KCOM SIP Trunking:

Cisco Unified Communications Manager 11.0.1 with Cisco Unified Border Element (CUBE 11.5.0) on ISR 4321 [IOS-XE 3.17 – 15.6(1)S] using SIP

May 24, 2016
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Introduction

Service Providers today, such as KCOM, 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.

KCOM is a service provider offering that allows connection to the PSTN and may offer the end customer a viable alternative to traditional PSTN connectivity. A demarcation device between these services and customer owned services is recommended. As an intermediary device between Cisco Unified Communications Manager and KCOM network, Cisco Unified Border Element (Cisco UBE) ISR 4321/K9 running IOS-XE 3.17 – 15.6(1)S can be used. The Cisco Unified Border Element 11.5.0 provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 11.0.1 connected to KCOM IP network.

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

- This application note describes how to configure a Cisco Unified Communications Manager (Cisco UCM) 11.0.1 and Cisco Unified Border Element (Cisco UBE) on ISR 4321/K9 [IOS-XE 3.17 – 15.6(1)S] for connectivity to KCOM SIP trunking service. The deployment model covered in this application note is CPE (Cisco UCM 11.0.1) to PSTN (KCOM).

- Testing was performed in accordance to KCOM generic SIP trunking test methodology and among features verified were – basic calls, DTMF transport, Music on Hold (MOH), unattended and attended transfers, call forward, conferences and interoperability with Cisco Unity Connection (CUC)

- The Cisco UCM configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between KCOM 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 KCOM SIP trunking network.

This application note does not cover the use of Calling Search Spaces (CSS) or partitions on Cisco UCM. To understand and learn how to apply CSS and partitions refer to the cisco.com link below:

Network Topology

Figure 1: Network Topology

- Cisco IP Phones 7971, 8961 and 7965 are the devices primarily used throughout the testing to place or receive calls.
- VentaFax Soft Client is used to perform all fax related scenarios. The fax client is connected to SIP Gateway via FXS port, which in turn, communicates with Cisco UCM over SIP.
System Components

Hardware Requirements
- Cisco UCS-C240-M3S VMWare host running ESXi 5.5 Standard
- Cisco ISR 4321/K9 router as CUBE
- Cisco 2851 Fax Gateway
- IP phones 8961 (SIP) and 7965, 7971 (SCCP)

Software Requirements
- Cisco Unified Communications Manager 11.0.1
- Cisco Unity Connection 11.0.1
- IOS-XE 3.17 for ISR 4321/K9 Cisco Unified Border Element (v 11.5.0)
- IOS 15.1(4)M5 for Cisco 2851 Fax Gateway

Features

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

Features Not Supported
- Cisco IP phones used in this test do not support blind transfer

Caveats
- CLID is not updated on PSTN phones for transfer (attended and unattended) Offnet PSTN scenarios. Caller ID is not updated at PSTN once transfer is completed by PBX. Cisco UBE modifies the PAI/PPI header and forwards to the network in the tested release. CISCO BUG ID: CSCuv04539.
- KCOM sends 4-digits number to Cisco UBE/UCM instead of the full DID, which is the default setting
Configuration
Configuring Cisco Unified Border Element

Network Interface
Ethernet IP address and sub interface are configured. The IP address and VLAN encapsulation used are for illustration only and the actual IP address can vary. For SIP trunks two IP addresses must be configured - for LAN and WAN.

interface GigabitEthernet0/0/0
description KCOM CUBE1 LAN MS4 1/0/5
ip address 10.80.13.21 255.255.255.0
negotiation auto
redundancy rii 3
redundancy group 2 ip 10.80.13.20 exclusive
!
interface GigabitEthernet0/0/1
description KCOM CUBE1 WAN MS4 1/0/6
ip address 192.XX.XX.XX 255.255.255.224
negotiation auto
redundancy rii 4
redundancy group 2 ip 192.XX.XX.XX exclusive
!
Global Cisco UBE Settings

