



# Connecting Cisco Unified Communication Manager [v11.5.1] to British Telecom Global SIP Trunking Service via Cisco Unified Border Element v12.0 [IOS-XE 16.06.01]

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## Introduction

Service Providers today, such as British Telecom, 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.

British Telecom 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(Cisco UCM) and British Telecom network, Cisco Unified Border Element (Cisco UBE) ISR 4321/K9 running IOS-XE 16.6.1 can be used. The Cisco Unified Border Element 12.0.0 provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 11.5.1 connected to British Telecom IP network.

This document assumes the reader is knowledgeable with the terminology and configuration of Cisco Unified Communications Manager. Only configuration settings specifically required for British Telecom interoperability are presented. Feature configuration and most importantly the dial plan are customer specific and need individual approach.

- This application note describes how to configure Cisco UCM 11.5.1, and Cisco UBE on ISR 4321/K9 [IOS-XE 16.6.1] for connectivity to British Telecom SIP Trunking service. The deployment model covered in this application note is CPE (Cisco UCM) to PSTN (British Telecom) via Cisco UBE v12.0 [IOS-XE 16.6.1]
- Testing was performed in accordance to British Telecom 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 High Availability.
- The Cisco UBE configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between British Telecom 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 British Telecom SIP Trunking network.



## Network Topology

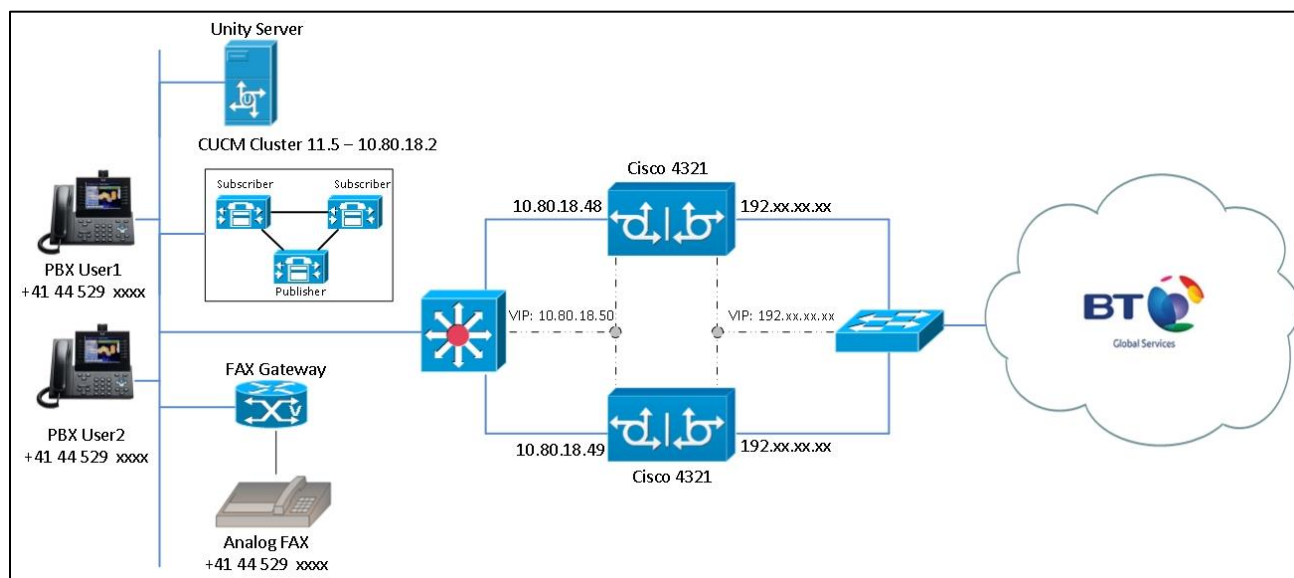


Figure 1 Network Topology

- The network topology includes the Cisco UCM Cluster, Unity Voicemail system, Cisco Fax gateway and 2 Cisco Endpoints. Cisco UCM has a trunk configured to Cisco UBE's Virtual IP Address. British Telecom was used as the service provider with SIP trunk to the Cisco UBE using the WAN Virtual IP Address.
- 2 Cisco Unified Border Elements are used here for High Availability.
- SIP Trunk transport type used between Cisco Unified Border Element and Cisco UCM is UDP and to British Telecom is UDP.

## Cisco UCM and Cisco UBE Settings:

Setting	Value
Transport from Cisco UBE to Cisco UCM	UDP with RTP
Transport from Cisco UBE to British Telecom	UDP with RTP
Voice Mail Support	YES
Session Refresh	YES
Early Media support with PRACK	YES



## System Components

### Hardware Requirements

- Cisco UBE on Cisco ISR 4321 router
- CUCM cluster on UCS C240, 1 Publisher node and 2 Subscriber nodes
- Cisco 2851 with FXS ports and Analog Fax machine
- Generic Cisco IP-Phones

### Software Requirements

- CUBE-Version: 12.0 running IOS-XE 16.6.1
- CUCM UCOS 11.5.1.12900-21 for 1 Publisher and 2 Subscriber
- Cisco IOS v15.1 for the fax gateway

## Features

### Features Supported

- Incoming and outgoing off-net calls using G711ulaw and G729 voice codecs
- International Calls and digit manipulations
- Call Conference
- CPE Voice Mail support
- Call hold & Resume with and without MoH
- Unattended and Attended Call transfer
- Call forward (all, busy and no answer)
- DTMF (RFC2833)
- Fax (T38 and G711 Pass-through)
- IP-PBX Calling number privacy
- High Availability

### Features Not Supported

- Cisco UCM does not support Blind Call transfer
- In HA Redundancy mode, the Primary Cisco UBE will not take over the Primary/Active role after a reboot/network outage
- British Telecom does not support Fax at Super G3 Speed



## Caveats

- Caller ID is not updated on attended and unattended transfer scenarios.
- Only one IP PBX used for the testing.
- The Cisco UBE HA tested is layer 2 box to box Cisco UBE redundancy.
- British Telecom does not support SG3 Fax. Fax test cases are executed only on G3. Maximum bit rate supported by British Telecom for T38 fax is 9600bps.
- Loopback fax over G711 Pass-Through was unsuccessful. Re-Invite for pass-through from the originating fax gateway is dropped in the BT network.
- British Telecom supports only E164 format for International calls.





## Configuration

### Configuring Cisco Unified Border Element

#### Network Interface

The IP address used are for illustration only, the actual IP address can vary. The Active/Standby pair share the same virtual IP address and continually exchange status messages.

