Application Note

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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Table of Contents

Table of Figures

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.

Network Topology

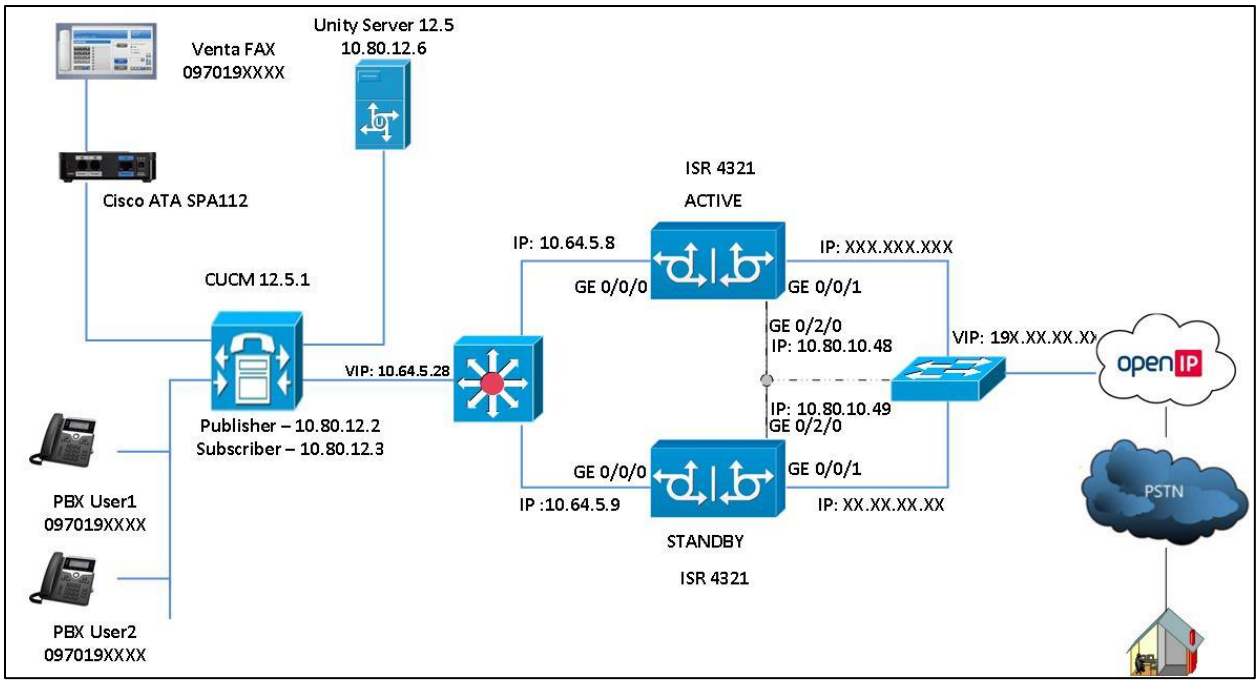

Figure 1 Network Topology

- 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. 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:

Setting	Value
Transport from Cisco UBE to Cisco UCM	UDP with RTP
Transport from Cisco UBE to OpenIP	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 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.

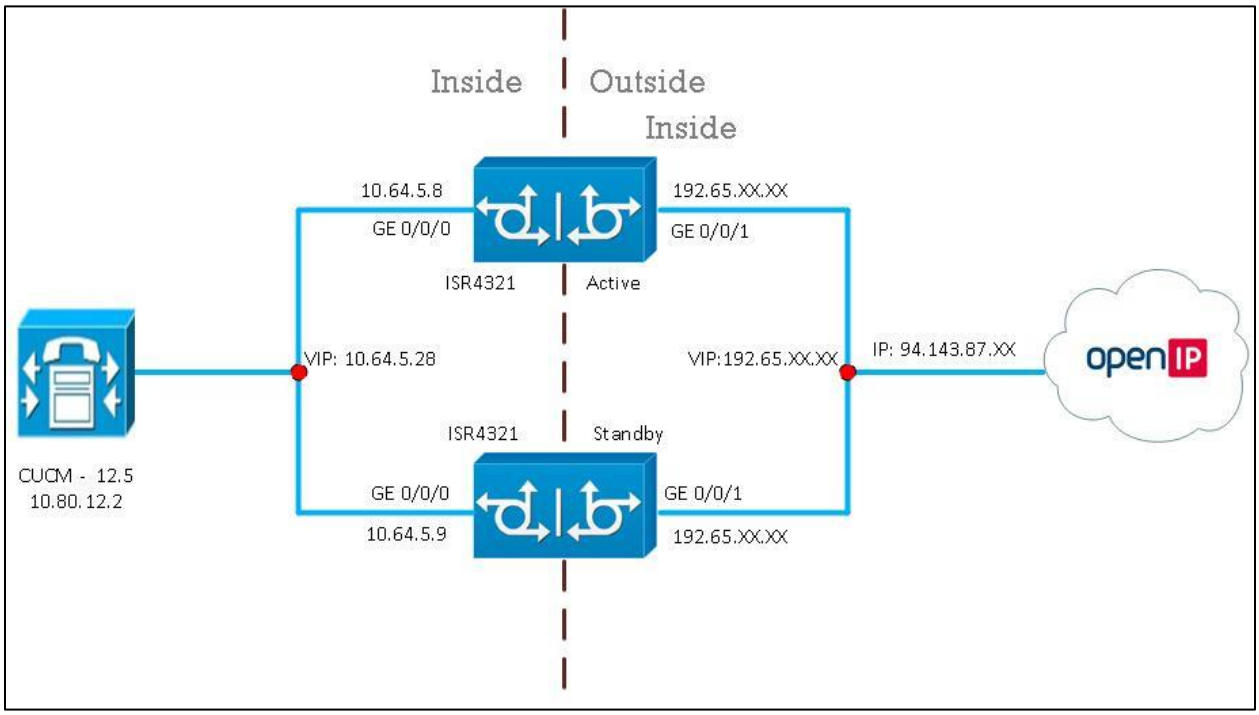

Figure 2 High Availability topology

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

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

Codecs

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-nte
 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-nte
 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 2D536967 6E65642D 43657274
  69666963 6174652D 31353832 37323832 3330301E 170D3139 30363034 31363134
  33315A17 0D333030 31303130 30303030 305A3031 312F302D 06035504 03132649
  4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D31 35383237
  32383233 30308201 22300D06 092A8648 86F70D01 01010500 0382010F 00308201
  0A028201 0100BF53 1863559D 9E9AAF55 78A65131 1B752C08 7FFBC2E2 BB438D7F
  ADE41EA3 FD3D74E7 29C085C7 000D25EE 40502487 DB8F1071 6BCB5A3F 2B796E14
  300AEC73 AB770F3D BC12B5FF 30829AC1 6B3BD338 1B077920 7847B3F3 E69F0D2C
  2BA8B15B 1D4DA010 50020018 49C1E6F5 AD8A484A 1DACA6B7 23FFB678 58F3450C
  A944DDE4 577894A9 D8274F78 F4A2DBD3 7966BBAC EBD00DC9 E8F613E5 13F209DA
  99780361 0BBDA5AB 773F016B 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
```