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

voice service voip
ip address trusted list
    ipv4 0.0.0.0 0.0.0.0
no ip address trusted authenticate
address-hiding
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 2
redirect ip2ip
fax protocol pass-through g711alaw
sip
    bind control source-interface GigabitEthernet0/0/1
    bind media source-interface GigabitEthernet0/0/1
    rel1xx supported "rel100"
    session refresh
    asserted-id pai
    privacy pstn
    early-offer forced
    midcall-signaling passthru
    privacy-policy passthru
    pass-thru subscribe-notify-events 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>fax protocol</td>
<td>Specifies the fax protocol</td>
</tr>
<tr>
<td>asserted-id</td>
<td>Specifies the type of privacy header in the outgoing SIP requests and response messages</td>
</tr>
<tr>
<td>early-offer forced</td>
<td>Enables SIP Delayed-Offer to Early-Offer globally</td>
</tr>
<tr>
<td>midcall-signaling passthru</td>
<td>Passes SIP messages from one IP leg to another IP leg</td>
</tr>
</tbody>
</table>
Media Passing through Cisco UBE (media flow-through vs. media flow-around)

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

- voice service voip
  - media flow-around

**Codecs**

G729 is used as the preferred codec for this testing

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

**Dial Peer**

Cisco UBE uses dial-peers to route the call accordingly based on the digits

- dial-peer voice 201 voip
  - description PBX to PBX call via KCOM - WAN facing
  - destination-pattern 01T
  - session protocol sipv2
  - session target sip-server
  - voice-class codec 1
  - voice-class sip early-offer forced
  - voice-class sip profiles 101
  - 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 disable
  - fax protocol pass-through g711alaw
  - no vad

!
dial-peer voice 202 voip
description PBX to PBX call via KCOM - LAN facing
session protocol sipv2
session target sip-server
incoming called-number 01T
voice-class codec 1
voice-class sip early-offer forced
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
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711alaw
no vad
!
dial-peer voice 999 voip
description "Emergency and Operator call"
destination-pattern [1,9]..
session protocol sipv2
session target sip-server
voice-class codec 1
voice-class sip early-offer forced
voice-class sip profiles 101
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
fax protocol pass-through g711alaw
no vad
!
dial-peer voice 118 voip
description "information service call"
destination-pattern 118218
session protocol sipv2
session target sip-server
voice-class codec 1
voice-class sip early-offer forced
voice-class sip profiles 101
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 protocol pass-through g711alaw
no vad
!
dial-peer voice 998 voip
description "Emergency and Operator call-LAN side"
session protocol sipv2
session target sip-server
incoming called-number [1,9].
voice-class codec 1
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 protocol pass-through g711alaw
no vad
!
dial-peer voice 218 voip
description "information service call-LAN side"
session protocol sipv2
session target sip-server
incoming called-number 118218
voice-class codec 1
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 protocol pass-through g711alaw
no vad
!
dial-peer voice 101 voip
description Inbound-from PSTN to IP PBX - LAN facing
destination-pattern 926.
session protocol sipv2
session target ipv4:XX.XX.XX.X:5060
voice-class codec 1
voice-class sip early-offer forced
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 disable
fax protocol pass-through g711alaw
no vad
!
dial-peer voice 100 voip
description Inbound-from PSTN to IP PBX - WAN facing
session protocol sipv2
session target sip-server
incoming called-number 926.
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nse
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711alaw
no vad
!
dial-peer voice 588 voip
description from IP PBX to PSTN - LAN facing
session protocol sipv2
session target sip-server
incoming called-number .T
voice-class codec 1
voice-class sip early-offer forced
voice-class sip profiles 101
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nse
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711alaw
no vad
!
dial-peer voice 589 voip
description from IP PBX to PSTN - WAN facing
destination-pattern .T
session protocol sipv2
session target sip-server
voice-class codec 1
voice-class sip profiles 101
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 disable
fax protocol pass-through g711alaw
no vad

! 
dial-peer voice 210 voip
description outgoing call to KCOM - facing CUCM
session protocol sipv2
session target sip-server
incoming called-number 001T
voice-class codec 1
voice-class sip early-offer forced
voice-class sip profiles 101
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 disable
fax protocol pass-through g711alaw
no vad

!
dial-peer voice 200 voip
description Outbound-from IP PBX to PSTN - WAN facing
destination-pattern 001T
session protocol sipv2
session target sip-server
voice-class codec 1
voice-class sip early-offer forced
voice-class sip profiles 101
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 disable
fax protocol pass-through g711alaw
no vad
!

