



Connecting Cisco Unified Communication Manager [12.5.1] to Peerless Network SIP Trunk via Cisco Unified Border Element v12.7.0 [IOS-XE 16.12.04]

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Introduction

Service Providers today, such as Peerless Network, 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.

Peerless Network 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 Peerless Network, Cisco Unified Border Element (Cisco UBE) ISR 4331/K9 running IOS-XE 16.12.04 can be used. The Cisco Unified Border Element v12.7.0 provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 12.5.1 connected to Peerless Network 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 Peerless Network 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 12.5.1, and Cisco UBE on ISR 4331/K9 [IOS-XE – 16.12.04] for connectivity to Peerless Network SIP Trunking service. The deployment model covered in this application note is CPE (Cisco UCM) to PSTN (Peerless Network) via Cisco UBE v12.7.0 [IOS-XE] 16.12.04.
- Testing was performed in accordance to Peerless Network 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 Peerless Network 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 Peerless Network SIP Trunking network.



Network Topology

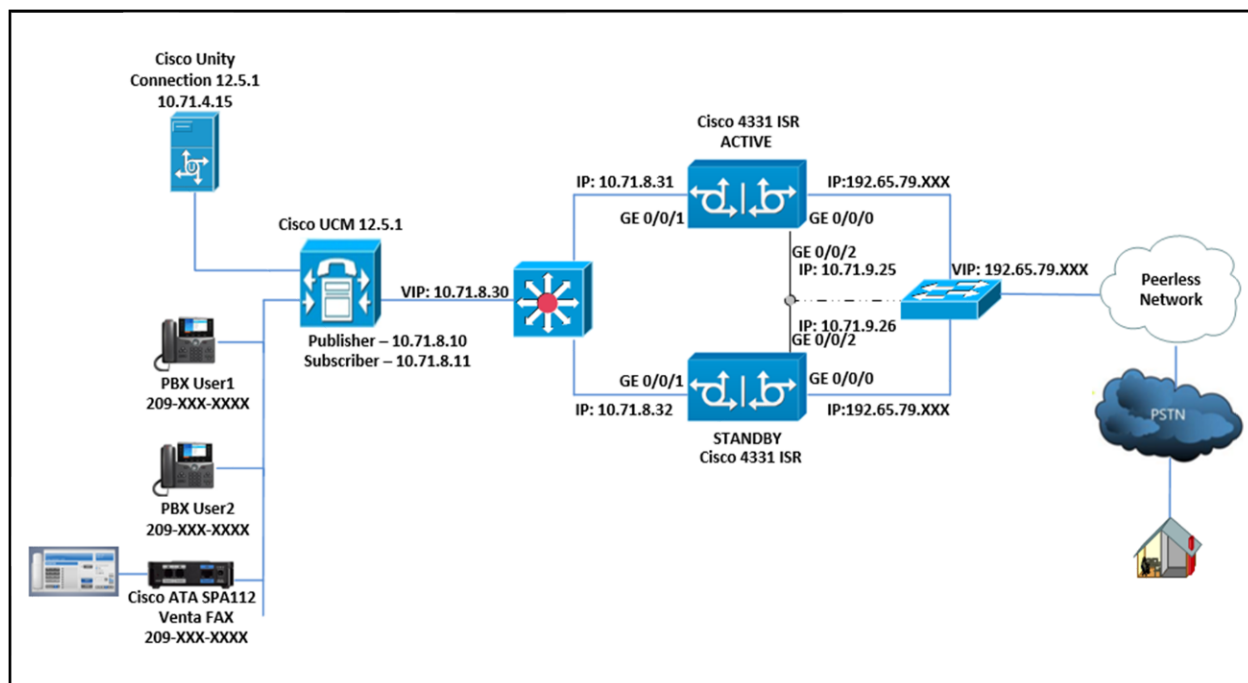


Figure 1 Network Topology

- The network topology includes the Cisco UCM Cluster, Unity Voicemail system and 2 Cisco Endpoints. Cisco UCM has a trunk configured to Cisco UBE's Virtual IP Address. Peerless Network 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 Peerless Network 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 Peerless Network IP	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 C240, 1 Publisher node and 1 Subscriber node.
- Generic Cisco IP-Phones.

Software Requirements

- Cisco UBE-Version: 12.7.0 running IOS-XE 16.12.4
- Cisco UCM-Version: 12.5.1.13900-152

Features

Features Supported

- Incoming, Outgoing and International calls using G711ulaw and G729 voice codecs
- Call Conference
- 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)
- Fax
 - G711 u-Law Pass-through
 - T.38
- IP-PBX Calling number privacy
- High Availability

Features Not Supported

- Cisco UCM does not support Blind Call transfer
- Peerless Network doesn't support 0,0+10 dial plan



Caveats

- None

Observations

- Only one IP PBX is used for testing.
- Cisco UCM phones used in this test does not support Blind Call transfer feature and hence the feature related test cases are not tested.
- Caller ID is not updated on attended and semi-attended transfer scenarios



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.