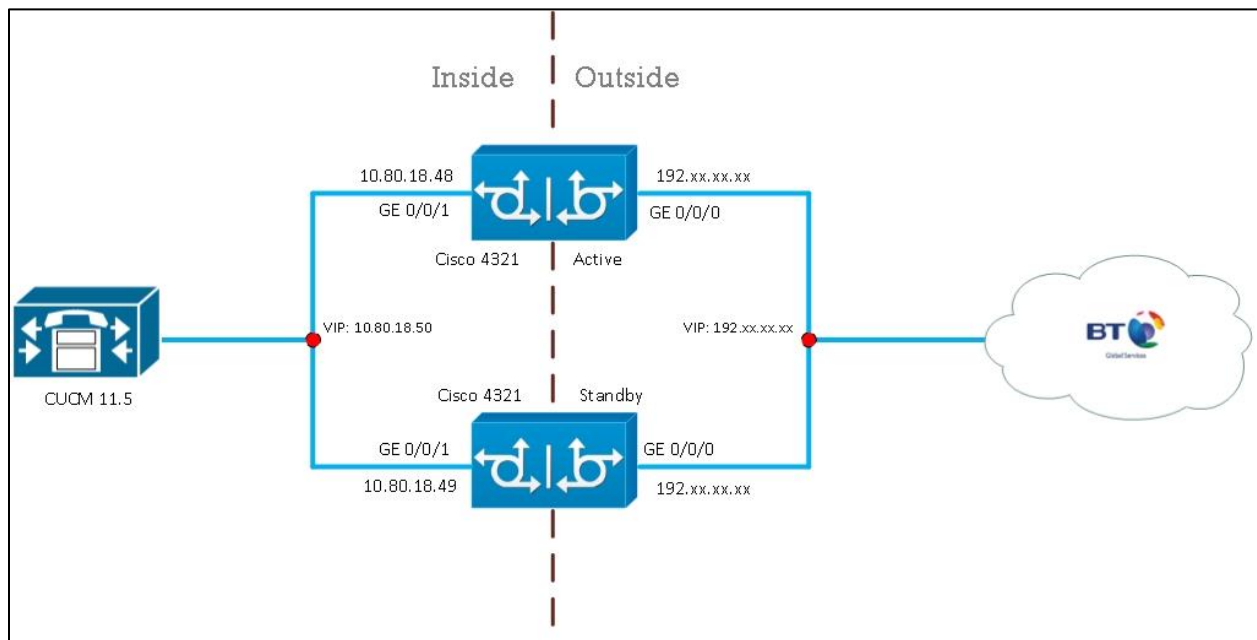


Figure 2 High Availability topology



#### Cisco UBE 1:

```
interface GigabitEthernet0/0/0
  description BT WAN Interface
  ip address 192.XX.XX.XX 255.255.255.128
  media-type rj45
  negotiation auto
  redundancy rii 15
  redundancy group 2 ip 192.XX.XX.XX exclusive
!
interface GigabitEthernet0/0/1
  description BT LAN Interface
  ip address 10.80.18.48 255.255.255.0
  negotiation auto
  redundancy rii 16
  redundancy group 2 ip 10.80.18.50 exclusive
!
interface GigabitEthernet0/1/0
  description BT CubeHA Interface
  ip address 10.70.50.100 255.255.255.0
  negotiation auto
!
interface GigabitEthernet0
  vrf forwarding Mgmt-intf
  no ip address
  negotiation auto
```



## Cisco UBE 2:

```
interface GigabitEthernet0/0/0
  description BT WAN Interface
  ip address 192.XX.XX.XX 255.255.255.128
  media-type rj45
  negotiation auto
  redundancy rii 15
  redundancy group 2 ip 192.XX.XX.XX exclusive
!
interface GigabitEthernet0/0/1
  description BT LAN Interface
  ip address 10.80.18.49 255.255.255.0
  negotiation auto
  redundancy rii 16
  redundancy group 2 ip 10.80.18.50 exclusive
!
interface GigabitEthernet0/1/0
  description BT CubeHA Interface
  ip address 10.70.50.110 255.255.255.0
  negotiation auto
!
interface GigabitEthernet0
  vrf forwarding Mgmt-intf
  no ip address
  shutdown
  negotiation auto
!
```



## Global Cisco UBE settings

In order to enable Cisco UBE IP2IP SBC functionality, following command has to be entered:

```
voice service voip
  no ip address trusted authenticate
  address-hiding
  mode border-element license capacity 20
  allow-connections sip to sip
  redundancy-group 2
  fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
  sip
    session refresh
    asserted-id pai
    privacy pstn
    early-offer forced
    midcall-signaling passthru
    g729 annexb-all
```

### Explanation

Command	Description
allow-connections sip to sip	Allow IP2IP connections between two SIP call legs
redundancy-group 2	Enable High Availability for the VoIP service
fax protocol	Specifies the fax protocol
asserted-id	Specifies the privacy header in the outgoing SIP requests and response messages



## Codecs

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

```
voice class codec 1
  codec preference 1 g729r8
  codec preference 2 g711ulaw
  codec preference 3 g711alaw
!
```

```
voice class codec 2
  codec preference 1 g711ulaw
  codec preference 2 g729r8
  codec preference 3 g711alaw
!
```

## Dial peer

### Outbound Dial-peer to British Telecom:

```
dial-peer voice 500 voip
  description Outgoing Call to BT - LAN facing
  huntstop
  session protocol sipv2
  session transport udp
  incoming called-number .T
  voice-class codec 1
  voice-class sip asserted-id pai
  voice-class sip options-keepalive
  voice-class sip bind control source-interface GigabitEthernet0/0/1
  voice-class sip bind media source-interface GigabitEthernet0/0/1
  dtmf-relay rtp-nte
  fax-relay ecm disable
  fax rate 14400
```



```
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 510 voip
description Outgoing call to BT - WAN facing
translation-profile outgoing E164dialing
huntstop
destination-pattern .T
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip profiles 1
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
```



### Inbound Dial-peer from British Telecom:

```
dial-peer voice 600 voip
description Incoming call to IP-PBX - WAN facing
huntstop
session protocol sipv2
session transport udp
incoming called-number +4144529....
incoming uri via PSTN
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 610 voip
description Incoming call to PBX - LAN facing
huntstop
destination-pattern +4144529....
session protocol sipv2
session target ipv4:10.80.18.2
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip profiles 1
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
```



```
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
```





## Configuration example

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

### *Active Cisco UBE:*

Current configuration : 6222 bytes

!

! Last configuration change at 09:44:23 UTC Mon Nov 6 2017

!

version 16.6

service timestamps debug datetime msec

service timestamps log datetime msec

platform qfp utilization monitor load 80

no platform punt-keepalive disable-kernel-core

!

hostname BTCube1

!

boot-start-marker

boot system flash isr4300-universalk9.16.06.01.SPA.bin

boot-end-marker

!

vrf definition Mgmt-intf

!

address-family ipv4

exit-address-family

!

address-family ipv6

exit-address-family

!



```
enable secret 5 xxxxxxxx
!
no aaa new-model
!
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 2
    fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
    sip
        session refresh
        asserted-id pai
        privacy pstn
        early-offer forced
        midcall-signaling passthru
        g729 annexb-all
    !
voice class uri PSTN sip
    host ipv4:213.XX.XX.XX
voice class codec 1
    codec preference 1 g729r8
    codec preference 2 g711ulaw
    codec preference 3 g711alaw
```



```
!  
voice class codec 2  
    codec preference 1 g711ulaw  
    codec preference 2 g729r8  
    codec preference 3 g711alaw  
!  
voice class sip-profiles 1  
    request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>"  
    "<sip:+4144529\1@\2>"  
!  
voice translation-rule 1  
    rule 1 /\(^.....$\)/ /+\1/  
    rule 2 /\(^.....$\)/ /+\1/  
!  
voice translation-profile E164dialing  
    translate called 1  
!  
  
license udi pid ISR4321/K9 sn FD019220MQ8  
diagnostic bootup level minimal  
spanning-tree extend system-id  
!  
username cisco password 0 *****  
!  
  
redundancy  
    mode none  
    application redundancy  
        group 2  
            name b2bhaBT
```



```
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
!
track 1 interface GigabitEthernet0/0/0 line-protocol
!
track 2 interface GigabitEthernet0/0/1 line-protocol
!
!
interface GigabitEthernet0/0/0
description BT WAN Interface
ip address 192.XX.XX.XX 255.255.255.128
media-type rj45
negotiation auto
redundancy rii 15
redundancy group 2 ip 192.XX.XX.XX exclusive
!
interface GigabitEthernet0/0/1
description BT LAN Interface
ip address 10.80.18.48 255.255.255.0
negotiation auto
redundancy rii 16
redundancy group 2 ip 10.80.18.50 exclusive
!
interface GigabitEthernet0/1/0
description BT CubeHA Interface
```



```
ip address 10.70.50.100 255.255.255.0
negotiation auto
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 negotiation auto
!
ip forward-protocol nd
no ip http server
no ip http secure-server
ip route 0.0.0.0 0.0.0.0 192.XX.XX.XX
ip route 10.70.50.0 255.255.255.0 10.80.18.1
ip route 172.16.0.0 255.255.0.0 10.80.18.1
!
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 500 voip
 description Outgoing Call to BT - LAN facing
```



```
huntstop
session protocol sipv2
session transport udp
incoming called-number .T
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 510 voip
description Outgoing call to BT - WAN facing
translation-profile outgoing E164dialing
huntstop
destination-pattern .T
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip profiles 1
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
```



```
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 600 voip
description Incoming call to IP-PBX - WAN facing
huntstop
session protocol sipv2
session transport udp
incoming called-number +4144529....
incoming uri via PSTN
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 610 voip
description Incoming call to PBX - LAN facing
huntstop
destination-pattern +4144529....
```



```
session protocol sipv2
session target ipv4:10.80.18.2
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip profiles 1
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 710 voip
description IP-PBX to IP-PBX loopbackcall
translation-profile outgoing E164dialing
huntstop
destination-pattern 4144529....
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip profiles 1
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/0
```





```
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
!

sip-ua
keepalive target ipv4:213.XX.XX.XX:5060
timers keepalive active 180
sip-server ipv4:213.XX.XX.XX
!

line con 0
exec-timeout 0 0
login local
transport input none
stopbits 1

line aux 0
stopbits 1

line vty 0 4
exec-timeout 0 0
login local
transport input telnet
!

end
```