Call Flow
In the sample configuration presented here, Cisco UCM is provisioned with four-digit directory numbers corresponding to the last four DID digits. No digit manipulation is performed on the Cisco UBE.

For incoming PSTN calls, KCOM presents 4-digits number to Cisco UBE which then forward to Cisco UCM. Voice calls are routed to IP phones; Fax calls are routed via a 4-digit route pattern over a SIP trunk that terminates on the Fax Gateway and in turn to the VentaFax client connected to the Fax Gateway.

CPE callers make out-bound PSTN calls by dialing an “8” prefix followed by the destination number. For outbound fax calls from the analog fax endpoint, Cisco fax Gateway sends to Cisco UCM the DID with leading access code “8”. An “8.!” route pattern strips the prefix and routes the call with the remaining digits via a SIP trunk terminating on the Cisco UBE for Voice call or Fax.

Figure 2: Outbound Voice Call
Figure 3: Inbound Voice Call

Figure 4: Outbound Fax Call

Figure 5: Inbound Fax Call
Configuration Example
The following configuration snippet contains a sample configuration of Cisco UBE with all parameters mentioned previously

Active Cisco UBE

KCOM_CUBE1#sh run
version 15.6
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname KCOM_CUBE1
!
boot-start-marker
boot system bootflash:isr4300-universalk9.03.17.00.S.156-1.S-std.SPA.bin
boot-end-marker
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
logging queue-limit 10000
logging buffered 100000000
logging rate-limit 10000
no logging console
!
aaa new-model
aaa session-id common
no ip domain lookup
ip domain name tekvizion.com
!
subscriber templating
multilink bundle-name authenticated
!
voice service voip
ip address trusted list
ipv4 0.0.0.0 0.0.0.0
no ip address trusted authenticate
address-hiding
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 2
redirect ip2ip
fax protocol pass-through g711alaw
sip
bind control source-interface GigabitEthernet0/0/1
bind media source-interface GigabitEthernet0/0/1
rel1xx supported "rel100"
session refresh
asserted-id pai
privacy pstn
early-offer forced
midcall-signaling passthru
privacy-policy passthru
pass-thru subscribe-notify-events all
!

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

voice class sip-profiles 101
request INVITE sip-header Diversion modify "<sip:(.*)@(.*)" "<sip:0113399\1@\2>"

license udi pid ISR4321/K9 sn FDO19230MQ8
license boot level appxk9
license boot level uck9
!
spanning-tree extend system-id
!
redundancy
mode none
application redundancy
group 2
    name Voice-b2bha_KCOM
priority 100 failover threshold 75
timers delay 30 reload 60
control GigabitEthernet0/1/0 protocol 1
data GigabitEthernet0/1/0
track 1 shutdown
track 2 shutdown
!
vlan internal allocation policy ascending
!
track 1 interface GigabitEthernet0/0/0 line-protocol
track 2 interface GigabitEthernet0/0/1 line-protocol
!
interface GigabitEthernet0/0/0
description KCOM CUBE1 LAN MS4 1/0/5
ip address 10.80.13.21 255.255.255.0
negotiation auto
redundancy rii 3
redundancy group 2 ip 10.80.13.20 exclusive
!
interface GigabitEthernet0/0/1
description KCOM CUBE1 WAN MS4 1/0/6
ip address 192.XX.XX.XX 255.255.255.224
negotiation auto
redundancy rii 4
redundancy group 2 ip 192.XX.XX.XX exclusive
!
interface GigabitEthernet0/1/0
description CUBE HA MS5 3/0/35
ip address 10.89.20.7 255.255.255.0
negotiation auto
!
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
negotiation auto
!
interface Vlan1
no ip address
shutdown
!
ip forward-protocol nd
no ip http server
no ip http secure-server
ip tftp source-interface GigabitEthernet0
ip rtcp report interval 3000
ip route 0.0.0.0 0.0.0.0 192.65.XX.XX
ip route 10.64.0.0 255.255.0.0 10.80.13.1
ip route 10.80.13.0 255.255.255.0 10.80.13.1
ip route 172.16.0.0 255.255.0.0 10.80.13.1