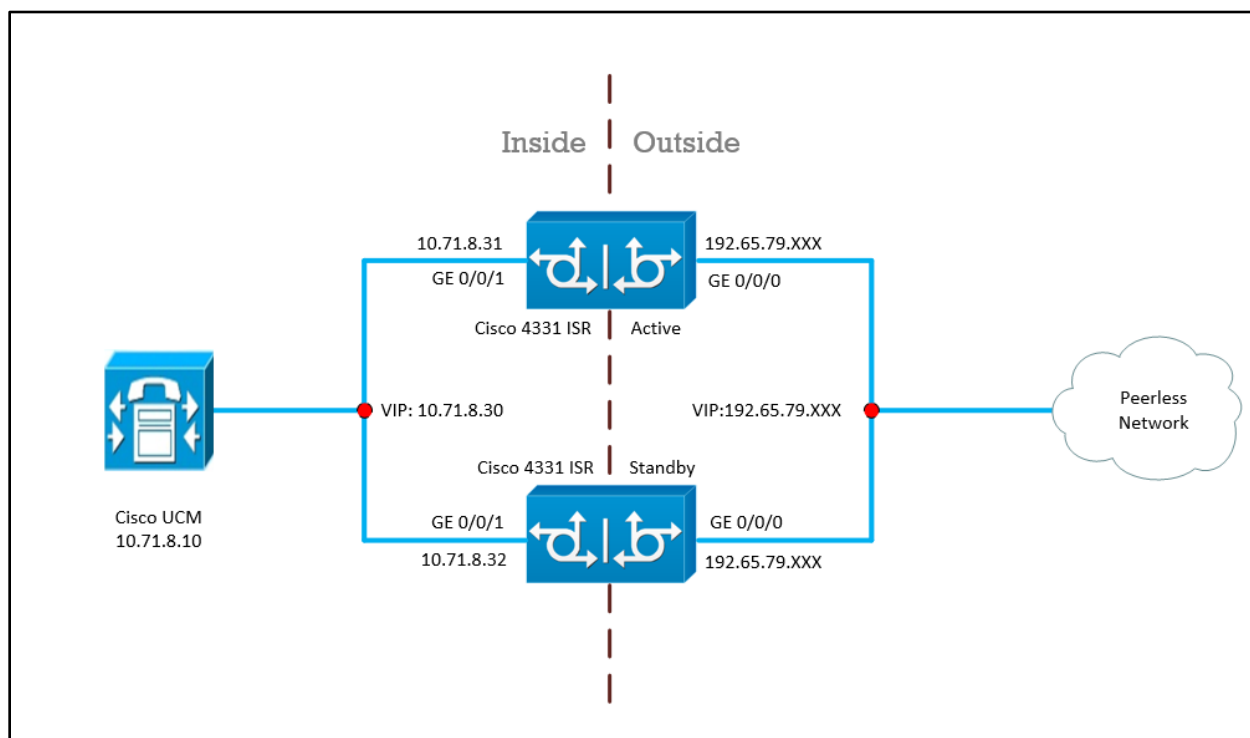


Figure 2 High Availability topology



Cisco UBE 1:

```
interface GigabitEthernet0/0/0
  description Peerless CUBE WAN interface
  ip address 192.65.79.XXX 255.255.255.128
  negotiation auto
  redundancy rii 10
  redundancy group 1 ip 192.65.79.XXX exclusive
!
interface GigabitEthernet0/0/1
  description Peerless CUBE LAN interface
  ip address 10.71.8.31 255.255.255.0
  negotiation auto
  redundancy rii 20
  redundancy group 1 ip 10.71.8.30 exclusive
!
interface GigabitEthernet0/0/2
  description Peerless CUBE HA Interface
  ip address 10.71.9.25 255.255.255.0
  negotiation auto
!
```



Cisco UBE 2:

```
interface GigabitEthernet0/0/0
  description Peerless CUBE WAN interface
  ip address 192.65.79.XXX 255.255.255.128
  negotiation auto
  redundancy rii 10
  redundancy group 1 ip 192.65.79.XXX exclusive
!
interface GigabitEthernet0/0/1
  description Peerless CUBE LAN interface
  ip address 10.71.8.32 255.255.255.0
  negotiation auto
  redundancy rii 20
  redundancy group 1 ip 10.71.8.30 exclusive
!
interface GigabitEthernet0/0/2
  description Peerless CUBE HA Interface
  ip address 10.71.9.26 255.255.255.0
  negotiation auto
!
```



Global Cisco UBE settings

To enable Cisco UBE IP2IP SBC functionality, following commands have to be entered:

```
voice service voip
  ip address trusted list
    ipv4 208.93.42.xxx 255.255.255.255
    ipv4 10.71.8.10 255.255.255.255
    ipv4 10.71.8.11 255.255.255.255
    ipv4 10.71.8.12 255.255.255.255
  address-hiding
  mode border-element license capacity 20
  allow-connections sip to sip
  redundancy-group 1
  fax protocol pass-through g711ulaw
  sip
    bind control source-interface GigabitEthernet0/0/0
    bind media source-interface GigabitEthernet0/0/0
  session refresh
  asserted-id pai
  early-offer forced
  midcall-signaling passthru
  privacy-policy passthru
  g729 annexb-all
!
```

