Application Note

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:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab10/collab10/dialplan.html

Network Topology

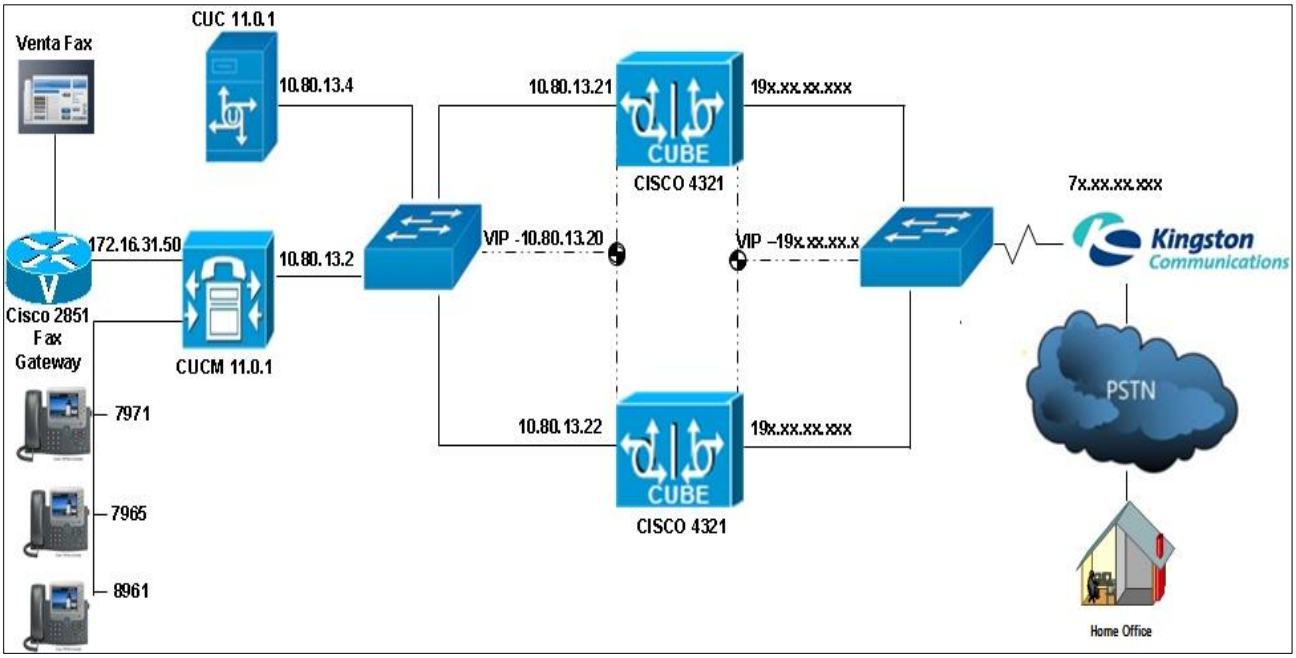

Figure 1: Network Topology

- 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

Command	Description
allow-connections sip to sip	Allow IP2IP connections between two SIP call legs
fax protocol	Specifies the fax protocol
asserted-id	Specifies the type of privacy header in the outgoing SIP requests and response messages
early-offer forced	Enables SIP Delayed-Offer to Early-Offer globally
midcall-signaling passthru	Passes SIP messages from one IP leg to another IP leg

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-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
!
```

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 outbound 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.

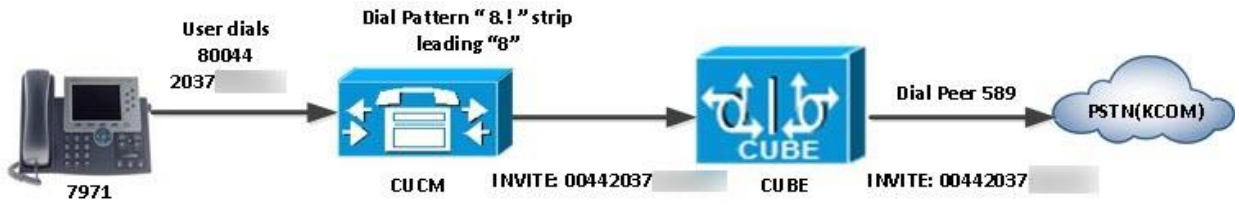

Figure 2: Outbound Voice Call

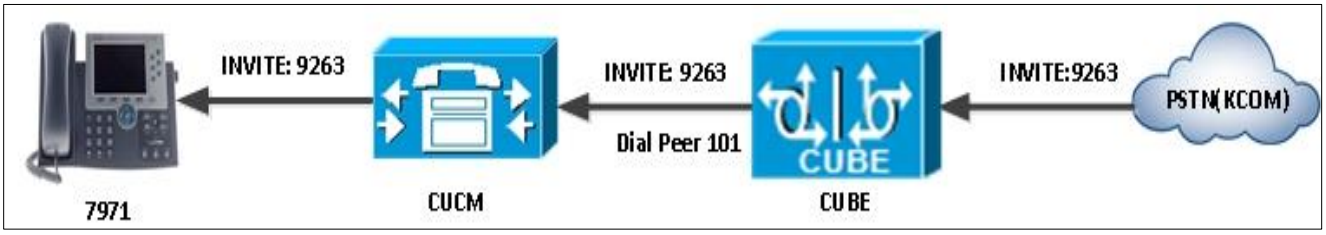

Figure 3: Inbound Voice Call

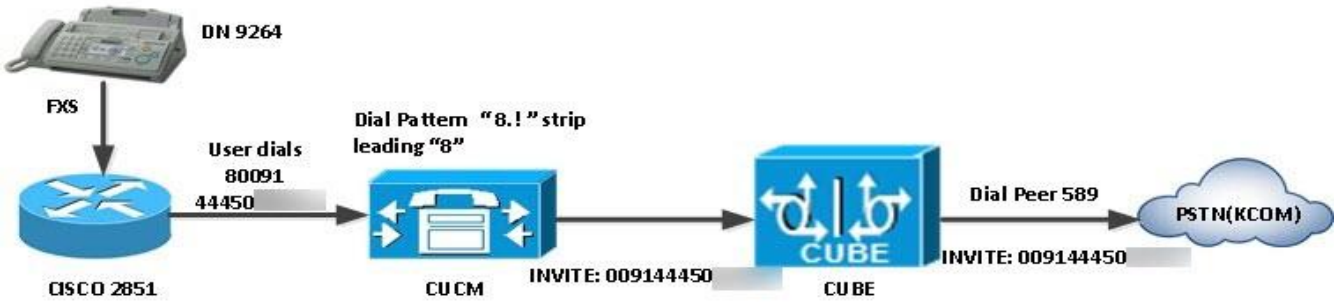

Figure 4: Outbound Fax Call

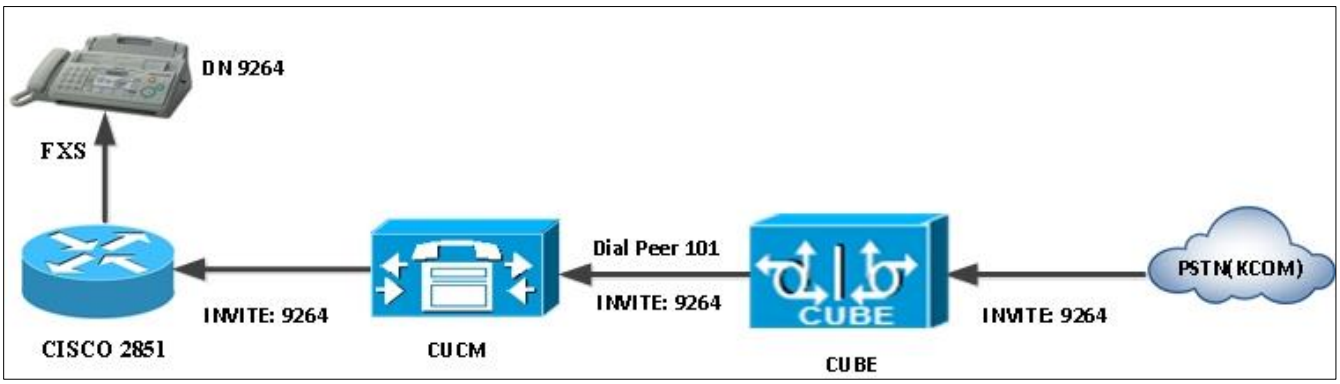

Figure 5: Inbound Fax Call

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
!
```

```
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
 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
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dial-peer voice 202 voip
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 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 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
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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 XXXXXXXX
 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
 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-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 XXXXXXXX
 transport input telnet ssh
!
end
```

