



Connecting Cisco Unified Communication Manager [v12.0.1] to Cox Business SIP Trunks via Cisco Unified Border Element v12.0 [IOS-XE 16.06.01]

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Introduction

Service Providers today, such as Cox, 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 centralized IP to TDM POP gateways 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. A demarcation device between these services and customer owned services is recommended. As an intermediary device between Cisco Unified Communications Manager (Cisco UCM) and Cox network, Cisco Unified Border Element (Cisco UBE) ISR 4331/K9 running IOS 16.6.1 can be used. The Cisco Unified Border Element 16.6.1 provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 12.0.1 connected to Cox network.

This document assumes the reader is knowledgeable with the terminology and configuration of Cisco Unified Communications Manager. Only configuration settings specifically required for Cox interoperability are presented. Feature configuration and most importantly the dial plan are customer specific and need individual approach.

- This application note describes how to configure Cisco UCM 12.0.1, and Cisco UBE on ISR 4331/K9 [IOS – 16.6.1] for connectivity to Cox SIP Trunking service. The deployment model covered in this application note is CPE (Cisco UCM) to PSTN (Cox) via Cisco Border UBE v12.0 [IOS-XE] 16.6.1.
- Testing was performed in accordance to Cox generic SIP Trunking test methodology and among features verified were – basic calls, DTMF transport, Music on Hold (MOH), unattended and attended transfers, call forward, conferences and High Availability.
- The Cisco UBE configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between Cox SIP network and Cisco Unified Communications. The configuration described in this document details the important configuration settings to have enabled for interoperability to be successful and care must be taken by the network administrator deploying Cisco UCM to interoperate to Cox SIP Trunking network.



Network Topology

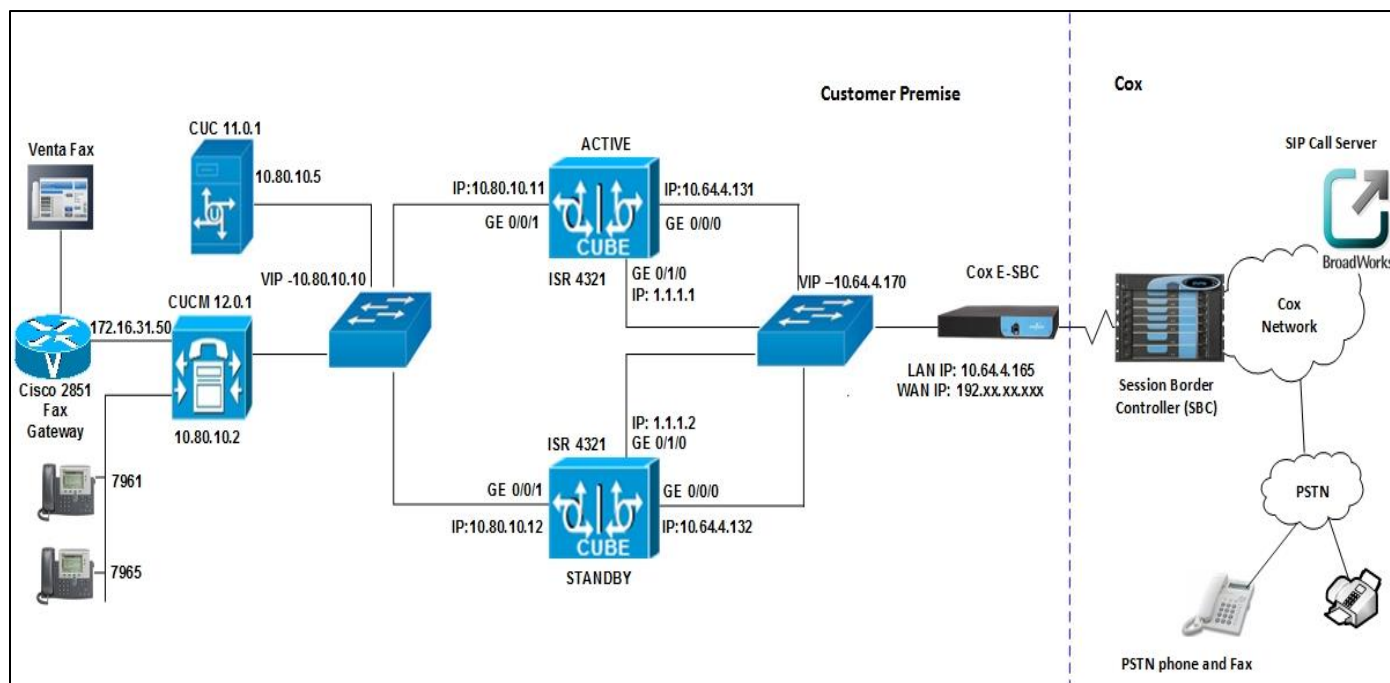


Figure 1 Network Topology

- The network topology includes the Cisco UCM Cluster, Unity Voicemail system, Cisco Fax gateway and 2 Cisco Endpoints. Cisco UCM has a trunk configured to Cisco UBE's Virtual IP Address. Cox was used as the service provider with SIP trunk to the Cisco UBE using the WAN Virtual IP Address.
- 2 Cisco Unified Border Elements are used here for High Availability.
- SIP Trunk transport type used between Cisco Unified Border Element and Cisco UCM is UDP and to Cox is UDP.

Cisco UCM and Cisco UBE Settings:

Setting	Value
Transport from Cisco UBE to Cisco UCM	UDP with RTP
Transport from Cisco UBE to Cox	UDP with RTP
Voice Mail Support	YES
Session Refresh	YES
Early Media support with PRACK	YES



System Components

Hardware Requirements

- Cisco UBE on Cisco ISR 4331 router
- CUCM cluster on UCS, 1 Publisher node and 2 Subscriber nodes
- Cisco 2851 with FXS ports and Analog Fax machine
- Generic Cisco IP-Phones

Software Requirements

- CUBE-Version: 12.0 running IOS-XE 16.6.1
- CUCM UCOS 12.0.1.10000-10 for 1 Publisher and 2 Subscriber
- Cisco IOS v15.1 for the fax gateway

Features

Features Supported

- Incoming and outgoing off-net calls using G711ulaw and G729 voice codecs
- International Calls and digit manipulations
- Call Conference
- CPE Voice Mail support
- Call hold & Resume with and without MoH
- Unattended and Attended Call transfer
- Call forward (all, busy and no answer)
- DTMF (RFC2833)
- Fax (T38 and G711 Pass-through)
- IP-PBX Calling number privacy
- High Availability

Features Not Supported

- Cisco UCM does not support Blind Call transfer
- In HA Redundancy mode, the Primary Cisco UBE will not take over the Primary/Active role after a reboot/network outage
- Cox does not support G729 codec



Caveats

- Caller ID does not get updated on attended and unattended transfer scenarios.
- Only one IP PBX used for the testing.
- The Cisco UBE HA tested is layer 2 box to box Cisco UBE redundancy.
- The maximum fax rate achieved using T.38 (G3 or SG3) fax protocol is only 14400 kbps
- For T38 test related scenario (G3 or SG3) 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



Configuration

Configuring Cisco Unified Border Element

The following configuration snippet contains a sample configuration of Cisco Unified Border Element.

Network Interface

The IP address used are for illustration only, the actual IP address can vary. The Active/Standby pair share the same virtual IP address and continually exchange status messages.

