Connecting Cisco Unified Communication Manager [v11.5.1] to JT Global SIP Trunks via Cisco Unified Border Element v12.0.0 [IOS-XE 16.06.01]

May 14, 2018
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Introduction

Service Providers today, such as JT Global, 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.

JT Global 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 JT Global network, Cisco Unified Border Element (Cisco UBE) ISR 4331/K9 running IOS 16.06.01 can be used. The Cisco Unified Border Element 16.06.01 provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 11.5.1 connected to JT Global 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 JT Global 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 4331/K9 [IOS – 16.06.01] for connectivity to JT Global SIP Trunking service. The deployment model covered in this application note is CPE (Cisco UCM) to PSTN (JT Global) via Cisco UBE v12.0 [IOS-XE] 16.06.01.

- Testing was performed in accordance to JT Global 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 JT Global 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 JT Global SIP Trunking network.
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. JT Global 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 UBE and Cisco UCM is TCP and to JT Global 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>TCP with RTP</td>
</tr>
<tr>
<td>Transport from Cisco UBE to JT Global</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>NO</td>
</tr>
</tbody>
</table>
System Components

Hardware Requirements
- Cisco UBE on Cisco ISR 4331/K9 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: 12.0.0 running IOS-XE 16.06.01
- CUCM UCOS 11.5.1.12900-21 for 1 Publisher and 2 Subscriber
- Cisco IOS v15.1(4)M5 for the fax gateway

Features

Features Supported
- Incoming and outgoing calls using G711alaw 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 (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
- JT Global does not support Fax at Super G3 Speed
- JT Global does not support T.38 Fax
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.
- JT Global does not support SG3 Fax. Fax test cases are executed only on G3.
- Transcoder has been enabled on Cisco UBE to support mid call codec renegotiation for Conference and transfer scenarios. Please refer the Cisco UBE configuration section for more details.
- For anonymous call test case, JT Global suggested to prefix +44153414100 with DID for the anonymous call
- While doing conference and transfer scenario, Mid-call-codec-renegotiation was not happening. Call established with PSTN 1 with G729 and then PBX initiate conference to PSTN 2 with G729. When complete the conference on PBX. Cisco UCM sends re-INVITE with G711alaw only on SDP for codec re negotiation but JT Global respond back with 200 OK/SDP contains only G729 due to this condition the CUBE disconnected the call
- For PRACK test case, SBC of JT Global strips the ‘100rel’ from the 183 Session Progress message.
- While JT Global was sending CANCEL message towards Cisco UCM, Request URI of CANCEL contains “CANCEL sip:+JESIPSRVXXXXX@192.XX.XX:5060 SIP/2.0” for that Cisco UBE has disconnected the call. To resolve this issue the SIP profile has been used. Please refer the Cisco UBE configuration section for more details.
- Tested with E.164 format for all 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
 ip address 192.xx.xx.xx 255.255.255.224
 negotiation auto
 redundancy rii 2
 redundancy group 1 ip 192.xx.xx.xx exclusive
!
interface GigabitEthernet0/0/1
 ip address 10.80.18.38 255.255.255.0
 negotiation auto
 redundancy rii 1
 redundancy group 1 ip 10.80.18.40 exclusive
!
interface GigabitEthernet0/0/2
 ip address 1.1.1.5 255.255.255.252
 negotiation auto
!
interface Service-Engine0/4/0
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 negotiation auto
Cisco UBE 2:

interface GigabitEthernet0/0/0
  ip address 192.xx.xx.xx 255.255.255.224
  speed 1000
  no negotiation auto
  redundancy rii 2
  redundancy group 1 ip 192.xx.xx.xx exclusive
!
interface GigabitEthernet0/0/1
  ip address 10.80.18.39 255.255.255.0
  negotiation auto
  redundancy rii 1
  redundancy group 1 ip 10.80.18.40 exclusive
!
interface GigabitEthernet0/0/2
  ip address 1.1.1.6 255.255.255.252
  negotiation auto
!
interface Service-Engine0/4/0
!
