COX Business:

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

October 2016

Note: Testing was conducted in tekVizion labs

Table of Contents

Introduction

Service Providers today, such as COX Business, 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. COX 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.

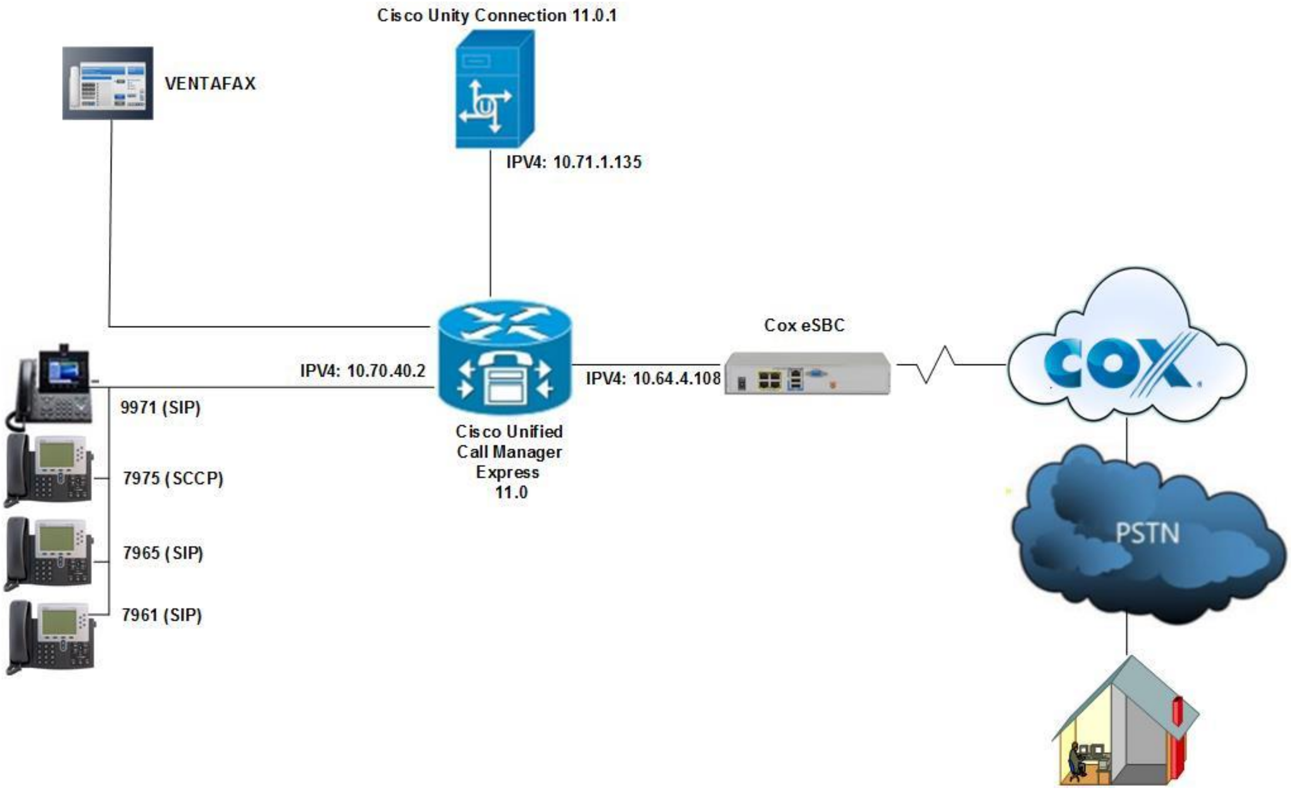
- This application note describes how to configure a Cisco Unified Communications Manager Express (Cisco Unified CME) 11.0, Cisco Unity Connection 11.0.1 with connectivity to Cox SIP trunk service. The application note also covers support and configuration example Cisco Unity Connection (CUC) messaging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

- Testing was performed in accordance to COX 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

Figure 1. Basic Call Setup



Hardware Components

- This solution was tested with Cisco ISR4431
- Cisco IP Phones. This solution was tested with 7961, 7965, 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 FTX1845AJ9S
- 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 IOS-XE 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 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

- Cisco IP phones used in this test do not support Blind Transfer, only Semi-attendant and Attendant transfers were tested
- COX doesn't support G729 codec at the moment
- Privacy ID feature is not supported by COX at the moment

Caveats

Fax

- The maximum fax rate achieved using T.38 (G3 or SG3) fax protocol is only 14400 kbps
- For T38 test related scenario (G3) achieved using "fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none" on dial peer
- For T38 test related scenario (SG3) achieved using "fax protocol t38 version 3 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 G711ulaw for auto attendant prompts. G.729 prompts can be used; however it was not tested here.