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	Specifies the privacy 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 g711ulaw
  codec preference 2 g729r8
!
voice class codec 2
  codec preference 1 g729r8
  codec preference 2 g711ulaw
!
```

Dial peer

Outbound Dial-peer to Peerless Network:

```
dial-peer voice 101 voip
  description *** Outbound Call from CUBE-LAN to CUCM****
  destination-pattern 120.....
  session protocol sipv2
  session target ipv4:10.71.8.10:5060
  session transport udp
  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
  no fax-relay sg3-to-g3
  fax protocol pass-through g711ulaw
  no vad
!
dial-peer voice 201 voip
  description *** Outbound Call from CUBE-WAN to Peerless ****
```



```
destination-pattern .T
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip profiles 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
no fax-relay sg3-to-g3
fax protocol pass-through g711ulaw
no vad
!
```

Inbound Dial-peer from Peerless Network:

```
dial-peer voice 100 voip
description *** Inbound Call from CUCM to CUBE-LAN ***
session protocol sipv2
session transport udp
incoming called-number .T
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
no fax-relay sg3-to-g3
fax protocol pass-through g711ulaw
no vad
!

dial-peer voice 200 voip
description *** Inbound Call from Peerless to CUBE-WAN ***
session protocol sipv2
```



```
session transport udp
incoming called-number 120.....
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
no fax-relay sg3-to-g3
fax protocol pass-through g711ulaw
no vad
!
```



Configuration example

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

Active Cisco UBE:

```
version 16.12
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
service call-home
platform qfp utilization monitor load 80
platform punt-keepalive disable-kernel-core
!
hostname CUBE7
!
boot-start-marker
boot system bootflash:isr4300-universalk9.16.12.04.SPA.bin
boot-end-marker
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
enable secret 9 $9$WqvswdeKDYk$aWfZx9jLuxYKTg1njANmuDAXLCRXapdky0zd9.KvFRc
!
no aaa new-model
call-home
profile "XXXX"
```



```
active
destination transport-method http
no destination transport-method email
!
ip name-server 8.8.8.8
!
login on-success log
subscriber templating
!
multilink bundle-name authenticated
!
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
```

```
!
```

```
voice service voip
```

```
ip address trusted list
```

```
ipv4 208.93.42.xxx 255.255.255.255
```

```
ipv4 10.71.8.10 255.255.255.255
```

```
ipv4 10.71.8.11 255.255.255.255
```

```
ipv4 10.71.8.12 255.255.255.255
```

```
address-hiding
```

```
mode border-element license capacity 20
```

```
allow-connections sip to sip
```

```
redundancy-group 1
```

```
fax protocol pass-through g711ulaw
```

```
sip
```

```
bind control source-interface GigabitEthernet0/0/0
```

```
bind media source-interface GigabitEthernet0/0/0
```

```
session refresh
```

```
asserted-id pai
```

```
early-offer forced
```

```
midcall-signaling passthru
```

```
privacy-policy 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 sip-profiles 1  
    request INVITE sip-header Diversion modify "<sip:(....)@" "<sip:209xxx\1@"  
    request INVITE sip-header From modify "<sip:(.*)@.*>"  
    "<sip:XXXXXXXXXX_CISCO661@gw1.peerlessnetwork.io:5060>"  
!  
voice translation-rule 1  
    rule 1 /9209xxxxxxx/ /209xxxxxxx/  
!  
!  
voice translation-profile loopback  
    translate called 1  
!  
voice-card 0/2  
    no watchdog  
!  
voice-card 0/4  
    no watchdog  
!  
no license feature hseck9  
license udi pid ISR4331/K9 sn XXXXXXXX  
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 200 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 Peerless CUBE WAN interface
    ip address 192.65.79.XXX 255.255.255.128
    negotiation auto
    redundancy rii 10
    redundancy group 1 ip 192.65.79.XXX exclusive
!
interface GigabitEthernet0/0/1
    description Peerless CUBE LAN interface
    ip address 10.71.8.31 255.255.255.0
    negotiation auto
    redundancy rii 20
```



```
redundancy group 1 ip 10.71.8.30 exclusive
!
interface GigabitEthernet0/0/2
  description Peerless CUBE HA Interface
  ip address 10.71.9.25 255.255.255.0
  negotiation auto
!
interface GigabitEthernet0/1/0
  no ip address
  negotiation auto
!
interface Service-Engine0/2/0
!
interface Service-Engine0/4/0
!
interface GigabitEthernet0
  vrf forwarding Mgmt-intf
  no ip address
  negotiation auto
!
ip forward-protocol nd
no ip http server
no ip http secure-server
ip route 0.0.0.0 0.0.0.0 192.65.79.xxx
ip route 10.64.0.0 255.255.0.0 10.64.1.1
ip route 10.71.8.0 255.255.255.0 10.64.1.1
ip route 10.71.9.0 255.255.255.0 10.64.1.1
ip route 172.16.0.0 255.255.0.0 10.71.8.1
ip route 172.17.0.0 255.255.0.0 10.71.8.1
!
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 Call from CUCM to CUBE-LAN ***
  session protocol sipv2
  session transport udp
  incoming called-number .T
  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
  no fax-relay sg3-to-g3
  fax protocol pass-through g711ulaw
  no vad
!
dial-peer voice 101 voip
  description *** Outbound Call from CUBE-LAN to CUCM****
  destination-pattern 120.....
  session protocol sipv2
  session target ipv4:10.71.8.10:5060
  session transport udp
  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
  no fax-relay sg3-to-g3
```