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
!
mgcp profile default
!
!
dial-peer voice 201 voip
description PBX to PBX call via KCOM - WAN facing
destination-pattern 01T
session protocol sipv2
session target sip-server
voice-class codec 1
voice-class sip early-offer forced
voice-class sip profiles 101
voice-class sip bind control source-interface GigabitEthernet0/0/1
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dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711alaw
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fax protocol pass-through g711alaw
no vad
!
dial-peer voice 998 voip
description "Emergency and Operator call-LAN side"
session protocol sipv2
session target sip-server
incoming called-number [1,9]..
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
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destination-pattern .T
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fax-relay ecm disable
fax rate disable
fax protocol pass-through g711alaw
no vad
!
dial-peer voice 210 voip
description outgoing call to KCOM - facing CUCM
session protocol sipv2
session target sip-server
incoming called-number 001T
voice-class codec 1
voice-class sip early-offer forced
voice-class sip profiles 101
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 disable
fax protocol pass-through g711alaw
no vad
!
dial-peer voice 200 voip
dercription Outbound-from IP PBX to PSTN - WAN facing
destination-pattern 001T
session protocol sipv2
session target sip-server
voice-class codec 1
voice-class sip early-offer forced
voice-class sip profiles 101
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 disable
fax protocol pass-through g711alaw
no vad
!
!
gateway
media-inactivity-criteria all
timer receive-rtcp 5
timer receive-rtp 86400
!
sip-ua
sip-server ipv4:XX.XX.XX.XX:5060
!
line con 0
stopbits 1
line aux 0
stopbits 1
line vty 0 4
password XXXXXXXXXXX
transport input telnet ssh
!
!
end
Standby Cisco UBE:

KCOM_CUBE2#show run
!
version 15.6
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname KCOM_CUBE2
!
boot-start-marker
boot system bootflash:isr4300-universalk9.03.17.00.S.156-1.S-std.SPA.bin
boot-end-marker
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
logging queue-limit 10000
logging buffered 100000000
logging rate-limit 10000
no logging console
!
aaa new-model
aaa session-id common
!
no ip domain lookup
ip domain name tekvizion.com

subscriber templating
multilink bundle-name authenticated

voice service voip
ip address trusted list
ipv4 0.0.0.0 0.0.0.0
no ip address trusted authenticate
address-hiding
mode border-element
allow-connections sip to sip
redundancy-group 2
redirect ip2ip
fax protocol pass-through g711alaw
sip
bind control source-interface GigabitEthernet0/0/1
bind media source-interface GigabitEthernet0/0/1
rel1xx supported "rel100"
session refresh
asserted-id pai
privacy pstn
early-offer forced
midcall-signaling passthru
privacy-policy passthru
pass-thru subscribe-notify-events all