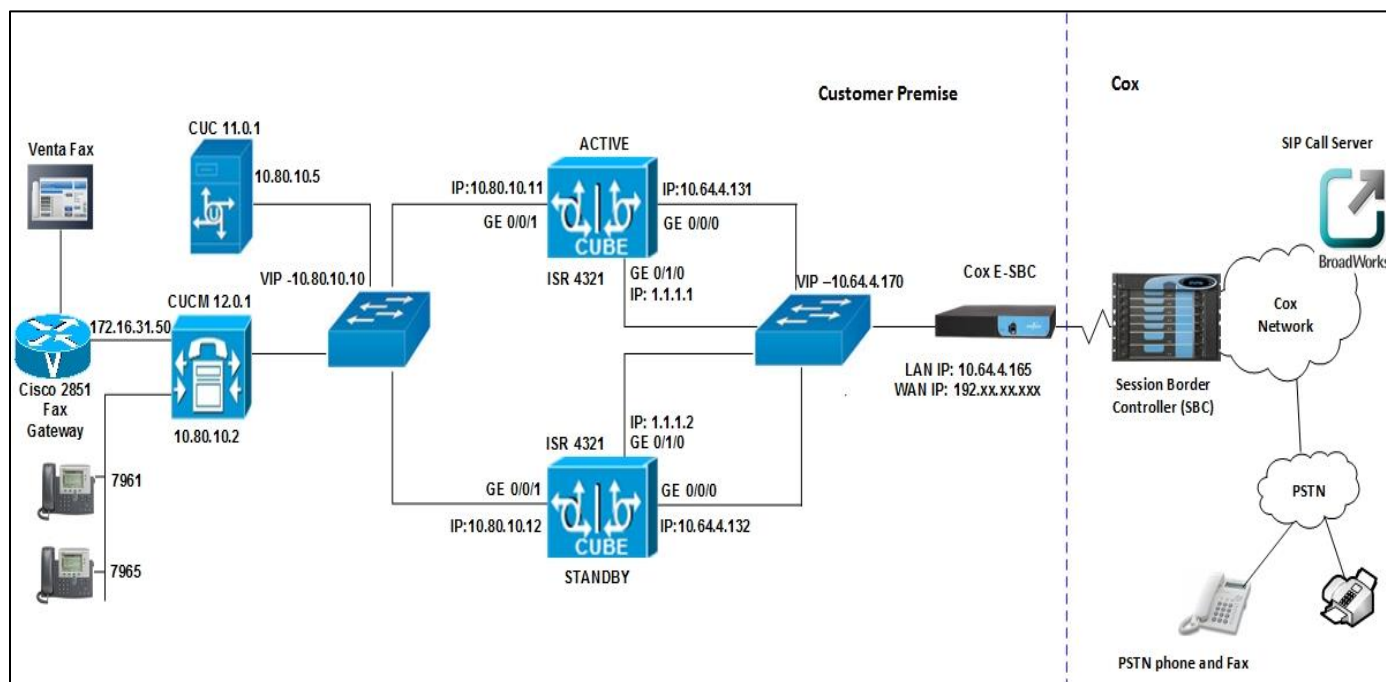


Figure 2 High Availability topology



Cisco UBE 1:

```
interface GigabitEthernet0/0/0
  description Cox WAN
  ip address 10.64.4.131 255.255.0.0
  negotiation auto
  redundancy rii 4
  redundancy group 1 ip 10.64.4.170 exclusive
!
!
interface GigabitEthernet0/0/1
  description Cox LAN
  ip address 10.80.10.11 255.255.255.0
  negotiation auto
  redundancy rii 3
  redundancy group 1 ip 10.80.10.10 exclusive
!
!
interface GigabitEthernet0/0/2
  description Cox CubeHA Interface
  ip address 1.1.1.1 255.255.255.252
  negotiation auto
!
interface GigabitEthernet0
  vrf forwarding Mgmt-intf
  no ip address
  negotiation auto
```



Cisco UBE 2:

```
interface GigabitEthernet0/0/0
  description Cox WAN
  ip address 10.64.4.132 255.255.0.0
  negotiation auto
  redundancy rii 4
  redundancy group 1 ip 10.64.4.170 exclusive
!
interface GigabitEthernet0/0/1
  description Cox LAN
  ip address 10.80.10.12 255.255.255.0
  negotiation auto
  redundancy rii 3
  redundancy group 1 ip 10.80.10.10 exclusive
!
interface GigabitEthernet0/0/2
  description Cox CubeHA Interface
  ip address 1.1.1.2 255.255.255.252
  negotiation auto
!
interface GigabitEthernet0
  vrf forwarding Mgmt-intf
  no ip address
  shutdown
  negotiation auto
!
```



Global Cisco UBE settings

In order to enable Cisco UBE IP2IP SBC functionality, following command has to be entered:

```
voice service voip
no ip address trusted authenticate
address-hiding
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
sip
bind control source-interface GigabitEthernet0/0/0
rel1xx disable
session refresh
header-passing
asserted-id pai
privacy pstn
early-offer forced
midcall-signaling passthru
g729 annexb-all
```

Explanation

Command	Description
allow-connections sip to sip	Allow IP2IP connections between two SIP call legs
redundancy-group 2	Enable High Availability for the VoIP service
fax protocol	Specifies the fax protocol
asserted-id	Specifies the privacy header in the outgoing SIP requests and response messages



Codecs

G711ulaw and G729 voice codecs are used for this testing. Codec preferences used to change according to the test plan description.

```
voice class codec 1
  codec preference 1 g711ulaw
  codec preference 2 g729r8
!
voice class codec 2
  codec preference 1 g729r8
  codec preference 2 g711ulaw
!
voice class codec 3
  codec preference 1 g729r8
!
```

Dial peer

Outbound Dial-peer to Cox:

```
dial-peer voice 510 voip
  description outgoing call to Cox - Lan Facing
  huntstop
  session protocol sipv2
  session transport udp
  incoming called-number [0-9]T
  voice-class codec 1
  voice-class sip asserted-id pai
  voice-class sip bind control source-interface GigabitEthernet0/0/1
  voice-class sip bind media source-interface GigabitEthernet0/0/1
  dtmf-relay rtp-nte
  no vad
!
!
```



```
dial-peer voice 520 voip
description outgoing call to Cox - WAN Facing
huntstop
destination-pattern .T
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
no voice-class sip outbound-proxy
voice-class sip profiles 101
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
no vad
!
```

Inbound Dial-peer from Cox:

```
dial-peer voice 610 voip
description incoming call to PBX-WAN facing
huntstop
session protocol sipv2
session transport udp
incoming called-number 402.....
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
no vad
!
```



```
!  
dial-peer voice 620 voip  
  description incoming call to PBX-LAN facing  
  huntstop  
  destination-pattern 402.....  
  session protocol sipv2  
  session target ipv4:10.80.10.2  
  session transport udp  
  voice-class codec 1  
  voice-class sip asserted-id pai  
  no voice-class sip outbound-proxy  
  voice-class sip options-keepalive  
  voice-class sip bind control source-interface GigabitEthernet0/0/1  
  voice-class sip bind media source-interface GigabitEthernet0/0/1  
  dtmf-relay rtp-nte  
  no vad  
!
```



Configuration example

The following configuration snippet contains a sample configuration of Cisco UBE with all parameters mentioned previously.

Active Cisco UBE:

User Access Verification

Username: cisco

Password:

