Application Note

Fusion Connect SIP Trunking:

Cisco Unified Communications Manager 11.5.1 with Cisco Unified Border Element (CUBE 11.5.2) on ISR4321/K9 [IOS 16.3(1)] using SIP

August 2016

Table of Contents

Introduction 4
Network Topology 5
System Components 6
Hardware Requirements 6
Software Requirements 6
Features 7
Features Supported 7
Features Not Supported 7
Caveats 7
Configuration 8
Configuring the Cisco Unified Border Element for Registration SIP Trunk Testing 8
Network interface 8
Global Cisco UBE settings 9
Codecs 10
Dial peer 10
Configuration example 13
Configuring the Cisco Unified Border Element for Static SIP Trunk Testing 31
Network interface 31
Global Cisco UBE settings 32
Codecs 33
Dial peer 33
Configuration example 35
Call flow 47
Configuring Cisco Unified Communications Manager 49
Cisco UCM Version 49
Cisco Call Manager Service Parameters 49
Offnet Calls via Fusion Connect SIP Trunk 51
Dial Plan 60
Acronyms 65
Important Information 66
Test Results 67

Table of Figures

Figure 1: Network Topology 5
Figure 2: Cisco UBE High Availability........ 5
Figure 3: Outbound Voice Call 47
Figure 4: Outbound Fax Call 47
Figure 5: Inbound Voice Call 48
Figure 6 : Inbound Fax Call 48
Figure 7 : PBX to PBX via Fusion Connect Call........ 48
Figure 8: Cisco UCM Version 49
Figure 9: Service Parameters........ 49
Figure 10: Service Parameters (Cont.)........ 50
Figure 11: SIP Trunk Security Profile 51
Figure 12: SIP Profile........ 52
Figure 13: SIP Profile (Cont.) 53
Figure 14: SIP Profile (Cont.) 54
Figure 15: SIP Trunks List........ 55
Figure 16: SIP Trunk to Cisco UBE........ 56
Figure 17: SIP Trunk to Cisco UBE (Cont.)........ 57
Figure 18: SIP Trunk to Cisco UBE (Cont.)........ 58
Figure 19: Route Patterns List........ 60
Figure 20: Route Pattern for Voice 61
Figure 21: Route Pattern for Voice (Cont.)........ 62
Figure 22: Route Pattern for Fax........ 63

Introduction

Service Providers today, such as Fusion Connect, 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.

Fusion Connect 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 and Fusion Connect network, Cisco Unified Border Element (Cisco UBE) ISR 4321/K9 running IOS 16.3.1 can be used. The Cisco Unified Border Element 16.3.1 provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 11.5.1 connected to Fusion Connect 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 Fusion Connect 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/K9 [IOS – 16.3.1] for connectivity to Fusion Connect SIP Trunking service. The deployment model covered in this application note is CPE (Cisco Unified Communications Manager 11.5.1 to PSTN (Fusion Connect).
- Testing was performed in accordance to Fusion Connect 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 Fusion Connect 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 Fusion Connect 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

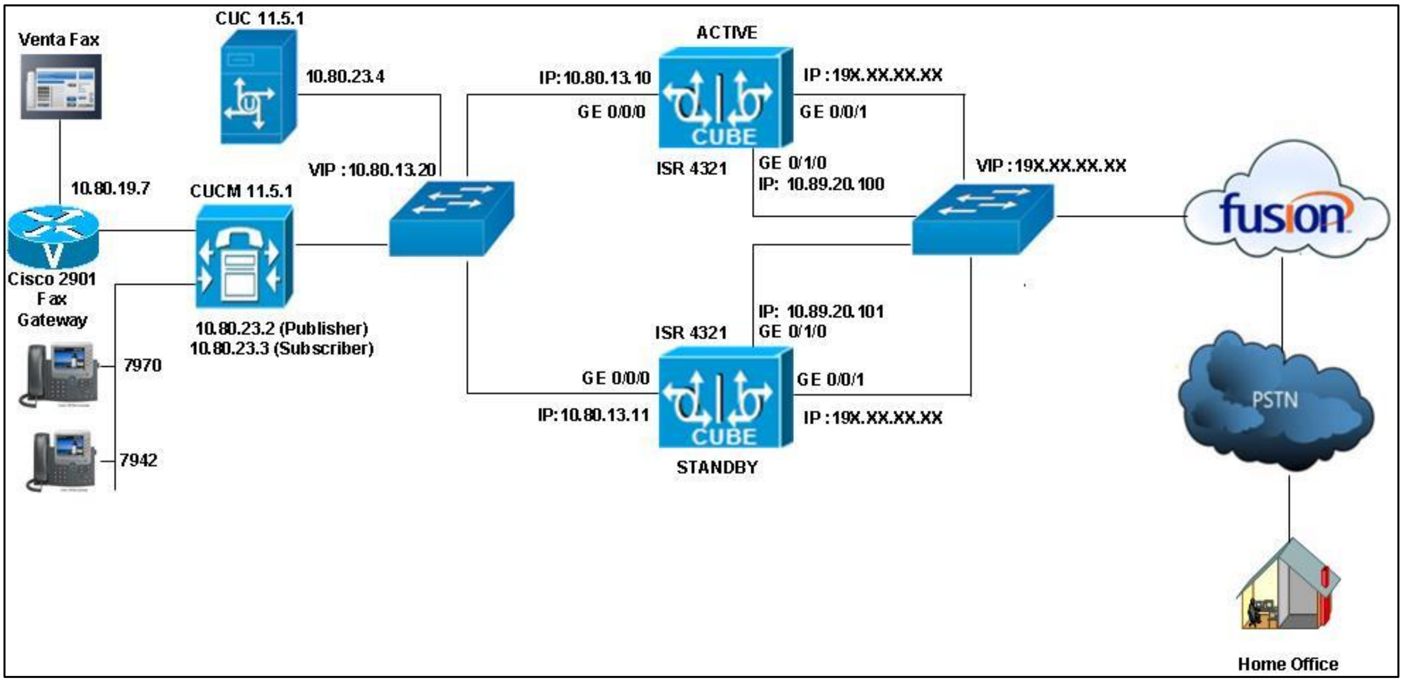


Figure 1: Network Topology

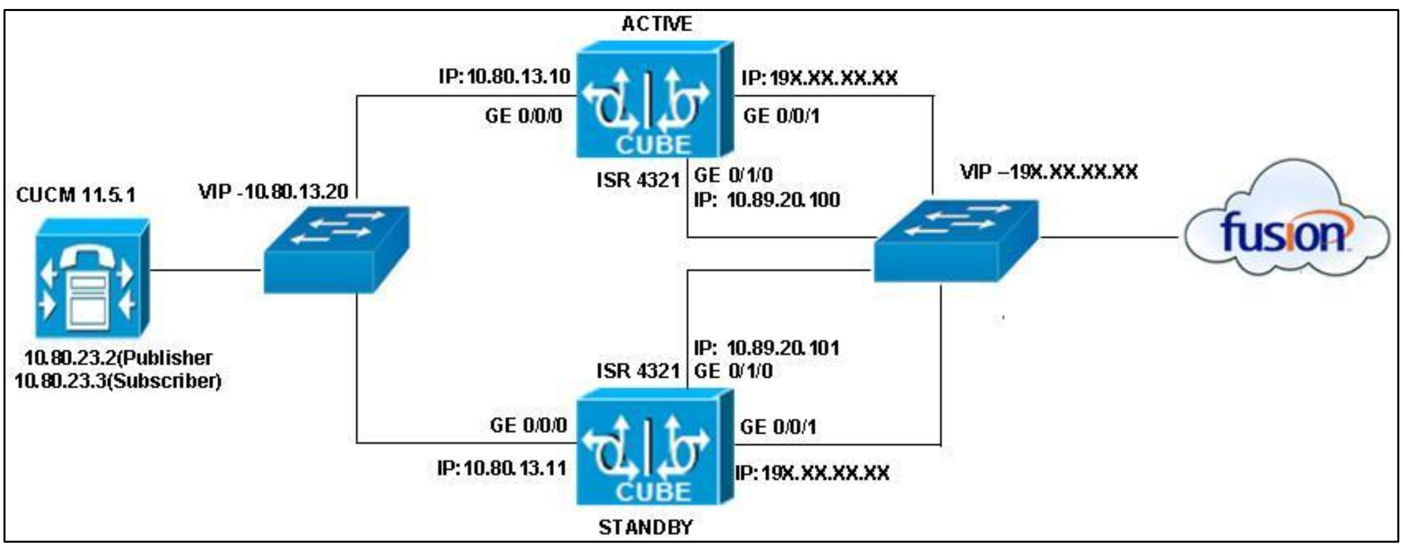


Figure 2: Cisco UBE High Availability

System Components

Hardware Requirements

- Cisco UCS-C240-M3S VMWare host running ESXi 5.5 Standard
- Cisco ISR 4321/K9 router as CUBE
- Cisco ISR 4321/K9 (1RU) processor with 1652953K/6147K bytes of memory with 3 Gigabit Ethernet interfaces
- Processor board ID FLM1925W0X0
- Cisco 2901 Fax Gateway
- IP phones 7970 (SIP), 7942 (SCCP)

Software Requirements

- Cisco Unified Communications Manager 11.5.1.10000-6
- Cisco Unity Connection 11.5.1.10000-6
- IOS 16.3.1 for ISR 4321/K9 Cisco Unified Border Element
- Cisco IOS Software, ISR Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 16.3.1, RELEASE SOFTWARE (fc3)
- Cisco IOS XE Software, Version 16.3.1
- IOS 15.4 XA for Cisco 2901 Fax Gateway

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)
- DTMF relay (both directions) (RFC2833)
- Media flow-through on Cisco UBE
- Fax (T38 & G711 pass-through)

Features Not Supported

- 0 and 0+10 digit dial plan - Operator assisted calls are not supported by Fusion Connect at the moment
- Privacy ID feature is not supported by Fusion Connect
- Rel100 feature is not supported by Fusion Connect

Caveats

- In HA Redundancy mode the Primary CUBE will not take over the Primary/Active role after a reboot/network outage
- 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.
- Fusion Connect supports Faxing up to G3/V.17 with T.38 Version 0. However, Fusion supports faxing with SuperG3/V.34 over G711.
- Early media calls that requires PRACK with SDP failed with enabling "require100rel" since Fusion Connect does not support "require100rel". However, the call was successful without PRACK.
- Fusion Connect does not support "Privacy=id". But Fusion Connect can support Privacy ID as per customer request.
- Fusion Connect has its own Music on Hold features and if the customer would like to use the PBX hold music, Fusion will support the Hold INVITE utilizing "a=sendreceive" instead of "a=sendonly"
- Fusion Connect requires 10 digit BTN number in From header to process the basic Outbound calls successfully

Configuration

Configuring the Cisco Unified Border Element for Registration SIP Trunk Testing

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 FusionConnect LAN
 ip address 10.80.13.10 255.255.255.0
 media-type rj45
 negotiation auto
 redundancy rii 8
 redundancy group 1 ip 10.80.13.20 exclusive
!
interface GigabitEthernet0/0/1
 description FusionConnect WAN
 ip address 1XX.XX.XX.XX 255.255.255.128
 negotiation auto
 redundancy rii 9
 redundancy group 1 ip 1XX.XX.XX.XX exclusive
!
```

Global Cisco UBE settings

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

```
voice service voip
 ip address trusted list
  ipv4 2XX.XX.XX.XX
  ipv4 2XX.XX.XX.XX
  address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 1
 fax protocol pass-through g711ulaw
 sip
  bind control source-interface GigabitEthernet0/0/1
   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
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 used to change according to the test plan description

```
 voice class codec 1
  codec preference 1 g711ulaw
  codec preference 2 g729r8
  codec preference 3 g722-56
```

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 from PBX to PSTN-LAN facing
 huntstop
 session protocol sipv2
 session target sip-server
 session transport udp
 incoming called-number [0-9]T
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 510 voip
```

```
 description Outgoing Call from PBX to PSTN-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 options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 520 voip
 description Incoming Call to PBX from PSTN-WAN facing
 huntstop
 session protocol sipv2
 session target sip-server
 session transport udp
 incoming called-number XXXXXX....
 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 disable
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 530 voip
 description Incoming Call to PBX from PSTN-LAN facing
 huntstop
 destination-pattern XXXXXX....
 session protocol sipv2
 session target ipv4:10.80.23.3:5060
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 no voice-class sip outbound-proxy
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
```

Configuration example

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

Active Cisco UBE

User Access Verification

Password:

fusionConnect1>en

Password:

fusionConnect1#sh run

Building configuration...

Current configuration : 7181 bytes

!

! Last configuration change at 15:10:58 UTC Mon Aug 29 2016

!

version 16.3

service timestamps debug datetime msec

service timestamps log datetime msec

service internal

service sequence-numbers

no platform punt-keepalive disable-kernel-core

!

hostname fusionConnect1

!

boot-start-marker

boot system flash bootflash:isr4300-universalk9.16.03.01.SPA.bin

boot-end-marker

!

vrf definition Mgmt-intf