Cisco UCM Version

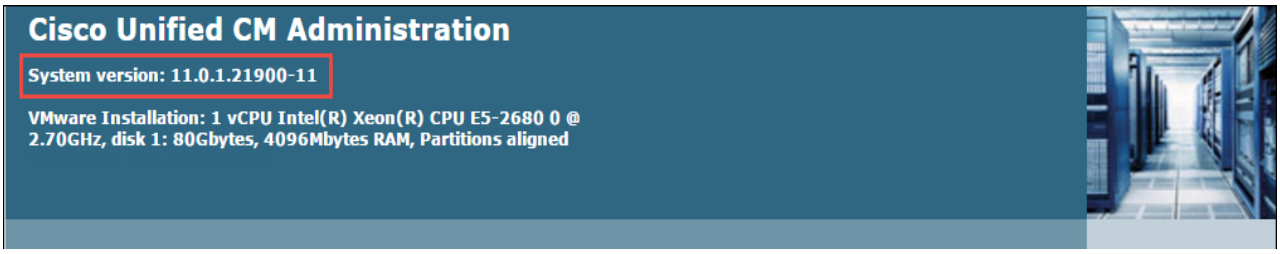


Figure 6: Cisco UCM Version

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

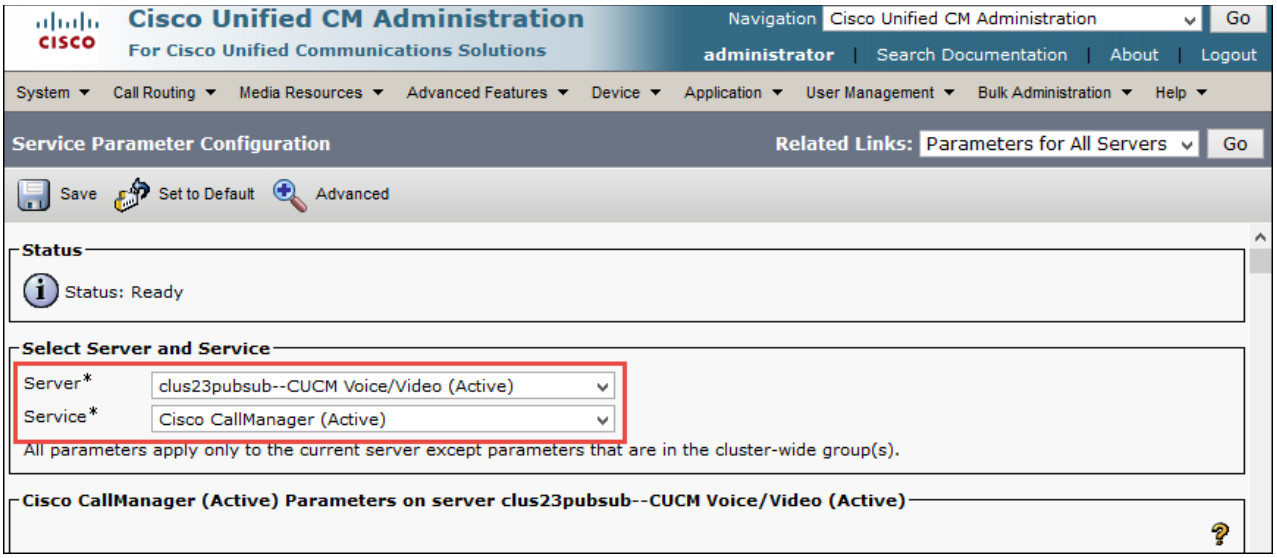


Figure 7: Service Parameters

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

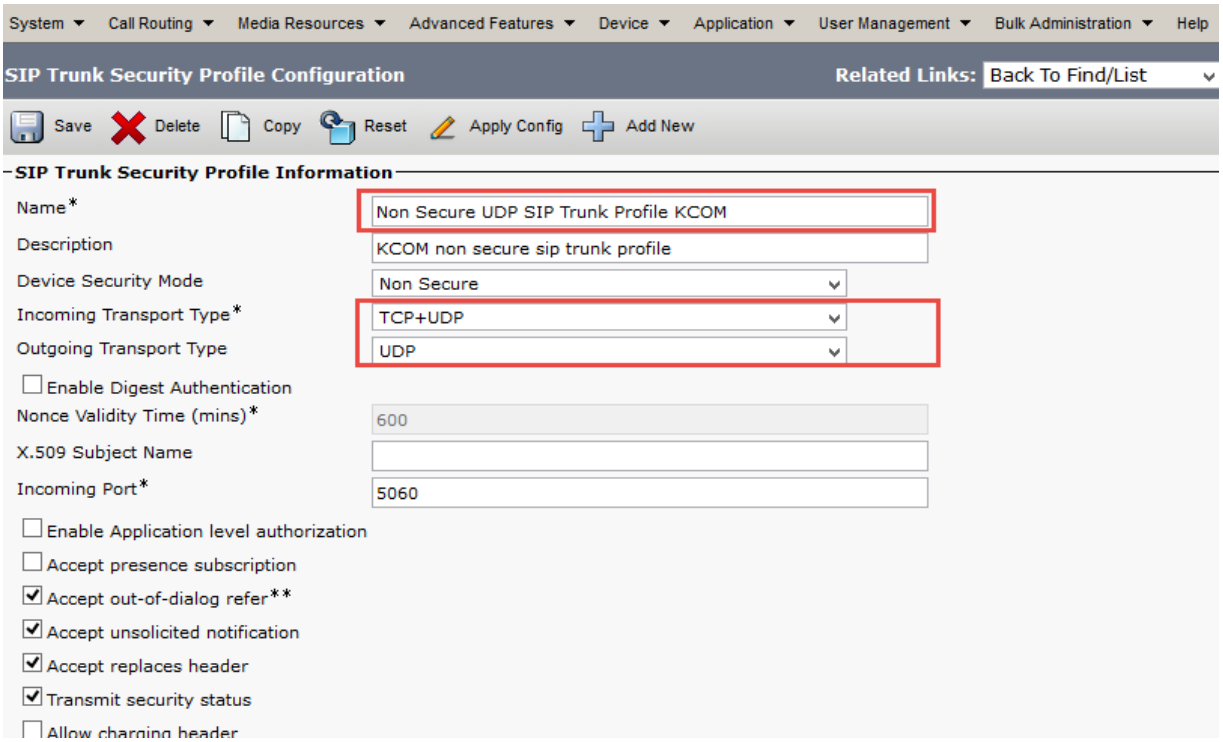


Figure 8: SIP Trunk Security Profile

Parameter	Value	Description
Incoming Transport Type	TCP + UDP	
Outgoing Transport Type	UDP	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.