CoxCertCube7#sh run

```
version 16.6
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname CoxCertCube7
!
boot-start-marker
boot-end-marker
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
enable secret 5 $1$k8fq$3FBrZ5Ub4lioq2gwvc3e0
```




```
no aaa new-model
no ip domain lookup
subscriber templating
!
multilink bundle-name authenticated
!
crypto pki trustpoint TP-self-signed-2194658987
  enrollment selfsigned
  subject-name cn=IOS-Self-Signed-Certificate-2194658987
  revocation-check none
  rsakeypair TP-self-signed-2194658987
!
!
crypto pki certificate chain TP-self-signed-2194658987
!
voice service voip
  no ip address trusted authenticate
  address-hiding
  mode border-element license capacity 20
  allow-connections sip to sip
  redundancy-group 1
  fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
  sip
    bind control source-interface GigabitEthernet0/0/0
    rel1xx disable
    session refresh
    header-passing
    asserted-id pai
    privacy pstn
```



```
early-offer forced
midcall-signaling passthru
g729 annexb-all
!
voice class codec 1
  codec preference 1 g711ulaw
  codec preference 2 g729r8
!
voice class codec 2
  codec preference 1 g729r8
  codec preference 2 g711ulaw
!
voice class codec 3
  codec preference 1 g729r8
!
!
voice class sip-profiles 100
!
voice class sip-profiles 101
  request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>"
  "<sip:402932\1@\2>"
!
voice translation-rule 1
  rule 1 /\^.*\(\.....\)/ /\1/
!
!
voice translation-profile Cox
  translate called 1
!
voice-card 0/4
```



```
no watchdog
!
license udi pid ISR4331/K9 sn FDO21381GMV
diagnostic bootup level minimal
spanning-tree extend system-id
!
username cisco privilege 15 password 7 0010160D320A11575F2F
!
redundancy
mode none
application redundancy
group 1
name coxcubecucm
priority 100 failover threshold 75
timers delay 30 reload 60
control GigabitEthernet0/0/2 protocol 1
data GigabitEthernet0/0/2
track 1 shutdown
track 2 shutdown
!
track 1 interface GigabitEthernet0/0/0 line-protocol
!
track 2 interface GigabitEthernet0/0/1 line-protocol
!
interface GigabitEthernet0/0/0
description Cox WAN
ip address 10.64.4.131 255.255.0.0
negotiation auto
redundancy rii 4
```



```
redundancy group 1 ip 10.64.4.170 exclusive
!
interface GigabitEthernet0/0/1
 ip address 10.80.10.11 255.255.255.0
 negotiation auto
 redundancy rii 3
 redundancy group 1 ip 10.80.10.10 exclusive
!
interface GigabitEthernet0/0/2
 ip address 1.1.1.1 255.255.255.252
 negotiation auto
!
interface Service-Engine0/4/0
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 negotiation auto
!
ip forward-protocol nd
ip http server
ip http authentication local
ip http secure-server
ip route 0.0.0.0 0.0.0.0 10.64.1.1
ip route 10.80.0.0 255.255.0.0 10.80.10.1
ip route 172.16.0.0 255.255.0.0 10.80.10.1
ip route 172.16.24.0 255.255.248.0 10.80.10.1
!
ip ssh server algorithm encryption aes128-ctr aes192-ctr aes256-ctr
```



```
ip ssh client algorithm encryption aes128-ctr aes192-ctr aes256-ctr
!
control-plane
!
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
!
dial-peer voice 510 voip
  description outgoing call to Cox - Lan Facing
  huntstop
  session protocol sipv2
  session transport udp
  incoming called-number [0-9]T
  voice-class codec 1
  voice-class sip asserted-id pai
  voice-class sip bind control source-interface GigabitEthernet0/0/1
  voice-class sip bind media source-interface GigabitEthernet0/0/1
  dtmf-relay rtp-nte
  no vad
!
dial-peer voice 520 voip
  description outgoing call to Cox - WAN Facing
  huntstop
  destination-pattern .T
  session protocol sipv2
```



```
session target sip-server
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
no voice-class sip outbound-proxy
voice-class sip profiles 101
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
no vad
!
dial-peer voice 610 voip
description incoming call to PBX-WAN facing
huntstop
session protocol sipv2
session transport udp
incoming called-number 402.....
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
no vad
!
dial-peer voice 620 voip
description incoming call to PBX-LAN facing
huntstop
```



```
destination-pattern 402.....
session protocol sipv2
session target ipv4:10.80.10.2
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
no voice-class sip outbound-proxy
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
no vad
!
dial-peer voice 700 voip
description PBX to PBX - LAN facing
huntstop
session protocol sipv2
session target sip-server
session transport udp
incoming called-number 1402.....
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
no vad
!
dial-peer voice 710 voip
description PBX to PBX - WAN facing
```



```
translation-profile outgoing Cox
huntstop
destination-pattern 142.....
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
no vad
!
sip-ua
keepalive target ipv4:10.64.4.165
retry invite 3
timers keepalive active 10
sip-server ipv4:10.64.4.165
connection-reuse
!
line con 0
exec-timeout 0 0
password 7 01100F175804
login
transport input none
stopbits 1
line aux 0
stopbits 1
line vty 0 4
```




```
password 7 070C285F4D06
login local
transport input telnet
!
ntp server 34.208.249.133
!
end
```



Standby Cisco UBE:

CoxCertCUBE8#sh run

```
version 16.6
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname CoxCertCUBE8
!
boot-start-marker
boot-end-marker
!
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
enable secret 5 $1$qL.R$XtXGz/0yrmPBmB15S1b.1
!
no aaa new-model
no ip domain lookup
subscriber templating
```



```
!  
multilink bundle-name authenticated  
!  
crypto pki trustpoint TP-self-signed-3616943619  
  enrollment selfsigned  
  subject-name cn=IOS-Self-Signed-Certificate-3616943619  
  revocation-check none  
  rsakeypair TP-self-signed-3616943619  
!  
crypto pki certificate chain TP-self-signed-3616943619  
!  
voice service voip  
  no ip address trusted authenticate  
  address-hiding  
  mode border-element license capacity 20  
  allow-connections sip to sip  
  redundancy-group 1  
  fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none  
  sip  
    bind control source-interface GigabitEthernet0/0/0  
    rel1xx disable  
    session refresh  
    header-passing  
    asserted-id pai  
    privacy pstn  
    early-offer forced  
    midcall-signaling passthru  
    g729 annexb-all  
!
```



```
voice class codec 1
  codec preference 1 g711ulaw
  codec preference 2 g729r8
!
voice class codec 2
  codec preference 1 g729r8
  codec preference 2 g711ulaw
!
voice class codec 3
  codec preference 1 g729r8
!
voice class sip-profiles 100
!
voice class sip-profiles 101
  request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>"
  "<sip:402932\1@\2>"
!
voice translation-rule 1
  rule 1 /^.*\(\.....\)/ /\1/
!
voice translation-profile Cox
  translate called 1
!
voice-card 0/4
  no watchdog
!
license udi pid ISR4331/K9 sn FDO21381FEY
diagnostic bootup level minimal
spanning-tree extend system-id
!
```



```
username cisco privilege 15 password 7 0010160D320A11575F2F
!
redundancy
mode none
application redundancy
group 1
name coxcubecucm
priority 100 failover threshold 75
timers delay 30 reload 60
control GigabitEthernet0/0/2 protocol 1
data GigabitEthernet0/0/2
track 1 shutdown
track 2 shutdown
!
track 1 interface GigabitEthernet0/0/0 line-protocol
track 2 interface GigabitEthernet0/0/1 line-protocol
!
interface GigabitEthernet0/0/0
description Cox WAN
ip address 10.64.4.132 255.255.0.0
negotiation auto
redundancy rii 4
redundancy group 1 ip 10.64.4.170 exclusive
!
interface GigabitEthernet0/0/1
description Cox LAN
ip address 10.80.10.12 255.255.255.0
negotiation auto
redundancy rii 3
```



```
redundancy group 1 ip 10.80.10.10 exclusive
!
interface GigabitEthernet0/0/2
 ip address 1.1.1.2 255.255.255.252
 negotiation auto
!
interface Service-Engine0/4/0
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 negotiation auto
!
ip forward-protocol nd
ip http server
ip http authentication local
ip http secure-server
ip http client source-interface GigabitEthernet0/0/0
ip route 0.0.0.0 0.0.0.0 10.64.1.1
ip route 10.64.0.0 255.255.0.0 10.64.1.1
ip route 10.80.0.0 255.255.0.0 10.80.10.1
ip route 172.16.24.0 255.255.248.0 10.80.10.1
!
ip ssh server algorithm encryption aes128-ctr aes192-ctr aes256-ctr
ip ssh client algorithm encryption aes128-ctr aes192-ctr aes256-ctr
!
control-plane
!
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
!
dial-peer voice 510 voip
  description outgoing call to Cox - Lan Facing
  huntstop
  session protocol sipv2
  session transport udp
  incoming called-number [0-9]T
  voice-class codec 1
  voice-class sip asserted-id pai
  voice-class sip bind control source-interface GigabitEthernet0/0/1
  voice-class sip bind media source-interface GigabitEthernet0/0/1
  dtmf-relay rtp-nte
  no vad
!
dial-peer voice 520 voip
  description outgoing call to Cox - WAN Facing
  huntstop
  destination-pattern .T
  session protocol sipv2
  session target sip-server
  session transport udp
  voice-class codec 1
  voice-class sip asserted-id pai
  no voice-class sip outbound-proxy
```



```
voice-class sip profiles 101
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
no vad
!
dial-peer voice 610 voip
description incoming call to PBX-WAN facing
huntstop
session protocol sipv2
session transport udp
incoming called-number 402.....
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
no vad
!
dial-peer voice 620 voip
description incoming call to PBX-LAN facing
huntstop
destination-pattern 402.....
session protocol sipv2
session target ipv4:10.80.10.2:5060
session transport udp
voice-class codec 1
```




```
voice-class sip asserted-id pai
no voice-class sip outbound-proxy
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
no vad
!
dial-peer voice 700 voip
description PBX to PBX - LAN facing
huntstop
session protocol sipv2
session target sip-server
session transport udp
incoming called-number 1402.....
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
no vad
!
dial-peer voice 710 voip
description PBX to PBX - WAN facing
translation-profile outgoing Cox
huntstop
destination-pattern 1402.....
session protocol sipv2
session target sip-server
```



```
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
no vad
!
sip-ua
keepalive target ipv4:10.64.4.165
retry invite 3
timers keepalive active 10
sip-server ipv4:10.64.4.165
connection-reuse
!
line con 0
exec-timeout 0 0
password 7 00071A150754
login
transport input none
stopbits 1
line aux 0
stopbits 1
line vty 0 4
password 7 104D000A0618
login local
transport input telnet
!
ntp server 34.202.215.187
```



```
ntp server pool.ntp.org
```

```
!
```

```
end
```



