

Connecting Cisco Unified Communication Manager [14.0.1] to Cox SIP Trunk via Cisco Unified Border Element v14.1 [IOS-XE 17.3.3]

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Introduction

Service Providers 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) ISR4331/K9 running IOS-XE 17.3.3 can be used. The Cisco Unified Border Element provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 14.0.1 connected to Cox IP 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 14.0.1, and Cisco UBE on ISR 4331/K9 [IOS-XE 17.3.3] for connectivity to Cox SIP Trunking service. The deployment model covered in this application note is CPE (Cisco UCM) to PSTN (Cox) via Cisco UBE v14.1 [IOS-XE 17.3.3].
- 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), semi-attended 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 with Cox SIP Trunking network.

Network Topology

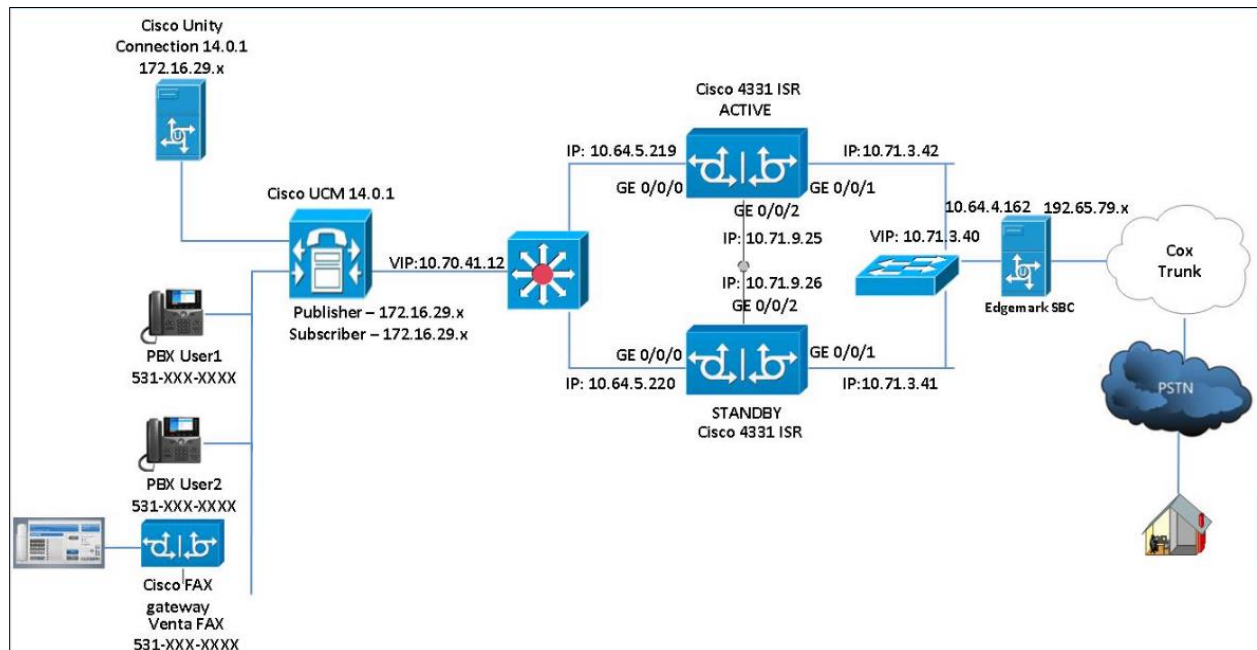


Figure 1 Network Topology

- The network topology includes the Cisco UCM Cluster, Cisco Unity Connection Voicemail system and Cisco Endpoints. Cisco UCM has a trunk configured to Cisco UBE's Virtual IP Address. Cox is used as the service provider with SIP trunk to the Cisco UBE using the WAN Virtual IP Address.
- 2 Cisco UBE's are used here for High Availability.
- SIP Trunk transport type used between Cisco UBE 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
- VMware ESXi v7.0.2
- Generic Cisco IP-Phones

Software Requirements

- Cisco UBE-Version: 14.1 running IOS-XE 17.3.3
- Cisco Unified Communications Manager release 14.0.1.10000-20
- Cisco Unity Connection release 14.0.1.10000-19

Features

Features Supported

- Incoming and outgoing national and international calls using G711alaw voice codecs
- Call Conference
- Fax
 - G711 A-Law Pass-through
 - T.38
- Voice Mail
- Auto Attendant
- Call Hold & Resume(MoH and ToH)
- Semi-attended and Attended Call transfer
- Call forward (all, busy and no answer)
- DTMF (RFC2833)
- IP-PBX Calling number privacy
- High Availability

Features Not Supported

- Cox supports only G711. G729 specific scenarios hence are not supported.
- Cisco UCM does not support Blind transfer feature.
- Cox does not support SG3. All the fax scenarios hence worked only on G3 speed.

Caveats

- Only one IP PBX is used for testing.
- Caller ID is not updated for attended and unattended transfer scenarios.
- For anonymous call between IP PBX phone numbers over Cox, Cox copies the caller number from the PAI header into the From header and sends it while looping back. This make the target IP PBX phone to display the actual caller ID instead of concealing the identity of the originator. Cox also discards Privacy ID parameter in the INVITE while looping back.
- For a call between IP PBX phone numbers over Cox, Cox changes the calling number in the From header from 10 digits DID to 4 digits number (ignoring the first 6 digits of the DID) and also modifies the display-name of the From and To headers as "Unassigned Unassigned".

Configuration

Configuring 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.