*Standby Cisco UBE:*

```
Current configuration : 6580 bytes
!
! Last configuration change at 09:53:15 UTC Mon Nov 6 2017
!
version 16.6
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname BTCube2
!
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 $1$aEKr$TYxs2kVsHhvpeqnj4bdYn.
!
no aaa new-model
```



```
!  
subscriber templating  
!  
!  
multilink bundle-name authenticated  
!  
!  
crypto pki trustpoint TP-self-signed-2548443246  
  enrollment selfsigned  
  subject-name cn=IOS-Self-Signed-Certificate-2548443246  
  revocation-check none  
  rsakeypair TP-self-signed-2548443246  
!  
!  
crypto pki certificate chain TP-self-signed-2548443246  
!  
!  
voice service voip  
  no ip address trusted authenticate  
  address-hiding  
  mode border-element license capacity 20  
  allow-connections sip to sip  
  redundancy-group 2  
  fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none  
sip  
  session refresh  
  asserted-id pai  
  privacy pstn  
  early-offer forced
```



```
midcall-signaling passthru
g729 annexb-all
!
voice class uri PSTN sip
  host ipv4:213.XX.XX.XX
voice class codec 1
  codec preference 1 g729r8
  codec preference 2 g711ulaw
  codec preference 3 g711alaw
!
voice class codec 2
  codec preference 1 g711ulaw
  codec preference 2 g729r8
  codec preference 3 g711alaw
!
voice class sip-profiles 1
  request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>"
"<sip:+4144529\1@\2>"
!
voice translation-rule 1
  rule 1 /\(^.....$\)/ /+\1/
  rule 2 /\(^.....$\)/ /+\1/
!
!
voice translation-profile E164dialing
  translate called 1
!
!
license udi pid ISR4321/K9 sn FDO19220MW3
license boot level appxk9
```



```
license boot level uck9
diagnostic bootup level minimal
spanning-tree extend system-id
!
username cisco privilege 15 password 7 131112193D5D1E7B7B2A
!
redundancy
mode none
application redundancy
group 2
name b2bhaBT
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
!
!
track 1 interface GigabitEthernet0/0/0 line-protocol
!
track 2 interface GigabitEthernet0/0/1 line-protocol
!
!
interface GigabitEthernet0/0/0
description BT WAN Interface
ip address 192.XX.XX.XX 255.255.255.128
media-type rj45
negotiation auto
```



```
redundancy rii 15
redundancy group 2 ip 192.XX.XX.XX exclusive
!
interface GigabitEthernet0/0/1
description BT LAN Interface
ip address 10.80.18.49 255.255.255.0
negotiation auto
redundancy rii 16
redundancy group 2 ip 10.80.18.50 exclusive
!
interface GigabitEthernet0/1/0
description BT CubeHA Interface
ip address 10.70.50.110 255.255.255.0
negotiation auto
!
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
shutdown
negotiation auto
!
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.XX.XX.XX
ip route 10.70.50.0 255.255.255.0 10.80.18.1
ip route 172.16.24.0 255.255.248.0 10.80.18.1
```



```
!  
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 500 voip  
  description Outgoing Call to BT - LAN facing  
  huntstop  
  session protocol sipv2  
  session transport udp  
  incoming called-number .T  
  voice-class codec 1  
  voice-class sip asserted-id pai  
  voice-class sip options-keepalive  
  voice-class sip bind control source-interface GigabitEthernet0/0/1  
  voice-class sip bind media source-interface GigabitEthernet0/0/1  
  dtmf-relay rtp-nte  
  fax-relay ecm disable  
  fax rate 14400  
  fax nsf 000000
```



```
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none  
no vad
```

```
!
```

```
dial-peer voice 510 voip
```

```
description Outgoing call to BT - WAN facing
```

```
translation-profile outgoing E164dialing
```

```
huntstop
```

```
destination-pattern .T
```

```
session protocol sipv2
```

```
session target sip-server
```

```
session transport udp
```

```
voice-class codec 1
```

```
voice-class sip asserted-id pai
```

```
voice-class sip profiles 1
```

```
voice-class sip options-keepalive
```

```
voice-class sip bind control source-interface GigabitEthernet0/0/0
```

```
voice-class sip bind media source-interface GigabitEthernet0/0/0
```

```
dtmf-relay rtp-nte
```

```
fax-relay ecm disable
```

```
fax rate 14400
```

```
fax nsf 000000
```

```
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none  
no vad
```

```
!
```

```
dial-peer voice 600 voip
```

```
description Incoming call to IP-PBX - WAN facing
```

```
huntstop
```

```
session protocol sipv2
```

```
session transport udp
```





```
incoming called-number +4144529....
incoming uri via PSTN
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 610 voip
description Incoming call to PBX - LAN facing
huntstop
destination-pattern +4144529....
session protocol sipv2
session target ipv4:10.80.18.2
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip profiles 1
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay ecm disable
```



```
fax rate 14400
```

```
fax nsf 000000
```

```
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
```

```
no vad
```

```
!
```

```
dial-peer voice 710 voip
```

```
description IP-PBX to IP-PBX loopbackcall
```

```
translation-profile outgoing E164dialing
```

```
huntstop
```

```
destination-pattern 4144529....
```

```
session protocol sipv2
```

```
session target sip-server
```

```
session transport udp
```

```
voice-class codec 1
```

```
voice-class sip asserted-id pai
```

```
voice-class sip profiles 1
```

```
voice-class sip options-keepalive
```

```
voice-class sip bind control source-interface GigabitEthernet0/0/0
```

```
voice-class sip bind media source-interface GigabitEthernet0/0/0
```

```
dtmf-relay rtp-nte
```

```
fax-relay ecm disable
```

```
fax rate 14400
```

```
fax nsf 000000
```

```
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
```

```
no vad
```

```
!
```

```
sip-ua
```

```
keepalive target ipv4:213.XX.XX.XX:5060
```

```
timers keepalive active 180
```



```

sip-server ipv4:213.XX.XX.XX
!
!
line con 0
  exec-timeout 0 0
  login local
  transport input none
  stopbits 1
line aux 0
  stopbits 1
line vty 0 4
  exec-timeout 0 0
  login local
  transport input telnet
!
End
```



