Application Note

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

August 2016

Note: Testing was conducted in tekVizion labs

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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.

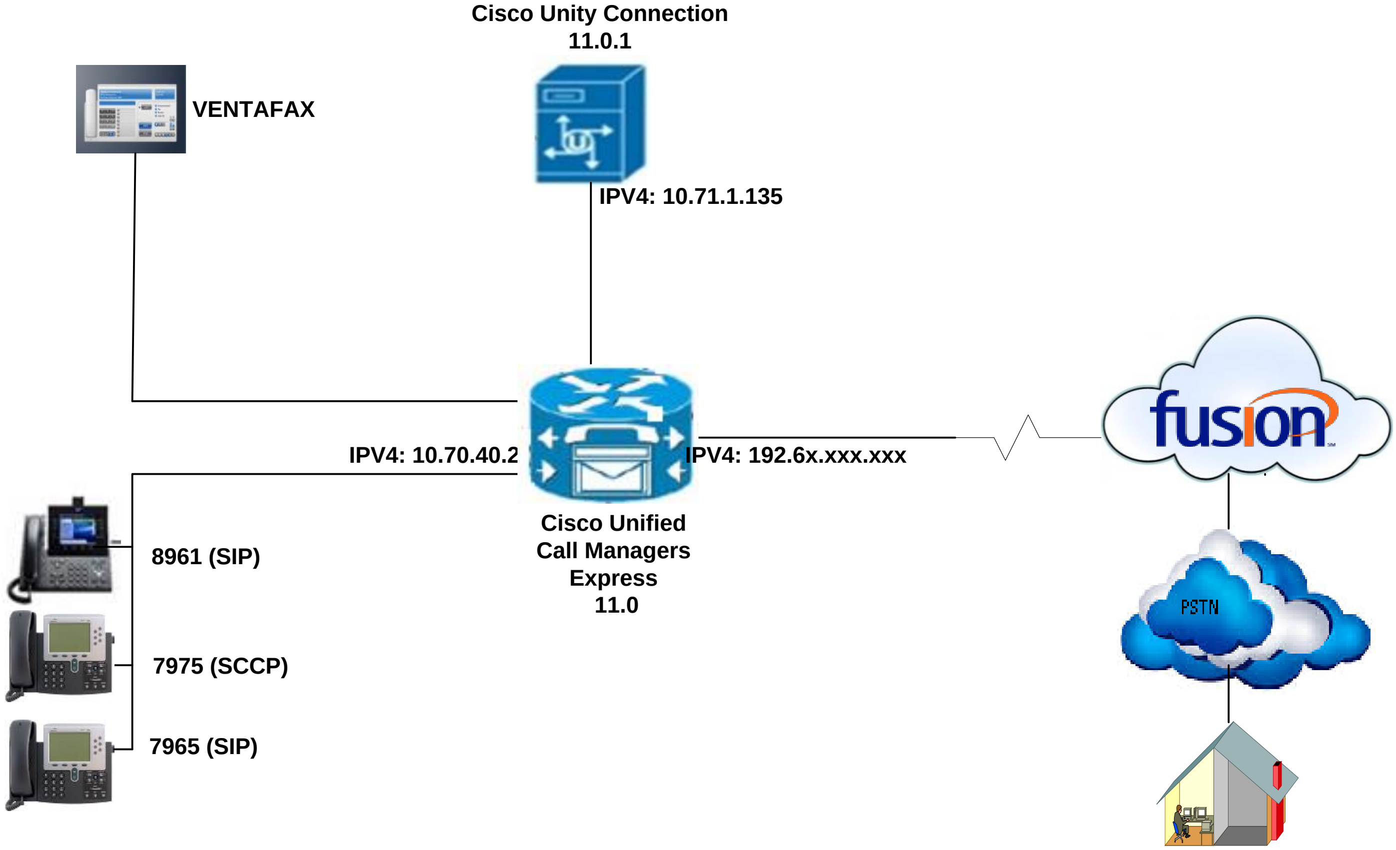
- This application note describes how to configure a Cisco Unified Communications Manager Express (CUCME) 11.0, Cisco Unity Connection (CUC) 11.0.1 with connectivity to Fusion Connect SIP trunk service. The application note also covers support and configuration examples for Cisco Unity Connection (CUC) 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.

