Connecting Cisco Unified Communication Manager [v11.5.1] to CenturyLink IQ SIP Trunks via Cisco Unified Border Element v11.5.2 [IOS-XE 16.3.2]

May 3, 2017
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Introduction

Service Providers today, such as CenturyLink, 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.

CenturyLink IQ SIP Trunk 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 CUCM 11.5.1 and CenturyLink network, Cisco Unified Border Element (Cisco UBE) v11.5.2 can be used. The Cisco Unified Border Element provides demarcation, security and inter-working and session control services for CUCM connected to CenturyLink IP network.

This document assumes the reader is knowledgeable with the terminology and configuration of CUCM. Only configuration settings specifically required for CenturyLink interoperability are presented. Feature configuration and most importantly the dial plan are customer specific and need individual approach.

- This application note describes how to configure a CUCM 11.5.1 and Cisco Unified Border Element (Cisco UBE) v11.5.2 for connectivity to CenturyLink IQ SIP Trunking service. The deployment model covered in this application note is CUCM to PSTN via Cisco Unified Border Element v11.5.2 [IOS-XE] 16.3.2.
- Testing was performed in accordance to Cisco 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.
- Testing was performed in accordance to CenturyLink Dual Trunk Test with Fail-over and Load Balancing among the trunks.
- The Cisco Unified Border Element (Cisco UBE) configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between CenturyLink SIP network and CUCM. 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 CUCM to interoperate to CenturyLink SIP Trunking network.
Network Topology

- The network topology includes the CUCM Cluster, Unity Voicemail system, Cisco Fax gateway and 2 Cisco Endpoints. CUCM has a trunk configured to CUBE’s Virtual IP Address. CenturyLink was used as the service provider with Dual SIP trunks 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 CUCM is TCP and to CenturyLink is UDP.

CUCM and CUBE Settings:

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Transport from CUBE to CUCM</td>
<td>TCP with RTP</td>
</tr>
<tr>
<td>Transport from CUBE to CenturyLink</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>
<tr>
<td>G729 Conference Support</td>
<td>YES</td>
</tr>
</tbody>
</table>
System Components

Hardware Requirements
- Cisco UBE on Cisco ISR 4431 router
- CUCM cluster on UCS, 1 Publisher node and 2 Subscriber nodes
- Cisco 2851 with FXS ports and Analog Fax machine
- Generic Cisco IP-Phones

Software Requirements
- CUBE-Version: 11.5.2 running IOS-XE 16.3.2
- CUCM UCOS for 1 Publisher and 2 Subscriber
- Cisco IOS v12.4 for the fax gateway

Features

Features Supported
- Incoming and outgoing off-net calls using G711 and G729
- Cisco Multi-Tenant with Failover and Load Balancing in SIP Trunks.
- International Calls and digit manipulations
- Call Conference with G729 support
- 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 Pass-through
- IP-PBX Calling number privacy
- High Availability

Features Not Supported
- Blind Call transfer
Caveats

➢ As of writing this application note, CenturyLink supports G.711 pass-through for faxing on IQ SIP Trunk. Currently T.38 is not supported. The Fax protocol is indicated and set at the time of implementation of the service and CenturyLink recommends that customers use our CenturyLink trial process to confirm Fax capability prior to full deployment.

➢ Caller ID updates are not observed on attended call transfer scenarios.

➢ Testing is done with only one IP PBX.

➢ Workaround is done for SIP header manipulations for Register, Call Forward and Call Transfer.

➢ The CUBE HA tested here is layer 2 box to box CUBE redundancy.
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
**CUBE 1:**

interface GigabitEthernet0/0/0
 ip address 10.64.4.19 255.255.0.0
 media-type rj45
 negotiation auto

interface GigabitEthernet0/0/1
 no ip address
 shutdown
 media-type rj45
 negotiation auto

interface GigabitEthernet0/0/2
 ip address 10.80.22.74 255.255.255.0
 media-type rj45
 negotiation auto
 redundancy rii 1
 redundancy group 1 ip 10.80.22.100 exclusive

interface GigabitEthernet0/0/3
 description Wan Interface
 ip address 192.65.79.140 255.255.255.128
 media-type rj45
 negotiation auto
 redundancy rii 2
 redundancy group 1 ip 192.65.79.155 exclusive