```
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
logging queue-limit 10000
logging buffered 10000000
no logging rate-limit
no logging console
no logging monitor
enable secret 5
!
no aaa new-model
!
subscriber templating
!
multilink bundle-name authenticated
!
voice service voip
 ip address trusted list
  ipv4 2XX.XX.XX.XX
  ipv4 2XX.XX.XX.XX
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 1
 fax protocol pass-through g711ulaw
 sip
```

```
  bind control source-interface GigabitEthernet0/0/1
  bind media source-interface GigabitEthernet0/0/1
  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
 codec preference 3 g722-56
!
voice class codec 2
 codec preference 1 g729r8
 codec preference 2 g711ulaw
 codec preference 3 g722-56
!
voice class sip-profiles 101
 request INVITE sip-header From modify "<(.*)>" "<sip:XXXXXXXXXX@1XX.XX.XX.XX>"
 request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:XXXXXX\1@\2"
!
license udi pid ISR4321/K9 sn FDO19220MQ8
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 b2bhafusionConnect
   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 FusionConnect LAN
 ip address 10.80.13.10 255.255.255.0
 media-type rj45
 negotiation auto
 redundancy rii 8
 redundancy group 1 ip 10.80.13.20 exclusive
!

interface GigabitEthernet0/0/1
```

```
 description FusionConnect WAN
 ip address 1XX.XX.XX.XX 255.255.255.128
 negotiation auto
 redundancy rii 9
 redundancy group 1 ip 1XX.XX.XX.XX exclusive
!
interface GigabitEthernet0/1/0
 description CUBE HA
 ip address 10.89.20.100 255.255.255.0
 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 route 0.0.0.0 0.0.0.0 1XX.XX.XX.XX
ip route 10.80.0.0 255.255.0.0 10.80.13.1
ip route 172.16.0.0 255.255.0.0 10.80.13.1
!
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 from PBX to PSTN-LAN facing
 huntstop
 session protocol sipv2
 session target sip-server
 session transport udp
 incoming called-number [0-9]T
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 510 voip
 description Outgoing Call from PBX to PSTN-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 options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 520 voip
 description Incoming Call to PBX from PSTN-WAN facing
 huntstop
 session protocol sipv2
 session target sip-server
 session transport udp
 incoming called-number XXXXXX....
 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 disable
```

```
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 530 voip
 description Incoming Call to PBX from PSTN-LAN facing
 huntstop
 destination-pattern XXXXXX....
 session protocol sipv2
 session target ipv4:10.80.23.3:5060
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 no voice-class sip outbound-proxy
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 540 voip
 description Incoming Call to PBX from PSTN-WAN facing (for alternate DID which differs from the given DID range)
 huntstop
 session protocol sipv2
 session target sip-server
 session transport udp
```

```
 incoming called-number 3XXXXX....
 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
 no vad
!
dial-peer voice 550 voip
 description Incoming Call to PBX from PSTN-LAN facing (for alternate DID which differs from the given DID range)
 huntstop
 destination-pattern 3XXXXX....
 session protocol sipv2
 session target ipv4:10.80.23.3:5060
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 no voice-class sip outbound-proxy
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 no vad
!
sip-ua
 credentials number XXXXXXXXXX username XXXXXXXXXX password 7 realm asterisk
 keepalive target ipv4:2XX.XX.XX.XX
 authentication username XXXXXXXXXX password 7
 timers keepalive active 10
```

```
 registrar ipv4:2XX.XX.XX.XX expires 60
 sip-server ipv4:2XX.XX.XX.XX:5060
!
line con 0
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 exec-timeout 0 0
 password XXXXXX
 login
!
end
```

Standby Cisco UBE

```
User Access Verification
Password:
fusionConnect2>en
Password:
fusionConnect2#sh run
Building configuration...
Current configuration : 7391 bytes
!
! Last configuration change at 08:03:26 UTC Fri Sep 2 2016
!
version 16.3
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
service internal
service sequence-numbers
no platform punt-keepalive disable-kernel-core
!
hostname fusionConnect2
!
boot-start-marker
boot system flash bootflash:isr4300-universalk9.16.03.01.SPA.bin
boot-end-marker
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
logging queue-limit 10000
logging buffered 10000000
no logging rate-limit
no logging console
```

```
enable secret 5
!
no aaa new-model
!
subscriber templating
!
multilink bundle-name authenticated
!
voice service voip
 ip address trusted list
  ipv4 2XX.XX.XX.XX
  ipv4 2XX.XX.XX.XX
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 1
 fax protocol pass-through g711ulaw
 sip
  bind control source-interface GigabitEthernet0/0/1
  bind media source-interface GigabitEthernet0/0/1
  rel1xx supported "rel100"
  session refresh
  asserted-id pai
  privacy pstn
  outbound-proxy ipv4:2XX.XX.XX.XX
  early-offer forced
  midcall-signaling passthru
  g729 annexb-all
!
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 g729r8
 codec preference 3 g722-56
!
voice class codec 2
 codec preference 1 g729r8
```

```
 codec preference 2 g711ulaw
 codec preference 3 g722-56
!
voice class sip-profiles 101
 request INVITE sip-header From modify "<(.*)>" "<sip:XXXXXXXXXX@1XX.XX.XX.XX>"
 request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:XXXXXX\1@\2"
!
license udi pid ISR4321/K9 sn FDO19220MQ9
!
diagnostic bootup level minimal
spanning-tree extend system-id
!
redundancy
 mode none
 application redundancy
  group 1
   name b2bhafusionConnect
   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 FusionConnect LAN
 ip address 10.80.13.11 255.255.255.0
 media-type rj45
 negotiation auto
 redundancy rii 8
```

```
 redundancy group 1 ip 10.80.13.20 exclusive
!
interface GigabitEthernet0/0/1
 description FusionConnect WAN
 ip address 1XX.XX.XX.XX 255.255.255.128
 negotiation auto
 redundancy rii 9
 redundancy group 1 ip 1XX.XX.XX.XX exclusive
!
interface GigabitEthernet0/1/0
 description CUBE HA
 ip address 10.89.20.101 255.255.255.0
 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 route 0.0.0.0 0.0.0.0 1XX.XX.XX.XX
ip route 10.80.0.0 255.255.0.0 10.80.13.1
ip route 172.16.0.0 255.255.0.0 10.80.13.1
!
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 from PBX to PSTN-LAN facing
 huntstop
 session protocol sipv2
 session target sip-server
 session transport udp
 incoming called-number [0-9]T
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 510 voip
 description Outgoing Call from PBX to PSTN-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 options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
```

```
 fax-relay ecm disable
 fax rate disable
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 520 voip
 description Incoming Call to PBX from PSTN-WAN facing
 huntstop
 session protocol sipv2
 session target sip-server
 session transport udp
 incoming called-number XXXXXX....
 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 disable
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 530 voip
 description Incoming Call to PBX from PSTN-LAN facing
 huntstop
 destination-pattern XXXXXX....
 session protocol sipv2
 session target ipv4:10.80.23.3:5060
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 no voice-class sip outbound-proxy
 voice-class sip options-keepalive
```

```
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
  fax-relay ecm disable
 fax rate disable
 fax nsf 000000
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 540 voip
 description Incoming Call to PBX from PSTN-WAN facing (for Alternate DID which differs from given range)
 huntstop
 session protocol sipv2
 session target sip-server
 session transport udp
 incoming called-number 3XXXXX....
 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
 no vad
!
dial-peer voice 550 voip
 description Incoming Call to PBX from PSTN-LAN facing (for Alternate DID which differs from given range)
 huntstop
 destination-pattern 3XXXXX....
 session protocol sipv2
 session target ipv4:10.80.23.3:5060
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 no voice-class sip outbound-proxy
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/0
```

```
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 no vad
!
sip-ua
 credentials number XXXXXXXXXX username XXXXXXXXXX password 7
 keepalive target ipv4:2XX.XX.XX.XX
 authentication username XXXXXXXXXX password 7
 timers keepalive active 10
 registrar ipv4:2XX.XX.XX.XX expires 60
 sip-server ipv4:2XX.XX.XX.XX:5060
!
line con 0
 exec-timeout 0 0
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 exec-timeout 0 0
 password XXXXXX
 login
!
end
```

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 FusionConnect LAN
 ip address 10.80.13.10 255.255.255.0
 media-type rj45
 negotiation auto
 redundancy rii 8
 redundancy group 1 ip 10.80.13.20 exclusive
!
interface GigabitEthernet0/0/1
 description FusionConnect WAN
 ip address 1XX.XX.XX.XX 255.255.255.128
 negotiation auto
 redundancy rii 9
 redundancy group 1 ip 1XX.XX.XX.XX exclusive
```

Global Cisco UBE settings

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

```
voice service voip
 ip address trusted list
  ipv4 2XX.XX.XX.XX
  ipv4 2XX.XX.XX.XX
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 1
 sip
  bind control source-interface GigabitEthernet0/0/1
  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 used to change according to the test plan description

```
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 g729r8
 codec preference 3 g722-56
```

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 FusionConnect - 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 GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 510 voip
 description Outgoing call to FusionConnect - 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 options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/1
```

```
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 520 voip
 description Incoming call to PBX - WAN facing
 huntstop
 session protocol sipv2
 session transport udp
 incoming called-number XXXXXX...
 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
 no vad
!
dial-peer voice 530 voip
 description Incoming call to PBX - LAN facing
 huntstop
 destination-pattern XXXXXX....
 session protocol sipv2
 session target ipv4:10.80.23.3
 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 GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 no vad
!
```

Configuration example

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

Active Cisco UBE

```
User Access Verification
Password:
fusionConnect1>en
Password:
fusionConnect1#sh run
Building configuration...

Current configuration : 5308 bytes
! Last configuration change at 06:47:07 UTC Thu Aug 11 2016
version 16.3
service timestamps debug datetime msec
service timestamps log datetime msec
service internal
service sequence-numbers
no platform punt-keepalive disable-kernel-core
!
hostname fusionConnect1
!
boot-start-marker
boot system flash bootflash:isr4300-universalk9.16.03.01.SPA.bin
boot-end-marker
!
vrf definition Mgmt-intf
 !

 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
logging queue-limit 10000
logging buffered 10000000
```

```
no logging rate-limit
no logging console
logging monitor notifications
enable secret 5
!
no aaa new-model
!
subscriber templating
!
multilink bundle-name authenticated
!
voice service voip
 ip address trusted list
  ipv4 2XX.XX.XX.XX
  ipv4 2XX.XX.XX.XX
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 1
 sip
  bind control source-interface GigabitEthernet0/0/1
  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
 codec preference 3 g722-56
!
voice class codec 2
 codec preference 1 g729r8
 codec preference 2 g711ulaw
 codec preference 3 g722-56
```

```
!
license udi pid ISR4321/K9 sn FDO19220MQ8
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 b2bhafusionConnect
   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 FusionConnect LAN
 ip address 10.80.13.10 255.255.255.0
 media-type rj45
 negotiation auto
 redundancy rii 8
 redundancy group 1 ip 10.80.13.20 exclusive
!
interface GigabitEthernet0/0/1
 description FusionConnect WAN
 ip address 1XX.XX.XX.XX 255.255.255.128
 negotiation auto
```

```
 redundancy rii 9
 redundancy group 1 ip 1XX.XX.XX.XX exclusive
!
interface GigabitEthernet0/1/0
 description CUBE HA
 ip address 10.89.20.100 255.255.255.0
 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 route 0.0.0.0 0.0.0.0 19X.XX.XX.XX
ip route 10.80.0.0 255.255.0.0 10.80.13.1
ip route 172.16.0.0 255.255.0.0 10.80.13.1
!
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 FusionConnect - 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 GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 510 voip
 description Outgoing call to FusionConnect - 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 options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 520 voip
 description Incoming call to PBX - WAN facing
 huntstop
 session protocol sipv2
 session transport udp
 incoming called-number XXXXXX...
 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
 no vad
!
dial-peer voice 530 voip
```

```
 description Incoming call to PBX - LAN facing
 huntstop
 destination-pattern XXXXXX....
 session protocol sipv2
 session target ipv4:10.80.23.3
 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 GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 no vad
!
sip-ua
 keepalive target ipv4:2XX.XX.XX.XX
 timers keepalive active 60
 sip-server ipv4:2XX.XX.XX.XX
!
line con 0
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 exec-timeout 0 0
 password XXXXXX
 login
!
end
```