```
  301F0603 551D2304 18301680 14133543 EC7CE4D0 55C632C4 3B61DBE9 8E22652D
  49301D06 03551D0E 04160414 133543EC 7CE4D055 C632C43B 61DBE98E 22652D49
  300D0609 2A864886 F70D0101 05050003 82010100 6B4F7F1C 9B4EBEC1 AC603DF9
  CC273FBB 1CF9F9A7 749F8343 A26EDFFD 65BC662D 1EE6AED7 64BF2E99 69E0EC40
  A160572A 11FFE002 27BD0EF9 6F4E71CC 7B265975 62C19144 90099020 AB1673BE
  6B1E01AB 6932F12C 5797ADDD 9FD1EC4C 632DFA96 245F04AB 19931437 01EE09B0
  E33951AA 6E723B25 6E4910BE 05725226 E08DBE07 D4816496 3E577BF3 8D146327
  14F4E4E0 5ACE7B56 9B61D88F C55DE5CF A932D86C 5A368560 D54DBEAB 55111000
  C9982300 2C4E792C BCDFCEB9 CC437BC8 0D1D82D9 48D358D1 DDE66F48 83DC9FCC
  C8FDEFC5 10228211 5EE900FC E5742E9B 747C2677 F737B685 C66DA37C 59A80DB3
  CB282CC9 A77E70DA 9F762ED0 B8FAEF95 E78E96FA
        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 FAO19220MSQ
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
!
spanning-tree extend system-id
!
username cisco privilege 15 password 0 XXXXXXX
!
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-nte
 fax protocol pass-through g711alaw
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 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 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-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
!
!
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$s28GnogC9ln1sa6Bxt7sa1
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
  69666963 6174652D 31303137 30353737 3439301E 170D3139 30363034 31353532
  31355A17 0D333030 31303130 30303030 305A3031 312F302D 06035504 03132649
  4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D31 30313730
  35373734 39308201 22300D06 092A8648 86F70D01 01010500 0382010F 00308201
  0A028201 0100C4DA E9377E1E 6D105A2B D515C69E 22CC009B 46C156B7 D3F1C010
  61B1DB54 2878D73E 204ED530 48F991A5 6D0F08D5 F3638BD6 663A86B7 6D7C41C7
  479025A2 09097DAB 79A0B1D7 472222B5 9ACB8653 1F47BFF5 F7A7D7D6 75B30B01
  2A248B37 88DD22ED B2B6D0F0 E3EFE2A7 798DBFFD DE44F671 DF9E977C EDA5AE59
  2391B823 71067148 B3831C27 4DDB5F44 3DE4A103 14C60BE4 93415748 3FD344AE
  BC3753F3 8AC88024 E9029E13 09167404 A49904D7 D002453C 2318A6DE B6BB50F2
  BC89528E 50E4F8DA 6C21D3C1 8602BF99 64167694 7B5BFD11 05CA2920 27853F45
  52A36462 4AB50B30 93193E9C 9EA71DD8 1203143C 21A1F85E DBC82294 580020E2
  B97748E5 8B830203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF
  301F0603 551D2304 18301680 14E9F109 FFFF0CAE 86A33886 AF2C94BB 46CBF705
  D7301D06 03551D0E 04160414 E9F109FF FF0CAE86 A33886AF 2C94BB46 CBF705D7
  300D0609 2A864886 F70D0101 05050003 82010100 535394A6 58E0A69A 9522BDEB
  8F94D32E 8078E94E 9E72DE89 67AA64EA FD55415D 32438623 E154CBEC 744796CA
  60FF9C63 E2E9ED6C C931076D 66268191 9BDE128B C96A1C87 9AF6AD35 46222EC8
  8EA05B9E 46B37CBC D44E0B79 138CBF23 1B350F63 031F6F2D D8BB5877 98729AE7
```

```
  1B3DD064 37C2F86E E1616719 D522E5F5 51AA9E19 8E873F2D 65E13DB8 B96A9683
  15503EFC CAA282F0 EA0EF75C 264B3B78 A1695147 D012FBA4 5B09ED47 6A000ACE
  C0DDF420 43E35313 46551702 C0E923E0 B74DAC50 543F071D B31BB287 D73AC0C8
  58D805AA 0F501DA3 2AFB7111 08E79855 28D2AACD E5645A02 6D325D72 154B8207
  F6389A36 B3B28626 74C19A37 F7D72A68 FD6C8ADD
        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
!
spanning-tree extend system-id
!
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 10.64.5.9 255.255.0.0
 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-nte
 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-nte
 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-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 XXXXXX
 login local
 transport input none
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 exec-timeout 0 0
 password XXXXXX
 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