voice class codec 2
codec preference 1 g729r8
codec preference 2 g711alaw
codec preference 3 g711ulaw
!
voice class codec 3
codec preference 1 g711alaw
codec preference 2 g711ulaw
!
voice class codec 1
codec preference 1 g729r8
codec preference 2 g711alaw
codec preference 3 g711ulaw
!
!
voice class sip-profiles 101
request INVITE sip-header Diversion modify "<sip:(.*)@(.*>)" "<sip:0113399\1@\2>"
!
license udi pid ISR4321/K9 sn FDO19220MW3
license boot level appxk9
license boot level uck9
!
spanning-tree extend system-id
!
redundancy
mode none
application redundancy
group 2
  name Voice-b2bha_KCOM
  priority 100 failover threshold 75
timers delay 30 reload 60
control GigabitEthernet0/1/0 protocol 1
data GigabitEthernet0/1/0
track 1 shutdown
track 2 shutdown
!
vlan internal allocation policy ascending
!
track 1 interface GigabitEthernet0/0/0 line-protocol
track 2 interface GigabitEthernet0/0/1 line-protocol
!
interface GigabitEthernet0/0/0
description KCOM CUBE2 LAN MS4 1/0/7
ip address 10.80.13.22 255.255.255.0
negotiation auto
redundancy rii 3
redundancy group 2 ip 10.80.13.20 exclusive
!
interface GigabitEthernet0/0/1
description KCOM CUBE2 WAN MS4 1/0/8
ip address 192.XX.XX.XX 255.255.255.224
negotiation auto
redundancy rii 4
redundancy group 2 ip 192.XX.XX.XX exclusive
!
interface GigabitEthernet0/1/0
description CUBE HA MS5 3/0/36
ip address 10.89.20.8 255.255.255.0
negotiation auto
!
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
negotiation auto
!
interface Vlan1
no ip address
shutdown
!
ip forward-protocol nd
no ip http server
no ip http secure-server
ip tftp source-interface GigabitEthernet0
ip rtcp report interval 3000
ip route 0.0.0.0 0.0.0.0 192.65.XX.XX
ip route 10.64.0.0 255.255.0.0 10.80.13.1
ip route 10.80.13.0 255.255.255.0 10.80.13.1
ip route 172.16.0.0 255.255.0.0 10.80.13.1
!
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
!
mgcp profile default
!
dial-peer voice 201 voip
description PBX to PBX call via KCOM - WAN facing
destination-pattern 01T
session protocol sipv2
session target sip-server
voice-class codec 1
voice-class sip early-offer forced
voice-class sip profiles 101
voice-class sip bind control source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711alaw
no vad
!
dial-peer voice 202 voip
description PBX to PBX call via KCOM - LAN facing
session protocol sipv2
session target sip-server
incoming called-number 01T
voice-class codec 1
voice-class sip early-offer forced
voice-class sip bind control source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711alaw
no vad
!
dial-peer voice 999 voip
description "Emergency and Operator call"
destination-pattern [1,9]..
session protocol sipv2
session target sip-server
voice-class codec 1
voice-class sip early-offer forced
voice-class sip profiles 101
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 protocol pass-through g711alaw
no vad
!
dial-peer voice 118 voip
description "information service call"
destination-pattern 118218
session protocol sipv2
session target sip-server
voice-class codec 1
voice-class sip early-offer forced
voice-class sip profiles 101
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 protocol pass-through g711alaw
no vad
!
dial-peer voice 998 voip
description "Emergency and Operator call-LAN side"
session protocol sipv2
session target sip-server
incoming called-number [1,9].
voice-class codec 1
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 protocol pass-through g711alaw
no vad
!
dial-peer voice 218 voip
description "information service call-LAN side"
  session protocol sipv2
  session target sip-server
incoming called-number 118218
voice-class codec 1
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 protocol pass-through g711alaw
no vad
!
dial-peer voice 101 voip
description Inbound-from PSTN to IP PBX - LAN facing
destination-pattern 926.
  session protocol sipv2
  session target ipv4:XX.XX.XX.X:5060
voice-class codec 1
voice-class sip early-offer forced
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 disable
fax protocol pass-through g711alaw
no vad
!
dial-peer voice 100 voip
description Inbound-from PSTN to IP PBX - WAN facing
session protocol sipv2
session target sip-server
incoming called-number 926.
voice-class codec 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
fax rate disable
fax protocol pass-through g711alaw
no vad
!
dial-peer voice 588 voip
description from IP PBX to PSTN - LAN facing
session protocol sipv2
session target sip-server
incoming called-number .T
voice-class codec 1
voice-class sip early-offer forced
voice-class sip profiles 101
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 disable
fax protocol pass-through g711alaw
no vad!