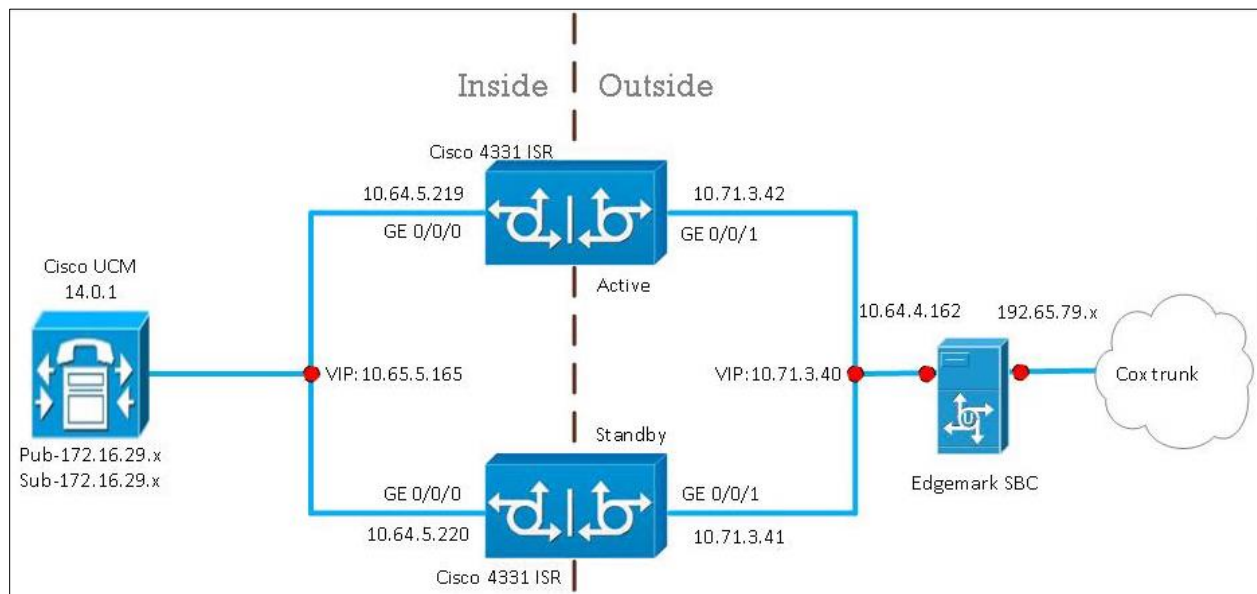


Figure 2 High Availability topology

Cisco UBE -Active:

```
interface GigabitEthernet0/0/0
  description CUBE LAN INTERFACE
  ip address 10.64.5.219 255.255.0.0
  negotiation auto
  redundancy rii 10
  redundancy group 1 ip 10.64.5.165 exclusive
!
interface GigabitEthernet0/0/1
  description CUBE WAN INTERFACE
  ip address 10.71.3.x 255.255.255.0
  negotiation auto
  redundancy rii 20
  redundancy group 1 ip 10.71.3.x exclusive
!
interface GigabitEthernet0/0/2
  description HA interface
  ip address 10.71.9.25 255.255.255.0
  negotiation auto
!
```

Cisco UBE -Standby:

```
interface GigabitEthernet0/0/0
  description CUBE LAN INTERFACE
  ip address 10.64.5.220 255.255.0.0
  negotiation auto
  no mop enabled
  redundancy rii 10
  redundancy group 1 ip 10.64.5.165 exclusive
!
interface GigabitEthernet0/0/1
  description CUBE WAN INTERFACE
  ip dhcp client client-id ascii cisco-ec1d.8bd2.9751-Gi0/0/1
  ip address 10.71.3.x 255.255.255.0
  negotiation auto
```

```
no mop enabled
redundancy rii 20
redundancy group 1 ip 10.71.3.x exclusive
!
interface GigabitEthernet0/0/2
description HA interface
ip address 10.71.9.26 255.255.255.0
negotiation auto
no mop enabled
!
```

Global Cisco UBE settings

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

```
voice service voip
  ip address trusted list
    ipv4 172.16.29.x
    ipv4 10.64.5.0 255.255.255.0
    ipv4 172.17.8.0 255.255.255.0
  no ip address trusted authenticate
  address-hiding
  mode border-element
  media bulk-stats
  allow-connections sip to sip
  redundancy-group 1
  no supplementary-service sip refer
  no supplementary-service sip handle-replaces
  supplementary-service media-renegotiate
  fax protocol t38 version 3 ls-redundancy 0 hs-redundancy 0 fallback none
  trace
  sip
    session refresh
    header-passing
    error-passthru
    asserted-id pai
    privacy pstn
    conn-reuse
    early-offer forced
    midcall-signaling passthru
    g729 annexb-all
    sip-profiles inbound
  no call service stop
```

Explanation

Command	Description
allow-connections sip to sip	Allow IP2IP connections between two SIP call legs
redundancy-group 1	Enable High Availability for the VoIP service
fax protocol	Specifies the fax protocol
asserted-id pai	Specifies the PAI header in the outgoing SIP requests and response messages

Codecs

G711u-Law and G729 voice codecs are configured for this testing. Codec preferences used to change according to the test plan description.

```
voice class codec 1
  codec preference 1 g711u-law
  codec preference 3 g729r8
```

Dial peer

Inbound calls from CUCM

!

```
dial-peer voice 100 voip
description *** Inbound Dial-Peer- from CUCM to CUBE ***
session protocol sipv2
session transport udp
incoming uri via CUCM
voice-class codec 2
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax protocol t38 version 3 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
```

!