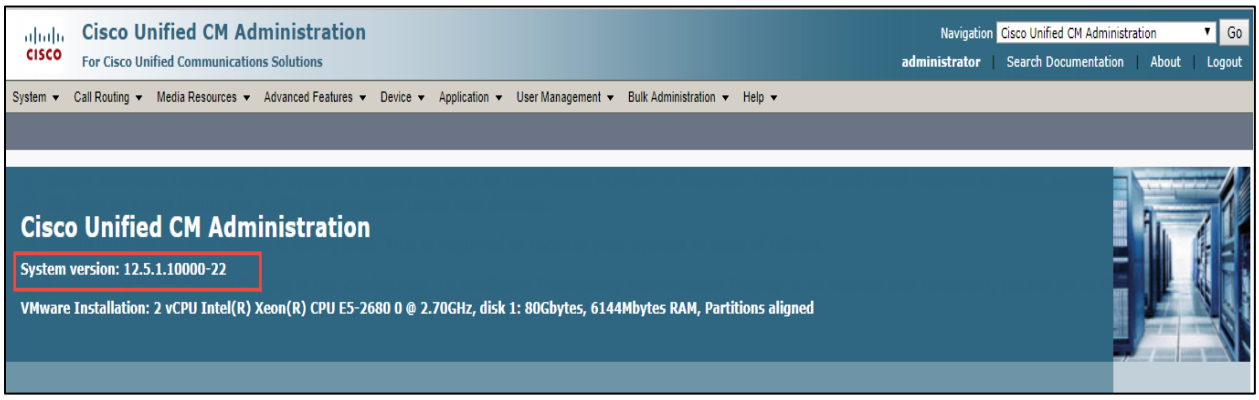

Figure 3: Cisco UCM Version

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

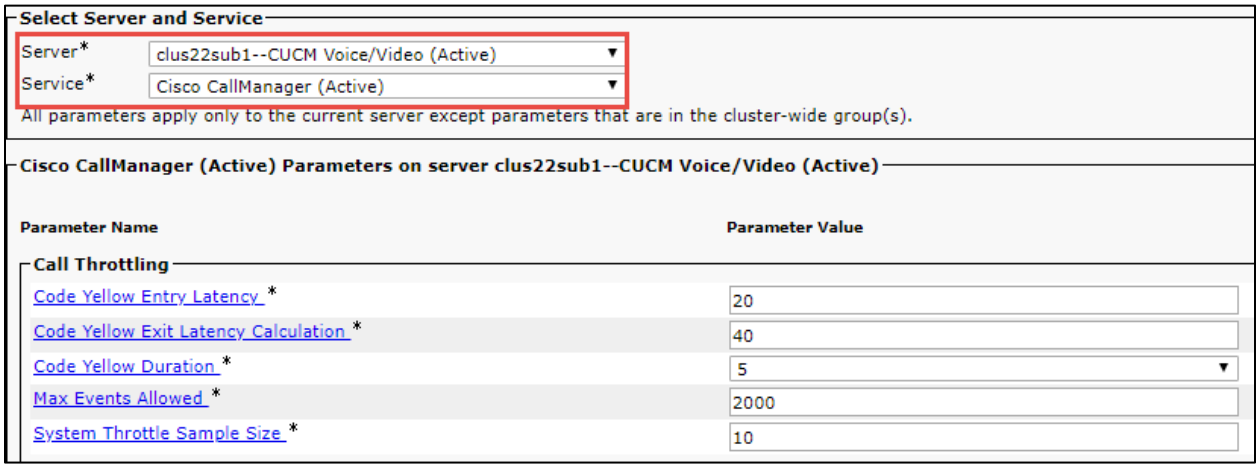

Figure 4: Service Parameters

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**

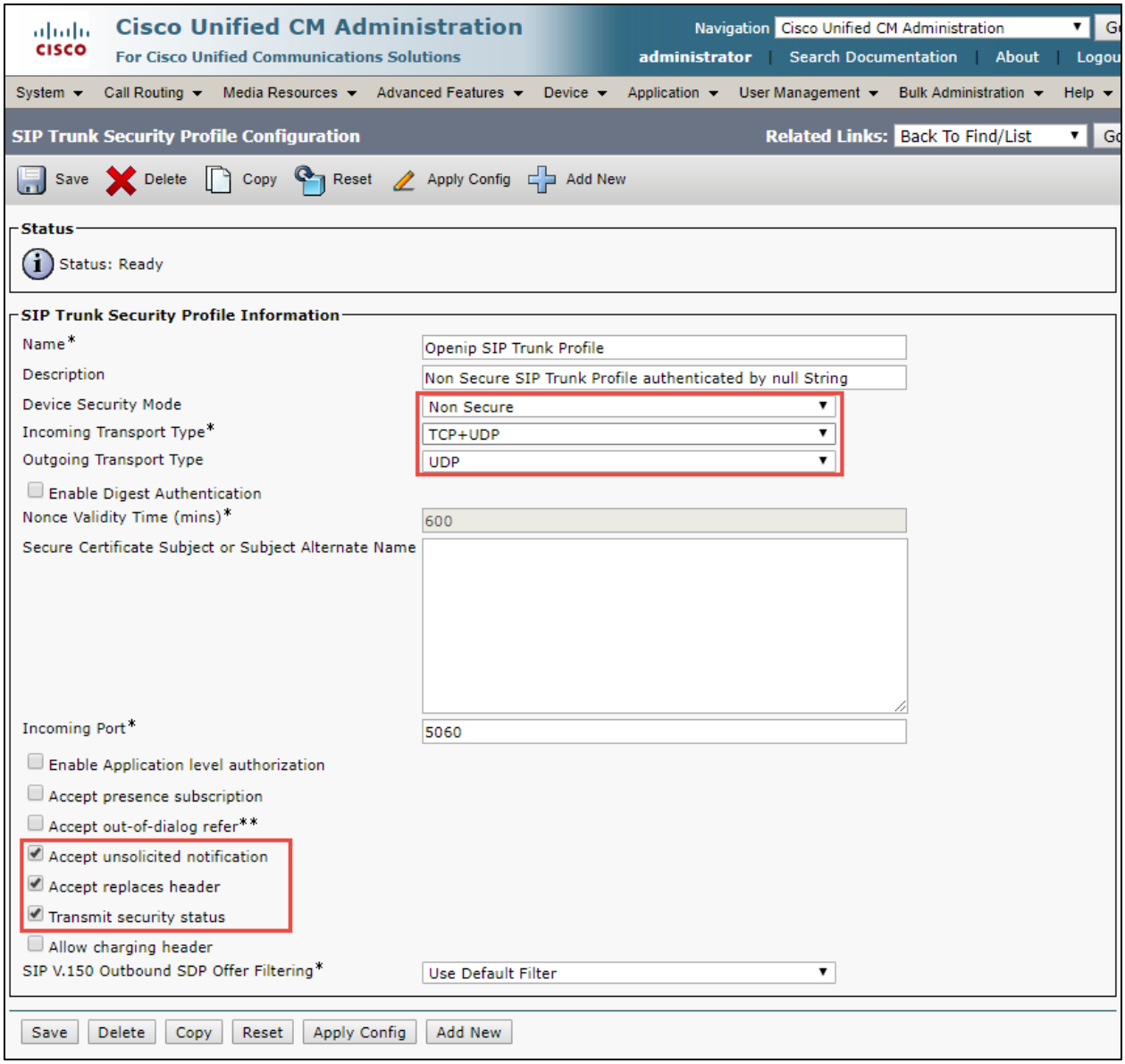

Figure 5:SIP Trunk Security Profile

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

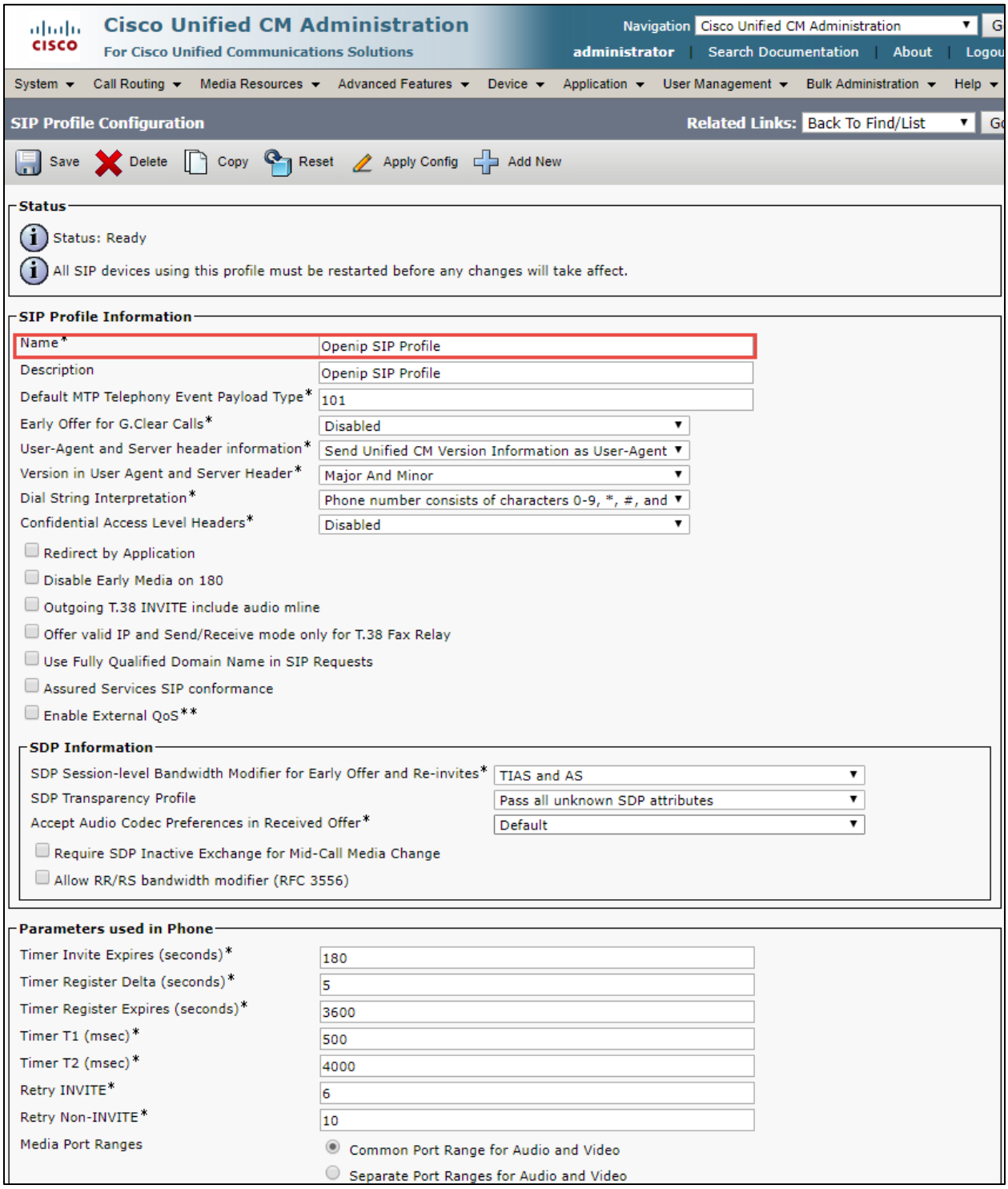

Figure 6: SIP Profile

Start Media Port*	16384
Stop Media Port*	32766
DSCP for Audio Calls	Use System Default
DSCP for Video Calls	Use System Default
DSCP for Audio Portion of Video Calls	Use System Default
DSCP for TelePresence Calls	Use System Default
DSCP for Audio Portion of TelePresence Calls	Use System Default
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off
Caller ID Blocking*	Off
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
Resource Priority Namespace	< None >
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70
Off Hook To First Digit Timer (milliseconds)*	15000
Call Forward URI*	x-cisco-serviceuri-cfwdall
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial

- [x] Conference Join Enabled
- [ ] RFC 2543 Hold
- [x] 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

Figure 6: SIP Profile (Cont.)

