



IntelePeer SIP Trunking:

**Cisco Unified Communications Manager 11.5.1 with
Cisco Unified Border Element (CUBE 11.6.0) on ISR
4321/K9 [IOS-XE 16.5.1b] using SIP**

November 15, 2017



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Introduction

Service Providers today, such as IntelPeer, are offering alternative methods to connect to the PSTN via their IP networks. 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.

A demarcation device between these services and customer owned services is recommended. As an intermediary device between Cisco Unified Communications Manager and IntelPeer network, Cisco Unified Border Element (Cisco UBE) ISR 4321/K9 running IOS 16.5.1b 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 IntelPeer network.

This document assumes the reader is knowledgeable with the terminology and configuration of Cisco UCM (Cisco Unified Communications Manager). Only configuration settings specifically required for IntelPeer 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 a Cisco Unified Communications Manager (Cisco UCM) 11.5.1 and Cisco Unified Border Element (Cisco UBE) on ISR 4321/K9 [IOS-XE 16.5.1b] for connectivity to IntelPeer SIP Trunking service available in the former IntelPeer Business service area¹. The deployment model covered in this application note is CPE (Cisco UCM 11.5.1) to PSTN (IntelPeer).
- Testing was performed in accordance to IntelPeer 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 IntelPeer 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 IntelPeer 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