## Configuring Cisco UCM 11.5 Cluster

### Cisco UCM Version

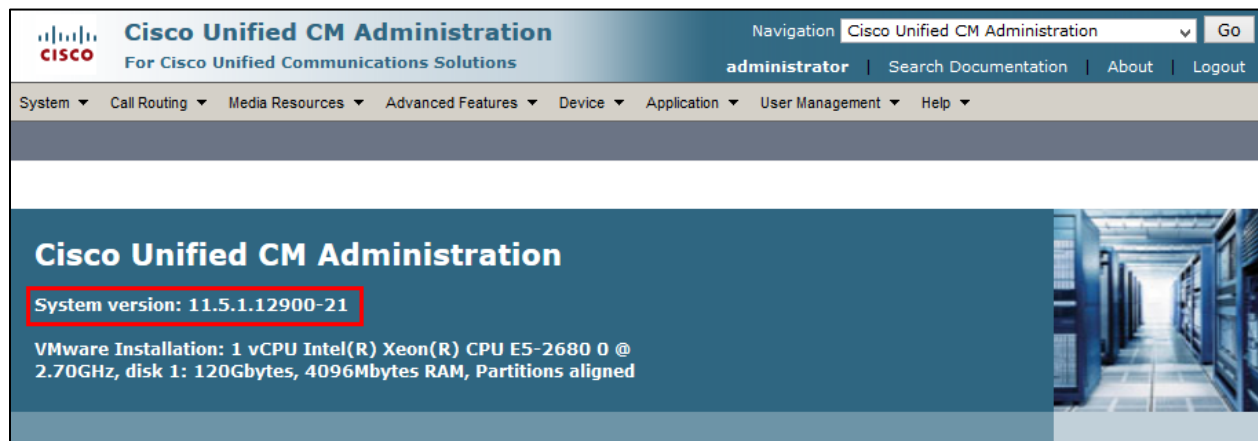


Figure 3: Cisco UCM Version



## Cisco Call Manager Service Parameters

**Navigation:** System → Service Parameters

- Select Server\* = Clus28pub--CUCM Voice/Video (Active)
- Select Service\* = Cisco CallManager (Active)
- Duplex Streaming Enabled\* = True
- All other fields are set to default values

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go  
administrator | Search Documentation | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

**Service Parameter Configuration** Related Links: Parameters for All Servers Go

Save Set to Default Advanced

**Status**  
Status: Ready

**Select Server and Service**

Server\* Clus28pub--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 Clus28pub--CUCM Voice/Video (Active)**

Parameter Name	Parameter Value	Suggested Value
<b>Call Throttling</b>		
<a href="#">Code Yellow Entry Latency</a> *	20	20
<a href="#">Code Yellow Exit Latency Calculation</a> *	40	40
<a href="#">Code Yellow Duration</a> *	5 ▾	5
<a href="#">Max Events Allowed</a> *	2000	2000
<a href="#">System Throttle Sample Size</a> *	10	10

Figure 4: Service Parameters



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

Figure 5: Service Parameters (Cont.)



## SIP Trunk Security Profile

**Navigation:** System → Security → SIP Trunk Security Profile

- Name\* = **Non Secure SIP Trunk Profile for BT** is used as an example
- Description = **Non Secure SIP Trunk Profile authenticated by null String** is used as an example
- Device Security Mode = **Non Secure**
- Incoming Transport Type\* = **TCP + UDP**
- Outgoing Transport Type = **UDP**

The screenshot displays the Cisco Unified CM Administration interface for the SIP Trunk Security Profile Configuration page. The page title is "SIP Trunk Security Profile Configuration" and it includes a "Related Links" section with a "Back To Find/List" link. The main configuration area is divided into several sections:

- Status:** Shows "Status: Ready".
- SIP Trunk Security Profile Information:** Contains the following fields and options:
  - Name\*:** Non Secure SIP Trunk Profile for BT
  - Description:** Non Secure SIP Trunk Profile authenticated by null String
  - Device Security Mode:** Non Secure (highlighted with a red box)
  - Incoming Transport Type\*:** TCP+UDP (highlighted with a red box)
  - Outgoing Transport Type:** UDP (highlighted with a red box)
  - Enable Digest Authentication:** ☐
  - Nonce Validity Time (mins)\*:** 600
  - X.509 Subject Name:** (Empty text area)
  - Incoming Port\*:** 5060
  - Enable Application level authorization:** ☐
  - Accept presence subscription:** ☐
  - Accept out-of-dialog refer\*\*:** ☒ (highlighted with a red box)
  - Accept unsolicited notification:** ☒ (highlighted with a red box)
  - Accept replaces header:** ☒ (highlighted with a red box)
  - Transmit security status:** ☐
  - Allow charging header:** ☐
  - SIP V.150 Outbound SDP Offer Filtering\*:** Use Default Filter

At the bottom of the page, there are buttons for "Save", "Delete", "Copy", "Reset", "Apply Config", and "Add New".

Figure 6: SIP Trunk Security Profile



## SIP Profile

**Navigation:** Device → Device Settings → SIP Profile

- Name\* = **Standard SIP Profile for BT** is used as an example
- Description = **Standard SIP Profile for BT** is used as an example

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go  
administrator | Search Documentation | About | Logou

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

**SIP Profile Configuration** Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

**Status**

Status: Ready  
All SIP devices using this profile must be restarted before any changes will take affect.

**SIP Profile Information**

Name\* Standard SIP Profile for BT  
Description Standard SIP Profile for BT  
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

☐ Redirect by Application  
☐ Disable Early Media on 180  
☐ Outgoing T.38 INVITE include audio mline  
☐ Offer valid IP and Send/Receive mode only for T.38 Fax Relay  
☐ Use Fully Qualified Domain Name in SIP Requests  
☐ Assured Services SIP conformance  
☐ 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

☐ Require SDP Inactive Exchange for Mid-Call Media Change  
☐ Allow RR/RS bandwidth modifier (RFC 3556)

Figure 7: SIP Profile





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
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	

Figure 8: SIP Profile (Cont.)



**Normalization Script**  
Normalization Script < None >  
☐ Enable Trace  

	Parameter Name	Parameter Value	
1			<span>+</span> <span>-</span>

**Incoming Requests FROM URI Settings**  
Caller ID DN  
Caller Name

**Trunk Specific Configuration**  
Reroute Incoming Request to new Trunk based on\* Never  
Resource Priority Namespace List < None >  
SIP Rel1XX Options\* Send PRACK if 1xx Contains SDP  
Video Call Traffic Class\* Mixed  
Calling Line Identification Presentation\* Default  
Session Refresh Method\* Invite  
Early Offer support for voice and video calls\* Disabled (Default value)  
☐ 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  
☐ Connect Inbound Call before Playing Queuing Announcement

**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)\* 60  
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 9: SIP Profile (Cont.)



## Trunk configuration

*Trunk configuration from Cisco UCM to the LAN side of the Cisco UBE:*

**Navigation:** Device → Trunk → Add New

The screenshot shows the 'Find and List Trunks' page in the Cisco Unified CM Administration interface. The page has a navigation bar at the top with the Cisco logo and 'Cisco Unified CM Administration For Cisco Unified Communications Solutions'. Below the navigation bar is a menu with options: System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled 'Find and List Trunks' and contains a search bar with the text 'Find Trunks where Device Name begins with'. There are buttons for 'Find', 'Clear Filter', and a plus/minus icon. Below the search bar is a message: 'No active query. Please enter your search criteria using the options above.' At the bottom left, the 'Add New' button is highlighted with a red box.

Figure 10: Add New Trunk to Cisco UBE

- Select 'Trunk Type' as SIP Trunk and 'Device Protocol' as SIP and select 'Next' as shown below.

The screenshot shows the 'Trunk Configuration' page in the Cisco Unified CM Administration interface. The page has a navigation bar at the top with the Cisco logo and 'Cisco Unified CM Administration For Cisco Unified Communications Solutions'. Below the navigation bar is a menu with options: System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled 'Trunk Configuration' and contains a 'Next' button with a green arrow. Below the 'Next' button is a 'Status' section with an information icon and the text 'Status: Ready'. Below the 'Status' section is a 'Trunk Information' section with three dropdown menus: 'Trunk Type\*' (SIP Trunk), 'Device Protocol\*' (SIP), and 'Trunk Service Type\*' (None(Default)). At the bottom left, the 'Next' button is highlighted with a red box. Below the 'Next' button is an information icon and the text '\*- indicates required item.'

