Application Note

DIDforSale SIP Trunking:

Cisco Unified Communications Manager 12.0.1 with Cisco Unified Border Element (CUBE 12.1.0) on ISR 4331 [IOS-XE – 16.08.01] using SIP

September 2018

Table of Contents

Table of Figures

Introduction

Service Providers today, such as DIDforSale, 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.

DIDforSale 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 DIDforSale network, Cisco Unified Border Element (Cisco UBE) ISR 4331/K9 running IOS-XE 16.08.01 can be used. The Cisco Unified Border Element 16.08.01 provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 12.0.1 connected to DIDforSale 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 DIDforSale 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) 12.0.1 and Cisco Unified Border Element (Cisco UBE) on ISR 4331/K9 [IOS-XE – 16.08.01] connectivity to DIDforSale SIP Trunking service. The deployment model covered in this application note is CPE (Cisco UCM 12.0.1) to PSTN (DIDforSale).
- Testing was performed in accordance to DIDforSale 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 DIDforSale 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 DIDforSale 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

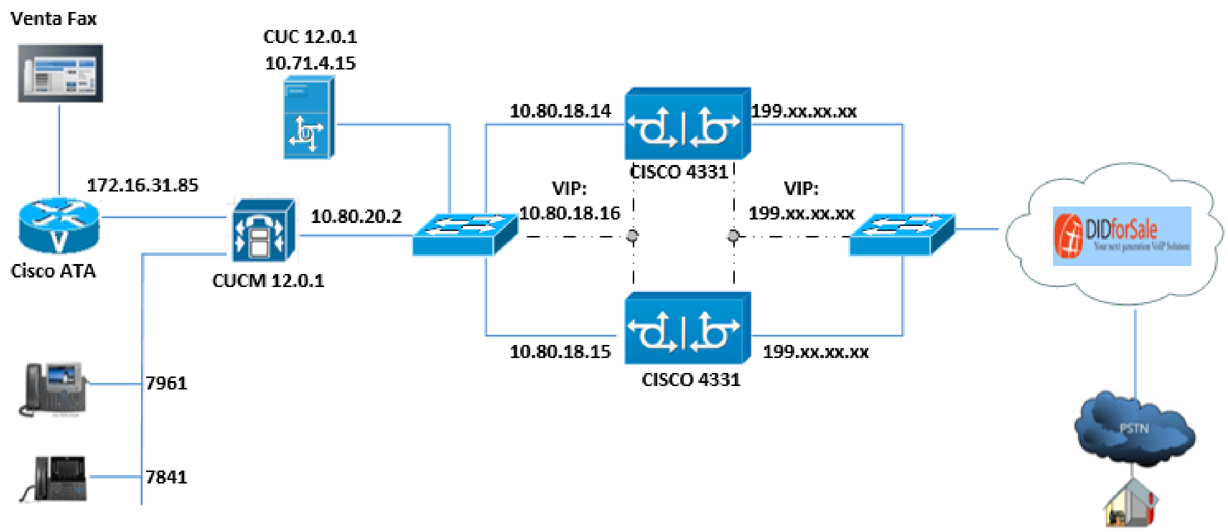


Figure 1: Network Topology

- Cisco IP Phones 7942, 7961 and 7841 phones 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 (Cisco ATA) 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 4331/K9 router as CUBE
- Cisco ATA SPA112
- IP phones 7841 (SIP), 7961 (SCCP) and 7942 (SCCP)

Software Requirements

- Cisco Unified Communications Manager 12.0.1
- Cisco Unity Connection 12.0.1
- IOS-XE 16.08.01 for ISR 4331/K9 Cisco Unified Border Element
- Firmware Version 1.3.5 (004p) for Cisco ATA SPA112

Features

Features Supported

- Incoming and outgoing off-net calls using G711ULaw and 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

- 911 emergency call has been tested with only with G.711 as voice codec.
- CLID is not updated on PSTN phones for call transfer (attended and unattended) to OffNet PSTN scenarios. Caller ID is not updated at PSTN once transfer is completed by PBX.
- Network sends invite without "+" for the inbound call. Network will supports E.164 Numbering Plan for outbound call.
- In midcall re-invite during conference call, CUCM sends invite without SDP to the network and the network responded back with "200 OK "without SDP. With that CUCM is sends "BYE" with cause code=47 (resource unavailable) and terminates the call. This issue has been resolved by adding "voice-class sip midcall-signaling pass-through media-change command in dial-peer. .
- For T.38 FAX, Invite coming from the network is adding "+1" in the "FROM" and "TO" header. For the outbound T.38 FAX, network responds back with "200 Ok " to the initial "INVITE" from the FAX ATA with adding "+1" in the "contact" header. However FAX pages transmitted successfully.

