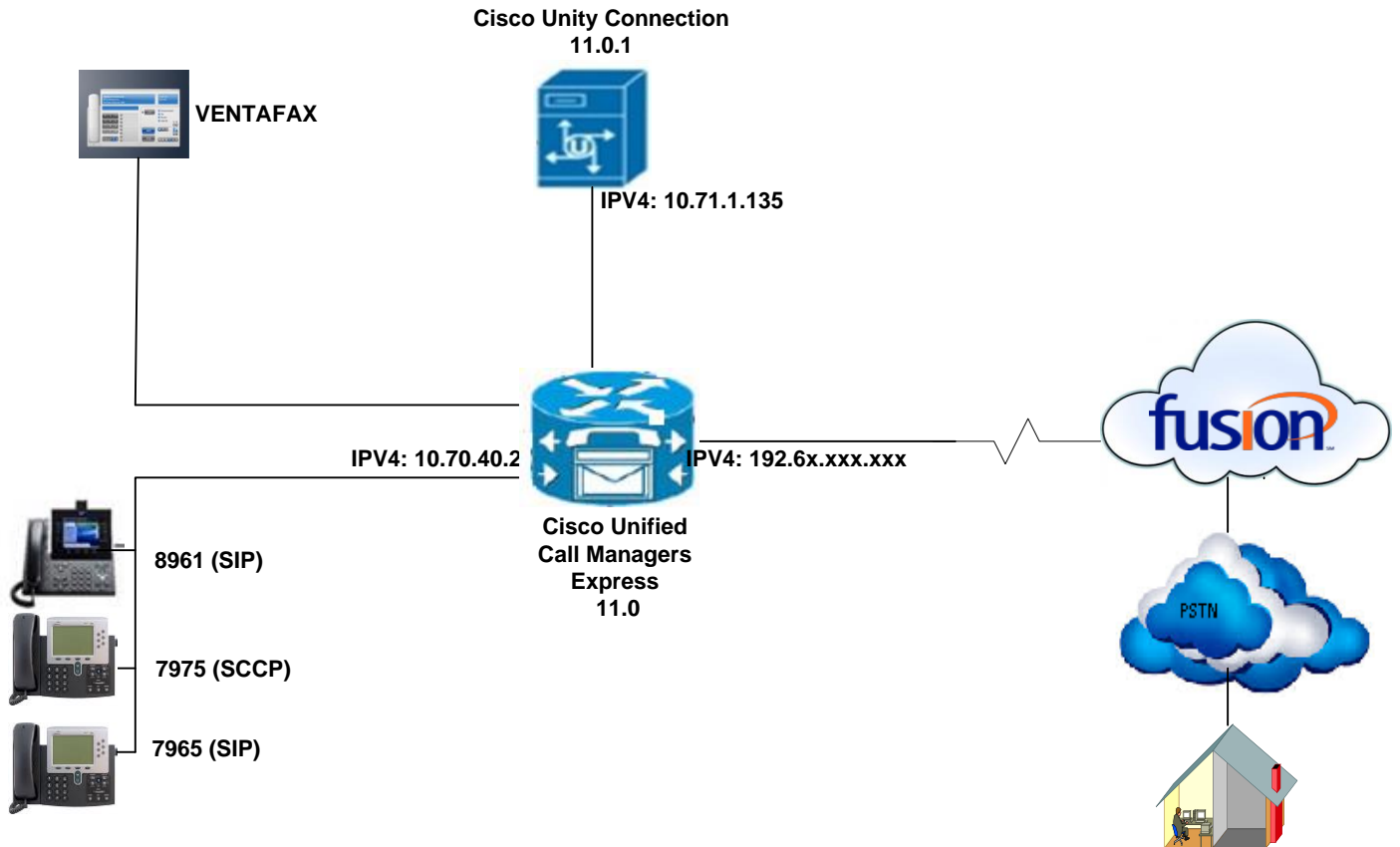
Introduction

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 mcpre
```

Cisco IOS-XE software, Copyright (c) 2005-2015 by cisco Systems, Inc. All rights reserved. Certain components of Cisco IOS-XE software are licensed under the GNU General Public License ("GPL") Version 2.0. The software code licensed under GPL Version 2.0 is free software that comes with ABSOLUTELY NO WARRANTY. You can redistribute and/or modify such GPL code under the terms of GPL Version 2.0. For more details, see the documentation or "License Notice" file accompanying the IOS-XE software, or the applicable URL provided on the flyer accompanying the IOS-XE software.