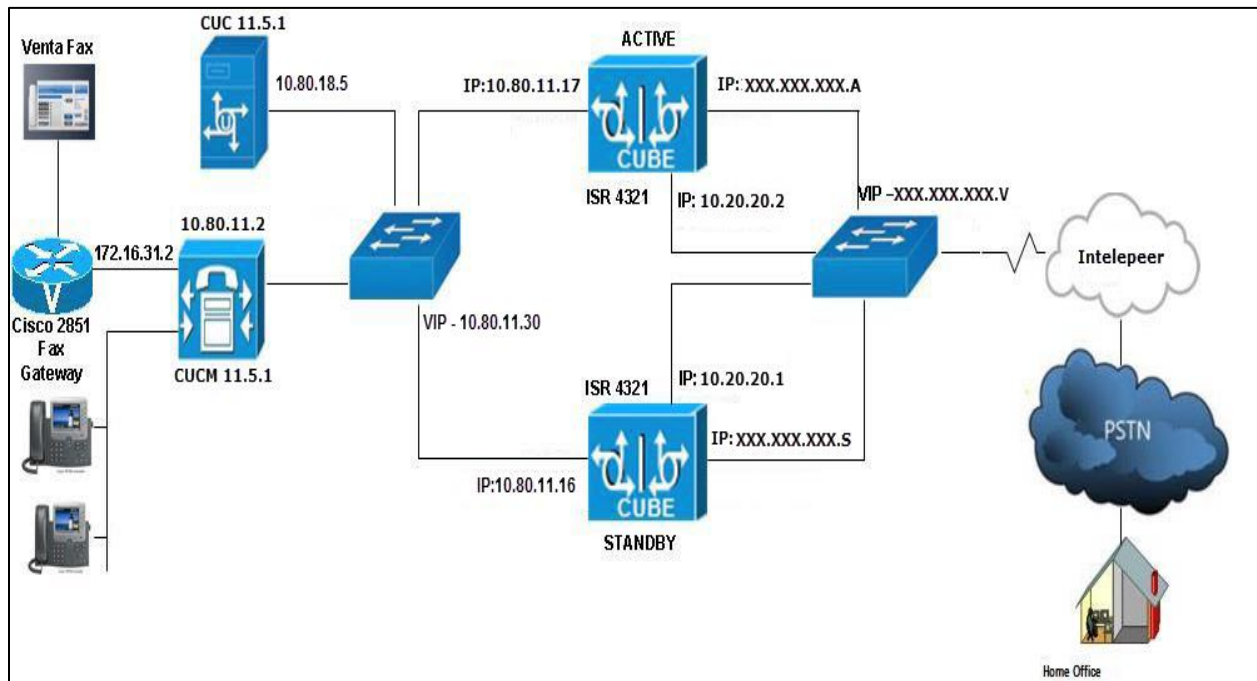


Figure 1: Network Topology

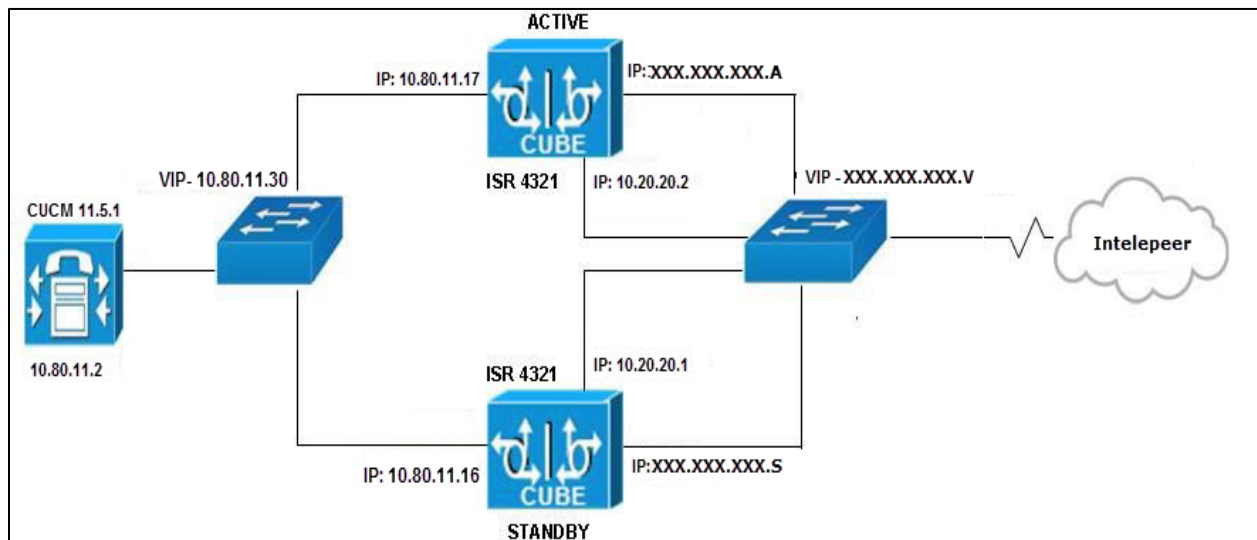


Figure 2: Cisco UBE High Availability



System Components

Hardware Requirements

- Cisco UCSC-C240-M3S VMWare host running ESXi 5.5 Standard
- Cisco ISR4431/K9 router as CUBE
- Cisco ISR4431/K9 (1RU) processor with 1684579K/6147K bytes of memory with 4 Gigabit Ethernet interfaces
- Processor board ID FTX1845AJ9S
- Cisco 2851 Fax Gateway
- IP phones 9971 (SIP) and 8945 (SIP)

Software Requirements

- Cisco Unified Communications Manager 11.5.1
- Cisco Unity Connection 11.5.1
- IOS 16.05.01b for ISR 4431/K9 Cisco Unified Border Element
- Cisco IOS-XE Software, ISR Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 16.05.01b, RELEASE SOFTWARE (fc1)
- IOS 15.1(4)M5 for Cisco 2851 Fax Gateway



Features

Features Supported

- Incoming and outgoing off-net calls using G711ULaw
- Call hold
- Call transfer (unattended and attended)
- Call conference
- Call forward (all, busy and no answer)
- Calling Line (number) Identification Presentation (CLIP)
- Calling Line (number) Identification Restriction (CLIR)
- DTMF relay (both directions) (RFC2833)
- Media flow-through on Cisco UBE
- Fax (G.711 pass-through and T38)

Features Not Supported

- Cisco IP phones used in this test do not support blind transfer
- In HA redundancy mode, the primary cube will not take over the primary/active role after a reboot/network outage

Caveats

- Caller ID is not updated after attended or unattended 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.



Configuration

Configuring Cisco Unified Border Element

Network Interface

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

```
interface GigabitEthernet0/0/0
ip address 192.65.79.134 255.255.255.128
negotiation auto
redundancy rii 2
redundancy group 1 ip 192.65.79.250 exclusive
!
interface GigabitEthernet0/0/1
ip address 10.80.11.17 255.255.255.0
negotiation auto
redundancy rii 1
redundancy group 1 ip 10.80.11.30 exclusive
```




Global Cisco UBE Settings

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

```
voice service voip
no ip address trusted authenticate
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 1
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
sip
bind control source-interface GigabitEthernet0/0/0
bind media source-interface GigabitEthernet0/0/0
rel1xx supported "rel100"
session refresh
header-passing
asserted-id pai
early-offer forced
midcall-signaling passthru
privacy-policy passthru
```

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 is used as the preferred codec for this testing and changed the preferences according to the test plan description

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

Dial Peer

Cisco UBE uses dial-peers to route the call accordingly based on the digits

```
dial-peer voice 10 voip
description Incoming from CUCM
huntstop
session protocol sipv2
incoming called-number [0-9]T
voice-class codec 1
voice-class sip profiles 100
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 sg3-to-g3
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 20 voip
description Outgoing to intelepeer
huntstop
destination-pattern [0-9]T
session protocol sipv2
session target ipv4:65.158.193.102
voice-class codec 1
voice-class sip conn-reuse
voice-class sip options-ping 60
voice-class sip profiles 100
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 sg3-to-g3
```



```
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 30 voip
description Incoming from intelepeer
huntstop
session protocol sipv2
incoming called-number 1980
voice-class codec 1
voice-class sip profiles 100
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 sg3-to-g3
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 40 voip
description Outgoing to CUCM
huntstop
destination-pattern 1980
session protocol sipv2
session target ipv4:10.80.11.2
voice-class codec 1
voice-class sip options-ping 60
voice-class sip profiles 100
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 sg3-to-g3
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
```



Call Flow

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

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

CPE callers make outbound PSTN calls by dialing a “7” 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 “7”. A “7.@" 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 IntelPeer, Caller dial 7 prefix followed by the target 10Digit DID no for that extension number, 7 was stripped and the 10 digits number was send to Cisco UBE, Cisco UBE sends the full 10 digits DID under Dial Peer 200 and send to IntelPeer network which will direct back to Cisco UBE and handled same as normal incoming PSTN call. For International calls same pattern 7 followed by 011, country code and calling no is used.

