



COX Business:

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

October 2016



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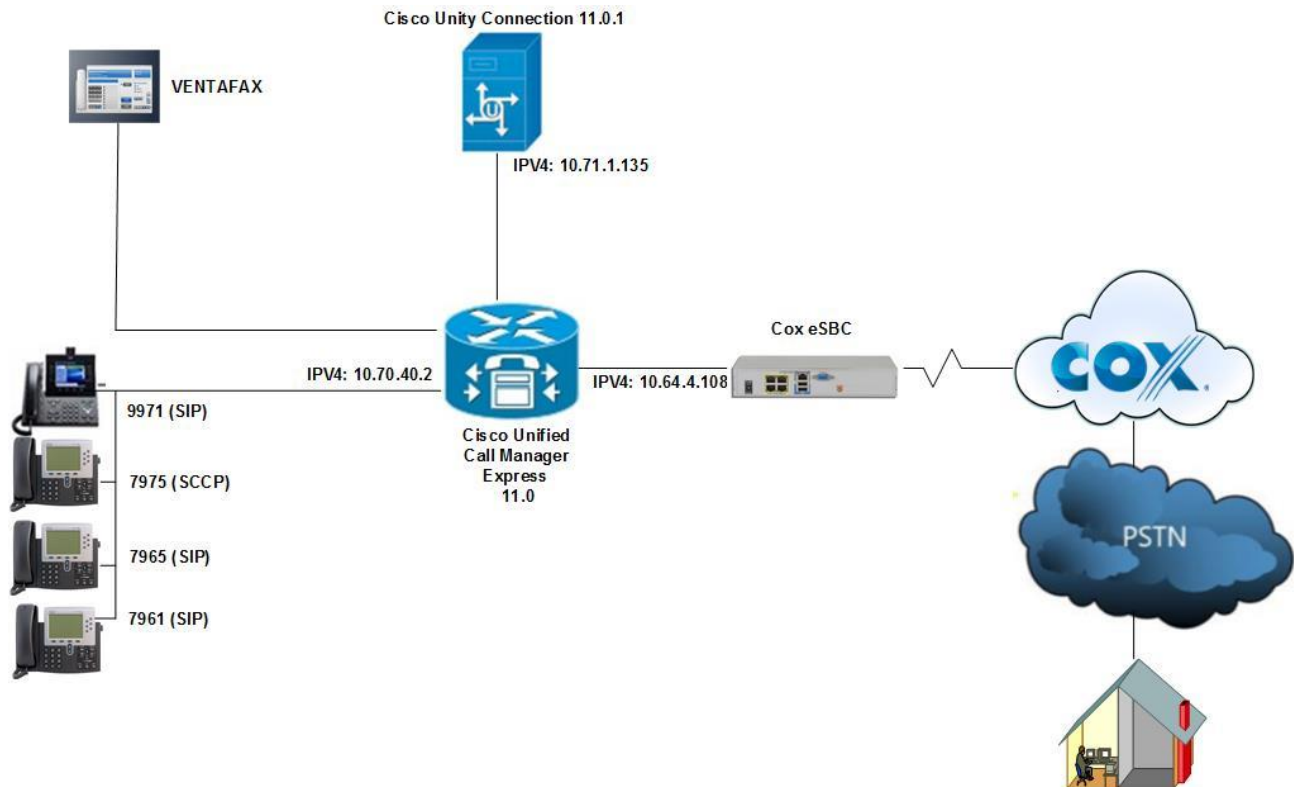
Introduction

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)



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
```

Cisco IOS-XE software, Copyright (c) 2005-2015 by cisco Systems, Inc.
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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 securityk9 appxk9	None	None	None
AdvUCSuiteK9 uck9 cme-srst cube	None	None	None

Technology Package License Information:

Technology	Technology-package Current	Technology-package Type	Technology-package 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

```
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 sip1
no supplementary-service sip moved-temporarily
no supplementary-service sip refer
redirect ip2ip
fax protocol pass-through g711ulaw2
sip
  session refresh3
  registrar server4
!
voice class codec 15
  codec preference 1 g729r8
  codec preference 2 g711ulaw
!
voice class sip-profiles 16
  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 cme9
  source-address 10.70.40.2 port 506010
  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 1011
max-pool 1012
load 7945 SIP45.9-4-2SR1-1S13
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 002442134433850315
!
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 116
  rule 1 /^40250(54...)$/ ^1/
  rule 2 /^40250(5XXXX)$/ ^1/
  rule 3 /^40261(4XX..)$/ ^1/
  rule 4 /^53122(2XX..)$/ ^1/
!
voice translation-rule 217
  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



```
!  
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
```



```
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 777718
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