```
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 200 voip
description *** Inbound Call from Peerless to CUBE-WAN ***
session protocol sipv2
session transport udp
incoming called-number 120.....
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
no fax-relay sg3-to-g3
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 201 voip
description *** Outbound Call from CUBE-WAN to Peerless ****
translation-profile outgoing loopback
destination-pattern .T
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip profiles 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
no fax-relay sg3-to-g3
fax protocol pass-through g711ulaw
no vad
!
```



```
!  
gateway  
    timer receive-rtp 1200  
!  
sip-ua  
    credentials username XXXXXXXX_CISCOC661 password 7 XXXXXX realm  
reg.peerlessnetwork.io  
    registrar dns:gw1.peerlessnetwork.io:5060 expires 60  
    sip-server dns:gw1.peerlessnetwork.io:5060  
!  
!  
line con 0  
    stopbits 1  
line aux 0  
    stopbits 1  
line vty 0 4  
    exec-timeout 30 30  
    password 7 XXXXXXXXX  
    login  
!  
!  
!  
end
```



Standby Cisco UBE:

```
version 16.12
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
service call-home
platform qfp utilization monitor load 80
platform punt-keepalive disable-kernel-core
!
hostname CUBE8
!
boot-start-marker
boot system bootflash:isr4300-universalk9.16.12.04.SPA.bin
boot-end-marker
!
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
enable secret 9 $9$s0ATbbL5j5ok4.mNCRebFwGyWcBoZunVuUhRIWv6SHw1dsDtrrj7nghc.
!
no aaa new-model
call-home
profile "XXXXXX"
active
destination transport-method http
no destination transport-method email
```



```
ip name-server 8.8.8.8
!
login on-success log
subscriber templating
multilink bundle-name authenticated
!
password encryption aes
!
crypto pki trustpoint SLA-TrustPoint
  enrollment terminal
  revocation-check crl
!
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 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 EEB7C7F9 8428787B 35202CDC 60E4616A B623CDBD 230E3AFB
418616A9 4093E049 4D10AB75 27E86F73 932E35B5 8862FDAE 0275156F 719BB2F0
D697DF7F 28
```

```
quit
```

```
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 170D3230 31303037 30383332
35305A17 0D333030 31303130 30303030 305A3031 312F302D 06035504 03132649
4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D33 36313639
34333631 39308201 22300D06 092A8648 86F70D01 01010500 0382010F 00308201
0A028201 0100D7B5 B01CB61F 405692AB 0367B5FD FFDD7DE7 F2779111 6742D4CC
729BA52D 38816BC7 BD076992 B8FB4CFF 9EF76D2F 2B170F33 58302925 E8BC82A3
2730377A 53B10B7E 887D003B 4F117214 2AB94356 BA35364F 01AF4B32 20AAE06E
33FC9D86 4E52101B 32BD500B FB8E154C 71602FC7 CF65C1FA 87BAF066 D4CA48ED
A19EA403 2DC6374B 719905A1 C6600AF2 AA0B64F5 6D00EE5B 4A165757 2F3C93C4
2C9C8E7B 54B4733C 89FCBBFF 663174C9 D6661110 9ACCC49C DE25143D 5DF87ECB
DC0F7183 E76D02D7 1F3E89FC D30D91C1 165483D6 AF5270B6 14C83073 6C2B5990
E0D66B36 6A98D663 C969A8BA 45F7F5CB FADD5C6F B15E13DB 4C5F0E72 9FB6B049
0F16EA21 44130203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF
301F0603 551D2304 18301680 14B2A0A6 33E67A00 23D6FD1E C788ADE1 2E569223
```



```
6D301D06 03551D0E 04160414 B2A0A633 E67A0023 D6FD1EC7 88ADE12E 5692236D
300D0609 2A864886 F70D0101 05050003 82010100 8C471DC9 C1CEB945 5C84C148
64938048 988D4F8A DE90C28C CD23386E 6BE64436 7AA84811 8CD3E5F6 93A4E8EF
8D8DC5AE 4D3129EA 9BC9773A 049E2D80 DEC3A3D5 54252CB3 AD73D570 B242240E
B934D489 FC547738 FF502D71 BB2A332E A06A2219 3F69FECA 1AEEFF2E D0C30C8A
C0F41FCE 4875BDE7 EF489676 67097E4D 07DEB502 1F4DE3E4 28E01CA0 30887BAD
8BAF7F58 C51E2F32 A2661908 8E4E9427 4D98BE1E 3B7E3565 97A60C1D D56692A7
480B7F5F 016BC74C 756740AE 9BCDABCA 0AAD574B 5AB3A1AB A7A3D570 4FDB4D7A
093CF60C B7C5CFAC 329CC205 804CBC91 46F93DDD 87311771 73EA8AF5 01E8DE4D
5CB04AEE 03A47061 9C3DF7E9 AAB0C7F1 276F29A6
```