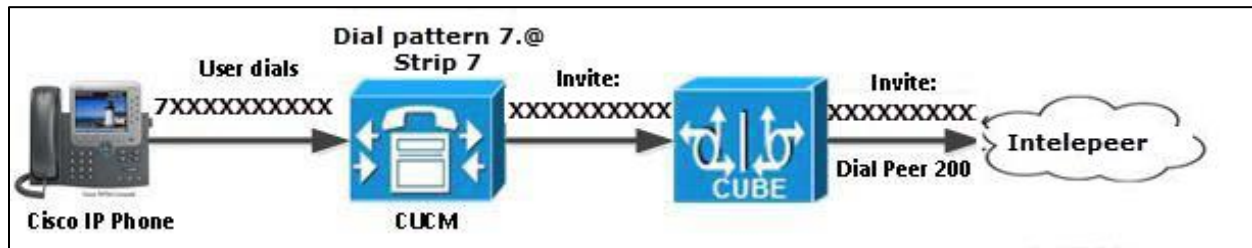


Figure 3: Outbound Voice Call

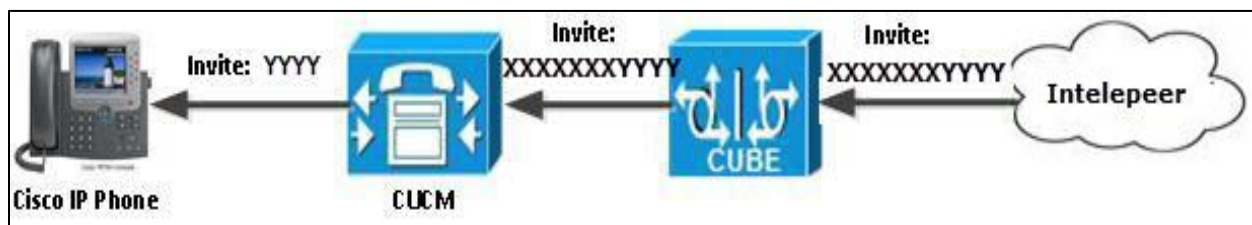


Figure 4: Inbound Voice Call

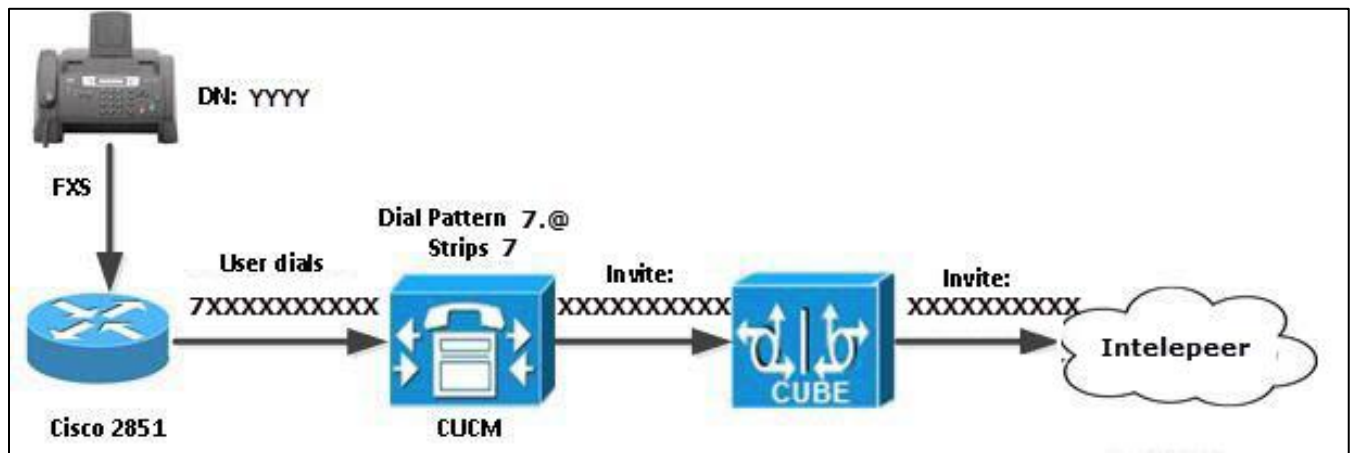


Figure 5: Outbound Fax Call

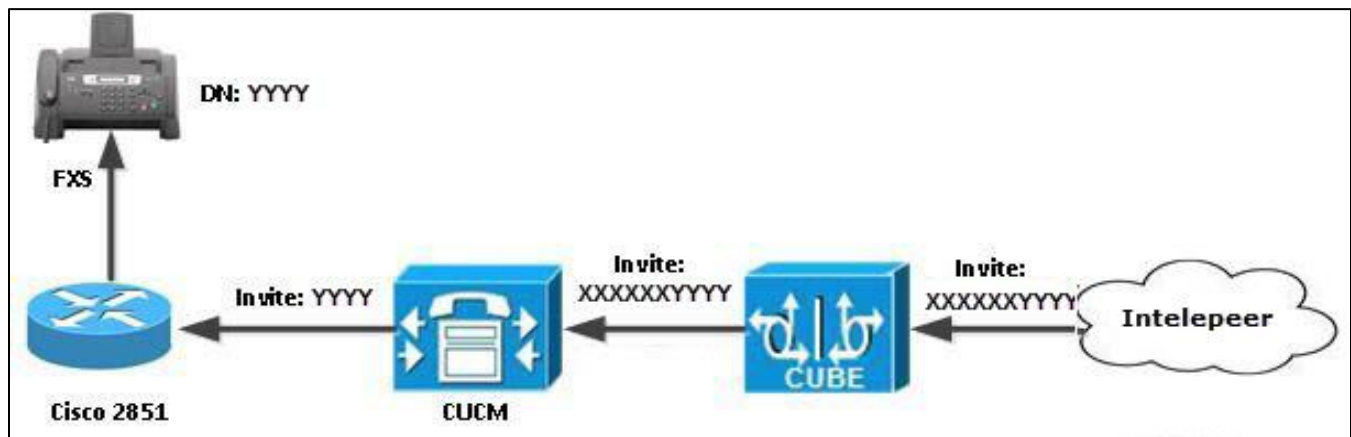


Figure 6: Inbound Fax Call

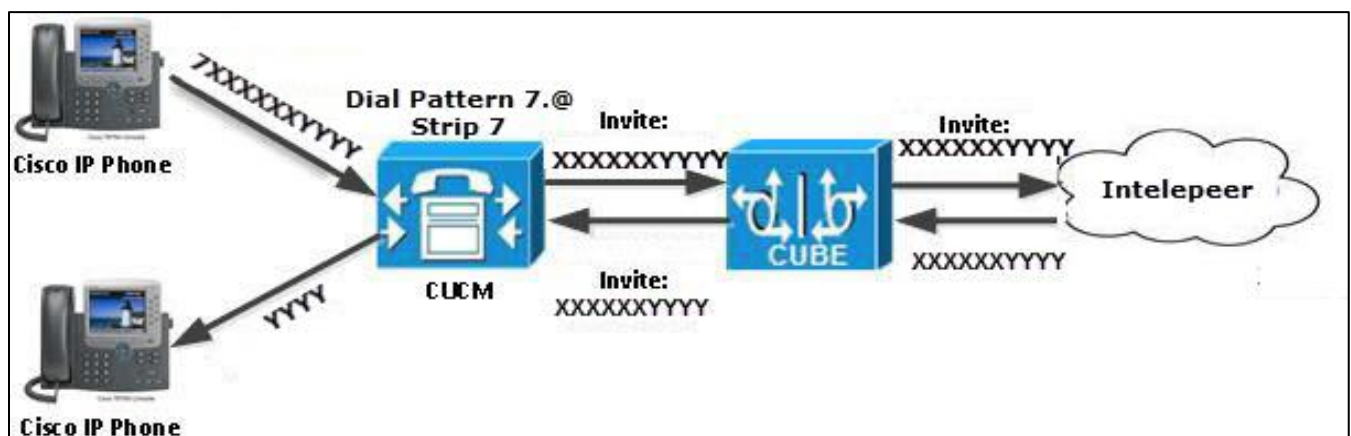


Figure 7: PBX to PBX via IntelePeer Call



Configuration Example

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

Active Cisco UBE

```
intelPeerCube2#sh run
Building configuration...
Current configuration : 5590 bytes
!
! Last configuration change at 08:13:43 UTC Thu Oct 5 2017 by cisco
!
version 16.5
service timestamps debug datetime msec
service timestamps log datetime msec
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname intelPeerCube2
!
boot-start-marker
boot-end-marker
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
enable secret 5
!
no aaa new-model
!
subscriber templating
!
multilink bundle-name authenticated
!
voice service voip
no ip address trusted authenticate
mode border-element license capacity 20
allow-connections sip to sip
```



```
redundancy-group 1
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
sip
bind control source-interface GigabitEthernet0/0/0
bind media source-interface GigabitEthernet0/0/0
rel1xx supported "rel100"
session refresh
header-passing
asserted-id pai
early-offer forced
midcall-signaling passthru
privacy-policy passthru
!
voice class codec 1
codec preference 1 g711ulaw
codec preference 2 g729r8
!
voice class codec 2
codec preference 1 g729r8
codec preference 2 g711ulaw
!
voice class sip-profiles 100
request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:98023311@12>"
!
license udi pid ISR4321/K9 sn FDO19220MQ9
license accept end user agreement
license boot suite AdvUCSuiteK9
license boot level appxk9
license boot level uck9
license boot level securityk9
!
diagnostic bootup level minimal
spanning-tree extend system-id
!
username cisco privilege 15 password 0 *****
!
redundancy
mode none
application redundancy
group 1
name voice-b2bha
timers delay 30 reload 60
control GigabitEthernet0/1/0 protocol 1
```



```
data GigabitEthernet0/1/0
track 1 shutdown
track 2 shutdown
!
track 1 interface GigabitEthernet0/0/1 line-protocol
!
track 2 interface GigabitEthernet0/0/0 line-protocol
!
interface GigabitEthernet0/0/0
