Application Note

Logix Communications SIP Trunking:

Cisco Unified Communications Manager 11.5.1 with Cisco Unified Border Element (CUBE 11.5.2) on ISR4321 [IOS-XE 16.3.2] using SIP

April 4, 2017

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Introduction

Service Providers today, such as Logix Communications, 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.

Logix Communications is a service provider 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 and Logix Communications Network, Cisco Unified Border Element (CUBE v11.5.2) on ISR 4321 running IOS-XE 16.3.2 can be used. The Cisco Unified Border Element provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 11.5.1 connected to Logix Communications 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 Logix Communications 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 Unified Communications Manager (Cisco UCM) 11.5.1, and Cisco Unified Border Element (Cisco UBE) on ISR 4321 [IOS-XE version 16.3.2] for connectivity to Logix Communications SIP Trunking service. The deployment model covered in this application note is CPE (Cisco Unified Communications Manager 11.5.1) to PSTN (Logix).
- Testing was performed in accordance to Logix Communications 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 interoperability with Cisco Unity Connection (CUC).
- The Cisco UCM configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between Logix Communications 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 Logix Communications SIP Trunking network.

This application note does not cover the use of Calling Search Spaces (CSS) or partitions on Cisco UCM. To understand and learn how to apply CSS and partitions refer to the cisco.com link below:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab10/collab10/dialplan.html

Network Topology

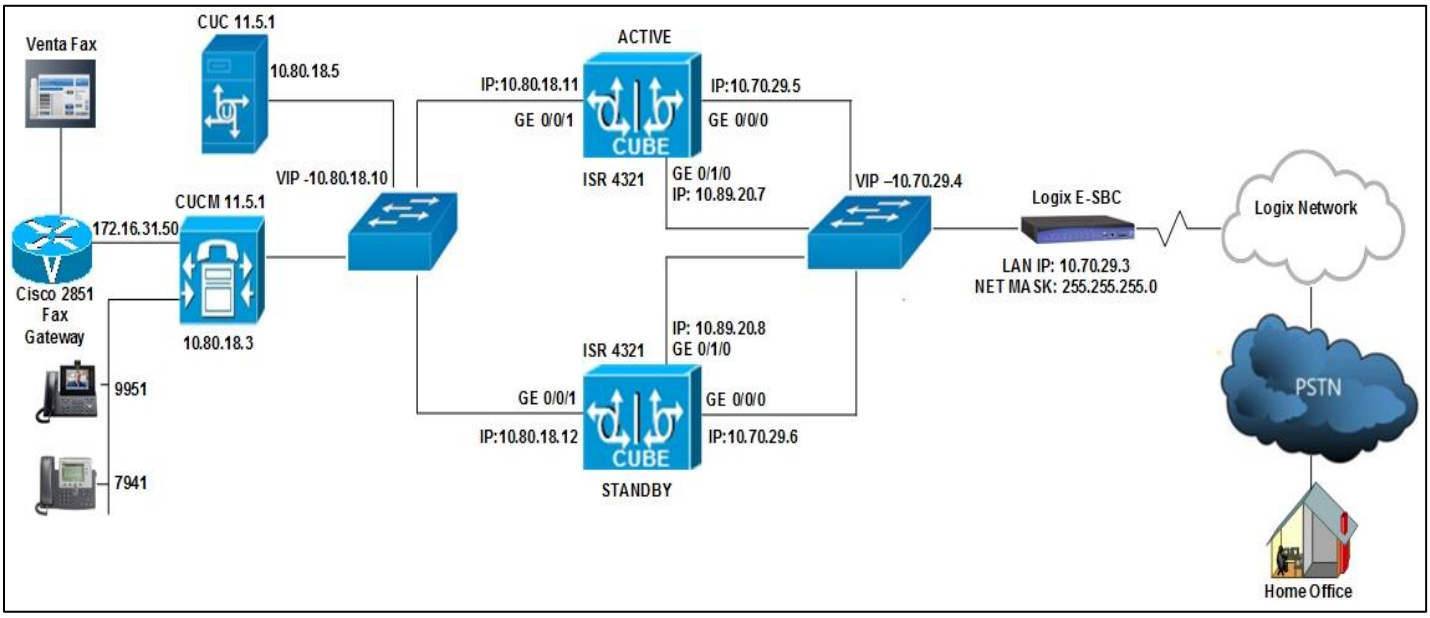


Figure 1 Network Topology

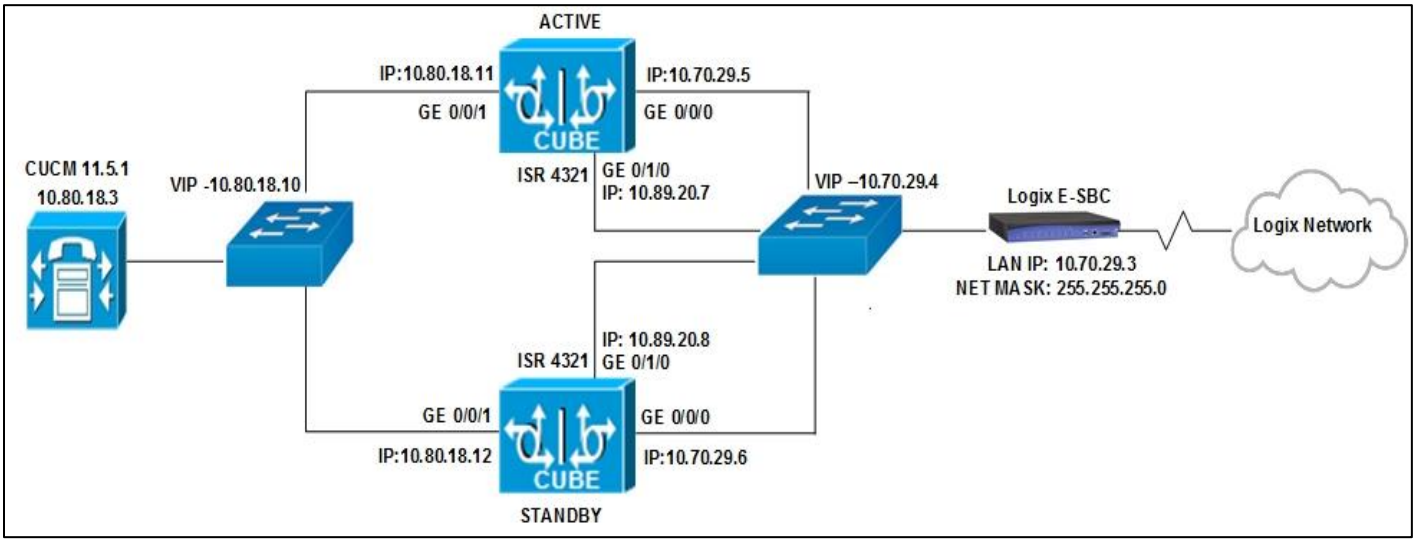


Figure 2: Cisco UBE High Availability

System Components

Hardware Requirements

- Cisco UCS-C240-M3S VMWare host running ESXi 5.5 Standard
- Cisco ISR 4321/K9 router as Cisco UBE
- Cisco ISR4321/K9 (1RU) processor with 1651964K/6147K bytes of memory with 3 Gigabit Ethernet interfaces
- Processor board ID FLM1925W0WZ
- Cisco 2851 Fax Gateway
- IP phones 9951 (SIP) and 7941 (SCCP)
- Adtran Total Access 900e 3rd Gen eSBC – Provided and Managed by Logix Communications

Software Requirements

- Cisco Unified Communications Manager 11.5.1.12900-21
- Cisco Unity Connection 11.5.1.12900-21
- Cisco IOS XE Software, Version 16.03.02 running as Cisco Unified Border Element
- IOS 15.1(4)M5 for Cisco 2851 Fax Gateway
- Adtran Total Access 900e 3rd Gen eSBC Release 11.6.0.SA.E– Provided and Managed by Logix Communications

Features

Features Supported

- Incoming and outgoing off-net calls using G711ULAW & G729 voice codecs
- Call Hold
- Call Transfer (unattended and attended)
- Call Conference
- Call Forward (all, busy, no answer)
- Calling Line (number) Identification Presentation (CLIP)
- Calling Line (number) Identification Restriction (CLIR)
- DTMF Relay (both directions) (RFC2833)
- Media flow-through on Cisco UBE
- Fax (T.38 and G711 Pass-through)

Features Not Supported

- Cisco IP phones used in this test do not support Blind Transfer
- 0 + 10 digit dial plan – Operator assisted call is not supported by Logix Communications
- In HA Redundancy mode, the Primary cube will not take over the Primary/Active role after a reboot/network outage

Caveats

- Caller ID is not updated after attended or semi-attended transfers to off-net phones. This is due to a limitation on Cisco UBE and will be resolved in the next release. The issue does not impact the calls.
- Logix Communications supports faxing up to G3/V.17 with T.38 Version 0 and G711 Pass-through
- Call Forward Unconditional to CPE User scenarios was executed by configuring Calling Party Selection* to “First Redirect number (External)” under Trunk Settings

Configuration

Configuring the Cisco Unified Border Element

Network interface

Configure Ethernet IP address and sub interface. The IP address and VLAN encapsulation used are for illustration only, the actual IP address can vary. For SIP trunks two IP addresses must be configured—LAN and WAN.

```
interface GigabitEthernet0/0/0
 description LogixCommunication WAN
 ip address 10.70.29.5 255.255.255.0
 media-type rj45
 negotiation auto
 redundancy rii 4
 redundancy group 1 ip 10.70.29.4 exclusive
!
interface GigabitEthernet0/0/1
 description LogixCommunication LAN
 ip address 10.80.18.11 255.255.255.0
 negotiation auto
 redundancy rii 3
 redundancy group 1 ip 10.80.18.10 exclusive
!
```

Global Cisco UBE settings

In order to enable Cisco UBE IP2IP gateway functionality, enter the following:

```
!
voice service voip
 no ip address trusted authenticate
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 1
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 sip
  rel1xx supported "rel100"
  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
fax protocol	Specifies the fax protocol
asserted-id	Specifies the type of privacy header in the outgoing SIP requests and response messages
early-offer forced	Enables SIP Delayed-Offer to Early-Offer globally
midcall-signaling passthru	Passes SIP messages from one IP leg to another IP leg

Codecs

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

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

Dial Peer

Cisco UBE uses dial-peer to route the call based on the digit to route the call accordingly.