```
quit
```

```
!
```

```
voice service voip
```

```
ip address trusted list
```

```
ipv4 10.71.8.10 255.255.255.255
```

```
ipv4 10.71.8.11 255.255.255.255
```

```
ipv4 10.71.8.12 255.255.255.255
```

```
ipv4 208.93.42.xxx 255.255.255.255
```

```
address-hiding
```

```
mode border-element license capacity 20
```

```
allow-connections sip to sip
```

```
redundancy-group 1
```

```
fax protocol pass-through g711ulaw
```

```
sip
```

```
bind control source-interface GigabitEthernet0/0/0
```

```
bind media source-interface GigabitEthernet0/0/0
```

```
session refresh
```

```
asserted-id pai
```

```
early-offer forced
```

```
midcall-signaling passthru
```

```
privacy-policy 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 sip-profiles 1
  request INVITE sip-header Diversion modify "<sip:(....)@" "<sip:209xxx\1@"
  request INVITE sip-header From modify "<sip:(.*)@.*>"
"<sip:XXXXXXXXXX_CISCOC661@gw1.peerlessnetwork.io:5060>"
!
voice translation-rule 1
  rule 1 /9209XXXXXXX/ /209XXXXXXX/
!
voice translation-profile loopback
  translate called 1
!
voice-card 0/2
  no watchdog
!
voice-card 0/4
  no watchdog
!
no license feature hseck9
license udi pid ISR4331/K9 sn XXXXXXXXXXXX
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 Peerless CUBE WAN interface
ip address 192.65.79.XXX 255.255.255.128
negotiation auto
redundancy rii 10
redundancy group 1 ip 192.65.79.XXX exclusive
!
interface GigabitEthernet0/0/1
description Peerless CUBE LAN interface
ip address 10.71.8.32 255.255.255.0
negotiation auto
redundancy rii 20
redundancy group 1 ip 10.71.8.30 exclusive
!
```



```
interface GigabitEthernet0/0/2
  description Peerless CUBE HA Interface
  ip address 10.71.9.26 255.255.255.0
  negotiation auto
!
interface Service-Engine0/2/0
!
interface Service-Engine0/4/0
!
interface GigabitEthernet0
  vrf forwarding Mgmt-intf
  no ip address
  negotiation auto
!
ip forward-protocol nd
no ip http server
no ip http secure-server
ip route 0.0.0.0 0.0.0.0 192.65.79.xxx
ip route 10.64.0.0 255.255.0.0 10.64.1.1
ip route 10.71.8.0 255.255.255.0 10.64.1.1
ip route 10.71.9.0 255.255.255.0 10.64.1.1
ip route 172.16.0.0 255.255.0.0 10.71.8.1
ip route 172.17.0.0 255.255.0.0 10.71.8.1
!
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 Call from CUCM to CUBE-LAN ***
  session protocol sipv2
  session transport udp
  incoming called-number .T
  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
  no fax-relay sg3-to-g3
  fax protocol pass-through g711ulaw
  no vad
!
dial-peer voice 101 voip
  description *** Outbound Call from CUBE-LAN to CUCM****
  destination-pattern 120.....
  session protocol sipv2
  session target ipv4:10.71.8.10:5060
  session transport udp
  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
  no fax-relay sg3-to-g3
  fax protocol pass-through g711ulaw
  no vad
!
dial-peer voice 200 voip
  description *** Inbound Call from Peerless to CUBE-WAN ***
```



```
session protocol sipv2
session transport udp
incoming called-number 120.....
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
no fax-relay sg3-to-g3
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 201 voip
description *** Outbound Call from CUBE-WAN to Peerless ****
translation-profile outgoing loopback
destination-pattern .T
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip profiles 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
no fax-relay sg3-to-g3
fax protocol pass-through g711ulaw
no vad
!
gateway
timer receive-rtcp 1200
!
sip-ua
```