Hold & Resume

- Re-invites for hold/resume from the network would potentially depend on the carrier/network the call is traversing

Configuration Considerations

- 10 Digit Pilot is used to register Cisco Unified CME to COX SIP trunk (See configuration section for details)

- COX 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 (5-digit assignments in this example) and map 10-digit DID number using Cisco Unified CME translation patterns. Because 5-digit extensions configured on Cisco Unified CME IP phones, it is necessary to expand the 5-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)

Configuration

Cisco IOS Version

coxCME#sh 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

coxCME uptime is 1 hour, 21 minutes
Uptime for this control processor is 1 hour, 23 minutes
System returned to ROM by reload at 08:59:28 UTC Fri Sep 9 2016
System restarted at 09:04:13 UTC Fri Sep 9 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     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
```

```
coxCME#sh run
!
version 15.6
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname coxCME
!
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
!
subscriber templating
!

multilink bundle-name authenticated
!
```

```
voice service voip
 ip address trusted list
 no ip address trusted authenticate
 allow-connections sip to sip[1]
 no supplementary-service sip moved-temporarily
 no supplementary-service sip refer
 redirect ip2ip
 fax protocol pass-through g711ulaw[2]
 sip
  session refresh[3]
  registrar server[4]
!
voice class codec 1[5]
 codec preference 1 g729r8
 codec preference 2 g711ulaw
!
voice class sip-profiles 1[6]
 request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:40250\1@\2>"[7]
 request INVITE sip-header Privacy add "Privacy: id"[8]
!
voice register global
 mode  cme[9]
 source-address 10.70.40.2 port 5060[10]
 no privacy
```

[1] This command enables the Cisco Unified CME to perform basic SIP to SIP voice communication

[2] This command enables G711 fax at global level, meaning all VoIP dial-peers not configured for specific fax protocol will use this setting. G711 fax protocol may be configured under appropriate dial-peers

[3] Use the SIP session refresh command to send the session refresh request

[4] Enable Local SIP Registrar which is required for SIP phones in Cisco Unified CME

[5] This command enables multiple codec support and performs codec filtering required for correct interoperability between COX SIP network and Cisco Unified CME

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

[7] SIP profile is used to convert 5 digit extension number to 10 digit DID for call forward

[8] "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

[9] Enables the mode for configuring SIP IP phones in Cisco Unified CME

[10] This is the source address for SIP phone registration

```
 max-dn 10[11]
 max-pool 10[12]
 load 7945 SIP45.9-4-2SR1-1S[13]
 load 7961 SIP41.9-2-1S.loads
 load 9971 sip9971.9-2-2SR1-9
 timezone 8
 tftp-path flash:[14]
 file text
 create profile sync 0024421344338503[15]
!
voice register dn  2
 number 56069
 call-forward b2bua busy 402505XXXX
 name name CPE-3
 mwi
!
voice register dn  3
 number 54092
 name CPE-4
!
voice register pool  2
 busy-trigger-per-button 1
 id mac 001D.A21A.2791
 type 7961
 number 1 dn 2
 dtmf-relay rtp-nte
 voice-class codec 1
 username 56069 password X
!
```

[11] Configuration for maximum extensions
[12] Configuration for maximum phones
[13] Specify phone loads for each phone type
[14] Setup path for tftp files
[15] Create configuration files for all phones

```
voice register pool  3
 busy-trigger-per-button 1
 id mac 44AD.D9D5.7114
 type 7965
 number 1 dn 3
 dtmf-relay rtp-nte
 voice-class codec 1
 username 54092 password X
!
!
voice translation-rule 1[16]
 rule 1 /^40250\(54...\)$/ /\1/
 rule 2 /^40250\(5XXXX\)$/ /\1/
 rule 3 /^40261\(4XX..\)$/ /\1/
 rule 4 /^53122\(2XX..\)$/ /\1/
!
voice translation-rule 2[17]
 rule 2 /\(^40959\)/ /4XXXX\1/
 rule 3 /\(^54092\)/ /4XXXX\1/
 rule 4 /\(^56069\)/ /4XXXX\1/
 rule 5 /\(^28568\)/ /5XXXX\1/
!
!
voice translation-profile to-10dig
translate calling 2
!
voice translation-profile to-ext
translate called 1
!
!
!
```

[16] This translation rule/profile is used to modify the called number
[17] This translation rule/profile is used to modify the calling number

Note: Testing was conducted in tekVizion labs

```
!
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
 no ip address
 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
```


Note: Testing was conducted in tekVizion labs