```
!
dial-peer voice 500 voip
 description Outgoing Call to Logix - LAN facing
 huntstop
 session protocol sipv2
 session transport udp
 incoming called-number [0-9]T
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 510 voip
 description Outgoing call to Logix- WAN facing
 huntstop
```

```
 destination-pattern [0-9]T
 session protocol sipv2
 session target sip-server
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip profiles 101
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 520 voip
 description Incoming call to PBX - WAN facing
 huntstop
 session protocol sipv2
 session transport udp
 incoming called-number 469708....
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax nsf 000000
```

```
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 530 voip
 description Incoming call to PBX - LAN facing
 translation-profile outgoing LogixCom
 huntstop
 destination-pattern 469708....
 session protocol sipv2
 session target ipv4:10.80.18.3:5060
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
```

Call flow

In the sample configuration presented here, Cisco UCM is provisioned with four-digit directory numbers corresponding to the last four DID digits. No digit manipulation is performed on the Cisco UBE.

For incoming PSTN calls, the Cisco UBE presents the full ten-digit DID number to Cisco UCM. The Cisco UCM picks up the last 4 significant Digits configured under SIP Trunk and routes the call based on those 4 digits. Voice calls are routed to IP phones; Fax calls are routed via a 4-digit route pattern over a SIP trunk that terminates on the Fax Gateway and in turn to the VentaFax client connected to the Fax Gateway.

CPE callers make outbound PSTN calls by dialing a “8” prefix followed by the destination number. For outbound fax calls from the analog fax endpoint, Cisco fax Gateway sends to Cisco UCM the DID with leading access code “8”. A “8.@” route pattern strips the prefix and routes the call with the remaining digits via a SIP trunk terminating on the Cisco UBE for Voice call or Fax. For PBX to PBX via Logix Communications, Caller dial 8 prefix followed by the target 1+10-digits number, 8 was stripped and the remaining digits were send to Cisco UBE, Cisco UBE pass the DID under Dial Peer 510 and send to Logix Communications network which will direct back to Cisco UBE and handled same as normal incoming PSTN call.

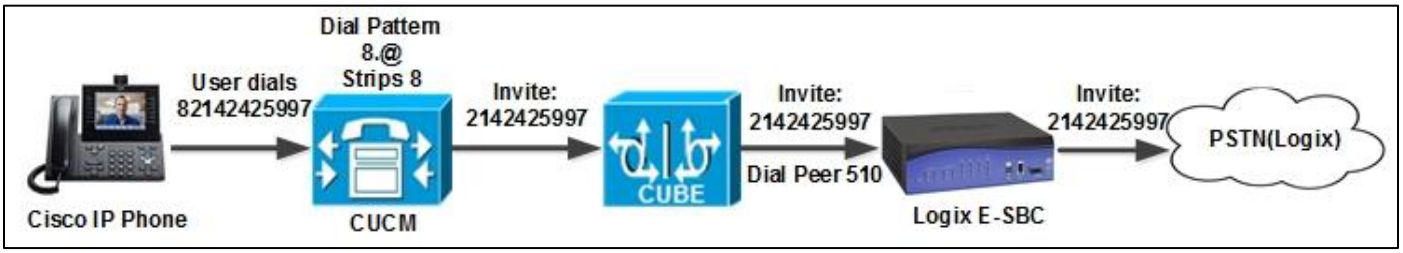


Figure 3: Outbound Voice Call

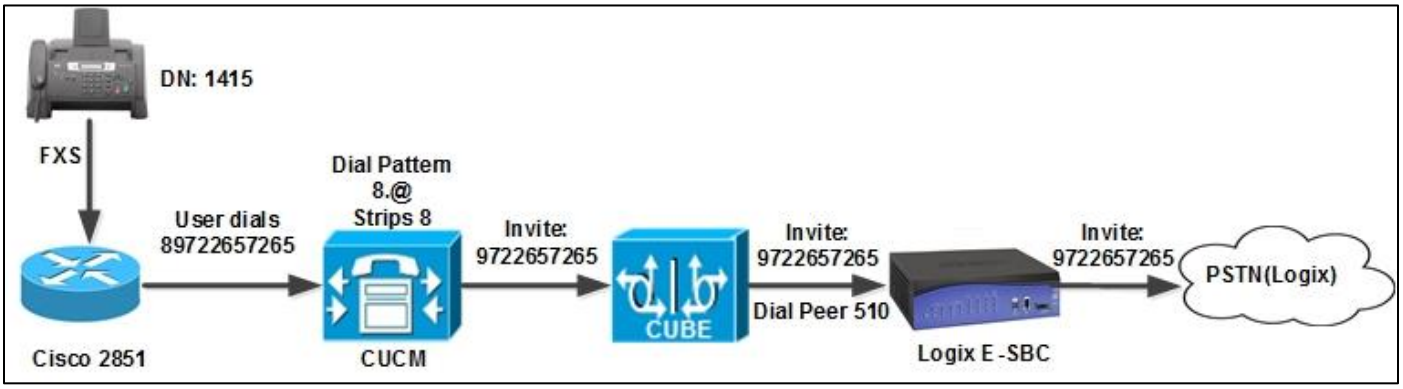


Figure 4: Outbound Fax Call

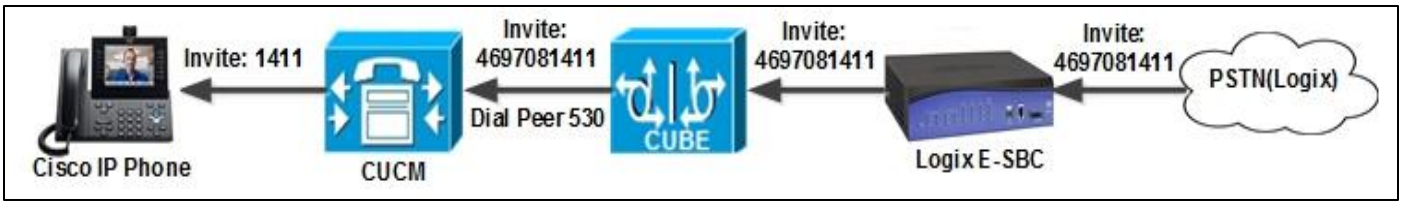


Figure 5: Inbound Voice Call

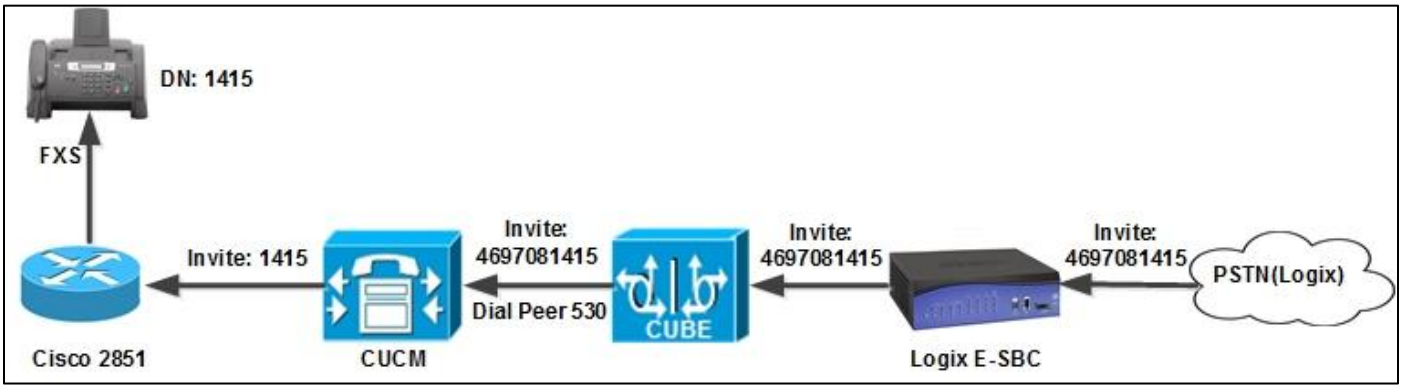


Figure 6 : Inbound Fax Call

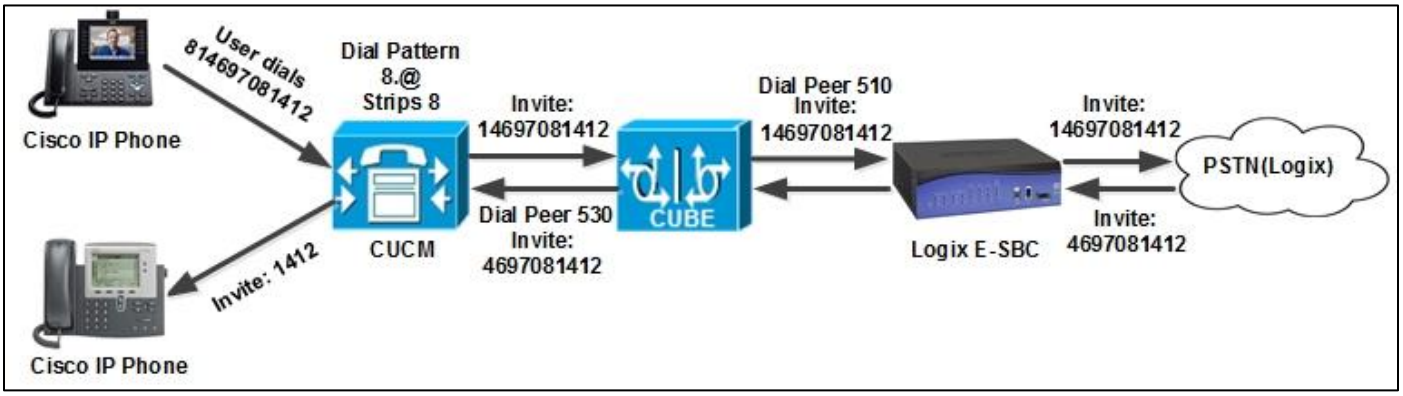


Figure 7 : PBX to PBX via Logix Call

Configuration Example

The following configuration snippet contains a sample configuration of Cisco Unified Border Element with all parameters mentioned previously.

Active Cisco UBE

```
Logix1#sh running-config
Building configuration...