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
  mode border-element license capacity 200
  allow-connections sip to sip
  redundancy-group 1
  fax protocol pass-through g711alaw
  sip
    bind control source-interface GigabitEthernet0/0/0
    bind media source-interface GigabitEthernet0/0/0
    rel1xx supported "rel100"
    session refresh
    header-passing
    asserted-id pai
    privacy pstn
    midcall-signaling passthru media-change
    early-offer forced
    privacy-policy passthru
    g729 annexb-all
    sip-profiles inbound
    sip-profiles 222 inbound
```
## 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 messages</td>
</tr>
<tr>
<td>midcall-signaling passtrhru media-change</td>
<td>Pass-through media change method optimizes or consumes mid-call, media-related signaling within the call</td>
</tr>
<tr>
<td>sip-profiles inbound</td>
<td>Enables inbound SIP profiles feature</td>
</tr>
<tr>
<td>sip-profiles 222 inbound</td>
<td>SIP profile can be configured with rules to add, remove, copy, or modify the SIP, Session Description Protocol (SDP), and peer headers that enter or leave CUBE.</td>
</tr>
<tr>
<td>g729 annexb-all</td>
<td>Specifies that the G.729br8 codec is treated as a superset of G.729r8 and G.729br8 codecs to communicate with Cisco UCM.</td>
</tr>
</tbody>
</table>
Codecs

G711alaw 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 g711alaw
!
voice class codec 2
  codec preference 1 g729r8
  codec preference 2 g711alaw
  codec preference 3 g711ulaw
!
voice class codec 3
  codec preference 1 g711alaw
!

Dial peer

Outbound Dial-peer to JT Global:

dial-peer voice 10 voip
description Incoming from CUCM
huntstop
session protocol sipv2
incoming called-number 1T
voice-class codec 3
voice-class sip profiles 1
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
no fax-relay sg3-to-g3
fax rate disable
fax protocol pass-through g711alaw
no vad
!
dial-peer voice 20 voip
description Outgoing to JT_Global
translation-profile outgoing e164
huntstop
destination-pattern .T
session protocol sipv2
session target ipv4:212.9.11.194
voice-class codec 3
voice-class sip options-ping 60
voice-class sip profiles 1
no voice-class sip pass-thru headers
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
no fax-relay sg3-to-g3
fax rate disable
fax protocol pass-through g711alaw
no vad

**Inbound Dial-peer from JT Global:**

dial-peer voice 30 voip
description Incoming from JT_Global
huntstop
session protocol sipv2
incoming called-number +44
voice-class codec 3
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
no fax-relay sg3-to-g3
fax rate disable
fax protocol pass-through g711alaw
no vad
!

no dial-peer voice 40 voip
description Outgoing to CUCM
huntstop
destination-pattern +44
session protocol sipv2
session target ipv4:10.80.18.2
voice-class codec 3
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
no fax-relay sg3-to-g3
fax rate disable
fax protocol pass-through g711alaw
no vad
Configuration example

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

**Active Cisco UBE:**

```plaintext
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 JTCertCube10
!
boot-start-marker
boot-end-marker
!
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
no aaa new-model
no ip domain lookup
!
subscriber templating
```

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multilink bundle-name authenticated
!
crypto pki trustpoint TP-self-signed-2930804041
  enrollment selfsigned
  subject-name cn=IOS-Self-Signed-Certificate-2930804041
  revocation-check none
  rsakeypair TP-self-signed-2930804041
!
!
crypto pki certificate chain TP-self-signed-2930804041
!
!
!
