

Application Note

IntelePeer SIP Trunking:

Cisco Unified Communications Manager 11.0.1 with Cisco Unified Border Element (CUBE 11.5.0) on ISR 4321 [IOS - 15.6(1) S] using SIP

April 21, 2016

Table of Contents

Introduction4
Network Topology5
System Components5
Hardware Requirements5
Software Requirements5
Features6
Features Supported6
Features Not Supported6
Caveats6
Configuration7
Configuring Cisco Unified Border Element7
Network Interface7
Global Cisco UBE Settings7
-
Media Passing through Cisco UBE (Media flow-through vs. Media flow-around)
Media Passing through Cisco UBE (Media flow-through vs. Media flow-around)
Media Passing through Cisco UBE (Media flow-through vs. Media flow-around)
Media Passing through Cisco UBE (Media flow-through vs. Media flow-around)
Media Passing through Cisco UBE (Media flow-through vs. Media flow-around)
Media Passing through Cisco UBE (Media flow-through vs. Media flow-around) 8 Codecs 8 Dial Peer 9 Call Flow 12 Configuration Example 13 Configuring Cisco Unified Communications Manager 37
Media Passing through Cisco UBE (Media flow-through vs. Media flow-around) 8 Codecs 8 Dial Peer 9 Call Flow 12 Configuration Example 13 Configuring Cisco Unified Communications Manager 37 Cisco UCM Version 37
Media Passing through Cisco UBE (Media flow-through vs. Media flow-around) 8 Codecs 8 Dial Peer 9 Call Flow 12 Configuration Example 13 Configuring Cisco Unified Communications Manager 37 Cisco UCM Version 37 Cisco Call Manager Service Parameters 37
Media Passing through Cisco UBE (Media flow-through vs. Media flow-around) 8 Codecs 8 Dial Peer 9 Call Flow 12 Configuration Example 13 Configuring Cisco Unified Communications Manager 37 Cisco UCM Version 37 Cisco Call Manager Service Parameters 37 Offnet Calls via IntelePeer SIP Trunk 38
Media Passing through Cisco UBE (Media flow-through vs. Media flow-around) 8 Codecs 8 Dial Peer 9 Call Flow 12 Configuration Example 13 Configuring Cisco Unified Communications Manager 37 Cisco UCM Version 37 Cisco Call Manager Service Parameters 37 Offnet Calls via IntelePeer SIP Trunk 38 Dial Plan 46
Media Passing through Cisco UBE (Media flow-through vs. Media flow-around)
Media Passing through Cisco UBE (Media flow-through vs. Media flow-around) .8 Codecs .8 Dial Peer .9 Call Flow .12 Configuration Example .13 Configuring Cisco Unified Communications Manager .37 Cisco UCM Version .37 Cisco Call Manager Service Parameters .37 Offnet Calls via IntelePeer SIP Trunk .38 Dial Plan .46 Acronyms .51 Important Information .52



Table of Figures

5
12
12
12
13
13
37
37
38
39
40
41
42
43
44
47

Introduction

Service Providers today, such as IntelePeer, 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.

IntelePeer 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 IntelePeer network, Cisco Unified Border Element (Cisco UBE) ISR 4321/K9 running IOS 15.6(1)S can be used. The Cisco Unified Border Element 15.6(1)S provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 11.0.1 connected to IntelePeer IP 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 IntelePeer 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) 11.0.1 and Cisco Unified Border Element (Cisco UBE) on ISR 4321/K9 [IOS 15.6(1)S] for connectivity to IntelePeer SIP Trunking service. The deployment model covered in this application note is CPE (Cisco UCM 11.0.1) to PSTN (IntelePeer).
- Testing was performed in accordance to IntelePeer 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 IntelePeer 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 IntelePeer 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

- Cisco IP Phones 7975,7965 and 9971 phones are the devices primarily used throughout the testing to place or receive calls
- VentaFax Soft Client is used to perform all fax related scenarios. The fax client is connected to SIP Gateway via FXS port which in turn communicates with Cisco UCM over SIP.

System Components

Hardware Requirements

- Cisco UCS-C240-M3S VMWare host running ESXi 5.5 Standard
- Cisco ISR 4321/K9 Router as CUBE
- Cisco 2851 Fax Gateway
- IP Phones 9971(SIP),7965 (SCCP) and 7975 (SIP)

Software Requirements

- Cisco Unified Communications Manager 11.0.1
- Cisco Unity Connection 11.0.1
- IOS 15.6(1)S for ISR 4321/K9 Cisco Unified Border Element
- IOS 15.1(4)M5 for Cisco 2851 Fax Gateway



Features

Features Supported

- Incoming and outgoing off-net calls using G711ULaw and G729
- Call Hold
- Call Transfer (unattended and attended)
- Call Forward (all, busy and no answer)
- Calling Line (number) Identification Presentation (CLIP)
- Calling Line (number) Identification Restriction (CLIR)
- DTMF (RFC2833)
- Media flow-through on Cisco UBE
- Fax (G.711 pass-through and T.38)

Features Not Supported

- Cisco IP phones used in this test do not support blind transfer
- Fax re-invite is not supported by Service Provider

Caveats

- CLID is not updated on PSTN phones for transfer (attended and unattended) OffNet PSTN scenarios. Caller ID is not updated at PSTN once transfer is completed by PBX. Cisco UBE modify PAI/PPI header and forward to network in the tested release. CISCO BUG ID: CSCuv04539
- Transfer scenario from CPE observed no way speech path issue between PSTN phones while mid call codec negotiated from g729 to g711ulaw
- Privacy: ID sent from PBX has stripped from IntelePeer while make call from PBX Phone A to PBX phone B. The caller ID shows as anonymous when making call from PBX to PSTN.
- Intermittent no way speech path observed for an inbound 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 10.80.18.21 255.255.255.0 media-type rj45 negotiation auto redundancy rii 1 redundancy group 1 ip 10.80.18.20 exclusive ! interface GigabitEthernet0/0/1 ip address 192.65.x.x 255.255.128 negotiation auto redundancy rii 2 redundancy group 1 ip 192.65.x.x exclusive

Global Cisco UBE Settings

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

voice service voip no ip address trusted authenticate address-hiding mode border-element license capacity 20 allow-connections sip to sip redundancy-group 1 fax protocol pass-through G711ULaw sip rel1xx supported "rel100" session refresh asserted-id pai privacy pstn



early-offer forced

midcall-signaling passthru

privacy-policy passthru

g729 annexb-all

!