Standby Cisco UBE

```
User Access Verification
Password:
fusionConnect2>en
Password:
fusionConnect2#sh run
Building configuration...

Current configuration : 5023 bytes
!
! Last configuration change at 07:04:55 UTC Tue Sep 6 2016
!
version 16.3
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
service internal
service sequence-numbers
no platform punt-keepalive disable-kernel-core
!
hostname fusionConnect2
!
boot-start-marker
boot system flash bootflash:isr4300-universalk9.16.03.01.SPA.bin
boot-end-marker
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
logging queue-limit 10000
logging buffered 10000000
no logging rate-limit
no logging console
```

```
enable secret 5
!
no aaa new-model
!
subscriber templating
!
multilink bundle-name authenticated
!
voice service voip
 ip address trusted list
  ipv4 2XX.XX.XX.XX
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 1
 sip
  bind control source-interface GigabitEthernet0/0/1
  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
 codec preference 3 g722-56
!
voice class codec 2
 codec preference 1 g729r8
 codec preference 2 g711ulaw
 codec preference 3 g722-56
!
license udi pid ISR4321/K9 sn FDO19220MQ9
!
diagnostic bootup level minimal
```

```
spanning-tree extend system-id
!
redundancy
 mode none
 application redundancy
  group 1
   name b2bhafusionConnect
   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 FusionConnect LAN
 ip address 10.80.13.11 255.255.255.0
 media-type rj45
 negotiation auto
 redundancy rii 8
 redundancy group 1 ip 10.80.13.20 exclusive
!
interface GigabitEthernet0/0/1
 description FusionConnect WAN
 ip address 1XX.XX.XX.XX 255.255.255.128
 negotiation auto
 redundancy rii 9
 redundancy group 1 ip 1XX.XX.XX.XX exclusive
!
interface GigabitEthernet0/1/0
 description CUBE HA
 ip address 10.89.20.101 255.255.255.0
```

```
 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 route 0.0.0.0 0.0.0.0 1XX.XX.XX.XX
ip route 10.80.0.0 255.255.0.0 10.80.13.1
ip route 172.16.0.0 255.255.0.0 10.80.13.1
!
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 FusionConnect - 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 GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
```

```
 no vad
!
dial-peer voice 510 voip
 description Outgoing call to FusionConnect - 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 options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 520 voip
 description Incoming call to PBX - WAN facing
 huntstop
 session protocol sipv2
 session transport udp
 incoming called-number XXXXXX....
 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
 no vad
!
dial-peer voice 530 voip
 description Incoming call to PBX - LAN facing
 huntstop
 destination-pattern XXXXXX....
 session protocol sipv2
 session target ipv4:10.80.23.3
 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 GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 no vad
!
sip-ua
 keepalive target ipv4:2XX.XX.XX.XX
 timers keepalive active 60
 sip-server ipv4:2XX.XX.XX.XX:5060
!
line con 0
 exec-timeout 0 0
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 exec-timeout 0 0
 password XXXXXX
 login
!
end
```

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 “6” 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 “6”. A “6.@” 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 Fusion Connect, Caller dial 7 prefix followed by the target 10-digits number, 6 was stripped and the remaining digits were send to Cisco UBE, Cisco UBE pass the DID under Dial Peer 510 and send to Fusion Connect network which will direct back to Cisco UBE and handled same as normal incoming PSTN call.

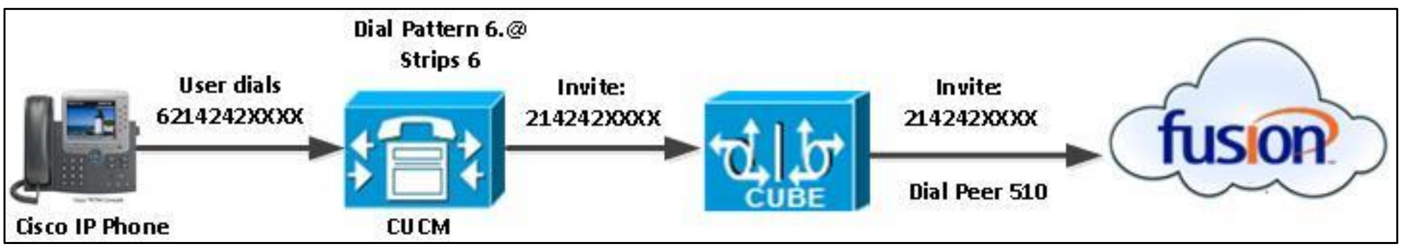


Figure 3: Outbound Voice Call

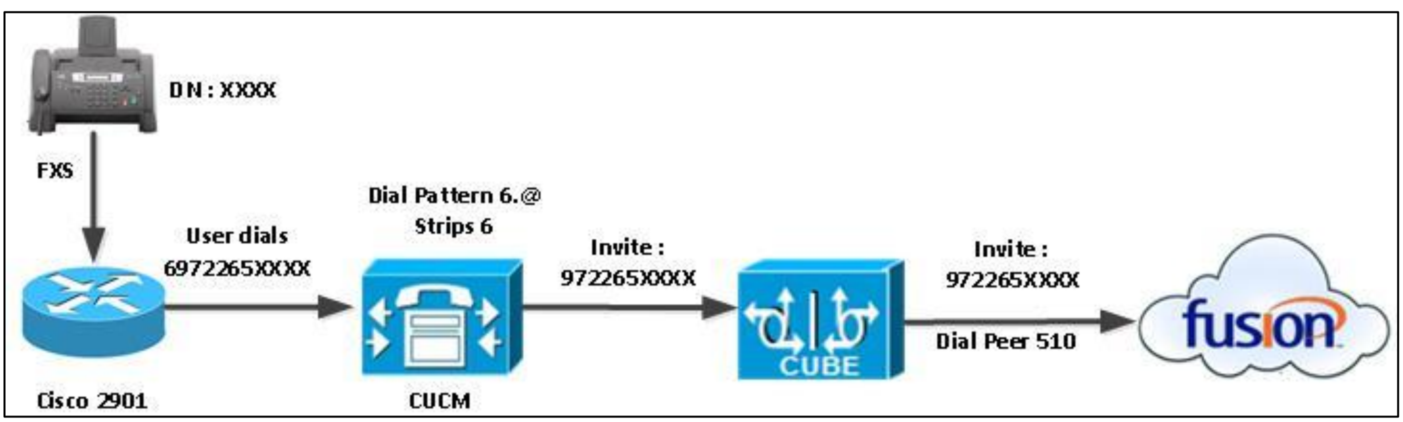


Figure 4: Outbound Fax Call

Figure 5: Inbound Voice Call

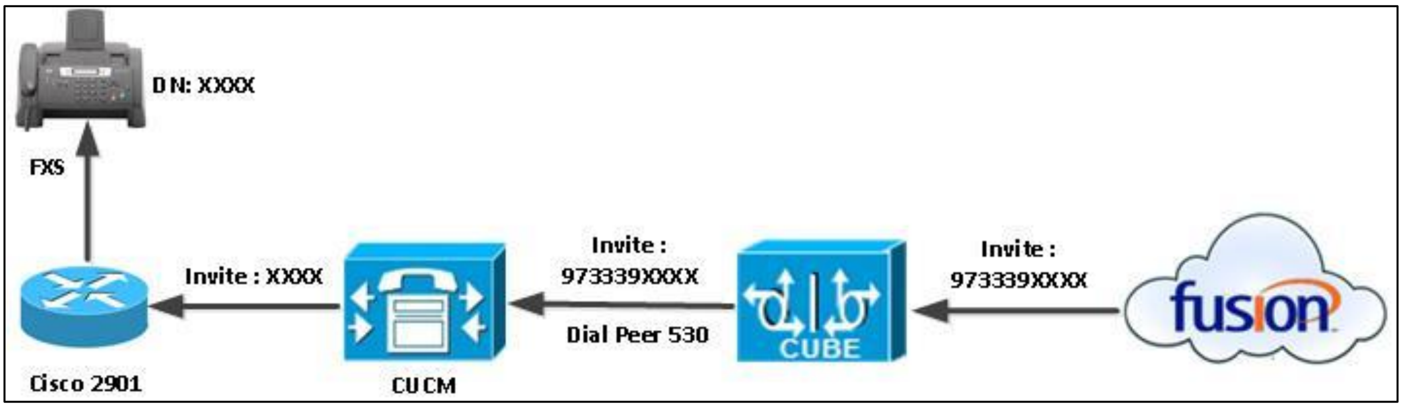


Figure 6 : Inbound Fax Call

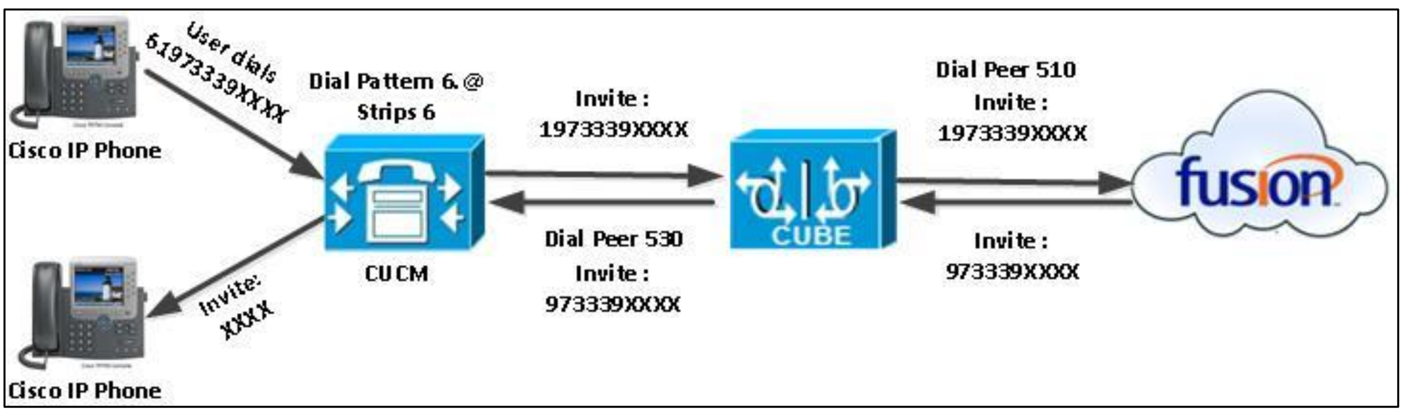


Figure 7 : PBX to PBX via Fusion Connect Call

Cisco UCM Version

Figure 8: Cisco UCM Version

Cisco Call Manager Service Parameters

Navigation: System → Service Parameters

1. **Server** = clus33pub--CUCM Voice/Video (Active)
2. **Service** = Cisco CallManager (Active)
3. **Duplex Streaming Enabled** = True
4. All other fields are set to default values

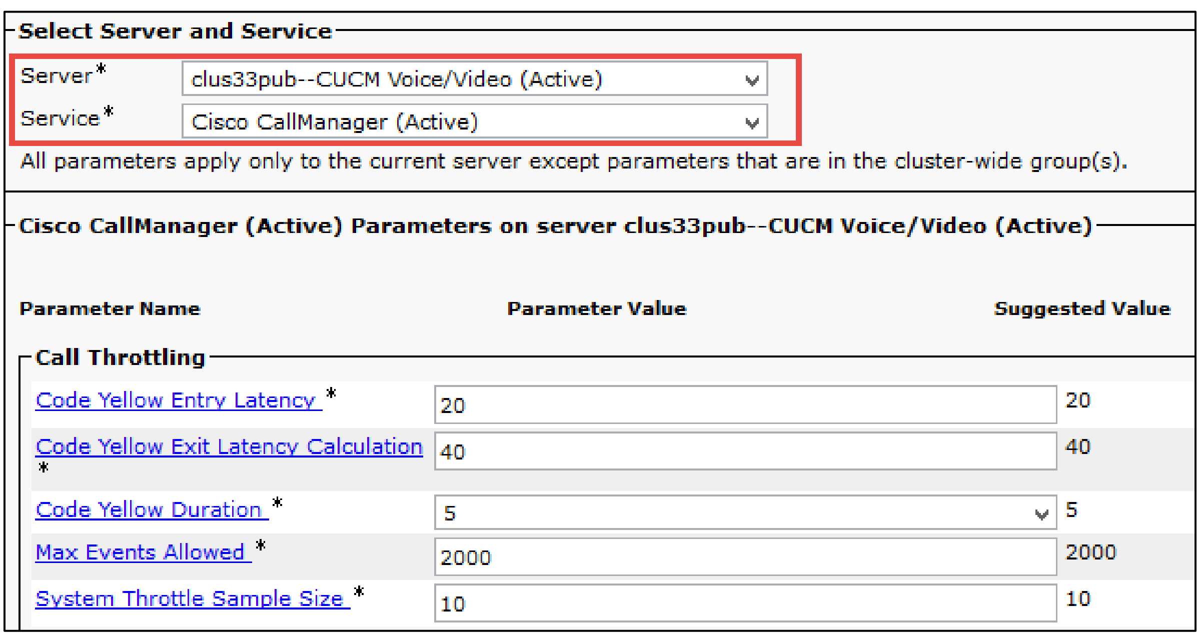


Figure 9: Service Parameters

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 *	False	False
Silence Suppression for Gateways *	False	False
Strip G.729 Annex B (Silence Suppression) from Capabilities *	False	False
Enable Source IP Address Verification for Software Media Devices *	True	True

Figure 10: Service Parameters (Cont.)