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

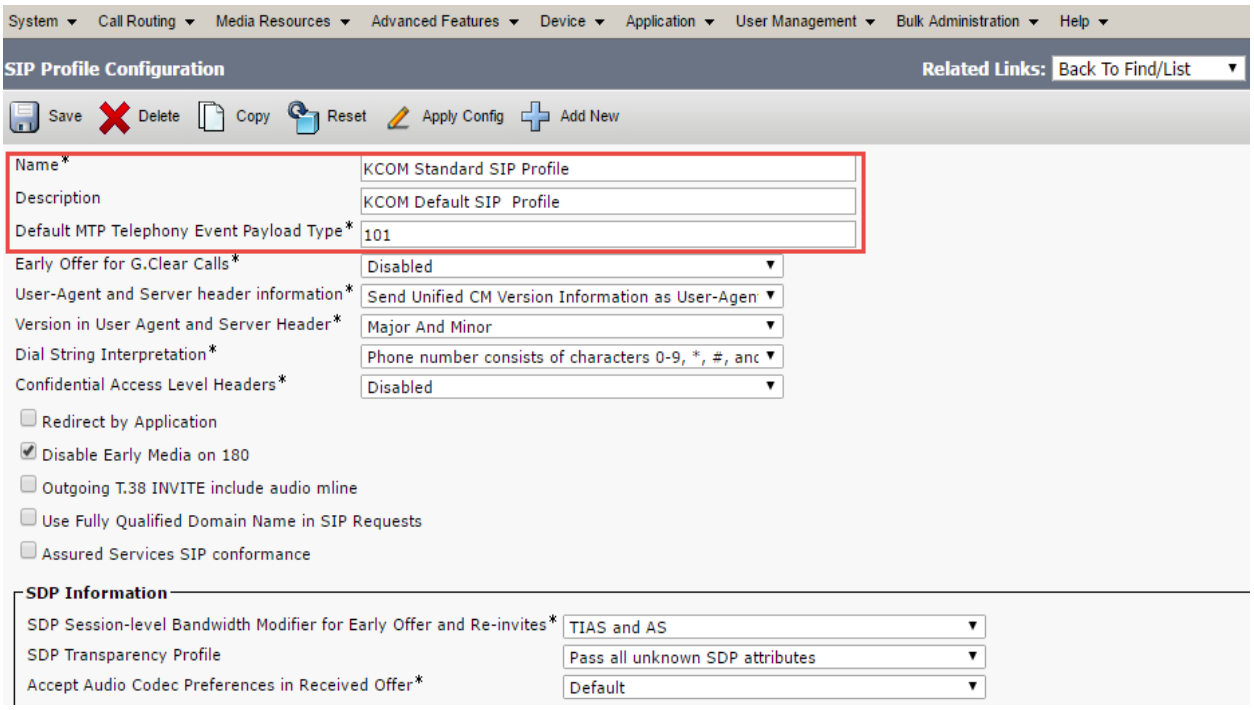


Figure 9: SIP Profile

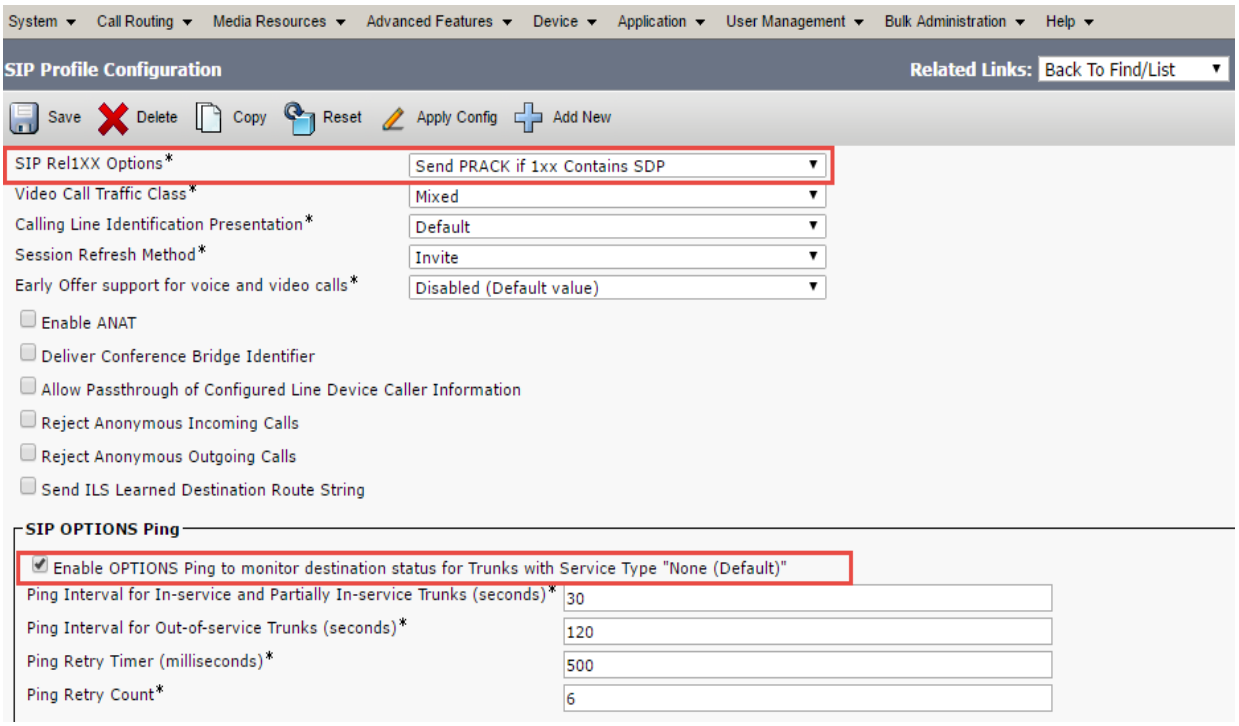


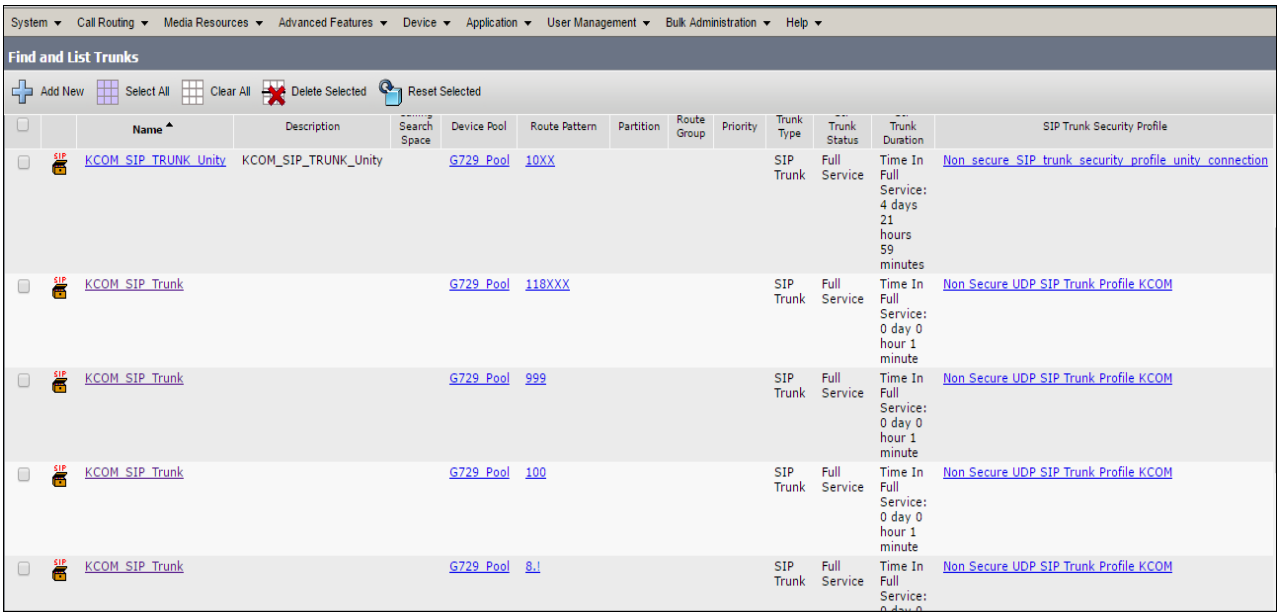
Figure 10: SIP Profile (Cont.)

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

SIP Trunk Configuration

SIP trunk between Cisco UCM and Cisco UBE is configured

Navigation Path: Device > Trunk



Name	Description	Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type	Trunk Status	Trunk Duration	SIP Trunk Security Profile
KCOM_SIP_TRUNK_Unity	KCOM_SIP_TRUNK_Unity		G729_Pool	10XX				SIP Trunk	Full Service	Time In Full Service: 4 days 21 hours 59 minutes	Non_secure_SIP_trunk_security_profile_unity_connection
KCOM_SIP_Trunk			G729_Pool	118XXX				SIP Trunk	Full Service	Time In Full Service: 0 day 0 hour 1 minute	Non Secure UDP SIP Trunk Profile KCOM
KCOM_SIP_Trunk			G729_Pool	999				SIP Trunk	Full Service	Time In Full Service: 0 day 0 hour 1 minute	Non Secure UDP SIP Trunk Profile KCOM