voice service voip
  no ip address trusted authenticate
  mode border-element license capacity 200
  allow-connections sip to sip
  redundancy-group 1
  fax protocol pass-through g711alaw
  sip
    bind control source-interface GigabitEthernet0/0/0
    bind media source-interface GigabitEthernet0/0/0
    rel1xx supported "rel100"
    session refresh
    header-passing
    asserted-id pai
    privacy pstn
    midcall-signaling passthru media-change
early-offer forced
privacy-policy passthru
g729 annexb-all
sip-profiles inbound
sip-profiles 222 inbound

voice class codec 1
codec preference 1 g729r8
codec preference 2 g711alaw

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

voice class codec 3
codec preference 1 g711alaw

voice class sip-profiles 222
request CANCEL sip-header To copy "sip:(.*)@u01"
request CANCEL sip-header SIP-Req-URI modify ".*@(.*)" "CANCEL sip:u01@1"
request ACK sip-header To copy "sip:(.*)@u03"
request ACK sip-header SIP-Req-URI modify ".*@(.*)" "ACK sip:u03@1"

voice class sip-profiles 1
request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>
"<sip:+44153471\1@2>"
voice translation-rule 10

rule 1 /\(\^1\...........\$\)/ /+\1/
rule 2 /\(\^...........\$\)/ /+44\1/
rule 3 /\(\^0\...........\$\)/ /\1/
rule 4 /\(\^441534141001\...........\$\)/ /+\1/

voice translation-profile e164

translate called 10

voice-card 0/4
dsp services dspfarm
no watchdog

license udi pid ISR4331/K9 sn FDO2876767
diagnostic bootup level minimal
spanning-tree extend system-id

redundancy
mode none
application redundancy
group 1
name voice-b2bha
timers delay 30 reload 60
control GigabitEthernet0/0/2 protocol 1
data GigabitEthernet0/0/2
track 1 shutdown
track 2 shutdown

track 1 interface GigabitEthernet0/0/1 line-protocol
track 2 interface GigabitEthernet0/0/0 line-protocol

interface GigabitEthernet0/0/0
  ip address 192.xx.xx.xx 255.255.255.224
  speed 1000
  no negotiation auto
  redundancy rii 2
  redundancy group 1 ip 192.xx.xx.xx exclusive

interface GigabitEthernet0/0/1
  ip address 10.80.18.39 255.255.255.0
  negotiation auto
  redundancy rii 1
  redundancy group 1 ip 10.80.18.40 exclusive

interface GigabitEthernet0/0/2
  ip address 1.1.1.6 255.255.255.252
  negotiation auto

interface Service-Engine0/4/0

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 http client source-interface GigabitEthernet0/0/0

ip tftp source-interface GigabitEthernet0/0/0

ip route 0.0.0.0 0.0.0.0 10.64.1.1

ip route 0.0.0.0 0.0.0.0 192.xx.xx.xx

ip route 10.64.0.0 255.255.0.0 10.80.18.1

ip route 172.16.24.0 255.255.248.0 10.80.18.1

!

ip 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

!

dspfarm profile 111 transcode

   codec g729abr8

   codec g729ar8

   codec g711alaw
codec g711ulaw
codec g729r8
codec g729br8
maximum sessions 10
associate application CUBE
!
dial-peer voice 10 voip
description Incoming from CUCM
huntstop
session protocol sipv2
incoming called-number 1T
voice-class codec 3
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-npe
fax-relay ecm disable
no fax-relay sg3-to-g3
fax rate disable
fax protocol pass-through g711alaw
no vad
!
dial-peer voice 30 voip
description Incoming from JT_Global
huntstop
session protocol sipv2
incoming called-number +44
voice-class codec 3
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
no fax-relay sg3-to-g3
fax rate disable
fax protocol pass-through g711alaw
no vad
!
dial-peer voice 40 voip
description Outgoing to CUCM
huntstop
destination-pattern +44
session protocol sipv2
session target ipv4:10.80.18.2
voice-class codec 3
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
no fax-relay sg3-to-g3
fax rate disable
fax protocol pass-through g711alaw
no vad
!
dial-peer voice 118 voip
description Operator number
destination-pattern 118534
session protocol sipv2
session target ipv4:212.XX.XX.XXX
voice-class codec 3
no voice-class sip pass-thru headers
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-n-te
no vad
!
dial-peer voice 999 voip
description Emergency
huntstop
session protocol sipv2
incoming called-number 999
voice-class codec 3
no voice-class sip pass-thru headers
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-n-te
no vad
!
dial-peer voice 20 voip
description Outgoing to JT_Global
translation-profile outgoing e164
huntstop
destination-pattern .T
session protocol sipv2
session target ipv4:212.xx.xx.xxx
voice-class codec 3
voice-class sip options-ping 60
voice-class sip profiles 1
no voice-class sip pass-thru headers
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
no fax-relay sg3-to-g3
fax rate disable
fax protocol pass-through g711alaw
no vad
!