Outbound calls to CUCM:

```
dial-peer voice 410 voip
description *** Outbound Dial-Peer towards CUCM ***
huntstop
destination-pattern 531.....
session protocol sipv2
session target ipv4:172.16.29.72
session transport udp
voice-class codec 2
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
fax-relay ecm disable
fax rate disable
```

```
fax protocol t38 version 3 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
```

Inbound from COX:

```
dial-peer voice 200 voip
description *** Inbound Dial-Peer- from COX to CUBE ***
translation-profile incoming for_FOPBX
session protocol sipv2
session transport udp
incoming uri via COX
voice-class codec 2
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax protocol t38 version 3 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
```

Outbound To COX:

```
dial-peer voice 310 voip
description *** Outbound Dial-Peer towards COX over EDGESBC-PSTN ***
huntstop
destination-pattern .T
session protocol sipv2
session target ipv4:10.64.4.162:5060
session transport udp
voice-class codec 2
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
fax-relay ecm disable
```

fax rate disable

fax protocol t38 version 3 ls-redundancy 0 hs-redundancy 0 fallback none
no vad

!

Dial peer group

voice class dpg 100

description incoming CUCM to Cox

dial-peer 310 preference 1

!

voice class dpg 300

description incoming COX to CUCM

dial-peer 410 preference 1

Configuration example

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

Active Cisco UBE:

```
version 17.3
service config
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
no service private-config-encryption
service call-home
platform qfp utilization monitor load 80
platform punt-keepalive disable-kernel-core
!
hostname CUBE7
!
boot-start-marker
boot system flash bootflash:isr4300-universalk9.17.03.03.SPA.bin
boot-end-marker
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
logging buffered 2000000
enable password 7 0010160D320A11575F2F
!
no aaa new-model
```

```
ip name-server x.x.x.x
login on-success log
!
subscriber templating
multilink bundle-name authenticated
no device-tracking logging theft
!
password encryption aes
!
crypto pki trustpoint SLA-TrustPoint
    enrollment pkcs12
    revocation-check crl
!
crypto pki certificate chain SLA-TrustPoint
    certificate ca 01
        30820321 30820209 A0030201 02020101 300D0609 2A864886 F70D0101 0B050030
        32310E30 0C060355 040A1305 43697363 6F312030 1E060355 04031317 43697363
        6F204C69 63656E73 696E6720 526F6F74 20434130 1E170D31 33303533 30313934
        3834375A 170D3338 30353330 31393438 34375A30 32310E30 0C060355 040A1305
        43697363 6F312030 1E060355 04031317 43697363 6F204C69 63656E73 696E6720
        526F6F74 20434130 82012230 0D06092A 864886F7 0D010101 05000382 010F0030
        82010A02 82010100 A6BCBD96 131E05F7 145EA72C 2CD686E6 17222EA1 F1EFF64D
        CBB4C798 212AA147 C655D8D7 9471380D 8711441E 1AAF071A 9CAE6388 8A38E520
        1C394D78 462EF239 C659F715 B98C0A59 5BBB5CBD 0CFEBEA3 700A8BF7 D8F256EE
        4AA4E80D DB6FD1C9 60B1FD18 FFC69C96 6FA68957 A2617DE7 104FDC5F EA2956AC
        7390A3EB 2B5436AD C847A2C5 DAB553EB 69A9A535 58E9F3E3 C0BD23CF 58BD7188
        68E69491 20F320E7 948E71D7 AE3BCC84 F10684C7 4BC8E00F 539BA42B 42C68BB7
        C7479096 B4CB2D62 EA2F505D C7B062A4 6811D95B E8250FC4 5D5D5FB8 8F27D191
        C55F0D76 61F9A4CD 3D992327 A8BB03BD 4E6D7069 7CBADF8B DF5F4368 95135E44
        DFC7C6CF 04DD7FD1 02030100 01A34230 40300E06 03551D0F 0101FF04 04030201
        06300F06 03551D13 0101FF04 05300301 01FF301D 0603551D 0E041604 1449DC85
        4B3D31E5 1B3E6A17 606AF333 3D3B4C73 E8300D06 092A8648 86F70D01 010B0500
        03820101 00507F24 D3932A66 86025D9F E838AE5C 6D4DF6B0 49631C78 240DA905
        604EDCDE FF4FED2B 77FC460E CD636FDB DD44681E 3A5673AB 9093D3B1 6C9E3D8B
```

```
D98987BF E40CBD9E 1AECA0C2 2189BB5C 8FA85686 CD98B646 5575B146 8DFC66A8
467A3DF4 4D565700 6ADF0F0D CF835015 3C04FF7C 21E878AC 11BA9CD2 55A9232C
7CA7B7E6 C1AF74F6 152E99B7 B1FCF9BB E973DE7F 5BDDEB86 C71E3B49 1765308B
5FB0DA06 B92AFE7F 494E8A9E 07B85737 F3A58BE1 1A48A229 C37C1E69 39F08678
80DDCD16 D6BACECA EEBC7CF9 8428787B 35202CDC 60E4616A B623CDBD 230E3AFB
418616A9 4093E049 4D10AB75 27E86F73 932E35B5 8862FDAE 0275156F 719BB2F0
D697DF7F 28
```

Quit

!

crypto pki certificate pool

! ('certificate ca' cmd has been deprecated. Downloaded

! Trustpool certificates should be re-downloaded

! using 'crypto pki trustpool import url <url>')

!

