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.
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:

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Transport from Cisco UBE to Cisco UCM</td>
<td>UDP with RTP</td>
</tr>
<tr>
<td>Transport from Cisco UBE to British Telecom</td>
<td>UDP with RTP</td>
</tr>
<tr>
<td>Voice Mail Support</td>
<td>YES</td>
</tr>
<tr>
<td>Session Refresh</td>
<td>YES</td>
</tr>
<tr>
<td>Early Media support with PRACK</td>
<td>YES</td>
</tr>
</tbody>
</table>
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:

```plaintext
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

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>allow-connections sip to sip</td>
<td>Allow IP2IP connections between two SIP call legs</td>
</tr>
<tr>
<td>redundancy-group 2</td>
<td>Enable High Availability for the VoIP service</td>
</tr>
<tr>
<td>fax protocol</td>
<td>Specifies the fax protocol</td>
</tr>
<tr>
<td>asserted-id</td>
<td>Specifies the privacy header in the outgoing SIP requests and response</td>
</tr>
<tr>
<td></td>
<td>messages</td>
</tr>
</tbody>
</table>
Codecs

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

voice class codec 1  
  codec preference 1 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-nlte  
  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-npe
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-nce
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 xxxxxxx
!
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 FDO19220MQ8
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-npe
defax-relay ecm disable
defax 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-npe
fax-relay ecm disable
dial-peer voice 600 voip
description Incoming call to IP-PBX - WAN facing
huntstop
session protocol sipv2
session transport udp
incoming called-number +4144529....
ingoing 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-npe
fax-relay ecm disable
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-nce
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-nce
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 FDO1922OMw3
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

![Service Parameters Configuration](image)

Figure 4: Service Parameters
### Clusterwide Parameters (Service)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value 1</th>
<th>Value 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default Network Hold MOH Audio Source ID</td>
<td>1</td>
<td>1</td>
</tr>
<tr>
<td>Default User Hold MOH Audio Source ID</td>
<td>1</td>
<td>1</td>
</tr>
<tr>
<td>Duplex Streaming Enabled</td>
<td>True</td>
<td>False</td>
</tr>
<tr>
<td>Media Exchange Interface Capability Timer</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>Send Multicast MOH in H.245 OLC Message</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>Media Exchange Timer</td>
<td>12</td>
<td>12</td>
</tr>
<tr>
<td>Media Exchange Stop Streaming Timer</td>
<td>8</td>
<td>8</td>
</tr>
<tr>
<td>Open Video Channel Response Timer for SIP Interop</td>
<td>500</td>
<td>500</td>
</tr>
<tr>
<td>Part Received Timer After Call Connection</td>
<td>200</td>
<td>200</td>
</tr>
<tr>
<td>Media Resource Allocation Timer</td>
<td>12</td>
<td>12</td>
</tr>
<tr>
<td>H.225 and Transcoder Resource Throttling Percentage</td>
<td>95</td>
<td>95</td>
</tr>
<tr>
<td>Intercluster Capabilities Mismatch Timer</td>
<td>1000</td>
<td>1000</td>
</tr>
<tr>
<td>Silence Suppression</td>
<td>True</td>
<td>False</td>
</tr>
<tr>
<td>Silence Suppression for Gateways</td>
<td>True</td>
<td>False</td>
</tr>
<tr>
<td>Strip G.729 Annex B (Silence Suppression) from Capabilities</td>
<td>True</td>
<td>False</td>
</tr>
<tr>
<td>Enable Source IP Address Verification for Software Media Devices</td>
<td>True</td>
<td>True</td>
</tr>
</tbody>
</table>

**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

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

![SIP Profile Configuration](image)

**Figure 7:** SIP Profile
**Figure 8: SIP Profile (Cont.)**
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

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.

Figure 11: Add SIP Trunk Type
Figure 12: SIP Trunk to Cisco UBE
- 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 Cells**

- Called Party Transformation CSS: `< None >`
- Use Device Pool Called Party Transformation CSS: `< None >`
- Calling Party Transformation CSS: `< None >`
- Use Device Pool Calling Party Transformation CSS: `< None >`
- 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: `< None >`

**Caller Information**

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

**SIP Information**

- Destination Address: 10.80.19.50
- Destination Address IPv6: `5060`
- Destination Port: 5060
- MTP Preferred Originating Codec: 711ulaw
- BU Presence Group: Standard Presence group
- SIP Trunk Security Profile: Non Secure SIP Trunk Profile for BT
- Recomputed Calling Search Space: `< None >`
- Out-Of-Dialing Calling Search Space: `< None >`
- SUBSCRIBE Calling Search Space: `< None >`
- SIP Profile: Standard SIP Profile for BT
- DTMF Signaling Method: RFC 2833

**Normalization Script**

- Normalization Script: `< None >`
- Enable Trace

**Recording Information**

- Recording: 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

---

Figure 14: SIP Trunk to Cisco UBE (Cont.)