dial-peer voice 589 voip
description from IP PBX to PSTN - WAN facing
destination-pattern .T
session protocol sipv2
session target sip-server
voice-class codec 1
voice-class sip profiles 101
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 disable
fax protocol pass-through g711alaw
no vad!

dial-peer voice 210 voip
description outgoing call to KCOM - facing CUCM
session protocol sipv2
session target sip-server
incoming called-number 001T
voice-class codec 1
voice-class sip early-offer forced
voice-class sip profiles 101
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 disable
fax protocol pass-through g711alaw
no vad
!
dial-peer voice 200 voip
description Outbound-from IP PBX to PSTN - WAN facing
destination-pattern 001T
session protocol sipv2
session target sip-server
voice-class codec 1
voice-class sip early-offer forced
voice-class sip profiles 101
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 disable
fax protocol pass-through g711alaw
no vad
!
!
gateway
media-inactivity-criteria all
timer receive-rtcp 5
timer receive-rtp 86400
!
sip-ua
sip-server ipv4:XX.XX.XX.XX:5060
!
! line con 0
  stopbits 1
line aux 0
  stopbits 1
line vty 0 4
  password Xxxxxxxxx
  transport input telnet ssh
!
end
Cisco Call Manager Service Parameters

**Navigation Path:** System > Service Parameters

Select Server* = clus23pubsub--CUCM Voice/Video (Active)
Select Service*= Cisco CallManager (Active)

All other fields are set to Suggested Values
Off-net Calls via KCOM SIP Trunk

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

**SIP Trunk Security Profile**

**Navigation Path:** System > Security > SIP Trunk Security Profile

![Image of SIP Trunk Security Profile Configuration]

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Incoming Transport Type</td>
<td>TCP + UDP</td>
<td>SIP trunks to KCOM SBC should use UDP as a transport protocol for SIP. This is configured using SIP Trunk Security profile, which is later assigned to the SIP trunk itself.</td>
</tr>
<tr>
<td>Outgoing Transport Type</td>
<td>UDP</td>
<td></td>
</tr>
</tbody>
</table>
**SIP Profile Configuration**

The SIP Profile will be later associated with the SIP trunk

**Navigation Path:** Device > Device Settings > SIP Profile

**Name** = KCOM Standard SIP Profile

**Description** = KCOM Default SIP Profile

![SIP Profile Configuration](image)

Figure 9: SIP Profile
Figure 10: SIP Profile (Cont.)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default MTP Telephony Event Payload Type</td>
<td>101</td>
<td>RFC2833 DTMF payload type</td>
</tr>
<tr>
<td>SIP Rel1XX Options</td>
<td>Send PRACK for 1xx Messages</td>
<td>Enable Provisional Acknowledgements (Reliable 100 messages)</td>
</tr>
<tr>
<td>Ping Interval for In-service and Partially In-service Trunks (seconds)</td>
<td>30</td>
<td>OPTIONS message parameters- interval time</td>
</tr>
<tr>
<td>Ping Interval for Out-of-service Trunks (seconds)</td>
<td>120</td>
<td>OPTIONS message parameters- interval time</td>
</tr>
</tbody>
</table>
SIP Trunk Configuration

SIP trunk between Cisco UCM and Cisco UBE is configured

**Navigation Path:** Device > Trunk

**Figure 11:** SIP Trunks List
Figure 12: SIP Trunk to Cisco UBE

Figure 13: SIP Trunk to Cisco UBE (Cont.)
Figure 14: SIP Trunk to Cisco UBE (Cont.)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Name</td>
<td>KCOM_SIP_Trunk</td>
<td>Name for the trunk</td>
</tr>
<tr>
<td>Device Pool</td>
<td>G729_Pool</td>
<td>Default Device Pool is used for this trunk</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>MRGL_MTP</td>
<td>MRG with resources: ANN, CFB, MOH and MTP</td>
</tr>
<tr>
<td>Significant Digits</td>
<td>4</td>
<td>4 digits Extension for all CPE phones</td>
</tr>
<tr>
<td>Destination Address</td>
<td>10.80.13.20</td>
<td>IP address of the Cisco UBE Virtual LAN</td>
</tr>
<tr>
<td>SIP Trunk Security Profile</td>
<td>Non Secure UDP SIP Trunk Profile KCOM</td>
<td>SIP Trunk Security Profile configured earlier</td>
</tr>
<tr>
<td>SIP Profile</td>
<td>KCOM Standard SIP Profile</td>
<td>SIP Profile configured earlier</td>
</tr>
</tbody>
</table>
Dial Plan