Configuring Cisco UCM 11.5 Cluster

Cisco UCM Version

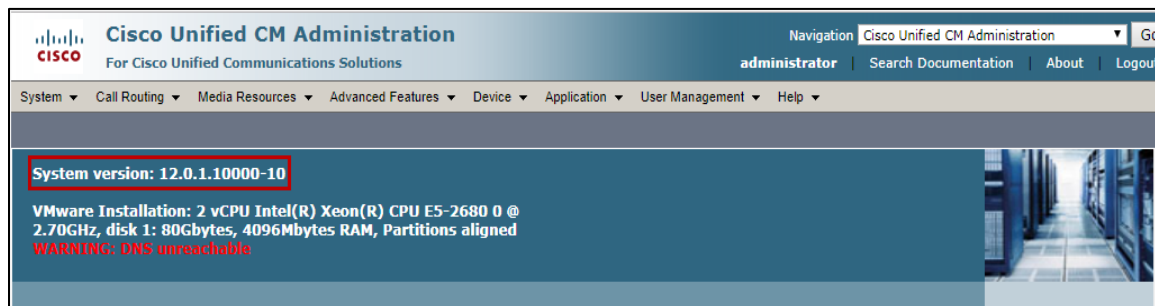


Figure 3: Cisco UCM Version

Cisco Call Manager Service Parameters

Navigation: System → Service Parameters

- Select Server* = **Clus20pub--CUCM Voice/Video (Active)**
- Select Service* = **Cisco CallManager (Active)**
- Duplex Streaming Enabled* = **True**
- All other fields are set to default values

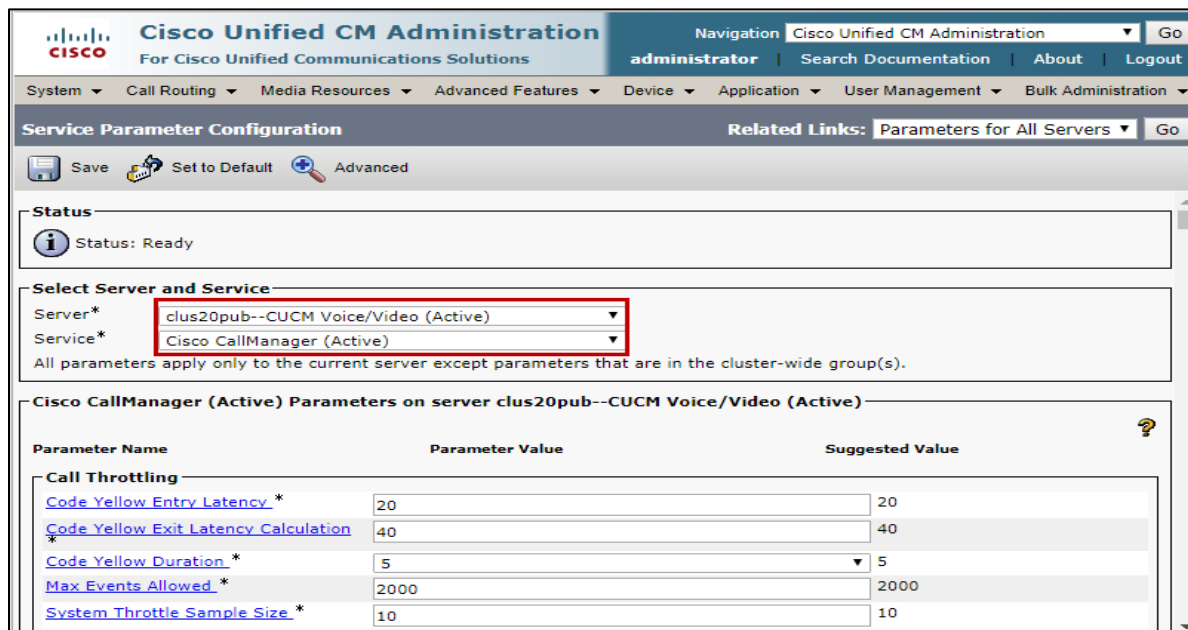


Figure 4: Service Parameters



Clusterwide Parameters (Service)		
Default Network Hold MOH Audio Source ID *	1	1
Default User Hold MOH Audio Source ID *	1	1
Duplex Streaming Enabled *	True	False
Media Exchange Interface Capability Timer *	8	8
Send Multicast MOH in H.245 OLC Message *	True	True
Media Exchange Timer *	12	12
Media Exchange Stop Streaming Timer *	8	8
Open Video Channel Response Timer for SIP Interop *	500	500
Port Received Timer After Call Connection *	500	500
Media Resource Allocation Timer *	12	12
MTP and Transcoder Resource Throttling Percentage *	95	95
Intercluster Capabilities Mismatch Timer *	1000	1000
Silence Suppression *	True	False
Silence Suppression for Gateways *	True	False
Strip G.729 Annex B (Silence Suppression) from Capabilities *	True	False
Enable Source IP Address Verification for Software Media Devices *	True	True

Figure 5: Service Parameters (Cont.)



SIP Trunk Security Profile

Navigation: System → Security → SIP Trunk Security Profile

- Name* = **Non Secure SIP Trunk Profile-Cox-cube-cucm** is used as an example
- Description = **Non Secure SIP Trunk Profile authenticated by null String** is used as an example
- Device Security Mode = **Non Secure**
- Incoming Transport Type* = **TCP + UDP**
- Outgoing Transport Type = **UDP**
- **Accept unsolicited Notification** is enabled

SIP Trunk Security Profile Configuration

Save Delete Copy Reset Apply Config Add New

SIP Trunk Security Profile Information

Name* Non Secure SIP Trunk Profile-Cox-cube-cucm

Description Non Secure SIP Trunk Profile authenticated by null String

Device Security Mode Non Secure

Incoming Transport Type* TCP+UDP

Outgoing Transport Type UDP

☐ Enable Digest Authentication

Nonce Validity Time (mins)* 600

X.509 Subject Name

Incoming Port* 5060

☐ Enable Application level authorization

☐ Accept presence subscription

☒ Accept out-of-dialog refer**

☒ Accept unsolicited notification

☒ Accept replaces header

☐ Transmit security status

☐ Allow charging header

SIP V.150 Outbound SDP Offer Filtering* Use Default Filter

Figure 6: SIP Trunk Security Profile



SIP Profile

Navigation: Device → Device Settings → SIP Profile

- Name* = **Standard SIP Profile_CUBE_CUCM_COX** is used as an example
- Description = **SIP Profile for cox-cucm-cube-certification** is used as an example

SIP Profile Configuration

Save Delete Copy Reset Apply Config Add New

All SIP devices using this profile must be restarted before any changes will take affect.

SIP Profile Information

Name*

Standard SIP Profile_CUBE_CUCM_COX

Description

SIP Profile for cox-cucm-cube-certification

Default MTP Telephony Event Payload Type*

101

Early Offer for G.Clear Calls*

Disabled

User-Agent and Server header information*

Send Unified CM Version Information as User-Agent

Version in User Agent and Server Header*

Major And Minor

Dial String Interpretation*

Phone number consists of characters 0-9, *, #, and

Confidential Access Level Headers*

Disabled

☐ Redirect by Application

☐ Disable Early Media on 180

☐ Outgoing T.38 INVITE include audio mline

☐ Offer valid IP and Send/Receive mode only for T.38 Fax Relay

☐ Use Fully Qualified Domain Name in SIP Requests

☐ Assured Services SIP conformance

☐ Enable External QoS**

SDP Information

SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*

TIAS and AS

SDP Transparency Profile

Pass all unknown SDP attributes

Accept Audio Codec Preferences in Received Offer*

Default

☐ Require SDP Inactive Exchange for Mid-Call Media Change

☐ Allow RR/RS bandwidth modifier (RFC 3556)

Figure 7: SIP Profile



Parameters used in Phone	
Timer Invite Expires (seconds)*	180
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10
Media Port Ranges	<input checked="" type="radio"/> Common Port Range for Audio and Video <input type="radio"/> Separate Port Ranges for Audio and Video
Start Media Port*	16384
Stop Media Port*	32766
DSCP for Audio Calls	Use System Default
DSCP for Video Calls	Use System Default
DSCP for Audio Portion of Video Calls	Use System Default
DSCP for TelePresence Calls	Use System Default
DSCP for Audio Portion of TelePresence Calls	Use System Default
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off
Caller ID Blocking*	Off
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
Resource Priority Namespace	< None >
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70
Off Hook To First Digit Timer (milliseconds)*	15000
Call Forward URI*	x-cisco-serviceuri-cfwdall
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial
<input checked="" type="checkbox"/> Conference Join Enabled <input type="checkbox"/> RFC 2543 Hold <input checked="" type="checkbox"/> Semi Attended Transfer <input type="checkbox"/> Enable VAD <input type="checkbox"/> Stutter Message Waiting <input type="checkbox"/> MLPP User Authorization	

Figure 8: SIP Profile (Cont.)