Current configuration: 8264 bytes
!
version 16.3
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
service password-encryption
service internal
service sequence-numbers
no platform punt-keepalive disable-kernel-core
!
hostname Logix1
!
boot-start-marker
boot system bootflash:isr4300-universalk9.16.03.02.SPA.bin
boot-end-marker
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
```

```
 address-family ipv6
 exit-address-family
!
no logging queue-limit
no logging buffered
no logging rate-limit
enable secret 5
!
no aaa new-model
no ip domain lookup
!
subscriber templating
multilink bundle-name authenticated
!
crypto pki trustpoint TP-self-signed-1270583006
 enrollment selfsigned
 subject-name cn=IOS-Self-Signed-Certificate-1270583006
 revocation-check none
 rsakeypair TP-self-signed-1270583006
!
!
crypto pki certificate chain TP-self-signed-1270583006
 certificate self-signed 01
  30820330 30820218 A0030201 02020101 300D0609 2A864886 F70D0101 05050030
  31312F30 2D060355 04031326 494F532D 53656C66 2D536967 6E65642D 43657274
  69666963 6174652D 31323730 35383330 3036301E 170D3137 30323130 31343237
  34345A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649
  4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D31 32373035
  38333030 36308201 22300D06 092A8648 86F70D01 01010500 0382010F 00308201
  0A028201 0100A34F 136DD7A1 E1815DA1 B05C5396 E3B88AC9 7DE1A1D7 12F1BEA7
  4985E12B C6858F9D 95E7082B 3BBC56CD AAFDEC4F 7250D4CE 892713BE 509A6DCE
```

```
 05FD3768 ED1EF293 B3C2C1CE 4684F8F9 E920AE8F 33F4DFE0 FF04BE27 B75A28C1
 6A2084C5 31BFF5C1 CD07916D 83FD56EF 9023C974 A9835AAF AC1AAF93 C0FB6856
 5CA7B10A AF9EFCE1 5DE8651F D30847FF 024D6EF3 3AADB77D 68519BA9 F21AC1FE
 5A50CA58 A00CDBB5 25C693E8 4D8C639D 6E5A3935 2F050F4D A3A7B2AD 47942BDD
 4D78EFEE 81FDAFE0 F26220A6 6AF1D505 C601A2B3 56B2D2FE 5DD60B95 7B149AC6
 EB0CACE9 CA5D42CC 4B0CA1DE 2895251A 4C1AFBC0 4FD54872 50BC69B2 445DF62E
 CA556655 9BEB0203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF
 301F0603 551D2304 18301680 140623A5 BE29331A B5FF4081 20569978 FB29842F
 58301D06 03551D0E 04160414 0623A5BE 29331AB5 FF408120 569978FB 29842F58
 300D0609 2A864886 F70D0101 05050003 82010100 9A8F4A49 4CC83788 4EC24211
 CE24EE7A B8552513 F9F34632 04B8119D 612FFA57 370471EF 123E1385 EBC74CBD
 92DD9795 086536A0 F2469390 219B288F 3D9EC787 48A4EE78 5A492BA1 1680D1C9
 F3A8A820 1D065DEB E8F0D00E 37A2A866 F759FB2D E30CD9B8 8900E25E 8171A288
 FB2BB185 B6A6ED29 3BAA4495 31FCC789 0305E830 6EBA491E 211F0B7C FE808066
 4E78E657 D16239B2 6E40B8A0 41631417 40EAB264 D31C2A10 24FEFD2B F0C5A1D9
 693C7384 D1C54B99 BAC2A7AA 5B646CED 6E31FEC1 EB64C663 9F703970 BFA72795
 06252993 E38182F3 7F760357 37556092 A5FE18F4 4FCC6BA3 716886FA 76106709
 8D4EF4C5 14A81EB9 F0A29EB9 DB41CAB7 F98A5D15
      quit
!
voice service voip
 no ip address trusted authenticate
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 1
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 sip
  rel1xx supported "rel100"
  session refresh
  asserted-id pai
```

```
 privacy pstn
 early-offer forced
 midcall-signaling passthru
 g729 annexb-all
!
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 g729r8
!
voice class codec 2
 codec preference 1 g729r8
 codec preference 2 g711ulaw
!
voice class codec 3
 codec preference 1 g729r8
!
!
voice class sip-profiles 101
 request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "sip:469708\1@\2"
!
voice translation-rule 1004
 rule 1 /4697081413/ /4000/
!
voice translation-profile LogixCom
translate called 1004
!
license udi pid ISR4321/K9 sn FDO17860MQ8
license boot level appxk9
license boot level uck9
!
no diagnostic bootup level
```

```
spanning-tree extend system-id
!
redundancy
 mode none
 application redundancy
  group 1
   name b2bhalogixcommunications
   priority 100 failover threshold 75
   timers delay 30 reload 60
   control GigabitEthernet0/1/0 protocol 1
   data GigabitEthernet0/1/0
   track 1 shutdown
   track 2 shutdown
!
vlan internal allocation policy ascending
!
track 1 interface GigabitEthernet0/0/0 line-protocol
track 2 interface GigabitEthernet0/0/1 line-protocol
!
interface GigabitEthernet0/0/0
 description LogixCommunication WAN
 ip address 10.70.29.5 255.255.255.0
 media-type rj45
 negotiation auto
 redundancy rii 4
 redundancy group 1 ip 10.70.29.4 exclusive
!
interface GigabitEthernet0/0/1
 description LogixCommunication LAN
 ip address 10.80.18.11 255.255.255.0
 negotiation auto
```

```
 redundancy rii 3
 redundancy group 1 ip 10.80.18.10 exclusive
!
interface GigabitEthernet0/1/0
 description CUBE HA
 ip address 10.89.20.7 255.255.255.0
 negotiation auto
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 negotiation auto
!
interface Vlan1
 no ip address
 shutdown
!
ip forward-protocol nd
no ip http server
no ip http secure-server
ip tftp source-interface GigabitEthernet0
ip route 0.0.0.0 0.0.0.0 10.70.29.1
ip route 10.64.0.0 255.255.0.0 10.80.18.1
ip route 10.70.0.0 255.255.0.0 10.70.29.1
ip route 10.80.0.0 255.255.0.0 10.80.18.1
ip route 172.16.0.0 255.255.0.0 10.80.18.1
!
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 500 voip
 description Outgoing Call to Logix - LAN facing
 huntstop
 session protocol sipv2
 session transport udp
 incoming called-number [0-9]T
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 510 voip
 description Outgoing call to Logix- WAN facing
 huntstop
 destination-pattern [0-9]T
 session protocol sipv2
 session target sip-server
 session transport udp
```

```
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip profiles 101
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 520 voip
 description Incoming call to PBX - WAN facing
 huntstop
 session protocol sipv2
 session transport udp
 incoming called-number 469708....
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 530 voip
 description Incoming call to PBX - LAN facing
```

```
 translation-profile outgoing LogixCom
 huntstop
 destination-pattern 469708....
 session protocol sipv2
 session target ipv4:10.80.18.3:5060
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
sip-ua
 keepalive target ipv4:10.70.29.3:5060
 timers keepalive active 60
 sip-server ipv4:10.70.29.3:5060
!
!
line con 0
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 exec-timeout 0 0
 password 7
```

```
 login
line vty 5
 exec-timeout 0 0
 password 7
 login
!
end
```

Standby Cisco UBE

```
version 16.3
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
no platform punt-keepalive disable-kernel-core
!
hostname Logix2
!
boot-start-marker
boot system bootflash:isr4300-universalk9.16.03.02.SPA.bin
boot-end-marker
!
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
no logging queue-limit
no logging buffered
no logging rate-limit
enable secret 5
!
no aaa new-model
!
no ip domain lookup
```