KCOM_SIP_Trunk			G729_Pool	100				SIP Trunk	Full Service	Time In Full Service: 0 day 0 hour 1 minute	Non Secure UDP SIP Trunk Profile KCOM
KCOM_SIP_Trunk			G729_Pool	8.!				SIP Trunk	Full Service	Time In Full Service:	Non Secure UDP SIP Trunk Profile KCOM

										minute	
KCOM_SIP_Trunk			G729_Pool	999				SIP Trunk	Full Service	Time In Full Service: 0 day 0 hour 1 minute	Non Secure UDP SIP Trunk Profile KCOM
KCOM_SIP_Trunk			G729_Pool	100				SIP Trunk	Full Service	Time In Full Service: 0 day 0 hour 1 minute	Non Secure UDP SIP Trunk Profile KCOM
KCOM_SIP_Trunk			G729_Pool	8.!				SIP Trunk	Full Service	Time In Full Service: 0 day 0 hour 1 minute	Non Secure UDP SIP Trunk Profile KCOM
KCOM_SIP_Trunk			G729_Pool	0113399926X				SIP Trunk	Full Service	Time In Full Service: 0 day 0 hour 1 minute	Non Secure UDP SIP Trunk Profile KCOM
SIP_Trunk_Fax_Gateway			G711Alaw	9264				SIP Trunk	Full Service	Time In Full Service: 0 day 1 hour 15 minutes	Non Secure UDP SIP Trunk Profile KCOM

