Fusion Connect SIP Trunking:

Cisco Unified Communications Manager 11.5.1 with Cisco Unified Border Element (CUBE 11.5.2) on ISR4321 [IOS-XE 16.3.1] using SIP

October 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  
    Call flow .......................................................................................................................................... 31  
  Configuring Cisco Unified Communications Manager .................................................................... 33  
    Cisco UCM Version .......................................................................................................................... 33  
    Cisco Call Manager Service Parameters ....................................................................................... 33  
    Offnet Calls via Fusion Connect SIP Trunk ..................................................................................... 35  
    Dial Plan ......................................................................................................................................... 43  
Acronyms ............................................................................................................................................. 49  
Important Information ............................................................................................................................ 50
# Table of Figures

| Figure 1: Network Topology                                                                 | 5 |
| Figure 2: Cisco UBE High Availability                                                     | 5 |
| Figure 3: Outbound Voice Call                                                             | 31 |
| Figure 4: Outbound Fax Call                                                              | 31 |
| Figure 5: Inbound Voice Call                                                             | 32 |
| Figure 6: Inbound Fax Call                                                               | 32 |
| Figure 7: PBX to PBX via Fusion Connect Call                                             | 32 |
| Figure 8: Cisco UCM Version                                                              | 33 |
| Figure 9: Service Parameters                                                             | 33 |
| Figure 10: Service Parameters (Cont.)                                                     | 34 |
| Figure 11: SIP Trunk Security Profile                                                     | 35 |
| Figure 12: SIP Profile                                                                   | 36 |
| Figure 13: SIP Profile (Cont.)                                                           | 37 |
| Figure 14: SIP Profile (Cont.)                                                           | 38 |
| Figure 15: SIP Trunks List                                                               | 39 |
| Figure 16: SIP Trunk to Cisco UBE                                                         | 40 |
| Figure 17: SIP Trunk to Cisco UBE (Cont.)                                                | 41 |
| Figure 18: SIP Trunk to Cisco UBE (Cont.)                                                | 42 |
| Figure 19: Route Patterns List                                                           | 44 |
| Figure 20: Route Pattern for Voice                                                       | 45 |
| Figure 21: Route Pattern for Voice (Cont.)                                               | 46 |
| Figure 22: Route Pattern for Fax                                                         | 47 |
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 (CUBE 11.5.2) ISR 4321/K9 running IOS-XE 16.3.1 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 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-XE 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:

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-XE 16.3.1 for ISR 4321/K9 Cisco Unified Border Element [CUBE 11.5.2]
- 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

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>allow-connections sip to sip</td>
<td>Allow IP2IP connections between two SIP call legs</td>
</tr>
<tr>
<td>asserted-id</td>
<td>Specifies the type of privacy header in the outgoing SIP requests and response messages</td>
</tr>
<tr>
<td>early-offer forced</td>
<td>Enables SIP Delayed-Offer to Early-Offer globally</td>
</tr>
<tr>
<td>midcall-signaling passthru</td>
<td>Passes SIP messages from one IP leg to another IP leg</td>
</tr>
</tbody>
</table>
**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
!
dlial-peer voice 530 voip
description Incoming Call to PBX from PSTN-LAN facing
huntstop
destination-pattern X...X...X....
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**

fusionConnect1#sh run

```plaintext
! 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 b2bhfusionConnect
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-nre
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-nre
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

fusionConnect2#sh run

!
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
!
vrfo 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:XXXXXXXXXXX@1XX.XX.XX.XX>"
request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:XXXXXXX\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-n-te
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-n-te
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-nge
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-nge
no vad
!
