Connecting Cisco Unified Communication Manager [12.5.1] to OpenIP All IP SIP Trunks via Cisco Unified Border Element v12.2.0 [IOS-XE 16.09.01]

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Introduction

Service Providers today, such as OpenIP, 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.

OpenIP 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 OpenIP network, Cisco Unified Border Element (CUBE) 12.2 with ISR 4321/K9 running IOS-XE 16.9.1 can be used. The Cisco Unified Border Element 12.2 provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 12.5.1 connected to OpenIP 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 OpenIP 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 4321/K9 [IOS-XE 16.9.1] for connectivity to OpenIP SIP Trunking service. The deployment model covered in this application note is CPE (Cisco UCM) to PSTN (OpenIP) via Cisco UBE v12.2 [IOS-XE 16.9.1].

- Testing was performed in accordance to OpenIP 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 OpenIP 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 OpenIP SIP Trunking network.
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. OpenIP 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 OpenIP is UDP.

Cisco UCM and Cisco UBE Settings:

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Transport from Cisco UBE to Cisco UCM</td>
<td>UDP with RTP</td>
</tr>
<tr>
<td>Transport from Cisco UBE to OpenIP</td>
<td>UDP with RTP</td>
</tr>
<tr>
<td>Voice Mail Support</td>
<td>YES</td>
</tr>
<tr>
<td>Session Refresh</td>
<td>YES</td>
</tr>
<tr>
<td>Early Media support with PRACK</td>
<td>YES</td>
</tr>
</tbody>
</table>
System Components

Hardware Requirements
- Cisco UBE on Cisco ISR 4321 router
- CUCM cluster on UCS C240, 1 Publisher node and 1 Subscriber nodes
- Generic Cisco IP-Phones

Software Requirements
- CUBE-Version: 12.1 running IOS-XE 16.9.1
- CUCM UCOS 12.5.1.10000-22 for 1 Publisher and 1 Subscriber

Features

Features Supported
- Incoming and outgoing off-net calls using G711alaw and G729 voice codecs
- International Calls and digit manipulations
- Call Conference
- CPE Voice Mail support
- Call hold & Resume with and without MoH
- Unattended and Attended Call transfer
- Call forward (all, busy and no answer)
- DTMF (RFC2833)
- Fax (g711alaw pass-through)
- IP-PBX Calling number privacy
- High Availability

Features Not Supported
- Cisco UCM does not support Blind Call transfer
- OpenIP does not support the following related to Fax scenario,
  - SG3 fax transmission.
  - Fax with T38 codec.
Caveats

- Caller ID is not updated on attended and unattended transfer scenarios.
- Only one IP PBX used for the testing.
- OpenIP does not support Fax at Super G3 Speed and Fax with T38 codec.
- Open IP does not support g711ulaw codec.
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 CUBE LAN
  ip address 10.64.5.8 255.255.0.0
  negotiation auto
  redundancy rii 4
  redundancy group 2 ip 10.64.5.28 exclusive
!
interface GigabitEthernet0/0/1
  description CUBE WAN
  ip address 1XX.XX.XX.XX 255.255.255.128
  negotiation auto
  redundancy rii 3
  redundancy group 2 ip 1XX.XX.XX.XX exclusive
!
interface GigabitEthernet0/2/0
  description CUBE HA
  ip address 10.80.10.48 255.255.255.0
  negotiation auto
!
Cisco UBE 2:

interface GigabitEthernet0/0/0
   description CUBE LAN
   ip address 10.64.5.9 255.255.0.0
   negotiation auto
   redundancy rii 4
   redundancy group 2 ip 10.64.5.28 exclusive
!
interface GigabitEthernet0/0/1
   description CUBE WAN
   ip address 1XX.XX.XX.XX 255.255.255.128
   negotiation auto
   redundancy rii 3
   redundancy group 2 ip 1XX.XX.XX.XX exclusive
!
interface GigabitEthernet0/2/0
   description CUBE HA
   ip address 10.80.10.49 255.255.255.0
   negotiation auto
!
Global Cisco UBE settings