Figure 11: SIP Trunks List

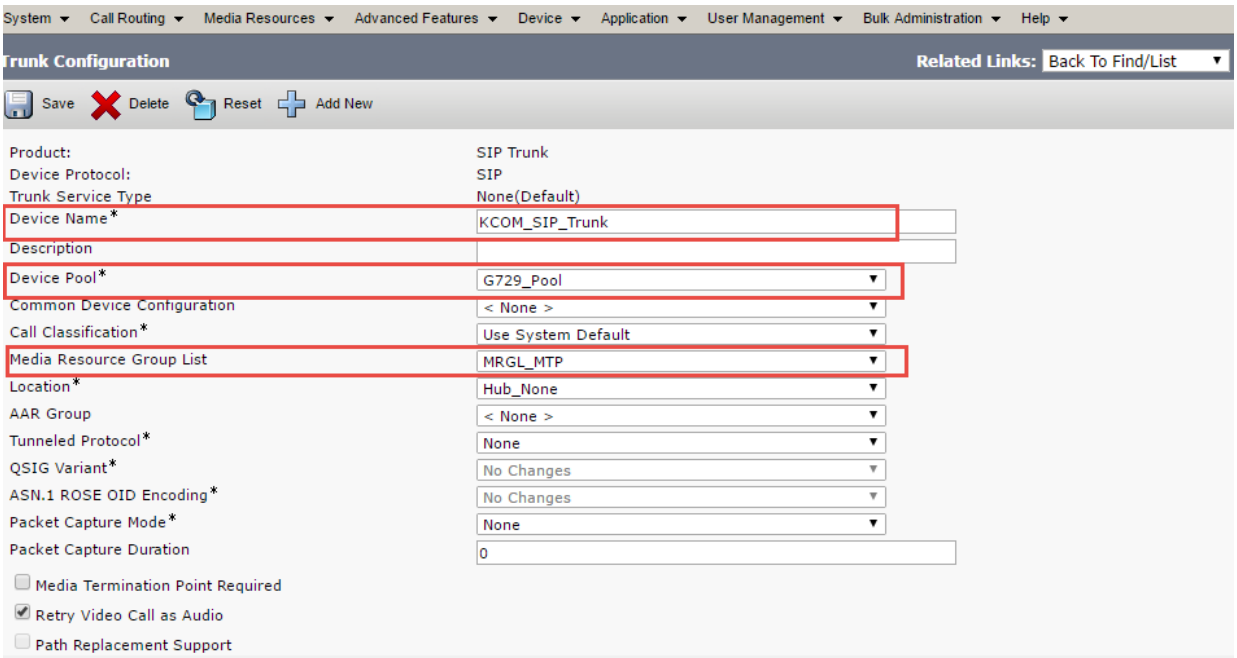


Figure 12: SIP Trunk to Cisco UBE

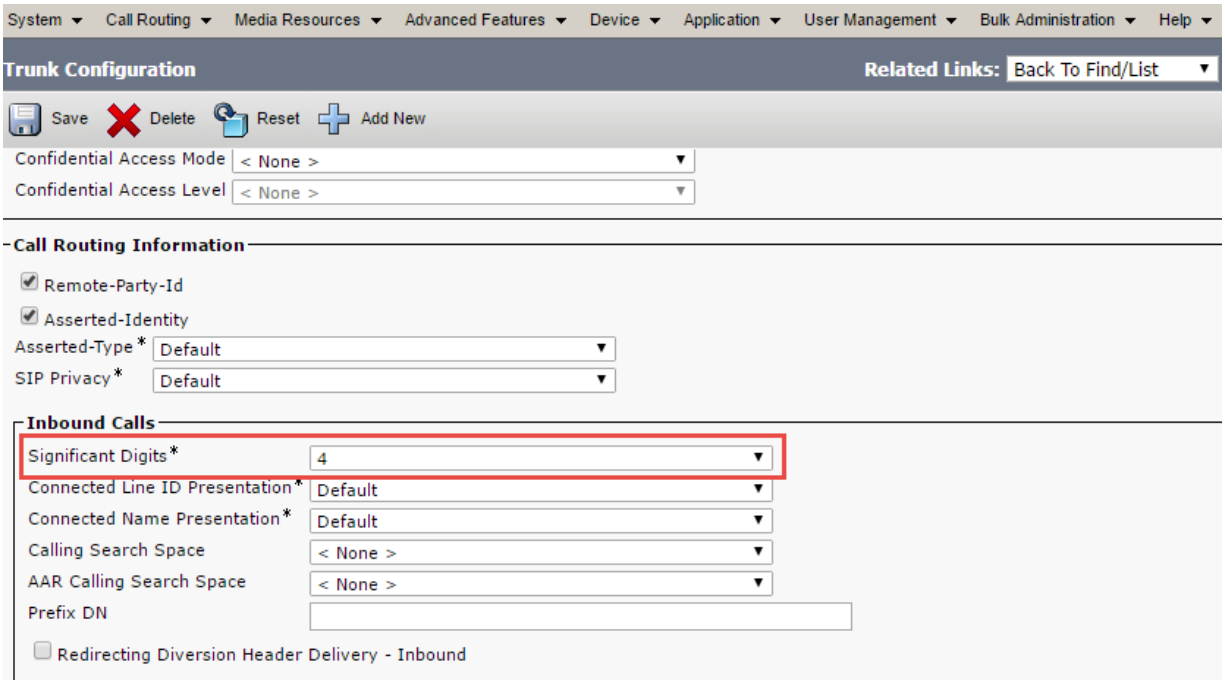


Figure 13: SIP Trunk to Cisco UBE (Cont.)

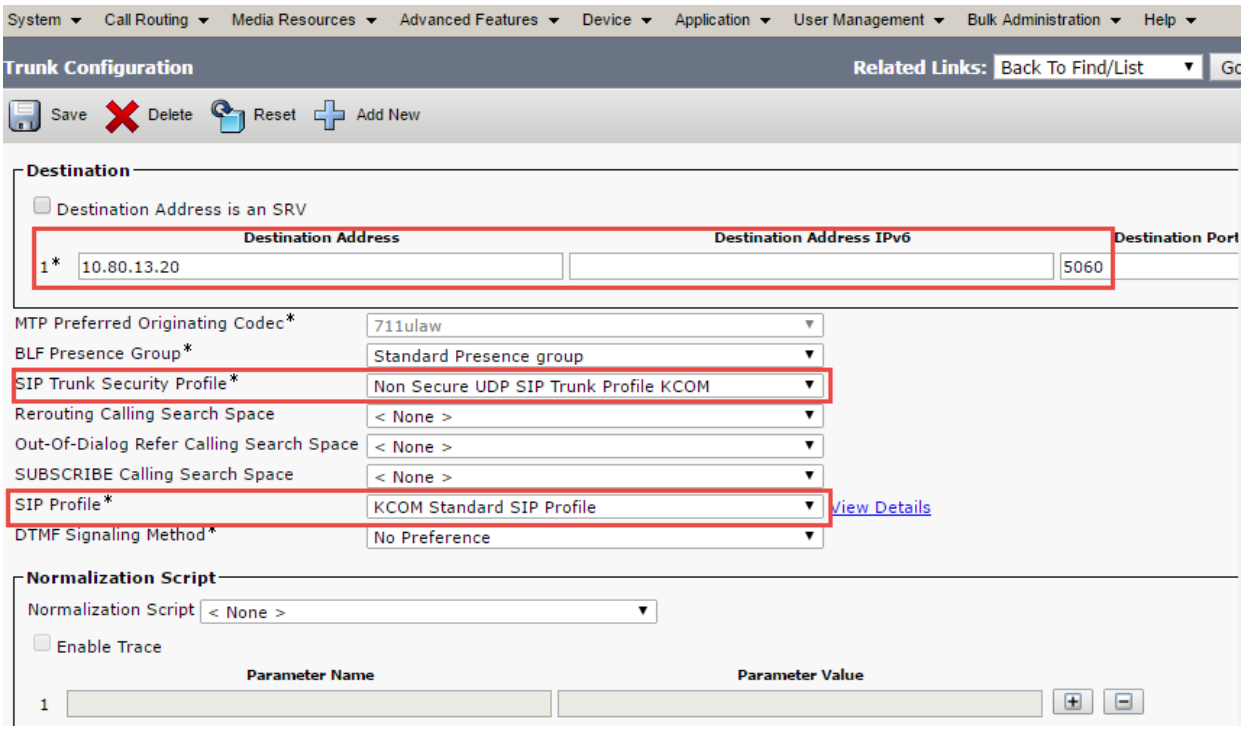


Figure 14: SIP Trunk to Cisco UBE (Cont.)

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

Dial Plan

Route Pattern Configuration

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

Route patterns are configured as follows:

- Cisco IP phone dial “8.!” (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