```
subscriber templating
multilink bundle-name authenticated
!
crypto pki trustpoint TP-self-signed-2548443246
 enrollment selfsigned
 subject-name cn=IOS-Self-Signed-Certificate-2548443246
 revocation-check none
 rsakeypair TP-self-signed-2548443246
!
crypto pki certificate chain TP-self-signed-2548443246
 certificate self-signed 01
  30820330 30820218 A0030201 02020101 300D0609 2A864886 F70D0101 05050030
  31312F30 2D060355 04031326 494F532D 53656C66 2D536967 6E65642D 43657274
  69666963 6174652D 32353438 34343332 3436301E 170D3137 30323130 31373132
  33365A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649
  4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D32 35343834
  34333234 36308201 22300D06 092A8648 86F70D01 01010500 0382010F 00308201
  0A028201 0100B73A 8AE876C0 62A381D9 4C331F21 6FBF60E9 20F9420A 6F2C3A5A
  2DA4B74B 1A9B55DB 65BC3A4B 016D4E96 3CB638A7 31C61AA1 A2E8EF3E FE7733F5
  A0035F13 9AE153CE D55D4F64 FBBCA3CE EC8D110A 6490B2CD 44509DEE 14A60E75
  66CF37C5 3DF0BBBE 7B27306D C2ACDBA2 A3497E3D 7EFDCC2B 1902A0A8 038AD01E
  68FD339C B2620BFF 00E703AB 88DD7796 B7C0351F 27BFF1EC 791ECF53 87B57E81
  166B26BA 1428E6F7 A7484680 ACF2B8F7 BB95977B C3854F78 D4295377 CE568896
  451A72F7 5D117423 6A69CEEE 8E13BDAB 96B61B29 1165C7C4 E7DB2BCB 1A2095F6
  E9C80DDD 9B1DCED7 CE87C3F0 BF726628 6AF272BA 9A3D33B0 4AD1444F 87DF933F
  9BCCF78D 3C6B0203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF
  301F0603 551D2304 18301680 14EADBB3 9378001B 592B6EC4 0CD4C83F 3FA14FA3
  AD301D06 03551D0E 04160414 EADBB393 78001B59 2B6EC40C D4C83F3F A14FA3AD
  300D0609 2A864886 F70D0101 05050003 82010100 A99C7D6F 325E52A8 9F0221FE
  3BC2460A 5383DF63 B1A66575 D8CE62F8 725B14FA F18879B1 38173AF8 3A05D05F
  B72318A1 23B11F12 E14DE814 CF93938D 41F75435 21999BC2 6FFAFBAC 84347F5F
```

```
  0B4787B0 2A54C190 8E4A7505 3957998E 8A58C6A1 D6BBD261 11C24F7F 61FA6931
  3A311AC7 E7D50544 7A6DA790 09A5E366 4635FE2F C5824130 AB3FDA56 AE0BEA53
  E674A90C C13EF3B9 2C0D7F9B 3ED7CD5A 3FABB4D6 A0B9E76A 180E8DC1 E1E8024F
  E431C813 64B5F09E A3BB9F8D 669E33F5 2BFB2295 34FDDE83 1290F246 61BA7FEB
  943F10C5 29499BFE 173CE107 6D938477 5A4DB917 B88D2CC5 6A2F77B1 3E605311
  B4A19B46 1A05F455 3E2816E0 59D13336 027F827D
        quit
!
voice service voip
 no ip address trusted authenticate
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 1
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 sip
  rel1xx supported "rel100"
  session refresh
  asserted-id pai
  privacy pstn
  early-offer forced
  midcall-signaling passthru
  g729 annexb-all
!
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 g729r8
!
voice class codec 2
 codec preference 1 g729r8
 codec preference 2 g711ulaw
```

```
!
voice class codec 3
 codec preference 1 g729r8
!
voice class sip-profiles 101
 request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "sip:469708\1@\2"
!
voice translation-rule 1004
 rule 1 /4697081413/ /4000/
!
voice translation-profile LogixCom
translate called 1004
!
license udi pid ISR4321/K9 sn FDO17860MW3
license boot level appxk9
license boot level uck9
!
diagnostic bootup level minimal
spanning-tree extend system-id
!
redundancy
 mode none
 application redundancy
  group 1
   name b2bhalogixcommunications
   priority 100 failover threshold 75
   timers delay 30 reload 60
   control GigabitEthernet0/1/0 protocol 1
   data GigabitEthernet0/1/0
   track 1 shutdown
   track 2 shutdown
```

```
!
vlan internal allocation policy ascending
!
track 1 interface GigabitEthernet0/0/0 line-protocol
track 2 interface GigabitEthernet0/0/1 line-protocol
!
interface GigabitEthernet0/0/0
 description LogixCommunication WAN
 ip address 10.70.29.6 255.255.255.0
 media-type rj45
 negotiation auto
 redundancy rii 4
 redundancy group 1 ip 10.70.29.4 exclusive
!
interface GigabitEthernet0/0/1
 description LogixCommunication LAN
 ip address 10.80.18.12 255.255.255.0
 negotiation auto
 redundancy rii 3
 redundancy group 1 ip 10.80.18.10 exclusive
!
interface GigabitEthernet0/1/0
 description CUBE HA
 ip address 10.89.20.8 255.255.255.0
 negotiation auto
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 negotiation auto
!
```

```
interface Vlan1
 no ip address
 shutdown
!
ip forward-protocol nd
no ip http server
no ip http secure-server
ip tftp source-interface GigabitEthernet0
ip route 0.0.0.0 0.0.0.0 10.70.29.1
ip route 0.0.0.0 0.0.0.0 10.64.1.1
ip route 10.70.0.0 255.255.0.0 10.70.29.1
ip route 10.80.0.0 255.255.0.0 10.80.18.1
ip route 172.16.0.0 255.255.0.0 10.80.18.1
!
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 500 voip
 description Outgoing Call to Logix - LAN facing
 huntstop
 session protocol sipv2
 session transport udp
 incoming called-number [0-9]T
 voice-class codec 1
 voice-class sip asserted-id pai
```

```
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 510 voip
 description Outgoing call to Logix- WAN facing
 huntstop
 destination-pattern [0-9]T
 session protocol sipv2
 session target sip-server
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip profiles 101
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 520 voip
 description Incoming call to PBX - WAN facing
 huntstop
```

```
 session protocol sipv2
 session transport udp
 incoming called-number 469708....
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 no vad
!
dial-peer voice 530 voip
 description Incoming call to PBX - LAN facing
 huntstop
 destination-pattern 469708....
 session protocol sipv2
 session target ipv4:10.80.18.3:5060
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
```

```
 no vad
!
!
sip-ua
 keepalive target ipv4:10.70.29.3:5060
 timers keepalive active 60
 sip-server ipv4:10.70.29.3:5060
!
!
line con 0
 stopbits 1
line aux 0
 stopbits 1
line vty 0 5
 exec-timeout 0 0
 password 7
 login
!