In order to enable Cisco UBE IP2IP SBC functionality, the following commands must be entered:

```
voice service voip
  no ip address trusted authenticate
  address-hiding
  mode border-element license capacity 20
  allow-connections sip to sip
  redundancy-group 2
  fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
  sip
    session refresh
    asserted-id pai
    privacy pstn
    early-offer forced
    midcall-signaling passthru
  g729 annexb-all
```

### Explanation

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>allow-connections sip to sip</td>
<td>Allow IP2IP connections between two SIP call legs</td>
</tr>
<tr>
<td>redundancy-group 2</td>
<td>Enable High Availability for the VoIP service</td>
</tr>
<tr>
<td>fax protocol</td>
<td>Specifies the fax protocol</td>
</tr>
<tr>
<td>asserted-id</td>
<td>Specifies the privacy header in the outgoing SIP requests and response messages</td>
</tr>
</tbody>
</table>
Codecs

G711alaw and G729 voice codecs are used for this testing. Codec preferences used to change according to the test plan description.

voice class codec 1
  codec preference 1 g729r8
  codec preference 2 g711alaw
  codec preference 3 g711ulaw

voice class codec 2
  codec preference 1 g711alaw
  codec preference 2 g729r8
  codec preference 3 g711ulaw
Dial peer

Outbound Dial-peer to OpenIP:

dial-peer voice 1 voip
description Ingress CUCM to PSTN - LAN Interface
session protocol sipv2
session transport udp
incoming called-number .T
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-nce
fax protocol pass-through g711alaw
no vad
!
dial-peer voice 2 voip
description Egress CUCM to PSTN - WAN Interface
destination-pattern .T
session protocol sipv2
session target dns:voip.myopenip.fr:5060
session transport udp
voice-class codec 2
voice-class sip options-ping 60
voice-class sip profiles 101
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nce
fax protocol pass-through g711alaw
no vad
!
Inbound Dial-peer from OpenIP:

dial-peer voice 3 voip
description openip PSTN to CUCM - WAN Interface
session protocol sipv2
session transport udp
incoming called-number 97....... 
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 protocol pass-through g711alaw
no vad
!
dial-peer voice 4 voip
description openip PSTN to CUCM - LAN Interface
destination-pattern 97....... 
session protocol sipv2
session target ipv4:10.80.12.2
session transport udp
voice-class codec 2
voice-class sip options-ping 60
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 protocol pass-through g711alaw
no vad
!
Configuration example

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

Active Cisco UBE:

version 16.9
service timestamps debug datetime msec
service timestamps log datetime msec
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname OpeniPCUBE1
!
boot-start-marker
boot system flash isr4300-universalk9.16.09.01.SPA.bin
boot-end-marker
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
enable secret 5 $1$OhVv$qYPCq3AtBxs6T6kiLSeL81
no aaa new-model
ip name-server 8.8.8.8
ip admission watch-list expiry-time 0
!
subscriber templating
!
multilink bundle-name authenticated
!
crypto pki trustpoint TP-self-signed-1582728230
  enrollment selfsigned
  subject-name cn=IOS-Self-Signed-Certificate-1582728230
  revocation-check none
  rsakeypair TP-self-signed-1582728230
!
crypto pki certificate chain TP-self-signed-1582728230
  certificate self-signed 01
    30820330 30820218 A0030201 02020101 300D0609 2A864886 F70D0101 05050030
    31312F30 2D060355 04031326 494F532D 53656C66 2D369676 6E65642D 43657274
    69666966 6174652D 31353832 37323832 33303031 170D3139 30363034 33313831
    33315A17 0D333030 31303030 305A3031 312F302D 06035504 03132649
    4F532D53 656C662D 53656C66 2D369676 6E65642D 43657274
    32383233 30308201 22300D06 092A8648 86F70D01 01010500 0382010F 00308201
    0A028201 0100BF53 1863559D 9E9AA5F5 78A65131 1B752C08 7FFBC2E2 BB438D7F
    ADE41EA3 FD3D74E7 29C085C7 00D25EE 40502487 DB8F1071 6BCB5A3F 2B796E14
    300AEC73 AB770F3D BC12B5FF 30829AC1 6B3BD338 1B077920 7847B3F3 E69F0D2C
    2BA8B15B 1D4DA010 50020018 49C1E6F5 AD8A484A 1DACA6B7 23FFB678 5B3F3450C
    A944DDE4 577894A9 D8274F78 F4A2DB03 7966BBAC EBD00DC9 E8F613E5 13F2098A
    99780361 0BBDA5AB 773F0168 90D67459 0E5807EF 4BE49CBA EA66725F CD13CC76
    C0B6E360 068A16B1 3A1AE0E4 943F22EB EAE4947E 8012D44C F261CB0F 54E461B3
    3F677D2A 4CDEF535 4CFE7835 7F409FD6 21CB48C5 CFF2A46D 3ACE0A0F 74C1F789
    2ACCD057 213F0203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF
voice service voip