```
 ip address 10.64.4.108 255.255.0.0
 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
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 10.64.1.1
ip route 10.70.40.0 255.255.255.0 10.70.40.1
ip route 172.16.24.0 255.255.248.0 10.70.40.1
!
tftp-server flash:.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:dkern9971.100609R2-9-2-2SR1-9.sebn
tftp-server flash:kern9971.9-2-2SR1-9.sebn
tftp-server flash:rootfs9971.9-2-2SR1-9.sebn
tftp-server flash:sboot9971.031610R1-9-2-2SR1-9.sebn
tftp-server flash:sip9971.9-2-2SR1-9.loads
tftp-server flash:skern9971.022809R2-9-2-2SR1-9.sebn
!
control-plane
!
!
voice-port 0/1/0
 cptone IN
 station-id number 402614XXXX
 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
 no privacy
 conference transfer-pattern
 max-ephones 20
 max-dn 20
 ip source-address 10.70.40.2 port 2000
 voicemail 7777[18]
 max-conferences 8 gain -6
 call-forward pattern .T
 moh enable-g711 "flash:wav"[19]
 transfer-system full-consult
 transfer-pattern .T
 create cnf-files version-stamp 7960 Sep 09 2016 09:27:28
!
!
dspfarm profile 2 mtp security
 codec g711ulaw
 maximum sessions software 20
 associate application SCCP
 shutdown
!
dial-peer voice 100 voip
 description inbound from Cox
 translation-profile incoming to-ext
 session protocol sipv2
```

[18] 7777 - Voice mail pilot number used in this example

[19] This is Music on Hold file configured in this example

Note: Testing was conducted in tekVizion labs

```
 session target sip-server
 session transport udp
 incoming called-number 402505....
 voice-class codec 1
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 200 voip
 description outbound to Cox
 translation-profile outgoing to-10dig
 destination-pattern .T
 session protocol sipv2
 session target sip-server[20]
 session transport udp
 voice-class codec 1[21]
 no voice-class sip asserted-id
 voice-class sip profiles 1[22]
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax protocol pass-through g711ulaw[23]
 no vad
!
dial-peer voice 101 voip
 description inbound from Cox
 translation-profile incoming to-ext
 session protocol sipv2
 session target sip-server
 session transport udp
 incoming called-number 402614....
```

[20] This command sets the SIP server target for outgoing SIP calls
[21] This command assigns the voice class codec setting to this dial-peer
[22] This commands assigns the applicable SIP profile to use for this dial-peer
[23] Example of configuring G711 as fax protocol per dial peer

```
 voice-class codec 1
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 7777 voip[24]
 description voice mail pilot
 max-conn 8
 destination-pattern 7777
 session protocol sipv2
 session target ipv4:10.71.1.135:5060[25]
 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
 no vad
!
dial-peer voice 102 voip
 description inbound from Cox
 translation-profile incoming to-ext
 huntstop
 session protocol sipv2
 session target sip-server
 session transport udp
 incoming called-number 531222....
 voice-class codec 1
 dtmf-relay rtp-nte
 no vad
```

[24] Dial peer used to route the calls to Cisco unity connection server for Voicemail access
[25] This command sets the Cisco unity connection server target for voicemail calls

```
!
dial-peer voice 999 voip
 description "Emergency call"
 translation-profile outgoing to-10dig
 destination-pattern 911
 session protocol sipv2
 session target sip-server
 session transport udp
 voice-class codec 1
 no voice-class sip asserted-id
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 1000 pots
 preference 2
 service session
 destination-pattern 40959
 no digit-strip
 port 0/1/0
!
!
sip-ua
 credentials number 402359XXXX username 402359XXXX password 7 402359XXXX realm
asterisk
 authentication username 402359XXXX password 7 402359XXXX realm asterisk
 mwi-server ipv4:10.71.1.135 expires 3600 port 5060 transport udp unsolicited
 registrar ipv4:10.64.4.164 expires 3600
 sip-server ipv4:10.64.4.164:5060
!
!
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 28568
 label sccp
 name sccp
!
!
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
!
!
!
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
```


Note: Testing was conducted in tekVizion labs