Configuration

Configuring Cisco Unified Border Element

Network Interface

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

```
interface GigabitEthernet0/0/0
 description DIDforsale  CUBE1 WAN
 ip address 199.XXX.XXX.XX 255.255.255.224
 negotiation auto
 redundancy rii 4
 redundancy group 2 ip 199.XXX.XXX.XX exclusive
!
interface GigabitEthernet0/0/1
 description DIDforsale  CUBE1 LAN
 ip address 10.80.18.14 255.255.255.0
 negotiation auto
 redundancy rii 3
 redundancy group 2 ip 10.80.18.16 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
 supplementary-service media-renegotiate
 redirect ip2ip
 fax protocol pass-through g711ulaw
 sip
  session refresh
  asserted-id pai
  privacy pstn
  early-offer forced
  privacy-policy passthru
  g729 annexb-all
  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
pass-thru subscribe-notify-events all	This command is to configure pass-through for all SUBSCRIBE-NOTIFY events

Codecs

G711ulaw is used as the preferred codec for this testing and changed the preferences according to the test plan description

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

Dial Peer

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

```
!
dial-peer voice 222 voip
 description outbound-from PBX-PSTN - LAN facing
 destination-pattern 1626313....
 session protocol sipv2
 session target ipv4:10.80.20.2:5060
 voice-class codec 1
 voice-class sip early-offer forced
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 fax-relay ecm disable
```

```
 fax rate disable
 fax protocol pass-through g711alaw
 no vad
!
dial-peer voice 111 voip
 description Inbound peer match FROM DIDFORSALE
 session protocol sipv2
 incoming uri via trunk1
 voice-class codec 1
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 201 voip
 description Inbound peer match FROM CUCM
 session protocol sipv2
 incoming called-number .T
 voice-class codec 1
 no voice-class sip early-offer forced
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 voice-class sip midcall-signaling passthru media-change
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate 14400
 fax nsf 000000
```

```
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 101 voip
 description Outgoing Call from PBX to PSTN-WAN facing
 destination-pattern .T
 session protocol sipv2
 session server-group 1
 voice-class codec 1
 voice-class sip early-offer forced
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 voice-class sip midcall-signaling passthru media-change
 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, the Cisco UBE presents the full ten-digit DID number to Cisco UCM. The Cisco UCM picks up the last 4 significant Digits configured under SIP Trunk and routes the call based on those 4 digits. 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 “21” 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 “2”. A “2.@” 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. For PBX to PBX via DIDforSale, Caller dial 2 prefix followed by the target 1+10-digits number, 2 was stripped and the remaining digits were send to Cisco UBE, Cisco UBE pass the DID under Dial Peer 101 and send to DIDforSale network which will direct back to Cisco UBE and handled same as normal incoming PSTN call.