no ip address trusted authenticate

address-hiding

mode border-element license capacity 20

allow-connections sip to sip

redundancy-group 2

fax protocol pass-through g711alaw

sip

bind control source-interface GigabitEthernet0/0/1

session refresh

asserted-id pai

privacy pstn

early-offer forced

midcall-signaling passthru

privacy-policy passthru

g729 annexb-all
voice class codec 1
  codec preference 1 g729r8
  codec preference 2 g711alaw
  codec preference 3 g711ulaw

voice class codec 2
  codec preference 1 g711alaw
  codec preference 2 g729r8
  codec preference 3 g711ulaw

voice class sip-profiles 101
  request INVITE sip-header Diversion modify "<sip:8(...)@" "<sip:970198\1@"

license udi pid ISR4321/K9 sn FA019220MSQ
license accept end user agreement
license boot suite AdvUCSuiteK9
license boot level appxk9
license boot level securityk9
no license smart enable
diagnostic bootup level minimal

username cisco privilege 15 password 0 XXXXXXXX

redundancy
  mode none
  application redundancy
group 2
  name openip
  priority 100 failover threshold 75
  timers delay 30 reload 60
  control GigabitEthernet0/2/0 protocol 1
  data GigabitEthernet0/2/0
  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
  ip address 10.64.5.8 255.255.0.0
  negotiation auto
  redundancy rii 4
  redundancy group 2 ip 10.64.5.28 exclusive

interface GigabitEthernet0/0/1
  description CUBE WAN
  ip address 1XX.XX.XX.XX 255.255.255.128
  negotiation auto
  redundancy rii 3
  redundancy group 2 ip 1XX.XX.XX.XX exclusive

interface GigabitEthernet0/2/0
  description CUBE HA
  ip address 10.80.10.48 255.255.255.0
negotiation auto
!
interface GigabitEthernet0
  vrf forwarding Mgmt-intf
  no ip address
  shutdown
  negotiation auto
!
ip forward-protocol nd
ip http server
ip http authentication local
ip http secure-server
ip tftp source-interface GigabitEthernet0
ip route 0.0.0.0 0.0.0.0 1XX.XX.XX.XX
ip route 10.64.0.0 255.255.0.0 10.64.1.1
ip route 10.80.12.0 255.255.255.0 10.64.1.1
ip route 172.16.24.0 255.255.248.0 10.64.1.1
!
control-plane
!
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
!
dial-peer voice 1 voip
  description Ingress CUCM to PSTN - LAN Interface
  session protocol sipv2
  session transport udp
  incoming called-number .T
  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-npe
  fax protocol pass-through g711alaw
  no vad
!
dial-peer voice 2 voip
  description Egress CUCM to PSTN - WAN Interface
  destination-pattern .T
  session protocol sipv2
  session target dns:voip.myopenip.fr:5060
  session transport udp
  voice-class codec 1
  voice-class sip options-ping 60
  voice-class sip profiles 101
  voice-class sip bind control source-interface GigabitEthernet0/0/1
  voice-class sip bind media source-interface GigabitEthernet0/0/1
  dtmf-relay rtp-npe
  fax protocol pass-through g711alaw
  no vad
!
dial-peer voice 3 voip
  description openip PSTN to CUCM - WAN Interface
  session protocol sipv2
  session transport udp
  incoming called-number 97.......
  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-nre
  fax protocol pass-through g711alaw
  no vad
!
dial-peer voice 4 voip
  description openip PSTN to CUCM - LAN Interface
  destination-pattern 97....... 
  session protocol sipv2
  session target ipv4:10.80.12.2
  session transport udp
  voice-class codec 1
  voice-class sip options-ping 60
  voice-class sip bind control source-interface GigabitEthernet0/0/0
  voice-class sip bind media source-interface GigabitEthernet0/0/0
  dtmf-relay rtp-nre
  fax protocol pass-through g711alaw
  no vad
!