Figure 11: Add SIP Trunk Type

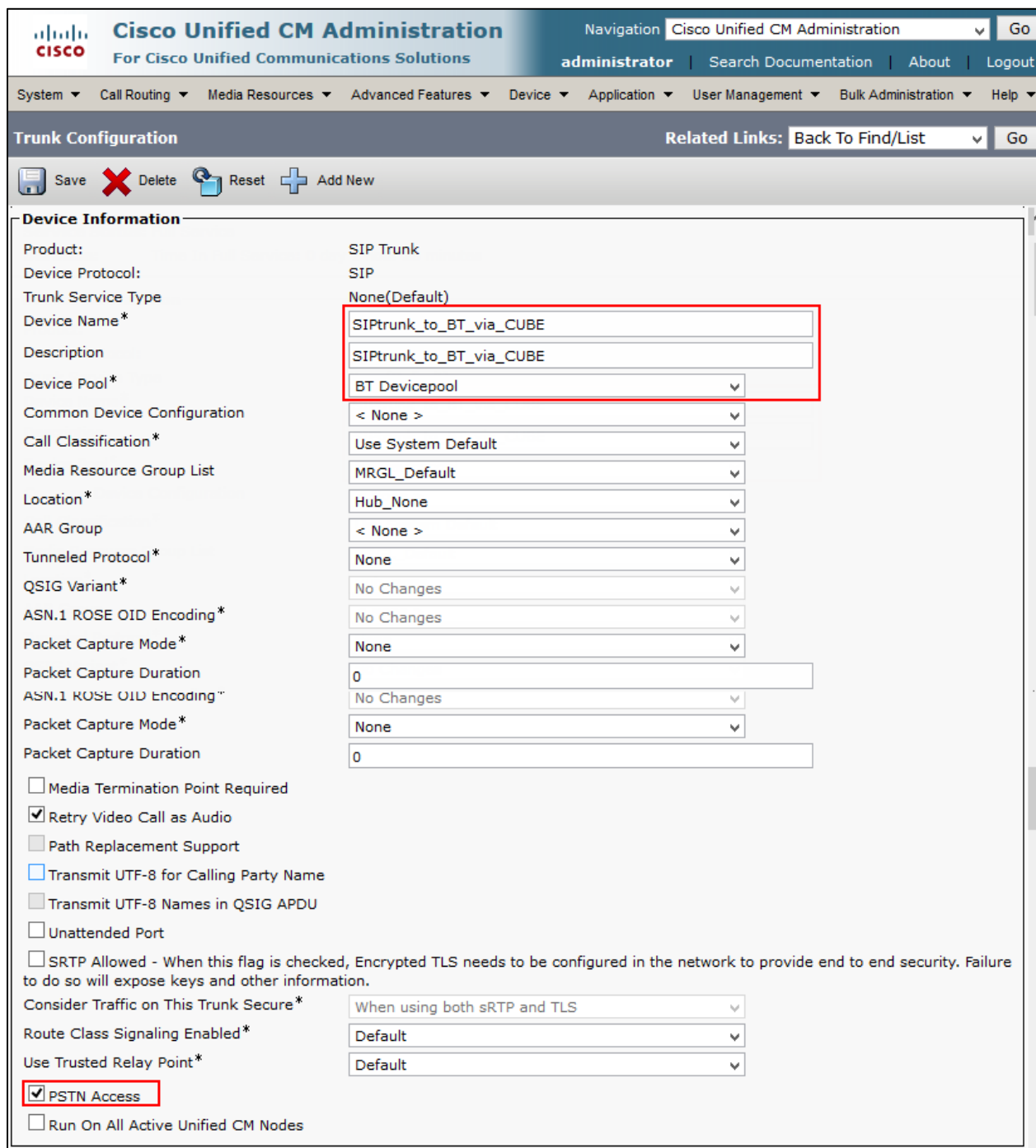


Figure 12: SIP Trunk to Cisco UBE

**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\* PAI  
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"/>

**Connected Party Settings**  
Connected Party Transformation CSS < None >  
☒ Use Device Pool Connected Party Transformation CSS

Figure 13: SIP Trunk to Cisco UBE (Cont.)

- Configure the Virtual LAN IP address of the Cisco UBE and the Destination Port
- Configure the SIP Trunk Security Profile and SIP Profile as shown below
- The rest of the configuration is all default
- Click Save and Reset after completion



**Outbound Calls**

Called Party Transformation CSS< None >

☒ Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS< None >

☒ Use Device Pool Calling Party Transformation CSS

Calling Party Selection\*OriginatorCalling Line ID Presentation\*DefaultCalling Name Presentation\*DefaultCalling and Connected Party Info Format\*Deliver DN only in connected party

☒ Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS< None >

☒ Use Device Pool Redirecting Party Transformation CSS

**Caller Information**

Caller ID DN

Caller Name

☐ Maintain Original Caller ID DN and Caller Name in Identity Headers

**SIP Information**

**Destination**

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1 *	10.80.18.50		5060

MTP Preferred Originating Codec\*711ulawBLF Presence Group\*Standard Presence groupSIP Trunk Security Profile\*Non Secure SIP Trunk Profile for BTRerouting Calling Search Space< None >Out-Of-Dialog Refer Calling Search Space< None >SUBSCRIBE Calling Search Space< None >SIP Profile\*Standard SIP Profile for BTView DetailsDTMF Signaling Method\*RFC 2833