Offnet Calls via Fusion Connect SIP Trunk

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

SIP Trunk Security Profile

Navigation: System → Security → SIP Trunk Security Profile

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

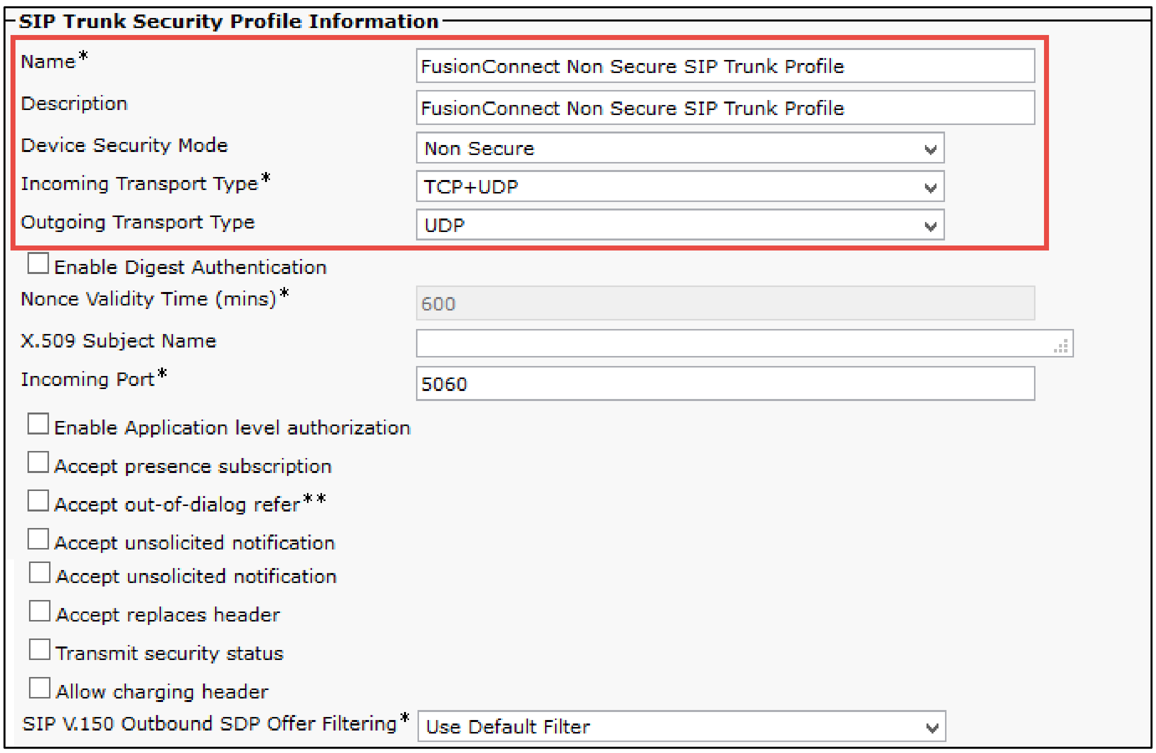


Figure 11: SIP Trunk Security Profile

Explanation

Parameter	Value	Description
Incoming Transport Type	TCP + UDP	
Outgoing Transport Type	UDP	SIP trunks to Fusion Connect 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** = *Fusion Connect SIP Profile* is used as an example
2. **Description** = *Fusion Connect SIP Profile* is used as an example

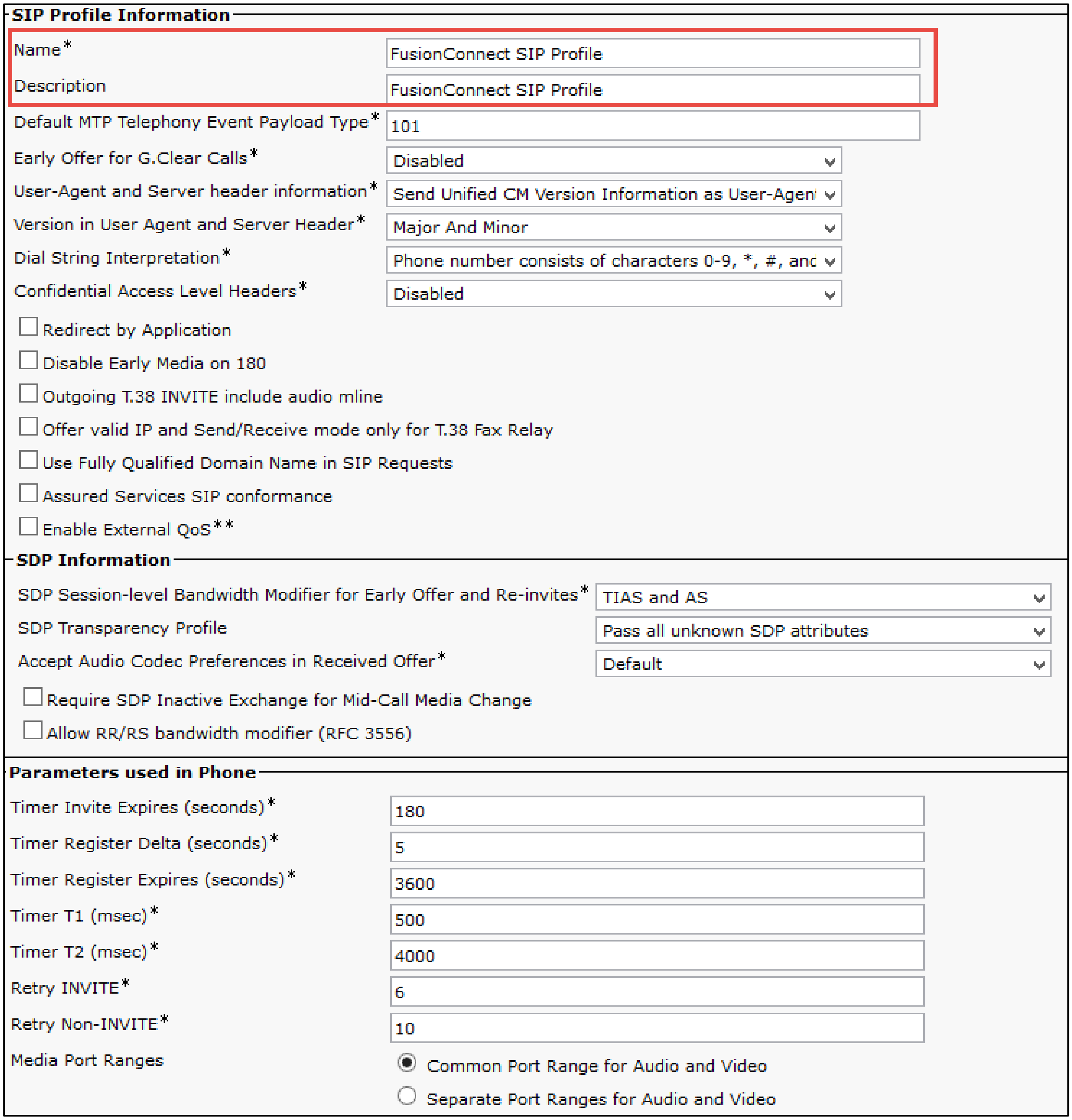


Figure 12: SIP Profile

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		

Incoming Requests FROM URI Settings

Caller ID DN	
Caller Name	

Figure 13: SIP Profile (Cont.)

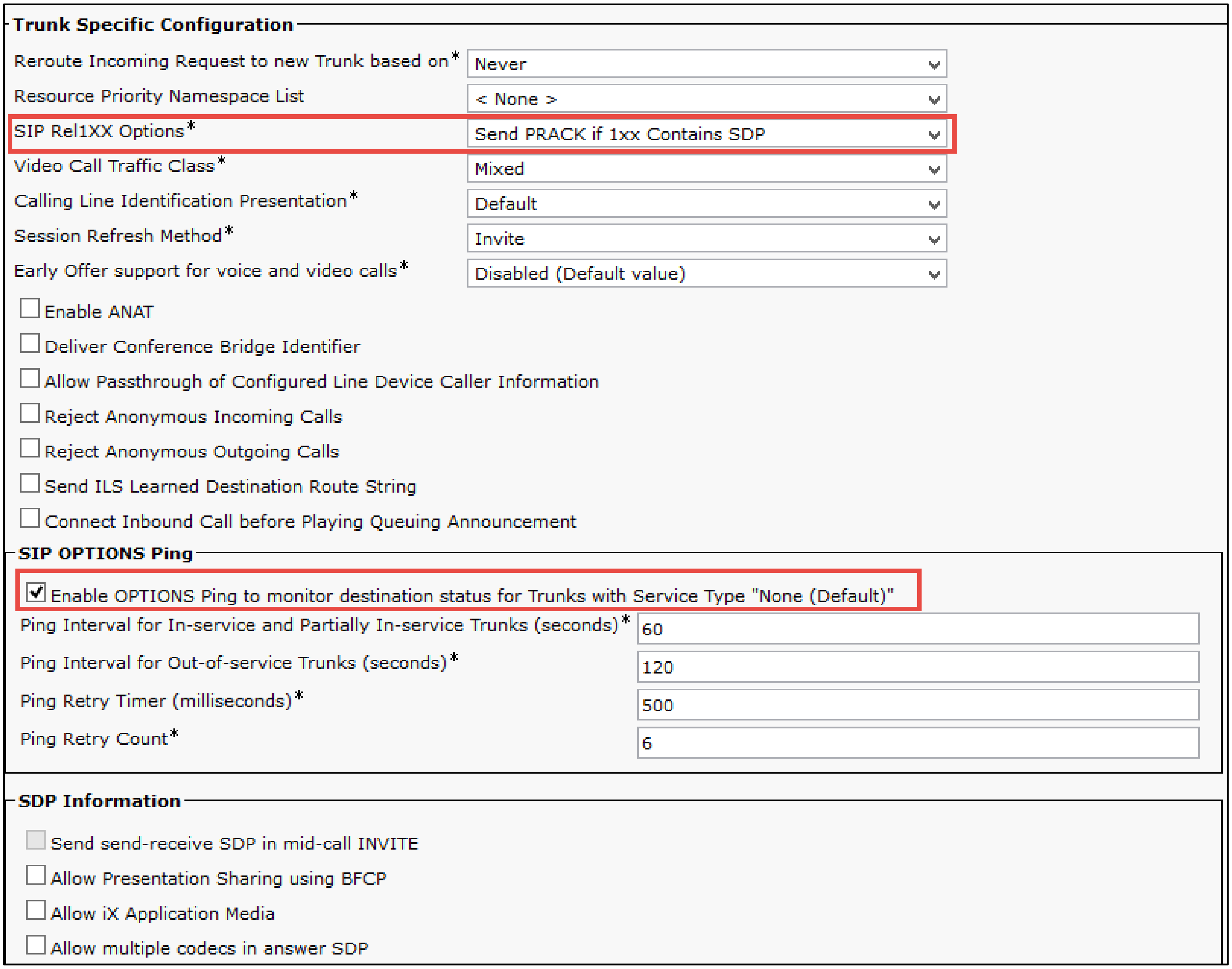


Figure 14: SIP Profile (Cont.)

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: Device → Trunk

Trunks ***(1 - 4 of 4)*** ***Rows per Page*** 50

Find Trunks where Device Name begins with Fusion 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
☐	SIP	FusionConnect	FusionConnect		FusionConnect Devicepool	6.@				SIP Trunk	Full Service	Time In Full Service: 0 day 0 hour 11 minutes	FusionConnect Non Secure SIP Trunk Profile
☐	SIP	FusionConnect	FusionConnect		FusionConnect Devicepool	*67.@				SIP Trunk	Full Service	Time In Full Service: 0 day 0 hour 11 minutes	FusionConnect Non Secure SIP Trunk Profile
☐	SIP	FusionConnect_SIP_trunk_to_Voice_gateway	SIP_trunk_to_Voice_gateway		FusionConnect Devicepool	7593				SIP Trunk	Full Service	Time In Full Service: 0 day 0 hour 0 minute	FusionConnect Non Secure SIP Trunk Profile
☐	SIP	FusionConnect_Trunk_to_UnityConnection	FusionConnect Trunk to UnityConnection		FusionConnect Devicepool	6XXX				SIP Trunk	Full Service	Time In Full Service: 0 day 0 hour 0 minutes	FusionConnect Non Secure SIP Trunk Profile