Explanation

Command	Description
allow-connections sip to sip	Allow IP2IP connections between two SIP call legs
fax protocol	Specifies the fax protocol
asserted-id	Specifies the type of privacy header in the outgoing SIP requests and response messages
early-offer forced	Enables SIP Delayed-Offer to Early-Offer globally
midcall-signaling passthru	Passes SIP messages from one IP leg to another IP leg

Media Passing through Cisco UBE (Media flow-through vs. Media flow-around)

Default Cisco UBE configuration enables Cisco UBE to work in flow-through mode (this test uses the flow-through mode). In order to enable flow-around mode, perform the following actions:

voice service voip

media flow-around

Codecs

G729 is used as the preferred codec for this testing and changed the codecs according to the test plan description

voice class codec 1

codec preference 1 g711ulaw

codec preference 2 g729r8

!

voice class codec 2

codec preference 1 g729r8

codec preference 2 g729br8

codec preference 3 g711ulaw



Dial Peer Cisco UBE uses dial-peers to route the call accordingly based on the digits dial-peer voice 200 voip description Outbound-from IP PBX to PSTN - WAN facing translation-profile outgoing e16four huntstop destination-pattern .T session protocol sipv2 session target sip-server session transport tcp voice-class codec 2 voice-class sip asserted-id pai voice-class sip profiles 100 voice-class sip options-keepalive voice-class sip bind control source-interface GigabitEthernet0/0/1 voice-class sip bind media source-interface GigabitEthernet0/0/1 dtmf-relay rtp-nte fax-relay ecm disable fax rate 14400 fax nsf 000000 fax protocol pass-through g711ulaw no vad L dial-peer voice 210 voip description outgoing call to intelepeer - LAN facing huntstop session protocol sipv2 session target sip-server incoming called-number .T voice-class codec 2

voice-class sip asserted-id pai



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 ecm disable fax rate 14400 fax nsf 000000 fax protocol pass-through g711ulaw no vad L dial-peer voice 500 voip description cube-dp incomming call from PSTN huntstop session protocol sipv2 session transport tcp incoming called-number +1949204.... voice-class codec 2 voice-class sip asserted-id pai 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 ecm disable fax rate 14400 fax nsf 000000 fax protocol pass-through g711ulaw no vad ! dial-peer voice 100 voip description Inbound-from PSTN to IP PBX - LAN facing translation-profile outgoing noe16four



huntstop destination-pattern +1949204.... session protocol sipv2 session target ipv4:10.80.18.3:5060 voice-class codec 2 voice-class sip asserted-id pai 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 ecm disable fax rate 14400 fax nsf 000000 fax protocol pass-through g711ulaw no vad 1



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 "9" 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 "9". A 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 IntelePeer, Caller dial 9 prefix followed by the 10 digit DID provided by IntelePeer, 9 was stripped and the ten-digit number was send to Cisco UBE, Cisco UBE translate the 10 digits extension number to its full ten-digits DID with E.164 format under Dial Peer 200 and send to IntelePeer network which will direct back to Cisco UBE and handled same as normal incoming PSTN call.



Figure 3: Inbound Voice Call



Figure 4: Outbound Fax Call



Figure 6: 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

User Access Verification

Username: cisco

Password:

IN_CUBE1#sh running

Building configuration...

Current configuration : 6624 bytes

!

! Last configuration change at 07:45:49 UTC Fri Apr 15 2016 by cisco

!

version 15.6

service timestamps debug datetime msec

```
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname IN_CUBE1
!
boot-start-marker
boot system bootflash:isr4300-universalk9.03.17.01.S.156-1.S1-std.SPA.bin
boot-end-marker
!
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
enable secret 5
enable password
!
no aaa new-model
!
!
!
!
l
1
!
```



! !

no ip domain lookup

ip domain name tekvizion.com

!
!
!
!
!
!
!
!
!
!
subscriber templating
!
!
!
multilink bundle-name authenticated
!
!
!
!
!
!
!
!
!
voice service voip

no ip address trusted authenticate address-hiding mode border-element license capacity 20 allow-connections sip to sip redundancy-group 1 fax protocol pass-through g711ulaw sip rel1xx supported "rel100" session refresh asserted-id pai privacy pstn early-offer forced midcall-signaling passthru g729 annexb-all ! voice class codec 1 codec preference 1 g711ulaw codec preference 2 g729r8 ! voice class codec 2 codec preference 1 g729r8 codec preference 2 g729br8 codec preference 3 g711ulaw ! ! voice class sip-profiles 100 request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:+1949204\1@\2" ! l

```
!
!
l
voice translation-rule 1
rule 1 /\(^.....$\)/ /+1\1/
!
voice translation-rule 2
rule 1 /\(^.....$\)/ /+1\1/
rule 2 /\(^.....$\)/ /+1\1/
rule 3 /\(^442037.....\)/ /+\1/
!
voice translation-rule 3
rule 1 /\+1\(.....\)/ /\1/
!
voice translation-rule 4
rule 1 /\+1\(.....\)/ /\1/
!
1
voice translation-profile e16four
translate calling 1
translate called 2
!
voice translation-profile noe16four
translate calling 3
translate called 4
!
1
1
license udi pid ISR4321/K9 sn FDO19220MSQ
```



```
!
spanning-tree extend system-id
!
username cisco privilege 15 password 0 tekV1z10n
!
redundancy
mode none
application redundancy
 group 1
 name voice-b2bhaIntelePeer
 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
!
l
T
I
l
vlan internal allocation policy ascending
!
track 1 interface GigabitEthernet0/0/0 line-protocol
!
track 2 interface GigabitEthernet0/0/1 line-protocol
!
L
translation-rule 1
```