!
CUBE 2:

interface GigabitEthernet0/0/0
ip address 10.64.4.20 255.255.0.0
media-type rj45
negotiation auto
!
interface GigabitEthernet0/0/1
no ip address
shutdown
media-type rj45
negotiation auto
!
interface GigabitEthernet0/0/2
description WAN
ip address 10.80.22.75 255.255.255.0
media-type rj45
negotiation auto
redundancy rii 1
redundancy group 1 ip 10.80.22.100 exclusive
!
interface GigabitEthernet0/0/3
description Wan Interface
ip address 192.65.79.141 255.255.255.128
media-type rj45
negotiation auto
redundancy rii 2
redundancy group 1 ip 192.65.79.155 exclusive
!
Global Cisco UBE settings

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

```
voice service voip
  ip address trusted list
no ip address trusted authenticate
rtcp keepalive
address-hiding
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 1
no supplementary-service sip moved-temporarily
no supplementary-service sip refer
supplementary-service media-renegotiate
redirect ip2ip
fax protocol pass-through g711ulaw
sip
  bind control source-interface GigabitEthernet0/0/2
  bind media source-interface GigabitEthernet0/0/2
session refresh
  asserted-id pai
early-offer forced
midcall-signaling passthru
  pass-thru subscribe-notify-events all
  pass-thru content unsupp
sip-profiles inbound
sip-profiles 1
```

**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 1</td>
<td>Enable High Availability for the VoIP service</td>
</tr>
<tr>
<td>rtcp keepalive</td>
<td>Enables the CUBE to send rtcp keepalive packets for the session keepalive</td>
</tr>
</tbody>
</table>
fax protocol | Specifies the fax protocol
--- | ---
early-offer forced | Enables SIP Delayed-Offer to Early-Offer globally
asserted-id | Specifies the privacy header in the outgoing SIP requests and response messages

**Codecs**

G729 is used primarily towards CenturyLink until specified otherwise.

voice class codec 1

codec preference 1 g711ulaw
codec preference 2 g729r8

voice class codec 2

codec preference 1 g729r8
codec preference 2 g711ulaw

**Dial peer**

**Outbound Dial-peer to CenturyLink:**

dial-peer voice 100 voip
description Outbound peer to CenturyLink Primary
translation-profile outgoing ctl
destination-pattern .T
session protocol sipv2
session target ipv4:<Trunk1_IP>:5100
voice-class codec 1
voice-class sip early-offer forced
voice-class sip profiles 100
voice-class sip tenant 10
voice-class sip bind control source-interface GigabitEthernet0/0/3
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-nte
fax protocol pass-through g711ulaw
no vad!

dial-peer voice 102 voip
description Outbound peer to CenturyLink Secondary
translation-profile outgoing ctl
destination-pattern .T
session protocol sipv2
session target ipv4:<Trunk2_IP>:5100
voice-class codec 1
voice-class sip early-offer forced
voice-class sip profiles 200
voice-class sip tenant 20
voice-class sip bind control source-interface GigabitEthernet0/0/3
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-npe
fax protocol pass-through g711ulaw
no vad

**Inbound Dial-peer from CenturyLink:**

dial-peer voice 111 voip
description Inbound peer match FROM CenturyLink
session protocol sipv2
session transport udp
incoming called-number 612356599.
voice-class codec 1
voice-class sip profiles 100
voice-class sip tenant 10
voice-class sip bind control source-interface GigabitEthernet0/0/3
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-npe
fax protocol pass-through g711ulaw
no vad
Outbound Dial-peer to CUCM:

! 

dial-peer voice 200 voip 
description Outbound peer to CUCM PUB 
translation-profile outgoing cucm 
destination-pattern 612356599. 
session protocol sipv2 
session target ipv4:10.80.20.2:5060 
session transport tcp 
voice-class codec 1 
voice-class sip bind control source-interface GigabitEthernet0/0/2 
voice-class sip bind media source-interface GigabitEthernet0/0/2 
dtmf-relay rtp-nte 
fax protocol pass-through g711ulaw 
no vad!