Figure 15: SIP Trunks List

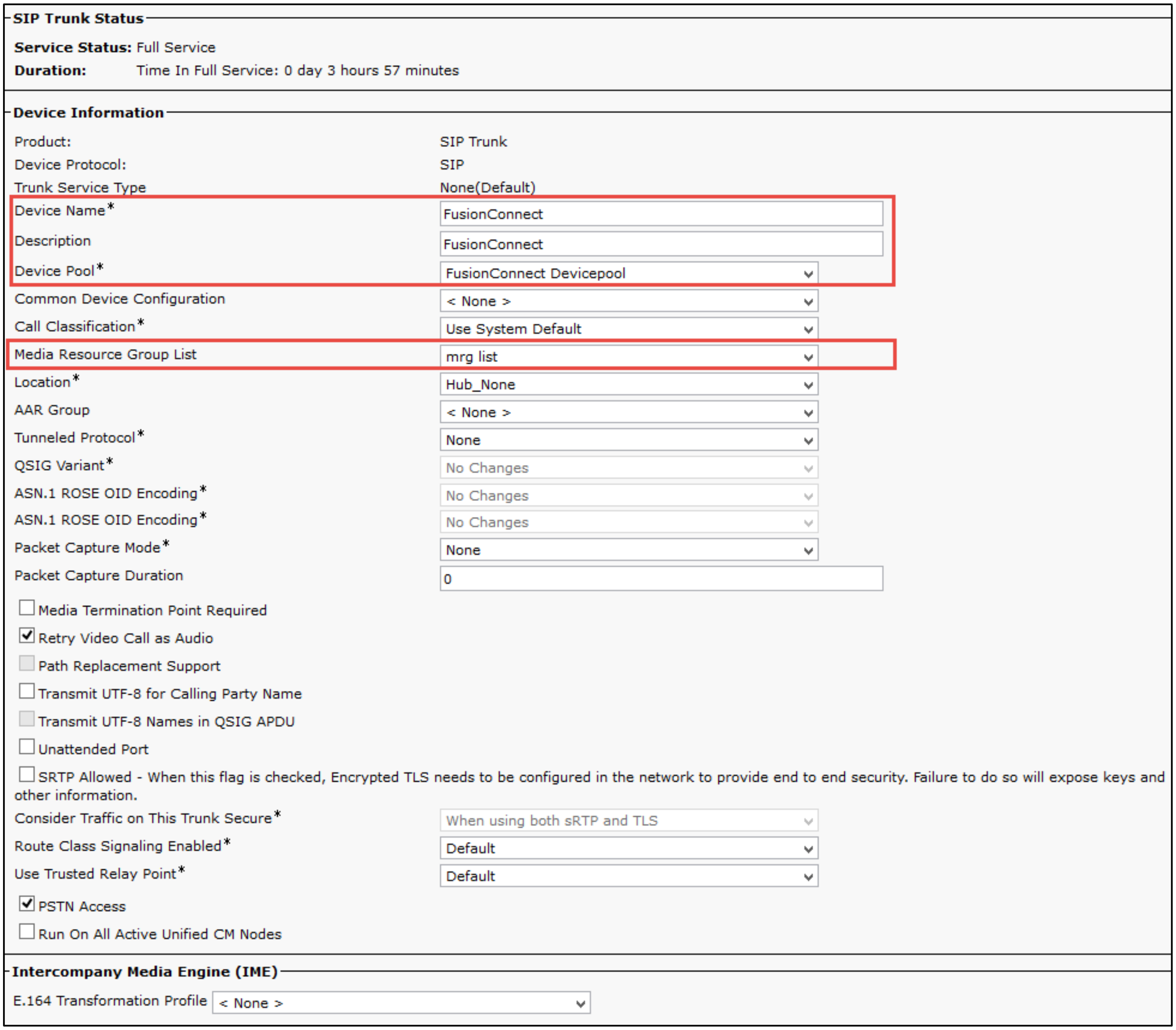
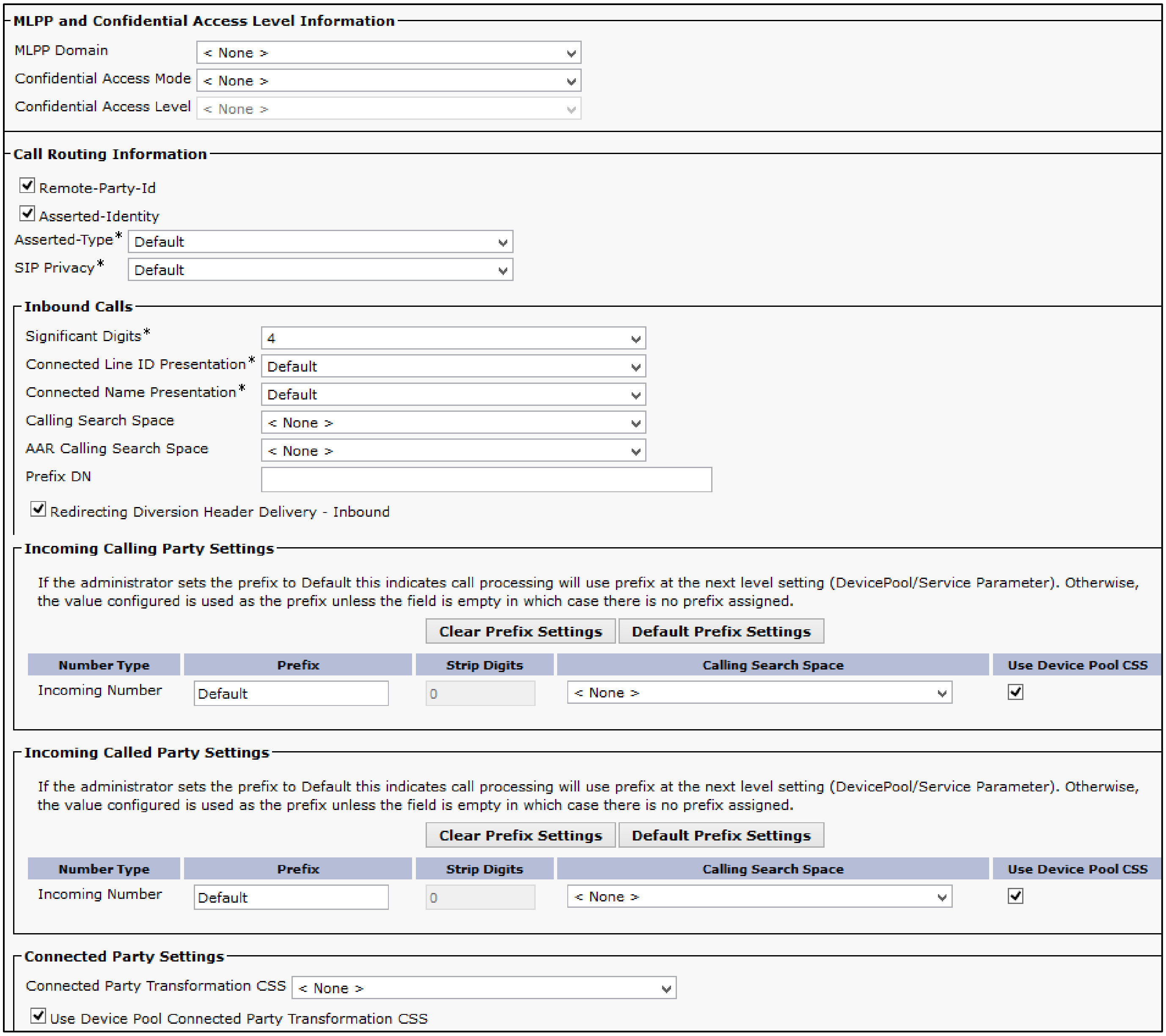


Figure 16: SIP Trunk to Cisco UBE


MLPP and Confidential Access Level Information

MLPP Domain: < None >

Confidential Access Mode: < None >

Confidential Access Level: < None >

Call Routing Information

☑ Remote-Party-Id

☑ Asserted-Identity

Asserted-Type*: Default

SIP Privacy*: Default

Inbound Calls

Significant Digits*: 4

Connected Line ID Presentation*: Default

Connected Name Presentation*: Default

Calling Search Space: < None >

AAR Calling Search Space: < None >

Prefix DN:

☑ Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings | Default Prefix Settings

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

Incoming Called Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings | Default Prefix Settings

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

Connected Party Settings

Connected Party Transformation CSS: < None >

☑ Use Device Pool Connected Party Transformation CSS

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

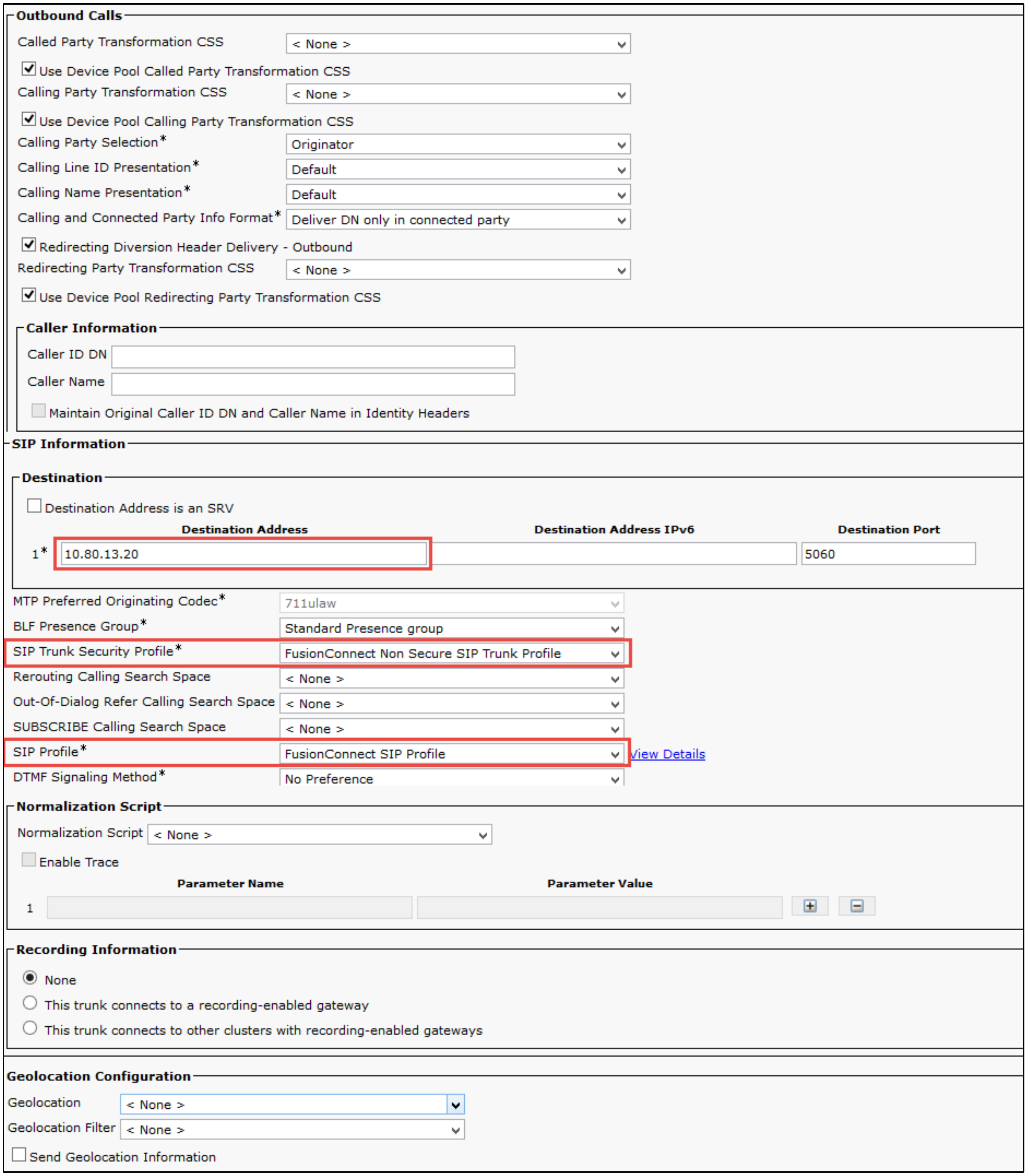


Figure 18: SIP Trunk to Cisco UBE (Cont.)