! ! l I ! l ! 1 ! ! ! 1 1 l 1 ! I I I ! ! ! interface GigabitEthernet0/0/0 ip address 10.80.18.21 255.255.255.0 media-type rj45 negotiation auto redundancy rii 1 redundancy group 1 ip 10.80.18.20 exclusive !

interface GigabitEthernet0/0/1 ip address 192.65.XX.XXX 255.255.255.128 negotiation auto redundancy rii 2 redundancy group 1 ip 192.65.x.x exclusive l interface GigabitEthernet0/1/0 description CUBE HA MS5 3/0/37 ip address 10.89.20.9 255.255.255.0 negotiation auto l interface GigabitEthernet0 vrf forwarding Mgmt-intf no ip address negotiation auto ! interface Vlan1 no ip address shutdown ! ip forward-protocol nd no ip http server no ip http secure-server ip tftp source-interface GigabitEthernet0 ip route 0.0.0.0 0.0.0.0 192.65.XX.XXX ip route 10.64.0.0 255.255.0.0 10.80.18.1 ip route 10.80.0.0 255.255.0.0 10.80.18.1 ip route 172.16.24.0 255.255.248.0 10.80.18.1 ip route 172.16.31.0 255.255.255.0 10.80.18.1

! ! l T ! l control-plane ! ! ! ! ! 1 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 ! l ! L dial-peer voice 200 voip description Outbound-from IP PBX to PSTN - WAN facing translation-profile outgoing e16four huntstop

destination-pattern .T

session protocol sipv2

session target sip-server session transport tcp voice-class codec 2 voice-class sip asserted-id pai voice-class sip profiles 100 voice-class sip options-keepalive voice-class sip bind control source-interface GigabitEthernet0/0/1 voice-class sip bind media source-interface GigabitEthernet0/0/1 dtmf-relay rtp-nte fax-relay ecm disable fax rate 14400 fax nsf 000000 fax protocol pass-through g711ulaw no vad ! dial-peer voice 210 voip description outgoing call to intelepeer - LAN facing huntstop session protocol sipv2 session target sip-server incoming called-number .T voice-class codec 2 voice-class sip asserted-id pai 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 ecm disable fax rate 14400



fax nsf 000000 fax protocol pass-through g711ulaw no vad L dial-peer voice 500 voip description cube-dp incomming call from PSTN huntstop session protocol sipv2 session transport tcp incoming called-number +19492..... voice-class codec 2 voice-class sip asserted-id pai 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 ecm disable fax rate 14400 fax nsf 000000 fax protocol pass-through g711ulaw no vad ! dial-peer voice 100 voip description Inbound-from PSTN to IP PBX - LAN facing translation-profile outgoing noe16four huntstop destination-pattern +19492..... session protocol sipv2 session target ipv4:10.80.18.3:5060



```
voice-class codec 2
voice-class sip asserted-id pai
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 ecm disable
fax rate 14400
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
!
sip-ua
keepalive target ipv4:68.68.XXX.xx:5060
timers keepalive active 180
sip-server ipv4:68.68.XX.xx:5060
!
!
line con 0
stopbits 1
line aux 0
stopbits 1
line vty 04
exec-timeout 0 0
password xxxx
login local
!
!
End
```

Standby Cisco UBE **User Access Verification** cisco Password: IN_CUBE2#sh run Building configuration... Current configuration : 5630 bytes ! ! Last configuration change ! version 15.6 service timestamps debug datetime msec service timestamps log datetime msec no platform punt-keepalive disable-kernel-core ! hostname IN_CUBE2 ! boot-start-marker boot system bootflash:isr4300-universalk9.03.17.01.S.156-1.S1-std.SPA.bin boot-end-marker ! 1 vrf definition Mgmt-intf ! address-family ipv4 exit-address-family ! address-family ipv6 exit-address-family



```
!
enable secret 5 $1$o64N$3mgK4kNZKFwv3VUAouCDm1
enable password cisco
!
```

no aaa new-model ! ! ! ! ! ! ! ! !

no ip domain lookup

ip domain name tekvizion.com

```
--More-- !
!
!
!
!
!
!
!
!
subscriber templating
!
!
```

! multilink bundle-name authenticated ! ! ! ! ! 1 ! ! l voice service voip no ip address trusted authenticate address-hiding mode border-element license capacity 20 allow-connections sip to sip redundancy-group 1 fax protocol pass-through g711ulaw sip rel1xx supported "rel100" session refresh asserted-id pai privacy pstn early-offer forced midcall-signaling passthru g729 annexb-all ! L voice class codec 1

```
codec preference 1 g711ulaw
codec preference 2 g729r8
!
voice class codec 2
codec preference 1 g729r8
codec preference 2 g729br8
codec preference 3 g711ulaw
!
!
!
!
l
voice class sip-profiles 100
request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:+1949204\1@\2"
!
!
voice translation-rule 1
rule 1 /\(^.....$\)/ /+1\1/
!
voice translation-rule 2
rule 1 /\(^.....$\)/ /+1\1/
rule 2 /\(^.....$\)/ /+1\1/
rule 3 /\(^442037.....\)/ /+\1/
!
voice translation-rule 3
rule 1 /\+1\(.....\)/ /\1/
!
voice translation-rule 4
rule 1 /\+1\(.....\)/ /1/!
```



```
!
voice translation-profile e16four
translate calling 1
translate called 2
!
voice translation-profile noe16four
translate calling 3
translate called 4
!
!
l
license udi pid ISR4321/K9 sn FDO19220MQ9
!
spanning-tree extend system-id
!
username cisco privilege 15 password 0
!
redundancy
mode none
application redundancy
 group 1
 name voice-b2bhaIntelePeer
 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
!
!
!
1
1
!
!
ļ
!
1
!
!
!
!
!
!
!
1
1
!
!
```

```
!
interface GigabitEthernet0/0/0
ip address 10.80.18.22 255.255.255.0
media-type rj45
negotiation auto
redundancy rii 1
redundancy group 1 ip 10.80.18.20 exclusive
L
interface GigabitEthernet0/0/1
ip address 192.65.XX.XXX 255.255.255.128
negotiation auto
redundancy rii 2
redundancy group 1 ip 192.65.XX.XXX exclusive
l
interface GigabitEthernet0/1/0
description CUBE HA MS5 3/0/38
ip address 10.89.20.10 255.255.255.0
negotiation auto
l
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
shutdown
negotiation auto
!
interface Vlan1
no ip address
shutdown
l
```

```
ip forward-protocol nd
no ip http server
no ip http secure-server
ip tftp source-interface GigabitEthernet0
ip route 0.0.0.0 0.0.0.0 192.65.XX.XXX
ip route 10.64.0.0 255.255.0.0 10.80.18.1
ip route 10.80.0.0 255.255.0.0 10.80.18.1
ip route 172.16.24.0 255.255.248.0 10.80.18.1
!
!
I
I
I
1
control-plane
!
!
!
!
--More--
             !
!
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
!
!
```