sip-ua
credentials number XXXXXXXXXXX username XXXXXXXXXXX password 7
keepalive target ipv4:2XX.XX.XX.XX
authentication username XXXXXXXXXXX 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
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
Configuring Cisco Unified Communications Manager

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)**

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

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

![SIP Trunk Security Profile Information](image)

Figure 11: SIP Trunk Security Profile

Explanation

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Incoming Transport Type</td>
<td>TCP + UDP</td>
<td>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.</td>
</tr>
<tr>
<td>Outgoing Transport Type</td>
<td>UDP</td>
<td></td>
</tr>
</tbody>
</table>
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

<table>
<thead>
<tr>
<th>SIP Profile Information</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Name</strong></td>
</tr>
<tr>
<td>FusionConnect SIP Profile</td>
</tr>
<tr>
<td><strong>Description</strong></td>
</tr>
<tr>
<td>FusionConnect SIP Profile</td>
</tr>
<tr>
<td>Default RTP Payload Type</td>
</tr>
<tr>
<td>101</td>
</tr>
<tr>
<td>Early Offer for G.729 Calls</td>
</tr>
<tr>
<td>Disabled</td>
</tr>
<tr>
<td>User-Agent and Server header information</td>
</tr>
<tr>
<td>Send Unified CM Version Information as User-Agent</td>
</tr>
<tr>
<td>Version in User Agent and Server Header</td>
</tr>
<tr>
<td>Major And Minor</td>
</tr>
<tr>
<td>Dial String Interpretation</td>
</tr>
<tr>
<td>Phone number consists of characters 0-9, *, #, etc.</td>
</tr>
<tr>
<td>Confidential Access Level Headers</td>
</tr>
<tr>
<td>Disabled</td>
</tr>
</tbody>
</table>

**SDP Information**

| SDP Session-level Bandwidth Modifier for Early Offer and Re-invites |
| TIAS and AS |
| SDP Transparency Profile |
| Pass all unknown SDP attributes |

**Parameters used in Phone**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Timer Invite Expires (seconds)</td>
<td>180</td>
</tr>
<tr>
<td>Timer Register Delta (seconds)</td>
<td>5</td>
</tr>
<tr>
<td>Timer Register Expires (seconds)</td>
<td>3600</td>
</tr>
<tr>
<td>Timer T1 (msec)</td>
<td>500</td>
</tr>
<tr>
<td>Timer T2 (msec)</td>
<td>4000</td>
</tr>
<tr>
<td>Retry INVITE</td>
<td>6</td>
</tr>
<tr>
<td>Retry Non-INVITE</td>
<td>10</td>
</tr>
</tbody>
</table>

- **Media Port Ranges**
  - Common Port Range for Audio and Video
  - Separate Port Ranges for Audio and Video

*Figure 12: SIP Profile*
<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Start Media Port</td>
<td>16384</td>
</tr>
<tr>
<td>Stop Media Port</td>
<td>32766</td>
</tr>
<tr>
<td>DSCP for Audio Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for Video Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for Audio Portion of Video Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for TelePresence Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for Audio Portion of TelePresence Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Call Pickup URI</td>
<td>x-cisco-serviceuri-pickup</td>
</tr>
<tr>
<td>Call Pickup Group Other URI</td>
<td>x-cisco-serviceuri-opickup</td>
</tr>
<tr>
<td>Call Pickup Group URI</td>
<td>x-cisco-serviceuri-gpickup</td>
</tr>
<tr>
<td>Meet Me Service URI</td>
<td>x-cisco-serviceuri-meetme</td>
</tr>
<tr>
<td>User Info</td>
<td>None</td>
</tr>
<tr>
<td>DTMF DB Level</td>
<td>Nominal</td>
</tr>
<tr>
<td>Call Hold Ring Back</td>
<td>Off</td>
</tr>
<tr>
<td>Anonymous Call Block</td>
<td>Off</td>
</tr>
<tr>
<td>Caller ID Blocking</td>
<td>Off</td>
</tr>
<tr>
<td>Do Not Disturb Control</td>
<td>User</td>
</tr>
<tr>
<td>Telnet Level for 7940 and 7960</td>
<td>Disabled</td>
</tr>
<tr>
<td>Resource Priority Namespace</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Timer Keep Alive Expires (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subcribe Expires (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subcribe Delta (seconds)</td>
<td>5</td>
</tr>
<tr>
<td>Maximum Redirects</td>
<td>70</td>
</tr>
<tr>
<td>Off Hook To First Digit Timer (milliseconds)</td>
<td>15000</td>
</tr>
<tr>
<td>Call Forward URI</td>
<td>x-cisco-serviceuri-cfnedl</td>
</tr>
<tr>
<td>Speed Dial (Abbreviated Dial) URI</td>