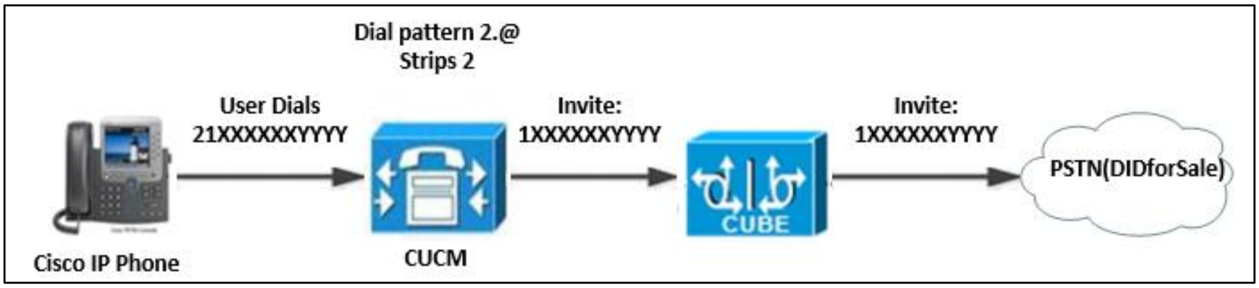


Figure 2: Outbound Voice Call

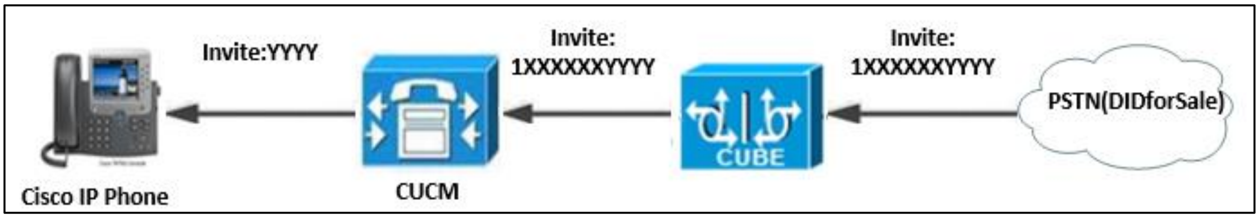


Figure 3: Inbound Voice Call

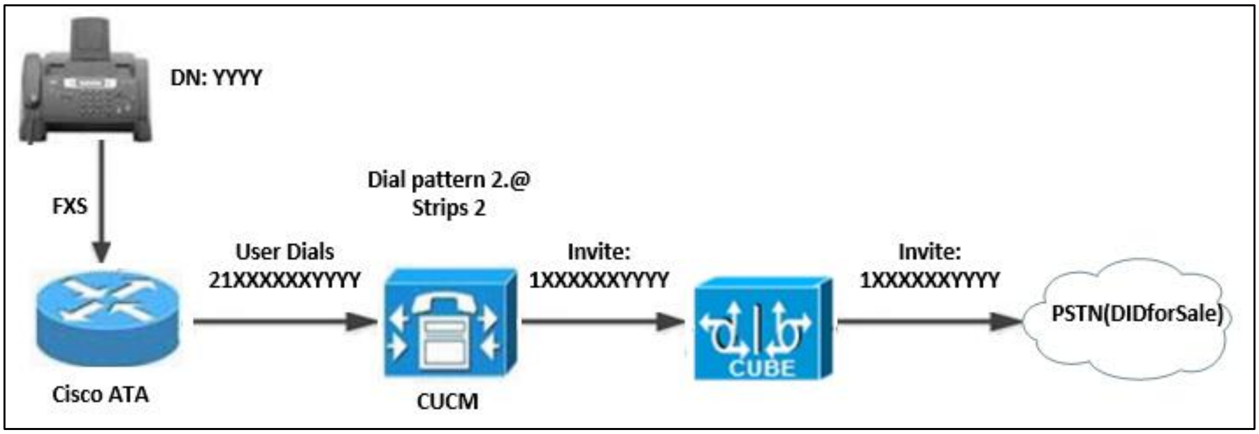


Figure 4: Outbound Fax Call

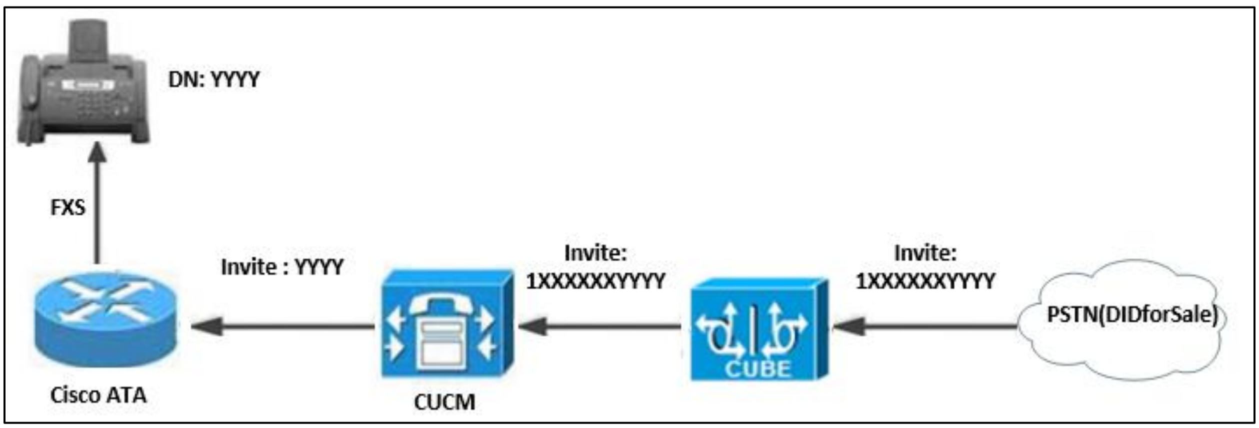


Figure 5: Inbound Fax Call

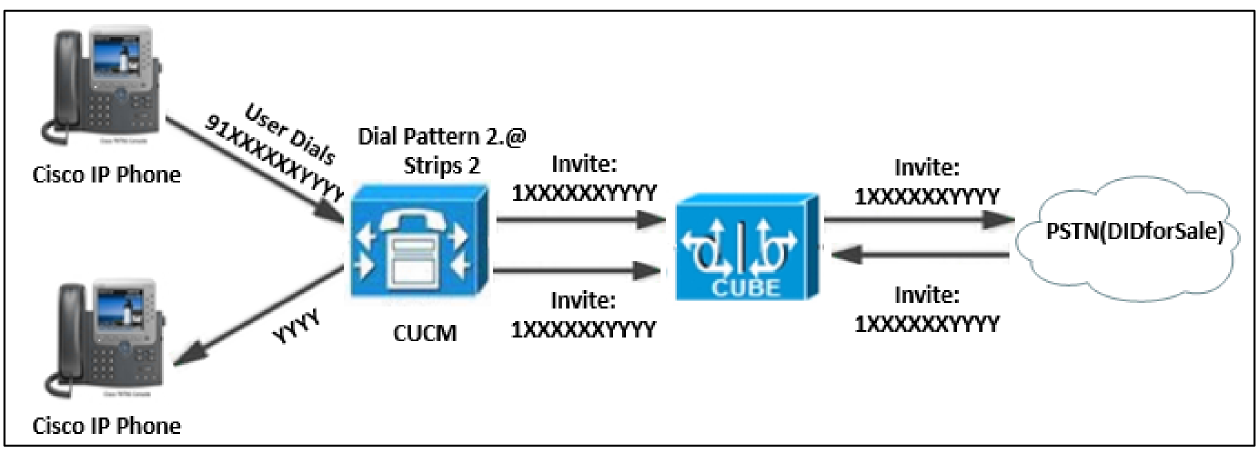


Figure 6: PBX to PBX via DIDforSale Call

Configuration Example

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

Active Cisco UBE

```
cube9#sh run
!
version 16.8
service timestamps debug datetime msec
service timestamps log datetime msec
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname cube9
!
boot-start-marker
boot-end-marker
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
no aaa new-model
subscriber templating
multilink bundle-name authenticated
!
crypto pki trustpoint TP-self-signed-3793611302
```

```
 enrollment selfsigned
 subject-name cn=IOS-Self-Signed-Certificate-3793611302
 revocation-check none
 rsakeypair TP-self-signed-3793611302
!
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
 supplementary-service media-renegotiate
 redirect ip2ip
 fax protocol pass-through g711ulaw
 sip
  session refresh
  asserted-id pai
  privacy pstn
  early-offer forced
  privacy-policy passthru
  g729 annexb-all
  pass-thru subscribe-notify-events all
!
voice class uri trunk1 sip
 host ipv4: 209.XXX.X.XXX
 host ipv4: 209.XXX.X.XXX
 host ipv4: 209.XXX.X.XXX
 host ipv4: 209.XXX.X.XXX
```

```
 host ipv4: 209.XXX.X.XXX
 host ipv4: 209.XXX.XX.XX
 host ipv4: 209.XXX.XX.XX
 host ipv4: 209.XXX.XX.XX
 host ipv4: 209.XXX.XX.XX
 host ipv4: 209.XXX.XX.XX
voice class codec 2
 codec preference 1 g711alaw
 codec preference 2 g711ulaw
!
voice class codec 1
 codec preference 2 g729r8
 codec preference 3 g711alaw
 codec preference 4 g711ulaw
!
voice class server-group 1
 ipv4 209.XXX.X.XXX preference 1
 ipv4 209.XXX.X.XXX preference 1
!
voice translation-rule 1
 rule 1 /\(^..........$\)/ /1\1/
!
!
voice translation-profile coverting_to_11Digits
 translate called 1
!
voice-card 0/4
 no watchdog
!
license udi pid ISR4331/K9 sn FDO41381F1G
```

```
no license smart enable
diagnostic bootup level minimal
!
spanning-tree extend system-id
!
!
!
!
redundancy
 mode none
 application redundancy
  group 2
   name Voice-b2bha_DIDforsale
   priority 100 failover threshold 75
   timers delay 30 reload 60
   control GigabitEthernet0/0/2 protocol 1
   data GigabitEthernet0/0/2
   track 1 shutdown
   track 2 shutdown
!
!
!
track 1 interface GigabitEthernet0/0/0 line-protocol
!
track 2 interface GigabitEthernet0/0/1 line-protocol
!
!
!
!
!
!
```

```
interface GigabitEthernet0/0/0
 description DIDforsale  CUBE1 WAN
 ip address 199.XXX.XXX.XX 255.255.255.224
 shutdown
 negotiation auto
 redundancy rii 4
 redundancy group 2 ip 199.XXX.XXX.XX.26 exclusive
!
interface GigabitEthernet0/0/1
 description DIDforsale  CUBE1 LAN
 ip address 10.80.18.14 255.255.255.0
 negotiation auto
 redundancy rii 3