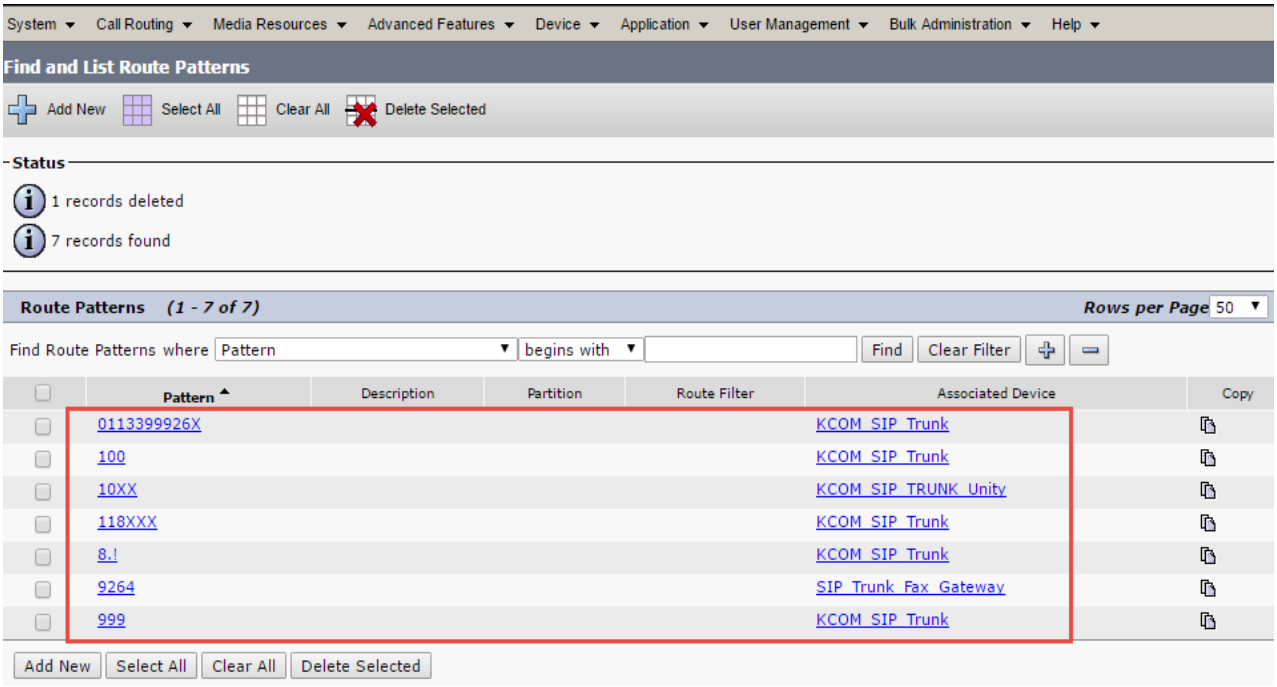


Figure 15: Route Patterns List

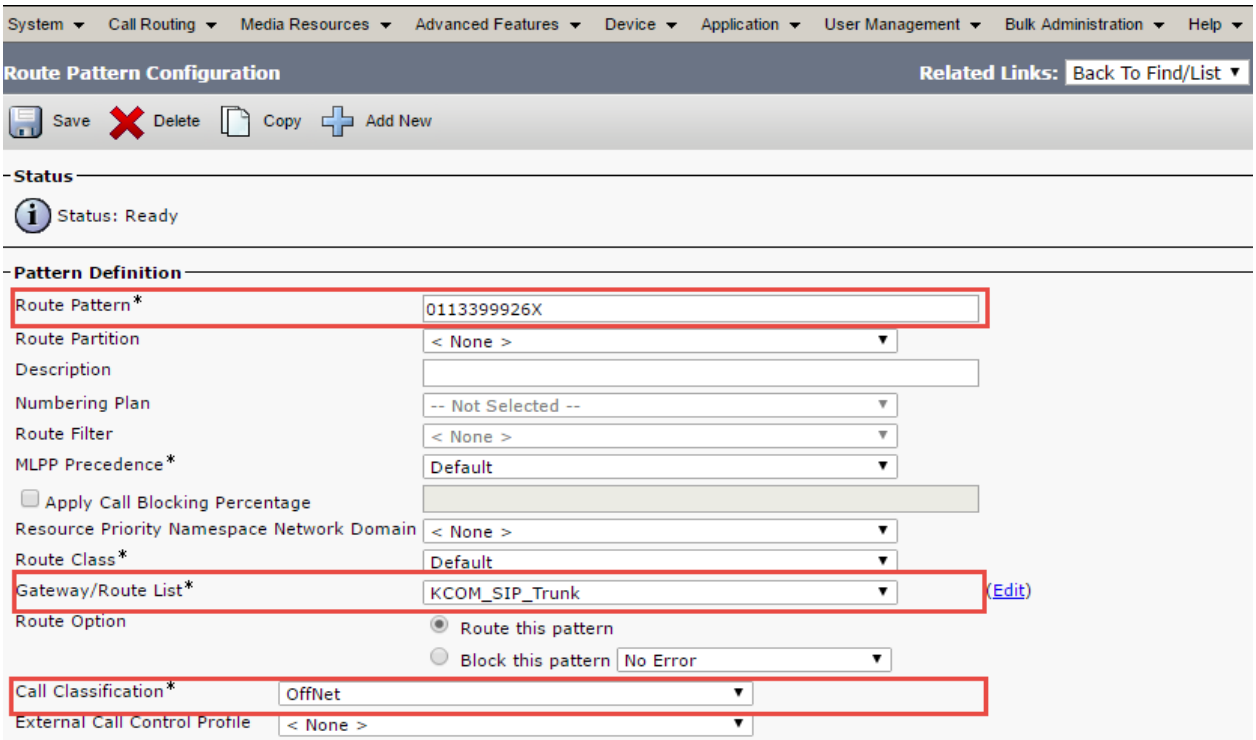


Figure 16: Route Pattern for Voice

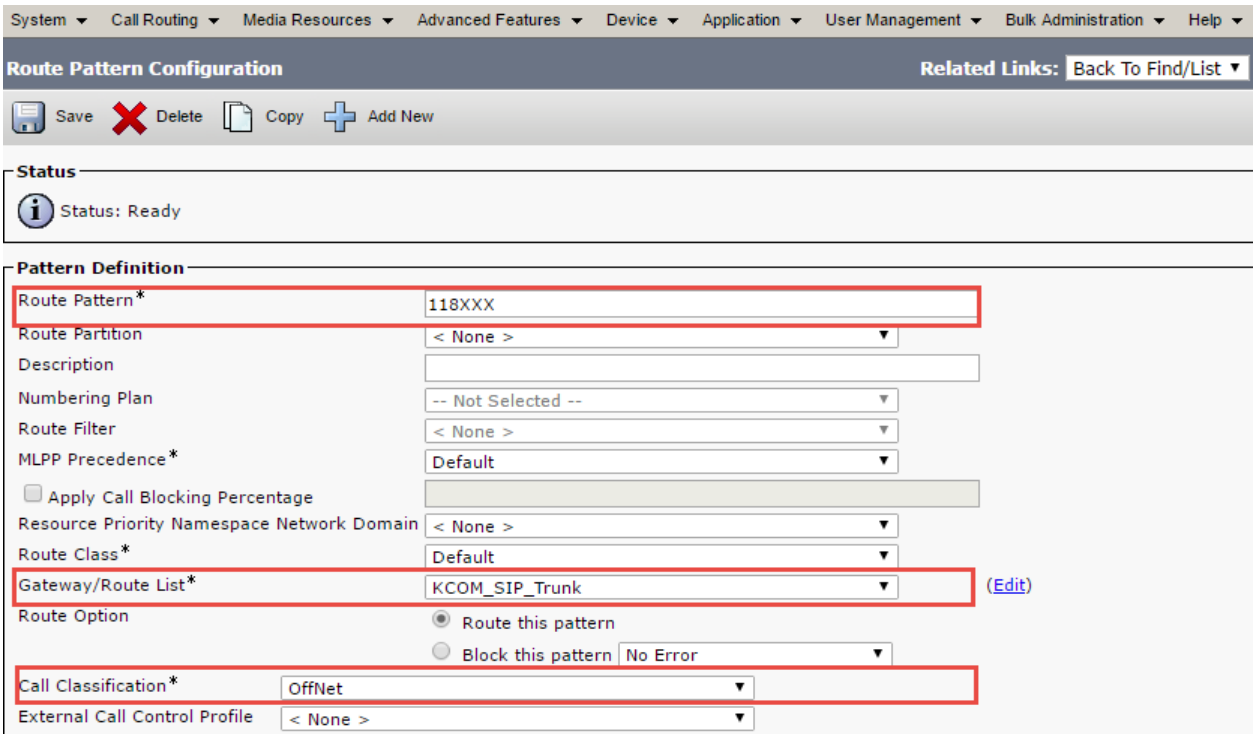


Figure 17: Route Pattern for Voice (Cont.)