<td>x-cisco-serviceuri-abrdial</td>
</tr>
</tbody>
</table>

**Normalization Script**

- **Normalization Script**: < None >
- **Enable Trace**: False

**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

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

Create SIP trunks to Cisco UBE

**Navigation:** Device → Trunk

<table>
<thead>
<tr>
<th>Name</th>
<th>Description</th>
<th>Device Pool</th>
<th>Route Pattern</th>
<th>Partition</th>
<th>Route Group</th>
<th>Priority</th>
<th>Trunk Type</th>
<th>SIP Trunk Status</th>
<th>SIP Trunk Duration</th>
<th>SIP Trunk Security Profile</th>
</tr>
</thead>
<tbody>
<tr>
<td>FusionConnect</td>
<td>FusionConnect</td>
<td>FusionConnect Defaultpool</td>
<td>6.0</td>
<td>SIP Trunk</td>
<td>Full Service</td>
<td>Time In Full Service: 0 day 0 hour 0 minutes</td>
<td>Full</td>
<td>FusionConnect Non Secure</td>
<td>SIP Trunk Profile</td>
<td></td>
</tr>
<tr>
<td>FusionConnect</td>
<td>FusionConnect</td>
<td>FusionConnect Defaultpool</td>
<td>67.0</td>
<td>SIP Trunk</td>
<td>Full Service</td>
<td>Time In Full Service: 0 day 0 hour 11 minutes</td>
<td>Full</td>
<td>FusionConnect Non Secure</td>
<td>SIP Trunk Profile</td>
<td></td>
</tr>
<tr>
<td>FusionConnect SIP trunk to Voice gateway</td>
<td>SIP trunk to Voice gateway</td>
<td>FusionConnect Defaultpool</td>
<td>7523</td>
<td>SIP Trunk</td>
<td>Full Service</td>
<td>Time In Full Service: 0 day 0 hour 0 minute</td>
<td>Full</td>
<td>FusionConnect Non Secure</td>
<td>SIP Trunk Profile</td>
<td></td>
</tr>
<tr>
<td>FusionConnect SIP trunk to UnityConnection</td>
<td>FusionConnect SIP trunk to UnityConnection</td>
<td>FusionConnect Defaultpool</td>
<td>800X</td>
<td>SIP Trunk</td>
<td>Full Service</td>
<td>Time In Full Service: 0 day 0 hour 0 minute</td>
<td>Full</td>
<td>FusionConnect Non Secure</td>
<td>SIP Trunk Profile</td>
<td></td>
</tr>
</tbody>
</table>

Figure 15: SIP Trunks List
### SIP Trunk Status

**Service Status:** Full Service  
**Duration:** Time In Full Service: 0 day 3 hours 57 minutes

### Device Information

<table>
<thead>
<tr>
<th>Product</th>
<th>SIP Trunk</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Protocol</td>
<td>SIP</td>
</tr>
<tr>
<td>Trunk Service Type</td>
<td>None (Default)</td>
</tr>
<tr>
<td><strong>Device Name</strong></td>
<td>FusionConnect</td>
</tr>
<tr>
<td><strong>Device Pool</strong></td>
<td>FusionConnect Devicepool</td>
</tr>
<tr>
<td>Common Device Configuration</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Call Classification</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>mrg list</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Location</th>
<th>Hub_None</th>
</tr>
</thead>
<tbody>
<tr>
<td>AAR Group</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Tunnelled Protocol</td>
<td>None</td>
</tr>
<tr>
<td>QSIG Variant</td>
<td>No Changes</td>
</tr>
<tr>
<td>ASN.1 ROSE OID Encoding</td>
<td>No Changes</td>
</tr>
<tr>
<td>Packet Capture Mode</td>
<td>None</td>
</tr>
<tr>
<td>Packet Capture Duration</td>
<td>0</td>
</tr>
<tr>
<td>Media Termination Point Required</td>
<td></td>
</tr>
<tr>
<td>Retry Video Call as Audio</td>
<td>✓</td>
</tr>
<tr>
<td>Path Replacement Support</td>
<td></td>
</tr>
<tr>
<td>Transmit UTF-8 for Calling Party Name</td>
<td></td>
</tr>
<tr>
<td>Transmit UTF-8 Names in QSIG APDU</td>
<td></td>
</tr>
<tr>
<td>Unattended Port</td>
<td></td>
</tr>
<tr>
<td>SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide and to end security. Failure to do so will expose keys and other information.</td>
<td></td>
</tr>
<tr>
<td>Consider Traffic on This Trunk Secure</td>
<td></td>
</tr>
<tr>
<td>Route Class Signaling Enabled</td>
<td>✓</td>
</tr>
<tr>
<td>Use Trusted Relay Point</td>
<td></td>
</tr>
<tr>
<td>PSTN Access</td>
<td>✓</td>
</tr>
<tr>
<td>Run On All Active Unified CM Nodes</td>
<td></td>
</tr>
</tbody>
</table>

### Intercompany Media Engine (IME)

| E.164 Transformation Profile | < None > |

---

**Figure 16: SIP Trunk to Cisco UBE**
**MLPP and Confidential Access Level Information**

<table>
<thead>
<tr>
<th>MLPP Domain</th>
<th>&lt; None &gt;</th>
</tr>
</thead>
<tbody>
<tr>
<td>Confidential Access Mode</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Confidential Access Level</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