 redundancy group 2 ip 10.80.18.16 exclusive
!
interface GigabitEthernet0/0/2
 description CUBE HA
 ip address 10.89.20.7 255.255.255.0
 negotiation auto
!
interface Service-Engine0/4/0
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 shutdown
 negotiation auto
!
ip forward-protocol nd
ip http server
ip http authentication local
```

```
ip http secure-server
ip tftp source-interface GigabitEthernet0
ip route 0.0.0.0 0.0.0.0 199.182.124.1
ip route 10.64.0.0 255.255.0.0 10.80.18.1
ip route 172.16.24.0 255.255.248.0 10.80.18.1
!
!
!
!
!
!
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 Inbound peer match FROM CUCM
 session protocol sipv2
 incoming called-number .T
 voice-class codec 1
 no voice-class sip early-offer forced
 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 101 voip
 description Outgoing Call from PBX to PSTN-WAN facing
 destination-pattern .T
 session protocol sipv2
 session server-group 1
 voice-class codec 1
 voice-class sip early-offer forced
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 voice-class sip midcall-signaling passthru media-change
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 222 voip
 description outbound-from PBX-PSTN - LAN facing
 destination-pattern 1626313....
 session protocol sipv2
 session target ipv4:10.80.20.2:5060
 voice-class codec 1
 voice-class sip early-offer forced
```

```
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 111 voip
 description Inbound peer match FROM DIDFORSALE
 session protocol sipv2
 incoming uri via trunk1
 voice-class codec 1
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 203 voip
 description Inbound peer match FROM CUCM
 shutdown
 session protocol sipv2
 incoming called-number .T
 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 protocol pass-through g711ulaw
 no vad
!
dial-peer voice 1010 voip
 description Outgoing Call from PBX to PSTN-WAN facing
 shutdown
 destination-pattern 12T
 session protocol sipv2
 session target ipv4:209.XXX.XX.XX:5060
 voice-class codec 1
 voice-class sip early-offer forced
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 1000 voip
 description PBX to PBX call via DIDforsale - WAN facing
 translation-profile outgoing coverting_to_11Digits
 destination-pattern 626313....
 session protocol sipv2
 session target ipv4: 209.XXX.XX.XX: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 g711ulaw
 no vad
!
dial-peer voice 1002 voip
 description PBX to PBX call via DIDforsale-SecondTrunk - WAN facing
 translation-profile outgoing coverting_to_11Digits
 destination-pattern 626313....
 session protocol sipv2
 session target ipv4: 209.XXX.XX.XX: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 g711ulaw
 no vad
!
sip-ua
!
line con 0
 exec-timeout 0 0
login
 transport input none
 stopbits 1
line aux 0
```

```
 stopbits 1
line vty 0 4
 exec-timeout 0 0
 password tekV1z10n
 login
 transport input telnet
!
wsma agent exec
!
wsma agent config
!
wsma agent filesys
!
wsma agent notify
!
!
end

cube9#
```