ip address xxx.xxx.xxx.xxx 255.255.255.128
negotiation auto
redundancy rii 2
redundancy group 1 ip xxx.xxx.xxx.xxx exclusive
!
interface GigabitEthernet0/0/1
ip address 10.80.11.17 255.255.255.0
negotiation auto
redundancy rii 1
redundancy group 1 ip 10.80.11.30 exclusive
!
interface GigabitEthernet0/1/0
ip address 20.0.0.2 255.255.255.252
negotiation auto
!
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
shutdown
negotiation auto
!
threat-visibility
ip forward-protocol nd
ip http server
ip http authentication local
ip http secure-server
ip tftp source-interface GigabitEthernet0
ip route 0.0.0.0 0.0.0.0 192.65.79.129
ip route 10.64.0.0 255.255.0.0 10.80.11.1
ip route 172.16.24.0 255.255.248.0 10.80.11.1
!
ip ssh server algorithm encryption aes128-ctr aes192-ctr aes256-ctr
ip ssh client algorithm encryption aes128-ctr aes192-ctr aes256-ctr
!
```




```
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 10 voip
description Incoming from CUCM
huntstop
session protocol sipv2
incoming called-number [0-9]T
voice-class codec 1
voice-class sip profiles 100
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 sg3-to-g3
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 20 voip
description Outgoing to intelepeer
huntstop
destination-pattern [0-9]T
session protocol sipv2
session target ipv4:xxx.xxx.xxx.xxx
voice-class codec 1
voice-class sip conn-reuse
voice-class sip options-ping 60
voice-class sip profiles 100
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 sg3-to-g3
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 30 voip
```



```
description Incoming from intelepeer
huntstop
session protocol sipv2
incoming called-number 1980
voice-class codec 1
voice-class sip profiles 100
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 sg3-to-g3
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 40 voip
description Outgoing to CUCM
huntstop
destination-pattern 1980
session protocol sipv2
session target ipv4:10.80.11.2
voice-class codec 1
voice-class sip options-ping 60
voice-class sip profiles 100
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 sg3-to-g3
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
sip-ua
!
line con 0
transport input none
stopbits 1
line aux 0
stopbits 1
line vty 0 4
logging synchronous
login local
!
no network-clock synchronization automatic
```