- “SIP Rel1XXX Options” is set to “Send PRACK if 1xx Contains SDP”

SIP Profile Configuration
Related Links

Save Delete Copy Reset Apply Config Add New

Normalization Script

Normalization Script < None >

☐ Enable Trace

	Parameter Name	Parameter Value
1		

Incoming Requests FROM URI Settings

Caller ID DN

Caller Name

Trunk Specific Configuration

Reroute Incoming Request to new Trunk based on* Never

Resource Priority Namespace List < None >

SIP Rel1XX Options* Send PRACK if 1xx Contains SDP

Video Call Traffic Class* Mixed

Calling Line Identification Presentation* Default

Session Refresh Method* Update

Early Offer support for voice and video calls* Disabled (Default value)

☐ Enable ANAT

☐ Deliver Conference Bridge Identifier

☐ Allow Passthrough of Configured Line Device Caller Information

☐ Reject Anonymous Incoming Calls

☐ Reject Anonymous Outgoing Calls

☐ Send ILS Learned Destination Route String

☐ Connect Inbound Call before Playing Queuing Announcement

SIP OPTIONS Ping

☒ Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"

Ping Interval for In-service and Partially In-service Trunks (seconds)* 60

Ping Interval for Out-of-service Trunks (seconds)* 120

Ping Retry Timer (milliseconds)* 500

Ping Retry Count* 6

SDP Information

☐ Send send-receive SDP in mid-call INVITE

☐ Allow Presentation Sharing using BFCP

☐ Allow iX Application Media

☒ Allow multiple codecs in answer SDP

Save
Delete
Copy
Reset
Apply Config
Add New

Figure 9: SIP Profile (Cont.)



Trunk configuration

Trunk configuration from Cisco UCM to the LAN side of the Cisco UBE:

Navigation: Device → Trunk → Add New

The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes the Cisco logo, the title 'Cisco Unified CM Administration For Cisco Unified Communications Solutions', and a navigation dropdown menu set to 'Cisco Unified CM Administration'. Below the navigation bar is a menu with options: System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled 'Find and List Trunks' and contains a search bar with the text 'Find Trunks where Device Name begins with'. Below the search bar is a message: 'No active query. Please enter your search criteria using the options above.' The 'Add New' button is highlighted with a red box.

Figure 10: Add New Trunk to Cisco UBE

- Select 'Trunk Type' as **SIP Trunk** and 'Device Protocol' as **SIP** and select '**Next**' as shown below.

The screenshot shows the Cisco Unified CM Administration interface for 'Trunk Configuration'. The top navigation bar is the same as in Figure 10. The main content area is titled 'Trunk Configuration' and includes a 'Next' button with a green arrow. Below the 'Next' button is a 'Status' section showing 'Status: Ready'. The 'Trunk Information' section contains three dropdown menus: 'Trunk Type*' (set to 'SIP Trunk'), 'Device Protocol*' (set to 'SIP'), and 'Trunk Service Type*' (set to 'None(Default)'). The 'Next' button is highlighted with a red box. A legend at the bottom indicates that '*' indicates a required item.

Figure 11: Add SIP Trunk Type

- "Media Resource Group List" is set to **MRGL_NoMTP** for resources with ANN, CFB, MOH



Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation **Cisco**
administrator | Search

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾

Trunk Configuration

Save Delete Reset Add New

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	Cox-Cube-CUCM
Description	Cox-Cube-CUCM
Device Pool*	Default ▾
Common Device Configuration	< None > ▾
Call Classification*	Use System Default ▾
Media Resource Group List	MRGL_NoMTP ▾
Location*	Hub_None ▾
AAR Group	< None > ▾
Tunneled Protocol*	None ▾
QSIG Variant*	No Changes ▾
ASN.1 ROSE OID Encoding*	No Changes ▾
Packet Capture Mode*	None ▾
Packet Capture Duration	0
<input type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Path Replacement Support	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Transmit UTF-8 Names in QSIG APDU	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security.	

Figure 12: SIP Trunk to Cisco UBE



- “Significant Digits” set to 4. 4 digits Extension for all CPE phones.

The screenshot displays the 'Trunk Configuration' page in the Cisco UBE interface. The top navigation bar includes tabs for System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, and Bu. The main configuration area is divided into several sections:

- Trunk Configuration:** Includes buttons for Save, Delete, Reset, and Add New. It contains three dropdown menus: 'Consider Traffic on This Trunk Secure*' (set to 'When using both sRTP and TLS'), 'Route Class Signaling Enabled*' (set to 'Default'), and 'Use Trusted Relay Point*' (set to 'Default'). A checkbox for 'PSTN Access' is checked and highlighted with a red box. Below it is an unchecked checkbox for 'Run On All Active Unified CM Nodes'.
- Intercompany Media Engine (IME):** Contains a dropdown for 'E.164 Transformation Profile' set to '< None >'.
- MLPP and Confidential Access Level Information:** Contains three dropdowns: 'MLPP Domain' (set to '< None >'), 'Confidential Access Mode' (set to '< None >'), and 'Confidential Access Level' (set to '< None >').
- Call Routing Information:** Contains four checkboxes: 'Remote-Party-Id' (checked), 'Asserted-Identity' (checked), 'Asserted-Type*' (set to 'Default'), 'SIP Privacy*' (set to 'Default'), and 'Trust Received Identity*' (set to 'Trust All (Default)').
- Inbound Calls:** Contains three dropdowns: 'Significant Digits*' (set to 4 and highlighted with a red box), 'Connected Line ID Presentation*' (set to 'Default'), and 'Connected Name Presentation*' (set to 'Default').

Figure 13: SIP Trunk to Cisco UBE (Cont.)

- Configure the Virtual LAN IP address of the Cisco UBE and the Destination Port
- Configure the SIP Trunk Security Profile and SIP Profile as shown below
- **Redirecting Diversion Header Delivery-Outbound** is enabled to send the Diversion information during outbound calls
- The rest of the configuration is all default
- Click Save and Reset after completion



System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Trunk Configuration

Related Links: [Back To Find/List](#) ▾

Save Delete Reset Add New

Calling Search Space
AAR Calling Search Space
Prefix DN
☐ Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

[Clear Prefix Settings](#) [Default Prefix Settings](#)

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	<input style="width: 100px;" type="text" value=" Default "/>	<input style="width: 50px;" type="text" value=" 0 "/>	<input style="width: 150px;" type="text" value=" < None > "/>	<input checked="" type="checkbox"/>

Incoming Called Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

[Clear Prefix Settings](#) [Default Prefix Settings](#)

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	<input style="width: 100px;" type="text" value=" Default "/>	<input style="width: 50px;" type="text" value=" 0 "/>	<input style="width: 150px;" type="text" value=" < None > "/>	<input checked="" type="checkbox"/>

Connected Party Settings

Connected Party Transformation CSS
☒ Use Device Pool Connected Party Transformation CSS

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Trunk Configuration

Related Links: [Back To Find/List](#) ▾

Save Delete Reset Add New

Outbound Calls

Called Party Transformation CSS
☒ Use Device Pool Called Party Transformation CSS
Calling Party Transformation CSS
☒ Use Device Pool Calling Party Transformation CSS
Calling Party Selection*
Calling Line ID Presentation*
Calling Name Presentation*
Calling and Connected Party Info Format*
☒ Redirecting Diversion Header Delivery - Outbound
Redirecting Party Transformation CSS
☒ Use Device Pool Redirecting Party Transformation CSS

Caller Information

Caller ID DN
Caller Name
☐ Maintain Original Caller ID DN and Caller Name in Identity Headers

SIP Information

Destination

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1 *	<input style="width: 150px;" type="text" value=" 10.80.10.10 "/>	<input style="width: 150px;" type="text"/>	<input style="width: 50px;" type="text" value=" 5060 "/>

Figure 14: SIP Trunk to Cisco UBE (Cont.)