!
sip-ua
  credentials username JESIPSRV00487 password 7 XXXXXXXXX realm 212.XX.XX.XXX
  authentication username JESIPSRV00487 password 7 XXXXXXXXX
  registrar ipv4:212.XX.XX.XXX expires 3600
!
!
line con 0
  transport input none
  stopbits 1
line aux 0
  stopbits 1
line vty 0 4
  password 7 15060E07327B3175783D
  login local
  transport input telnet
!
end
Standby Cisco UBE:

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 JTCertCube9

!
boot-start-marker
boot-end-marker
!
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
no aaa new-model
no ip domain lookup
subscriber templating
multilink bundle-name authenticated
!
crypto pki trustpoint TP-self-signed-3793611302
  enrollment selfsigned
subject-name cn=IOS-Self-Signed-Certificate-3793611302
revocation-check none
rsakeypair TP-self-signed-3793611302

! crypto pki certificate chain TP-self-signed-3793611302
!

voice service voip
no ip address trusted authenticate
mode border-element license capacity 200
allow-connections sip to sip
redundancy-group 1
fax protocol pass-through g711alaw
sip
  bind control source-interface GigabitEthernet0/0/0
  bind media source-interface GigabitEthernet0/0/0
  rel1xx supported "rel100"
  session refresh
  header-passing
  asserted-id pai
  privacy pstn
  midcall-signaling passthru media-change
  early-offer forced
  privacy-policy passthru
  g729 annexb-all
  sip-profiles inbound
  sip-profiles 222 inbound
!

voice class codec 1
codec preference 1 g729r8
codec preference 2 g711alaw
!
voice class codec 2
codec preference 1 g729r8
codec preference 2 g711alaw
codec preference 3 g711ulaw
!
voice class codec 3
codec preference 1 g711alaw
!
!
voice class sip-profiles 222
request CANCEL sip-header To copy "sip:(.*)" u01
request CANCEL sip-header SIP-Req-URI modify ".*@(\.*)" "CANCEL sip:\u01\l"
request ACK sip-header To copy "sip:(\.*)@" u03
request ACK sip-header SIP-Req-URI modify ".*@(\.*)" "ACK sip:\u03\l"
!
voice class sip-profiles 1
request INVITE sip-header Diversion modify "<sip:(\.*)@(\.*)>"
"<sip:+44153471\l@2>"
!
!
voice translation-rule 10
rule 1 /\(^1............\$\)/ +\1/
rule 2 /\(^.........\$\)/ +44\1/
rule 3 /\(^0.........\$\)/ \1/
rule 4 /\(^441534141001............\$\)/ +\1/
!
voice translation-profile e164
translate called 10
voice-card 0/4
dsp services dspfarm
no watchdog

license udi pid ISR4331/K9 sn FDO213777G
diagnostic bootup level minimal
spanning-tree extend system-id

redundancy
mode none
application redundancy
group 1
  name voice-b2bha
  timers delay 30 reload 60
  control GigabitEthernet0/0/2 protocol 1
  data GigabitEthernet0/0/2
  track 1 shutdown
  track 2 shutdown

interface GigabitEthernet0/0/0
  ip address 192.xx.xx.xx 255.255.255.224
  negotiation auto
  redundancy rii 2
  redundancy group 1 ip 192.65.79.92 exclusive
interface GigabitEthernet0/0/1
ip address 10.80.18.38 255.255.255.0
negotiation auto
redundancy rii 1
redundancy group 1 ip 10.80.18.40 exclusive

interface GigabitEthernet0/0/2
ip address 1.1.1.5 255.255.255.252
negotiation auto

interface Service-Engine0/4/0

interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
negotiation auto

ip forward-protocol nd
ip http server
ip http authentication local
ip http secure-server
ip http client source-interface GigabitEthernet0/0/0
ip route 0.0.0.0 0.0.0.0 192.xx.xx.xx
ip route 10.64.0.0 255.255.0.0 10.80.18.1
ip route 10.80.0.0 255.255.0.0 10.80.18.1
ip route 172.16.24.0 255.255.248.0 10.80.18.1

ip 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
!