```
credentials username XXXXXXXX_CISCO661 password 7 XXXXXX realm
reg.peerlessnetwork.io
registrar dns:gw1.peerlessnetwork.io:5060 expires 60
sip-server dns:gw1.peerlessnetwork.io:5060
!
line con 0
  stopbits 1
line aux 0
  stopbits 1
line vty 0 4
  exec-timeout 30 30
  password 7 XXXXXXXXXX
  login
!
!
End
```



Configuring Cisco UCM 12.5

Cisco UCM Version

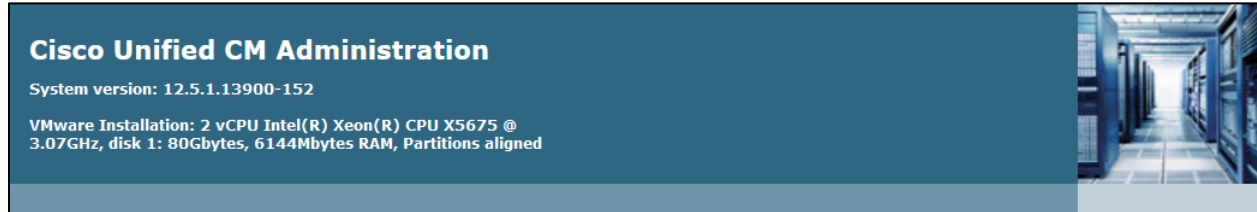


Figure 3: Cisco UCM Version

Cisco Call Manager Service Parameters

Navigation: System → Service Parameters

- Select Server* = cucm1-XXXXXX--CUCM Voice/Video (Active)
- Select Service* = Cisco CallManager (Active)
- All other fields are set to default values

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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

Service Parameter Configuration

Save Set to Default Advanced

Status
Status: Ready

Select Server and Service

Server* cucm1- --CUCM Voice/Video (Ac ▾
Service* Cisco CallManager (Active) ▾

All parameters apply only to the current server except parameters that are in the cluster-wide group(s).

Cisco CallManager (Active) Parameters on server cucm1-cucmpub.tekvoice.in--CUCM Voice/Video (Active)

Parameter Name	Parameter Value
Call Throttling	
Code Yellow Entry Latency *	20
Code Yellow Exit Latency Calculation *	40
Code Yellow Duration *	5 ▾
Max Events Allowed *	2000
System Throttle Sample Size *	10
Memory Throttling	
Enable Memory Throttling *	True ▾

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

Figure 4: Service Parameters



SIP Trunk Security Profile

Navigation: System → Security → SIP Trunk Security Profile

- Name* = **Peerless Network SIP Trunk Profile** 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**

Figure 5: SIP Trunk Security Profile



SIP Profile

Navigation: Device → Device Settings → SIP Profile

- Name* = **Peerless SIP Profile** is used as an example
- Description = **Peerless SIP Profile** is used as an example

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 Profile Information

Name* Peerless SIP Profile
Description Peerless 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)

Parameters used in Phone

Timer Invite Expires (seconds)* 180

Figure 6: SIP Profile



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	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

Figure 7: SIP Profile (Cont.)

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				
☒ Conference Join Enabled					
☐ RFC 2543 Hold					
☒ Semi Attended Transfer					
☐ Enable VAD					
☐ Stutter Message Waiting					
☐ MLPP User Authorization					
Normalization Script					
Normalization Script < None >					
☐ Enable Trace					
1	<table><tr><th>Parameter Name</th><th>Parameter Value</th></tr><tr><td></td><td></td></tr></table>	Parameter Name	Parameter Value		
Parameter Name	Parameter Value				
External Presentation Information					
☐ Anonymous External Presentation					
External Presentation Number					
External Presentation Name					

Figure 8: SIP Profile (Cont.)



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)

☐ Enable ANAT

☐ Deliver Conference Bridge Identifier

☐ Enable External Presentation Name and Number

☐ 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

Figure 9: SIP Profile (Cont.)

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 10: 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 features a search bar with a dropdown for 'Device Name' and a 'Find' button. A red box highlights the 'Add New' button at the bottom left of the search area.

Figure 11: 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 11. The main content area is titled 'Trunk Configuration' and includes a 'Next' button with a green arrow. Below this 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)'). A red box highlights the 'Next' button at the bottom left of the form.

Figure 12: Add SIP Trunk Type



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