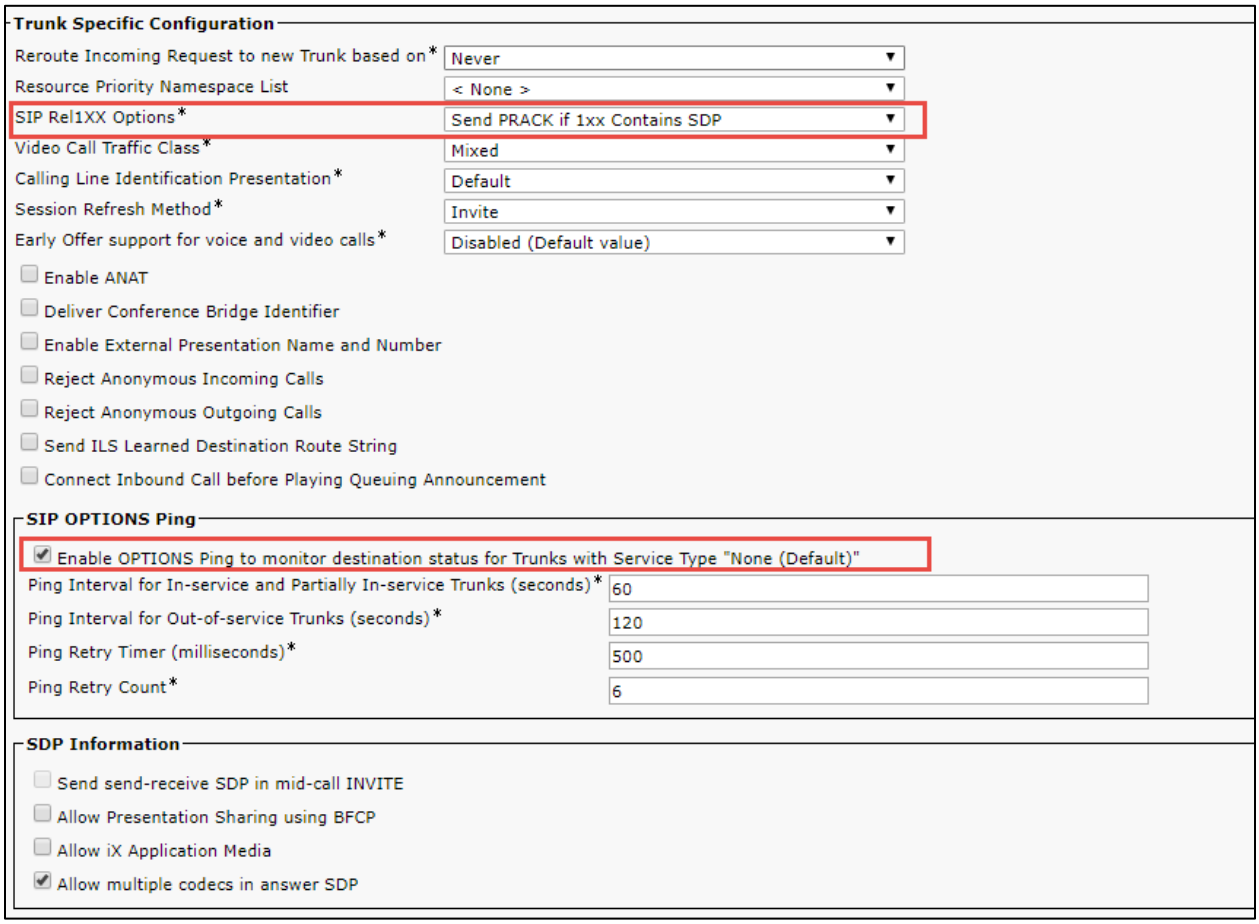

Figure 6: SIP Profile (Cont.)