dspfarm profile 111 transcode
  codec g729abr8
  codec g729ar8
  codec g711alaw
  codec g711ulaw
  codec g729r8
  codec g729br8
  maximum sessions 10
  associate application CUBE
!
dial-peer voice 10 voip
  description Incoming from CUCM
  huntstop
  session protocol sipv2
  incoming called-number 1T
  voice-class codec 3
  voice-class sip profiles 1
  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
no fax-relay sg3-to-g3
fax rate disable
fax protocol pass-through g711alaw
no vad
!
dial-peer voice 20 voip
description Outgoing to JT_Global
translation-profile outgoing e164
huntstop
destination-pattern .T
session protocol sipv2
session target ipv4:212.XX.XX.XXX
voice-class codec 3
voice-class sip options-ping 60
voice-class sip profiles 1
no voice-class sip pass-thru headers
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
no fax-relay sg3-to-g3
fax rate disable
fax protocol pass-through g711alaw
no vad
!
dial-peer voice 30 voip
description Incoming from JT_Global
huntstop
session protocol sipv2
incoming called-number +44
voice-class codec 3
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
no fax-relay sg3-to-g3
fax rate disable
fax protocol pass-through g711alaw
no vad
!
dial-peer voice 40 voip
  description Outgoing to CUCM
  huntstop
  destination-pattern +44
  session protocol sipv2
  session target ipv4:10.80.18.2
  voice-class codec 3
  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
no fax-relay sg3-to-g3
fax rate disable
fax protocol pass-through g711alaw
no vad

! 

dial-peer voice 118 voip 
description Operator number 
destination-pattern 118534 
session protocol sipv2 
session target ipv4:212.xx.xx.xxx 
voice-class codec 3 
no voice-class sip pass-thru headers 
voice-class sip bind control source-interface GigabitEthernet0/0/0 
voice-class sip bind media source-interface GigabitEthernet0/0/0 
dtmf-relay rtp-nre 
no vad 
! 
dial-peer voice 999 voip 
description Emergency 
huntstop 
session protocol sipv2 
incoming called-number 999 
voice-class codec 3 
no voice-class sip pass-thru headers 
voice-class sip bind control source-interface GigabitEthernet0/0/1 
voice-class sip bind media source-interface GigabitEthernet0/0/1 
dtmf-relay rtp-nre 
no vad 
! 
! 

sip-ua 
credentials username JESIPSRV00487 password 7 xxxxxxxxxx realm 212.xx.xx.xxx 
authentication username JESIPSRV00487 password 7 xxxxxxxxxx
registrar ipv4:212.xx.xx.xxx expires 60
!
!
line con 0
  exec-timeout 0 0
  password 7 00071A150754
  login
  transport input none
  stopbits 1
line aux 0
  stopbits 1
line vty 0 4
  password 7 05000000
  login local
  transport input telnet
!
ntp server 34.202.215.187
ntp server pool.ntp.org
!
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](image)

**Figure 4: Service Parameters**
<table>
<thead>
<tr>
<th>Clusterwide Parameters (Service)</th>
<th></th>
<th></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>8</td>
<td>8</td>
</tr>
<tr>
<td>Send Multicast MOH in H245 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>Port Received Timer After Call Connection</td>
<td>500</td>
<td>500</td>
</tr>
<tr>
<td>Media Resource Allocation Timer</td>
<td>12</td>
<td>12</td>
</tr>
<tr>
<td>RTP and Transcoder Resource Throttling Percentage</td>
<td>95</td>
<td>95</td>
</tr>
<tr>
<td>Intercaller Capabilities (Mismatch Timer)</td>
<td>1000</td>
<td>1000</td>
</tr>
<tr>
<td>Silence Suppression</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Silence Suppression for Gateways</td>
<td>False</td>
<td>False</td>
</tr>
<tr>
<td>Strip G.729 Annex B (Silence Suppression) from Capabilities</td>
<td>False</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 JT 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 = TCP