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	Trunk_to_CUBE_Peerless
Description	Trunk_to_CUBE_Peerless
Device Pool*	Peerless_DevicePool
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	< None >
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. Failure to do so will expose keys and 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

Figure 13: SIP Trunk to Cisco UBE



☒ Run On All Active Unified CM Nodes

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](#)

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



Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	☒

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 15: SIP Trunk to Cisco UBE (Cont.)

- 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



SIP Information

Destination

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port	Status	Stat
1 *	10.71.8.30		5060	up	

MTP Preferred Originating Codec*
711ulaw

BLF Presence Group*
Standard Presence group

SIP Trunk Security Profile*
Peerless trunk security profile

Rerouting Calling Search Space
< None >

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

SUBSCRIBE Calling Search Space
< None >

SIP Profile*
Peerless SIP Profile [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

Save

Delete

Reset

Add New

Figure 16: SIP Trunk to Cisco UBE (Cont.)



Routing configuration

Route Pattern for Cisco UBE:

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

The screenshot shows the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title 'Cisco Unified CM Administration', and a navigation dropdown menu. Below the navigation bar is a secondary menu with options like 'System', 'Call Routing', 'Media Resources', etc. The main content area is titled 'Find and List Route Patterns'. It features a '+ Add New' button at the top left. Below this is a search section with a dropdown menu set to 'Pattern' and a 'begins with' filter. A 'Find' button is next to the search input. At the bottom of the search section, a message states 'No active query. Please enter your search criteria using the options above.' Below this message, the 'Add New' button is highlighted with a red rectangular box.

Figure 17: Add New Route Pattern for Cisco UBE

- Route Pattern= **9.@** (Create Route patterns based on the dial plan requirement)
- Description = **Route Pattern for calls to CUBE - Peerless**
- Gateway/Route List = **Trunk_to_CUBE_Peerless** (Associate the SIP Trunk created earlier)

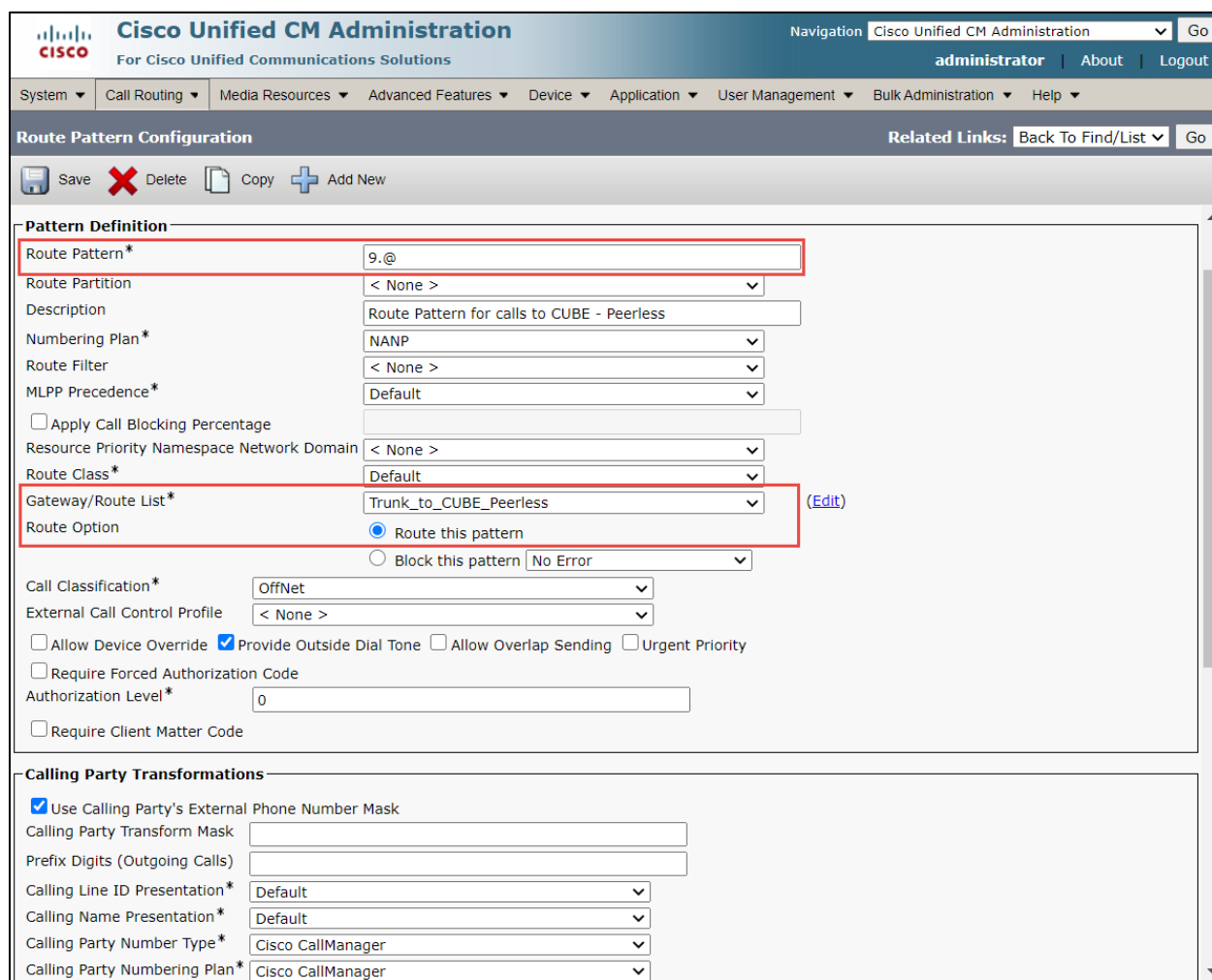


Figure 18: Route Pattern for PSTN dialing



- Discard Digits **PreDot**

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

*- Indicates required item.

Figure 19: Route Pattern for PSTN dialing (Cont.)



Fax Configuration in Cisco ATA SPA112

Quick Setup

- Access the IP address of Cisco ATA SPA112 using the web browser
- Enter the credentials to authenticate
- Click on **Quick Setup**
- **Line 1:** 10.71.8.10 (Enter the IP address of Cisco UCM)
- **User ID:** 209XXXXXXX (End User Configured in Cisco UCM)
- **Dial Plan:** Use the default values
- Click on **Submit** to save the configuration

The screenshot shows the 'Quick Setup' page of the Cisco Phone Adapter Configuration Utility. The page has a blue header with the Cisco logo and the title 'Phone Adapter Configuration Utility'. Below the header is a navigation bar with tabs: 'Quick Setup' (selected), 'Network Setup', 'Voice', 'Administration', and 'Status'. The main content area is titled 'Quick Setup' and contains two sections, 'Line 1' and 'Line 2'. Each section has fields for 'Proxy', 'Display Name', 'Password', 'Dial Plan', and 'User ID'. In the 'Line 1' section, the 'Proxy' field is set to '10.71.8.10:5060', the 'Display Name' is '9663', and the 'User ID' is '209XXXXXXX'. The 'Dial Plan' field contains a default value: '(*xx|[3469]11|0|00|[2-9]xxxxxx|1xxx[2-9]xxxxxxS0)xxxxxxxxxxxxxx'. The 'Submit' button is located at the bottom of the page, along with 'Cancel' and 'Refresh' buttons. Red boxes are drawn around the 'Proxy' field, 'Display Name' field, 'User ID' field, and the 'Submit' button.