end
```

Configuring Cisco Unified Communications Manager

Cisco UCM Version

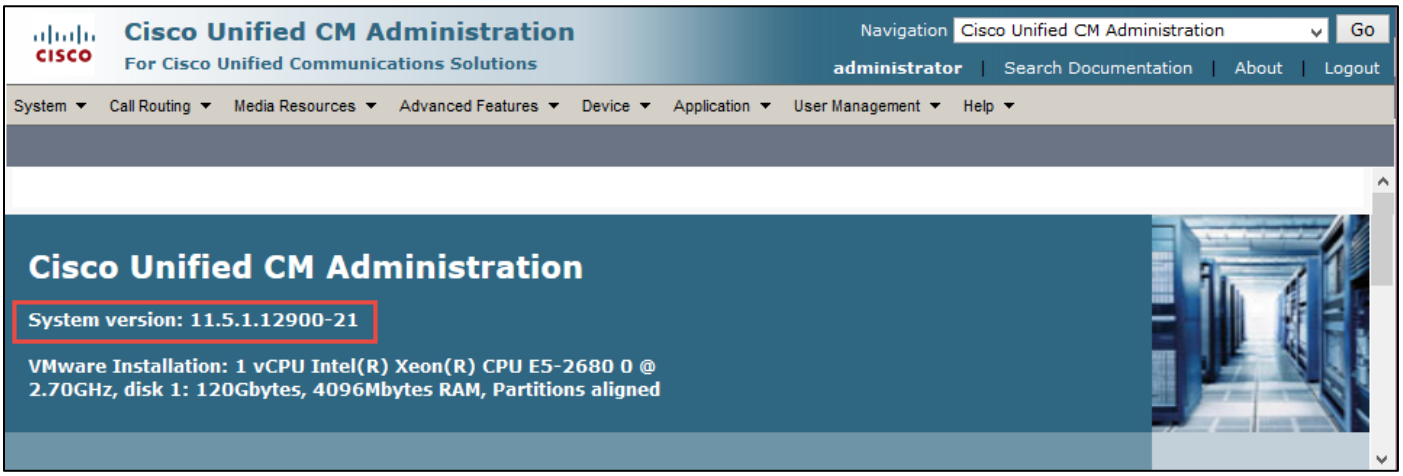


Figure 8: Cisco UCM Version

Cisco Call Manager Service Parameters

Navigation: System > Service Parameters

1. Select Server* = **Clus28Sub1--CUCM Voice/Video (Active)**
2. Select Service*= **Cisco CallManager (Active)**
3. Duplex Streaming Enabled = **True**
4. All other fields are set to default values

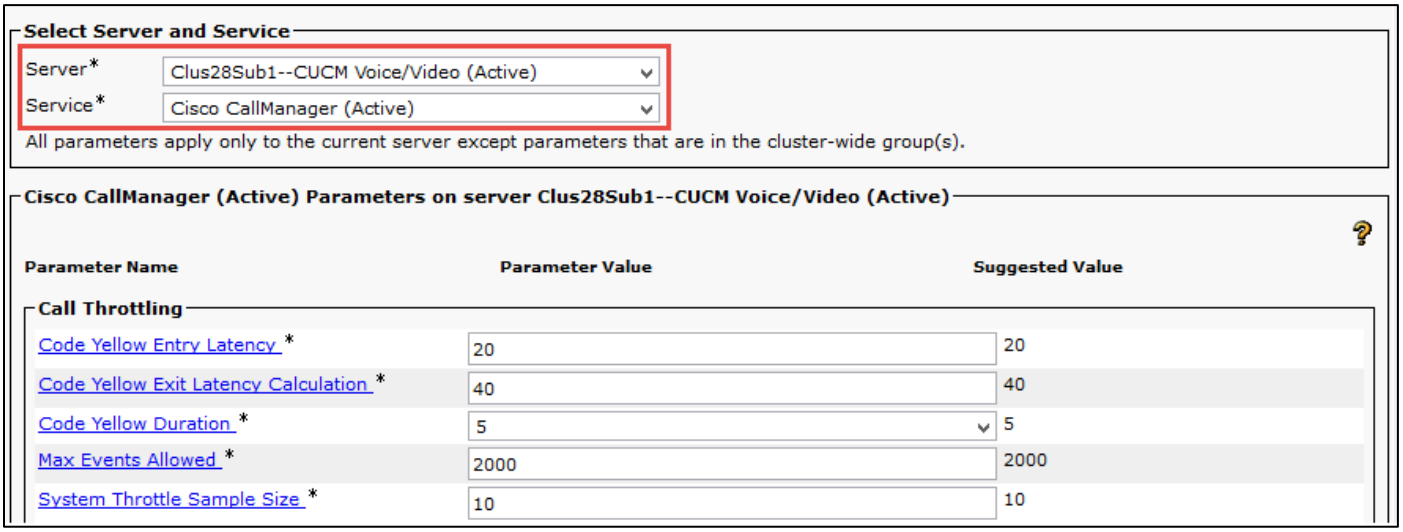


Figure 9: Service Parameters

Clusterwide Parameters (Service)

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

Figure 10: Service Parameters (Cont.)

Off-net Calls via Logix Communications SIP Trunk

Off-net calls are served by SIP trunks configured between Cisco UCM and Logix Communications Network and calls are routed via Cisco UBE

SIP Trunk Security Profile

Navigation: System → Security → SIP Trunk Security Profile

1. Name*= **LogixCommunications Non Secure SIP Trunk Profile** is used as an example
2. Description = **Non Secure SIP Trunk Profile authenticated by null String** is used as an example
3. Device Security Mode = **Non Secure**
4. Incoming Transport Type* = **TCP + UDP**
5. Outgoing Transport Type = **UDP**

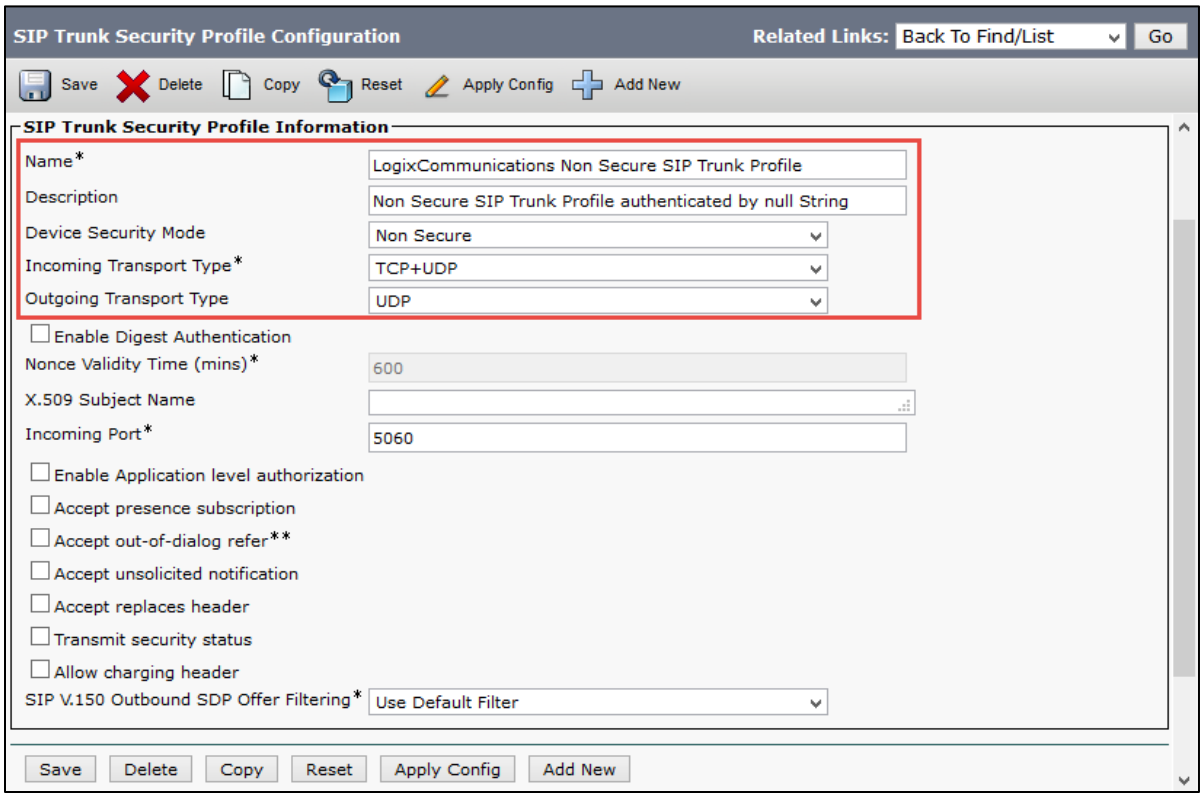


Figure 11: SIP Trunk Security Profile

Explanation

Parameter	Value	Description
Incoming Transport Type	TCP + UDP	
Outgoing Transport Type	UDP	SIP trunks to Logix Communications SBC should use UDP as a transport protocol for SIP. This is configured using SIP Trunk Security profile, which is later assigned to the SIP trunk itself.

SIP Profile Configuration

Note: SIP Profile will be later associated with the SIP trunk

Navigation: Device → Device Settings → SIP Profile

1. Name*= **LogixCommunications Standard SIP Profile** is used as an example
2. Description = **Default SIP Profile** is used as an example

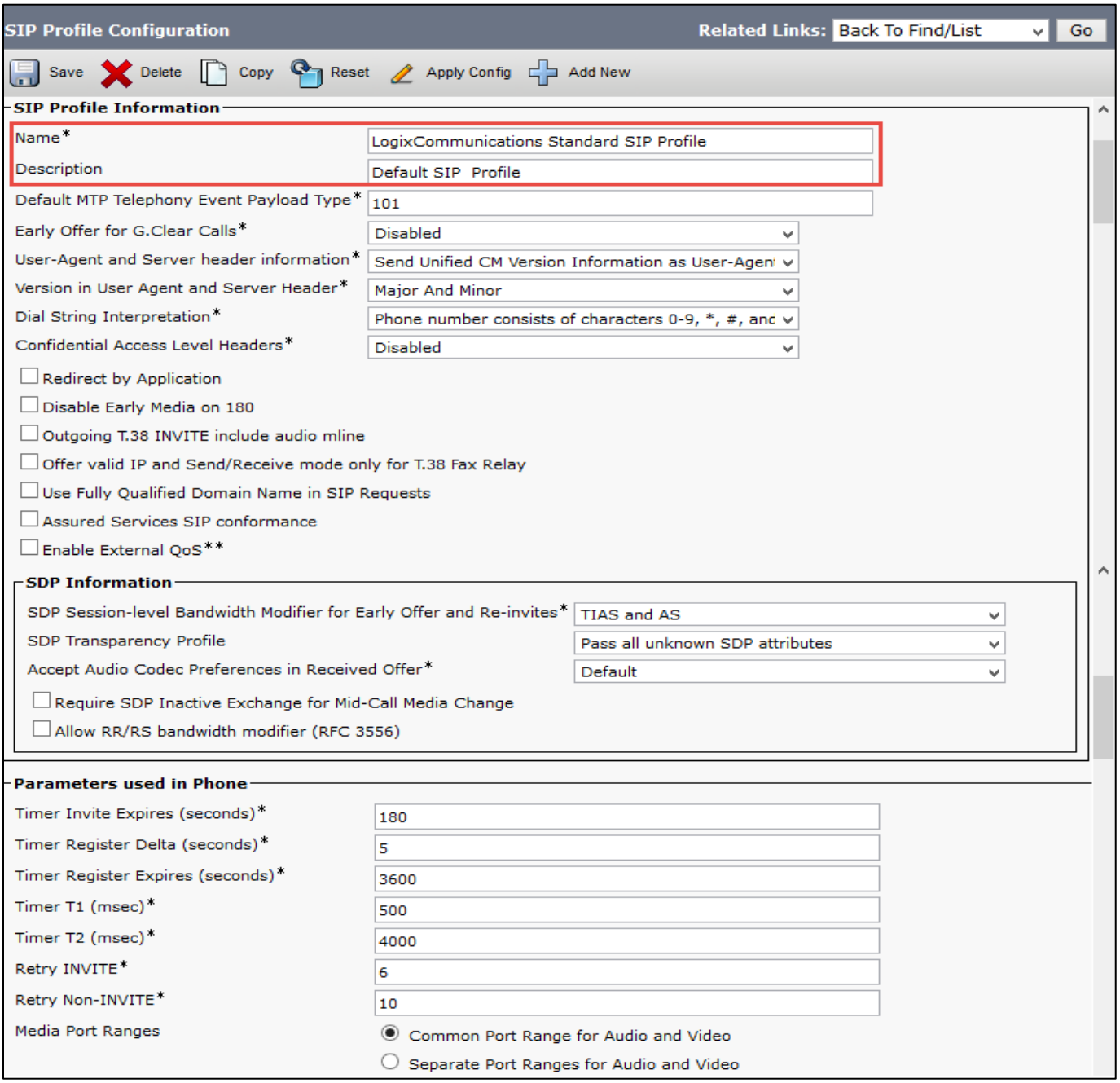


Figure 12: SIP Profile

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		

Figure 13: SIP Profile (Cont.)

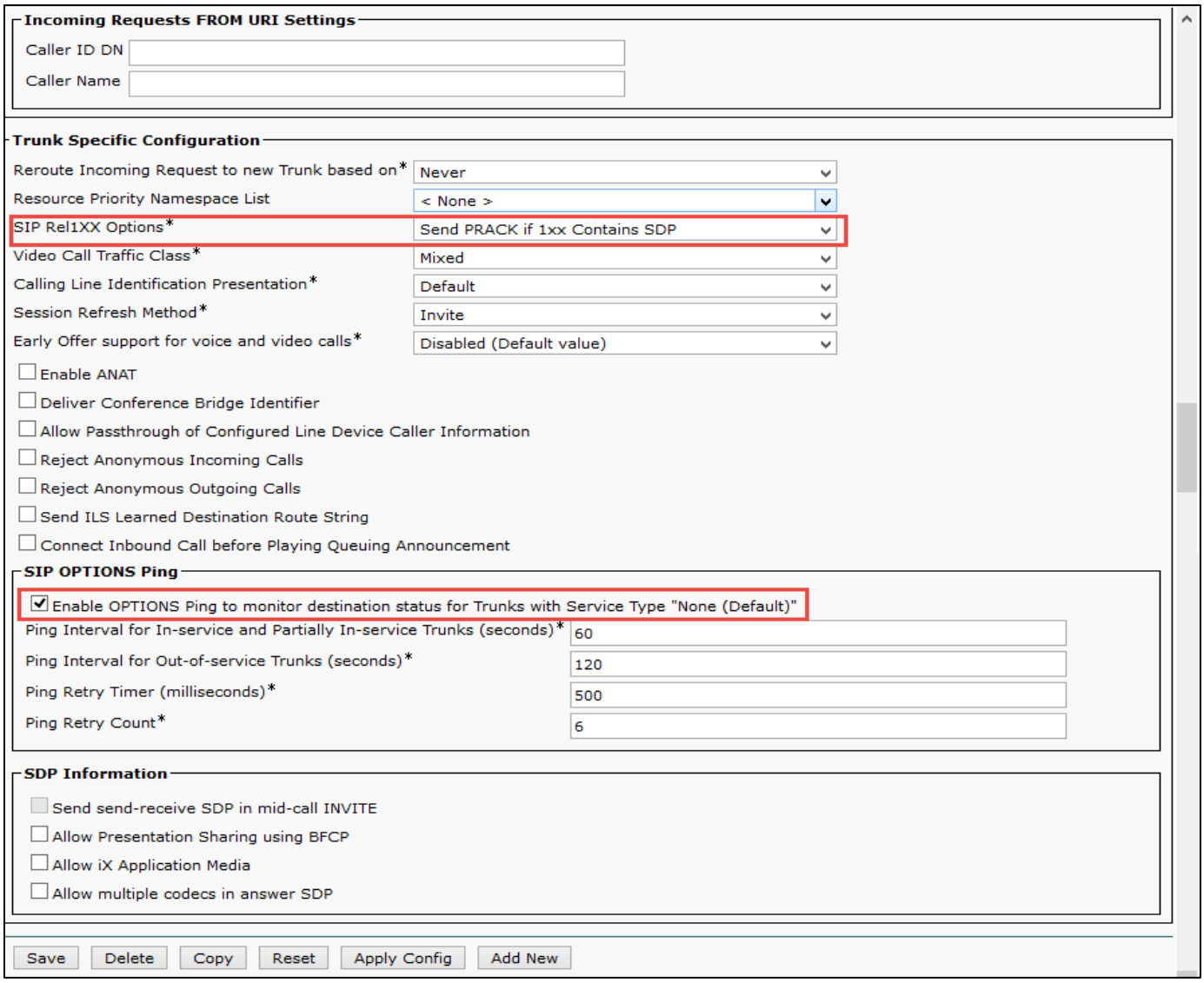


Figure 14: SIP Profile (Cont.)

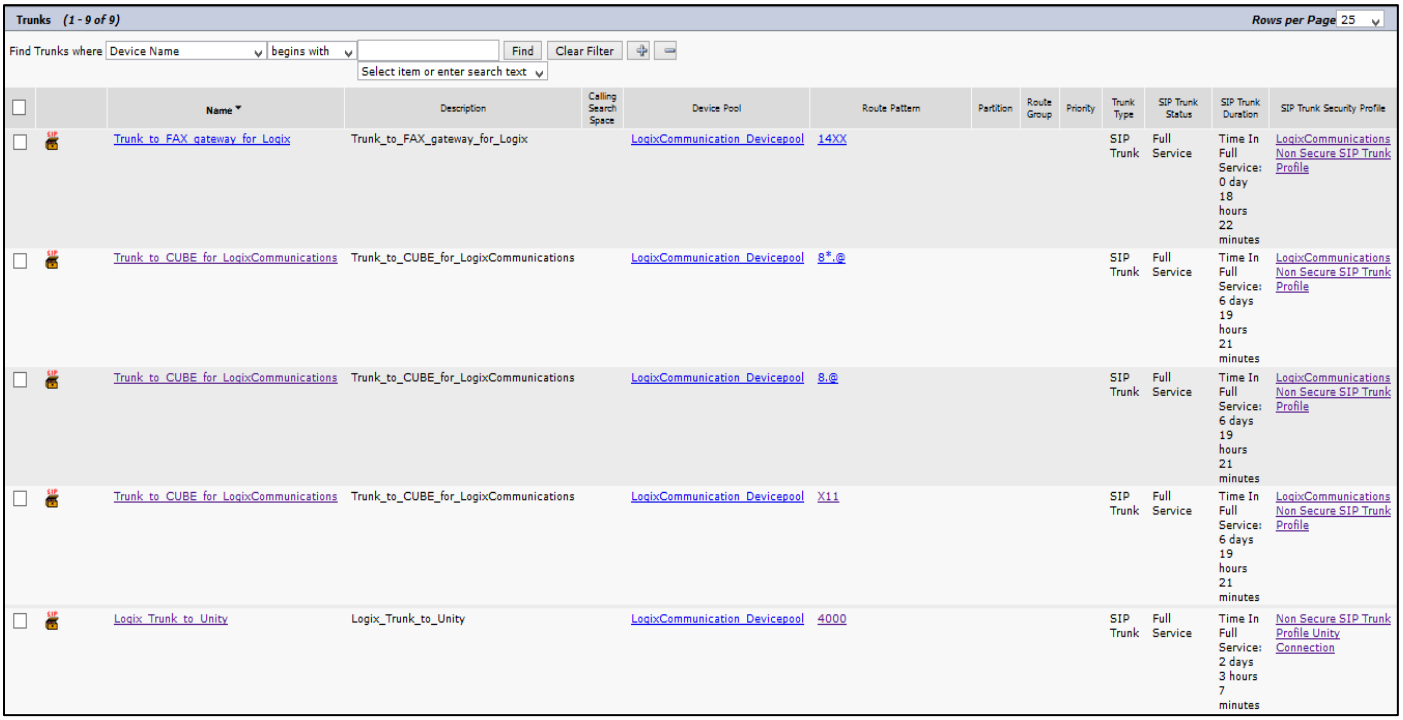
Explanation

Parameter	Value	Description
Default MTP Telephony Event Payload Type	101	RFC2833 DTMF payload type
SIP Rel1XX Options	Send PRACK for 1xx Contains SDP	Enable Provisional Acknowledgements (Reliable 100 messages)