**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.

<table>
<thead>
<tr>
<th>Number Type</th>
<th>Prefix</th>
<th>Strip Digits</th>
<th>Calling Search Space</th>
<th>Use Device Pool CSS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Incoming Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
<td>✓</td>
</tr>
</tbody>
</table>

**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.

<table>
<thead>
<tr>
<th>Number Type</th>
<th>Prefix</th>
<th>Strip Digits</th>
<th>Calling Search Space</th>
<th>Use Device Pool CSS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Incoming Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
<td>✓</td>
</tr>
</tbody>
</table>

**Connected Party Settings**

- Connected Party Transformation CSS: < None >
- Use Device Pool Connected Party Transformation CSS

---

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

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Called Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Use Device Pool Called Party Transformation CSS</td>
<td></td>
</tr>
<tr>
<td>Calling Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Use Device Pool Calling Party Transformation CSS</td>
<td></td>
</tr>
<tr>
<td>Calling Party Selection</td>
<td>Originator</td>
</tr>
<tr>
<td>Calling Line ID Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Calling Name Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Calling and Connected Party Info Format</td>
<td>Deliver DN only in connected party</td>
</tr>
<tr>
<td>Redirecting Diversion Header Delivery - Outbound</td>
<td></td>
</tr>
<tr>
<td>Redirecting Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Use Device Pool Redirecting Party Transformation CSS</td>
<td></td>
</tr>
</tbody>
</table>

### Caller Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Caller ID DN</td>
<td></td>
</tr>
<tr>
<td>Caller Name</td>
<td></td>
</tr>
<tr>
<td>Maintain Original Caller ID DN and Caller Name in Identity Headers</td>
<td>Box checked</td>
</tr>
</tbody>
</table>

### SIP Information

#### Destination

<table>
<thead>
<tr>
<th>Address</th>
<th>IPv6</th>
<th>Port</th>
</tr>
</thead>
<tbody>
<tr>
<td>10.0.0.23.20</td>
<td></td>
<td>5060</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>MTP Preferred Originating Code</td>
<td>7111law</td>
</tr>
<tr>
<td>BIF Presence Group</td>
<td>Standard Presence Group</td>
</tr>
<tr>
<td>SIP Trunk Security Profile</td>
<td>FusionConnect Non Secure SIP Trunk Profile</td>
</tr>
<tr>
<td>Redirecting Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Out-Of-Dialog Refer Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>SUBSCRIBE Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>SIP Profile</td>
<td>FusionConnect SIP Profile</td>
</tr>
<tr>
<td>DTMF Signaling Method</td>
<td>No Preference</td>
</tr>
</tbody>
</table>

#### Normalization Script

<table>
<thead>
<tr>
<th>Script</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Enable Trace</td>
<td>Check box</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Parameter Name</th>
<th>Parameter Value</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
</tr>
</tbody>
</table>

#### Recording Information

- None
- This trunk connects to a recording-enabled gateway
- This trunk connects to other clusters with recording-enabled gateways

#### Geolocation Configuration

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Geolocation</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Geolocation Filter</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

---

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

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

**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
   o “6” is removed before sending to Cisco UBE
2. For FAX call, Access Code “6”+10 digits number is used at Cisco Fax gateway
   o “6” is removed at Cisco UCM
   o 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
   o “*67” is removed at Cisco UCM
   o The rest of the number is sent to Cisco UBE to Fusion Connect network

![Route Patterns List](image)