dial-peer voice 202 voip 
description Outbound peer to CUCM SUB 
translation-profile outgoing cucm 
destination-pattern 612356599. 
session protocol sipv2 
session target ipv4:10.80.20.3:5060 
session transport tcp 
voice-class codec 1 
voice-class sip bind control source-interface GigabitEthernet0/0/2 
voice-class sip bind media source-interface GigabitEthernet0/0/2 
dtmf-relay rtp-nte 
fax protocol pass-through g711ulaw 
no vad
Inbound Dial-peer from CUCM:

dial-peer voice 201 voip
  description Inbound peer match FROM CUCM
  session protocol sipv2
  session transport tcp
  incoming called-number .T
  voice-class codec 1
  voice-class sip bind control source-interface GigabitEthernet0/0/2
  voice-class sip bind media source-interface GigabitEthernet0/0/2
  dtmf-relay rtp-nte
  fax protocol pass-through g711ulaw
  no vad
Configuration example
The following configuration snippet contains a sample configuration of Cisco UBE with all parameters mentioned previously.

Active Cisco UBE:

vrf definition Mgmt-intf

enable password *******
no aaa new-model

ip name-server 10.64.4.10

ip domain name voip.centurylink.com

subscriber templating
multilink bundle-name authenticated

voice service voip
  ip address trusted list
  no ip address trusted authenticate
  rtcp keepalive
  address-hiding
  mode border-element license capacity 20
  allow-connections sip to sip
  redundancy-group 1
  no supplementary-service sip moved-temporarily
  no supplementary-service sip refer
  supplementary-service media-renegotiate
  redirect ip2ip
  fax protocol pass-through g711ulaw
  sip
    bind control source-interface GigabitEthernet0/0/2
    bind media source-interface GigabitEthernet0/0/2
session refresh
asserted-id pai
early-offer forced
midcall-signaling passthru
pass-thru subscribe-notify-events all
pass-thru content unsupp
sip-profiles inbound
sip-profiles 1

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

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

voice class sip-profiles 1
  request REGISTER sip-header From modify "({.*:.}(@.*)")" "\1@voip.centurylink.com>"

  request REGISTER sip-header To modify "({.*:.}(@.*)")" "\1@voip.centurylink.com>"

voice class sip-profiles 100
  request ANY sip-header Cisco-Guid remove
  response ANY sip-header Cisco-Guid remove
  request ANY sip-header User-Agent remove
  response ANY sip-header User-Agent remove
voice class sip-profiles 200

voice class tenant 10
registrar ipv4:<Trunk1_IP>:5100 expires 3600
credentials number 7209254720 username 263160-7209254720 password 7
******************** realm voip.centurylink.com
authentication username 263160-7209254720 password 7 ********************
realm voip.centurylink.com
no remote-party-id
retry invite 2
timers connect 100
sip-server ipv4:<Trunk1_IP>:5100
connection-reuse
bind control source-interface GigabitEthernet0/0/3
bind media source-interface GigabitEthernet0/0/3
!
voice class tenant 20
registrar ipv4:<Trunk2_IP>:5100 expires 3600
credentials number 7209254721 username 263160-7209254721 password 7
******************** realm voip.centurylink.com
authentication username 263160-7209254721 password 7 ********************
realm voip.centurylink.com
no remote-party-id
retry invite 2
timers connect 100
sip-server ipv4:<Trunk2_IP>:5100
connection-reuse
bind control source-interface GigabitEthernet0/0/3
bind media source-interface GigabitEthernet0/0/3
!

!
voice translation-rule 1

voice translation-rule 2
  rule 1 /6123565991/ /5991/
  rule 2 /6123565992/ /5992/
  rule 3 /6123565993/ /5993/

voice translation-rule 3

voice translation-rule 4
  rule 1 /^9\((............)$\)/ /\1/
  rule 2 /^9\((................)$\)/ /\1/
  rule 3 /^9\(...$\)/ /\1/