sip-ua
  credentials username BWr9dOQ4NFGMpgJN password 7030C431E0D5B0C671A503E
  realm voip.myopenip.fr
authentication username BWr9d0Q4NFGMpgJN password 7020E1C4E0052220A18172E realm anonymous.invalid

authentication username BWr9d0Q4NFGMpgJN password 7020E1C4E0052220A18172E realm voip.myopenip.fr

registrar dns:voip.myopenip.fr expires 60

! 

line con 0
    exec-timeout 0 0
    password xxxxxxx
    login local
    transport input none
    stopbits 1
line aux 0
    stopbits 1
line vty 0 4
    exec-timeout 0 0
    password xxxxxxx
    login local

! 

end
Standby Cisco UBE:

version 16.9
service timestamps debug datetime msec
service timestamps log datetime msec
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname OpenipCUBE2
!
boot-start-marker
boot system flash isr4300-universalk9.16.09.01.SPA.bin
boot-end-marker
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
enable secret 5 $1$m56n$s28GnogC91n1sa6Bxt7sal
no aaa new-model
!
ip name-server 8.8.8.8
ip admission watch-list expiry-time 0
subscriber templating
multilink bundle-name authenticated
!

crypto pki trustpoint TP-self-signed-1017057749
enrollment selfsigned
subject-name cn=IOS-Self-Signed-Certificate-1017057749
revocation-check none
rsakeypair TP-self-signed-1017057749
!

crypto pki certificate chain TP-self-signed-1017057749
certificate self-signed 01
30820330 30820218 A0030201 02020101 300D0609 2A864886 F70D0101 05050030
31312F30 2D060355 04031326 494F532D 53656C66 2D536967 6E65642D 43657274
6966696D 6174652D 31303137 30353737 3439300E 170D3139 30363034 31353532
31355A17 0D33030A 31303130 30303030 303A3031 312F302D 06035504 3132649
4F532D53 656C662D 53656C66 65642D43 65727469 66696361 74652D31 30313730
35373734 39308201 22300D06 092A8648 86F70D01 01010500 0382010F 00308201
0A028201 0100C4DA E9377E1E 6D105A2B D515C69E 22CC009B 46C15687 D3F1C010
61B1DB54 2878D73E 204ED530 48F991A5 6D0F08D5 F3638B6D 663A86B7 6D7C41C7
479025A2 09097DAB 79A0B7D7 472222B5 9ACB8653 1F47BFF5 F7A7D7D6 75B30B01
2A248B37 88DD22ED B2B60F00 E3EFE2A7 798DBFFD DE44F671 DF9E977C EDA5A5E9
2391B823 71067148 B3831C27 4DD5F44 3DE4A103 14C60BE4 93415748 3F3D44AE
BC3753F3 8AC88024 E9029E13 09167404 A49904D7 D002453C 2318A6DE B6BB50F2
BC89528E 50E4F8DA 6C21D3C1 8602BF99 64167694 7B5BFD11 05C2920 2785F345
52A36462 4AB50B30 93193E9C 9E711DD8 1203143C 21A1F85E DBC82294 580020E2
B97748E5 8BB30203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF
301F0603 551D2304 18301680 14E9F109 FFF0CAE 86A33886 AF2C94BB 46CBF705
D7301D06 03551DOE 04160414 E9F109FF FF0CA8E6 A33886AF 2C94BB46 CBF705D7
300D0609 2A864886 F70D0101 05050030 82010100 535394A6 58E0A69A 9522BDEB
8F94032E 8078E94E 9E72DE89 67AA64EA FD55415D 32438623 E154CBEC 744796CA
60FF9C63 E2E9ED6C C931076D 66268191 9BDE128B C96A1C87 9AF6AD35 46222EC8
8EA05B9E 46B37C6C D44087B9 138CBF23 1B350F63 031F6F2D D88B5877 98729AE7
quit