!

end

Standby Cisco UBE

intelPeerCube1#sh run

Building configuration...

Current configuration : 5617 bytes

!

! Last configuration change at 09:12:23 UTC Thu Oct 5 2017 by cisco

!

version 16.5

service timestamps debug datetime msec

service timestamps log datetime msec

platform qfp utilization monitor load 80

no platform punt-keepalive disable-kernel-core

!

hostname intelPeerCube1

!

boot-start-marker

boot system flash isr4300-universalk9.16.05.01b.SPA.bin

boot-end-marker

!

vrf definition Mgmt-intf

!

address-family ipv4

exit-address-family

!

address-family ipv6

exit-address-family

!

enable secret 5

!

no aaa new-model

!

subscriber templating

!

multilink bundle-name authenticated

!

voice service voip

no ip address trusted authenticate



```
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 1
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
sip
bind control source-interface GigabitEthernet0/0/0
bind media source-interface GigabitEthernet0/0/0
rel1xx supported "rel100"
session refresh
header-passing
asserted-id pai
early-offer forced
midcall-signaling passthru
privacy-policy passthru
!
voice class codec 1
codec preference 1 g711ulaw
codec preference 2 g729r8
!
voice class codec 2
codec preference 1 g729r8
codec preference 2 g711ulaw
!
voice class sip-profiles 100
request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:98023311@12>"
!
license udi pid ISR4321/K9 sn FDO19220MSQ
license accept end user agreement
license boot suite AdvUCSuiteK9
license boot level appxk9
license boot level securityk9
!
diagnostic bootup level minimal
spanning-tree extend system-id
!
username cisco privilege 15 password 0
!
redundancy
mode none
application redundancy
```



```
group 1
name voice-b2bha
timers delay 30 reload 60
control GigabitEthernet0/1/0 protocol 1
data GigabitEthernet0/1/0
track 1 shutdown
track 2 shutdown
!
track 1 interface GigabitEthernet0/0/1 line-protocol
!
track 2 interface GigabitEthernet0/0/0 line-protocol
!
interface GigabitEthernet0/0/0
ip address xxx.xxx.xxx.xxx 255.255.255.128
negotiation auto
redundancy rii 2
redundancy group 1 ip xxx.xxx.xxx.xxx exclusive
!
interface GigabitEthernet0/0/1
ip address 10.80.11.16 255.255.255.0
negotiation auto
redundancy rii 1
redundancy group 1 ip 10.80.11.30 exclusive
!
interface GigabitEthernet0/1/0
ip address 20.0.0.1 255.255.255.0
negotiation auto
!
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
negotiation auto
!
threat-visibility
ip forward-protocol nd
ip http server
ip http authentication local
ip http secure-server
ip route 0.0.0.0 0.0.0.0 192.65.79.129
ip route 10.64.0.0 255.255.0.0 10.80.11.1
```



```
ip route 172.16.24.0 255.255.248.0 10.80.11.1
!
ip ssh server algorithm encryption aes128-ctr aes192-ctr aes256-ctr
ip ssh client algorithm encryption aes128-ctr aes192-ctr aes256-ctr
!
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 10 voip
description Incoming from CUCM
huntstop
session protocol sipv2
incoming called-number [0-9]T
voice-class codec 1
voice-class sip profiles 100
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 sg3-to-g3
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 20 voip
description Outgoing to intelepeer
huntstop
destination-pattern [0-9]T
session protocol sipv2
session target ipv4:xxx.xxx.xxx.xxx
voice-class codec 1
voice-class sip conn-reuse
voice-class sip options-ping 60
voice-class sip profiles 100
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 sg3-to-g3
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 30 voip
description Incoming from intelepeer
huntstop
session protocol sipv2
incoming called-number 1980
voice-class codec 1
voice-class sip profiles 100
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 sg3-to-g3
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 40 voip
description Outgoing to CUCM
huntstop
destination-pattern 1980
session protocol sipv2
session target ipv4:10.80.11.2
voice-class codec 1
voice-class sip options-ping 60
voice-class sip profiles 100
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 sg3-to-g3
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad!
sip-ua
!
```



```
line con 0
exec-timeout 0 0
transport input none
stopbits 1
line aux 0
stopbits 1
line vty 0 4
logging synchronous
login local
!
no network-clock synchronization automatic
ntp server time-pnp.cisco.com
!
end
```