Explanation

Parameter	Value	Description
Device Name	Fusion Connect	Name for the trunk
Device Pool	Fusion Connect Devicepool	Default Device Pool is used for this trunk
Media Resource Group List	mrg list	MRG with resources: ANN, CFB, MOH and MTP
Significant Digits	4	4 digits Extension for all CPE phones
Destination Address	10.80.13.20	IP address of the Cisco UBE Virtual LAN
SIP Trunk Security Profile	Fusion Connect Non Secure SIP Trunk Profile	SIP Trunk Security Profile configured earlier
SIP Profile	Fusion Connect SIP Profile	SIP Profile configured earlier

Dial Plan

Route Pattern Configuration

Navigation: Call Routing → Route/Hunt → Route Pattern

Route patterns are configured as below:

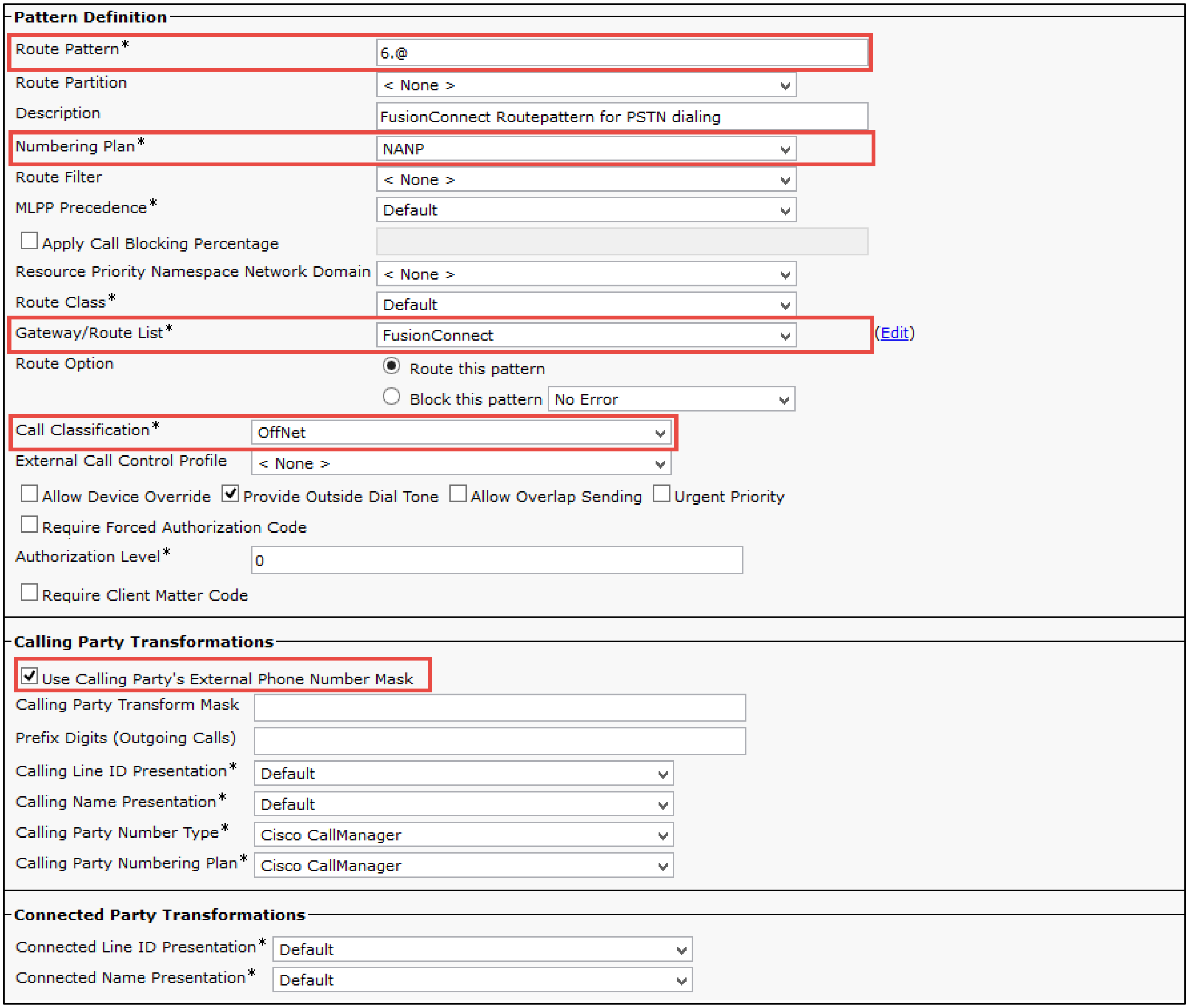
1. Cisco IP phone dial “6”+10 digits number to access PSTN via Cisco UBE
    - “6” is removed before sending to Cisco UBE
2. For FAX call, Access Code “6”+10 digits number is used at Cisco Fax gateway
    - “6” is removed at Cisco UCM
    - The rest of the number is sent to Cisco UBE to Fusion Connect network
3. Incoming fax call to 7XXX will be sent to Cisco Fax gateway
4. For Anonymous call, access code “*67”+10 digits number is used
    - “*67” is removed at Cisco UCM
    - The rest of the number is sent to Cisco UBE to Fusion Connect network

Route Patterns *(1 - 4 of 4)* *Rows per Page* 50

Find Route Patterns where Pattern begins with Find Clear Filter

	Pattern	Description	Partition	Route Filter	Associated Device	Copy
	*67.@	FusionConnect Rout Pattern for Anonymous Call			FusionConnect	
	7593	FusionConnect RoutePattern for FAX			FusionConnect_SIP_trunk_to_Voice_gateway	
	6.@	FusionConnect Routepattern for PSTN dialing			FusionConnect	

Figure 19: Route Patterns List



Pattern Definition

Route Pattern*	6.@
Route Partition	< None >
Description	FusionConnect Routepattern for PSTN dialing
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*	FusionConnect (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

Figure 20: Route Pattern for Voice

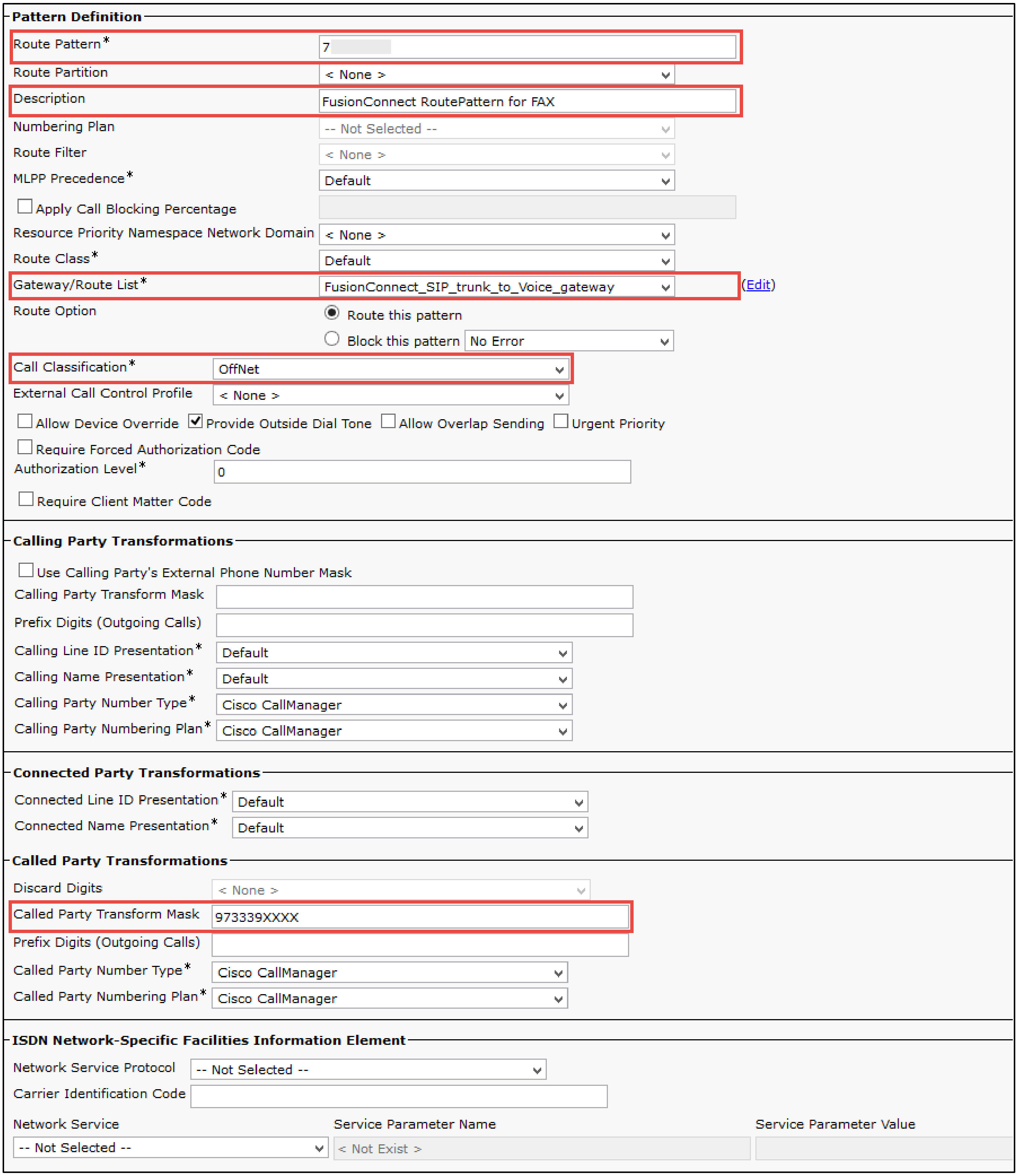


Figure 21: Route Pattern for Voice (Cont.)

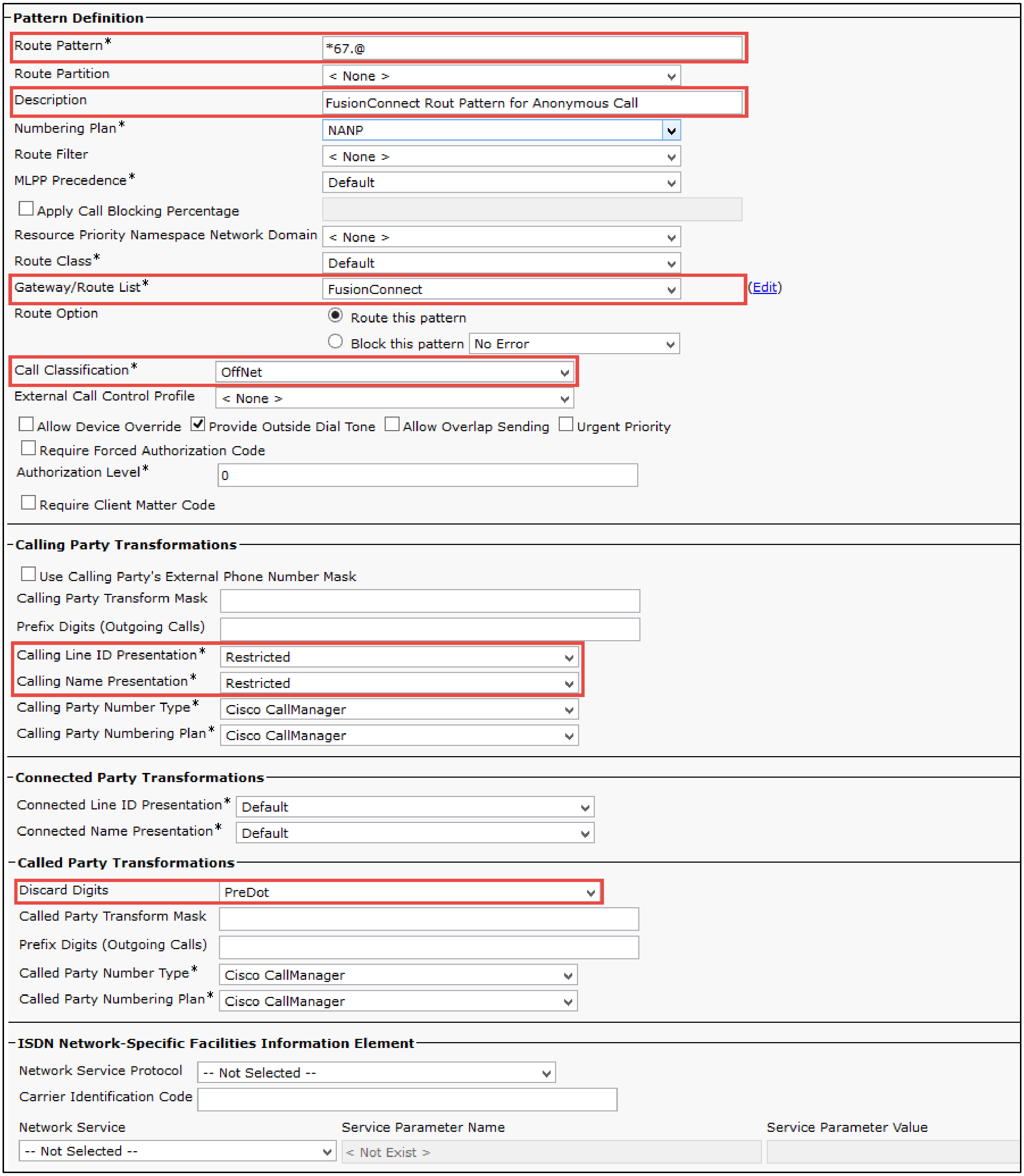


Figure 22: Route Pattern for Fax

Explanation

Setting	Value	Description
Route Pattern	6.@ for Voice & International Calls, 7XXX for Fax Call and *67.@ for Anonymous Call.	Specify appropriate Route Pattern
Gateway/Route List	Fusion Connect for Route Pattern 6.@, *67.@ and SIP_Trunk_To_Voice_Gateway for Route Pattern 7XXX	SIP Trunk name configured earlier
Numbering Plan	NANP for Route Pattern 6.@ and *67.@	North American Numbering Plan
Call Classification	OffNet for Route Pattern 6.@, 7XXX and *67.@	Restrict the transferring of an external call to an external device
Discard Digits	PreDot for Route Pattern 6.@ and *67.@	Specifies how to modify digit before they are sending to Fusion Connect network
Calling Line ID Presentation & Calling Name Presentation	Restricted for Route Pattern *67.@	Restrict the Caller ID Display of Calling party's at external user endpoint.