! l dial-peer voice 200 voip description Outbound-from IP PBX to PSTN - WAN facing translation-profile outgoing e16four huntstop destination-pattern .T session protocol sipv2 session target sip-server session transport tcp voice-class codec 2 voice-class sip asserted-id pai voice-class sip profiles 100 voice-class sip options-keepalive voice-class sip bind control source-interface GigabitEthernet0/0/1 voice-class sip bind media source-interface GigabitEthernet0/0/1 dtmf-relay rtp-nte fax-relay ecm disable fax rate 14400 fax nsf 000000 fax protocol pass-through g711ulaw no vad L dial-peer voice 210 voip description outgoing call to intelepeer - LAN facing huntstop session protocol sipv2 session target sip-server incoming called-number .T



voice-class codec 2 voice-class sip asserted-id pai 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 ecm disable fax rate 14400 fax nsf 000000 fax protocol pass-through g711ulaw no vad l dial-peer voice 500 voip description cube-dp incomming call from PSTN huntstop session protocol sipv2 session transport tcp incoming called-number +19492..... voice-class codec 2 voice-class sip asserted-id pai 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 ecm disable fax rate 14400 fax nsf 000000 fax protocol pass-through g711ulaw no vad



! dial-peer voice 100 voip description Inbound-from PSTN to IP PBX - LAN facing translation-profile outgoing noe16four huntstop destination-pattern +19492..... session protocol sipv2 session target ipv4:10.80.18.3:5060 voice-class codec 2 voice-class sip asserted-id pai 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 ecm disable fax rate 14400 fax nsf 000000 fax protocol pass-through g711ulaw no vad sip-ua keepalive target ipv4:68.68.123.174:5060 timers keepalive active 180 sip-server ipv4:68.68.123.174:5060 ! ! line con 0 stopbits 1 line aux 0 stopbits 1



line vty 0 4 password xxxx login local !

end

IN_CUBE2#

Configuring Cisco Unified Communications Manager

Cisco UCM Version



Figure 7: Cisco UCM Version

Cisco Call Manager Service Parameters

Navigation Path: System > Service Parameters

Select Server* = Clus28Sub1--CUCM Voice/Video (Active)

Select Service*= Cisco CallManager (Active)

All other fields are set to default values

System 👻 Ca	Il Routing 🔻 Media Resources 👻 Advanced	Features - Device - Application - User Manager	nent 🔻 Help 🔻
Service Para	meter Configuration		Related Links: Parameters for All Servers 🗸 Go
🔚 Save 🧬	🔊 Set to Default 🔍 Advanced		
Status			^
i Status:	Ready		
Select Serv	er and Service		
Server*	Clus28PubCUCM Voice/Video (Active) 🗸	
Service*	Cisco CallManager (Active)	~	
All parameter	rs apply only to the current server excep	parameters that are in the cluster-wide group(s).	
Cisco CallMa	anager (Active) Parameters on serve	er Clus28PubCUCM Voice/Video (Active)—	
			<u>ş</u>
Parameter Na	ame	Parameter Value	Suggested Value
Call Thrott	tling —		
Code Yellow	v Entry Latency *	20	20
Code Yellow	<pre>v Exit Latency Calculation *</pre>	40	40
Code Yellow	v Duration_*	5	✓ 5
Max Events	Allowed *	2000	2000
System Thr	ottle Sample Size *	10	10

Figure 8: Service Parameters

© 2016 Cisco Systems, Inc. All rights reserved. Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com Page **37** of **53**

Offnet Calls via IntelePeer SIP Trunk

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

SIP Trunk Security Profile

Navigation Path: System > Security > SIP Trunk Security Profile

Name*= Intelepeer Non Secure SIP Trunk Profile

Description = non Secure SIP Trunk Profile authenticated by null String

System	s ▼ Advanced Features ▼ Device ▼ Application ▼ User Management ▼	Bulk Administration - H
SIP Trunk Security Profile Configura	tion	Relate
🔚 Save 🗙 Delete 🗋 Copy 嗋	Reset 🥒 Apply Config 🕂 Add New	
–SIP Trunk Security Profile Informati	on	
Name*	Intelepeer 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 T	
Outgoing Transport Type	UDP	
Enable Digest Authentication		
Nonce Validity Time (mins)*	600	
X.509 Subject Name		
Incoming Port*	5060	
Enable Application level authorization	1	
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 9: SIP Trunk Security Profile

Explanation

Parameter	Value	Description
Incoming Transport Type	TCP + UDP	
Outgoing Transport Type	UDP	SIP trunks to IntelePeer 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 Path: Device > Device Settings > SIP Profile