voice service voip

ip address trusted list

ipv4 172.16.29.x

ipv4 10.64.5.0 255.255.255.0

ipv4 172.17.8.0 255.255.255.0

no ip address trusted authenticate

address-hiding

mode border-element

media bulk-stats

allow-connections sip to sip

redundancy-group 1

no supplementary-service sip refer

no supplementary-service sip handle-replaces

supplementary-service media-renegotiate

fax protocol t38 version 3 ls-redundancy 0 hs-redundancy 0 fallback none

trace

sip

session refresh

header-passing

error-passthru

```
asserted-id pai
privacy pstn
conn-reuse
early-offer forced
midcall-signaling passthru
g729 annexb-all
sip-profiles inbound
no call service stop
!
!
voice class uri CUCM sip
  host 172.16.29.72
!
voice class uri COX sip
  host ipv4:10.64.4.x
voice class codec 1
  codec preference 1 g711ulaw
  codec preference 3 g729r8
!
!
voice class sip-profiles 101
request INVITE sip-header Diversion modify "<.*:.*>(@.*>)" "\1@10.71.3.40>"
!
voice class dpg 100
  description incoming CUCM to Cox
  dial-peer 310 preference 1
!
voice class dpg 300
  description incoming COX to CUCM
  dial-peer 410 preference 1
!
voice class sip-options-keepalive 1
  description ** global options pings settings **
  up-interval 30
```

```
retry 3
!
voice-card 0/2
  no watchdog
!
voice-card 0/4
  no watchdog
!
no license feature hseck9
license udi pid ISR4331/K9 sn FDO21381GMV
license accept end user agreement
license boot level appxk9
license boot level uck9
license boot level securityk9
memory free low-watermark processor 67123
!
diagnostic bootup level minimal
!
spanning-tree extend system-id
!
!
redundancy
  mode none
  application redundancy
  group 1
    priority 150 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 CUBE LAN INTERFACE  
  ip address 10.64.5.x 255.255.0.0  
  negotiation auto  
  redundancy rii 10  
  redundancy group 1 ip 10.64.5.x exclusive  
!  
interface GigabitEthernet0/0/1  
  description CUBE WAN INTERFACE  
  ip address 10.71.3.42 255.255.255.0  
  negotiation auto  
  redundancy rii 20  
  redundancy group 1 ip 10.71.3.40 exclusive  
!  
interface GigabitEthernet0/0/2  
  description Access01 g4/0/9  
  ip address 10.71.9.25 255.255.255.0  
  negotiation auto  
!  
interface GigabitEthernet0/1/0  
  no ip address  
  shutdown  
  negotiation auto  
!  
interface Service-Engine0/2/0  
!  
interface Service-Engine0/4/0  
!  
interface GigabitEthernet0  
  vrf forwarding Mgmt-intf  
  no ip address  
  negotiation auto  
!
```

```
no ip http server
ip http authentication local
no ip http secure-server
ip http client source-interface GigabitEthernet0/0/0
ip forward-protocol nd
ip route 0.0.0.0 0.0.0.0 10.71.3.1
ip route 10.64.4.0 255.255.255.0 10.71.3.1
ip route 10.64.5.0 255.255.255.0 10.64.1.1
!
control-plane
!
!
voice-port 0/2/0
!
voice-port 0/2/1
!
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
!
dial-peer voice 100 voip
  description *** Inbound Dial-Peer- from CUCM to CUBE ***
  session protocol sipv2
  session transport udp
  incoming uri via CUCM
  voice-class codec 2
  voice-class sip bind control source-interface GigabitEthernet0/0/0
  voice-class sip bind media source-interface GigabitEthernet0/0/0
  dtmf-relay rtp-nte
  fax-relay ecm disable
  fax rate disable
```

```
fax protocol t38 version 3 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
```

```
!
```

```
dial-peer voice 200 voip
```

```
description *** Inbound Dial-Peer- from COX to CUBE ***
```

```
translation-profile incoming for_FOPBX
```

```
session protocol sipv2
```

```
session transport udp
```

```
incoming uri via COX
```

```
voice-class codec 2
```

```
voice-class sip bind control source-interface GigabitEthernet0/0/1
```

```
voice-class sip bind media source-interface GigabitEthernet0/0/1
```

```
dtmf-relay rtp-nte
```

```
fax-relay ecm disable
```

```
fax rate disable
```

```
fax protocol t38 version 3 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
```

```
!
```

```
dial-peer voice 310 voip
```

```
description *** Outbound Dial-Peer towards COX over EDGESBC-PSTN ***
```

```
huntstop
```

```
destination-pattern .T
```

```
session protocol sipv2
```

```
session target ipv4:10.64.4.162:5060
```

```
session transport udp
```

```
voice-class codec 2
```

```
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
```

```
fax-relay ecm disable
```

```
fax rate disable
```

```
fax protocol t38 version 3 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
```

```
!  
dial-peer voice 410 voip  
  description *** Outbound Dial-Peer towards CUCM ***  
  huntstop  
  destination-pattern 531.....  
  session protocol sipv2  
  session target ipv4:172.16.29.72  
  session transport udp  
  voice-class codec 2  
  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  
  fax-relay ecm disable  
  fax rate disable  
  fax protocol t38 version 3 ls-redundancy 0 hs-redundancy 0 fallback none  
  no vad  
!  
!  
gateway  
  timer receive-rtp 1200  
!  
!  
line con 0  
  exec-timeout 0 0  
  logging synchronous  
  stopbits 1  
line aux 0  
  stopbits 1  
line vty 0 4  
  exec-timeout 30 30  
logging synchronous  
login  
transport input telnet
```

```
!  
call-home  
  ! If contact email address in call-home is configured as sch-smart-  
  licensing@cisco.com  
  ! the email address configured in Cisco Smart License Portal will be used as  
  contact email address to send SCH notifications.  
  contact-email-addr sch-smart-licensing@cisco.com  
  profile "CiscoTAC-1"  
  active  
  destination transport-method http  
!  
End
```