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

Important Information

THE SPECIFICATIONS AND INFORMATION REGARDING THE PRODUCTS IN THIS MANUAL ARE SUBJECT TO CHANGE WITHOUT NOTICE. ALL STATEMENTS, INFORMATION, AND RECOMMENDATIONS IN THIS MANUAL ARE BELIEVED TO BE ACCURATE BUT ARE PRESENTED WITHOUT WARRANTY OF ANY KIND, EXPRESS OR IMPLIED. USERS MUST TAKE FULL RESPONSIBILITY FOR THEIR APPLICATION OF ANY PRODUCTS. IN NO EVENT SHALL CISCO OR ITS SUPPLIERS BE LIABLE FOR ANY INDIRECT, SPECIAL, CONSEQUENTIAL, OR INCIDENTAL DAMAGES, INCLUDING, WITHOUT LIMITATION, LOST PROFITS OR LOSS OR DAMAGE TO DATA ARISING OUT OF THE USE OR INABILITY TO USE THIS MANUAL, EVEN IF CISCO OR ITS SUPPLIERS HAVE BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGES.

cisco

Test Results

N/A = Not Applicable | N/S = Not Supported | N/T = Not Tested

Note: Cisco Test Plan has been executed with Registration SIP Trunk test set up as per Customer requirement.

ID	External Test Case Type	Description	Test Result	Observations
1.1	IP-PBX outbound to SP Off-net gateway(PSTN) (G.729 is offered first)	Call is established from IP-PBX (A-party) and terminated on PSTN (B-party). Allow for ringback to occur and cancel the call from the IP-PBX.	PASSED	
1.2	IP-PBX outbound to SP Off-net gateway(PSTN) (G.729 is offered first)	Call is established from PBX (A-party) and terminated on PSTN. Allow for ringback during call setup and answer the call. After at least 30 seconds disconnect the call from the PBX (A-party).	PASSED	
1.3	IP-PBX outbound to SP Off-net gateway(PSTN) (G.729 is offered first)	Call is established from IP-PBX (A-party) and terminated on PSTN (B-party). Allow for ringback to occur and cancel the call from the IP-PBX.	PASSED	
1.4	IP-PBX outbound to SP Off-net gateway(PSTN) (G.729 is offered first)	Call is established from IP-PBX and terminated on PSTN. Keep the call open for at least 1 hour to allow for Session Refresh to occur.	PASSED	
1.5	IP-PBX outbound to SP Off-net gateway(PSTN) (G.729 is offered first)	DTMF relay (both directions)	PASSED	
1.6	IP-PBX outbound to SP Off-net gateway(PSTN) (G.729 is offered first)	Call is established from IP-PBX (A-party) and terminated on PSTN (B-party). Allow for ringback to occur and cancel the call from the PSTN.	PASSED	

1.7	IP-PBX outbound to SP Off-net gateway(PSTN) (G.729 is offered first)	Call is established from IP-PBX (A-party) and terminated on PSTN (B-party). Allow for ringback to occur and cancel the call from the IP-PBX.	PASSED	
2.1	SP off-net gateway(PSTN) inbound to IP-PBX (G.729 offered first)	Call is established from PSTN and terminated on PBX. Allow for ringback to occur and cancel the call from the PBX.	PASSED	
2.2	SP off-net gateway(PSTN) inbound to IP-PBX (G.729 offered first)	Call is established from PSTN and terminated on PBX. Allow for ringback during call setup and answer the call. After at least 30 seconds disconnect the call from the PSTN.	PASSED	
2.3	SP off-net gateway(PSTN) inbound to IP-PBX (G.729 offered first)	Call is established from PSTN and terminated on PBX. Allow for ringback to occur and cancel the call from the PSTN.	PASSED	
2.4	SP off-net gateway(PSTN) inbound to IP-PBX (G.729 offered first)	Call is established from PSTN and terminated on PBX. Keep the call open for at least 1 hour to allow for Session Refresh to occur.	PASSED	
2.5	SP off-net gateway(PSTN) inbound to IP-PBX (G.729 offered first)	DTMF relay (both directions)	PASSED	
2.6	SP off-net gateway(PSTN) inbound to IP-PBX (G.729 offered first)	Call is established from PSTN and terminated on PBX. Allow for ringback to occur and cancel the call from the PBX.	PASSED	
2.7	SP off-net gateway(PSTN) inbound to IP-PBX (G.729 offered first)	Call is established from PSTN and terminated on PBX. Allow for ringback to occur and cancel the call from the PSTN.	PASSED	
3.1	IP-PBX to IP-PBX (place call out to the SP network and back) (G.729 is offered first)	Call is established from IP-PBX (A -party) to IP-PBX (B-party). Allow for ringback to occur and cancel the call from the IP-PBX (A -party).	PASSED	

3.2	IP-PBX to IP-PBX (place call out to the SP network and back) (G.729 is offered first)	Call is established from IP-PBX (A -party) to IP-PBX (B-party). Allow for ringback during call setup and answer the call. After at least 30 seconds disconnect the call from the PBX (A-party).	PASSED	
3.3	IP-PBX to IP-PBX (place call out to the SP network and back) (G.729 is offered first)	Call is established from IP-PBX (A -party) to IP-PBX (B-party). Allow for ringback to occur and cancel the call from the IP-PBX.	PASSED	
3.4	IP-PBX to IP-PBX (place call out to the SP network and back) (G.729 is offered first)	Call is established from IP-PBX (A -party) to IP-PBX (B-party). Keep the call open for at least 1 hour to allow for Session Refresh to occur.	PASSED	
3.5	IP-PBX to IP-PBX (place call out to the SP network and back) (G.729 is offered first)	DTMF relay (both directions)	PASSED	
3.6	IP-PBX to IP-PBX (place call out to the SP network and back) (G.729 is offered first)	Call is established from IP-PBX (A -party) to IP-PBX (B-party). Allow for ringback to occur and cancel the call from the IP-PBX (B-party).	PASSED	
3.7	IP-PBX to IP-PBX (place call out to the SP network and back) (G.729 is offered first)	Call is established from IP-PBX (A -party) to IP-PBX (B-party). Allow for ringback to occur and cancel the call from the IP-PBX (A -party).	PASSED	
4.1	IP-PBX Calling number privacy	Call is established from the IP-PBX and terminated on PSTN. IP-PBX sends out an INVITE containing header Privacy set to ID	N/S	Fusion Connect does not support “Privacy=ID”
4.2	IP-PBX Calling number privacy	Call is established from the IP-PBX (A-party) to IP-PBX (B-party). IP PBX sends out an INVITE containing header Privacy set to ID	N/A	Testing has been executed with only one IP-PBX
5.1	IP-PBX Telephone Number Support	Call is established from the IP-PBX and terminated on PSTN. Check the IP-PBX translating IP-PBX extension to 10 DID calling number.	PASSED	

5.2	IP-PBX Telephone Number Support	Call is established from the PSTN and terminated on IP-PBX. Check the IP-PBX translating 10 DID called number to private extension.	PASSED	
5.3	IP-PBX Telephone Number Support	Call is established from the IP-PBX to IP-PBX. Check the IP-PBX translating private extension to 10 DID calling number.	PASSED	
5.4	IP-PBX Telephone Number Support	Call is established from the IP-PBX to IP-PBX. Check the IP-PBX translating private extension to 10 DID calling number.	PASSED	
5.5	IP-PBX Telephone Number Support	IP-PBX Telephone Number Support	PASSED	
5.6	IP-PBX Telephone Number Support	IP-PBX Telephone Number Support	PASSED	
6.1	IP-PBX Calling Name Delivery	Call is established from IP-PBX to IP-PBX and check the call setup, teardown and correct CLI presentation	PASSED	
7.1	IP-PBX off-net Call Conference	Call is established from the PSTN 1 and terminated on PBX user. PBX user 1 conferences PSTN2 user.	PASSED	
7.2	IP-PBX off-net Call Conference	Call is established from PBX user and terminated on PSTN user 1. PBX user1 conferences PSTN 2 user	PASSED	
7.3	IP-PBX off-net Call Conference	Call is established from PBX user 1 to PBX user 2. PBX user1 conferences PSTN 2 user. Check the RTP path and release the call from PSTN 1.	PASSED	
8.1	IP-PBX Intra-Site Call Conference	IP-PBX Intra-Site Call Conference	N/A	Testing has been executed with only one IP-PBX
8.2	IP-PBX Intra-Site Call Conference	IP-PBX Intra-Site Call Conference	N/A	Testing has been executed with only one IP-PBX
8.3	IP-PBX Intra-Site Call Conference	Call is established from PBX user1 and terminated on PSTN user 1. PBX user1 conferences PBX user2 user. Check the RTP path and release the call from PSTN 1.	PASSED	

8.4	IP-PBX Intra-Site Call Conference		N/A	Testing has been executed with only one IP-PBX
8.5	IP-PBX Intra-Site Call Conference	Call is established from PSTN user and terminated on PBX user 1. PBX user1 conferences PBX user 2 user. Check the RTP path and release the call from PSTN 1.	PASSED	
9.1	CPE Intra-Site Attended Call Transfer	Call is established from PSTN user 1 and terminated on PBX user. PBX user transfer the call to PSTN user 2. Verify the CLID.	PASSED	Caller ID is not updated after Attended /Un-attended transfers to off-net phones. This is due a limitation on Cisco UBE and will be resolved in the next release.
9.2	CPE Intra-Site Attended Call Transfer	Call is established from PBX user 1 and terminated on PSTN user 1. PBX user transfer the call to PSTN user 2. Verify the CLID.	PASSED	Caller ID is not updated after Attended /Un-attended transfers to off-net phones. This is due a limitation on Cisco UBE and will be resolved in the next release.
9.3	CPE Intra-Site Attended Call Transfer		N/A	Testing has been executed with only one IP-PBX
9.4	CPE Intra-Site Attended Call Transfer	Call is established from PBX user 1 to PBX user 2. PBX user 1 transfer the call to PSTN user. Verify the CLID.	PASSED	Caller ID updated on Phone 2 (B-Party) with PSTN number (C-Party).
9.5	CPE Intra-Site Attended Call Transfer		N/A	Testing has been executed with only one IP-PBX
9.6	CPE Intra-Site Attended Call Transfer	Call is established from PBX user 1 to PSTN user. PBX user 1 transfer the call to PBX user 2. Verify the CLID.	PASSED	Caller ID updated on Phone 2 (C-Party) with PSTN number (B-Party).
9.7	CPE Intra-Site Attended Call Transfer		N/A	Testing has been executed with only one IP-PBX

9.8	CPE Intra-Site Attended Call Transfer	Call is established from PSTN user 1 and terminated on PBX user 1. PBX user1 transfer the call to PBX user 2. Verify the CLID.	PASSED	Caller ID updated on Phone 3 (C-Party) with PSTN Number (B-Party)
10.1	CPE Intra-Site Unattended Call Transfer	Call is established from PSTN user 1 and terminated on PBX user. PBX user transfer the call to PSTN user 2.Verify the CLID.	PASSED	Caller ID is not updated after Attended /Un-attended transfers to off-net phones. This is due a limitation on Cisco UBE and will be resolved in the next release.
10.2	CPE Intra-Site Unattended Call Transfer	Call is established from PBX user 1 and terminated on PSTN user 1. PBX user transfer the call to PSTN user 2. Verify the CLID.	PASSED	Caller ID is not updated after Attended /Un-attended transfers to off-net phones. This is due a limitation on Cisco UBE and will be resolved in the next release.
10.3	CPE Intra-Site Unattended Call Transfer	CPE Intra-Site Unattended Call Transfer	N/A	Testing has been executed with only one IP-PBX
10.4	CPE Intra-Site Unattended Call Transfer	Call is established from PBX user1 to PBX user 2. PBX user 1 transfer the call to PSTN user. Verify the CLID.	PASSED	Caller ID is not updated after Attended /Un-attended transfers to off-net phones. Due a limitation on Cisco UBE and will be resolved in the next release.
10.5	CPE Intra-Site Unattended Call Transfer	CPE Intra-Site Unattended Call Transfer	N/A	Testing has been executed with only one IP-PBX
10.6	CPE Intra-Site Unattended Call Transfer	Call is established from PBX user 1 to PSTN user. PBX user 1 transfer the call to PBX user 2. Verify the CLID.	PASSED	Caller ID is not updated after Attended /Un-attended transfers to PBX User-2. This is due a limitation on Cisco UBE and will be resolved in the next release.
10.7	CPE Intra-Site Unattended Call Transfer		N/A	Testing has been executed with only one IP-PBX