System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help

Trunk Configuration Related Links: [Back](#)

Save Delete Reset Add New

MTP Preferred Originating Codec*	711ulaw
BLF Presence Group*	Standard Presence group
SIP Trunk Security Profile*	Non Secure SIP Trunk Profile-Cox-cube-cucm
Rerouting Calling Search Space	< None >
Out-Of-Dialog Refer Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	Standard SIP Profile_CUBE_CUCM_COX View Details
DTMF Signaling Method*	No Preference

Normalization Script

Normalization Script < None >

☐ Enable Trace

	Parameter Name	Parameter Value	
1			

Recording Information

☒ None

☐ This trunk connects to a recording-enabled gateway

☐ This trunk connects to other clusters with recording-enabled gateways

Geolocation Configuration

Geolocation < None >

Geolocation Filter < None >

☐ Send Geolocation Information

Figure 15: SIP Trunk to Cisco UBE (Cont.)



Trunk configuration from Cisco UCM to Fax Gateway:

Navigation: Devices → Trunk → Add New

The screenshot shows the 'Find and List Trunks' page in the Cisco Unified CM Administration interface. The page has a header with the Cisco logo and navigation links. Below the header, there are tabs for 'System', 'Call Routing', 'Media Resources', 'Advanced Features', 'Device', 'Application', 'User Management', 'Bulk Administration', and 'Help'. The 'Device' tab is selected. The main content area is titled 'Find and List Trunks' and contains a search bar with a dropdown menu for 'Device Name' and a 'Find' button. Below the search bar, there is a message: 'No active query. Please enter your search criteria using the options above.' At the bottom of the page, the 'Add New' button is highlighted with a red box.

Figure 16: Add New Trunk to Fax Gateway




- Select 'Trunk Type' as **SIP Trunk** and 'Device Protocol' as **SIP** and select '**Next**' as shown below.

The screenshot shows the 'Trunk Configuration' page in the Cisco Unified CM Administration interface. The page has a header with the Cisco logo and navigation links. Below the header, there are tabs for 'System', 'Call Routing', 'Media Resources', 'Advanced Features', 'Device', 'Application', 'User Management', 'Bulk Administration', and 'Help'. The 'Device' tab is selected. The main content area is titled 'Trunk Configuration' and contains a 'Next' button. Below the 'Next' button, there is a 'Status' section with a message: 'Status: Ready'. Below the 'Status' section, there is a 'Trunk Information' section with three dropdown menus: 'Trunk Type*' (set to 'SIP Trunk'), 'Device Protocol*' (set to 'SIP'), and 'Trunk Service Type*' (set to 'None(Default)'). At the bottom of the page, the 'Next' button is highlighted with a red box.

Figure 17: Add SIP Trunk Type

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administ

Trunk Configuration Relate

Save  Delete  Reset  Add New

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	FaxGateway
Description	FaxGateway
Device Pool*	Default ▾
Common Device Configuration	< None > ▾
Call Classification*	Use System Default ▾
Media Resource Group List	MRGL_NoMTP ▾
Location*	Hub_None ▾
AAR Group	< None > ▾
Tunneled Protocol*	None ▾
QSIG Variant*	No Changes ▾
ASN.1 ROSE OID Encoding*	No Changes ▾
Packet Capture Mode*	None ▾
Packet Capture Duration	0

☐ Media Termination Point Required
☒ Retry Video Call as Audio
☐ Path Replacement Support
☐ Transmit UTF-8 for Calling Party Name
☐ Transmit UTF-8 Names in QSIG APDU
☐ Unattended Port

☐ SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security.. other information.
 Consider Traffic on This Trunk Secure* When using both sRTP and TLS ▾
 Route Class Signaling Enabled* Default ▾
 Use Trusted Relay Point* Default ▾
☒ PSTN Access
☐ Run On All Active Unified CM Nodes

Figure 18: SIP Trunk to FAX Gateway



System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Manag

Trunk Configuration

Save Delete Reset Add New

Intercompany Media Engine (IME)

E.164 Transformation Profile < None >

MLPP and Confidential Access Level Information

MLPP Domain < None >
Confidential Access Mode < None >
Confidential Access Level < None >

Call Routing Information

☒ Remote-Party-Id
☒ Asserted-Identity
Asserted-Type* Default
SIP Privacy* Default
Trust Received Identity* Trust All (Default)

Inbound Calls

Significant Digits* All
Connected Line ID Presentation* Default
Connected Name Presentation* Default
Calling Search Space < None >
AAR Calling Search Space < None >
Prefix DN
☐ Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

Incoming Called Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

Connected Party Settings

Connected Party Transformation CSS < None >
☒ Use Device Pool Connected Party Transformation CSS

Figure 19: SIP Trunk to FAX Gateway (Cont.)

- Configure the IP address of Fax Gateway and the Destination Port

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- Configure the SIP Trunk Security Profile and SIP Profile as shown below
- The rest of the configuration is all default
- Click Save and Reset after completion

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Trunk Configuration

Related Links: [Back To Find/List](#) ▾

Save Delete Reset Add New

Outbound Calls

Called Party Transformation CSS ▾ < None >

☒ Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS ▾ < None >

☒ Use Device Pool Calling Party Transformation CSS

Calling Party Selection* ▾ Originator ▾

Calling Line ID Presentation* ▾ Default ▾

Calling Name Presentation* ▾ Default ▾

Calling and Connected Party Info Format* ▾ Deliver DN only in connected party ▾

☐ Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS ▾ < None >

☒ Use Device Pool Redirecting Party Transformation CSS

Caller Information

Caller ID DN

Caller Name

☐ Maintain Original Caller ID DN and Caller Name in Identity Headers

SIP Information

Destination

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1 *	<input type="text" value="172.16.31.50"/>	<input type="text"/>	<input type="text" value="5060"/>

Other Settings

MTP Preferred Originating Codec* ▾ 711ulaw ▾

BLF Presence Group* ▾ Standard Presence group ▾

SIP Trunk Security Profile* ▾ **Non Secure SIP Trunk Profile-Cox-cube-cucm** ▾

Rerouting Calling Search Space ▾ < None > ▾

Out-Of-Dialog Refer Calling Search Space ▾ < None > ▾

SUBSCRIBE Calling Search Space ▾ < None > ▾

SIP Profile* ▾ **Standard SIP Profile_CUBE_CUCM_COX** ▾ [View Details](#)

DTMF Signaling Method* ▾ No Preference ▾

Figure 20: SIP Trunk to FAX Gateway (Cont.)



Normalization Script					
Normalization Script < None >					
<input type="checkbox"/> Enable Trace					
1	<table border="1"><thead><tr><th>Parameter Name</th><th>Parameter Value</th></tr></thead><tbody><tr><td></td><td></td></tr></tbody></table>	Parameter Name	Parameter Value		
Parameter Name	Parameter Value				
Recording Information					
<input checked="" type="radio"/> None					
<input type="radio"/> This trunk connects to a recording-enabled gateway					
<input type="radio"/> This trunk connects to other clusters with recording-enabled gateways					
Geolocation Configuration					
Geolocation < None >					
Geolocation Filter < None >					
<input type="checkbox"/> Send Geolocation Information					
<div>Save Delete Reset Add New</div>					

Figure 21: SIP Trunk to FAX Gateway (Cont.)



Routing configuration

Route Pattern for Cisco UBE:

Navigation: Call Routing → Route/Hunt → Route Pattern → Add New. Route patterns are configured as below.

- Cisco IP phone dial “9”+10 digits number to access PSTN via Cisco UBE.”9” is removed before sending to Cisco UBE.
- For FAX call, Access Code “9”+10 digits number is used at Cisco Fax gateway. “9” is removed at Cisco UCM. The rest of the number is sent to Cisco UBE to COX.
- Incoming fax call to 46XX will be sent to Cisco Fax gateway.
- For Anonymous call, access code “8*”+10 digits number is used. “8*” is removed at Cisco UCM. The rest of the number is sent to Cisco UBE to COX.
- Cisco IP phones dial “9” + X11 for emergency call and will send all digits to Cisco UBE to COX.

The screenshot shows the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title 'Cisco Unified CM Administration For Cisco Unified Communications Solutions', and a navigation dropdown menu. Below the navigation bar, there is a breadcrumb trail: System > Call Routing > Media Resources > Advanced Features > Device > Application > User Management > Bulk Administration > Help. The main content area is titled 'Find and List Route Patterns'. It features a '+ Add New' button. Below this, there is a section for 'Route Patterns' with a search bar. The search bar has a dropdown menu set to 'Pattern' and a radio button set to 'begins with'. There are 'Find', 'Clear Filter', and a plus-minus icon. Below the search bar, a message states: 'No active query. Please enter your search criteria using the options above.' At the bottom of the section, there is a red-bordered 'Add New' button.

Figure 22: Add New Route Pattern for Cisco UBE

- Route Pattern: Specify appropriate **Route pattern**.
- Gateway/Route List: Select **Cox-Cube-CUCM**
- **Use Calling Party's External Phone Number Mask** is enabled to send the 10-Digit Calling Number as DID when the call is routed via this route.