Figure 19: Route Patterns List
### Pattern Definition

<table>
<thead>
<tr>
<th><strong>Route Pattern</strong></th>
<th>6.0.0</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Route Partition</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>Description</strong></td>
<td>FusionConnect RoutePattern for PSTN dialing</td>
</tr>
<tr>
<td><strong>Numbering Plan</strong></td>
<td>NANP</td>
</tr>
<tr>
<td><strong>Route Filter</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>MLPP Precedence</strong></td>
<td>Default</td>
</tr>
<tr>
<td></td>
<td>Apply Call Blocking Percentage</td>
</tr>
<tr>
<td></td>
<td>Resource Priority Namespace Network Domain</td>
</tr>
<tr>
<td></td>
<td><strong>Route Class</strong></td>
</tr>
<tr>
<td><strong>Gateway/Route List</strong></td>
<td>FusionConnect</td>
</tr>
<tr>
<td><strong>Route Option</strong></td>
<td>Route this pattern</td>
</tr>
<tr>
<td></td>
<td>Block this pattern</td>
</tr>
<tr>
<td></td>
<td>No Error</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th><strong>Call Classification</strong></th>
<th>OffNet</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>External Call Control Profile</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Allow Device Override</td>
<td>Provide Outside Dial Tone</td>
</tr>
<tr>
<td>Allow Overlap Sending</td>
<td>Urgent Priority</td>
</tr>
<tr>
<td>Require Forced Authorization Code</td>
<td></td>
</tr>
<tr>
<td>Authorization Level</td>
<td>0</td>
</tr>
<tr>
<td>Require Client Matter Code</td>
<td></td>
</tr>
</tbody>
</table>

### Calling Party Transformations

- **Use Calling Party's External Phone Number Mask**

<table>
<thead>
<tr>
<th><strong>Calling Party Transform Mask</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Prefix Digits (Outgoing Calls)</strong></td>
</tr>
</tbody>
</table>

| **Calling Line ID Presentation** | Default |
| **Calling Name Presentation** | Default |
| **Calling Party Number Type** | Cisco CallManager |
| **Connected Party Transformations** | **Connected Line ID Presentation** |
| **Connected Name Presentation** | Default |

---

**Figure 20: Route Pattern for Voice**
Figure 21: Route Pattern for Voice (Cont.)
**Figure 22: Route Pattern for Fax**
## Explanation

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Pattern</td>
<td>6.@ for Voice &amp; International Calls, 7XXX for Fax Call and *67.@ for Anonymous Call.</td>
<td>Specify appropriate Route Pattern</td>
</tr>
<tr>
<td>Gateway/Route List</td>
<td>Fusion Connect for Route Pattern 6.@, *67.@ and SIP_Trunk_To_Voice_Gateway for Route Pattern 7XXX</td>
<td>SIP Trunk name configured earlier</td>
</tr>
<tr>
<td>Numbering Plan</td>
<td>NANP for Route Pattern 6.@ and *67.@</td>
<td>North American Numbering Plan</td>
</tr>
<tr>
<td>Call Classification</td>
<td>OffNet for Route Pattern 6.@, 7XXX and *67.@</td>
<td>Restrict the transferring of an external call to an external device</td>
</tr>
<tr>
<td>Discard Digits</td>
<td>PreDot for Route Pattern 6.@ and *67.@</td>
<td>Specifies how to modify digit before they are sending to Fusion Connect network</td>
</tr>
<tr>
<td>Calling Line ID Presentation</td>
<td>Restricted for Route Pattern *67.@</td>
<td>Restrict the Caller ID Display of Calling party’s at external user endpoint.</td>
</tr>
<tr>
<td>Acronym</td>
<td>Definitions</td>
<td></td>
</tr>
<tr>
<td>---------</td>
<td>--------------------------------------------</td>
<td></td>
</tr>
<tr>
<td>CPE</td>
<td>Customer Premise Equipment</td>
<td></td>
</tr>
<tr>
<td>CUBE</td>
<td>Cisco Unified Border Element</td>
<td></td>
</tr>
<tr>
<td>CUCM</td>
<td>Cisco Unified Communications Manager</td>
<td></td>
</tr>
<tr>
<td>MTP</td>
<td>Media Termination Point</td>
<td></td>
</tr>
<tr>
<td>POP</td>
<td>Point of Presence</td>
<td></td>
</tr>
<tr>
<td>PSTN</td>
<td>Public Switched Telephone Network</td>
<td></td>
</tr>
<tr>
<td>SCCP</td>
<td>Skinny Client Control Protocol</td>
<td></td>
</tr>
<tr>
<td>SIP</td>
<td>Session Initiation Protocol</td>
<td></td>
</tr>
</tbody>
</table>
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