Trunk configuration

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

Navigation: Device → Trunk → Add New

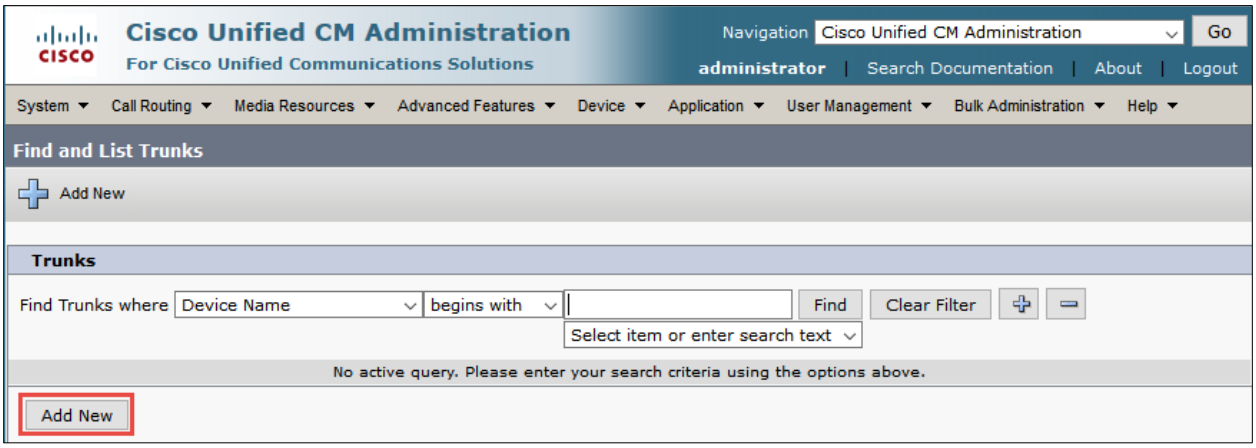

Figure 7: Add New Trunk to Cisco UBE

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

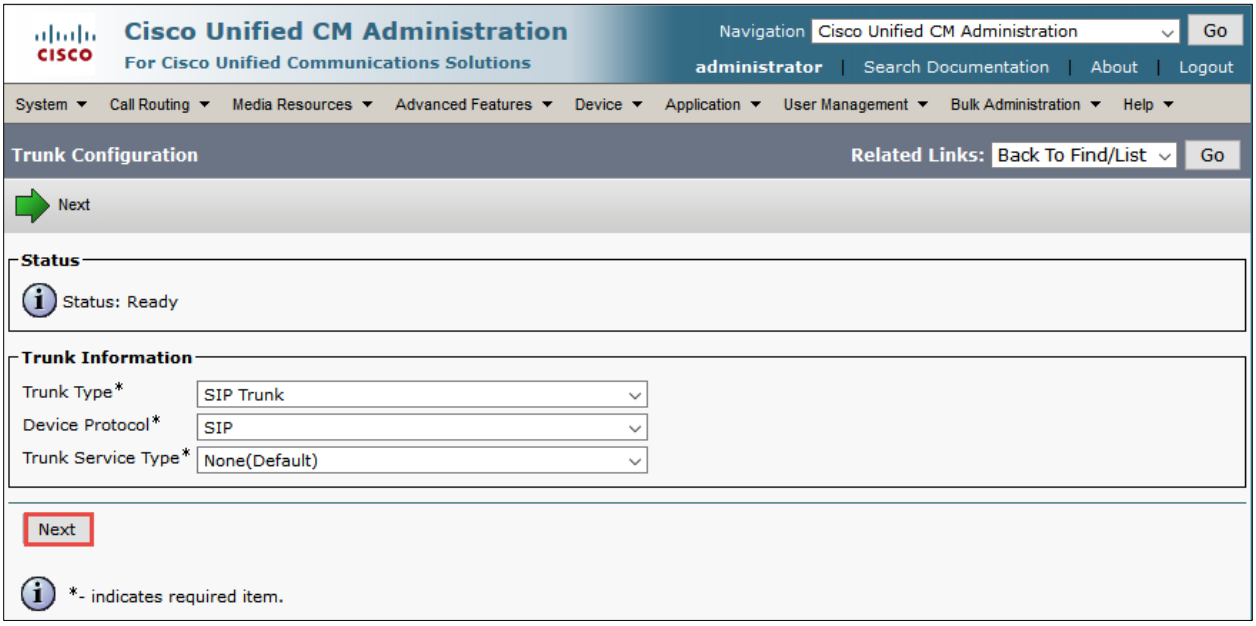

Figure 8: Add SIP Trunk Type

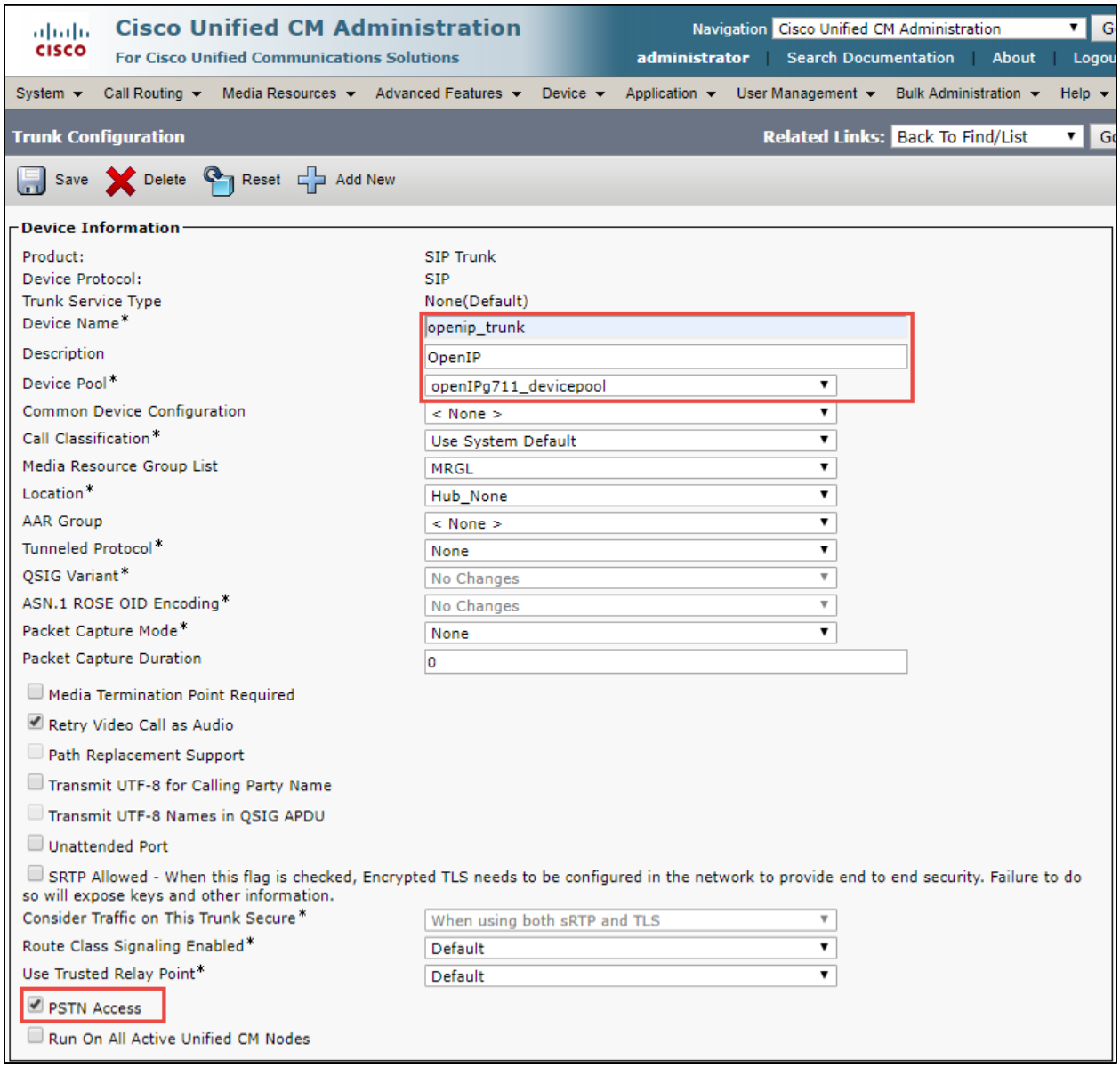