Ping Interval for In-service and Partially In-service Trunks (seconds)	60	OPTIONS message parameters- interval time
Ping Interval for Out-of-service Trunks (seconds)	120	OPTIONS message parameters- interval time

SIP Trunk Configuration

Create SIP trunks to Cisco UBE

Navigation Path: Device →Trunk



Trunks (1 - 9 of 9) Rows per Page 25

Find Trunks where Device Name begins with Find Clear Filter

Select item or enter search text

Name	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type	SIP Trunk Status	SIP Trunk Duration	SIP Trunk Security Profile
Trunk_to_FAX_gateway_for_Logix	Trunk_to_FAX_gateway_for_Logix		LogixCommunication_Devicepool	14XX				SIP Trunk	Full Service	Time In Full Service: 0 day 18 hours 22 minutes	LogixCommunications Non Secure SIP Trunk Profile
Trunk_to_CUBE_for_LogixCommunications	Trunk_to_CUBE_for_LogixCommunications		LogixCommunication_Devicepool	8*.@				SIP Trunk	Full Service	Time In Full Service: 6 days 19 hours 21 minutes	LogixCommunications Non Secure SIP Trunk Profile
Trunk_to_CUBE_for_LogixCommunications	Trunk_to_CUBE_for_LogixCommunications		LogixCommunication_Devicepool	8.@				SIP Trunk	Full Service	Time In Full Service: 6 days 19 hours 21 minutes	LogixCommunications Non Secure SIP Trunk Profile
Trunk_to_CUBE_for_LogixCommunications	Trunk_to_CUBE_for_LogixCommunications		LogixCommunication_Devicepool	X11				SIP Trunk	Full Service	Time In Full Service: 6 days 19 hours 21 minutes	LogixCommunications Non Secure SIP Trunk Profile
Logix_Trunk_to_Unity	Logix_Trunk_to_Unity		LogixCommunication_Devicepool	4000				SIP Trunk	Full Service	Time In Full Service: 2 days 3 hours 7 minutes	Non Secure SIP Trunk Profile Unity Connection

Figure 15: SIP Trunks List

SIP Trunk Status

Service Status: Full Service
Duration: Time In Full Service: 6 days 19 hours 27 minutes

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	Trunk_to_CUBE_for_LogixCommunications
Description	Trunk_to_CUBE_for_LogixCommunications
Device Pool*	LogixCommunication_Devicepool
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	MRGL_Default
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
- [x] 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

- [x] PSTN Access
- [ ] 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 >

Figure 16: SIP Trunk to Cisco UBE

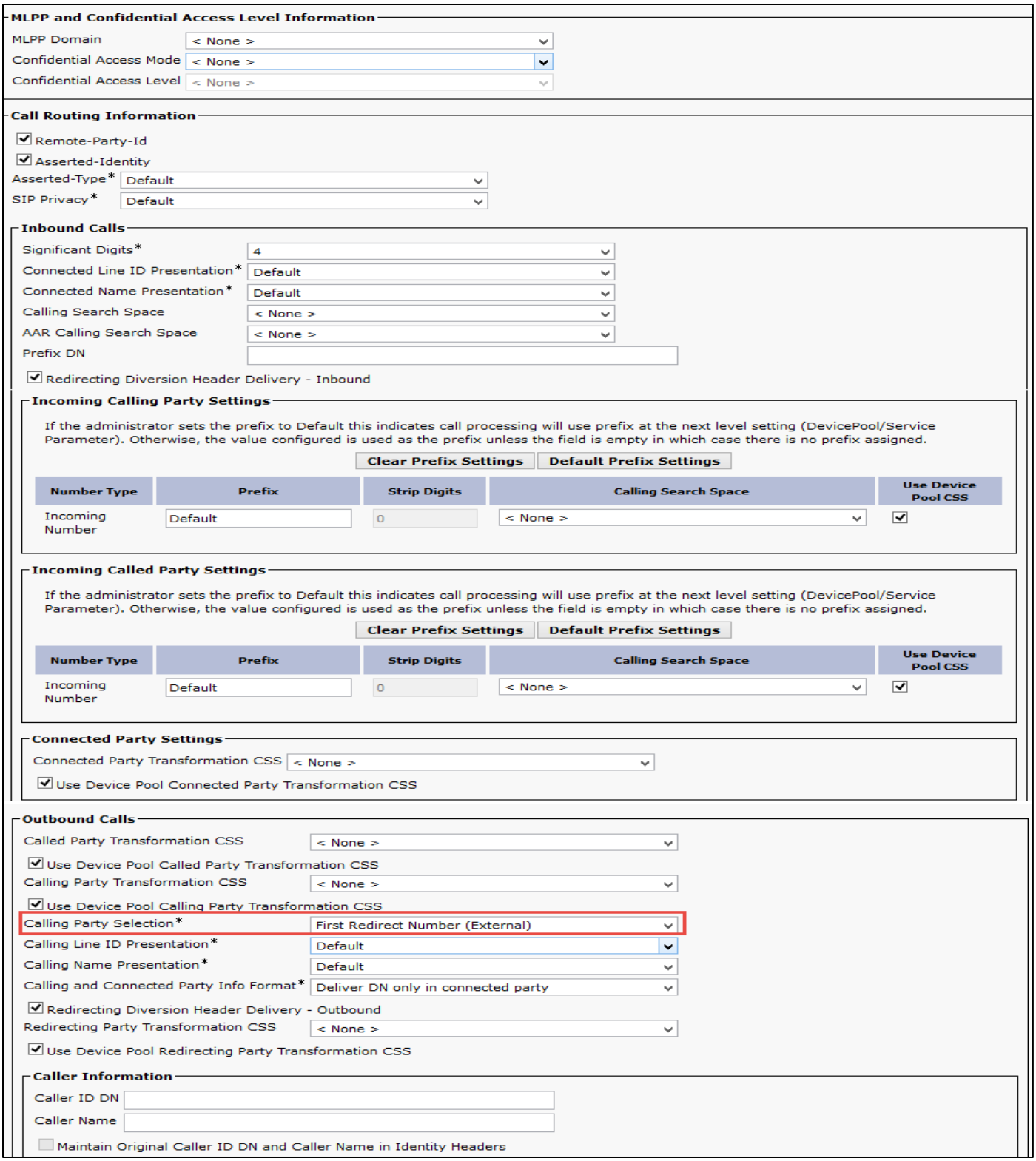


Figure 17: SIP Trunk to Cisco UBE (Cont.)

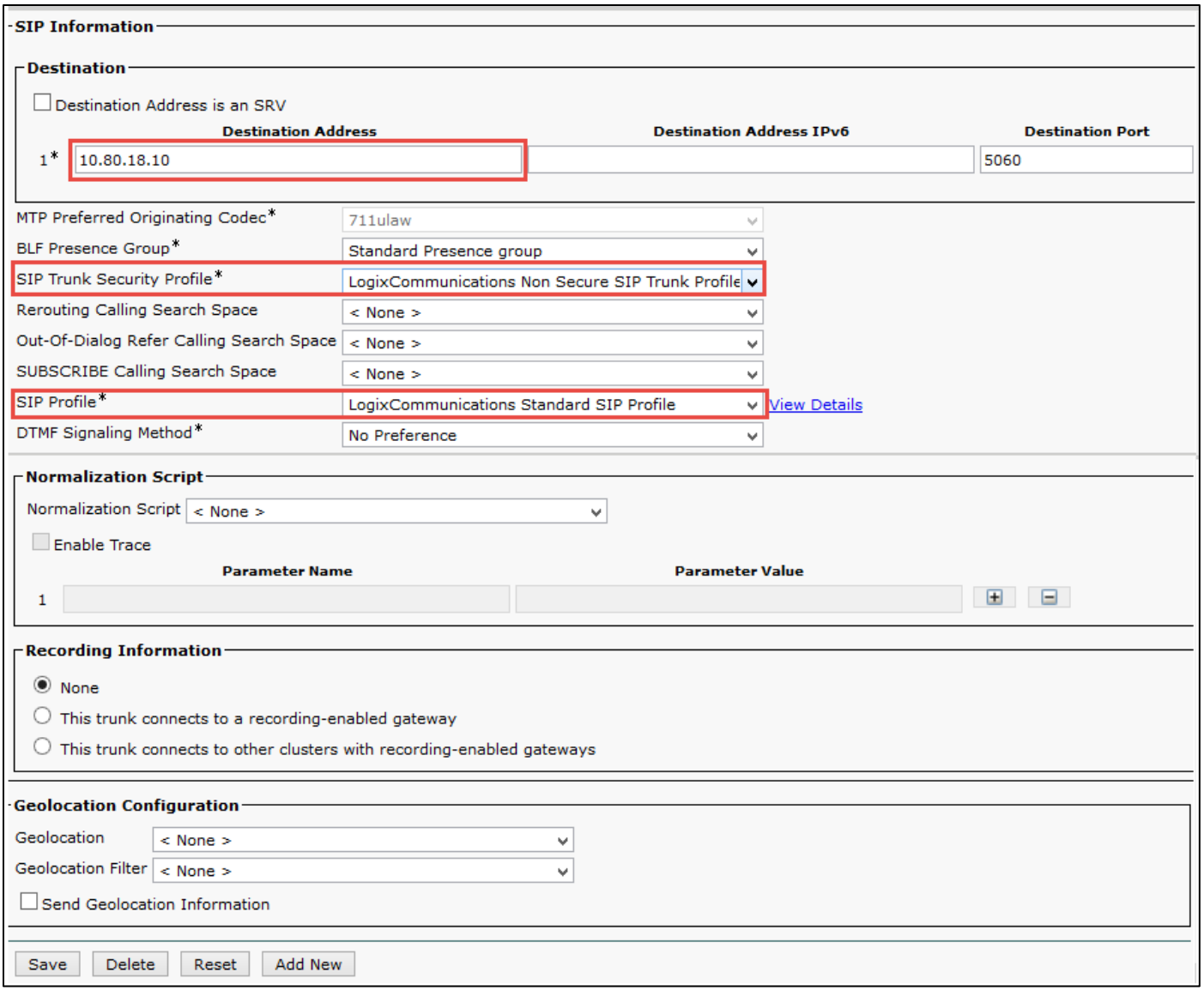