voice service voip
no ip address trusted authenticate
address-hiding
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 2
fax protocol pass-through g711alaw
sip
bind control source-interface GigabitEthernet0/0/1
session refresh
asserted-id pai
privacy pstn
early-offer forced
midcall-signaling passthru
privacy-policy passthru
g729 annexb-all
!
voice class codec 1
  codec preference 1 g729r8
  codec preference 2 g711alaw
  codec preference 3 g711ulaw

voice class codec 2
  codec preference 1 g711alaw
  codec preference 2 g729r8
  codec preference 3 g711ulaw

voice class sip-profiles 101
  request INVITE sip-header Diversion modify "<sip:8(...)@" "<sip:970198\1@"

license udi pid ISR4321/K9 sn FAO19220MQ9
license accept end user agreement
license boot suite AdvUCSuiteK9
license boot level appxk9
license boot level securityk9
no license smart enable
diagnostic bootup level minimal

username cisco privilege 15 password 0 xxxxxxxxx

redundancy
  mode none
  application redundancy
    group 2
name openip
priority 100 failover threshold 75
timers delay 30 reload 60
control GigabitEthernet0/2/0 protocol 1
data GigabitEthernet0/2/0
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
   ip address 10.64.5.9 255.255.0.0
   negotiation auto
   redundancy rii 4
   redundancy group 2 ip 10.64.5.28 exclusive
!
interface GigabitEthernet0/0/1
   description CUBE WAN
   ip address 1XX.XX.XX.XX 255.255.255.128
   negotiation auto
   redundancy rii 3
   redundancy group 2 ip 1XX.XX.XX.XX exclusive
!
interface GigabitEthernet0/2/0
   description CUBE HA
   ip address 10.80.10.49 255.255.255.0
   negotiation auto
! interface GigabitEthernet0
  vrf forwarding Mgmt-intf
  no ip address
  negotiation auto
!
  ip forward-protocol nd
  ip http server
  ip http authentication local
  ip http secure-server
  ip http client source-interface GigabitEthernet0/0/0
  ip route 0.0.0.0 0.0.0.0 1XX.XX.XX.XX
  ip route 10.64.0.0 255.255.0.0 10.64.1.1
  ip route 10.80.12.0 255.255.255.0 10.64.1.1
  ip route 172.16.24.0 255.255.248.0 10.64.1.1
!
! control-plane
!
  mgcp behavior rsip-range tgcp-only
  mgcp behavior comedia-role none
  mgcp behavior comedia-check-media-src disable
  mgcp behavior comedia-sdp-force disable
!
  mgcp profile default
!
!
dial-peer voice 1 voip
  description Ingress CUCM to PSTN - LAN Interface
  session protocol sipv2
  session transport udp
  incoming called-number .T
  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-npe
  fax protocol pass-through g711a
  no vad
!
dial-peer voice 2 voip
  description Egress CUCM to PSTN - WAN Interface
  destination-pattern .T
  session protocol sipv2
  session target dns:voip.myopenip.fr:5060
  session transport udp
  voice-class codec 1
  voice-class sip options-ping 60
  voice-class sip profiles 101
  voice-class sip bind control source-interface GigabitEthernet0/0/1
  voice-class sip bind media source-interface GigabitEthernet0/0/1
  dtmf-relay rtp-npe
  fax protocol pass-through g711a
  no vad
!
dial-peer voice 3 voip
  description openip PSTN to CUCM - WAN Interface
session protocol sipv2
session transport udp
incoming called-number 97.......
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 protocol pass-through g711alaw
no vad
!
dial-peer voice 4 voip
description openip PSTN to CUCM - LAN Interface
destination-pattern 97.......
session protocol sipv2
session target ipv4:10.80.12.2
session transport udp
voice-class codec 1
voice-class sip options-ping 60
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 protocol pass-through g711alaw
no vad
!
!
sip-ua

credentials username BWr9dOQ4NFGMpgJN password 7 030C431E0D5B0C671A503E realm voip.myopenip.fr

authentication username BWr9dOQ4NFGMpgJN password 7 020E1C4E0052220A18172E realm anonymous.invalid

authentication username BWr9dOQ4NFGMpgJN password 7 020E1C4E0052220A18172E realm voip.myopenip.fr

registrar dns:voip.myopenip.fr expires 60

!