Standby Cisco UBE:

```
version 17.3
service config
service timestamps debug datetime msec
service timestamps log datetime msec
service call-home
platform qfp utilization monitor load 80
platform punt-keepalive disable-kernel-core
!
hostname CUBE8
!
boot-start-marker
boot system flash bootflash:isr4300-universalk9.17.03.03.SPA.bin
boot-end-marker
!
!
vrf definition Mgmt-intf
!
  address-family ipv4
  exit-address-family
!
  address-family ipv6
  exit-address-family
!
no aaa new-model
!
ip vrf forwarding
!
ip name-server x.x.x.x
login on-success log
!
subscriber templating
!
```

```

multilink bundle-name authenticated
no device-tracking logging theft
!
password encryption aes
!
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 trustpoint SLA-TrustPoint
  enrollment pkcs12
  revocation-check crl
!
crypto pki certificate chain TP-self-signed-3616943619
certificate self-signed 01
  30820330 30820218 A0030201 02020101 300D0609 2A864886 F70D0101 05050030
  31312F30 2D060355 04031326 494F532D 53656C66 2D536967 6E65642D 43657274
  69666963 6174652D 33363136 39343336 3139301E 170D3231 30333131 31383336
  31385A17 0D333130 33313131 38333631 385A3031 312F302D 06035504 03132649
  4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D33 36313639
  34333631 39308201 22300D06 092A8648 86F70D01 01010500 0382010F 00308201
  0A028201 0100CC32 F6D8D39C A0BC8806 96974E12 4390FF0C 9EFD8638 C41CD0F7
  E13A6180 ABF9F775 F5E0680E D9441928 11492752 81251D9F 8FB05970 6144AB81
  0AAAB492 9C3F8533 330DF760 E5D0CBCD BDA740B2 046A0D9E EFA02C07 68A65747
  5507EEAE A91258A4 C116FBB9 B77A6A80 83AA6562 5A026D02 8B71578E 4D7FE57E
  E742AC93 DCF4BA4B 0A7F0F61 95E50CD2 2DB9A832 DF21404A BBCC4026 FC121099
  23B93665 C45BC25B 66BF11DA 9C9F52C6 22172878 3C973DC4 A96BB055 88776359
  BA6621DC ECE7162C 8C3C0079 09613EC6 0E96C6BD 7A3AED3C 4BA1E681 C7CDE874
  1F32AE69 22A3BE86 FD0EBC19 858E6015 66226E81 61169B97 8B01C38C 84ABBFAB
  30A92E56 EBA90203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF
  301F0603 551D2304 18301680 14A9AA09 C6C70997 473BB054 F8B6F282 35C2E198
  43301D06 03551D0E 04160414 A9AA09C6 C7099747 3BB054F8 B6F28235 C2E19843

```

```
300D0609 2A864886 F70D0101 05050003 82010100 1451CDA7 97C50823 04FFBBBD
49040A2C 84AC3CF0 1370FC6B B1DB5D54 E93A999F 9CB12E4B EEF7EC69 0892C4C1
48932BA1 FC4B4EBF 7420DFFD C777038A 8F6BC679 FC87C3BE 1D8FF0D8 6B75477B
219FD6B3 B8277864 8179E6DF 6A48242C 4C38A48E 081E1737 9A8DDA86 4BE7577F
6E467B9C B729B607 F4E7D941 2E857F70 C892BCE4 D21696AB 19889BC7 F9C33106
9BCA8181 3B390EED 7AEE9A0A 61E68CA0 AB8B48CC 0E69DB56 D9340B75 CC3372EA
F38B8270 406FB86F 57222154 959701FC C0855F08 E8751094 1AAD4E63 B66457E7
F311EE8A 619E7E49 409E65CC C4D99221 D7D12A3A 51BB134D 9A0077BC 5FCA0F69
2CF41AF5 E9DCEB60 442FB6F5 1FDB376E 4D2BD28C
```

quit

crypto pki certificate chain SLA-TrustPoint

certificate ca 01

```
30820321 30820209 A0030201 02020101 300D0609 2A864886 F70D0101 0B050030
32310E30 0C060355 040A1305 43697363 6F312030 1E060355 04031317 43697363
6F204C69 63656E73 696E6720 526F6F74 20434130 1E170D31 33303533 30313934
3834375A 170D3338 30353330 31393438 34375A30 32310E30 0C060355 040A1305
43697363 6F312030 1E060355 04031317 43697363 6F204C69 63656E73 696E6720
526F6F74 20434130 82012230 0D06092A 864886F7 0D010101 05000382 010F0030
82010A02 82010100 A6BCBD96 131E05F7 145EA72C 2CD686E6 17222EA1 F1EFF64D
CBB4C798 212AA147 C655D8D7 9471380D 8711441E 1AAF071A 9CAE6388 8A38E520
1C394D78 462EF239 C659F715 B98C0A59 5BBB5CBD 0CFE8EA3 700A8BF7 D8F256EE
4AA4E80D DB6FD1C9 60B1FD18 FFC69C96 6FA68957 A2617DE7 104FDC5F EA2956AC
7390A3EB 2B5436AD C847A2C5 DAB553EB 69A9A535 58E9F3E3 C0BD23CF 58BD7188
68E69491 20F320E7 948E71D7 AE3BCC84 F10684C7 4BC8E00F 539BA42B 42C68BB7
C7479096 B4CB2D62 EA2F505D C7B062A4 6811D95B E8250FC4 5D5D5FB8 8F27D191
C55F0D76 61F9A4CD 3D992327 A8BB03BD 4E6D7069 7CBADF8B DF5F4368 95135E44
DFC7C6CF 04DD7FD1 02030100 01A34230 40300E06 03551D0F 0101FF04 04030201
06300F06 03551D13 0101FF04 05300301 01FF301D 0603551D 0E041604 1449DC85
4B3D31E5 1B3E6A17 606AF333 3D3B4C73 E8300D06 092A8648 86F70D01 010B0500
03820101 00507F24 D3932A66 86025D9F E838AE5C 6D4DF6B0 49631C78 240DA905
604EDCDE FF4FED2B 77FC460E CD636FDB DD44681E 3A5673AB 9093D3B1 6C9E3D8B
D98987BF E40CBD9E 1AECA0C2 2189BB5C 8FA85686 CD98B646 5575B146 8DFC66A8
467A3DF4 4D565700 6ADF0F0D CF835015 3C04FF7C 21E878AC 11BA9CD2 55A9232C
7CA7B7E6 C1AF74F6 152E99B7 B1FCF9BB E973DE7F 5BDDEB86 C71E3B49 1765308B
```