```
 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
```

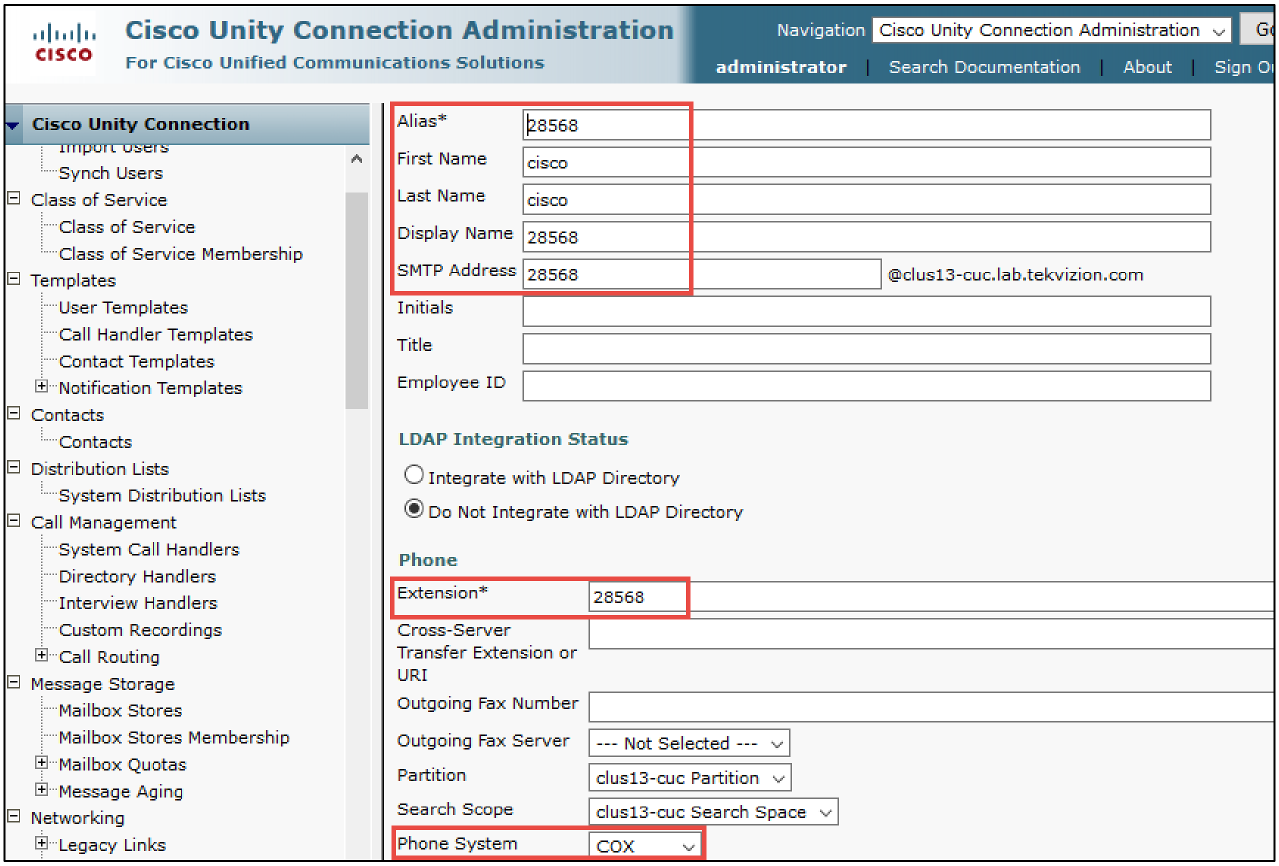
Cisco Unity Connection

Version Details

Cisco Unity Connection User Configuration

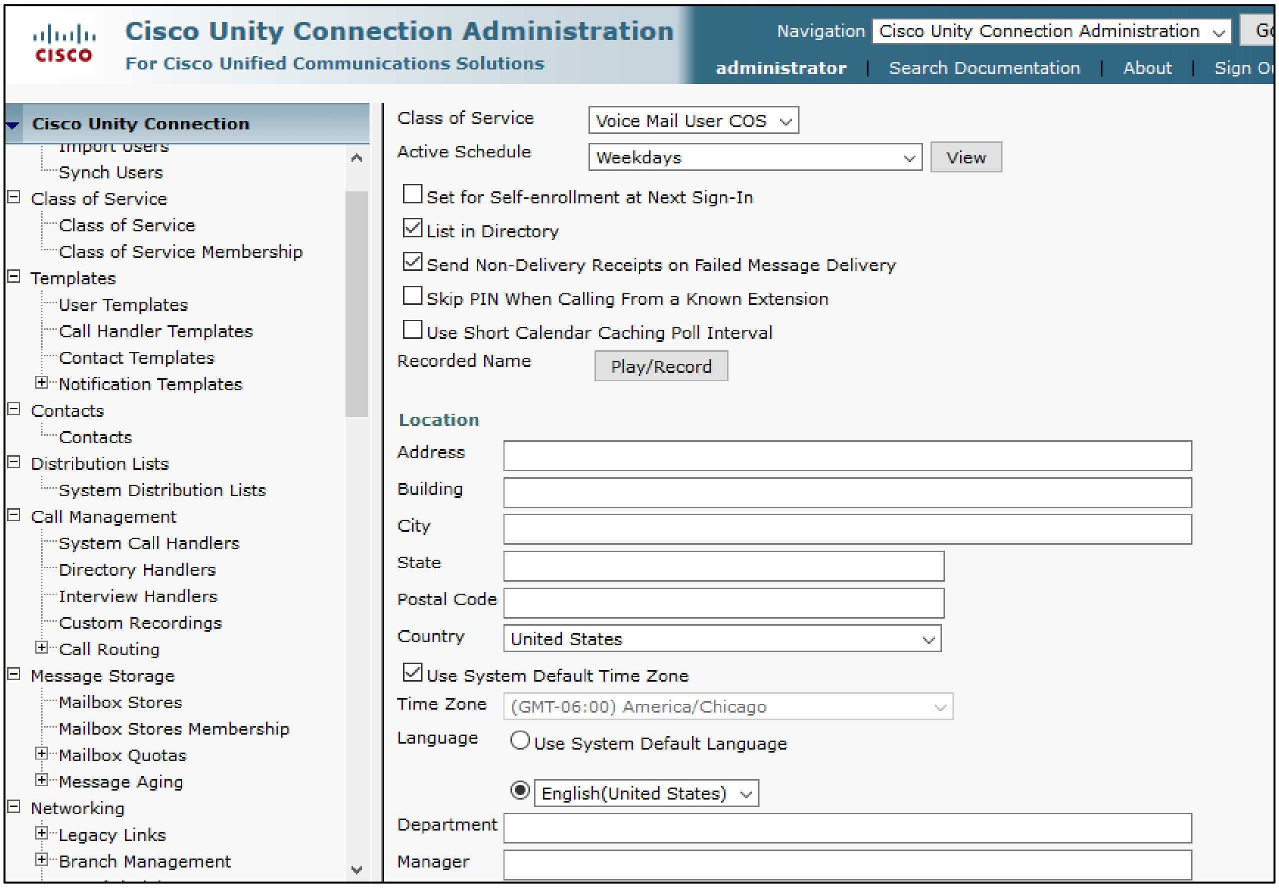
Navigation: Cisco Unity Connection → Users → Users

1. Set **Alias**= 28568 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** = 28568 is used in this example
5. Set **SMTP Address** = 28568 is used in this example
6. Set **Extension** = 28568 is used in this example