**Normalization Script**

Normalization Script< None >

☐ Enable Trace

	Parameter Name	Parameter Value	
1			+ -

**Recording Information**

☒ None

☐ This trunk connects to a recording-enabled gateway

☐ This trunk connects to other clusters with recording-enabled gateways

**Geolocation Configuration**

Geolocation< None >Geolocation Filter< None >

☐ Send Geolocation Information

SaveDeleteResetAdd New

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



## Trunk configuration from Cisco UCM to Fax Gateway:

**Navigation:** Devices → Trunk → Add New

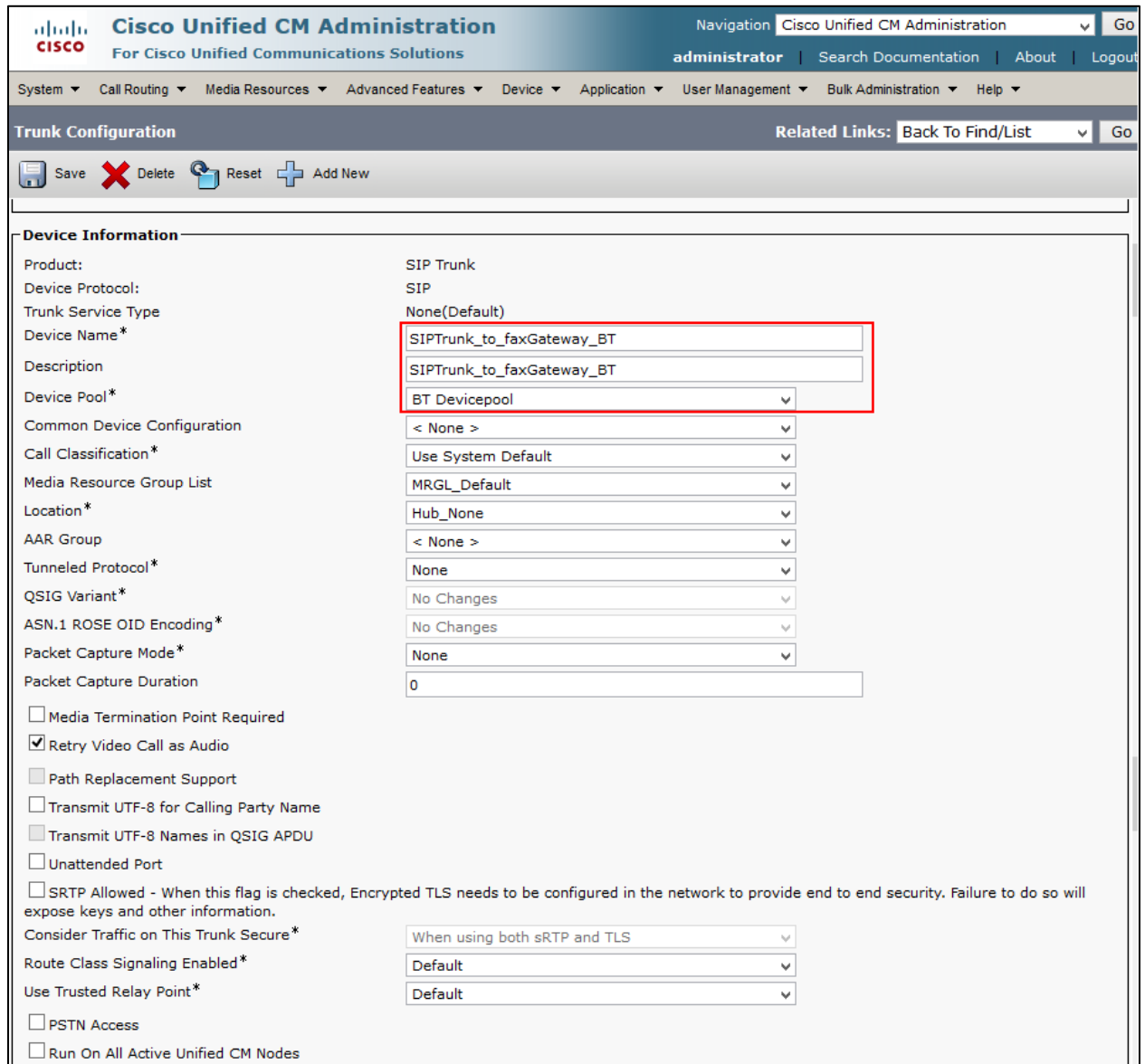
The screenshot shows the 'Find and List Trunks' page in the Cisco Unified CM Administration interface. The page has a header with the Cisco logo and navigation links. Below the header is a menu bar with options like System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled 'Find and List Trunks' and includes a '+ Add New' button. Below this is a search section with a 'Find Trunks where' dropdown set to 'Device Name', a 'begins with' dropdown, and a search input field. There are 'Find', 'Clear Filter', and '+ -' buttons. A message states 'No active query. Please enter your search criteria using the options above.' At the bottom, the 'Add New' button is highlighted with a red box.

Figure 15: Add New Trunk to Fax Gateway

- Select 'Trunk Type' as SIP Trunk and 'Device Protocol' as SIP and select 'Next' as shown below.

The screenshot shows the 'Trunk Configuration' page in the Cisco Unified CM Administration interface. The page has a header with the Cisco logo and navigation links. Below the header is a menu bar with options like System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled 'Trunk Configuration' and includes a 'Related Links: Back To Find/List' link. Below this is a 'Next' button. The 'Status' section shows 'Status: Ready'. The 'Trunk Information' section has three dropdown menus: 'Trunk Type\*' set to 'SIP Trunk', 'Device Protocol\*' set to 'SIP', and 'Trunk Service Type\*' set to 'None(Default)'. At the bottom, the 'Next' button is highlighted with a red box. A note at the bottom states '\*- indicates required item.'

Figure 16: Add SIP Trunk Type







**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\* PAI  
SIP Privacy\* Default

**Inbound Calls**  
Significant Digits\* All  
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"/>

**Connected Party Settings**  
Connected Party Transformation CSS < None >  
☒ Use Device Pool Connected Party Transformation CSS

Figure 18: SIP Trunk to FAX Gateway (Cont.)



- Configure the IP address of Fax Gateway and the Destination Port
- Configure the SIP Trunk Security Profile and SIP Profile as shown below
- The rest of the configuration is all default
- Click Save and Reset after completion

**Outbound Calls**  
Called Party Transformation CSS < None >  
☒ Use Device Pool Called Party Transformation CSS  
Calling Party Transformation CSS < None >  
☒ 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  
☐ Redirecting Diversion Header Delivery - Outbound  
Redirecting Party Transformation CSS < None >  
☒ Use Device Pool Redirecting Party Transformation CSS  
**Caller Information**  
Caller ID DN  
Caller Name  
☐ Maintain Original Caller ID DN and Caller Name in Identity Headers

**SIP Information**  
**Destination**  
☐ Destination Address is an SRV  

	Destination Address	Destination Address IPv6	Destination Port
1 *	172.16.31.50		5060

  
MTP Preferred Originating Codec\* 711ulaw  
BLF Presence Group\* Standard Presence group  
SIP Trunk Security Profile\* Non Secure SIP Trunk Profile for BT  
Rerouting Calling Search Space < None >  
Out-Of-Dialog Refer Calling Search Space < None >  
SUBSCRIBE Calling Search Space < None >  
SIP Profile\* Standard SIP Profile for BT [View Details](#)  
DTMF Signaling Method\* No Preference

Figure 19: SIP Trunk to FAX Gateway (Cont.)



## Routing configuration

*Route Pattern for Cisco UBE:*

**Navigation:** Call Routing → Route/Hunt → Route Pattern → Add New

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go  
administrator | Search Documentation | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

**Find and List Route Patterns**

+ Add New

**Route Patterns**

Find Route Patterns where Pattern ▾ begins with ▾ Find Clear Filter + -

No active query. Please enter your search criteria using the options above.

Add New

Figure 20: Add New Route Pattern for Cisco UBE



Cisco

Cisco Unified CM Administration

For Cisco Unified Communications Solutions

Navigation Cisco Unified CM AdministrationGo

administrator | Search Documentation | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Route Pattern Configuration

Related Links: Back To Find/ListGo

SaveDeleteCopyAdd New

Status

Status: Ready

Pattern Definition

Route Pattern\*

[0-9]![0-9#]

Route Partition

< None >

Description

PSTN access-national

Numbering Plan

-- Not Selected --

Route Filter

< None >

MLPP Precedence\*

Default

Apply Call Blocking Percentage

Resource Priority Namespace Network Domain

< None >

Route Class\*

Default

Gateway/Route List\*

SIPtrunk\_to\_BT\_via\_CUBE

(Edit)