**Trunk configuration from Cisco UCM to Fax Gateway:**

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

---

**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.

**Figure 16:** Add SIP Trunk Type
Figure 17: SIP Trunk to FAX Gateway
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

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

Route Pattern for Cisco UBE:

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

![Add New Route Pattern for Cisco UBE](image)

Figure 20: Add New Route Pattern for Cisco UBE
**Figure 21: Route Pattern Configuration for Cisco UBE-PSTN Access**

<table>
<thead>
<tr>
<th>Route Pattern</th>
<th>[0-9][0-9#]</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Partition</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Description</td>
<td>PSTN access-national</td>
</tr>
<tr>
<td>Numbering Plan</td>
<td>Not Selected</td>
</tr>
<tr>
<td>Route Filter</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>MLPP Precedence</td>
<td>Default</td>
</tr>
<tr>
<td>Resource Priority</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Route Class</td>
<td>Default</td>
</tr>
<tr>
<td>Gateway/Route List</td>
<td>S[Trunk_to_BT_vio_CUBE]</td>
</tr>
<tr>
<td>Route Option</td>
<td>Route this pattern</td>
</tr>
<tr>
<td>Call Classification</td>
<td>OffNet</td>
</tr>
<tr>
<td>External Call Control</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Allow Device Override</td>
<td>Provide Outside Dial Tone</td>
</tr>
<tr>
<td>Require Forced Authorization Code</td>
<td></td>
</tr>
<tr>
<td>Authorization Level</td>
<td>0</td>
</tr>
<tr>
<td>Use Calling Party’s External Phone Number Mask</td>
<td></td>
</tr>
</tbody>
</table>

**Calling Party Transformations**

<table>
<thead>
<tr>
<th>Calling Party Transform Mask</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Prefix Digits (Outgoing Calls)</td>
<td></td>
</tr>
<tr>
<td>Calling Line ID Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Calling Name Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Calling Party Number Type</td>
<td>Cisco CallManager</td>
</tr>
<tr>
<td>Calling Party Numbering Plan</td>
<td>Cisco CallManager</td>
</tr>
<tr>
<td>Connected Party Transformsations</td>
<td></td>
</tr>
<tr>
<td>----------------------------------</td>
<td>---</td>
</tr>
<tr>
<td>Connected Line ID Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Connected Name Presentation</td>
<td>Default</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Called Party Transformations</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Discard Digits</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Called Party Transform Mask</td>
<td></td>
</tr>
<tr>
<td>Prefix Digits (Outgoing Calls)</td>
<td></td>
</tr>
<tr>
<td>Called Party Number Type*</td>
<td>Cisco CallManager</td>
</tr>
<tr>
<td>Called Party Numbering Plan*</td>
<td>Cisco CallManager</td>
</tr>
</tbody>
</table>

**ISDN Network-Specific Facilities Information Element**

<table>
<thead>
<tr>
<th>Network Service Protocol</th>
<th>-- Not Selected --</th>
</tr>
</thead>
<tbody>
<tr>
<td>Carrier Identification Code</td>
<td>Service Parameter Name</td>
</tr>
<tr>
<td>Network Service</td>
<td></td>
</tr>
<tr>
<td>-- Not Selected --</td>
<td>&lt; Not Exist &gt;</td>
</tr>
</tbody>
</table>

**Figure 22: Route Pattern Configuration for Cisco UBE-PSTN Access (Cont.)**
Figure 23: Route Pattern Configuration for Cisco UBE-PSTN Access-International
Figure 24: Route Pattern Configuration for Cisco UBE-PSTN Access - International (Cont.)

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

![Figure 27: Add New Route pattern to Fax Gateway](image)

Figure 27: Add New Route pattern to Fax Gateway

![Figure 28: Route Pattern Configuration for Fax Gateway](image)

Figure 28: Route Pattern Configuration for Fax Gateway
Figure 29: Route Pattern Configuration for Fax Gateway (Cont.)

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

### Calling Party Transformations
- Use Calling Party's External Phone Number Mask
- Calling Party Transform Mask: <none>
- 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: 44144529XXXX
- 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: 

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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
boot-start-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

multilink bundle-name authenticated

isdn switch-type primary-qsig

voice rtp send-recev

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 CISCO2851 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-nce
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-nce
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
Acronyms

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definitions</th>
</tr>
</thead>
<tbody>
<tr>
<td>CPE</td>
<td>Customer Premise Equipment</td>
</tr>
<tr>
<td>Cisco UBE</td>
<td>Cisco Unified Border Element</td>
</tr>
<tr>
<td>Cisco UCM</td>
<td>Cisco Unified Communications Manager</td>
</tr>
<tr>
<td>MTP</td>
<td>Media Termination Point</td>
</tr>
<tr>
<td>POP</td>
<td>Point of Presence</td>
</tr>
<tr>
<td>PSTN</td>
<td>Public Switched Telephone Network</td>
</tr>
<tr>
<td>ESBC</td>
<td>Enterprise Session Border Controller</td>
</tr>
<tr>
<td>SCCP</td>
<td>Skinny Client Control Protocol</td>
</tr>
<tr>
<td>SIP</td>
<td>Session Initiation Protocol</td>
</tr>
</tbody>
</table>
Important Information

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