```
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-server20
session transport udp
voice-class codec 121
no voice-class sip asserted-id
voice-class sip profiles 122
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711ulaw23
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 voip24
description voice mail pilot
max-conn 8
destination-pattern 7777
session protocol sipv2
session target ipv4:10.71.1.135:506025
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 Cfdall 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
```

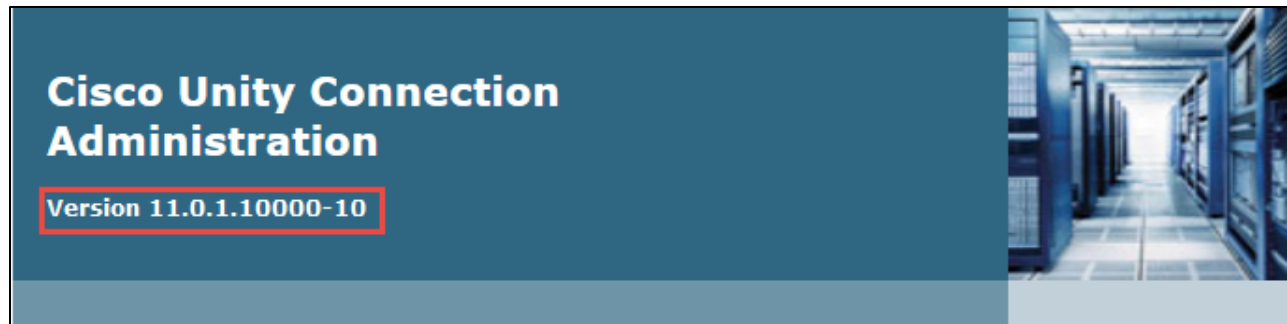