System
Call Routing
Media Resources
Advanced Features
Device
Application
User Management
Bulk A

Route Pattern Configuration

Save
Delete
Copy
Add New

Pattern Definition

Route Pattern* 9.@
Route Partition < None >
Description
Numbering Plan* NANP
Route Filter < None >
MLPP Precedence* Default
☐ Apply Call Blocking Percentage
Resource Priority Namespace Network Domain < None >
Route Class* Default
Gateway/Route List* Cox-Cube-CUCM (Edit)
Route Option
☒ Route this pattern
☐ Block this pattern No Error
Call Classification* OffNet
External Call Control Profile < None >
☐ Allow Device Override
☒ Provide Outside Dial Tone
☐ Allow Overlap Sending
☐ Urgent Priority
☐ Require Forced Authorization Code
Authorization Level* 0
☐ Require Client Matter Code

Calling Party Transformations

☒ Use Calling Party's External Phone Number Mask
Calling Party Transform Mask

Prefix Digits (Outgoing Calls)
Calling Line ID Presentation* Default
Calling Name Presentation* Default
Calling Party Number Type* Cisco CallManager
Calling Party Numbering Plan* Cisco CallManager

Figure 23: Route Pattern Configuration for Cisco UBE-PSTN Access



Connected Party Transformations

Connected Line ID Presentation* Default

Connected Name Presentation* Default

Called Party Transformations

Discard Digits PreDot

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type* Cisco CallManager

Called Party Numbering Plan* Cisco CallManager

ISDN Network-Specific Facilities Information Element

Network Service Protocol -- Not Selected --

Carrier Identification Code

Network Service	Service Parameter Name	Service Parameter Value
-- Not Selected --	< Not Exist >	

Figure 24: Route Pattern Configuration for Cisco UBE-PSTN Access (Cont.)

Route Pattern Configuration

Save Delete Copy Add New

Status

Status: Ready

Pattern Definition

Route Pattern* 8*.*

Route Partition < None >

Description

Numbering Plan* NANP

Route Filter < None >

MLPP Precedence* Default

☐ Apply Call Blocking Percentage

Resource Priority Namespace Network Domain < None >

Route Class* Default

Gateway/Route List* Cox-Cube-CUCM (Edit)

Route Option

☒ Route this pattern

☐ Block this pattern No Error

Call Classification* OffNet

External Call Control Profile < None >

☐ Allow Device Override ☒ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority

☐ Require Forced Authorization Code

Authorization Level* 0

☐ Require Client Matter Code

Figure 25 Route Pattern Configuration for Cisco UBE-PSTN Access National- Anonymous Call



Calling Party Transformations

☒ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask:

Prefix Digits (Outgoing Calls):

Calling Line ID Presentation*: **Restricted**

Calling Name Presentation*: **Restricted**

Calling Party Number Type*: Cisco CallManager

Calling Party Numbering Plan*: Cisco CallManager

Connected Party Transformations

Connected Line ID Presentation*: Default

Connected Name Presentation*: Default

Called Party Transformations

Discard Digits: **PreDot**

Called Party Transform Mask:

Prefix Digits (Outgoing Calls):

Called Party Number Type*: Cisco CallManager

Called Party Numbering Plan*: Cisco CallManager

ISDN Network-Specific Facilities Information Element

Network Service Protocol: -- Not Selected --

Carrier Identification Code:

Network Service	Service Parameter Name	Service Parameter Value
-- Not Selected --	< Not Exist >	

Figure 26: Route Pattern Configuration for Cisco UBE-PSTN Access National- Anonymous Call (Cont.)

Route Pattern for Fax Gateway:

- Navigation: Call Routing → Route/Hunt → Route Pattern → Add New

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go

administrator | Search Documentation | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Find and List Route Patterns

+ Add New

Route Patterns

Find Route Patterns where Pattern ▾ begins with ▾ Find Clear Filter + -

No active query. Please enter your search criteria using the options above.

Add New

Figure 27: Add New Route patter to Fax Gateway



Route Pattern Configuration

Save Delete Copy Add New

Status
Status: Ready

Pattern Definition

Route Pattern *

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence *

☐ Apply Call Blocking Percentage

Resource Priority Namespace Network Domain

Route Class *

Gateway/Route List * [\(Edit\)](#)

Route Option
☒ Route this pattern
☐ Block this pattern

Call Classification *

External Call Control Profile

☐ Allow Device Override ☒ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority

☐ Require Forced Authorization Code

Authorization Level *

☐ Require Client Matter Code

Figure 28: Route Pattern Configuration for Fax Gateway

Calling Party Transformations

☐ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation *

Calling Name Presentation *

Calling Party Number Type *

Calling Party Numbering Plan *

Connected Party Transformations

Connected Line ID Presentation *

Connected Name Presentation *

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type *

Called Party Numbering Plan *

ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

Network Service	Service Parameter Name	Service Parameter Value
<input style="width: 100px;" type="text" value=" -- Not Selected -- "/>	<input style="width: 100px;" type="text" value=" < Not Exist > "/>	<input type="text" value=""/>

Figure 29: Route Pattern Configuration for Fax Gateway (Cont.)



Configuring Cisco Voice Gateway for Fax

The following configuration snippet contains a sample configuration of Cisco Voice gateway.

Global Settings

```
voice service voip
  allow-connections sip to sip
  no supplementary-service sip handle-replaces
  redirect ip2ip
  fax protocol pass-through g711ulaw
  no fax-relay sg3-to-g3
  sip
    rel1xx disable
    midcall-signaling passthru
    g729 annexb-all
  !
```

Codecs

G711ulaw and G729 voice codecs are used for this testing. Codec preferences used to change according to the test plan description.

```
voice class codec 3
  codec preference 1 g711ulaw
  !
voice class codec 2
  codec preference 1 g711alaw
  codec preference 2 g729br8
  codec preference 3 g729r8
  !
voice class codec 1
  codec preference 1 g729br8
  codec preference 2 g729r8
  codec preference 3 g711ulaw!
```



Dial peer

Outbound Dial-peer to Cisco UCM:

```
dial-peer voice 11 voip
description Gateway to CUCM for COXCUBECUCM
service session
destination-pattern 9T
session protocol sipv2
session target ipv4:10.80.10.2
session transport udp
voice-class codec 3
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
```

Inbound Dial-peer from Cisco UCM:

```
dial-peer voice 12 voip
description COXCUBECUCM Inbound Fax
service session
session protocol sipv2
session transport udp
incoming called-number 4962
voice-class codec 3
dtmf-relay rtp-nte
fax-relay ecm disable
fax protocol pass-through g711ulaw
no vad
```

POTS and Port Configuration:

Based on configured POTS destination pattern, gateway forwards the call to designated voice port.



```
dial-peer voice 10 pots
  service session
  destination-pattern 4029324962
  no digit-strip
  port 0/0/1
  forward-digits all
```

```
voice-port 0/0/1
  no echo-cancel enable
  no vad
  cptone IN
  station-id name CoxFAXDIDname
  station-id number 4029324962
  caller-id enable
!
```