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:
<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.



Suite License Information for Module:'esg'

Suite	Suite Current	Type	Suite Next reboot
FoundationSuiteK9 securityk9 appxk9	None	None	None
AdvUCSuiteK9 uck9 cme-srst cube	None	None	None

Technology Package License Information:

Technology	Technology-package Technology-package Current	Type	Next reboot
appxk9	appxk9	RightToUse	appxk9
uck9	uck9	RightToUse	uck9
securityk9	securityk9	RightToUse	securityk9
ipbase	ipbasek9	Permanent	ipbasek9

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.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 1⁶

request INVITE sip-header From modify "< sip:(.*)@(.*)" "< sip:9733397580@12"⁷

request INVITE sip-header Contact modify "< sip:(.*)@(.*)" "< sip:9733397580@12"⁸

request INVITE sip-header Diversion modify "< sip:(.*)@(.*)>" "< sip:973339\1 @12>"⁹

request INVITE sip-header Privacy add "Privacy: id"¹⁰

!

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-nte
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 /973339\(...\)/ ^1/
rule 2 /408470\(...\)/ ^1/
!
voice translation-rule 2
```



```
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.XXX
ip route 10.64.0.0 255.255.0.0 10.70.40.1
ip route 10.70.40.0 255.255.255.0 10.70.40.1
ip route 10.85.0.232 255.255.255.255 GigabitEthernet0/0/3
ip route 172.16.24.0 255.255.248.0 10.70.40.1
!
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
conference transfer-pattern
no auto-reg-ephone
max-ephones 20
max-dn 20
ip source-address 10.70.40.2 port 2000
voicemail 777711
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 outboud to Fusion
translation-profile outgoing to-10dig
destination-pattern [2-9].....
session protocol sipv2
session target sip-server12
session transport udp
voice-class codec 113
no voice-class sip asserted-id
voice-class sip profiles 114
dtmf-relay rtp-nte
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none15
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-nte
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-nte
no vad
!
dial-peer voice 7777 voip16
description voice mail pilot
huntstop
max-conn 8
destination-pattern 7777
session protocol sipv2
session target ipv4:10.71.1.135:506017
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-nte
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-rtcp 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 Cfdall Join Pickup Login HLog Dnd Gpickup

softkeys seized Endcall Redial Cfdall Meetme Pickup Callback

softkeys alerting Endcall Callback

softkeys connected Hold Endcall Confm 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

Alias*	7583
First Name	cisco
Last Name	cisco
Display Name	7583
SMTP Address	7583 @clus13-cuc.lab.tekvizion.com
Initials	
Title	
Employee ID	
LDAP Integration Status	
<input type="radio"/> Integrate with LDAP Directory	
<input checked="" type="radio"/> Do Not Integrate with LDAP Directory	
Phone	
Extension*	7583
Cross-Server Transfer Extension or URI	
Outgoing Fax Number	
Outgoing Fax Server	--- Not Selected ---
Partition	clus13-cuc Partition
Search Scope	clus13-cuc Search Space
Phone System	Fusion



Cisco Unity Connection User Configuration (Continued)

9. All values are default

Class of Service	Voice Mail User COS	▼
Active Schedule	All Hours	▼ <input type="button" value="View"/>
<input type="checkbox"/> Set for Self-enrollment at Next Sign-In		
<input checked="" type="checkbox"/> List in Directory		
<input checked="" type="checkbox"/> Send Non-Delivery Receipts on Failed Message Delivery		
<input type="checkbox"/> Skip PIN When Calling From a Known Extension		
<input type="checkbox"/> Use Short Calendar Caching Poll Interval		