Note: Testing was conducted in tekVizion labs

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)

Note: Testing was conducted in tekVizion labs

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.

Note: Testing was conducted in tekVizion labs

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

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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.

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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     None                  None           None
securityk9
appxk9

AdvUCSuiteK9          None                  None           None
uck9
cme-srst
cube
```

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

```
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 sip[2]
 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 refresh[4]
  registrar server
!
voice class codec 1[5]
 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[6]
 request INVITE sip-header From modify "<sip:(.*)@(.*)" "<sip:9733397580@\2"[7]
 request INVITE sip-header Contact modify "<sip:(.*)@(.*)" "<sip:9733397580@\2"[8]
 request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:973339\1@\2>"[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 7777[11]
 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-server[12]
 session transport udp
 voice-class codec 1[13]
 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-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 voip[16]
 description voice mail pilot
 huntstop
 max-conn 8
 destination-pattern 7777
 session protocol sipv2
 session target ipv4:10.71.1.135:5060[17]
 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-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
!
!
```


Note: Testing was conducted in tekVizion labs

```
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
```

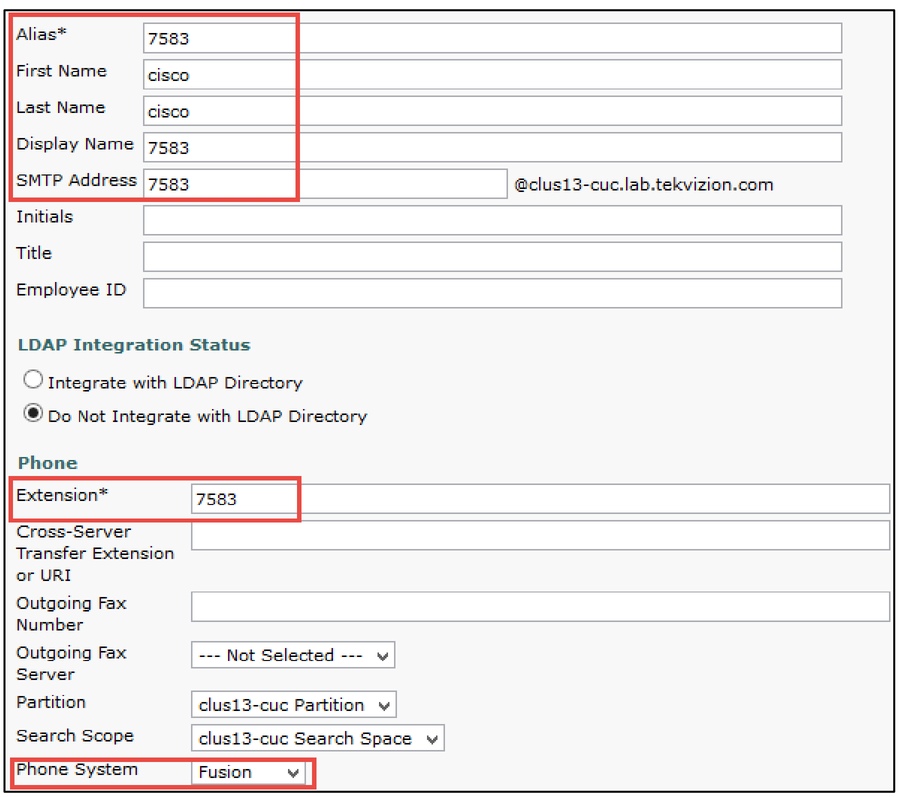
Cisco Unity Connection

Version Details

Cisco Unity Connection User Configuration

Navigation: Cisco Unity Connection → Users → Users

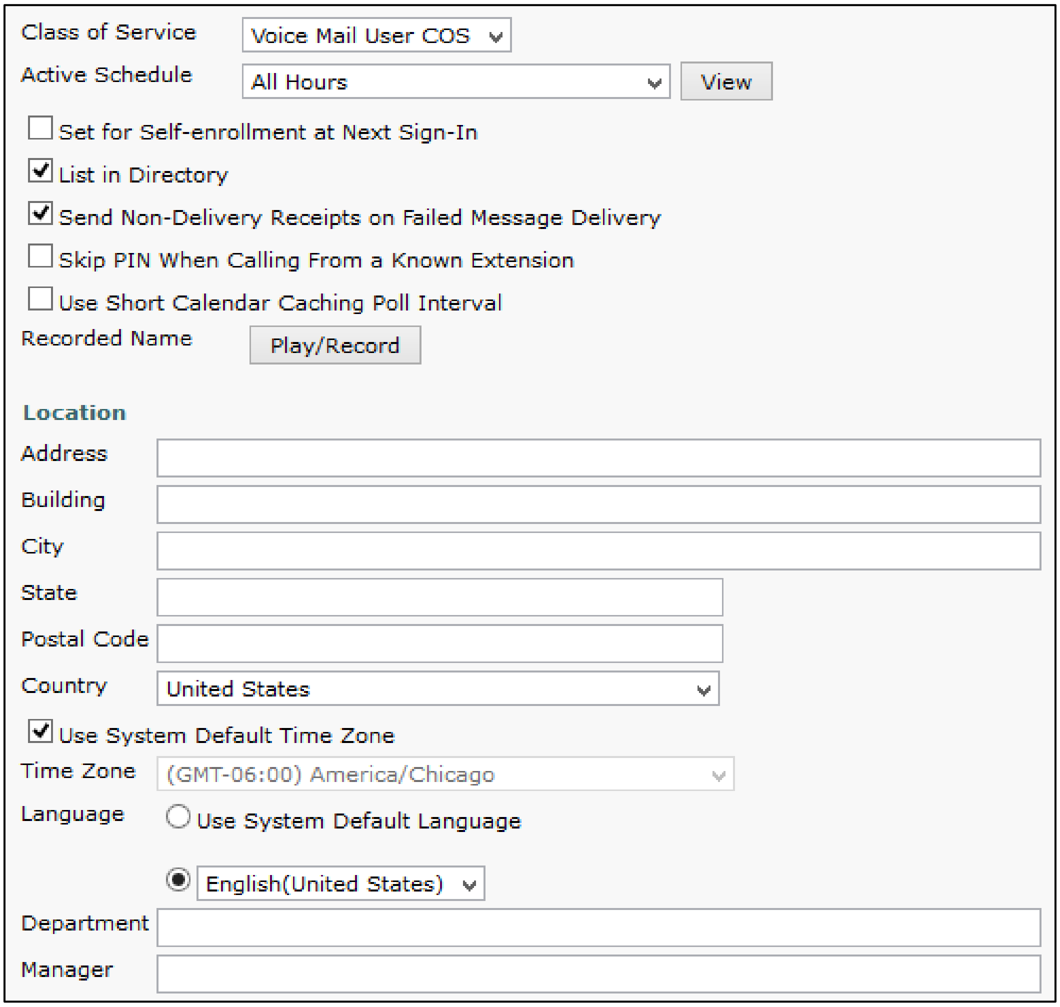
1. Set **Alias**= *7583* is used for this example