Standby Cisco UBE

```
CUBE10#sh run
!
version 16.8
service timestamps debug datetime msec
service timestamps log datetime msec
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname CUBE10
!
boot-start-marker
boot-end-marker
!
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
no aaa new-model
subscriber templating
multilink bundle-name authenticated
!
```

```
crypto pki trustpoint TP-self-signed-2930804041
 enrollment selfsigned
 subject-name cn=IOS-Self-Signed-Certificate-2930804041
 revocation-check none
 rsakeypair TP-self-signed-2930804041
!
crypto pki certificate chain TP-self-signed-2930804041
!
voice service voip
 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
 supplementary-service media-renegotiate
 redirect ip2ip
 fax protocol pass-through g711ulaw
 sip
  session refresh
  asserted-id pai
  privacy pstn
  early-offer forced
  privacy-policy passthru
  g729 annexb-all
  pass-thru subscribe-notify-events all
!
```

```
!
voice class uri trunk1 sip
 host ipv4: 209.XXX.X.XXX
 host ipv4: 209.XXX.X.XXX
 host ipv4: 209.XXX.X.XXX
 host ipv4: 209.XXX.X.XXX
 host ipv4: 209.XXX.X.XXX
 host ipv4: 209.XXX.XX.XX
 host ipv4: 209.XXX.XX.XX
 host ipv4: 209.XXX.XX.XX
 host ipv4: 209.XXX.XX.XX
 host ipv4: 209.XXX.XX.XX
voice class codec 1
 codec preference 1 g729r8
 codec preference 2 g711alaw
 codec preference 3 g711ulaw
!
voice class codec 2
 codec preference 1 g711alaw
 codec preference 2 g711ulaw
!
voice class server-group 1
 ipv4 209.XXX.XX.XX preference 1
 ipv4 209.XXX.X.XXX preference 1
!
voice translation-rule 1
 rule 1 /\(^..........$\)/ /1\1/
!
```

```
!
voice translation-profile coverting_to_11Digits
 translate called 1
!
voice-card 0/4
 no watchdog
!
license udi pid ISR4331/K9 sn FDO41381F17
no license smart enable
diagnostic bootup level minimal
!
spanning-tree extend system-id
!
!
!
!
redundancy
 mode none
 application redundancy
  group 2
   name Voice-b2bha_DIDforsale
   priority 100 failover threshold 75
   timers delay 30 reload 60
   control GigabitEthernet0/0/2 protocol 1
   data GigabitEthernet0/0/2
   track 1 shutdown
   track 2 shutdown
!
```

```
!
!
track 1 interface GigabitEthernet0/0/0 line-protocol
!
track 2 interface GigabitEthernet0/0/1 line-protocol
!
!
!
!
!
!
interface GigabitEthernet0/0/0
 description DIDforsale  CUBE1 WAN
 ip address 199.XXX.XXX.XX 255.255.255.224
 negotiation auto
 redundancy rii 4
 redundancy group 2 ip 199.XXX.XXX.XX exclusive
!
interface GigabitEthernet0/0/1
 description DIDforsale CUBE1 LAN
 ip address 10.80.18.15 255.255.255.0
 negotiation auto
 redundancy rii 3
 redundancy group 2 ip 10.80.18.16 exclusive
!
interface GigabitEthernet0/0/2
 description CUBE HA
 ip address 10.89.20.8 255.255.255.0
```

```
 negotiation auto
!
interface Service-Engine0/4/0
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 shutdown
 negotiation auto
!
ip forward-protocol nd
ip http server
ip http authentication local
ip http secure-server
ip http client source-interface GigabitEthernet0/0/0
ip route 0.0.0.0 0.0.0.0 199.182.124.1
ip route 10.64.0.0 255.255.0.0 10.80.18.1
ip route 172.16.24.0 255.255.248.0 10.80.18.1
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 1010 voip
 description Outgoing Call from PBX to PSTN-WAN facing
 shutdown
 destination-pattern .T
 session protocol sipv2
 session target ipv4:209.XXX.XX.XX:5060
 voice-class codec 1
 voice-class sip early-offer forced
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 101 voip
 description Outgoing Call from PBX to PSTN-WAN facing
 destination-pattern .T
 session protocol sipv2
 session server-group 1
 incoming called-number 0T
 voice-class codec 1
 voice-class sip asserted-id pai
```

```
 voice-class sip privacy-policy passthru
 voice-class sip early-offer forced
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 voice-class sip midcall-signaling passthru media-change
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 222 voip
 description outbound-from PBX-PSTN - LAN facing
 destination-pattern 1626313....
 session protocol sipv2
 session target ipv4:10.80.20.2:5060
 voice-class codec 1
 voice-class sip early-offer forced
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 voice-class sip midcall-signaling passthru media-change
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 111 voip
```

```
 description Inbound peer match FROM DIDFORSALE
 session protocol sipv2
 incoming uri via trunk1
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 voice-class sip midcall-signaling passthru media-change
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 1000 voip
 description PBX to PBX call via DIDforsale - WAN facing
 translation-profile outgoing coverting_to_11Digits
 destination-pattern 626313....
 session protocol sipv2
 session server-group 1
 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 g711ulaw
!
dial-peer voice 201 voip
```

```
 description Inbound peer match FROM CUCM
 session protocol sipv2
 incoming called-number .T
 voice-class codec 1
 voice-class sip bind control source-interface GigabitEthernet0/0/1
 voice-class sip bind media source-interface GigabitEthernet0/0/1
 voice-class sip midcall-signaling passthru media-change
dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax protocol pass-through g711ulaw
 no vad
!
dial-peer voice 203 voip
 description Inbound peer match FROM CUCM
 session protocol sipv2
 incoming called-number .T
 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 g711ulaw
 no vad
!
!
sip-ua
!
```

```
!
line con 0
 exec-timeout 0 0
 password tekV1z10n
 login
 transport input none
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 exec-timeout 0 0
login
 transport input telnet
!
ntp server 34.208.249.133
wsma agent exec
!
wsma agent config
!
wsma agent filesys
wsma agent notify
!
pnp profile pnp_redirection_profile
 transport http ipv4 127.0.0.1 port 80
end
```