Figure 20: Fax-Quick Setup



Voice

- Click on the tab **Voice**
- Click on **Line 1** from the menu options available in the left panel
- In the SIP Settings section ensure the following and the rest can be set to default values,
 - **SIP Transport:** UDP
 - **SIP Port:** 5060

The screenshot displays the Cisco Phone Adapter Configuration Utility interface. The 'Voice' tab is selected in the top navigation bar. In the left-hand menu, 'Line 1' is highlighted under the 'Regional' section. The main content area shows the configuration for 'Line 1'. The 'SIP Settings' section is expanded, showing various parameters. Two specific settings are highlighted with red boxes: 'SIP Transport' is set to 'UDP' and 'SIP Port' is set to '5060'. Other settings include SIP 100REL Enable (no), Auth Resync-Reboot (no), SIP Remote-Party-ID (yes), RTP Log Intvl (0), Referor Bye Delay (4), Referee Bye Delay (0), Sticky 183 (no), Reply 182 On Call Waiting (no), Use Local Addr In FROM (no), EXT SIP Port (empty), SIP Proxy-Require (empty), SIP GUID (no), Restrict Source IP (no), Refer Target Bye Delay (0), Refer-To Target Contact (no), Auth INVITE (no), Use Anonymous With RPID (yes), Broadsoft ALTC (no), Blind Attn-Xfer Enable (no), Xfer When Hangup Conf (yes), Conference Bridge Ports (3), Emergency Number (empty), Feature Key Sync (no), MOH Server (empty), Conference Bridge URL (empty), Enable IP Dialing (no), Mailbox ID (empty), Proxy (10.71.8.10:5060), Outbound Proxy (empty), Use Outbound Proxy (no), Use OB Proxy In Dialog (yes), Register (yes), and Make Call Without Reg (yes).

Figure 21: Fax-Line Setup-SIP Settings



- In the **Audio Configuration** section ensure the following and the rest can be set to default values,
 - **Preferred Code:** G711u/G729 (Depending on the Fax codec requirements)
 - **Silence Supp Enable:** no
 - **FAX Passthru Codec:** G711u
 - **FAX Enable T38:** yes/no (Depending on the Fax codec requirements)

The screenshot displays the Cisco Phone Adapter Configuration Utility interface. The 'Voice' tab is selected, and the 'Line 1' configuration page is shown. The 'Audio Configuration' section is expanded, revealing various settings. The 'Preferred Code' is set to 'G729a', and the 'FAX Passthru Method' is set to 'ReINVITE'. The 'Submit' button at the bottom left is highlighted with a red box. The 'Dial Plan' section shows a default pattern: (*xx[3469]11[0]00[2-9]xxxxxx1xxx[2-9]xxxxxxS0)xxxxxxxxxxxxxx. The 'FXS Port Polarity Configuration' section shows 'Idle Polarity' and 'Caller Conn Polarity' both set to 'Forward'.

Setting	Value
Preferred Code	G729a
Third Preferred Code	Unspecified
Codec Negotiation	Default
Silence Supp Enable	no
Silence Threshold	medium
Echo Canc Enable	yes
FAX Passthru Codec	G711u
DTMF Process INFO	yes
DTMF Process AVT	yes
DTMF Tx Method	Auto
DTMF Tx Mode	Strict
FAX Enable T38	yes
FAX T38 Redundancy	0
FAX Tone Detect Mode	caller or callee
FAX T38 Return to Voice	no
RTP to Proxy in Remote Hold	no
Second Preferred Code	Unspecified
Use Pref Codec Only	yes
G729a Enable	yes
G726-32 Enable	no
FAX V21 Detect Enable	yes
FAX CNG Detect Enable	yes
FAX Codec Symmetric	yes
FAX Passthru Method	ReINVITE
FAX PROCESS NSE	no
FAX Disable ECAN	no
DTMF Tx Strict Hold Off Time	70
Hook Flash Tx Method	None
FAX T38 ECM Enable	yes
Symmetric RTP	no
Modem Line	no

Dial Plan
Dial Plan: (*xx[3469]11[0]00[2-9]xxxxxx1xxx[2-9]xxxxxxS0)xxxxxxxxxxxxxx)

FXS Port Polarity Configuration
Idle Polarity: Forward
Caller Conn Polarity: Forward
Callee Conn Polarity: Forward

Buttons: Submit, Cancel, Refresh

Figure 22: Fax-Line Setup-Audio Configuration



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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