7. Set **Phone System** = COX is used in this example
8. All other values are default


Cisco Unity Connection Administration
For Cisco Unified Communications Solutions
Navigation Cisco Unity Connection Administration
administrator | Search Documentation | About | Sign Out
Cisco Unity Connection
Import Users
Synch Users
Class of Service
Class of Service
Class of Service Membership
Templates
User Templates
Call Handler Templates
Contact Templates
Notification Templates
Contacts
Contacts
Distribution Lists
System Distribution Lists
Call Management
System Call Handlers
Directory Handlers
Interview Handlers
Custom Recordings
Call Routing
Message Storage
Mailbox Stores
Mailbox Stores Membership
Mailbox Quotas
Message Aging
Networking
Legacy Links
Alias* 28568
First Name cisco
Last Name cisco
Display Name 28568
SMTP Address 28568 @clus13-cuc.lab.tekvizion.com
Initials
Title
Employee ID
LDAP Integration Status
Integrate with LDAP Directory
Do Not Integrate with LDAP Directory
Phone
Extension* 28568
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 COX

Cisco Unity Connection User Configuration (Continued)

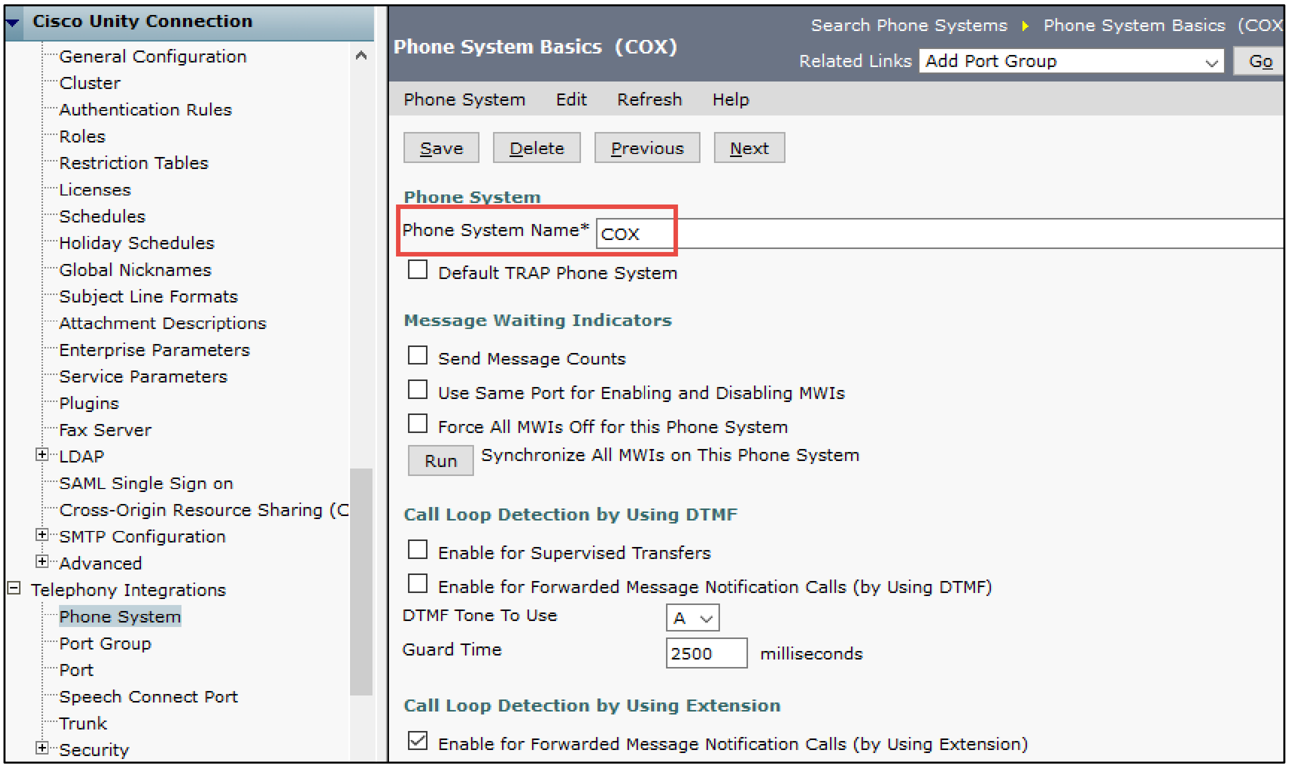
9. All values are default

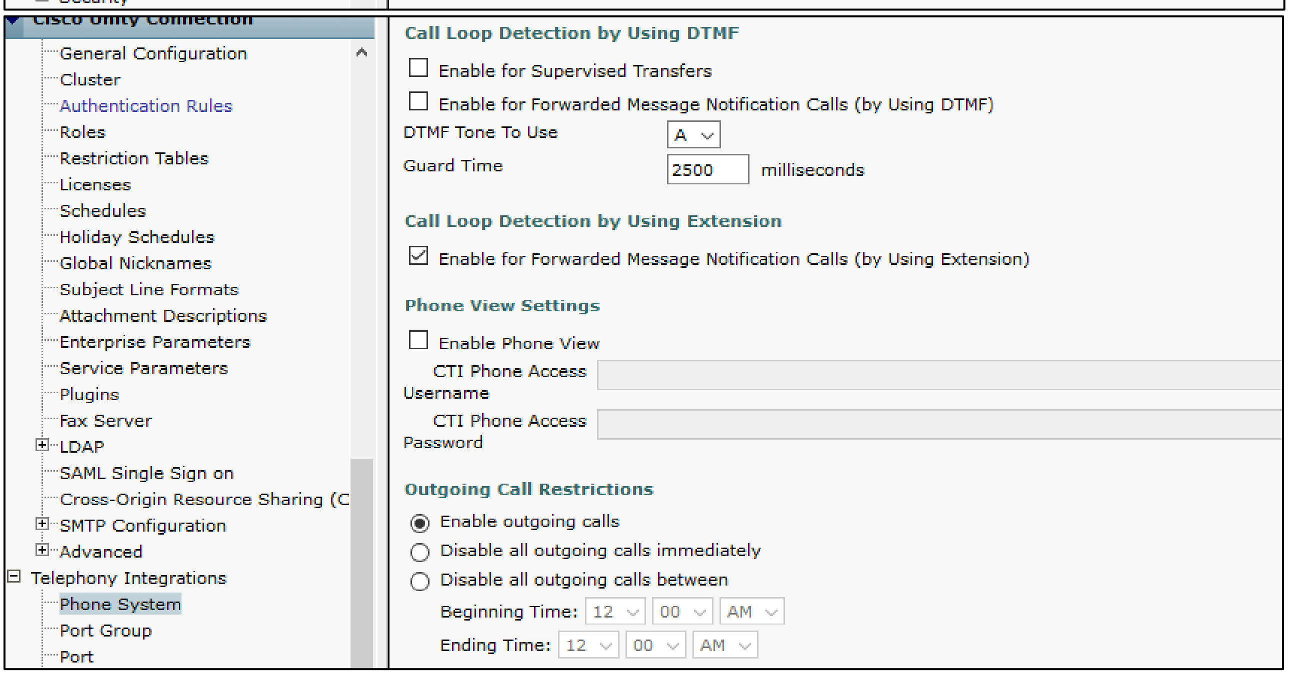


Cisco Unity Connection Telephony Integration

Navigation: Telephony Integrations → Phone system

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

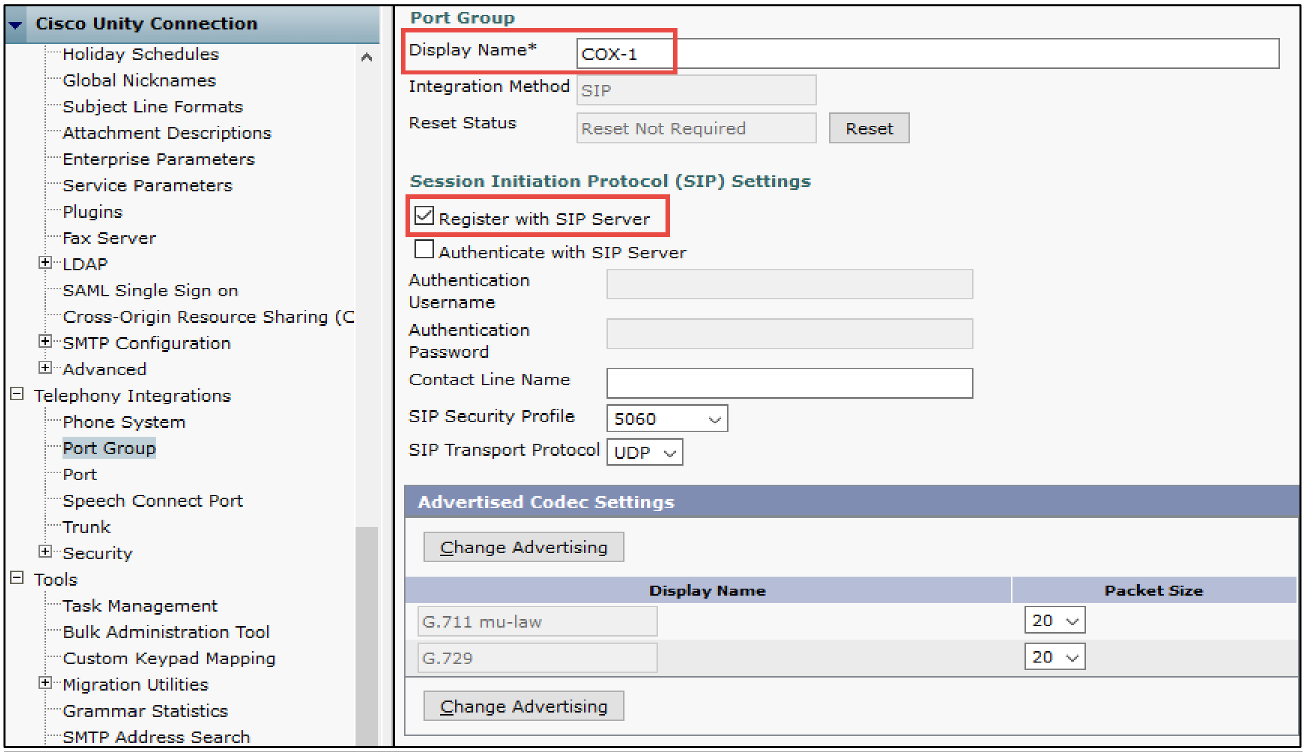


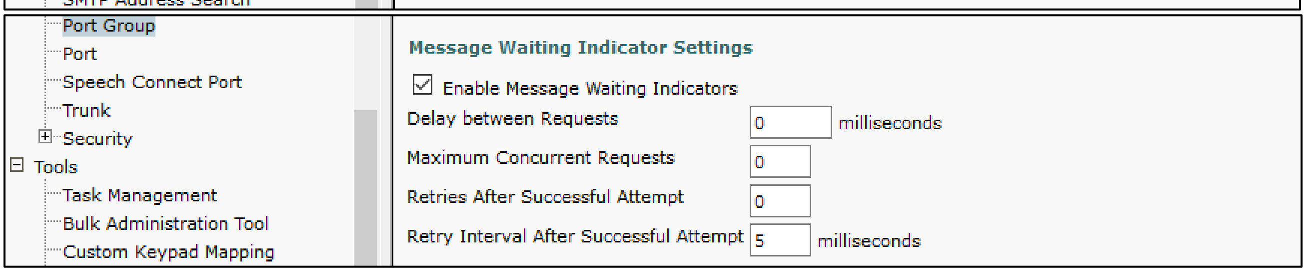


Port Group

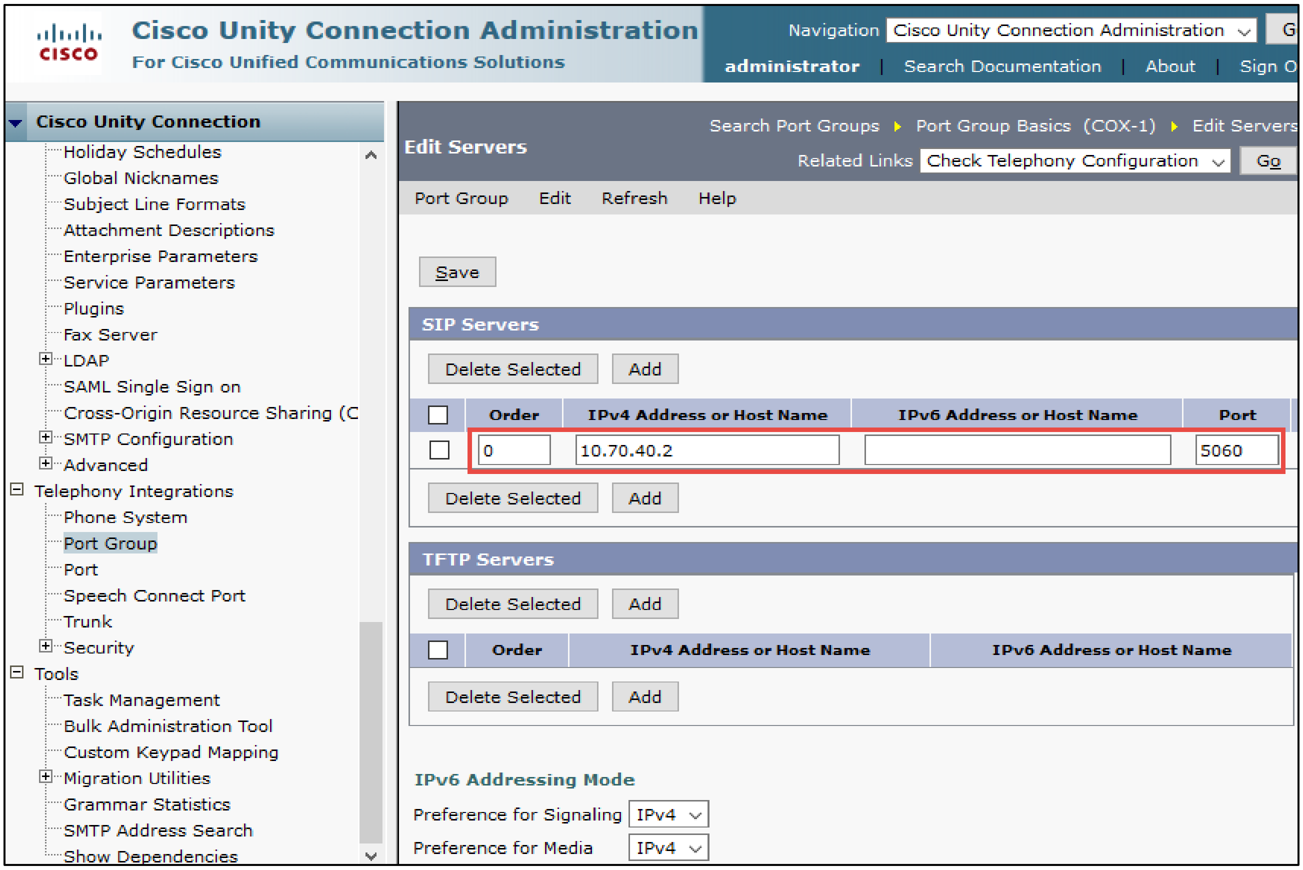
Navigation: Telephony Integration → Port Group