Figure 9: SIP Trunk to Cisco UBE

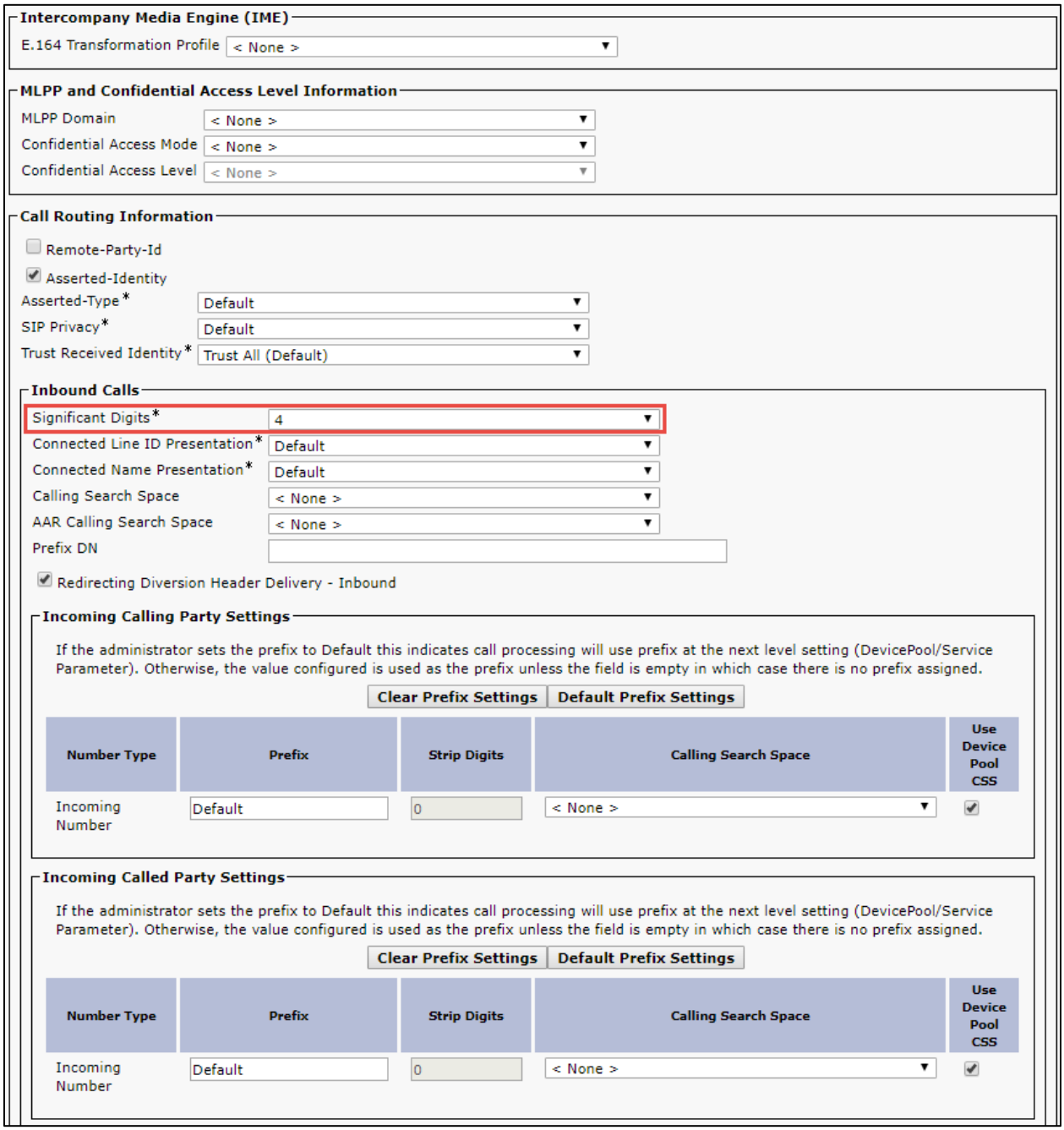

Figure 9: SIP Trunk to Cisco UBE (Cont.)

- 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

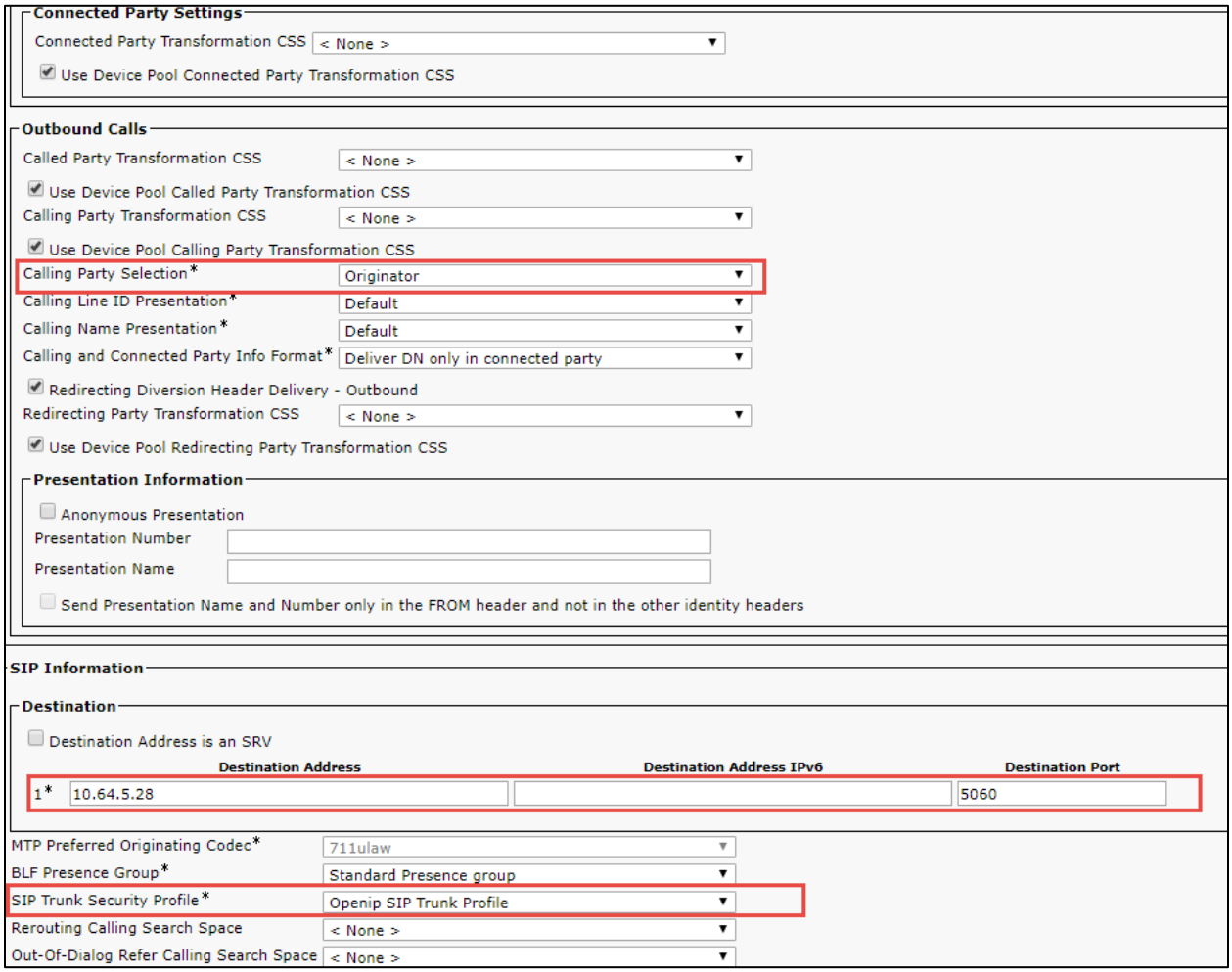

Figure 9: SIP Trunk to Cisco UBE (Cont.)

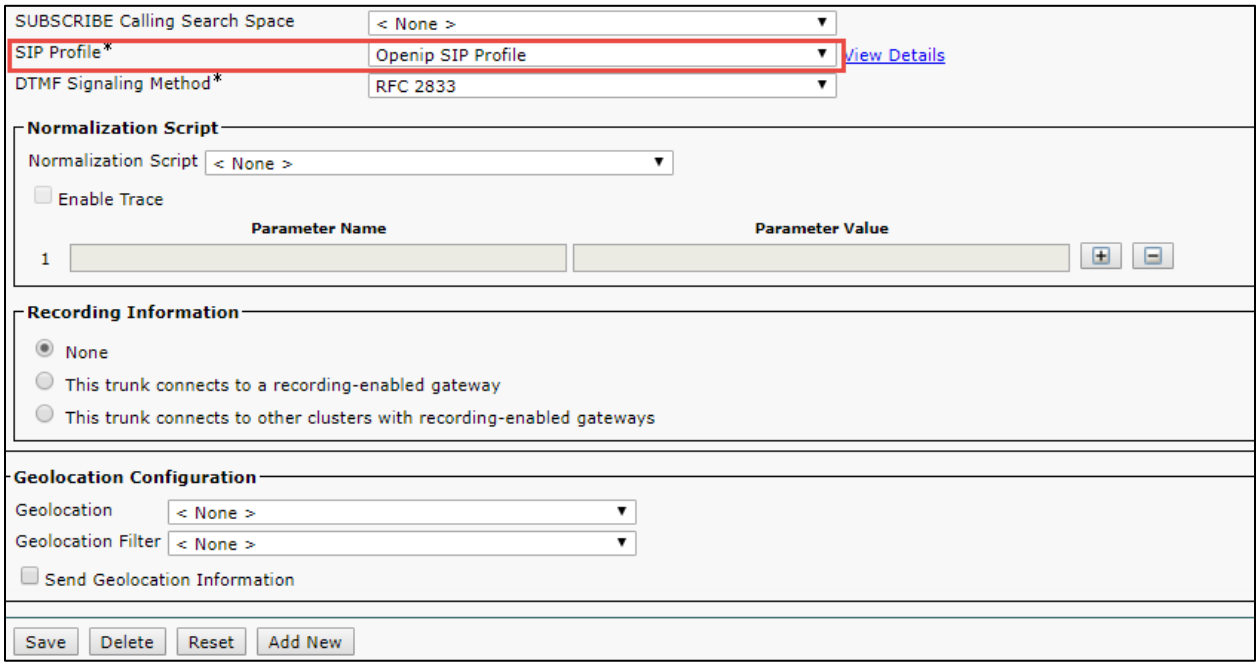

Figure 9: SIP Trunk to Cisco UBE (Cont.)