Configuring Cisco Unified Communications Manager

Cisco UCM Version

Figure 7: Cisco UCM Version

Cisco Call Manager Service Parameters

Navigation path: System > Service Parameters

Select Server* = Clus20pub1--CUCM Voice/Video (Active)

Select Service*= Cisco CallManager (Active)

All other fields are set to default values

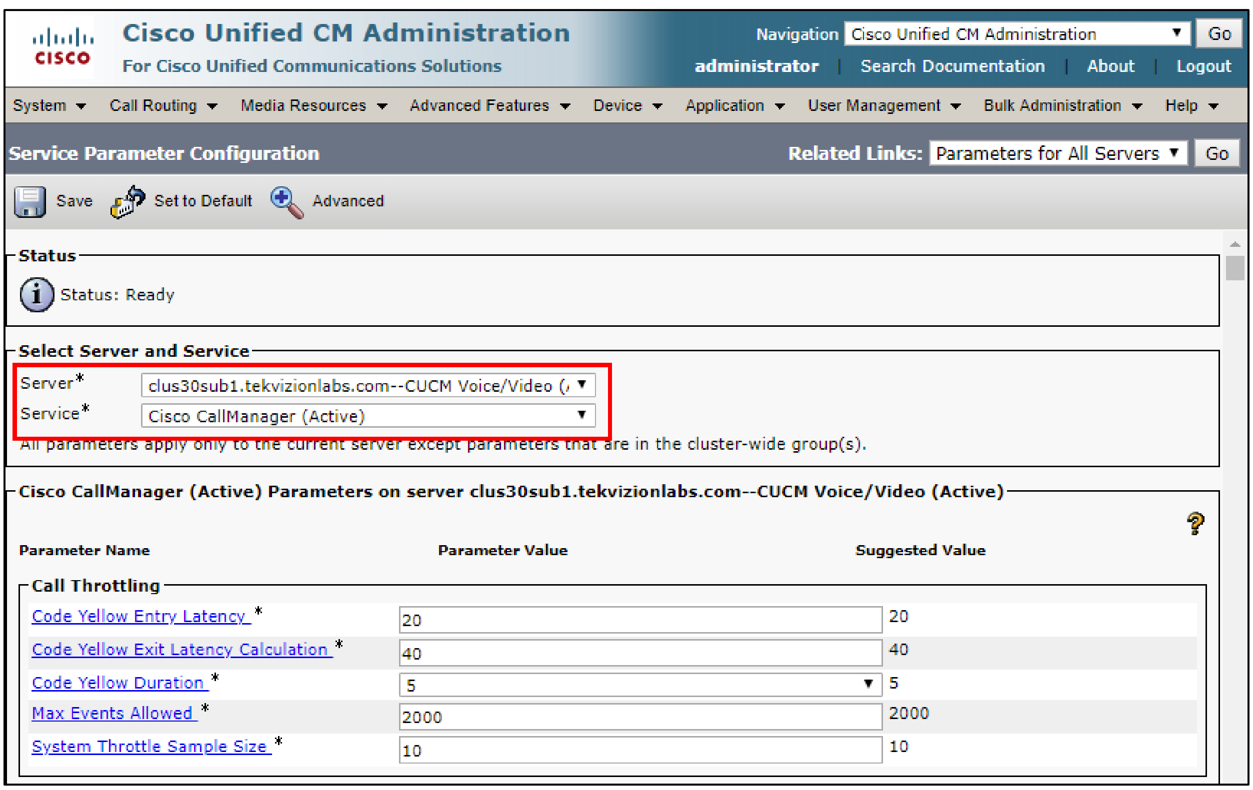


Figure 8: Service Parameters

Offnet Calls via DIDforSale SIP Trunk

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

SIP Trunk Security Profile

Navigation Path: System > Security > SIP Trunk Security Profile
Name*= non secure UDP sip trunk profile for DIDforSale

Description = Non Secure SIP Trunk Profile

(Note:- For this test the Outgoing Transport type was provisioned as UDP. DIDforsale supports TCP as well.)