Name*= Intelepeer SIP Profile

Description = Default SIP Profile

System ▼ Call Routing ▼ Media Resources ▼	Advanced Features 👻 De	evice 👻 Application 👻 Us	er Management 👻	Bulk Administration 👻	Help 🗸
SIP Profile Configuration				Rel	ated Links: Back To Find/List
🔚 Save 🗶 Delete 🗋 Copy 省 Rese	et 🧷 Apply Config 🕂	Add New			
-SIP Profile Information					
Name*	Intelepeer SIP Profile				
Description	Default SIP Profile				
Default MTP Telephony Event Payload Type*	101				
Early Offer for G.Clear Calls*	Disabled		T		
User-Agent and Server header information*	Send Unified CM Version	Information as User-Ager	•		
Version in User Agent and Server Header*	Major And Minor	Information as osci Ager	T		
Dial String Interpretation*	Phone number consists o	f characters 0-9 * # an	•		
Confidential Access Level Headers*	Disabled		• •		
Redirect by Application	Disabled				
Disable Early Media on 180					
Outpoine T 28 INV/ITE include audio mline					
Ose Fully Qualified Domain Name in SIP F	Requests				
Assured Services SIP conformance SDP Information					
SDP Session-level Bandwidth Modifier for E	arly Offer and Re-invites*	TIAS and AS		•	
SDP Transparency Profile		Pass all unknown SDP at	tributes	T	
Accept Audio Codec Preferences in Receive	d Offer*	Default		•	
Require SDP Inactive Exchange for Mid-	Call Media Change				
Allow RR/RS bandwidth modifier (RFC 3	556)				
-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	Common Port Range	for Audio and Video			
	Separate Port Ranges	s for Audio and Video			
Start Media Port*	16384				
Stop Media Port*	32766				

Figure 10: SIP Profile

DSCP for Audio Calls	Use System Default	V						
DSCP for Video Calls	Use System Default	T						
DSCP for Audio Portion of Video Calls	Use System Default	T						
DSCP for TelePresence Calls	Use System Default	Y						
DSCP for Audio Portion of TelePresence Calls	Use System Default	T						
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	T	d					
DTMF DB Level*	Nominal	Y						
Call Hold Ring Back*	Off	V						
Anonymous Call Block*	Off	V						
Caller ID Blocking*	Off	T						
Do Not Disturb Control*	User	Y						
Telnet Level for 7940 and 7960*	Disabled	T						
Resource Priority Namespace	< None >	V						
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) st	15000]					
Call Forward URI*	x-cisco-serviceuri-cfwdall							
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial							
Conference Join Enabled								
RFC 2543 Hold								
🗹 Semi Attended Transfer								
Enable VAD								
Stutter Message Waiting								
MLPP User Authorization								
Normalization Script								
Normalization Script < None >	•							
Enable Trace								
Parameter Name	Paramete	r Value						
1			± =					
- Incoming Requests EDOM UDI Catting								
Caller ID DN								
Caller Name								

Figure 11: SIP Profile (Cont.)



-Trunk Specific Configuration								
Reroute Incoming Request to new Trunk based on $\!\!\!*$	Never T							
Resource Priority Namespace List	<pre>None > </pre>							
SIP Rel1XX Options*	Send PRACK if 1xx	end PRACK if 1xx Contains SDP						
Video Call Traffic Class*	Mixed	lixed T						
Calling Line Identification Presentation*	Default ▼							
Session Refresh Method*	Invite		T					
Early Offer support for voice and video calls st	Disabled (Default v	/alue)	T					
Enable ANAT								
Deliver Conference Bridge Identifier								
\square Allow Passthrough of Configured Line Device Cal	ler Information							
Reject Anonymous Incoming Calls								
Reject Anonymous Outgoing Calls								
Send ILS Learned Destination Route String								
SIP OPTIONS Ping								
Enable OPTIONS Ping to monitor destination sta	atus for Trunks with	Service Type "None (Default)"						
Ping Interval for In-service and Partially In-service	• Trunks (seconds)*	30						
Ping Interval for Out-of-service Trunks (seconds)*		120						
Ping Retry Timer (milliseconds)*		500						
Ping Retry Count*		6						
⊂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								
Save Delete Copy Reset Apply Config Add New								

Figure 12: SIP Profile (Cont.)

Explanation

Parameter	Value	Description
Default MTP Telephony Event	101	RFC2833 DTMF payload type
Payload Type		
SIP Rel1XX Options	Send PRACK for	Enable Provisional Acknowledgements (Reliable 100
	1xx Messages	messages)
Ping Interval for In-service and	30	OPTIONS message parameters- interval time- This is
Partially In-service Trunks		used in this example
(seconds)		
Ping Interval for Out-of-service	120	OPTIONS message parameters- interval time
Trunks (seconds)		



SIP Trunk Configuration

Create SIP trunks to Cisco UBE

Navigation Path: Device > Trunk

System 👻 🛛	Call Routing 🔻 Media Resources 🔻 🖌	Advanced Features 👻 Device 👻	Applicatio	on 🔻 User Ma	inagement	 Bulk Adr 	ninistration		-			
Find and Li	ist Trunks											
Add Net	w 🔛 Select All 🔛 Clear All 🛔	Delete Selected 🏻 🍟 Reset Se	elected									
Trunks	(1 - 4 of 4)									Ro	ws per Pa <u>c</u>	je 50 ∨
Find Trunks	where Device Name	✓ begins with ✓		Find	d Clea	r Filter	÷ –					
		Select iter	n or enter	r search text	¥							
	Name [*]	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type	SIP Trunk Status	SIP Trunk Duration	SIP Trunk Security Profile
	Intelepeer	SIP trunk to Intelepeer CUBE		<u>G711 pool</u>	<u>*6.!</u>				SIP Trunk	Unknown - OPTIONS Ping not enabled		Intelepeer Non Secure SIP Trunk Profile
	Intelepeer	SIP trunk to Intelepeer CUBE		<u>G711 pool</u>	<u>9.1</u>				SIP Trunk	Unknown - OPTIONS Ping not enabled		Intelepeer Non Secure SIP Trunk Profile
	<u>SIP trunk to fax gateway</u>	SIP_trunk_to_fax_gateway		<u>G711 pool</u>	<u>5421</u>				SIP Trunk	Full Service	Time In Full Service: 2 days 2 hours 14 minutes	<u>Intelepeer</u> <u>Non</u> <u>Secure</u> <u>SIP Trunk</u> <u>Profile</u>
	<u>sip Unity</u>			<u>G711 pool</u>	<u>2900</u>				SIP Trunk	Full Service	Time In Full Service: 2 days 3 hours	<u>unity</u> profile