5FB0DA06 B92AFE7F 494E8A9E 07B85737 F3A58BE1 1A48A229 C37C1E69 39F08678
80DDCD16 D6BACECA EEBC7CF9 8428787B 35202CDC 60E4616A B623CDBD 230E3AFB
418616A9 4093E049 4D10AB75 27E86F73 932E35B5 8862FDAE 0275156F 719BB2F0
D697DF7F 28

quit

!

crypto pki certificate pool
cabundle nvram:ios_core.p7b

!

voice service voip
mode border-element
media bulk-stats
allow-connections sip to sip
redundancy-group 1
no supplementary-service sip refer
no supplementary-service sip handle-replaces
fax protocol t38 version 3 ls-redundancy 0 hs-redundancy 0 fallback none
trace

sip

rel1xx disable
conn-reuse
early-offer forced
midcall-signaling passthru
privacy-policy passthru
g729 annexb-all

!

voice class uri CUCM sip
host 172.16.29.72

!

voice class uri COX sip
host ipv4:10.64.4.x

voice class codec 1
codec preference 1 g729r8
codec preference 2 g711ulaw

```
!  
!  
voice class sip-profiles 200  
  request INVITE sip-header Diversion modify "<.*:.*>(@.*>)" "\1@10.71.3.40>"  
!  
voice class dpg 100  
  description incoming CUCM to Cox  
  dial-peer 310 preference 1  
!  
voice class dpg 300  
  description incoming COX to CUCM  
  dial-peer 410 preference 1  
!  
voice-card 0/2  
  no watchdog  
!  
voice-card 0/4  
  dsp services dspfarm  
  no watchdog  
!  
no license feature hseck9  
license udi pid ISR4331/K9 sn FD021381FEY  
license accept end user agreement  
license boot level appxk9  
license boot level uck9  
license boot level securityk9  
memory free low-watermark processor 67123  
!  
diagnostic bootup level minimal  
!  
spanning-tree extend system-id  
!  
username cisco privilege 15 password 6 `egTW_bFQEGWfSiSiVgM_]H^VRFAAB  
!
```

```
redundancy
mode none
application redundancy
group 1
priority 150 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 CUBE LAN INTERFACE
ip address 10.64.5.220 255.255.0.0
negotiation auto
no mop enabled
redundancy rii 10
redundancy group 1 ip 10.64.5.165 exclusive
!
interface GigabitEthernet0/0/1
description CUBE WAN INTERFACE
ip dhcp client client-id ascii cisco-ec1d.8bd2.9751-Gi0/0/1
ip address 10.71.3.x 255.255.255.0
negotiation auto
no mop enabled
redundancy rii 20
redundancy group 1 ip 10.71.3.x exclusive
!
interface GigabitEthernet0/0/2
description HA interface
```

```
ip address 10.71.9.26 255.255.255.0
negotiation auto
no mop enabled
!
interface Service-Engine0/2/0
!
interface Service-Engine0/4/0
!
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
negotiation auto
no mop enabled
!
no ip http server
ip http authentication local
no ip http secure-server
ip http client source-interface GigabitEthernet0/0/1
ip forward-protocol nd
ip route 0.0.0.0 0.0.0.0 10.71.3.1
ip route 10.64.4.0 255.255.255.0 10.71.3.1
ip route 10.64.5.0 255.255.255.0 10.64.1.1
!
control-plane
!
voice-port 0/2/0
!
voice-port 0/2/1
!
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
!
dspfarm profile 1 transcode
  codec g729abr8
  codec g729ar8
  codec g711alaw
  codec g711ulaw
  codec g729r8
  codec g729br8
  maximum sessions 10
  associate application CUBE
  shutdown
!
dial-peer voice 100 voip
  description *** Inbound Dial-Peer- from CUCM to CUBE ***
  session protocol sipv2
  session transport udp
  destination dpq 100
  incoming uri via CUCM
  voice-class codec 1
  voice-class sip bind control source-interface GigabitEthernet0/0/0
  voice-class sip bind media source-interface GigabitEthernet0/0/0
  dtmf-relay rtp-nte
  fax-relay ecm disable
  fax protocol t38 version 3 ls-redundancy 0 hs-redundancy 0 fallback none
  no vad
!
dial-peer voice 200 voip
  description *** Inbound Dial-Peer- from COX to CUBE ***
  translation-profile incoming for_FOPBX
  session protocol sipv2
  session transport udp
  destination dpq 300
  incoming uri via COX
```

```
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax protocol t38 version 3 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 310 voip
description *** Outbound Dial-Peer towards COX over EDGESBC-PSTN ***
huntstop
destination-pattern .T
session protocol sipv2
session target ipv4:10.64.4.162:5060
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip profiles 200
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
fax-relay ecm disable
fax protocol t38 version 3 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 410 voip
description *** Outbound Dial-Peer towards CUCM ***
huntstop
destination-pattern 531999....
session protocol sipv2
session target ipv4:172.16.29.72
session transport udp
voice-class codec 1
```

```
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
fax-relay ecm disable
fax protocol t38 version 3 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 320 voip
description *** Outbound Dial-Peer towards COX over EDGESBC-PSTN ***
huntstop
shutdown
destination-pattern 031.....
session protocol sipv2
session target ipv4:10.64.4.162:5060
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip profiles 200
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
!
gateway
timer receive-rtp 1200
!
!
line con 0
logging synchronous
stopbits 1
line aux 0
stopbits 1
```

```
line vty 0 4
  exec-timeout 30 30
  logging synchronous
  login
  transport input telnet
!
call-home
  ! If contact email address in call-home is configured as sch-smart-
  licensing@cisco.com
  ! the email address configured in Cisco Smart License Portal will be used as
  contact email address to send SCH notifications.
  contact-email-addr sch-smart-licensing@cisco.com
  profile "CiscoTAC-1"
  active
  destination transport-method http
!
end
```