Configuration example

The following configuration snippet contains a sample configuration of Cisco Voice Gateway with all parameters mentioned previously.

```
cme.in.tekvizion.com#sh run
```

```
version 15.1
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname cme.in.tekvizion.com
!
boot-start-marker
boot-end-marker
!
aaa new-model
!
aaa authentication login local_auth local
!
aaa session-id common
clock timezone IST 5 30
network-clock-participate wic 2
network-clock-participate wic 3
!
dot11 syslog
ip source-route
!
ip cef
!
isdn switch-type primary-qsig
!
voice rtp send-recv
```



```
!  
voice service pots  
!  
voice service voip  
    allow-connections sip to sip  
    no supplementary-service sip handle-replaces  
    redirect ip2ip  
    no fax-relay sg3-to-g3  
    sip  
        rel1xx disable  
        fax protocol pass-through g711ulaw  
        midcall-signaling passthru  
        g729 annexb-all  
!  
voice class codec 3  
    codec preference 1 g711ulaw  
!  
voice class codec 2  
    codec preference 1 g711alaw  
    codec preference 2 g729br8  
    codec preference 3 g729r8  
!  
voice class codec 1  
    codec preference 1 g729br8  
    codec preference 2 g729r8  
    codec preference 3 g711ulaw  
!  
voice class codec 4  
    codec preference 1 g711alaw  
    codec preference 2 g722-64  
!  
voice class codec 5  
    codec preference 1 g729r8
```



```
!  
voice-card 0  
    dspfarm  
    dsp services dspfarm  
!  
crypto pki token default removal timeout 0  
!  
license udi pid CISCO2851 sn FHK1137F4LY  
!  
interface GigabitEthernet0/0  
    ip address 172.16.31.50 255.255.255.0  
    duplex auto  
    speed auto  
!  
ip forward-protocol nd  
!  
ip http server  
no ip http secure-server  
!  
ip route 0.0.0.0 0.0.0.0 172.16.31.1  
ip route 0.0.0.0 0.0.0.0 172.16.29.1  
!  
ipv6 route ::/0 2620:96:c000:8::1  
!  
snmp-server community public RO  
snmp-server location Chennai  
!  
ipv6 access-list ipv6  
    permit ipv6 any any  
!  
control-plane  
!  
voice-port 0/0/0
```



```
no echo-cancel enable
no vad
cptone IN
station-id name COX user2
station-id number 26XX
caller-id enable
!
voice-port 0/0/1
no echo-cancel enable
no vad
cptone IN
station-id name CoxFAXDIDname
station-id number 402XXXXXXX
caller-id enable
!
voice-port 0/3/0:15
!
voice-port 0/2/0:15
!
voice-port 0/1/0
!
ccm-manager mgcp
no ccm-manager fax protocol cisco
ccm-manager music-on-hold
ccm-manager config server 172.16.26.2
ccm-manager config
!
mgcp
mgcp call-agent 172.16.26.2 2427 service-type mgcp version 0.1
mgcp rtp unreachable timeout 1000 action notify
mgcp modem passthrough voip mode nse
mgcp package-capability rtp-package
mgcp package-capability sst-package
```



```
mgcp package-capability pre-package
mgcp default-package mt-package
no mgcp package-capability fxr-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax t38 inhibit
mgcp rtp payload-type g726r16 static
!
mgcp profile default
!
sccp local GigabitEthernet0/0
sccp ccm 10.71.3.10 identifier 1 version 7.0
!
sccp ccm group 6
  bind interface GigabitEthernet0/0
  associate ccm 1 priority 1
!
dial-peer voice 10 pots
  service session
  destination-pattern 402XXXXXXX
  no digit-strip
  port 0/0/1
  forward-digits all
!
dial-peer voice 11 voip
  description Gateway to CUCM for COXCUBECUCM
  service session
  destination-pattern 1T
  session protocol sipv2
  session target ipv4:10.80.10.2
  session transport udp
  voice-class codec 3
  dtmf-relay rtp-nte
```




```
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 12 voip
description COXCUBECUCM Inbound Fax
service session
session protocol sipv2
session transport udp
incoming called-number 49XX
voice-class codec 3
dtmf-relay rtp-nte
fax-relay ecm disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 20 pots
service session
destination-pattern 26XX
no digit-strip
port 0/0/0
forward-digits all
!
sip-ua
no remote-party-id
retry register 5
no timers hold
protocol mode dual-stack
!
```



Configuring Cisco Unity Connection

Cisco Unity Connection Version

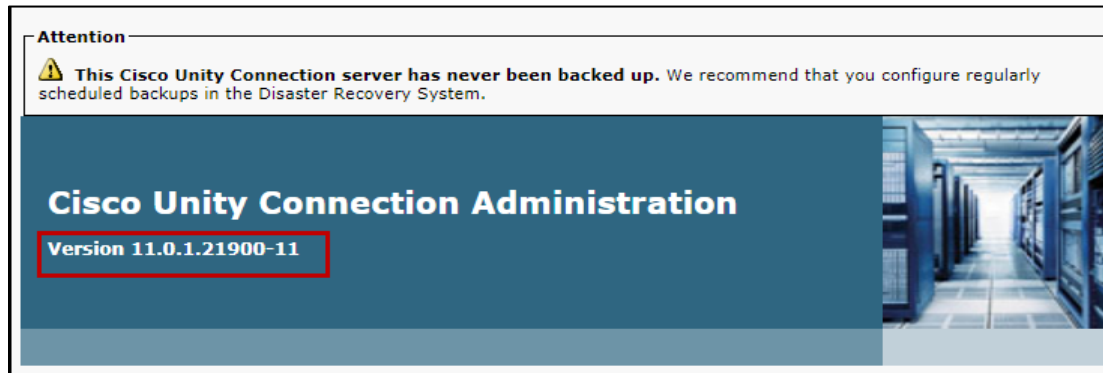


Figure 30: Version information

User Configuration

Navigation: Cisco Unity Connection → Users → Users

- Set Alias= **cox-cube-cucm** is used for this example
- Set First Name = **cox** is used to identify this User
- Set Extension = **4563** is used in this example
- Set Phone System = **cox-cube-cucm** is used in this example
- All other values are default

Figure 31: User Configuration

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Cisco Unity Connection

- Users
 - Users
 - 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
 - Branch Management
 - HTTP(S) Links
 - Locations

☐ 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

Location

Address

Building

City

State

Postal Code

Country

☒ Use System Default Time Zone

Time Zone

Language ☒ Use System Default Language

☐ English(United States)

Department

Manager

Billing ID

Corporate Email Address

☐ Generate SMTP Proxy Address From Corporate Email Address

Directory URI

Corporate Phone Number

Figure 32: User Configuration (cont.)



Telephony Integration

Navigation: Telephony Integrations → Phone system

- Set Phone System Name = **Cox-cube-cucm** is used for this example

Cisco Unity Connection

- System Settings
 - General Configuration
 - Cluster
 - Authentication Rules
 - Roles
 - Restriction Tables
 - Licenses
 - Schedules
 - 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 (CORS)
 - 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

Phone System

Phone System Name*

☐ 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

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

Guard Time 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:

Ending Time:

Figure 33: Telephone Integration – Phone System



Port Group

Navigation: Telephony Integration → Port Group

- Set Display Name = COX-cube-cucm-1 is used for this example

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"/>

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

Fields marked with an asterisk (*) are required.

Figure 34: Port Group



Navigation: Telephony Integration → Port Group → Edit → Servers

- Set IPV4 Address = **10.80.10.2** (Cisco UCM IP is used for this example)

Edit Servers

Search Port Groups ▶ Port Group Basics (Cox-cube-cucm-1) ▶ Edit Servers

Related Links [Check Telephony Configuration](#) ▼ [Go](#)

Port Group Edit Refresh Help

[Save](#)

SIP Servers

[Delete Selected](#) [Add](#)

<input type="checkbox"/>	Order	IPv4 Address or Host Name	IPv6 Address or Host Name	Port	TLS Port
<input type="checkbox"/>	0	10.80.10.2		5060	5061

[Delete Selected](#) [Add](#)

TFTP Servers

[Delete Selected](#) [Add](#)

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

[Delete Selected](#) [Add](#)

IPv6 Addressing Mode

Preference for Signaling [IPv4](#) ▼

Preference for Media [IPv4](#) ▼

[Save](#)

Figure 35: Port Group (cont.)



Port

Navigation: Telephony Integration → Port

- Set Port Name = **Cox-cube-cucm-1-001** is used for this example
- Set Phone System = **Cox-cube-cucm**
- Set Port Group = **Cox-cube-cucm-1**
- Set Server = **clus20unity.skypelabsj.local** is used for this example

Cisco Unity Connection

- System Settings
 - General Configuration
 - Cluster
 - Authentication Rules
 - Roles
 - Restriction Tables
 - Licenses
 - Schedules
 - 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 (CORS)
- SMTP Configuration
- Advanced
- Telephony Integrations
 - Phone System
 - Port Group
 - Port**
 - Speech Connect Port

Port Basics (Cox-cube-cucm-1-001)

Search Ports Related Links

Port Refresh Help

[Save](#) [Delete](#) [Previous](#) [Next](#)

Phone System Port

☒ Enabled

Port Name: Cox-cube-cucm-1-001 [Restart](#)

Phone System: Cox-cube-cucm

Port Group: Cox-cube-cucm-1

Server: clus20unity.skypelabsj.local ▼

Port Behavior

☒ Answer Calls

☒ Perform Message Notification

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

☒ Allow TRAP Connections

[Save](#) [Delete](#) [Previous](#) [Next](#)

Figure 36: Port



Acronyms

Acronym	Definitions
CPE	Customer Premise Equipment
Cisco UBE	Cisco Unified Border Element
Cisco UCM	Cisco Unified Communications Manager
MTP	Media Termination Point
POP	Point of Presence
PSTN	Public Switched Telephone Network
ESBC	Enterprise Session Border Controller
SCCP	Skinny Client Control Protocol
SIP	Session Initiation Protocol

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