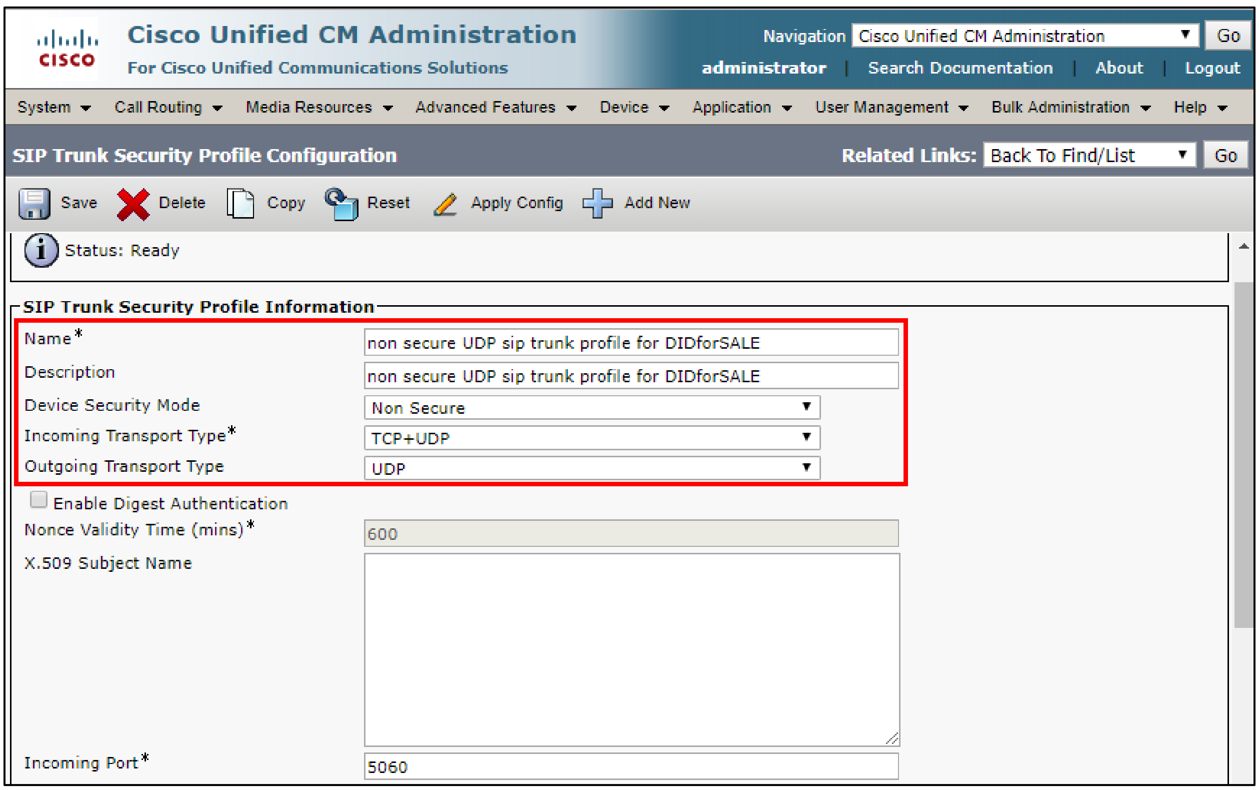


Figure 9: SIP Trunk Security Profile

Explanation

Parameter	Value	Description
Incoming Transport Type	TCP + UDP	
Outgoing Transport Type	UDP	SIP trunks to DIDforSale, SBC should use UDP as a transport protocol for SIP. This is configured