voice translation-profile ctl
translate calling 3
translate called 4

voice translation-profile cucm
translate calling 1
translate called 2

voice-card 0/1
  no watchdog

license udi pid ISR4431/K9 sn FOC18261KJL
license accept end user agreement
license boot level appxk9
license boot level uck9
license boot level securityk9
!
diagnostic bootup level minimal
spanning-tree extend system-id
!
!
username cisco privilege 15 secret 5 ************************
username user password 0 user
!
redundancy
  mode none
  application redundancy
    group 1
      name voice-b2bha
      priority 100 failover threshold 75
      timers delay 30 reload 60
      control GigabitEthernet0/0/0 protocol 1
      data GigabitEthernet0/0/0
      track 1 shutdown
      track 2 shutdown
!
!
!
vlan internal allocation policy ascending
!
track 1 interface GigabitEthernet0/0/2 line-protocol
!
track 2 interface GigabitEthernet0/0/3 line-protocol
!
!
interface GigabitEthernet0/0/0
  ip address 10.64.4.19 255.255.0.0
  media-type rj45
  negotiation auto
!
interface GigabitEthernet0/0/1
  no ip address
  shutdown
  media-type rj45
  negotiation auto
!
interface GigabitEthernet0/0/2
  description WAN
  ip address 10.80.22.74 255.255.255.0
  media-type rj45
  negotiation auto
  redundancy rii 1
  redundancy group 1 ip 10.80.22.100 exclusive
!
interface GigabitEthernet0/0/3
  description Wan Interface
  ip address 192.65.79.141 255.255.255.128
media-type rj45
negotiation auto
redundancy rii 2
redundancy group 1 ip 192.65.79.155 exclusive
!
interface Service-Engine0/1/0

interface GigabitEthernet0
  vrf forwarding Mgmt-intf
  no ip address
  negotiation auto
!
interface Vlan1
  no ip address
  shutdown

ip forward-protocol nd
ip http server
no ip http secure-server
ip route 0.0.0.0 0.0.0.0 192.65.79.129
ip route 10.64.0.0 255.255.0.0 10.64.1.1
ip route 172.16.0.0 255.255.0.0 10.80.22.1
!
!
!
!
!
!
control-plane

voice-port 0/1/0
captone IN
shutdown
caller-id enable

voice-port 0/1/1

dial-peer voice 100 voip
description Outbound peer to CenturyLink Primary
translation-profile outgoing ctl
destination-pattern .T
session protocol sipv2
session target ipv4:<Trunk1_IP>:5100
voice-class codec 1
voice-class sip early-offer forced
voice-class sip profiles 100
voice-class sip tenant 10
voice-class sip bind control source-interface GigabitEthernet0/0/3
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-nte
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 102 voip
  description Outbound peer to CenturyLink Secondary
  translation-profile outgoing ctl
  destination-pattern .T
  session protocol sipv2
  session target ipv4:<Trunk2_IP>::5100
  voice-class codec 1
  voice-class sip early-offer forced
  voice-class sip profiles 200
  voice-class sip tenant 20
  voice-class sip bind control source-interface GigabitEthernet0/0/3
  voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-nte
fax protocol pass-through g711ulaw
no vad
!

dial-peer voice 111 voip
description Inbound peer match FROM Centurylink
session protocol sipv2
session transport udp
incoming called-number 612356599.
voice-class codec 1
voice-class sip profiles 100
voice-class sip tenant 10
voice-class sip bind control source-interface GigabitEthernet0/0/3
dtmf-relay rtp-nte
fax protocol pass-through g711ulaw
no vad
!

dial-peer voice 200 voip
description Outbound peer to CUCM PUB
translation-profile outgoing cucm
destination-pattern 612356599.
session protocol sipv2
session target ipv4:10.80.20.2:5060
session transport tcp
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/2
dtmf-relay rtp-nte
fax protocol pass-through g711ulaw
no vad
dial-peer voice 201 voip
description Inbound peer match FROM CUCM
session protocol sipv2
session transport tcp
incoming called-number .T
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
dtmf-relay rtp-nre
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 202 voip
description Outbound peer to CUCM SUB
translation-profile outgoing cucm
destination-pattern 612356599.
session protocol sipv2
session target ipv4:10.80.20.3:5060
session transport tcp
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
dtmf-relay rtp-nre
fax protocol pass-through g711ulaw
no vad!
!
sip-ua
no remote-party-id
retry invite 1
timers options 1000
connection-reuse

line con 0
exec-timeout 0 0
logging synchronous
stopbits 1
line aux 0
privilege level 15
stopbits 1
line vty 0
exec-timeout 0 0
password *********
no activation-character
logging synchronous
login
transport preferred ssh
transport input telnet ssh
line vty 1 4
exec-timeout 0 0
password ************
logging synchronous
login
transport input telnet ssh

ntp server 10.10.10.5


Standby Cisco UBE:

Current configuration: 9385 bytes

Last configuration change at 15:17:17 UTC Mon Mar 13 2017

version 16.3
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core

hostname ISR4KR2

boot-start-marker
boot system flash isr4400-universalk9.16.03.02.SPA.bin
boot-end-marker

vrf definition Mgmt-intf

address-family ipv4
exit-address-family

address-family ipv6
exit-address-family
! logging buffered 9999999
no logging console
enable password ***********
!
no aaa new-model
!
ipc zone default
association 1
  no shutdown
!
!
!
!
!
!
!
!
!
!
!
!
!
ip name-server 10.64.4.10

no ip domain lookup
!
!
!
!
subscriber templating

multilink bundle-name authenticated

c ts logging verbose

voice rtp send-receive

voice service voip
  ip address trusted list
  no ip address trusted authenticate
  rtcp keepalive
  address-hiding
  mode border-element license capacity 20
  allow-connections sip to sip
  redundancy-group 1
  no supplementary-service sip moved-temporarily
  no supplementary-service sip refer
  supplementary-service media-renegotiate
  redirect ip2ip
fax protocol pass-through g711ulaw
sip
bind control source-interface GigabitEthernet0/0/2
bind media source-interface GigabitEthernet0/0/2
session refresh
asserted-id pai
early-offer forced
midcall-signaling passthru
pass-thru subscribe-notify-events all
pass-thru content unsupp
sip-profiles inbound
sip-profiles 1

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

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

voice class sip-profiles 1
request REGISTER sip-header From modify "(<.*:.*)(@.*)>"
"\1@voip.centurylink.com>"
request REGISTER sip-header To modify "(<.*:.*)(@.*)>"
"\1@voip.centurylink.com>"

voice class sip-profiles 100
request ANY sip-header Cisco-Guid remove
response ANY sip-header Cisco-Guid remove
request ANY sip-header User-Agent remove
response ANY sip-header User-Agent remove

request INVITE sip-header Diversion modify "(<.*::*)(@.*)"
"<sip:7209254720@voip.centurylink.com>"

request INVITE sip-header From modify "(<.*::*)(@.*)"
"\1@voip.centurylink.com>"

request INVITE sip-header P-Asserted-Identity modify "(<.*::*)(@.*)"
"<sip:7209254720@voip.centurylink.com>"

request REINVITE sip-header Diversion modify "(<.*::*)(@.*)"
"<sip:7209254720@voip.centurylink.com>"

request REINVITE sip-header From modify "(<.*::*)(@.*)"
"\1@voip.centurylink.com>"

request REINVITE sip-header P-Asserted-Identity modify "(<.*::*)(@.*)"
"<sip:7209254720@voip.centurylink.com>"

!

voice class sip-profiles 200

request ANY sip-header Cisco-Guid remove
response ANY sip-header Cisco-Guid remove
request ANY sip-header User-Agent remove
response ANY sip-header User-Agent remove

request INVITE sip-header Diversion modify "(<.*::*)(@.*)"
"<sip:7209254721@voip.centurylink.com>"

request INVITE sip-header From modify "(<.*::*)(@.*)"
"\1@voip.centurylink.com>"

request INVITE sip-header P-Asserted-Identity modify "(<.*::*)(@.*)"
"<sip:7209254721@voip.centurylink.com>"

request REINVITE sip-header Diversion modify "(<.*::*)(@.*)"
"<sip:7209254721@voip.centurylink.com>"

request REINVITE sip-header From modify "(<.*::*)(@.*)"
"\1@voip.centurylink.com>"
request REINVITE sip-header P-Asserted-Identity modify "(.*::*)(@.*)" 
"<sip:7209254721@voip.centurylink.com>"

!

voice class tenant 10

  registrar ipv4:<Trunk1_IP>:5100 expires 3600
  credentials number 7209254720 username 263160-7209254720 password 7
  ******************* realm voip.centurylink.com
  authentication username 263160-7209254720 password 7 *******************
  realm voip.centurylink.com

  no remote-party-id
  retry invite 2
  timers connect 100
  sip-server ipv4:<Trunk1_IP>:5100
  connection-reuse
  bind control source-interface GigabitEthernet0/0/3
  bind media source-interface GigabitEthernet0/0/3

!

voice class tenant 20

  registrar ipv4:<Trunk1_IP>:5100 expires 3600
  credentials number 7209254721 username 263160-7209254721 password 7
  ******************* realm voip.centurylink.com
  authentication username 263160-7209254721 password 7 *******************
  realm voip.centurylink.com

  no remote-party-id
  retry invite 2
  timers connect 100
  sip-server ipv4:<Trunk1_IP>:5100
  connection-reuse
  bind control source-interface GigabitEthernet0/0/3
bind media source-interface GigabitEthernet0/0/3
!