Figure 13: SIP Trunks List

System ▼ Call Routing ▼ Media Resources ▼ Advanced Features ▼ I	Device ▼ Application ▼ User Management ▼ Bulk Administration ▼	Help 🔻
Trunk Configuration		Related Links: Back To Find/List 🗸
🔚 Save 🗶 Delete 👇 Reset 🟳 Add New		
- Device Information		
Products	SIP Truck	
Device Protocol:	SIP	
Trunk Service Type	None(Default)	_
Device Name*	Intelepeer	
Description	SIP trunk to Intelepeer CUBE	
Device Pool*	G711_pool v	
Common Device Configuration	< None > V	-
Call Classification*	Use System Default	_
Media Resource Group List	MRGL_Default	
Location*	Hub_None v	
AAR Group	< None > V	
Tunneled Protocol*	None v	
QSIG Variant*	No Changes 🗸	
ASN.1 ROSE OID Encoding*	No Changes 🗸	
Packet Capture Mode*	None v	
Packet Capture Duration	0	
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 information.	to be configured in the network to provide end to end security. F	ailure to do so will expose keys and other
Consider Traffic on This Trunk Secure*	When using both sRTP and TLS	
Route Class Signaling Enabled*	Default	
Use Trusted Relay Point*	Default v	
PSTN Access		
Run On All Active Unified CM Nodes		
- Intercompany Media Engine (IME)		
E.164 Transformation Profile < None >	~	
-MLPP and Confidential Access Level Information		
MLPP Domain < None >	~	
Confidential Access Mode < None >	~	
Confidential Access Level < None >		
< None >	Ŧ	

Figure 14: SIP Trunk to Cisco UBE (Cont.)

Call Routing Information								
Remote-Party-Id								
Asserted-Identity								
Asserted-Type* Default			~					
SIP Privacy* Default			~					
-Inbound Calls								
Significant Digits*	4			~				
Connected Line ID Present	ation* Defai	ult		~				
Connected Name Presentat	tion* Defai	ult		~				
Calling Search Space	< No	ne >		~				
AAR Calling Search Space	< No	ne >		~				
Prefix DN								
Redirecting Diversion H	leader Delive	ry - Inbound						
_Incoming Calling Party	Settings —							
If the administrator sets	s the prefix to	o Default this indicate	es call processing will (use prefix	at the next level setting (Device	ePool/Service Parameter). Otherwise, the value	
configured is used as th	e prefix unle	ss the field is empty	in which case there is	no prefix a	assigned.			
			Clear Prefix S	Settings	Default Prefix Settings			
Number Type		Prefix	Strip Digits		Calling Search Sp	ace	Use Device Pool CSS	s
Incoming Number	Default		0	< None	>	~		
Incoming Called Party	Settings—							
If the administrator sets	s the prefix to	o Default this indicat	tes call processing will	use prefix	at the next level setting (Devi	cePool/Service Paramete	r). Otherwise, the value	1
			Clear Prefix	Settings	Default Prefix Settings			
Number Trees		Des fin	Chain Dinite	Settings	Colling Country	l 		
Incoming Number	Default	Prefix	Strip Digits		Calling Search S	pace		,55
incoming names.	Derault		0		6 ×	*	V	
- Connected Party Settin								
Connected Party Transfor	mation CSS							
✓ Use Device Pool Conr	ected Party	Transformation CSS			•			
Outbound Calls								
Called Party Transformatio	n CSS	< None >			¥			
✓ Use Device Pool Called	Party Transf	ormation CSS						
Calling Party Transformation	on CSS	< None >			¥			
Use Device Pool Calling	Party Transf	formation CSS						
Calling Party Selection*		Originator			×			
Calling Line ID Presentatio	n*	Default			×			
Calling Name Presentation	*	Default			¥			
Calling and Connected Par	ty Info Form	at* Deliver DN only	y in connected party		~			
Redirecting Diversion H	leader Delive	ery - Outbound						
Redirecting Party Transform	mation CSS	< None >			v			
Use Device Pool Redire	cting Party T	ransformation CSS						
Caller Information								
Caller ID DN								
Caller Name								
Maintain Original Call	er ID DN and	d Caller Name in Ide	ntity Headers					
SIP Information								
Destination								
Destination Address is	an SRV							
	estination A	ddress		Destinatio	on Address IPv6	Destination Por	t Status	-
1* 10.80.18.20						5060	down	

Figure 15: SIP Trunk to Cisco UBE (Cont.)

1	1]1,	1		1
	C	IS	C	0	TM

MTP Preferred Originating Codec*	711ulaw	v			
BLF Presence Group*	Standard Presence group	v			
SIP Trunk Security Profile*	Intelepeer 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	✓			
Normalization Script					
Normalization Script < None >					
Enable Trace					
Parameter Nam	e P	arameter Value			
1					
Recording Information					
None					
O This trunk connects to a recording-er	nabled gateway				
O This trunk connects to other clusters	O This trunk connects to other clusters with recording-enabled gateways				
-Geolocation Configuration					
Geolocation <pre></pre> <pre></pre> <pre></pre>					
Geolocation Filter < None > v					
Send Geolocation Information					
Save Delete Reset Add Ner	N				
Figure 16: SIP Trunk to Cisco UBE (Cont.)					