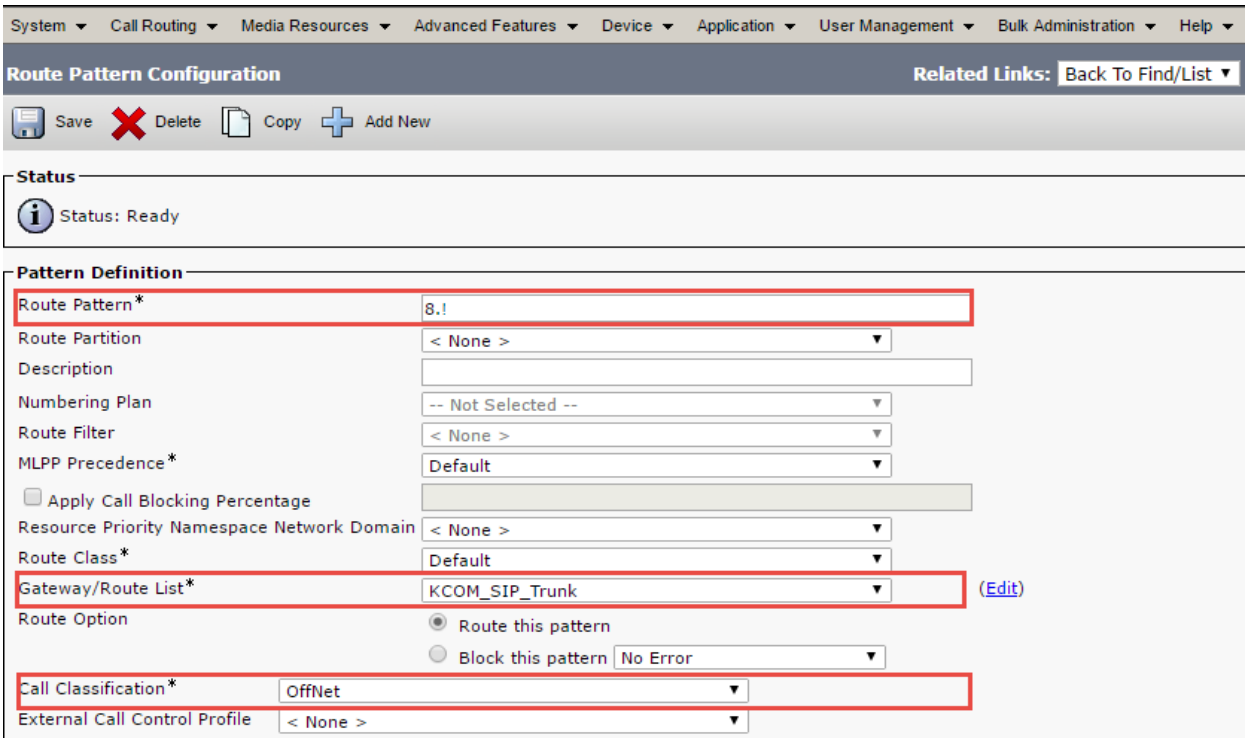


Figure 18: Route Pattern for Voice (Cont.)

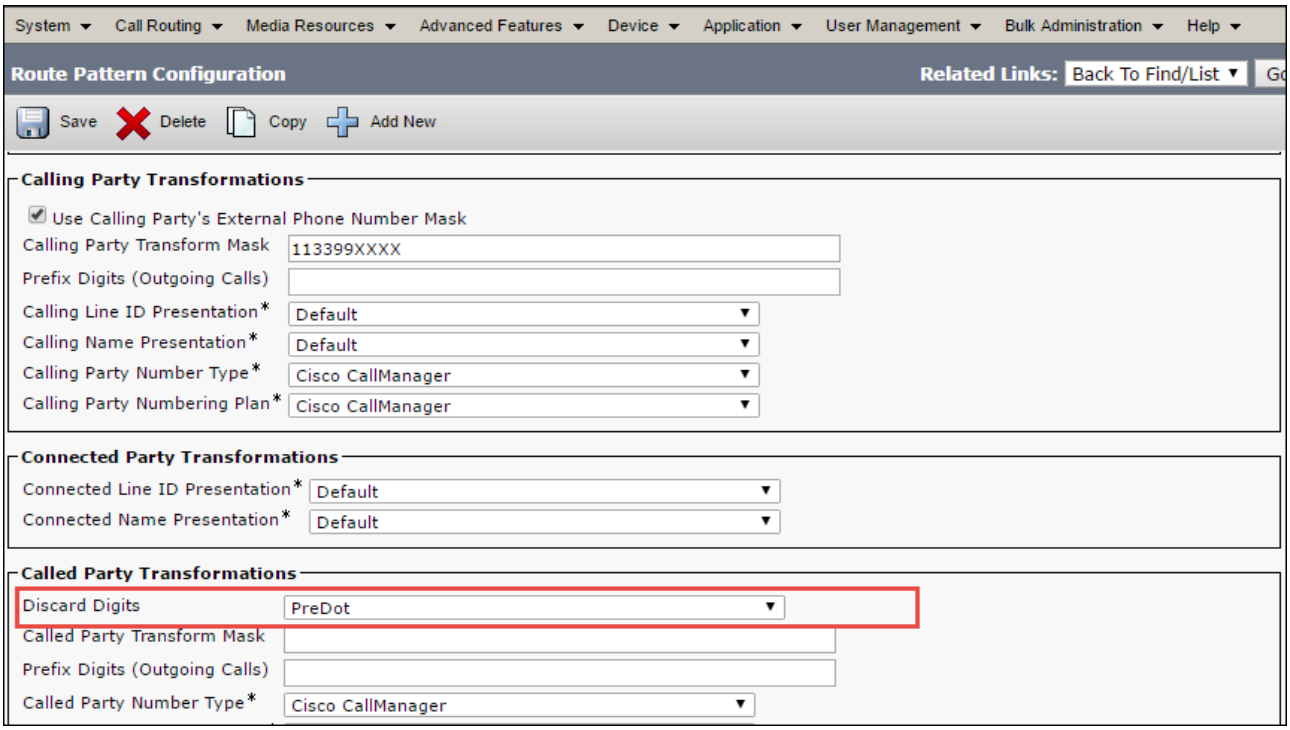


Figure 19: Route Pattern for Voice (Cont.)