Configuring Cisco UCM 14.0

Cisco UCM Version

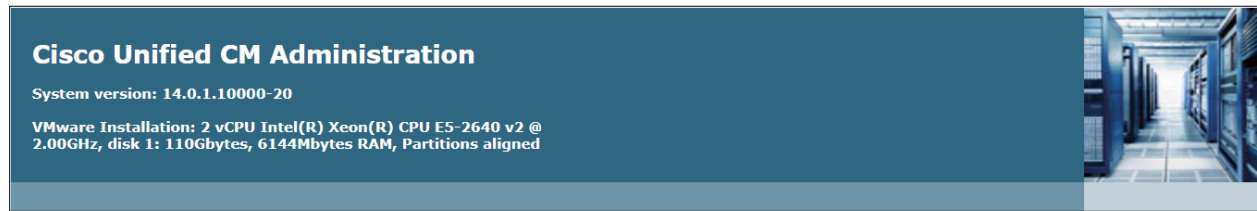


Figure 3: Cisco UCM Version

Cisco Call Manager Service Parameters

Navigation: System → Service Parameters

- Select Server* = cucm4-XXXX-CUCM Voice/Video (Active)
- Select Service* = Cisco CallManager (Active)
- All fields are set to default values

The screenshot displays the "Service Parameter Configuration" page in the Cisco Unified CM Administration interface. The page has a navigation bar at the top with "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", "Bulk Administration", and "Help". Below the navigation bar, the "Service Parameter Configuration" section is active, showing "Related Links: Parameters for All Servers" and "Go". The "Status" section indicates "Status: Ready". The "Select Server and Service" section shows "Server*" as "cucm4-XXXX-CUCM Voice/Video (Active)" and "Service*" as "Cisco CallManager (Active)". Below this, a note states: "All parameters apply only to the current server except parameters that are in the cluster-wide group(s)". The "Cisco CallManager (Active) Parameters on server 172.16.29.33--CUCM Voice/Video (Active)" section contains a table of parameters. The table has three columns: "Parameter Name", "Parameter Value", and "Suggested Value". The parameters listed are "Code Yellow Entry Latency", "Code Yellow Exit Latency Calculation", "Code Yellow Duration", and "Max Events Allowed".

Parameter Name	Parameter Value	Suggested Value
Code Yellow Entry Latency *	20	20
Code Yellow Exit Latency Calculation *	40	40
Code Yellow Duration *	5	5
Max Events Allowed *	2000	2000

Figure 4: Service Parameters

SIP Trunk Security Profile

Navigation: System → Security → SIP Trunk Security Profile

- Name* = **Non Secure SIP Trunk Profile Cox**
- Description = **Non Secure SIP Trunk Profile authenticated by null String**
- Device Security Mode = **Non Secure**
- Incoming Transport Type* = **TCP + UDP**
- Outgoing Transport Type = **UDP**
- Incoming port = **5060**


Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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

SIP Trunk Security Profile Configuration

 Save
 Delete
 Copy
 Reset
 Apply Config
 Add New

SIP Trunk Security Profile Information

Name*
Description
Device Security Mode
Incoming Transport Type*
Outgoing Transport Type
☐ Enable Digest Authentication
Nonce Validity Time (mins)*
Secure Certificate Subject or Subject Alternate Name

Incoming Port*
☐ 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*

Cox-Non Secure SIP Trunk Profile
Non Secure SIP Trunk Profile authenticated by null String
Non Secure
TCP+UDP
UDP
600

5060

Use Default Filter

Save Delete Copy Reset Apply Config Add New

Figure 5: SIP Trunk Security Profile

SIP Profile

Navigation: Device → Device Settings → SIP Profile

- Name* = **Standard SIP Profile -PING**
- Description = **Default SIP Profile -PING**
- Highlighted fields are to be focused

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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

SIP Profile Configuration

Save X Delete Copy Reset Apply Config Add New

Status

Status: Ready

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

SIP Profile Information

Name* Cox-Standard SIP Profile

Description Default SIP Profile

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 < None > ▾

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 6: SIP Profile


Cisco Unified CM Administration
 For Cisco Unified Communications Solutions

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

SIP Profile Configuration

 Save
  Delete
  Copy
  Reset
  Apply Config
  Add New

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	☒ Common Port Range for Audio and Video ☐ 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	16 (010000 - CS2)
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

Figure 7: SIP Profile Contd.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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