2. Set **First Name** = *cisco* is used to identify this User
3. Set **Last Name** = *cisco* is used for this example
4. Set **Display Name** = *7583* is used in this example
5. Set **SMTP Address** =*7583* is used in this example
6. Set **Extension**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



Note: Testing was conducted in tekVizion labs

Cisco Unity Connection User Configuration (Continued)

9. All values are default

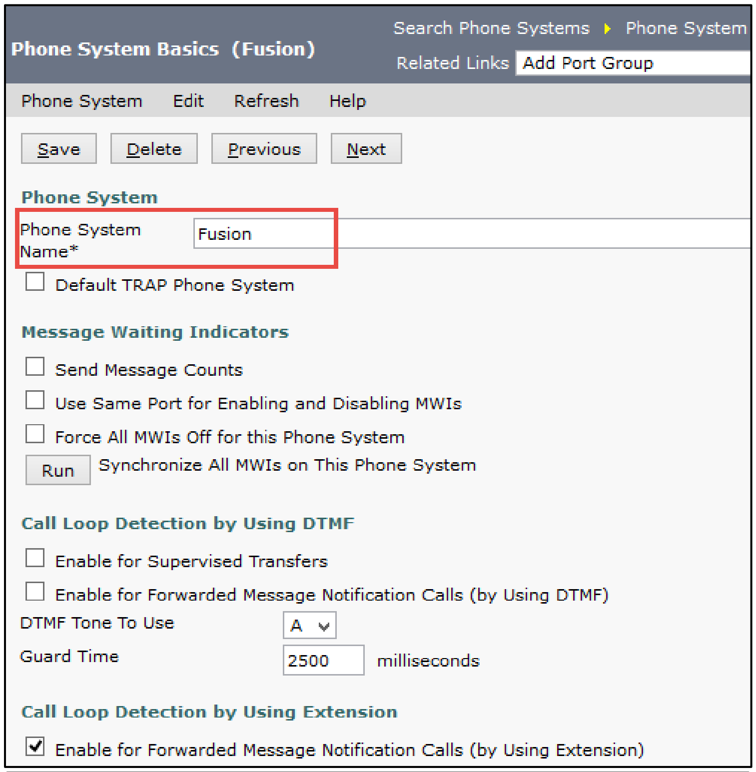


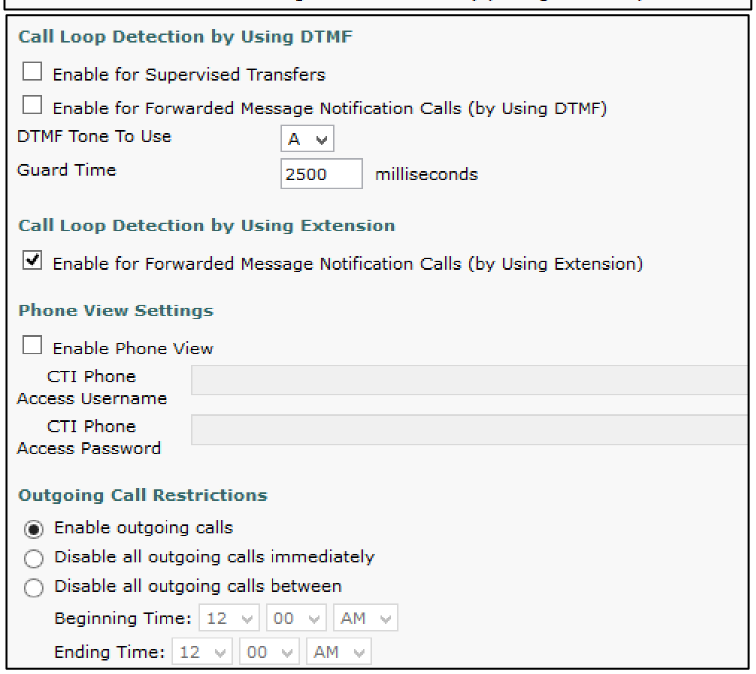
Note: Testing was conducted in tekVizion labs

Cisco Unity Connection Telephony Integration

Navigation: Telephony Integrations → Phone System

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