voice translation-rule 1
!

voice translation-rule 2
  rule 1 /6123565991/ /5991/
  rule 2 /6123565992/ /5992/
  rule 3 /6123565993/ /5993/
  rule 4 /6123565994/ /2302/
!

voice translation-rule 3
!

voice translation-rule 4
  rule 1 /^9\(...........\)/ /1/
  rule 2 /^9\(.............\)/ /1/
  rule 3 /^9\(...\)/ /1/
!

voice translation-profile ctl
translate calling 3
translate called 4
!

voice translation-profile cucm
translate calling 1
translate called 2
!
!
license udi pid ISR4431/K9 sn FOC18232988
license boot level appxk9
license boot level uck9
license boot level securityk9
!
diagnostic bootup level minimal
spanning-tree extend system-id
!
!
username cisco privilege 15 secret 5 ****************************
!
redundancy
  mode none
  application redundancy
    group 1
    name voice-b2bha
    control GigabitEthernet0/0/0 protocol 1
    data GigabitEthernet0/0/0
    track 1 shutdown
    track 2 shutdown
!
!
vlan internal allocation policy ascending
!
track 1 interface GigabitEthernet0/0/2 line-protocol
!
track 2 interface GigabitEthernet0/0/3 line-protocol
!
!
interface GigabitEthernet0/0/0
   ip address 10.64.4.20 255.255.0.0
   media-type rj45
   negotiation auto

interface GigabitEthernet0/0/1
   no ip address
   shutdown
   media-type rj45
   negotiation auto

interface GigabitEthernet0/0/2
   ip address 10.80.22.75 255.255.0.0
   media-type rj45
   negotiation auto
   redundancy rii 1
   redundancy group 1 ip 10.80.22.100 exclusive

interface GigabitEthernet0/0/3
   description Wan Interface
   ip address 192.65.79.140 255.255.128
   media-type rj45
   negotiation auto
   redundancy rii 2
   redundancy group 1 ip 192.65.79.155 exclusive

interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
shutdown
negotiation auto
!
interface Vlan1
no ip address
shutdown
!
ip forward-protocol nd
no ip http server
no ip http secure-server
ip route 0.0.0.0 0.0.0.0 192.65.79.129
ip route 10.64.0.0 255.255.0.0 10.64.1.1
ip route 10.80.10.0 255.255.255.0 10.80.22.1
ip route 10.80.22.0 255.255.255.0 10.80.22.1
ip route 172.16.0.0 255.255.0.0 10.80.22.1
!
!
!
!
!
!
!
!
!
!
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

dspfarm profile 1 conference security
  shutdown

dial-peer voice 100 voip
  description Outbound peer to CenturyLink Primary
  translation-profile outgoing ctl
  destination-pattern .T
  session protocol sipv2
  session target ipv4:<Trunk1_IP>:5100
  voice-class codec 1
  voice-class sip early-offer forced
  voice-class sip profiles 100
  voice-class sip tenant 10
  voice-class sip bind control source-interface GigabitEthernet0/0/3
  voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-nte
fax protocol pass-through g711ulaw
no vad!

dial-peer voice 102 voip
description Outbound peer to CenturyLink Secondary
translation-profile outgoing ctl
destination-pattern .T
session protocol sipv2
session target ipv4:<Trunk2_IP>:5100
voice-class codec 1
voice-class sip early-offer forced
voice-class sip profiles 200
voice-class sip tenant 20
voice-class sip bind control source-interface GigabitEthernet0/0/3
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-nte
fax protocol pass-through g711ulaw
no vad!

dial-peer voice 111 voip
description Inbound peer match FROM Centurylink
session protocol sipv2
session transport udp
incoming called-number 612356599.
voice-class codec 1
voice-class sip profiles 100
voice-class sip tenant 10
voice-class sip bind control source-interface GigabitEthernet0/0/3
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-nte
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 200 voip
description Outbound peer to CUCM PUB
translation-profile outgoing cucm
destination-pattern 612356599.
session protocol sipv2
session target ipv4:10.80.20.2:5060
session transport tcp
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
dtmf-relay rtp-nte
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 201 voip
description Inbound peer match FROM CUCM
session protocol sipv2
session transport tcp
incoming called-number .T
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
dtmf-relay rtp-nte
fax protocol pass-through g711ulaw
no vad
no vad
!  