Configuring Cisco Unified Communications Manager

Cisco UCM Version



Figure 8: Cisco UCM Version

Cisco Call Manager Service Parameters

Navigation: System → Service Parameters

1. Select **Server***: clus21sub1--CUCM Voice/Video (Active)
2. Select **Service***: Cisco CallManager (Active)
3. All other fields are set to default values

Select Server and Service	
Server*	clus21sub1--CUCM Voice/Video (Active)
Service*	Cisco CallManager (Active)

All parameters apply only to the current server except parameters that are in the cluster-wide group(s).

Cisco CallManager (Active) Parameters on server clus21sub1--CUCM Voice/Video (Active)	
Parameter Name	Parameter Value
Call Throttling	
Code Yellow Entry Latency *	20
Code Yellow Exit Latency Calculation *	40
Code Yellow Duration *	5
Max Events Allowed *	2000
System Throttle Sample Size *	10

Figure 9: Service Parameters



Offnet Calls via IntelPeer SIP Trunk

Off-net calls are served by SIP trunks configured between Cisco UCM and the IntelPeer network and calls are routed via Cisco UBE

SIP Trunk Security Profile

Navigation: System → Security → SIP Trunk Security Profile

1. **Name*:** Non Secure SIP Trunk Profile
2. **Description:** non Secure SIP Trunk Profile authenticated by null String

SIP Trunk Security Profile Information

Name* Non Secure SIP Trunk Profile

Description Non Secure SIP Trunk Profile authenticated by null String

Device Security Mode Non Secure

Incoming Transport Type* TCP+UDP

Outgoing Transport Type UDP

☐ Enable Digest Authentication

Nonce Validity Time (mins)* 600

X.509 Subject Name

Incoming Port* 5060

☐ Enable Application level authorization

☐ Accept presence subscription

☐ Accept out-of-dialog refer**

☐ Accept unsolicited notification

☐ Accept replaces header

☐ Transmit security status

☐ Allow charging header

SIP V.150 Outbound SDP Offer Filtering* Use Default Filter

Figure 10: SIP Trunk Security Profile

Explanation

Parameter	Value	Description
Incoming Transport Type	TCP + UDP	
Outgoing Transport Type	UDP	SIP trunks to IntelPeer 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

SIP Profile will be later associated with the SIP trunk

Navigation: Device → Device Settings → SIP Profile

1. **Name*:** Intelepeer SIP Profile, for example
2. **Description:** Intelepeer SIP Profile, for example

SIP Profile Information	
Name*	Intelepeer sip profile
Description	Intelepeer sip profile
Default MTP Telephony Event Payload Type*	101
Early Offer for G.Clear Calls*	Disabled
User-Agent and Server header information*	Send Unified CM Version Information as User-Agent
Version in User Agent and Server Header*	Major And Minor
Dial String Interpretation*	Phone number consists of characters 0-9, *, #, and
Confidential Access Level Headers*	Disabled
<input type="checkbox"/> Redirect by Application	
<input type="checkbox"/> Disable Early Media on 180	
<input type="checkbox"/> Outgoing T.38 INVITE include audio mline	
<input type="checkbox"/> Offer valid IP and Send/Receive mode only for T.38 Fax Relay	
<input type="checkbox"/> Use Fully Qualified Domain Name in SIP Requests	
<input type="checkbox"/> Assured Services SIP conformance	
<input type="checkbox"/> Enable External QoS**	

SDP Information	
SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*	TIAS and AS
SDP Transparency Profile	Pass all unknown SDP attributes
Accept Audio Codec Preferences in Received Offer*	Default
<input type="checkbox"/> Require SDP Inactive Exchange for Mid-Call Media Change	
<input type="checkbox"/> Allow RR/RS bandwidth modifier (RFC 3556)	

Parameters used in Phone	
Timer Invite Expires (seconds)*	180
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10
Media Port Ranges	<input checked="" type="radio"/> Common Port Range for Audio and Video <input type="radio"/> Separate Port Ranges for Audio and Video
Start Media Port*	16384
Stop Media Port*	32766

Figure 11: SIP Profile



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
<input checked="" type="checkbox"/> Conference Join Enabled	
<input type="checkbox"/> RFC 2543 Hold	
<input checked="" type="checkbox"/> Semi Attended Transfer	
<input type="checkbox"/> Enable VAD	
<input type="checkbox"/> Stutter Message Waiting	
<input type="checkbox"/> 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 12: SIP Profile (Cont.)