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



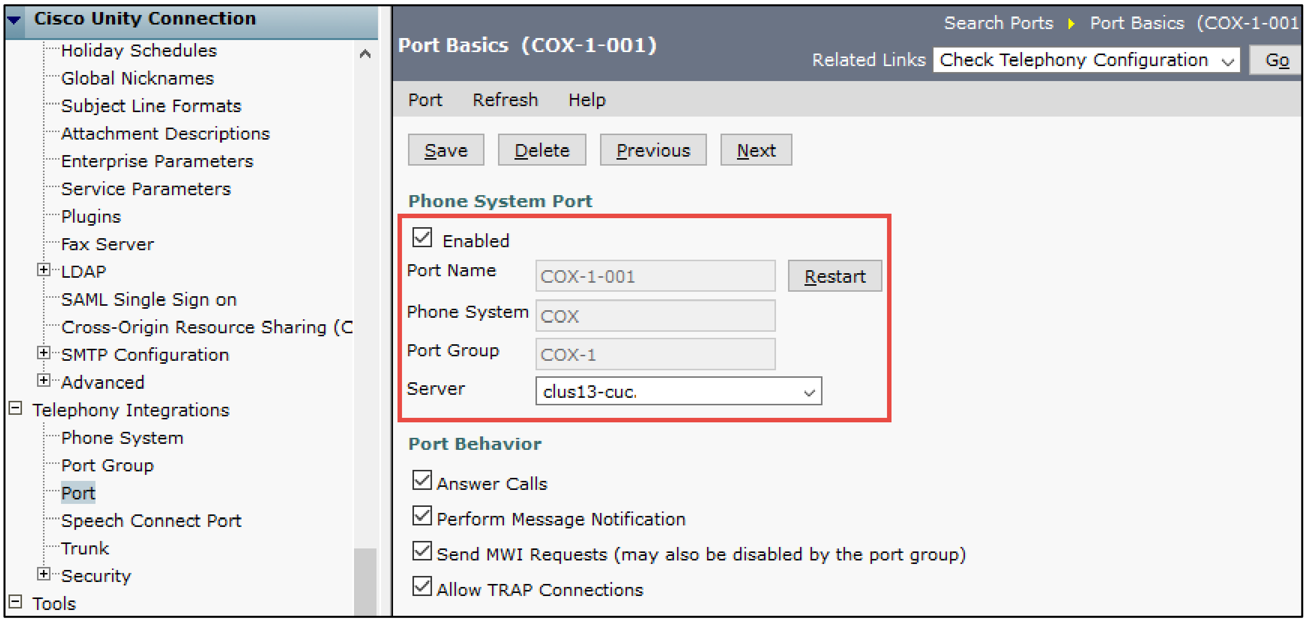


Navigation: Telephony Integration→ Port Group → Edit → Servers



Port

1. Set **Port Name** = COX-1-001 is used for this example
2. Set **Phone System** = COX
3. Set **Port Group** = COX-1
4. Set **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
CODEC	Coder-Decoder (in this document a device used to digitize and un-digitize voice signals)

Important Information

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