Explanation

Parameter	Value	Description
Device Name	Intelepeer	Name for the trunk
Device Pool	G711_pool	Default Device Pool is used for this trunk
Media Resource Group List	MRGL_Default	MRG with resources: ANN, CFB, MOH and MTP
Significant Digits	4	4 digits Extension for all CPE phones
Destination Address	10.80.18.20	IP address of the Cisco UBE Virtual LAN
SIP Trunk Security Profile	Intelepeer Non Secure SIP	SIP Trunk Security Profile configured earlier
	Trunk Profile	
SIP Profile	Intelepeer Standard SIP Profile	SIP Profile configured earlier



Dial Plan Route Pattern Configuration

Navigation Path: Call Routing > Route/Hunt > Route Pattern

Route patterns are configured as below:

- Cisco IP phone dial "9".10 digits number to access PSTN via Cisco UBE
 - "9" is removed before sending to Cisco UBE
- For FAX call, Access Code "9"+ 10 digits number is used at Cisco Fax gateway
 - o "9" is removed at Cisco UCM
 - The rest of the number is sent to Cisco UBE to IntelePeer network
- Incoming fax call to xxxx will be sent to the Cisco Fax gateway

սիսի	Cisco Un	ified CM Administratio	n	Navig	ation Cisco Unified CM Adn	ninistration 🗸 🗸
cisco	For Cisco Un	ified Communications Solutions		admini	strator Search Docume	ntation About
System 🔻	Call Routing 🔻 🛛	Media Resources 🔻 Advanced Features 🔻	Device 🔻 Applic	ation 🔻 User Manager	ment 👻 Bulk Administration 👻	Help 🔻
Find and	List Route Patt	erns				
🕂 Add	New Select A	All 🔛 Clear All 🙀 Delete Selected				
Status-						
(i) 4 re	ecords found					
0						
Route	Patterns (1 - 4	of 4)				Rows per Page 50
Find Rout	te Patterns where	Pattern	✓ begins with ✓		Find Clear Filte	r 수 =
	Pattern 📩	Description	Partition	Route Filter	Associated De	vice
	*6.!	PSTN call restricted			Intelepeer	ß
	<u>2900</u>	VM pilot Number			sip Unity	ß
	<u>5421</u>	Fax Gw			SIP trunk to fax gateway	. Б
	<u>9.1</u>	PSTN calling			Intelepeer	ß
Add N	ew Select All	Clear All Delete Selected				

Figure 17: Route Patterns List

System - Call Routing - Media F	Resources 🔻 A	Ivanced Features - Device - Application - User Manageme	nt 🔻 Bulk Administration 👻 Help 👻	
Route Pattern Configuration			Related Links: Back To Find/List 🗸	
🔚 Save 🗶 Delete 🗋 Cop	oy 🕂 Add Nev	v		
Pattern Definition				
Route Pattern*		9.!		
Route Partition		< None >		
Description		PSTN calling		
Numbering Plan		Net Selected		
Route Filter				
MI PP Precedence*		C None > V		
		Verault 🗸		
Apply Call Blocking Percenta	age			
Resource Priority Namespace N	etwork Domain	< None > V		
Route Class		Default v		
Gateway/Route List*		Intelepeer v	(Edit)	
Route Option		Route this pattern		
		O Block this pattern No Error 🗸		
Call Classification*	OffNet	✓		
External Call Control Profile	< None >	~		
Allow Device Override 🗹 Pr	rovide Outside E	ial Tone Allow Overlap Sending Urgent Priority		
Require Forced Authorization	n Code			
Authorization Level*	0			
Bequire Client Matter Code				
Calling Party Transformation	ns			
Use Calling Party's External	Phone Number	Mask		
Calling Party Transform Mask				
Prefix Digits (Outgoing Calls)				
Calling Line ID Presentation*	Default			
Calling Name Presentation*				
Calling Party Number Type*	Circo CallManager			
Calling Party Numbering Plan*				
video call bridger v				
Connected Party Transform	ations			
Connected Line ID Presentation	* Default	~		
Connected Name Presentation*	Default			
Called Party Transformation	15			
Discard Digits	PreDot	✓		
Called Party Transform Mask				
Prefix Digits (Outgoing Calls)				
Called Party Number Type* Cisco CallManager				
Called Party Numbering Plan* Cisco CallManager				
		•		
SDN Network-Specific Faci	ilities Informa	tion Element		
Network Service Protocol	Not Selected	¥		
Carrier Identification Code				
Network Service		Service Parameter Name	Service Parameter Value	
Not Selected		▼ < Not Exist >		
Save Delete Copy	Add New			

Figure 16: Route Pattern for Voice

🔚 Save 🗶 Delete 🗈 Copy 🕂 Add New					
-Pattern Definition					
Route Pattern*		5421			
Route Partition		< None > V			
Description		Fax Gw			
Numbering Plan		Not Selected V			
Route Filter		< None > V			
MLPP Precedence*		Default v			
Apply Call Blocking Percent	age				
Resource Priority Namespace N	Network Domain	< None >			
Route Class*		Default v	_		
Gateway/Route List*		SIP_trunk_to_fax_gateway v	(<u>Edit</u>)		
Route Option		Route this pattern			
		O Block this pattern No Error 🗸			
Call Classification*	OffNet				
External Call Control Profile	< None >				
Allow Device Override 🗹 P	Provide Outside [ial Tone 🗌 Allow Overlap Sending 🗌 Urgent Priority			
Require Client Matter Code					
-Calling Party Transformatio	ons				
Use Calling Party's External	Phone Number	lask			
Calling Party Transform Mask					
Prefix Digits (Outgoing Calls)					
Calling Line ID Presentation*	Default				
Calling Name Presentation*	Default	v			
Calling Party Number Type*	Cisco CallMana	Cisco CallManager v			
Calling Party Numbering Plan*	Cisco CallManager v				
- Connected Party Transformations					
Connected Line ID Presentation	n* Default	✓			
Connected Name Presentation	* Default	×			
-Called Party Transformatio	ns				
Discard Digits	< None >	✓			
Called Party Transform Mask	form Mask				
Prefix Digits (Outgoing Calls)					
Called Party Number Type* Cisco CallManager					
- ISDN Network-Specific Facilities Information Element					
Network Service Protocol Not Selected V					
Carrier Identification Code					
Network Service		Service Parameter Name	Service Parameter Value		
Not Selected		< Not Exist >			
Save Delete Conv	Add New				