Trunk Specific Configuration

Reroute Incoming Request to new Trunk based on*

Resource Priority Namespace List

SIP Rel1XX Options*

Video Call Traffic Class*

Calling Line Identification Presentation*

Session Refresh Method*

Early Offer support for voice and video calls*

☐ Enable ANAT

☐ Deliver Conference Bridge Identifier

☐ Allow Passthrough of Configured Line Device Caller Information

☐ Reject Anonymous Incoming Calls

☐ Reject Anonymous Outgoing Calls

☐ Send ILS Learned Destination Route String

SIP OPTIONS Ping

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

Ping Interval for In-service and Partially In-service Trunks (seconds)*

Ping Interval for Out-of-service Trunks (seconds)*

Ping Retry Timer (milliseconds)*

Ping Retry Count*

SDP Information

☒ Send send-receive SDP in mid-call INVITE

☐ Allow Presentation Sharing using BFCP

☐ Allow iX Application Media

☐ Allow multiple codecs in answer SDP

Figure 13: 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 Messages	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

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
cucm-faxgateway	cucm-faxgateway		G711 Pool	8021				SIP Trunk	Full Service	Time In Full Service: 0 day 9 hours 9 minutes	Non Secure SIP Trunk Profile
Intelepeer_Trunk	Intelepeer_Trunk		G729 Pool	7.0				SIP Trunk	Unknown - OPTIONS Ping not enabled		Non Secure SIP Trunk Profile

Figure 14: SIP Trunks List

Device Information

Product:

SIP Trunk

Device Protocol:

SIP

Trunk Service Type

None(Default)

Device Name*

Intelepeer_Trunk

Description

Intelepeer_Trunk

Device Pool*

G711 Pool ▾

Common Device Configuration

< None > ▾

Call Classification*

Use System Default ▾

Media Resource Group List

MRGL_MTP ▾

Location*

Hub_None ▾

AAR Group

< None > ▾

Tunneled Protocol*

None ▾

QSIG Variant*

No Changes ▾

ASN.1 ROSE OID Encoding*

No Changes ▾

Packet Capture Mode*

None ▾

Packet Capture Duration

0

☐ Media Termination Point Required

☒ Retry Video Call as Audio

☐ Path Replacement Support

☐ Transmit UTF-8 for Calling Party Name

☐ Transmit UTF-8 Names in QSIG APDU

☐ Unattended Port

☐ SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will and other information.

Consider Traffic on This Trunk Secure*

When using both sRTP and TLS ▾

Route Class Signaling Enabled*

Default ▾

Use Trusted Relay Point*

Default ▾

☒ PSTN Access

☒ Run On All Active Unified CM Nodes

Figure 15: SIP Trunk to Cisco UBE



Intercompany Media Engine (IME)
E.164 Transformation Profile < None >

MLPP and Confidential Access Level Information
MLPP Domain < None >
Confidential Access Mode < None >
Confidential Access Level < None >

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 >	<input checked="" type="checkbox"/>

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 >	<input checked="" type="checkbox"/>

Figure 16: SIP Trunk to Cisco UBE (Cont.)



Connected Party Settings		
Connected Party Transformation CSS < None >		
<input checked="" type="checkbox"/> Use Device Pool Connected Party Transformation CSS		
Outbound Calls		
Called Party Transformation CSS < None >		
<input checked="" type="checkbox"/> Use Device Pool Called Party Transformation CSS		
Calling Party Transformation CSS < None >		
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS		
Calling Party Selection*	Originator	
Calling Line ID Presentation*	Default	
Calling Name Presentation*	Default	
Calling and Connected Party Info Format*	Deliver DN only in connected party	
<input checked="" type="checkbox"/> Redirecting Diversion Header Delivery - Outbound		
Redirecting Party Transformation CSS < None >		
<input checked="" type="checkbox"/> Use Device Pool Redirecting Party Transformation CSS		
Caller Information		
Caller ID DN <input type="text"/>		
Caller Name <input type="text"/>		
<input type="checkbox"/> Maintain Original Caller ID DN and Caller Name in Identity Headers		
SIP Information		
Destination		
<input type="checkbox"/> Destination Address is an SRV		
	Destination Address	Destination Address IPv6
1*	<input type="text" value="10.80.11.30"/>	<input type="text" value="5060"/>
MTP Preferred Originating Codec* 711ulaw		
BLF Presence Group* Standard Presence group		
SIP Trunk Security Profile* Non Secure SIP Trunk Profile		
Rerouting Calling Search Space < None >		
Out-Of-Dialog Refer Calling Search Space < None >		
SUBSCRIBE Calling Search Space < None >		
SIP Profile* Intelepeer sip profile View Details		
DTMF Signaling Method* No Preference		
Recording Information		
<input checked="" type="radio"/> None		
<input type="radio"/> This trunk connects to a recording-enabled gateway		
<input type="radio"/> This trunk connects to other clusters with recording-enabled gateways		
Geolocation Configuration		
Geolocation < None >		
Geolocation Filter < None >		
<input type="checkbox"/> Send Geolocation Information		

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



Explanation

Parameter	Value	Description
Device Name	Intelepeer _Trunk	Name for the trunk
Device Pool	G711pool	Default Device Pool is used for this trunk
Media Resource Group List	MRGL_MTP	MRG with resources: ANN, CFB, MOH and MTP
Significant Digits	4	4 digits Extension for all CPE phones
Destination Address	10.80.11.30	IP address of the Cisco UBE Virtual LAN
SIP Trunk Security Profile	Non Secure SIP Trunk Profile	SIP Trunk Security Profile configured earlier
SIP Profile	Intelepeer SIP Profile	SIP Profile configured earlier