Routing configuration

Route Pattern for Cisco UBE:

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

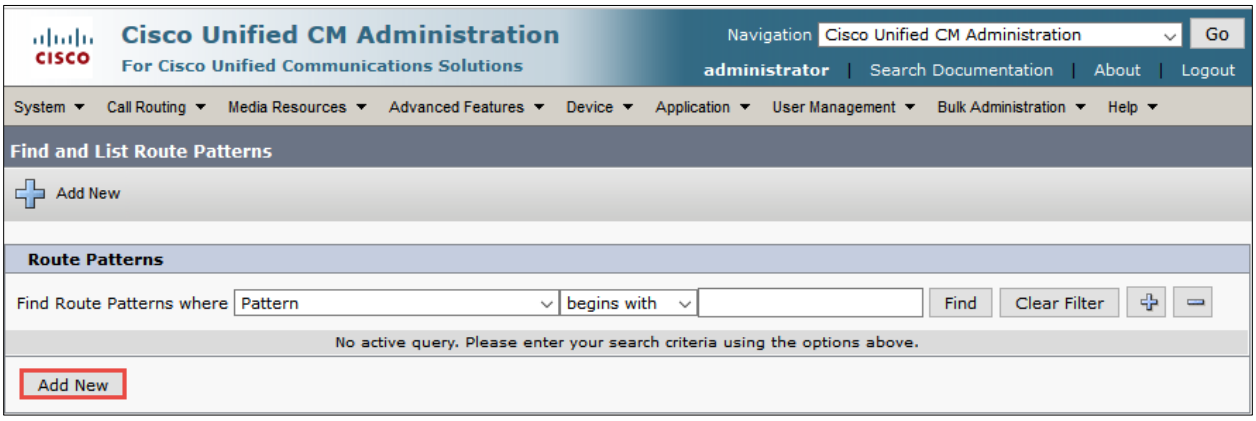

Figure 10: Add New Route Pattern for Cisco UBE

Pattern Definition

Route Pattern*	8.XXXXXXXXXX
Route Partition	< None >
Description	OPENIP_National_calls_route_pattern
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*	OpenIP (Edit)
Route Option	Route this pattern / Block this pattern No Error
Call Classification*	OffNet
External Call Control Profile	< None >

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

Require Forced Authorization Code

Authorization Level* 0

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask	
Prefix Digits (Outgoing Calls)	
Calling Line ID Presentation*	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

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

Figure 11:Route Pattern for National dialing

Pattern Definition

Route Pattern*	9.XXXXXXXXXXXXX
Route Partition	< None >
Description	OPENIP_International_calls_route_pattern
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*	OpenIP (Edit)
Route Option	Route this pattern / Block this pattern No Error
Call Classification*	OffNet
External Call Control Profile	< None >

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

Require Forced Authorization Code

Authorization Level* 0

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask	
Prefix Digits (Outgoing Calls)	
Calling Line ID Presentation*	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

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

Figure 12:Route Pattern for International dialing

Pattern Definition

Route Pattern* 6.XXXXXXXXXXXXX
Route Partition < None >
Description OPENIP_Anonymous_calls_route_pattern
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* OpenIP (Edit)
Route Option Route this pattern
Block this pattern No Error
Call Classification* OffNet
External Call Control Profile < None >
Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority
Require Forced Authorization Code
Authorization Level* 0
Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask
Calling Party Transform Mask
Prefix Digits (Outgoing Calls)
Calling Line ID Presentation* 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 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