Route Pattern Configuration

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

Route patterns are configured as follows:

- Cisco IP phone dial "8.1" (8+10 digits number to access PSTN via Cisco UBE)
- "8" is removed before send to Cisco UBE
- For FAX call, Access Code "8" is used at Cisco Fax gateway
- "8" is removed at Cisco UCM
- 00+Country Code+10 digits number is send to Cisco UBE to KCOM network
- Incoming fax call to 9264 will be sent to Cisco Fax gateway

Figure 15: Route Patterns List
Figure 16: Route Pattern for Voice

Figure 17: Route Pattern for Voice (Cont.)
Figure 18: Route Pattern for Voice (Cont.)

<table>
<thead>
<tr>
<th><strong>Route Pattern</strong></th>
<th>8.1</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Partition</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Description</td>
<td></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>MPP Precedence</td>
<td></td>
</tr>
<tr>
<td>Apply Call Blocking Percentage</td>
<td></td>
</tr>
<tr>
<td>Resource Priority Namespace Network Domain</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Route Class</td>
<td>Default</td>
</tr>
<tr>
<td>Gateway/Route List</td>
<td>KCON_SIP_Trunk</td>
</tr>
<tr>
<td>Route Option</td>
<td>Route this pattern</td>
</tr>
<tr>
<td>Call Classification</td>
<td></td>
</tr>
<tr>
<td>External Call Control Profile</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

Figure 19: Route Pattern for Voice (Cont.)

**Calling Party Transformations**

- Use Calling Party’s External Phone Number Mask
- Calling Party Transform Mask: 113399XXXX
- Prefix Digits (Outgoing Calls): 113399XXXX
- 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: Predot
- Called Party Transform Mask:     
- Prefix Digits (Outgoing Calls):     
- Called Party Number Type: Cisco CallManager
**Figure 20:** Route Pattern for Voice (Cont.)

<table>
<thead>
<tr>
<th>Route Pattern</th>
<th>999</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Partition</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Description</td>
<td></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>MLPF Precedence</td>
<td>Default</td>
</tr>
</tbody>
</table>

**Figure 21:** Route Pattern for Voice (Cont.)

<table>
<thead>
<tr>
<th>Route Pattern</th>
<th>100</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Partition</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Description</td>
<td></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>MLPF Precedence</td>
<td>Default</td>
</tr>
</tbody>
</table>

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### Figure 22: Route Pattern for Voice (Cont.)

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Pattern</td>
<td>8.! for Voice call, 9264 for fax call &amp;0113399926X for CPE to CPE call via KCOM, 118XXX for information services, 100 &amp; 999 for operator call, 10XX for Unity and emergency services</td>
<td>Specify appropriate Route Pattern</td>
</tr>
<tr>
<td>Gateway/Route List</td>
<td>KCOM_SIP_Trunk for Route Pattern 8.!, 100, 999, 118XXX, 0113399926X, SIP_Trunk_Fax_Gateway for Route Pattern 9264, KCOM_SIP_TRUNK_Unity for Route Pattern 10XX.</td>
<td>SIP trunk name configured earlier</td>
</tr>
<tr>
<td>Call Classification</td>
<td>Offnet for Route Pattern 8.!, 9264, 118XXX, 0113399926X, 100 &amp; 999</td>
<td>Restrict the transferring of an external call to an external device</td>
</tr>
<tr>
<td>Discard Digits</td>
<td>PreDot for Route Pattern 8.!</td>
<td>Specifies how to modify digit before they are sending to KCOM network</td>
</tr>
</tbody>
</table>
### 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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