10.8	CPE Intra-Site Unattended Call Transfer	Call is established from PSTN user 1 and terminated on PBX user 1. PBX user 1 transfer the call to PBX user 2. Verify the CLID.	PASSED	Caller ID is not updated after Attended /Un-attended transfers to PBX User-2. Due a limitation on Cisco UBE and will be resolved in the next release.
11.1	CPE Intra-Site Blind Call Transfer	Call is established from PSTN user 1 and terminated on PBX user	N/S	Cisco does not support Blind Call Transfer
11.2	CPE Intra-Site Blind Call Transfer	Call is established from PBX user 1 and terminated on PSTN user1	N/S	Cisco does not support Blind Call Transfer
11.3	CPE Intra-Site Blind Call Transfer	CPE Intra-Site Blind Call Transfer	N/S	Cisco does not support Blind Call Transfer
11.4	CPE Intra-Site Blind Call Transfer	Call is established from PBX user 1 to PBX user	N/S	Cisco does not support Blind Call Transfer
11.5	CPE Intra-Site Blind Call Transfer	CPE Intra-Site Blind Call Transfer	N/S	Cisco does not support Blind Call Transfer
11.6	CPE Intra-Site Blind Call Transfer	Call is established from PBX user 1 to PSTN user	N/S	Cisco does not support Blind Call Transfer
11.7	CPE Intra-Site Blind Call Transfer	CPE Intra-Site Blind Call Transfer	N/S	Cisco does not support Blind Call Transfer
11.8	CPE Intra-Site Blind Call Transfer	Call is established from PSTN user 1 and terminated on PBX user	N/S	Cisco does not support Blind Call Transfer
12.1	CPE Call Hold and Resume	Make call from CPE to PSTN and CPE puts call on hold, verify MOH	PASSED	
12.2	CPE Call Hold and Resume	Make call from PBX user 1 to PBX user 2, verify MOH	PASSED	
12.3	CPE Call Hold and Resume	Make call from PSTN to CPE. CPE puts call on hold, verify MOH	PASSED	
12.4	CPE Call Hold and Resume	Disable MOH and make call from CPE to PSTN. CPE puts call on hold, verify TOH	PASSED	

12.5	CPE Call Hold and Resume	Disabled MOH and make call from PBX user 1 to PBX user 2 and verify TOH.	PASSED	
12.6	CPE Call Hold and Resume	Disabled MOH and make call from PSTN to CPE. Verify TOH.	PASSED	
13.1	CPE Voicemail (e.g. using Unity or Unity Connection)	Call is established from PSTN (A-party) and terminated on IP-PBX (B-party). Call should forward to Voicemail after 4 rings. Leave message in mailbox and verify that BYE is received	PASSED	
13.2	CPE Voice Mail (e.g. using Unity or Unity Connection)	Call from PSTN to Voicemail hunt group number, enter the voicemail account and retrieve the messages.	PASSED	
14.1	SP Voice Mail (e.g. using mobile phone)	Call is established from IP-PBX and terminated on Mobile. Call should forward to Voicemail after 4 rings. Leave message in mailbox	PASSED	
14.2	SP Voice Mail (e.g. using mobile phone)	CPE to PSTN (mobile VM): retrieve voicemail	PASSED	
15.1	CPE Find Me (Call Forward Unconditionally)	Call is established from PSTN user 1 and terminated on PBX user	PASSED	
15.2	CPE Find Me (Call Forward Unconditionally)	Make call from PBX user 1 to PBX user 2. PBX user 2 configured unconditional transfer to PBX user3	PASSED	
15.3	CPE Find Me (Call Forward Unconditionally)	Call is established from PSTN user 1 and terminated on PBX user	PASSED	
15.4	CPE Find Me (Call Forward Unconditionally)	Make call from PBX user 1 to PBX user 2. PBX user 2 configured unconditional transfer to PSTN user	PASSED	
16.1	CPE T.38 FAX G3 (G.729 is offered first)	Place FAX call from PBX-FAX machine to Remote PSTN FAX	PASSED	
16.2	CPE T.38 FAX G3 (G.729 is offered first)	Place FAX call from PBX-FAX machine to Remote PSTN FAX	PASSED	

16.3	CPE T.38 FAX G3 (G.729 is offered first)	Place FAX call from PSTN-FAX machine to PBX-FAX machine	PASSED	
16.4	CPE T.38 FAX G3 (G.729 is offered first)	Place FAX call from PSTN-FAX machine to PBX-FAX machine	PASSED	
16.5	CPE T.38 FAX G3 (G.729 is offered first)	Place FAX call from PBX-FAX1 machine to PBX-FAX2 machine	N/A	Testing has been executed with only one PBX
16.6	CPE T.38 FAX G3 (G.729 is offered first)	Place FAX call from PBX-FAX1 machine to PBX-FAX2 machine	N/A	Testing has been executed with only one PBX
17.1	CPE T.38 FAX SG3 (G.729 is offered first)	Place FAX call from PBX-FAX machine to Remote PSTN FAX	PASSED	Test case has been executed in G3 since Fusion Connect does not support SG3
17.2	CPE T.38 FAX SG3 (G.729 is offered first)	Place FAX call from PBX-FAX machine to Remote PSTN FAX	PASSED	Test case has been executed in G3 since Fusion Connect does not support SG3
17.3	CPE T.38 FAX SG3 (G.729 is offered first)	Place FAX call from PSTN-FAX machine to PBX-FAX machine	PASSED	Test case has been executed in G3 since Fusion Connect does not support SG3
17.4	CPE T.38 FAX SG3 (G.729 is offered first)	Place FAX call from PSTN-FAX machine to PBX-FAX machine	PASSED	Test case has been executed in G3 since Fusion Connect does not support SG3
17.5	CPE T.38 FAX SG3 (G.729 is offered first)	Place FAX call from PBX-FAX1 machine to PBX-FAX2 machine	N/A	Testing has been executed with only one IP-PBX
17.6	CPE T.38 FAX SG3 (G.729 is offered first)	Place FAX call from PBX-FAX1 machine to PBX-FAX2 machine	N/A	Testing has been executed with only one IP-PBX
18.1	Simultaneous Calls	Make simultaneous call from PBX users to PSTN	PASSED	
18.2	Simultaneous Calls	Make simultaneous call from PSTN to PBX	PASSED	
18.3	Simultaneous Calls	Make simultaneous call from PBX to PBX	PASSED	
19.1	CPE Auto Attendant	Make call from PSTN to auto attendant and verify the voice prompt	PASSED	

19.2	CPE Auto Attendant	Make call from PSTN to auto attendant and verify the voice prompt	PASSED	
20.1	CPE to PSTN Off-net gateway international call	Call is established from IP-PBX (A-party) and terminated on PSTN (B-party)	PASSED	
20.2	CPE to PSTN Off-net gateway international call	Call is established from PBX (A-party) and terminated on PSTN	PASSED	
20.3	CPE to PSTN Off-net gateway international call	Call is established from IP-PBX (A-party) and terminated on PSTN (B-party)	PASSED	
20.4	CPE to PSTN Off-net gateway international call	DTMF relay (both directions) (RFC2833)	PASSED	
20.5	CPE to PSTN Off-net gateway international call	Call is established from IP-PBX (A-party) and terminated on PSTN (B-party)	PASSED	
20.6	CPE to PSTN Off-net gateway international call	Call is established from IP-PBX (A-party) and terminated on PSTN (B-party)	PASSED	
21.1	CPE G.711 FAX G3	Place FAX call from PBX-FAX machine to Remote PSTN FAX	PASSED	
21.2	CPE G.711 FAX G3	Place FAX call from PBX-FAX machine to Remote PSTN FAX	PASSED	
21.3	CPE G.711 FAX G3	Place FAX call from PSTN-FAX machine to PBX-FAX machine	PASSED	
21.4	CPE G.711 FAX G3	Place FAX call from PSTN-FAX machine to PBX-FAX machine	PASSED	
21.5	CPE G.711 FAX G3	Place FAX call from PBX-FAX1 machine to PBX-FAX2 machine	N/A	Testing has been executed with only one IP-PBX
21.6	CPE G.711 FAX G3	Place FAX call from PBX-FAX1 machine to PBX-FAX2 machine	N/A	Testing has been executed with only one IP-PBX
22.1	CPE G.711 FAX SG3	Place FAX call from PBX-FAX machine to Remote PSTN FAX	PASSED	

22.2	CPE G.711 FAX SG3	Place FAX call from PBX-FAX machine to Remote PSTN FAX	PASSED	
22.3	CPE G.711 FAX SG3	Place FAX call from PSTN-FAX machine to PBX-FAX machine	PASSED	
22.4	CPE G.711 FAX SG3	Place FAX call from PSTN-FAX machine to PBX-FAX machine	PASSED	
22.5	CPE G.711 FAX SG3	Place FAX call from PBX-FAX1 machine to PBX-FAX2 machine	N/A	Testing has been executed with only one IP-PBX
22.6	CPE G.711 FAX SG3	Place FAX call from PBX-FAX1 machine to PBX-FAX2 machine	N/A	Testing has been executed with only one IP-PBX
23.1	CPE Find Me (Call Forward On Busy)	Call is established from PSTN and terminated on PBX user 1	PASSED	
23.2	CPE Find Me (Call Forward On Busy)	Call is established from PSTN and terminated on PBX user 1	PASSED	
23.3	CPE Find Me (Call Forward On Busy)	Make call from PBX user 1 to PBX user 2	PASSED	
23.4	CPE Find Me (Call Forward On Busy)	Make call from PBX user 1 to PBX user 2	PASSED	
24.1	CPE Find Me (Call Forward Don't Answer)	Call is established from PSTN and terminated on PBX user 1	PASSED	
24.2	CPE Find Me (Call Forward Don't Answer)	Call is established from PSTN and terminated on PBX user 1	PASSED	
24.3	CPE Find Me (Call Forward Don't Answer)	Make call from PBX user 1 to PBX user 2	PASSED	
24.4	CPE Find Me (Call Forward Don't Answer)	Make call from PBX user 1 to PBX user 2	PASSED	
25.1	Codec mid-call re-negotiation (to be tested without transcoder)	Off-net calls IP PBX phone 1 (G729), phone 1 transfers to UM/ gateway (g711u)	PASSED	
25.2	Codec mid-call re-negotiation (to be tested without transcoder)	IP PBX phone 1 calls Off-net phone (call is G711), off-net phone transfers call to IP PBX phone 2 (G729 region)	PASSED	

26.1	Dial Plans	Make call from PBX based on dial plan	PASSED	0 & 0+10 digits – Fusion Connect does not support Operator Assistance dialing
27.1	PRACK with SDP (early-media cut-through with DTMF (RFC2833) navigation before 200OK)	Make call from IP PBX to call 800-864-8331 - United Airlines	N/S	Fusion Connect does not support "require100rel"
28.1	CUBE HA	Unplug the LAN side cable of Primary CUBE, verify both incoming and outgoing calls work through secondary CUBE	PASSED	
28.2	CUBE HA	Plug the LAN cable back for Primary CUBE, verify the incoming/ outgoing call going through the Primary CUBE and show status of both CUBE	PASSED	
28.3	CUBE HA	Unplug the WAN side cable of Primary CUBE, verify both incoming and outgoing calls work through secondary CUBE	PASSED	
28.4	CUBE HA	Plug the WAN cable back for Primary CUBE, verify the incoming/ outgoing call going through the Primary CUBE and show status of both CUBE	PASSED	
28.5	CUBE HA	Shutdown Primary CUBE and verify the traffic go through viz=a secondary CUBE	PASSED	
28.6	CUBE HA	Bring up the Primary CUBE and verify the traffic take over by the Primary CUBE	N/S	Once the Primary CUBE comes back it will not take over its role as Primary (i.e. Active) and it will be in standby role as per Cisco's statement.

Corporate Headquarters
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-4000
800 553-NETS (6387)
Fax: 408 526-4100

European Headquarters
Cisco Systems International BV
Haarlerbergpark
Haarlerbergweg 13-19
1101 CH Amsterdam
The Netherlands
www-europe.cisco.com
Tel: 31 0 20 357 1000
Fax: 31 0 20 357 1100

Americas Headquarters
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-7660
Fax: 408 527-0883

Asia Pacific Headquarters
Cisco Systems, Inc.
Capital Tower
168 Robinson Road
#22-01 to #29-01
Singapore 068912
www.cisco.com
Tel: +65 317 7777
Fax: +65 317 7799

Cisco Systems has more than 200 offices in the following countries and regions. Addresses, phone numbers, and fax numbers are listed on the Cisco Web site at www.cisco.com/go/offices.

Argentina • Australia • Austria • Belgium • Brazil • Bulgaria • Canada • Chile • China PRC • Colombia • Costa Rica • Croatia • Czech Republic • Denmark • Dubai, UAE • Finland • France • Germany • Greece • Hong Kong SAR • Hungary • India • Indonesia • Ireland • Israel • Italy • Japan • Korea • Luxembourg • Malaysia • Mexico • The Netherlands • New Zealand • Norway • Peru • Philippines • Poland • Portugal • Puerto Rico • Romania • Russia • Saudi Arabia • Scotland • Singapore • Slovakia • Slovenia • South Africa • Spain • Sweden • Switzerland • Taiwan • Thailand • Turkey Ukraine • United Kingdom • United States • Venezuela • Vietnam • Zimbabwe