Dial Plan

Route Pattern Configuration

Navigation: Call Routing → Route/Hunt → Route Pattern

Route patterns are configured as below:

- Cisco IP phone dial “7”+10 digits number to access PSTN via Cisco UBE
 - “7” is removed before sending to Cisco UBE
- For FAX call, Access Code “7”+10 digits number is used at Cisco Fax gateway
 - “7” is removed at Cisco UCM
 - The rest of the number is sent to Cisco UBE to IntelPeer network
- Incoming fax call to 8021 will be sent to Cisco Fax gateway

Find Route Patterns where	Pattern	begins with	Find	Clear Filter	+	-
<input type="checkbox"/>	Pattern ^	Description	Partition	Route Filter	Associated Device	Copy
<input type="checkbox"/>	7.@				Intelepeer_Trunk	
<input type="checkbox"/>	8021				cucm-faxgateway	

Figure 18: Route Patterns List



Pattern Definition		
Route Pattern*	7.@	
Route Partition	< None >	
Description		
Numbering Plan*	NANP	
Route Filter	< None >	
MLPP Precedence*	Default	
<input type="checkbox"/> Apply Call Blocking Percentage		
Resource Priority Namespace Network Domain	< None >	
Route Class*	Default	
Gateway/Route List*	Intelepeer_Trunk	
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error	
Call Classification*	OffNet	
External Call Control Profile	< None >	
<input type="checkbox"/> Allow Device Override	<input checked="" type="checkbox"/> Provide Outside Dial Tone	
<input type="checkbox"/> Require Forced Authorization Code	<input type="checkbox"/> Allow Overlap Sending	
Authorization Level*	0	
<input type="checkbox"/> Require Client Matter Code		
Calling Party Transformations		
<input checked="" type="checkbox"/> Use Calling Party's External Phone Number Mask		
Calling Party Transform Mask		
Prefix Digits (Outgoing Calls)		
Calling Line ID Presentation*	Default	
Calling Name Presentation*	Default	
Calling Party Number Type*	Cisco CallManager	
Calling Party Numbering Plan*	Cisco CallManager	
Connected Party Transformations		
Connected Line ID Presentation*	Default	
Connected Name Presentation*	Default	
Called Party Transformations		
Discard Digits	PreDot	
Called Party Transform Mask		
Prefix Digits (Outgoing Calls)		
Called Party Number Type*	Cisco CallManager	
Called Party Numbering Plan*	Cisco CallManager	
ISDN Network-Specific Facilities Information Element		
Network Service Protocol	-- Not Selected --	
Carrier Identification Code		
Network Service	Service Parameter Name	Service Parameter Value
-- Not Selected --	< Not Exist >	

Figure 19: Route Pattern for Voice



Pattern Definition		
Route Pattern *	8021	
Route Partition	< None >	
Description		
Numbering Plan	-- Not Selected --	
Route Filter	< None >	
MLPP Precedence *	Default	
<input type="checkbox"/> Apply Call Blocking Percentage		
Resource Priority Namespace Network Domain	< None >	
Route Class *	Default	
Gateway/Route List *	cucm-faxgateway	(Edit)
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error	
Call Classification *	OffNet	
External Call Control Profile	< None >	
<input type="checkbox"/> Allow Device Override	<input checked="" type="checkbox"/> Provide Outside Dial Tone	<input type="checkbox"/> Allow Overlap Sending
<input type="checkbox"/> Require Forced Authorization Code	<input type="checkbox"/> Urgent Priority	
Authorization Level *	0	
<input type="checkbox"/> Require Client Matter Code		

Calling Party Transformations	
<input type="checkbox"/> 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

Connected Party Transformations	
Connected Line ID Presentation *	Default
Connected Name Presentation *	Default

Called Party Transformations	
Discard Digits	< None >
Called Party Transform Mask	
Prefix Digits (Outgoing Calls)	
Called Party Number Type *	Cisco CallManager
Called Party Numbering Plan *	Cisco CallManager

ISDN Network-Specific Facilities Information Element		
Network Service Protocol	-- Not Selected --	
Carrier Identification Code		
Network Service	Service Parameter Name	Service Parameter Value
-- Not Selected --	< Not Exist >	

Figure 20: Route Pattern for Fax



Explanation

Setting	Value	Description
Route Pattern	7.@ for Voice & International Calls and 8021 for Fax Call	Specify appropriate Route Pattern
Gateway/Route List	IntelPeer for Route Pattern 7.@ and 8021 for SIP Trunk To Fax Gateway.	SIP Trunk name configured earlier
Numbering Plan	NANP for Route Pattern 7.@	North American Numbering Plan
Call Classification	OffNet for Route Pattern 7.@ and 8021	Restrict the transferring of an external call to an external device
Discard Digits	PreDot for Route Pattern 7.@	Specifies how to modify digit before they are sent to IntelPeer network



Acronyms

Acronym	Definition
CPE	Customer Premise Equipment
Cisco UBE	Cisco Unified Border Element
Cisco UCM	Cisco Unified Communications Manager
MTP	Media Termination Point
POP	Point of Presence
PSTN	Public Switched Telephone Network
ESBC	Enterprise Session Border Controller
SCCP	Skinny Client Control Protocol
SIP	Session Initiation Protocol

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