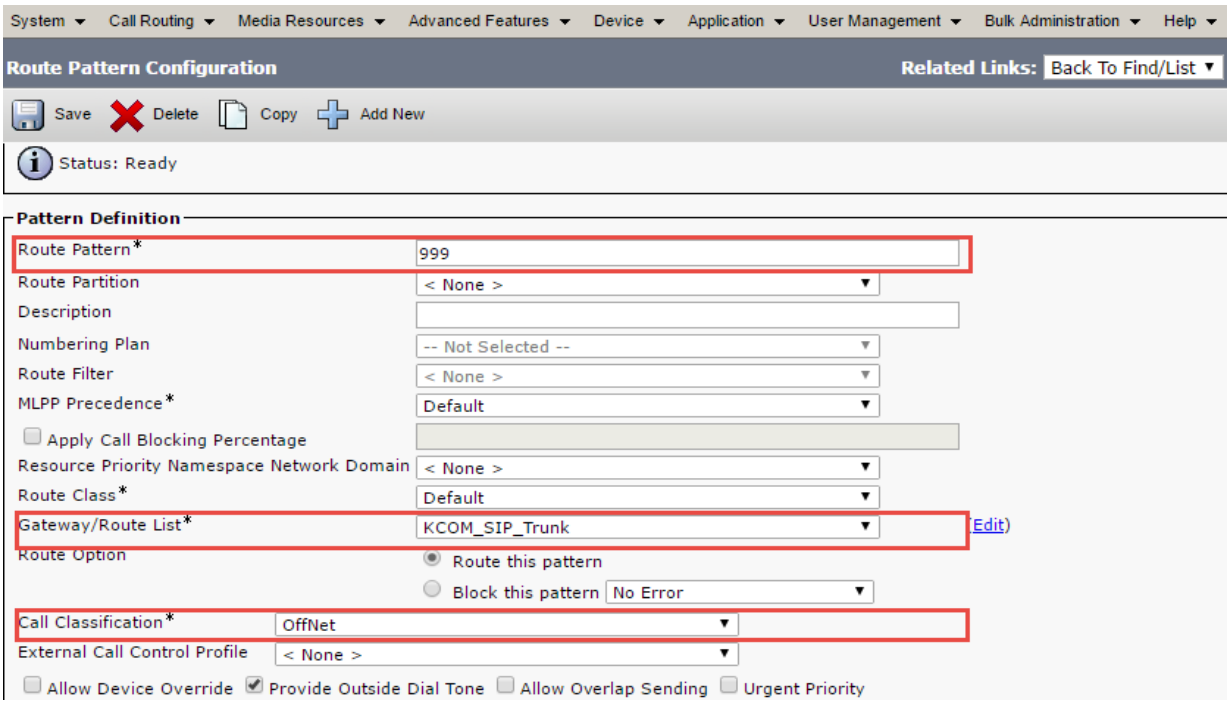


Figure 20: Route Pattern for Voice (Cont.)

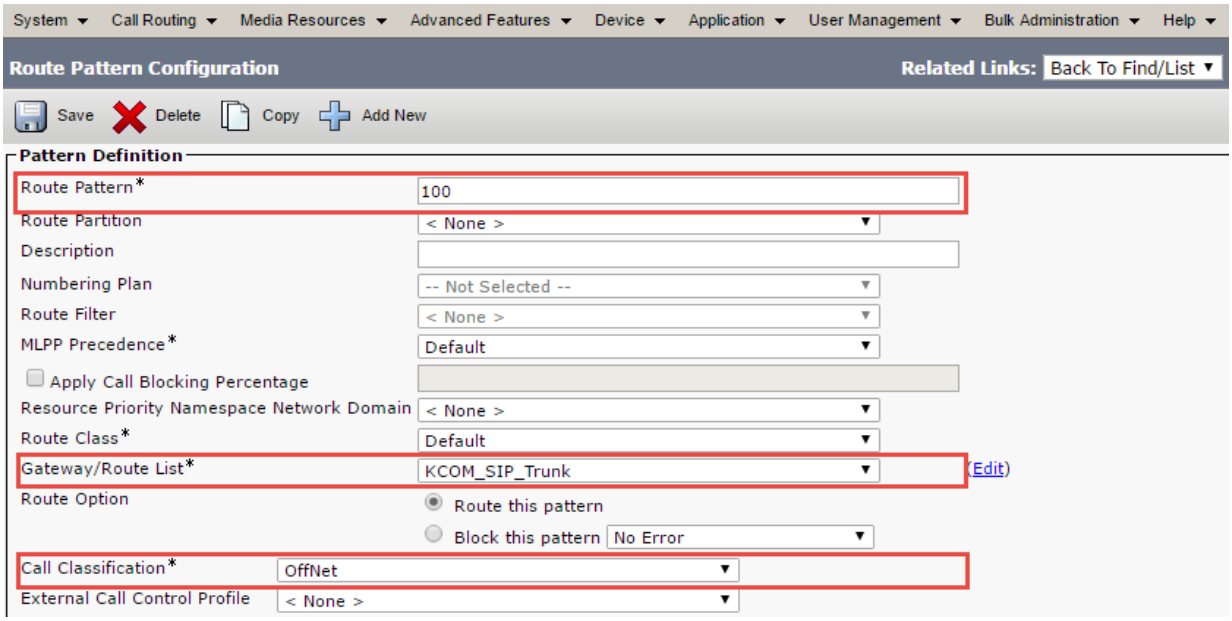


Figure 21: Route Pattern for Voice (Cont.)

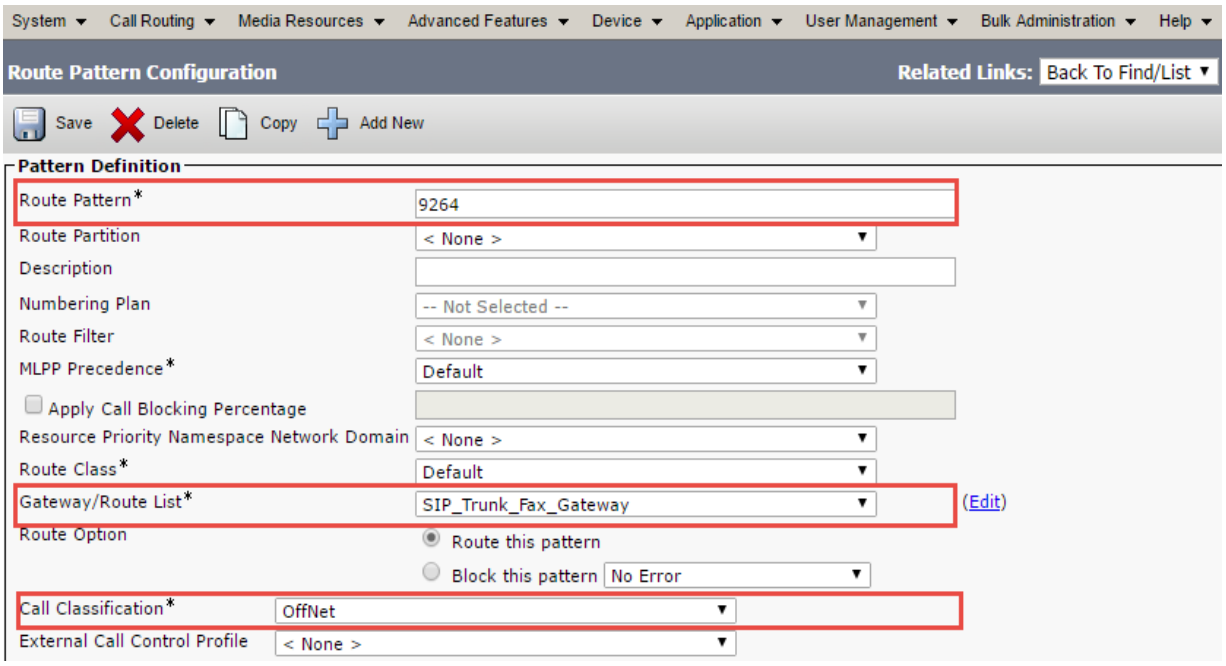


Figure 22: Route Pattern for Voice (Cont.)

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

Acronyms

Acronym	Definitions
CPE	Customer Premise Equipment
Cisco UBE	Cisco Unified Border Element
Cisco UCM	Cisco Unified Communications Manager
MTP	Media Termination Point
POP	Point of Presence
PSTN	Public Switched Telephone Network
ESBC	Enterprise Session Border Controller
SCCP	Skinny Client Control Protocol
SIP	Session Initiation Protocol

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