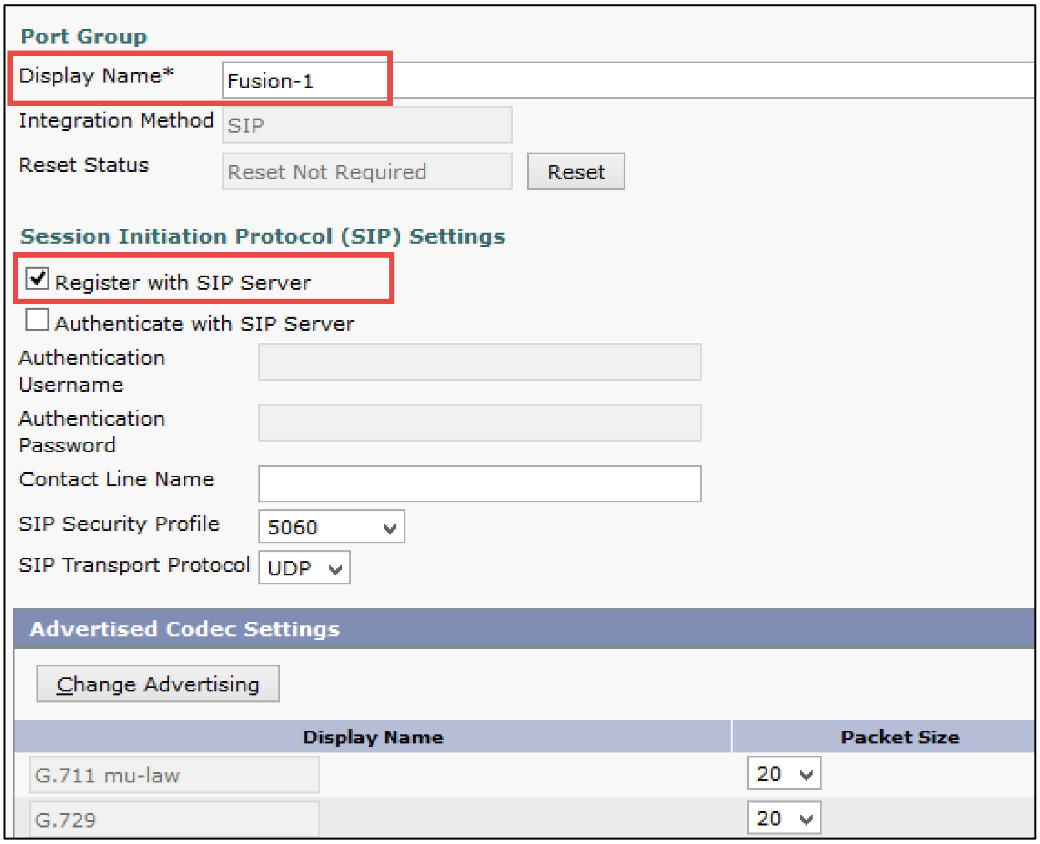


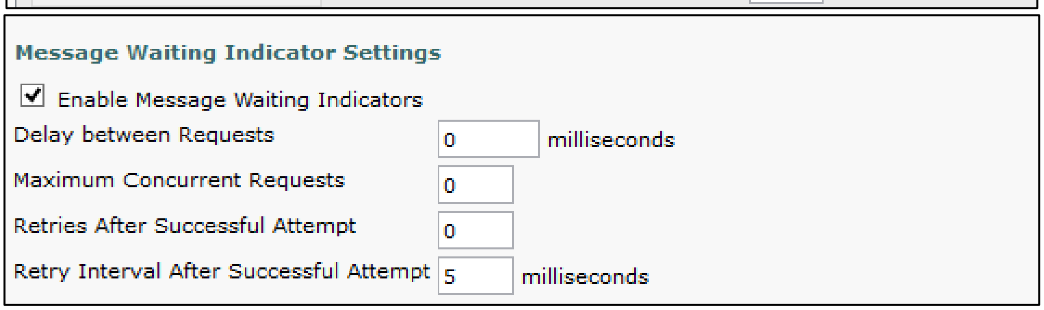
Note: Testing was conducted in tekVizion labs

Port Group

Navigation: Telephony Integrations → Port Group

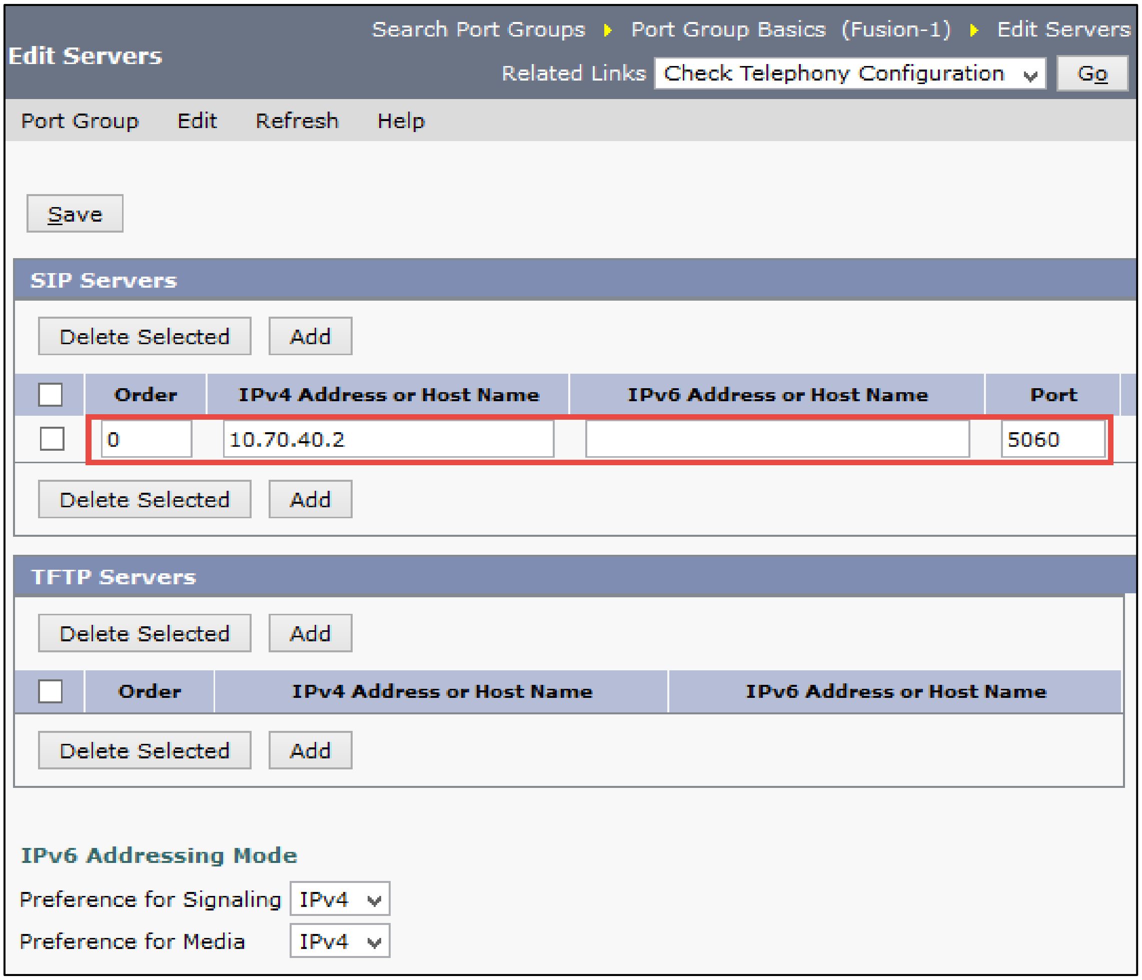
1. Set **Display Name** = *Fusion-1* is used for this example
2. Check **Register with SIP Server**





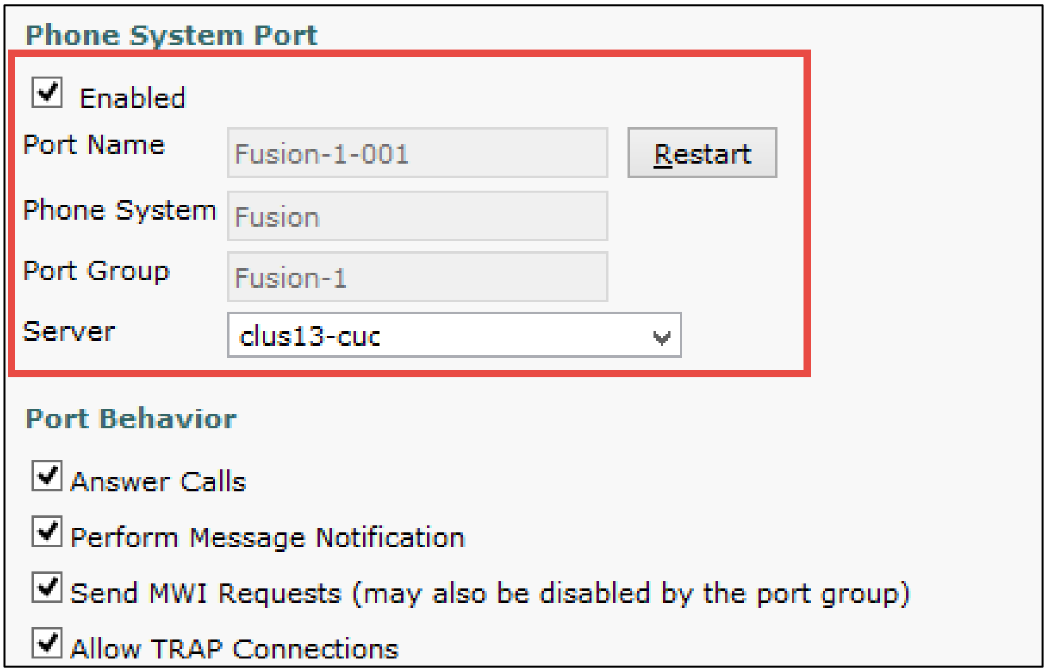
Note: Testing was conducted in tekVizion labs

Navigation: Telephony Integrations → Port Group → Edit → Servers



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

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)

Important Information

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