SIP Profile Configuration

Save
 Delete
 Copy
 Reset
 Apply Config
 Add New

10000

Call Forward URI*
Speed Dial (Abbreviated Dial) URI*

x-cisco-serviceuri-cfwdall

x-cisco-serviceuri-abbrdial

☒ Conference Join Enabled
☐ RFC 2543 Hold
☒ Semi Attended Transfer
☐ Enable VAD
☐ Stutter Message Waiting
☐ MLPP User Authorization

Normalization Script
Normalization Script < None > ▾
☐ Enable Trace

	Parameter Name	Parameter Value	
1			

External Presentation Information
☐ Anonymous External Presentation
External Presentation Number
External Presentation Name

Trunk Specific Configuration
Reroute Incoming Request to new Trunk based on* Never ▾
Resource Priority Namespace List < None > ▾
SIP Rel1XX Options* Disabled ▾
Video Call Traffic Class* Mixed ▾
Calling Line Identification Presentation* Default ▾
Session Refresh Method* Invite ▾
Early Offer support for voice and video calls* Disabled (Default value) ▾

Figure 8: SIP Profile Contd.


Cisco Unified CM Administration
 For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go
 administrator | About | Logout

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

SIP Profile Configuration Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

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

i *- indicates required item.

i **- setting only takes effect if the External QoS Enabled Service Parameter is set to true.

Figure 9: SIP Profile Contd.

Trunk configuration

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

Navigation: Device → Trunk

- Device Name= **Coxtrunk**
- Description = **Coxtrunk**
- Device Pool = **G711_codec**

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo and the title 'Cisco Unified CM Administration'. Below this, a secondary navigation bar lists various system components. The main content area is titled 'Trunk Configuration' and includes a 'Related Links' section with a 'Back To Find/List' link. The configuration page is divided into several sections: 'Status' (showing 'Ready'), 'SIP Trunk Status' (showing 'Full Service' and 'Time In Full Service: 0 day 4 hours 13 minutes'), and 'Device Information'. The 'Device Information' section contains a table of configuration parameters. The 'Device Name' field is set to 'Cox-Trunk' and the 'Device Pool' is set to 'Devicepool-1', both of which are highlighted with red boxes. Other fields include 'Product' (SIP Trunk), 'Device Protocol' (SIP), 'Trunk Service Type' (None(Default)), 'Description' (C-Trunk), 'Common Device Configuration' (< None >), 'Call Classification' (Use System Default), 'Media Resource Group List' (MGRL), 'Location' (Hub_None), 'AAR Group' (< None >), 'Tunneled Protocol' (None), 'QSIG Variant' (No Changes), 'ASN.1 ROSE OID Encoding' (No Changes), 'Packet Capture Mode' (None), and 'Packet Capture Duration' (0). There are also checkboxes for 'Media Termination Point Required', 'Retry Video Call as Audio' (checked), 'Path Replacement Support', and 'Transmit UTF-8 for Calling Party Name'.

Field	Value
Product	SIP Trunk
Device Protocol	SIP
Trunk Service Type	None(Default)
Device Name*	Cox-Trunk
Description	C-Trunk
Device Pool*	Devicepool-1
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	MGRL
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	☐

Figure 10: SIP Trunk

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go
administrator About Logout

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

Trunk Configuration Related Links: Back To Find/List Go

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*: 4
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

Figure 11 SIP Trunk Contd.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go
administrator About Logout

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

Trunk Configuration Related Links: Back To Find/List Go

Save Delete Reset Add New

Connected Party Settings

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

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
☐ Use original calling line's Calling Line ID Presentation for diverted calls

Presentation Information

☐ Anonymous Presentation
Presentation Number:
Presentation Name:
☐ Send Presentation Name and Number only in the FROM header and not in the other identity headers

Figure 12 SIP Trunk Contd.

- Configure the **Virtual LAN IP address of the Cisco UBE** and the **Destination Port**
- Associate the **SIP Trunk Security Profile** and **SIP Profile** as created earlier
- DTMF Signaling Method: **RFC2833**
- The rest of the configuration are set to default

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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Save | Delete | Reset | Add New

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port	Status	Status Reason	Duration
1*	10.64.5.165		5060	up		Time Up: 0 day 4 hou 13 minutes

MTP Preferred Originating Codec* 711ulaw

BLF Presence Group* Standard Presence group

SIP Trunk Security Profile* C-Non Secure SIP Trunk Profile

Rerouting Calling Search Space < None >

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

SUBSCRIBE Calling Search Space < None >

SIP Profile* Cox-Standard SIP Profile [View Details](#)

DTMF Signaling Method* RFC 2833

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 13 SIP Trunk Contd.

Routing configuration

Route Pattern for Cisco UBE:

Navigation: Call Routing → Route/Hunt → Route Pattern

- Route Pattern= **7.[0-9]![0-9#]** (Create Route patterns based on the dial plan requirement)
- Gateway/Route List = **Coxtrunk**(Associate the SIP Trunk created earlier)

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Route Pattern Configuration

Save Delete Copy Add New

Status
Status: Ready

Pattern Definition

Route Pattern* 7.[0-9]![0-9#]

Route Partition < None >

Description

Numbering Plan -- Not Selected --

Route Filter < None >

MLPP Precedence* Default

☐ Apply Call Blocking Percentage

Resource Priority Namespace Network Domain < None >

Route Class* Default

Gateway/Route List* Cox-Trunk (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

Figure 14: Route Pattern

- Discard Digits **PreDot**

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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 >	

Save
Delete
Copy
Add New

i *- indicates required item.

Figure 15 Route Pattern Contd.

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
SP	Service Provider

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