Figure 13: Route Pattern for Anonymous dialing

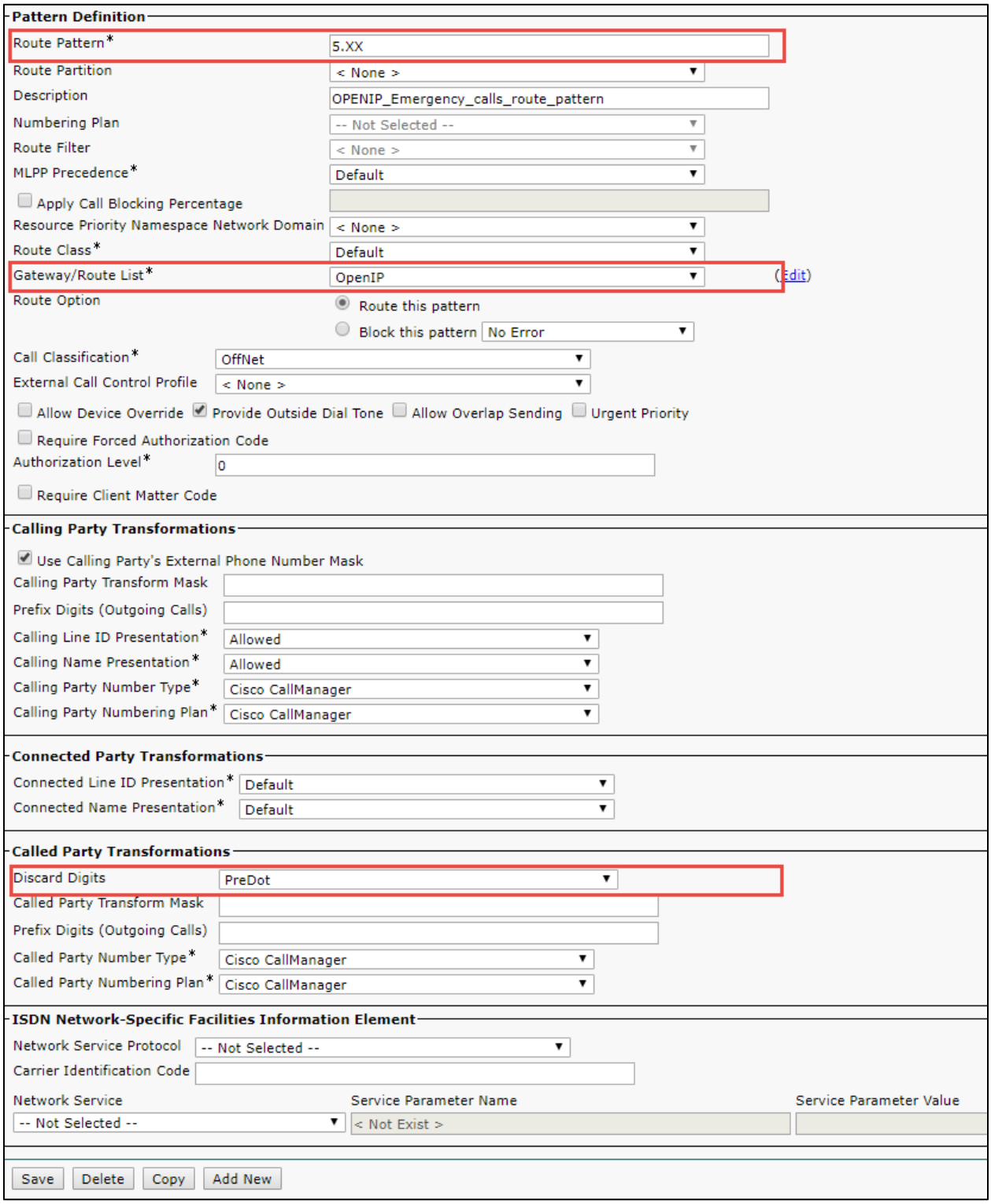

Figure 14: Route Pattern for Emergency calls

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

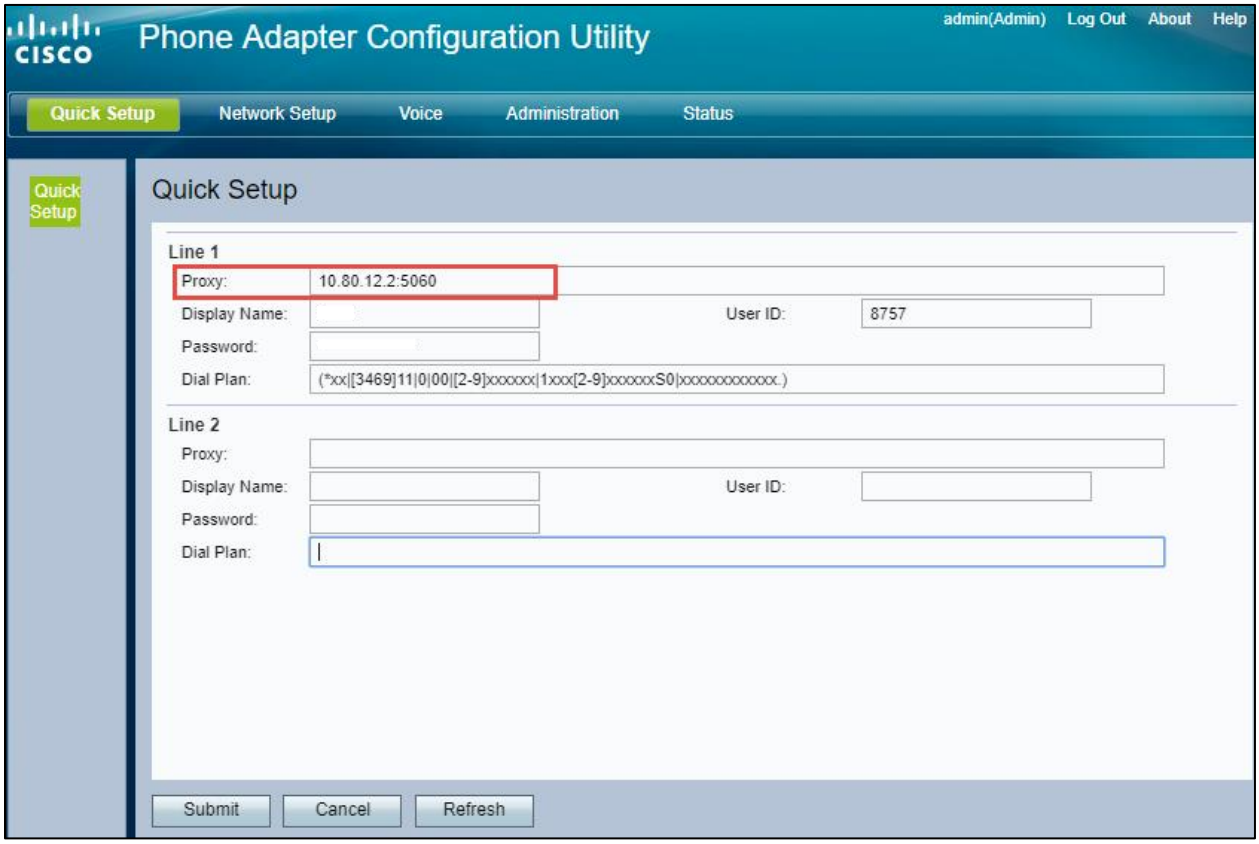

Figure 15: Fax-Quick Setup

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

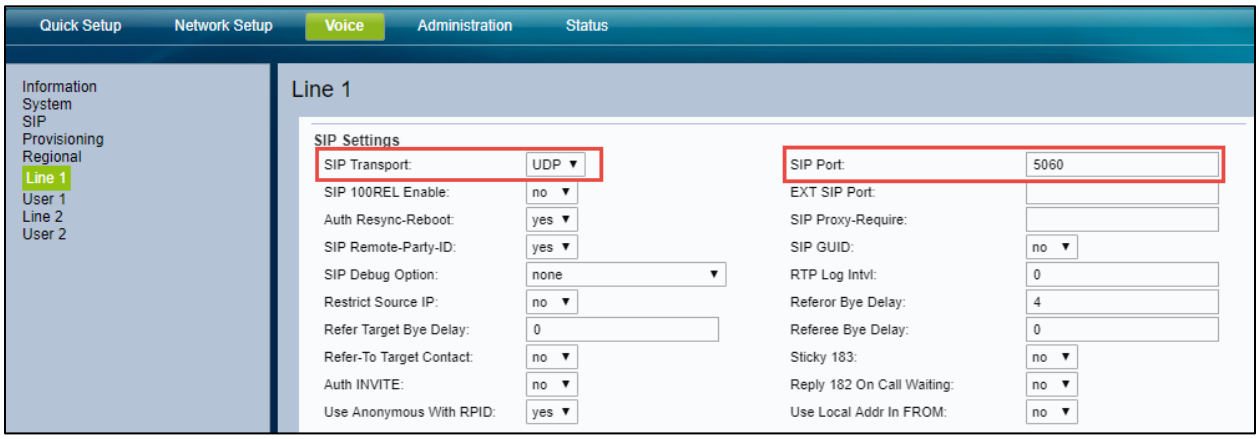

Figure 16: Fax-Line Setup-SIP Settings

- 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

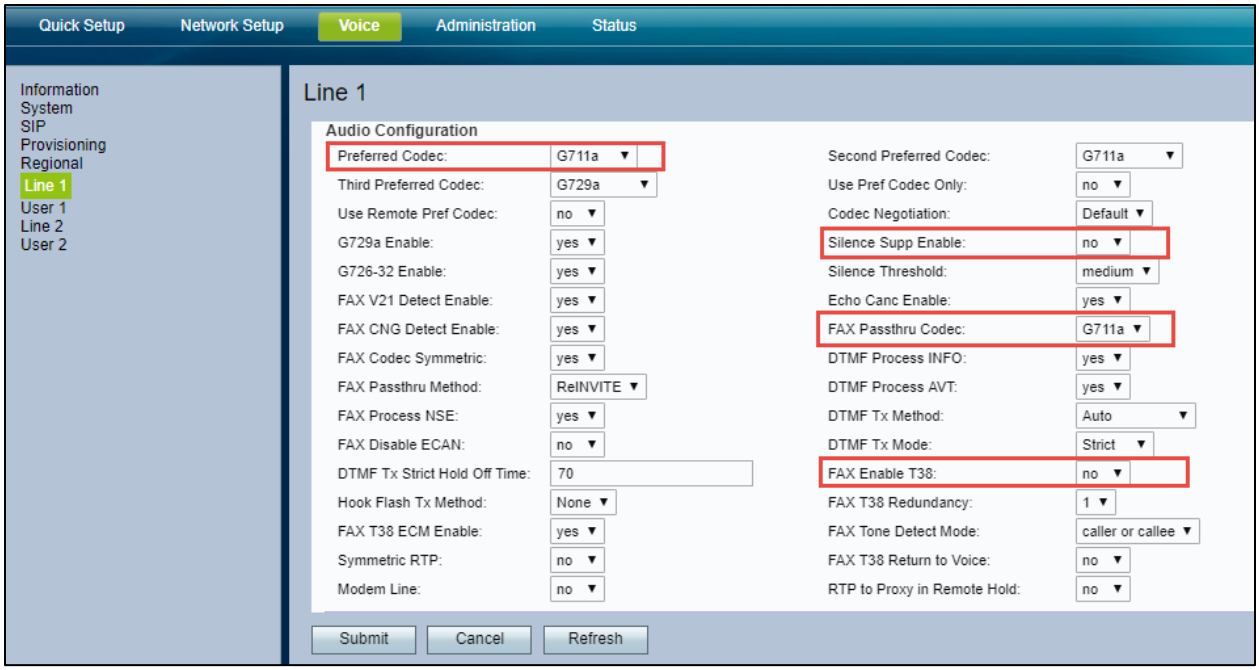

Figure 17: Fax-Line Setup-Audio Configuration

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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