Figure 19: Route Pattern for Voice

-Pattern Definition				
Route Pattern*		*6.!		
Route Partition		< None >	~	1
Description		PSTN call restricted		
Numbering Plan		Not Selected	~	
Route Filter		< None >	~	
MLPP Precedence*		Default	~	
Apply Call Blocking Percent	tage			
Resource Priority Namespace N	Network Domain	< None >	~	
Route Class*		Default	~	
Gateway/Route List*		Intelepeer	v (<u>(E dit</u>)
Route Option		Route this pattern		-
		O Block this pattern No Error	~	
Call Classification*	OffNet			
External Call Control Profile	< None >			
Allow Device Override 🗹 F	Provide Outside [Dial Tone Allow Overlap Sending Urgent Prio	rity	
Require Forced Authorizatio	on Code			
Authorization Level*	0			
Require Client Matter Code	L			
-Calling Party Transformation	15			
Calling Party Transform Mask	Phone Number N	lask		
Prefix Digits (Outgoing Calls)				
Colling Line ID Presentation*	Restricted			
Calling Name Presentation*	estricted v			
Calling Name Presentation	Cisco CallManager			
Calling Party Numbering Plan*	Cisco CallManager Y			
Calling Party Numbering Plan Cisco CallManager				
Connected Party Transforma	ations			
Connected Line ID Presentation* Default				
Connected Name Presentation*	Default	×		
-Called Party Transformation	s			
Discard Digits	PreDot	~		
Called Party Transform Mask				
Prefix Digits (Outgoing Calls)	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	lot Selected			
Carrier Identification Code				
Natwork Sanuice December Value				
Not Selected	~	< Not Exist >		
Save Delete Conv	Add New			

Figure 21: Route Pattern for Voice – Cont.

🔚 Save 🗙 Delete 🗈 Copy 🕂 Add New				
Pattern Definition				
Route Pattern*	2900]		
Route Partition	< None > V			
Description	VM pilot Number			
Numbering Plan	Not Selected V			
Route Filter	< None >			
MLPP Precedence*	Default			
Apply Call Blocking Percentage				
Resource Priority Namespace Network Domain	< None >			
Route Class*	Default			
Gateway/Route List*		(Edit)		
Route Option	Route this pattern	·		
Call Classification*				
External Call Control Brofile	↓			
	Dial Tone 🗀 Allow Overlap Sending 🗀 Urgent Priority			
Require Forced Authorization Code	1			
Calling Party Transformations				
Use Calling Party's External Phone Number	Mask			
Calling Party Transform Mask				
Prefix Digits (Outgoing Calls)				
Calling Line ID Presentation* Default	Default			
Calling Name Presentation* Default	Default			
Calling Party Number Type* Cisco CallMan	Cisco CallManager V			
Calling Party Numbering Plan*				
- Connected Party Transformations				
Connected Line ID Presentation* Default				
Connected Name Presentation* Default	v			
-Called Party Transformations				
Discard Digits < None >				
Called Party Transform Mask				
Prefix Digits (Outgoing Calls)				
Called Party Number Type* Cisco CallMana	ager V			
Called Party Numbering Plan* Cisco CallManager				
-ISDN Network-Specific Facilities Informa	tion Element			
Network Service Protocol Not Selected	~			
Carrier Identification Code				
Network Service	Service Parameter Name	Service Parameter Value		
Not Selected	Not Exist >			
	Levies -			
Save Delete Copy Add New				

Figure 22: Route Pattern for Voice



Explanation

Setting	Value	Description
Route Pattern	9.! For Voice call, 5421 for fax call,*6.! For restricted	Specify appropriate Route
	calls,2900 for Voice mail pilot number	Pattern
Gateway/Route	Routed the calls to the appropriate trunk and	SIP Trunk name configured
List	gateways	earlier
Discard Digits	PreDot for Route Pattern 9.!,*6.! .	Specifies how to modify digit
		before they are sending to
		IntelePeer network

Acronyms

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



Important Information

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

Appendix A: Test Results



ılıılı cısco.

Corporate Headquarters Cisco Systems, Inc. 170 West Tasman Drive San Jose, CA 95134-1706 USA www.cisco.com Tel: 408 526-4000 800 553-NETS (6387) Fax: 408 526-4100 European Headquarters CiscoSystems International BV Haarlerbergpark Haarlerbergweg 13-19 1101 CH Amsterdam The Netherlands www-europe.cisco.com Tel: 31 0 20 357 1000 Fax: 31 0 20 357 1100 Americas Headquarters Cisco Systems, Inc. 170 West Tasman Drive San Jose, CA 95134-1706 USA www.cisco.com Tel: 408 526-7660 Fax: 408 527-0883 AsiaPacific Headquarters Cisco Systems, Inc. Capital Tower 168 Robinson Road #22-01 to #29-01 Singapore 068912 www.cisco.com Tel: +65 317 7777 Fax: +65 317 7799

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

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

© 2016 Cisco Systems, Inc. All rights reserved.

CCENT, Cisco Lumin, Cisco Nexus, the Cisco logo and the Cisco Square Bridge logo are trademarks of Cisco Systems, Inc.; Changing the Way We Work, Live, Play, and Learn is a service mark of Cisco Systems, Inc.; and Access Registrar, Aironet, BPX, Catalyst, CCDA, CCDP, CCVP, CCIE, CCIP, CCNA, CCNP, CCSP, Cisco, the Cisco Certified Internetwork Expert logo, Cisco IOS, Cisco Press, Cisco Systems, Cisco Systems Capital, the Cisco Systems logo, Cisco Unity, EtherFast, EtherSwitch, Fast Step, Follow Me Browsing, FormShare, GigaDrive, HomeLink, Internet Quotient, IOS, iPhone, iQ Expertise, the iQ logo, iQ Net Readiness Scorecard, iQuick Study, LightStream, Linksys, Meeting Place, MGX, Networking Academy, Network Registrar, Packet, PIX, ProConnect, ScriptShare, SMARTnet, StackWise, The Fastest Way to Increase Your Internet Quotient, and TransPath are registered trademarks of Cisco Systems, Inc. and/or its affiliates in the United States and certain other countries.

All other trademarks mentioned in this document or Website are the property of their respective owners. The use of the word partner does not imply a partnership relationship between Cisco and any other company. (0705R) Printed in the USA