Route Option

Route this pattern

Block this pattern

No Error

Call Classification\*

OffNet

External Call Control Profile

< None >

Allow Device Override

Provide Outside Dial Tone

Allow Overlap Sending

Urgent Priority

Require Forced Authorization Code

Authorization Level\*

0

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation\*

Default

Calling Name Presentation\*

Default

Calling Party Number Type\*

Cisco CallManager

Calling Party Numbering Plan\*

Cisco CallManager

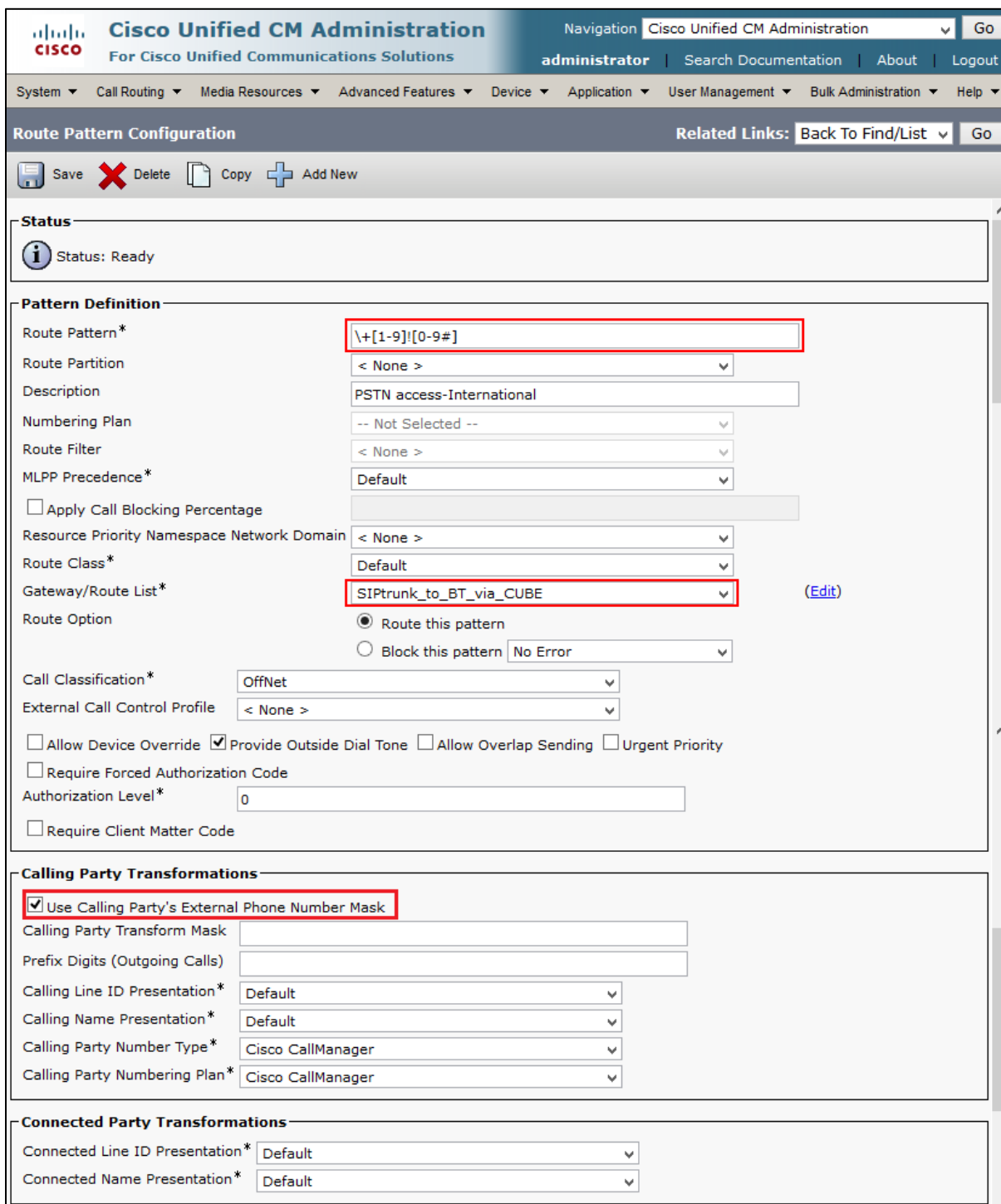
Figure 21: Route Pattern Configuration for Cisco UBE-PSTN Access

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<b>Connected Party Transformations</b>		
Connected Line ID Presentation*	Default	
Connected Name Presentation*	Default	
<b>Called Party Transformations</b>		
Discard Digits	< None >	
Called Party Transform Mask		
Prefix Digits (Outgoing Calls)		
Called Party Number Type*	Cisco CallManager	
Called Party Numbering Plan*	Cisco CallManager	
<b>ISDN Network-Specific Facilities Information Element</b>		
Network Service Protocol	-- Not Selected --	
Carrier Identification Code		
Network Service	Service Parameter Name	Service Parameter Value
-- Not Selected --	< Not Exist >	
<div>Save Delete Copy Add New</div>		

Figure 22: Route Pattern Configuration for Cisco UBE-PSTN Access (Cont.)



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**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: -- Not Selected --

Service Parameter Name: < Not Exist >

Service Parameter Value:

Save Delete Copy Add New

Figure 24: Route Pattern Configuration for Cisco UBE-PSTN Access-International (Cont.)

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**Route Pattern Configuration** Related Links: Back To Find/List Go

Save Delete Copy Add New

**Status**

Status: Ready

**Pattern Definition**

Route Pattern\*: 6.[0-9]!([0-9#])

Route Partition: < None >

Description: For Anonymous Call

Numbering Plan: -- Not Selected --

Route Filter: < None >

MLPP Precedence\*: Default

☐ Apply Call Blocking Percentage

Resource Priority Namespace Network Domain: < None >

Route Class\*: Default

Gateway/Route List\*: SIPtrunk\_to\_BT\_via\_CUBE (Edit)

Route Option

☒ Route this pattern

☐ Block this pattern No Error

Call Classification\*: OffNet

External Call Control Profile: < None >

Figure 25 Route Pattern Configuration for Cisco UBE-PSTN Access National- Anonymous Call



<input type="checkbox"/> Allow Device Override	<input checked="" type="checkbox"/> Provide Outside Dial Tone	<input type="checkbox"/> Allow Overlap Sending	<input type="checkbox"/> Urgent Priority
<input type="checkbox"/> Require Forced Authorization Code			
Authorization Level*	<input type="text" value="0"/>		
<input type="checkbox"/> Require Client Matter Code			
<b>Calling Party Transformations</b>			
<input checked="" type="checkbox"/> Use Calling Party's External Phone Number Mask			
Calling Party Transform Mask	<input type="text"/>		
Prefix Digits (Outgoing Calls)	<input type="text"/>		
Calling Line ID Presentation*	Restricted		
Calling Name Presentation*	Restricted		
Calling Party Number Type*	Cisco CallManager		
Calling Party Numbering Plan*	Cisco CallManager		
<b>Connected Party Transformations</b>			
Connected Line ID Presentation*	Default		
Connected Name Presentation*	Default		
<b>Called Party Transformations</b>			
Discard Digits	PreDot		
Called Party Transform Mask	<input type="text"/>		
Prefix Digits (Outgoing Calls)	<input type="text"/>		
Called Party Number Type*	Cisco CallManager		
Called Party Numbering Plan*	Cisco CallManager		
<b>ISDN Network-Specific Facilities Information Element</b>			
Network Service Protocol	-- Not Selected --		
Carrier Identification Code	<input type="text"/>		
Network Service	Service Parameter Name	Service Parameter Value	
-- Not Selected --	< Not Exist >		
<input type="button" value="Save"/> <input type="button" value="Delete"/> <input type="button" value="Copy"/> <input type="button" value="Add New"/>			

Figure 26: Route Pattern Configuration for Cisco UBE-PSTN Access National- Anonymous Call (Cont.)





### Route Pattern for Fax Gateway:

- **Navigation:** Call Routing → Route/Hunt → Route Pattern → Add New

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### Find and List Route Patterns

+ Add New

**Route Patterns**

Find Route Patterns where Pattern ▾ begins with ▾ Find Clear Filter + -

No active query. Please enter your search criteria using the options above.

Add New

Figure 27: Add New Route patter to Fax Gateway

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### Route Pattern Configuration

Related Links: Back To Find/List Go

Save X Delete Copy + Add New

**Status**

i Status: Ready

**Pattern Definition**

Route Pattern\* 91XX

Route Partition < None >

Description for BT Fax

Numbering Plan -- Not Selected --

Route Filter < None >

MLPP Precedence\* Default

☐ Apply Call Blocking Percentage

Resource Priority Namespace Network Domain < None >

Route Class\* Default

Gateway/Route List\* SIPTrunk\_to\_faxGateway\_BT (Edit)

Route Option

☒ Route this pattern

☐ Block this pattern No Error

Figure 28: Route Pattern Configuration for Fax Gateway



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"/> Urgent Priority	
<input type="checkbox"/> Require Forced Authorization Code	
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

### Called Party Transformations

Discard Digits	< None >
Called Party Transform Mask	+4144529XXXX
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 >		

Save Delete Copy Add New

Figure 29: Route Pattern Configuration for Fax Gateway (Cont.)