dial-peer voice 202 voip  
description Outbound peer to CUCM SUB  
translation-profile outgoing cucm  
destination-pattern 612356599.  
session protocol sipv2  
session target ipv4:10.80.20.3:5060  
session transport tcp  
voice-class codec 1  
voice-class sip bind control source-interface GigabitEthernet0/0/2  
voice-class sip bind media source-interface GigabitEthernet0/0/2  
dtmf-relay rtp-nre  
fax protocol pass-through g711ulaw  
no vad!  
sip-ua  
no remote-party-id  
retry invite 1  
timers options 1000  
connection-reuse  
!  
!  
line con 0  
stopbits 1  
line aux 0  
stopbits 1  
line vty 0  
session-timeout 90  
exec-timeout 960 0  
password ********
no activation-character
logging synchronous
login
transport preferred ssh
transport input all
stopbits 1
line vty 1 4
   exec-timeout 960 0
   password ********
logging synchronous
login
transport input all
!
!
!
end
Configuring CUCM 11.5 cluster

Trunk configuration from CUCM to the LAN side of the CUBE.

**Trunk configuration**

**Trunk to CUBE**

- Open Cisco Unified CM Administration and navigate to Devices → Trunk → Add New

![Cisco Unified CM Administration](image)

**Figure 3 Add new Trunk in CUCM**

- Click on “Add New” button and select the Trunk Type and Device Protocol as shown below.
Figure 4 Add SIP Trunk Type

- Select ‘Trunk Type’ as SIP Trunk and ‘Protocol’ as SIP and select ‘Next’.
### Device Information

<table>
<thead>
<tr>
<th>Product</th>
<th>SIP Trunk</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Protocol</td>
<td>SIP</td>
</tr>
<tr>
<td>Trunk Service Type</td>
<td>None (Default)</td>
</tr>
<tr>
<td>Device Name</td>
<td>Trunk-to-CUBE</td>
</tr>
<tr>
<td>Description</td>
<td>Trunk-to-CUBE</td>
</tr>
<tr>
<td>Device Pool 3rd</td>
<td>Default</td>
</tr>
<tr>
<td>Common Device Configuration</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Call Classification</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Location</td>
<td>Hub_None</td>
</tr>
<tr>
<td>AAR Group</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Tunneled Protocol</td>
<td>None</td>
</tr>
<tr>
<td>QSIG Variant 6</td>
<td>No Changes</td>
</tr>
<tr>
<td>ASN.1 ROSE OID Encoding</td>
<td>No Changes</td>
</tr>
<tr>
<td>Packet Capture Mode</td>
<td>None</td>
</tr>
<tr>
<td>Packet Capture Duration</td>
<td>0</td>
</tr>
</tbody>
</table>

| Media Termination Point Required | □                  |
| Retry Video Call as Audio       | □                  |
| Path Replacement Support        | □                  |
| Transmit UTF-8 for Calling Party Name | □                  |

---

**Figure 5** SIP Trunk Device name and Device Pool selection
• Please configure the Inbound and Outbound call settings as shown below

![Inbound call Configuration](image)

**Figure 6: Inbound call Configuration**
Configure the Virtual LAN IP address of the CUBE and the port number
Configure the SIP Profile and SIP trunk security profiles as shown below.
The rest of the configuration is all default.
Click Save and Reset after completion.

**Trunk to FAX gateway**
• Open Cisco Unified CM Administration and navigate to Devices → Trunk → Add New

![Cisco Unified CM Administration](image)

Figure 8 Add a new SIP Trunk to FAX gateway

• Click on “Add New” button and select the Trunk Type and Device Protocol as shown below.

![Trunk Configuration](image)

Figure 9 Trunk Type and Device Protocol configuration

• Select ‘Trunk Type’ as SIP Trunk and ‘Protocol’ as SIP and select ‘Next’.
- Figure 10 SIP Trunk Device name and Device Pool selection

- Please configure the Inbound and Outbound call settings as shown below
Figure 11 Inbound call Routing Configuration
Figure 12 Outbound Call Routing and Trunk Security Settings

- Configure IP address of the FAX Gateway and the port number
- Configure the SIP Profile and SIP trunk security profiles as shown below.
- The rest of the configuration is all default.
- Click Save and Reset after completion.