Figure 18: SIP Trunk to Cisco UBE (Cont.)

Explanation

Parameter	Value	Description
Device Name	Trunk_to_CUBE_for_LogixCommunications	Name for the trunk
Device Pool	LogixCommunication_Devicepool	Default Device Pool is used for this trunk
Media Resource Group List	MRGL_Default	MRG with resources: ANN, CFB, MOH and MTP
Significant Digits	4	4 digits Extension for all CPE phones
Destination Address	10.80.18.10	IP address of the Cisco UBE Virtual LAN
SIP Trunk Security Profile	LogixCommunications Non Secure SIP Trunk Profile	SIP Trunk Security Profile configured earlier
SIP Profile	LogixCommunications Standard SIP Profile	SIP Profile configured earlier

Dial Plan

Route Pattern Configuration

Navigation: Call Routing → Route/Hunt → Route Pattern

Route patterns are configured as below:

1. Cisco IP phone dial “8”+10 digits number to access PSTN via Cisco UBE
   - “8” is removed before sending to Cisco UBE
2. For FAX call, Access Code “8”+10 digits number is used at Cisco Fax gateway
   - “8” is removed at Cisco UCM
   - The rest of the number is sent to Cisco UBE to Logix Communications network
3. Incoming fax call to 14XX will be sent to Cisco Fax gateway
4. For Anonymous call, access code “8*”+10 digits number is used
   - “8*” is removed at Cisco UCM
   - The rest of the number is sent to Cisco UBE to Logix Communications network
5. Cisco IP phones dial X11 for emergency call and will send all digits to Cisco UBE to Logix Communications Network

Route Patterns *(1 - 8 of 8)* *Rows per Page* 50

Find Route Patterns where Pattern begins with Find Clear Filter

Pattern	Description	Partition	Route Filter	Associated Device	Copy
14XX	RP_for_LogixCommunicationsFAX			Trunk_to_FAX_gateway_for_Logix	
8*.@	RP_for_LogixCommunications-for Anonymous Calls			Trunk_to_CUBE_for_LogixCommunications	
8.@	RP_for_LogixCommunications			Trunk_to_CUBE_for_LogixCommunications	
X11	Emergency Calling			Trunk_to_CUBE_for_LogixCommunications	

Figure 19: Route Patterns List

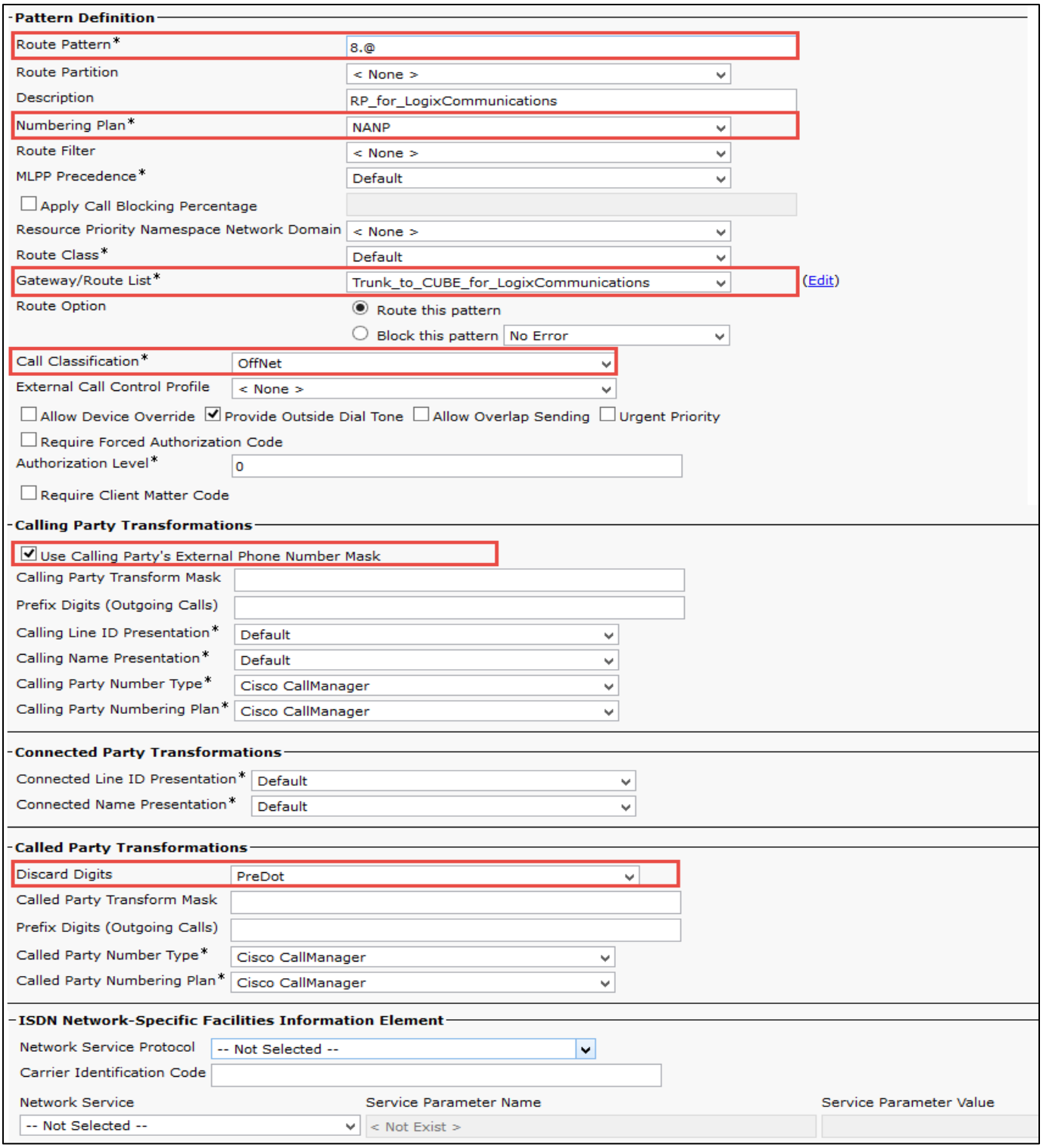


Figure 20: Route Pattern for Voice

Pattern Definition

Route Pattern*	14XX
Route Partition	< None >
Description	RP_for_LogixCommunicationsFAX
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*	Trunk_to_FAX_gateway_for_Logix (Edit)
Route Option	◉ Route this pattern
	○ Block this pattern No Error
Call Classification*	OffNet
External Call Control Profile	< None >