## Configuring Cisco Voice Gateway for Fax

### Global Settings

```
voice service voip
  allow-connections sip to sip
  no supplementary-service sip handle-replaces
  redirect ip2ip
  fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
  no fax-relay sg3-to-g3
  sip
    rel1xx disable
    midcall-signaling passthru
    g729 annexb-all
!
```

### Codecs

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

```
voice class codec 1
  codec preference 1 g729r8
  codec preference 2 g711ulaw
  codec preference 3 g711alaw
!
voice class codec 2
  codec preference 1 g711ulaw
  codec preference 2 g729r8
  codec preference 3 g711alaw
!
```



## Dial peer

### Outbound Dial-peer to Cisco UCM:

```
dial-peer voice 4000 voip
description BT Outbound FAX
translation-profile outgoing E164dialing
destination-pattern 1972265....
session protocol sipv2
session target ipv4:10.80.18.2
session transport udp
voice-class codec 1
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
```

### Inbound Dial-peer from Cisco UCM:

```
dial-peer voice 5000 voip
description BT Inbound FAX
service session
session protocol sipv2
session transport udp
incoming called-number +4144529....
voice-class codec 1
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
```



## POTS and Port Configuration:

Based on configured POTS destination pattern, gateway forwards the call to designated voice port.

```
dial-peer voice 45 pots
  huntstop
  service session
  destination-pattern +4144529XXXX
  no digit-strip
  port 0/0/0
  forward-digits all
```

```
voice-port 0/0/0
  no echo-cancel enable
  no vad
  cptone IN
  station-id name BT Fax user1
  station-id number +4144529XXXX
  caller-id enable
!
```

## Configuration example

The following configuration snippet contains a sample configuration of Cisco Voice Gateway with all parameters mentioned previously.

```
Current configuration : 13111 bytes
```

```
!
```

```
! Last configuration change at 16:30:57 IST Mon Nov 20 2017 by cisco
```

```
version 15.1
```

```
service timestamps debug datetime msec
```

```
service timestamps log datetime msec
```

```
no service password-encryption
```

```
!
```

```
hostname cme.in.tekvizion.com
```

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```
!  
boot-start-marker  
boot-end-marker  
!  
!  
enable password *****  
!  
aaa new-model  
!  
aaa authentication login local_auth local  
!  
aaa session-id common  
clock timezone IST 5 30  
network-clock-participate wic 2  
network-clock-participate wic 3  
!  
dot11 syslog  
ip source-route  
!  
ip cef  
!  
multilink bundle-name authenticated  
!  
isdn switch-type primary-qsig  
!  
voice rtp send-recv  
!  
voice service pots  
!  
voice service voip  
    allow-connections sip to sip  
    no supplementary-service sip handle-replaces  
    redirect ip2ip
```



```
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no fax-relay sg3-to-g3
sip
    rel1xx disable
    midcall-signaling passthru
    g729 annexb-all
!
voice class codec 1
    codec preference 1 g729r8
    codec preference 2 g711ulaw
    codec preference 3 g711alaw
!
voice class codec 2
    codec preference 1 g711ulaw
    codec preference 2 g729r8
    codec preference 3 g711alaw
!
voice translation-rule 1
    rule 1 /\(^.....$\)/ /+1/
!
voice translation-profile E164dialing
    translate called 1
!
voice-card 0
    dspfarm
    dsp services dspfarm
!
crypto pki token default removal timeout 0
!
license udi pid CISC02851 sn FHK1137F4LY
username cisco password 0 *****
!
```



```
controller E1 0/2/0
  shutdown
  pri-group timeslots 1-31 service mgcp
!
controller E1 0/3/0
  clock source internal
  pri-group timeslots 1-31
!
interface GigabitEthernet0/0
  ip address 172.16.31.50 255.255.255.0
  duplex auto
  speed auto
!
interface Service-Engine0/0
  no ip address
  shutdown
!
interface GigabitEthernet0/1
  no ip address
  ip nat outside
  ip virtual-reassembly in
  shutdown
  duplex auto
  speed auto
!
interface Serial0/2/0:15
  no ip address
!
interface Serial0/3/0:15
  no ip address
!
interface Service-Engine1/0
  no ip address
```





```
shutdown
!
ip forward-protocol nd
!
ip http server
no ip http secure-server
!
ip route 0.0.0.0 0.0.0.0 172.16.31.1
!
ipv6 route ::/0 2620:96:c000:8::1
!
snmp-server community public RO
snmp-server location Chennai
!
ipv6 access-list ipv6
 permit ipv6 any any
!
control-plane
!
!
voice-port 0/0/0
 no echo-cancel enable
 no vad
 cptone IN
 station-id name BT Fax user1
 station-id number +4144529XXXX
 caller-id enable
!
voice-port 0/0/1
!
voice-port 0/3/0:15
!
voice-port 0/2/0:15
```



```
!  
voice-port 0/1/0  
  no echo-cancel enable  
  no vad  
  cptone IN  
  station-id name BT Fax user2  
  station-id number +4144529XXXX  
  caller-id enable  
!  
voice-port 0/1/1  
  !  
  ccm-manager mgcp  
  no ccm-manager fax protocol cisco  
  ccm-manager music-on-hold  
  ccm-manager config  
  !  
  mgcp  
  mgcp rtp unreachable timeout 1000 action notify  
  mgcp modem passthrough voip mode nse  
  mgcp package-capability rtp-package  
  mgcp package-capability sst-package  
  mgcp package-capability pre-package  
  mgcp default-package mt-package  
  no mgcp package-capability fxr-package  
  no mgcp timer receive-rtcp  
  mgcp sdp simple  
  mgcp fax t38 inhibit  
  mgcp rtp payload-type g726r16 static  
  !  
  mgcp profile default  
  !  
  sccp local GigabitEthernet0/0  
  sccp ccm 10.71.3.10 identifier 1 version 7.0
```



```
!  
sccp ccm group 6  
  bind interface GigabitEthernet0/0  
  associate ccm 1 priority 1  
!  
dial-peer voice 5000 voip  
  description BT Inbound FAX  
  service session  
  session protocol sipv2  
  session transport udp  
  incoming called-number +4144529....  
  voice-class codec 1  
  dtmf-relay rtp-nte  
  fax-relay ecm disable  
  fax rate 14400  
  fax nsf 000000  
  fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none  
  no vad  
!  
dial-peer voice 4000 voip  
  description BT Outbound FAX  
  translation-profile outgoing E164dialing  
  destination-pattern 1972265....  
  session protocol sipv2  
  session target ipv4:10.80.18.2  
  session transport udp  
  voice-class codec 1  
  dtmf-relay rtp-nte  
  fax-relay ecm disable  
  fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none  
  no vad  
!  
dial-peer voice 45 pots
```



```
huntstop
service session
destination-pattern +4144529XXXX
no digit-strip
port 0/0/0
forward-digits all
!
dial-peer voice 4001 voip
description BT Outbound FAX loopback
destination-pattern 4144529....
session protocol sipv2
session target ipv4:10.80.18.2
session transport udp
voice-class codec 1
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 46 pots
huntstop
service session
destination-pattern +4144529XXXX
no digit-strip
port 0/1/0
forward-digits all
!
!
telephony-service
max-ephones 50
max-dn 60
```



```
ip source-address 172.16.31.50 port 2000
service phone sshAccess 0
cnf-file perphone
max-conferences 8 gain -6
web admin system name user password *****
transfer-system full-consult
create cnf-files version-stamp 7960 Nov 22 2013 19:05:58
!
banner login ^CC
!
line con 0
line aux 0
line 66
  no activation-character
  no exec
  transport preferred none
  transport input all
  transport output pad telnet rlogin lapb-ta mop udptn v120 ssh
line 194
  no activation-character
  no exec
  transport preferred none
  transport input all
  transport output all
line vty 0 4
  session-timeout 180
  exec-timeout 0 0
  password *****
  login authentication local_auth
  transport input all
!
scheduler allocate 20000 1000
ntp server 103.6.16.254
```



end

cme.in.tekvizion.com#

## 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



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