line con 0

exec-timeout 0 0
password xxxxxxx
login local
transport input none
stopbits 1

line aux 0
stopbits 1

line vty 0 4
exec-timeout 0 0
password xxxxxxx
login local
!
pnp profile pnp_cco_profile
transport https ipv4 52.203.231.173 port 443
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* = Clus22ub1--CUCM Voice/Video (Active)
- Select Service* = Cisco CallManager (Active)
- All other fields are set to default values

Figure 4: Service Parameters
SIP Trunk Security Profile

**Navigation:** System → Security → SIP Trunk Security Profile

- Name* = **Openip 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](image-url)
SIP Profile

**Navigation**: Device → Device Settings → SIP Profile

- Name* = **Openip SIP Profile** is used as an example
- Description = **Openip SIP Profile** is used as an example
Figure 6: SIP Profile
Figure 6: SIP Profile (Cont.)
### SIP Profile (Cont.)

**Trunk Specific Configuration**

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Reroute Incoming Request to new Trunk based on</td>
<td>Never</td>
</tr>
<tr>
<td>Resource Priority Namespace List</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>SIP 1xx Options</td>
<td>Send PRACK if 1xx Contains SDP</td>
</tr>
<tr>
<td>Video Call Traffic Class</td>
<td>Mixed</td>
</tr>
<tr>
<td>Calling Line Identification Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Session Refresh Method</td>
<td>Invite</td>
</tr>
<tr>
<td>Early Offer support for voice and video calls</td>
<td>Disabled (Default value)</td>
</tr>
<tr>
<td>Enable ANAT</td>
<td></td>
</tr>
<tr>
<td>Deliver Conference Bridge Identifier</td>
<td></td>
</tr>
<tr>
<td>Enable External Presentation Name and Number</td>
<td></td>
</tr>
<tr>
<td>Reject Anonymous Incoming Calls</td>
<td></td>
</tr>
<tr>
<td>Reject Anonymous Outgoing Calls</td>
<td></td>
</tr>
<tr>
<td>Send ILS Learned Destination Route String</td>
<td></td>
</tr>
<tr>
<td>Connect Inbound Call before Playing Queuing Announcement</td>
<td></td>
</tr>
</tbody>
</table>

**SIP OPTIONS Ping**

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

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ping Interval for In-service and Partially In-service Trunks (seconds)*</td>
<td>60</td>
</tr>
<tr>
<td>Ping Interval for Out-of-service Trunks (seconds)*</td>
<td>120</td>
</tr>
<tr>
<td>Ping Retry Timer (milliseconds)*</td>
<td>500</td>
</tr>
<tr>
<td>Ping Retry Count</td>
<td>6</td>
</tr>
</tbody>
</table>

**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

---

**Figure 6:** SIP Profile (Cont.)
Trunk configuration

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

Navigation: Device → Trunk → Add New

Figure 7: Add New Trunk to Cisco UBE

- Select ‘Trunk Type’ as SIP Trunk and ‘Device Protocol’ as SIP and select ‘Next’ as shown below.

Figure 8: Add SIP Trunk Type
Figure 9: SIP Trunk to Cisco UBE
Figure 9: SIP Trunk to Cisco UBE (Cont.)