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

☐ Require Forced Authorization Code

Authorization Level* 0

☐ Require Client Matter Code

Calling Party Transformations

☑ Use Calling Party's External Phone Number Mask

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

Connected Party Transformations

Connected Line ID Presentation*	Default
Connected Name Presentation*	Default

Called Party Transformations

Discard Digits	< None >
Called Party Transform Mask	469708XXXX
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 21: Route Pattern for Fax

Pattern Definition

Route Pattern*	8*.@
Route Partition	< None >
Description	RP_for_LogixCommunications-for Anonymous Calls
Numbering Plan*	NANP
Route Filter	< None >
MLPP Precedence*	Default
☐ Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	Trunk_to_CUBE_for_LogixCommunications (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*	Restricted
Calling Party Number Type*	Cisco CallManager
Calling Party Numbering Plan*	Cisco CallManager

Connected Party Transformations

Connected Line ID Presentation*	Default
Connected Name Presentation*	Default

Called Party Transformations

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

ISDN Network-Specific Facilities Information Element

Network Service Protocol	-- Not Selected --
Carrier Identification Code	

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

Figure 22: Route Pattern for Anonymous Call

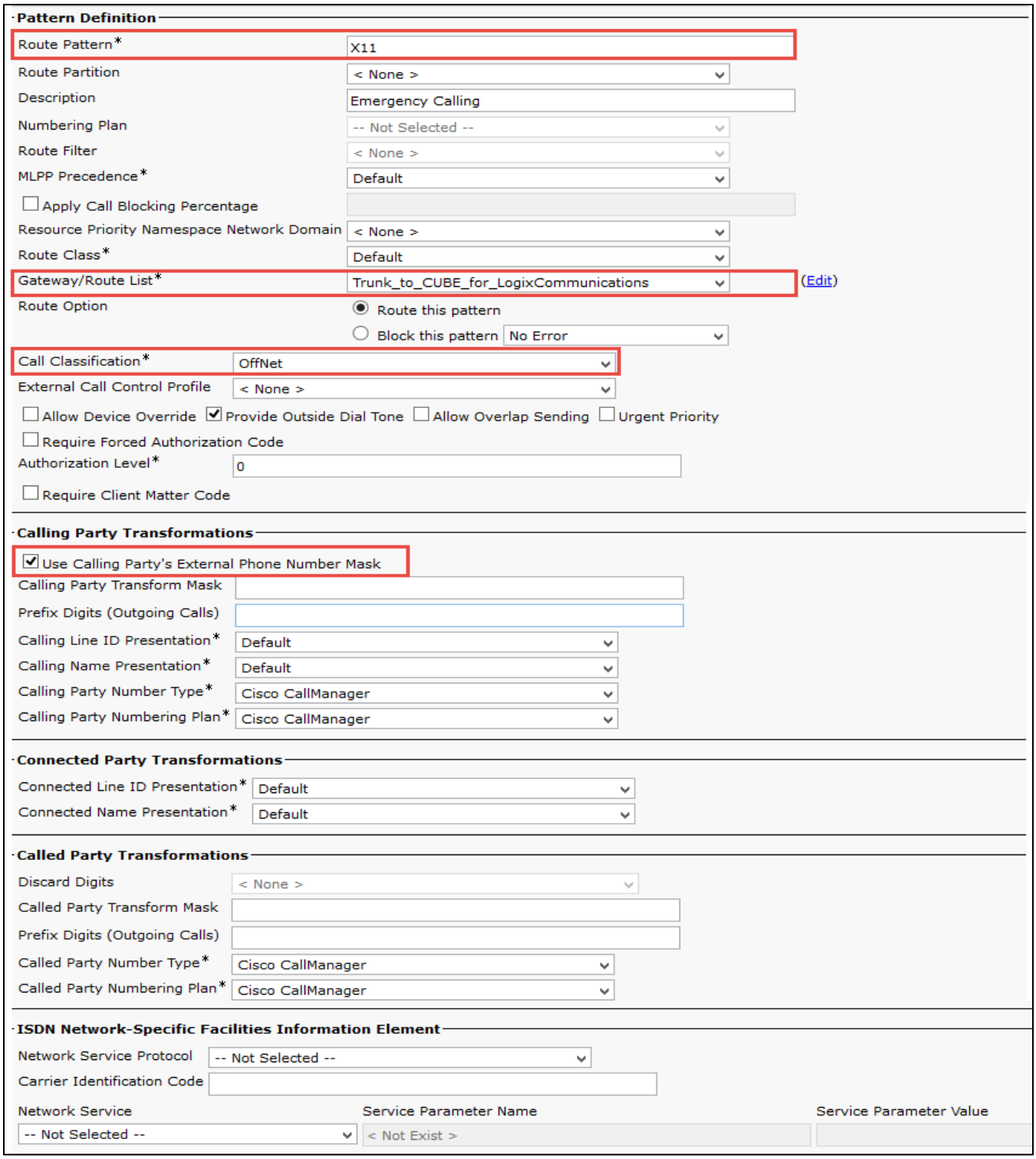


Figure 23: Route Pattern for Emergency Call

Explanation

Setting	Value	Description
Route Pattern	8.@ for Voice & International Calls, 14XX for Fax Call, 8*.@ for Anonymous Calls and X11 for Operator Call and Emergency Services	Specify appropriate Route Pattern
Gateway/Route List	Trunk_to_CUBE_for_LogixCommunications for Route Pattern 8.@, 8*.@, X11 and Trunk_to_FAX_gateway_for_Logix for Route Pattern 14XX	SIP Trunk name configured earlier
Numbering Plan	NANP for Route Pattern 8.@, 8*.@	North American Numbering Plan
Call Classification	Off-Net for Route Pattern 8.@, 8*.@, 14XX and X11	Restrict the transferring of an external call to an external device
Discard Digits	PreDot for Route Pattern 8.@, 8*.@	Specifies how to modify digit before they are sending to Logix network

Acronyms

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

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