Recorded Name	<input type="button" value="Play/Record"/>	
Location		
Address	<input type="text"/>	
Building	<input type="text"/>	
City	<input type="text"/>	
State	<input type="text"/>	
Postal Code	<input type="text"/>	
Country	United States ▼	
<input checked="" type="checkbox"/> Use System Default Time Zone		
Time Zone	(GMT-06:00) America/Chicago ▼	
Language	<input type="radio"/> Use System Default Language	
	<input checked="" type="radio"/> English(United States) ▼	
Department	<input type="text"/>	
Manager	<input type="text"/>	



Cisco Unity Connection Telephony Integration

Navigation: Telephony Integrations → Phone System

1. Set **Phone System Name** = *Fusion* is used for this example

Phone System Basics (Fusion) Search Phone Systems ▶ Phone System
Related Links Add Port Group

Phone System Edit Refresh Help

Save Delete Previous Next

Phone System

Phone System Name* Fusion

Default TRAP Phone System

Message Waiting Indicators

Send Message Counts
 Use Same Port for Enabling and Disabling MWIs
 Force All MWIs Off for this Phone System

Run Synchronize All MWIs on This Phone System

Call Loop Detection by Using DTMF

Enable for Supervised Transfers
 Enable for Forwarded Message Notification Calls (by Using DTMF)

DTMF Tone To Use A ▾
Guard Time 2500 milliseconds

Call Loop Detection by Using Extension

Enable for Forwarded Message Notification Calls (by Using Extension)

Call Loop Detection by Using DTMF

Enable for Supervised Transfers
 Enable for Forwarded Message Notification Calls (by Using DTMF)

DTMF Tone To Use A ▾
Guard Time 2500 milliseconds

Call Loop Detection by Using Extension

Enable for Forwarded Message Notification Calls (by Using Extension)

Phone View Settings

Enable Phone View

CTI Phone
Access Username
CTI Phone
Access Password

Outgoing Call Restrictions

Enable outgoing calls
 Disable all outgoing calls immediately
 Disable all outgoing calls between

Beginning Time: 12 ▾ 00 ▾ AM ▾
Ending Time: 12 ▾ 00 ▾ AM ▾



Port Group

Navigation: Telephony Integrations → Port Group

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

Port Group

Display Name* Fusion-1

Integration Method SIP

Reset Status Reset Not Required

Session Initiation Protocol (SIP) Settings

Register with SIP Server

Authenticate with SIP Server

Authentication Username

Authentication Password

Contact Line Name

SIP Security Profile 5060 ▼

SIP Transport Protocol UDP ▼

Advertised Codec Settings

Display Name	Packet Size
G.711 mu-law	20 ▼
G.729	20 ▼

Message Waiting Indicator Settings

Enable Message Waiting Indicators

Delay between Requests milliseconds

Maximum Concurrent Requests

Retries After Successful Attempt

Retry Interval After Successful Attempt milliseconds



Navigation: Telephony Integrations → Port Group → Edit → Servers

Search Port Groups ▶ Port Group Basics (Fusion-1) ▶ Edit Servers

Edit Servers Related Links [Check Telephony Configuration](#) Go

Port Group Edit Refresh Help

SIP Servers

<input type="checkbox"/>	Order	IPv4 Address or Host Name	IPv6 Address or Host Name	Port
<input type="checkbox"/>	0	10.70.40.2		5060

TFTP Servers

<input type="checkbox"/>	Order	IPv4 Address or Host Name	IPv6 Address or Host Name
--------------------------	-------	---------------------------	---------------------------

IPv6 Addressing Mode

Preference for Signaling

Preference for Media



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

Phone System Port

Enabled

Port Name

Phone System

Port Group

Server ▼

Port Behavior

Answer Calls

Perform Message Notification

Send MWI Requests (may also be disabled by the port group)

Allow TRAP Connections



Acronyms

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

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