![SIP Trunk Security Profile Configuration](image)

**Figure 6:** SIP Trunk Security Profile
SIP Profile

Navigation: Device → Device Settings → SIP Profile

- Name* = JT Global SIP Profile is used as an example
- Description = JT Global SIP Profile is used as an example

Figure 7: SIP Profile
### Parameters used in Phone

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Timer Invite Expires (seconds)</td>
<td>180 *</td>
</tr>
<tr>
<td>Timer Register Delta (seconds)</td>
<td>5</td>
</tr>
<tr>
<td>Timer Register Expires (seconds)</td>
<td>3600</td>
</tr>
<tr>
<td>Timer T1 (msec)</td>
<td>500</td>
</tr>
<tr>
<td>Timer T2 (msec)</td>
<td>4000</td>
</tr>
<tr>
<td>Retry INVITE</td>
<td>6</td>
</tr>
<tr>
<td>Retry Non-INvITE</td>
<td>10</td>
</tr>
<tr>
<td>Start Media Port</td>
<td>16384</td>
</tr>
<tr>
<td>Stop Media Port</td>
<td>12766</td>
</tr>
<tr>
<td>DSCP for Audio Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for Video Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for Audio Portion of Video Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for Telepresence Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Call Pickup URI</td>
<td>x-cisco-serviceuri-pickup</td>
</tr>
<tr>
<td>Call Pickup Group Other URI</td>
<td>x-cisco-serviceuri-pickup</td>
</tr>
<tr>
<td>Call Pickup Group URI</td>
<td>x-cisco-serviceuri-pickup</td>
</tr>
<tr>
<td>Meet Me Service URI</td>
<td>x-cisco-serviceuri-meetme</td>
</tr>
<tr>
<td>User Info</td>
<td>None</td>
</tr>
<tr>
<td>DTMF DB Level</td>
<td>Nominal</td>
</tr>
<tr>
<td>Call Hold Ring Back</td>
<td>Off</td>
</tr>
<tr>
<td>Anonymous Call Block</td>
<td>Off</td>
</tr>
<tr>
<td>Caller ID Blocking</td>
<td>Off</td>
</tr>
<tr>
<td>Do Not Disturb Control</td>
<td>User</td>
</tr>
<tr>
<td>Telnet Level for 7940 and 7960</td>
<td>Disabled</td>
</tr>
<tr>
<td>Resource Priority Namespace</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Timer Keep Alive Expires (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Expires (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Delta (seconds)</td>
<td>5</td>
</tr>
<tr>
<td>Maximum Redirects</td>
<td>70</td>
</tr>
<tr>
<td>Off Hook To First Digit Timer (milliseconds)</td>
<td>15000</td>
</tr>
<tr>
<td>Call Forward URI</td>
<td>x-cisco-serviceuri-cfwdall</td>
</tr>
<tr>
<td>Speed Dial (Abbreviated Dial) URI</td>
<td>x-cisco-serviceuri-abbrdial</td>
</tr>
<tr>
<td>Conference Join Enabled</td>
<td></td>
</tr>
<tr>
<td>RFC 2543 Hold</td>
<td></td>
</tr>
<tr>
<td>Semi Attended Transfer</td>
<td></td>
</tr>
<tr>
<td>Enable VAD</td>
<td></td>
</tr>
<tr>
<td>Stutter Message Waiting</td>
<td></td>
</tr>
<tr>
<td>MLPP User Authorization</td>
<td></td>
</tr>
</tbody>
</table>

**Figure 8: SIP Profile (Cont.)**
**Normalization Script**

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

**Incoming Requests FROM URI Settings**

<table>
<thead>
<tr>
<th>Caller ID DN</th>
<th>Caller Name</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Trunk Specific Configuration**

- **SIP RelLXX 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)

**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

---

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

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

**Figure 10: Add New Trunk to Cisco UBE**

**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

Figure 13: SIP Trunk to Cisco UBE (Cont.)
- 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 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
Figure 22: Route Pattern Configuration for Cisco UBE-PSTN Access (Cont.)
Figure 23: Route Pattern Configuration for Cisco UBE-PSTN Access - Anonymous Call
Figure 24: Route Pattern Configuration for Cisco UBE-PSTN Access - Anonymous Call (Cont.)