```
login local
transport preferred ssh
transport input telnet ssh
line vty 1 4
exec-timeout 0 0
logging synchronous
login local
transport input telnet ssh
!
ntp server 10.10.10.5
end
```



Cisco Unity Connection

Version Details



Cisco Unity Connection Administration

Version 11.0.1.10000-10

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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 our [Export Compliance Product Report](#) web site.

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 On

- ▼ 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
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 - 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

The screenshot displays the Cisco Unity Connection Administration web interface. The left sidebar shows a navigation tree with 'Cisco Unity Connection' expanded. The main content area shows configuration options for a user, with most values set to their defaults. The 'Class of Service' is 'Voice Mail User COS', and the 'Active Schedule' is 'Weekdays'. Several checkboxes are present, with 'List in Directory' and 'Send Non-Delivery Receipts on Failed Message Delivery' checked. The 'Location' section includes fields for Address, Building, City, State, Postal Code, and Country (set to 'United States'). The 'Language' is set to 'English(United States)'. The 'Department' and 'Manager' fields are empty.

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
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 - 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
 - Branch Management

Class of Service: Voice Mail User COS

Active Schedule: Weekdays [View]

Set for Self-enrollment at Next Sign-In

List in Directory

Send Non-Delivery Receipts on Failed Message Delivery

Skip PIN When Calling From a Known Extension

Use Short Calendar Caching Poll Interval

Recorded Name: [Play/Record]

Location

Address: []

Building: []

City: []

State: []

Postal Code: []

Country: United States

Use System Default Time Zone

Time Zone: (GMT-06:00) America/Chicago

Language: Use System Default Language

English(United States)

Department: []

Manager: []



Cisco Unity Connection Telephony Integration

Navigation: Telephony Integrations → Phone system

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

The screenshot displays the Cisco Unity Connection configuration interface for a Phone System named 'COX'. The left sidebar shows the navigation tree with 'Telephony Integrations' expanded to 'Phone System'. The main content area is titled 'Phone System Basics (COX)' and includes the following sections:

- Phone System Basics (COX)**: Search Phone Systems, Phone System Basics (COX), Related Links, Add Port Group, Go.
- Phone System**: Save, Delete, Previous, Next. Phone System Name* COX (highlighted with a red box).
 - 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** (repeated):
 - 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** (repeated):
 - Enable for Forwarded Message Notification Calls (by Using Extension)
- Phone View Settings**:
 - Enable Phone View
 - CTI Phone Access Username: [text field]
 - CTI Phone Access Password: [text field]
- 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 Integration → Port Group

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

Cisco Unity Connection

- ... Holiday Schedules
- ... Global Nicknames
- ... Subject Line Formats
- ... Attachment Descriptions
- ... Enterprise Parameters
- ... Service Parameters
- ... Plugins
- ... Fax Server
- ⊕ LDAP
- ... SAML Single Sign on
- ... Cross-Origin Resource Sharing (C
- ⊕ SMTP Configuration
- ⊕ Advanced
- ⊖ Telephony Integrations
 - ... Phone System
 - Port Group**
 - ... Port
 - ... Speech Connect Port
 - ... Trunk
- ⊕ Security
- ⊖ Tools
 - ... Task Management
 - ... Bulk Administration Tool
 - ... Custom Keypad Mapping
- ⊕ Migration Utilities
- ... Grammar Statistics
- ... SMTP Address Search

Port Group

Display Name*

Integration Method

Reset Status

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

Display Name	Packet Size
G.711 mu-law	<input type="text" value="20"/> ▾
G.729	<input type="text" value="20"/> ▾

Port Group

- ... Port
- ... Speech Connect Port
- ... Trunk
- ⊕ Security
- ⊖ Tools
 - ... Task Management
 - ... Bulk Administration Tool
 - ... Custom Keypad Mapping

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 Integration → Port Group → Edit → Servers

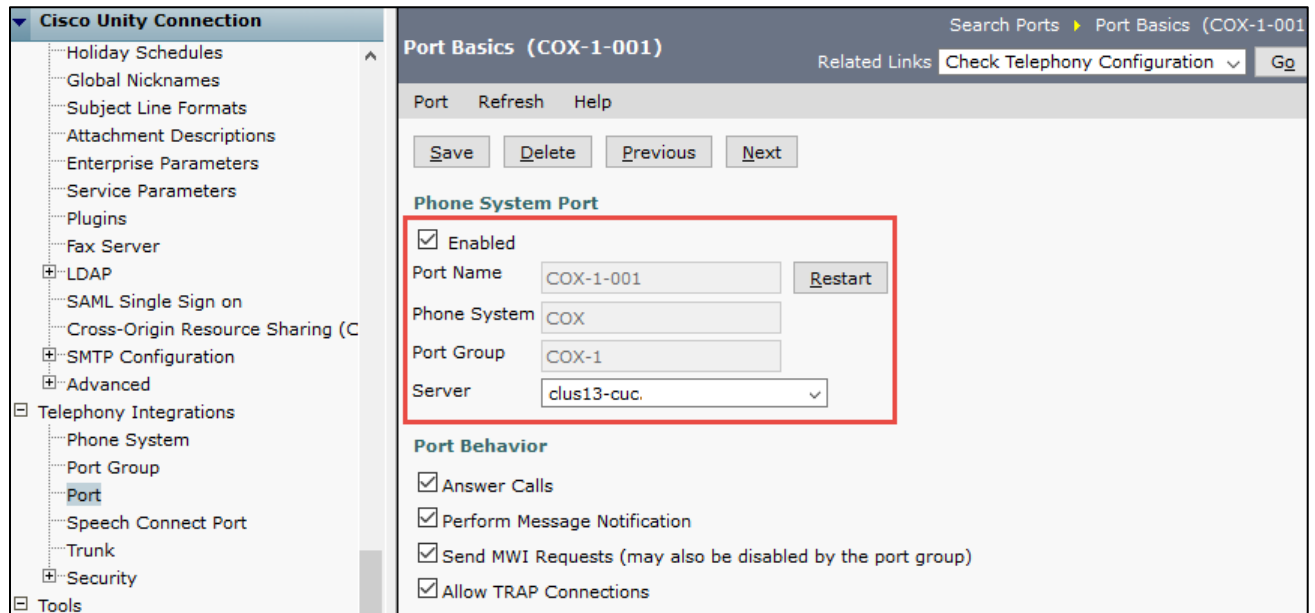
The screenshot displays the Cisco Unity Connection Administration interface. The left sidebar shows the navigation menu with 'Telephony Integrations' expanded to 'Port Group'. The main content area is titled 'Edit Servers' and includes a breadcrumb trail: 'Search Port Groups > Port Group Basics (COX-1) > Edit Servers'. Below the breadcrumb, there are buttons for 'Port Group', 'Edit', 'Refresh', and 'Help', and a 'Save' button. The 'SIP Servers' section contains a 'Delete Selected' and 'Add' button, followed by a table with the following data:

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

Below the table are 'Delete Selected' and 'Add' buttons. The 'TFTP Servers' section also has 'Delete Selected' and 'Add' buttons. At the bottom, the 'IPv6 Addressing Mode' section shows 'Preference for Signaling' and 'Preference for Media' both set to 'IPv4' via dropdown menus.

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



The screenshot shows the Cisco Unity Connection interface for configuring a port. The left sidebar shows a tree view with 'Port' selected under 'Telephony Integrations'. The main content area is titled 'Port Basics (COX-1-001)'. It includes a search bar, a 'Related Links' section with a 'Check Telephony Configuration' link, and navigation buttons: 'Save', 'Delete', 'Previous', and 'Next'. The 'Phone System Port' section is highlighted with a red box and contains the following configuration:

- Enabled
- Port Name: COX-1-001 (with a 'Restart' button)
- Phone System: COX
- Port Group: COX-1
- Server: clus13-cuc (dropdown menu)

The 'Port Behavior' section below contains the following checked options:

- 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
CODEC	Coder-Decoder (in this document a device used to digitize and un-digitize voice signals)

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Printed in the USA