Routing configuration
Route Pattern to CUBE

- Open Cisco Unified CM Administration and navigate to Call Routing → Route/Hunt → Route Pattern
- Click on “Add New” button to add a new route pattern to CUBE

Figure 13 Add New Route Pattern to CUBE
Figure 14 Route Pattern Configuration for CUBE

- Click on “Save” button to complete the route pattern configuration

Route Pattern to Fax Gateway
• Open Cisco Unified CM Administration and navigate to Call Routing → Route/Hunt → Route Pattern
• Click on “Add New” button to add a new route pattern to Fax Gateway

Figure 15 Add New Route pattern to Fax Gateway
Figure 16 Route Pattern Configuration to Fax Gateway
SIP Trunk Security Profile

- We have used the default Non-Secure SIP Trunk Security Profile.

Figure 17 SIP Trunk Security Profile

Fax Gateway Configuration
The following configuration snippet contains a sample configuration of Cisco FAX Gateway.

Current configuration : 9949 bytes

!  
! Last configuration change at 16:34:05 IST Wed Apr 19 2017 by cisco
version 15.1

service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption

!  
hostname faxgateway

!  
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
  address-hiding
  allow-connections sip to sip
  no supplementary-service sip handle-replaces
  redirect ip2ip
  fax protocol pass-through g711ulaw
  no fax-relay sg3-to-g3
sip
midcall-signaling passthru
g729 annexb-all
!
voice class codec 3
  codec preference 1 g711ulaw
!
voice class codec 2
  codec preference 1 g711alaw
  codec preference 2 g729br8
  codec preference 3 g729r8
!
voice class codec 1
  codec preference 1 g729r8
  codec preference 2 g711ulaw
!
voice class codec 4
  codec preference 1 g711alaw
  codec preference 2 g722-64
!
!
!
!
!
!
!
voice-card 0
!
crypto pki token default removal timeout 0
!
license udi pid CISCO2851 sn FHK1137F4LY

controller E1 0/2/0
  pri-group timeslots 1-31 service mgcp

controller E1 0/3/0
  clock source internal
  pri-group timeslots 1-31

ip tftp source-interface GigabitEthernet0/0

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
  encapsulation hdlc
  isdn switch-type primary-qsig
  isdn timer T310 120000
  isdn protocol-emulate network
  isdn incoming-voice voice
  isdn map address .* plan isdn type national
  isdn bind-l3 ccm-manager
  isdn send-alerting
  isdn sending-complete
  no cdp enable

!
interface Serial0/3/0:15
  no ip address
  encapsulation hdlc
  isdn switch-type primary-qsig
  isdn timer T310 120000
  isdn protocol-emulate network
  isdn incoming-voice voice
  no cdp enable
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 number 6123565991
caller-id enable
!
voice-port 0/0/1
   no echo-cancel enable
   no vad
!
voice-port 0/3/0:15
!
voice-port 0/2/0:15
!
voice-port 0/1/0
   no echo-cancel enable
   no vad
!
voice-port 0/1/1
   no echo-cancel enable
   no vad
!
dspfarm profile 1 transcode
   shutdown
!
!
dial-peer voice 111 pots
   service session
   destination-pattern 5991
   no digit-strip
   port 0/0/0
   forward-digits all
!  
!

dial-peer voice 113 voip

description Gateway to CUCM

shutdown

destination-pattern .T

session protocol sipv2

session target ipv4:10.80.20.2:5060

session transport udp

incoming called-number 5991

voice-class codec 1

dtmf-relay rtp-nre

fax protocol pass-through g711ulaw

!
!
sip-ua

no remote-party-id

retry register 5

no timers hold

protocol mode dual-stack

!
!
line con 0

line aux 0

line 66

no activation-character

no exec

transport preferred none

transport input all
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
  login authentication local_auth
  transport input all

!
scheduler allocate 20000 1000

end
!
!
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>

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<th>Americas Headquarters</th>
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<td>International BV</td>
<td>170 West Tasman Drive</td>
<td>Capital Tower</td>
</tr>
<tr>
<td>San Jose, CA 95134-1706</td>
<td>Haarlebergpark</td>
<td>San Jose, CA 95134-1706</td>
<td>168 Robinson Road</td>
</tr>
<tr>
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