Route Pattern for Fax Gateway:

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

![Add New Route pattern to Fax Gateway](image-url)
Figure 26: Route Pattern Configuration for Fax Gateway
**Figure 27: Route Pattern Configuration for Fax Gateway (Cont.)**

---

**Called Party Transformations**

- Discord 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**

<table>
<thead>
<tr>
<th>Network Service Protocol</th>
<th>Service Parameter Name</th>
<th>Service Parameter Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>-- Not Selected --</td>
<td>&lt; Not Exisit &gt;</td>
<td></td>
</tr>
</tbody>
</table>

---

Save  Delete  Copy  Add New

* indicates required item.
Configuring Cisco Voice Gateway for Fax

Global Settings

```plaintext
voice service voip
  allow-connections sip to sip
no supplementary-service sip handle-replaces
redirect ip2ip
fax protocol pass-through g711alaw
no fax-relay sg3-to-g3
sip
  rel1xx disable
  midcall-signaling passthru
  g729 annexb-all

!
```

Codecs

G711alaw and G729 voice codecs are used for this testing. Codec preferences used to change according to the test plan description.

```plaintext
voice class codec 1
  codec preference 1 g711alaw
!
voice class codec 2
  codec preference 1 g729r8
  codec preference 2 g711alaw
```
Dial peer

**Outbound Dial-peer to Cisco UCM:**

dial-peer voice 800 voip
description Outbound Gateway to CUCM
service session
destination-pattern 9T
session protocol sipv2
session target ipv4:10.80.18.2
session transport udp
voice-class codec 1
dtmf-relay rtp-npe
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711alaw
no vad

**Inbound Dial-peer from Cisco UCM:**
dial-peer voice 900 voip
description Inbound from JT Fax
service session
session protocol sipv2
session transport udp
incoming called-number 621.
voice-class codec 1
dtmf-relay rtp-npe
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711alaw
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 6215
  no digit-strip
  port 0/0/1
  forward-digits all

voice-port 0/0/1
  no echo-cancel enable
  no vad
cptone IN
  station-id name JT Global Fax
  station-id number +441534716215
caller-id enable

Configuration example

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

cme.in.tekvizion.com>en
Password:
cme.in.tekvizion.com#sh run

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
boot-end-marker
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-receive
!
voice service pots
!
voice service voip
  allow-connections sip to sip
  no supplementary-service sip handle-replaces
  redirect ip2ip
  fax protocol pass-through g711alaw
  no fax-relay sg3-to-g3
  sip
  rel1xx disable
  midcall-signaling passthru
  g729 annexb-all
!
!
voice class codec 1
  codec preference 1 g711alaw
!
voice class codec 2
  codec preference 1 g729r8
  codec preference 2 g711alaw
!
!
!
voice translation-rule 1
  rule 1 /\([^\................\$\)/ \(+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
ip route 0.0.0.0 0.0.0.0 172.16.29.1
!
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
!
voice-port 0/0/1
  no echo-cancel enable
  no vad
cptone IN
station-id name JT Global Fax
station-id number +441534716215
caller-id enable
!
voice-port 0/3/0:15
!
voice-port 0/2/0:15
!
voice-port 0/1/0
!
voice-port 0/1/1
  no echo-cancel enable
  no vad
cptone IN
station-id name JT Global Fax 2
station-id number +441534716212
caller-id enable
!
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
scrp ccm 10.71.3.10 identifier 1 version 7.0
!
scrp ccm group 6
  bind interface GigabitEthernet0/0
  associate ccm 1 priority 1
!
!
dial-peer voice 45 pots
huntstop
service session
destination-pattern 6215
no digit-strip
port 0/0/1
forward-digits all

dial-peer voice 46 pots
service session
destination-pattern 6212
no digit-strip
port 0/1/1
forward-digits all

!  
!
dial-peer voice 800 voip
  description Outbound Gateway to CUCM
  service session
destination-pattern 9T
  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 disable
  fax protocol pass-through g711alaw
  no vad

!  
dial-peer voice 900 voip
  description Inbound from JT Fax
  service session
  session protocol sipv2
session transport udp
incoming called-number 621.
voice-class codec 1
dtmf-relay rtp-nate
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711alaw
no vad
!
!
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 Administrator password tekV1z10n
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

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definitions</th>
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<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>
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</tbody>
</table>

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