- Configure the Virtual LAN IP address of the Cisco UBE and the Destination Port
- Configure the SIP Trunk Security Profile and SIP Profile as shown below
- The rest of the configuration is all default
- Click Save and Reset after completion
Figure 9: SIP Trunk to Cisco UBE (Cont.)
Figure 9: SIP Trunk to Cisco UBE (Cont.)
Routing configuration

Route Pattern for Cisco UBE:

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

Figure 10: Add New Route Pattern for Cisco UBE
### Pattern Definition

- **Route Pattern**: B.XXXXXXXX
- **Route Partition**: < None >
- **Description**: OPENIPNational_calls_route_pattern
- **Numbering Plan**: < Not Selected >
- **Route Filter**: < None >
- **MLPP Precedence**: Default
- **Apply Call Blocking Percentage**: False
- **Resource Priority Namespace Network Domain**: < None >
- **Route Class**: Default
- **Gateway/Route List**: OpenIP

#### Calling Party Transformations

- **Use Calling Party's External Phone Number Mask**: False
- **Calling Party Transform Mask**: 
- **Prefix Digits (Outgoing Calls)**: 
- **Calling Line ID Presentation**: Allowed
- **Calling Name Presentation**: Allowed
- **Calling Party Number Type**: Cisco CallManager
- **Calling Party Numbering Plan**: Cisco CallManager

#### Connected Party Transformations

- **Connected Line ID Presentation**: Default
- **Connected Name Presentation**: Default

#### Called Party Transformations

- **Discard Digits**: PreDot
- **Called Party Transform Mask**: 
- **Prefix Digits (Outgoing Calls)**: 
- **Called Party Number Type**: Cisco CallManager
- **Called Party Numbering Plan**: Cisco CallManager

#### ISDN Network-Specific Facilities Information Element

<table>
<thead>
<tr>
<th>Network Service Protocol</th>
<th>Service Parameter Name</th>
<th>Service Parameter Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>&lt; Not Selected &gt;</td>
<td>&lt; Not Exist &gt;</td>
<td></td>
</tr>
</tbody>
</table>
Figure 12: Route Pattern for International dialing
**Figure 13: Route Pattern for Anonymous dialing**

<table>
<thead>
<tr>
<th>Pattern Definition</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Route Pattern</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Route Partition</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>Description</strong></td>
<td>OPENIP_Anonymous_calls_route_pattern</td>
</tr>
<tr>
<td><strong>Numbering Plan</strong></td>
<td>-- Not Selected --</td>
</tr>
<tr>
<td><strong>Route Filter</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>MLPP Precedence</strong></td>
<td>Default</td>
</tr>
<tr>
<td><strong>Gateway/Route List</strong></td>
<td>OpenIP</td>
</tr>
<tr>
<td><strong>Route Option</strong></td>
<td>Route this pattern</td>
</tr>
<tr>
<td><strong>Call Classification</strong></td>
<td>OffNet</td>
</tr>
<tr>
<td><strong>External Call Control Profile</strong></td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

**Calling Party Transformations**

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

**Connected Party Transformations**

| **Connected Line ID Presentation** | Default |
| **Connected Name Presentation** | Default |

**Called Party Transformations**

| **Discard Digits** | Predet |
| **Called Party Transform Mask** |  |
| **Prefix Digits (Outgoing Calls)** |  |
| **Called Party Number Type** | Cisco CallManager |
| **Called Party Numbering Plan** | Cisco CallManager |

**ISDN Network-Specific Facilities Information Element**

| **Network Service Protocol** | -- Not Selected -- |
| **Carrier Identification Code** |  |
| **Network Service** | Service Parameter Name | Service Parameter Value |
| -- Not Selected -- | < Not Exist > |
Figure 14: Route Pattern for Emergency calls
Fax Configuration in Cisco ATA SPA112- G711a-Law Pass-through

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:** 10.80.12.2 (Enter the IP address of Cisco UCM)
- **User ID:** 8757 (End User Configured in Cisco UCM)
- **Dial Plan:** Use the default values
- Click on Submit to save the configuration

![Quick Setup Screenshot](image)

**Figure 15:** 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

![SIP Settings](image)

**Figure 16**: Fax-Line Setup-SIP Settings
• In the **Audio Configuration** section ensure the following and the rest can be set to default values,
  
  - **Preferred Code**: G711a
  - **Silence Supp Enable**: no
  - **FAX Passthru Codec**: G711a
  - **FAX Enable T38**: no

![Figure 17: Fax-Line Setup-Audio Configuration](image-url)
## Acronyms

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