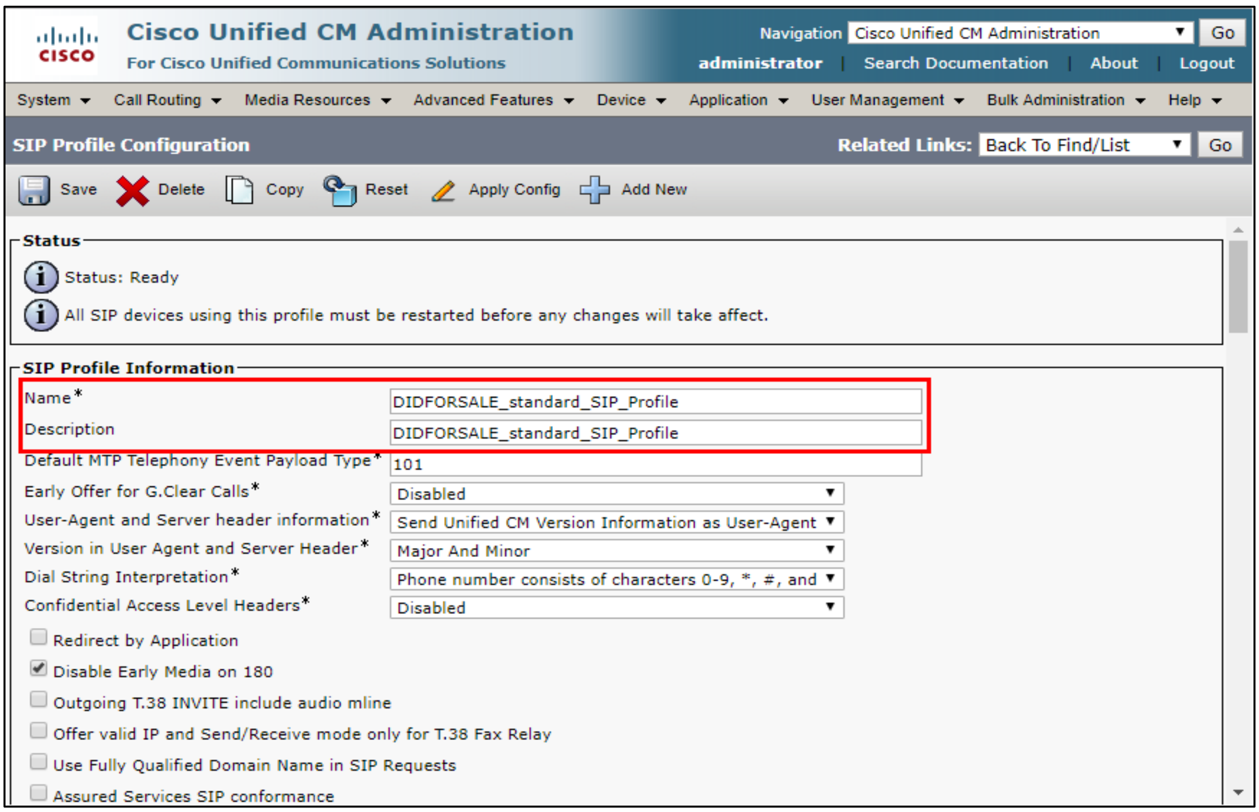
		using SIP Trunk Security profile, which is later assigned to the SIP trunk itself.

SIP Profile Configuration

SIP Profile will be later associated with the SIP trunk
Navigation Path: Device > Device Settings > SIP Profile

Name*= DIDFORSALE_standard_SIP_profile

Description = DIDFORSALE_standard_SIP_profile



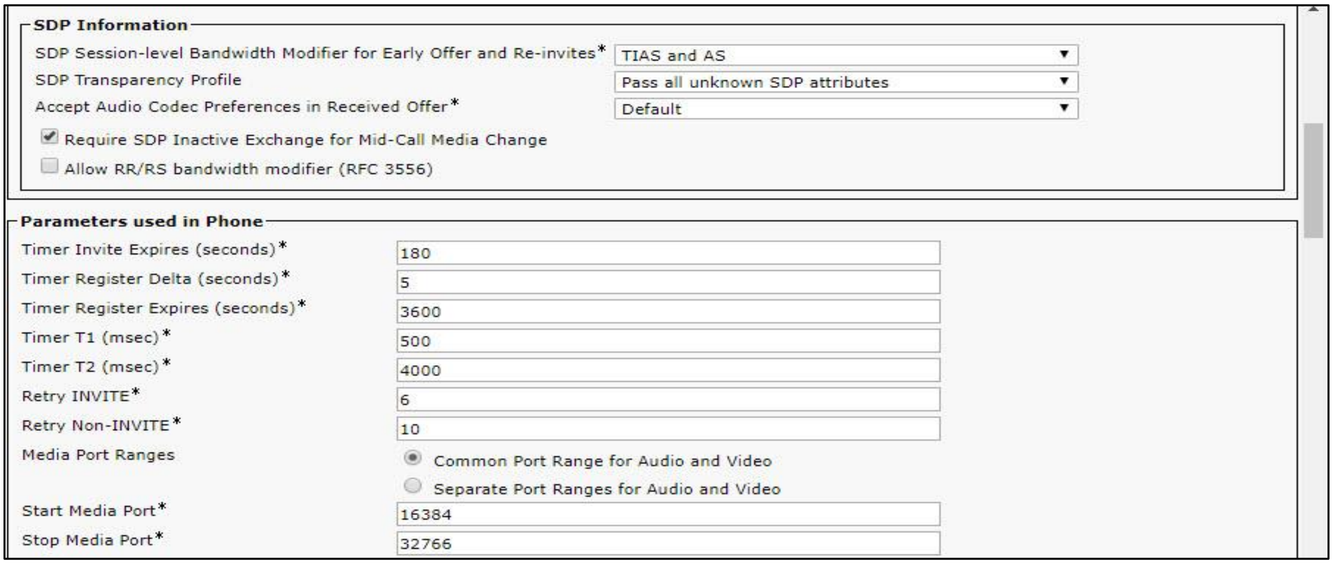


Figure 10: SIP Profile

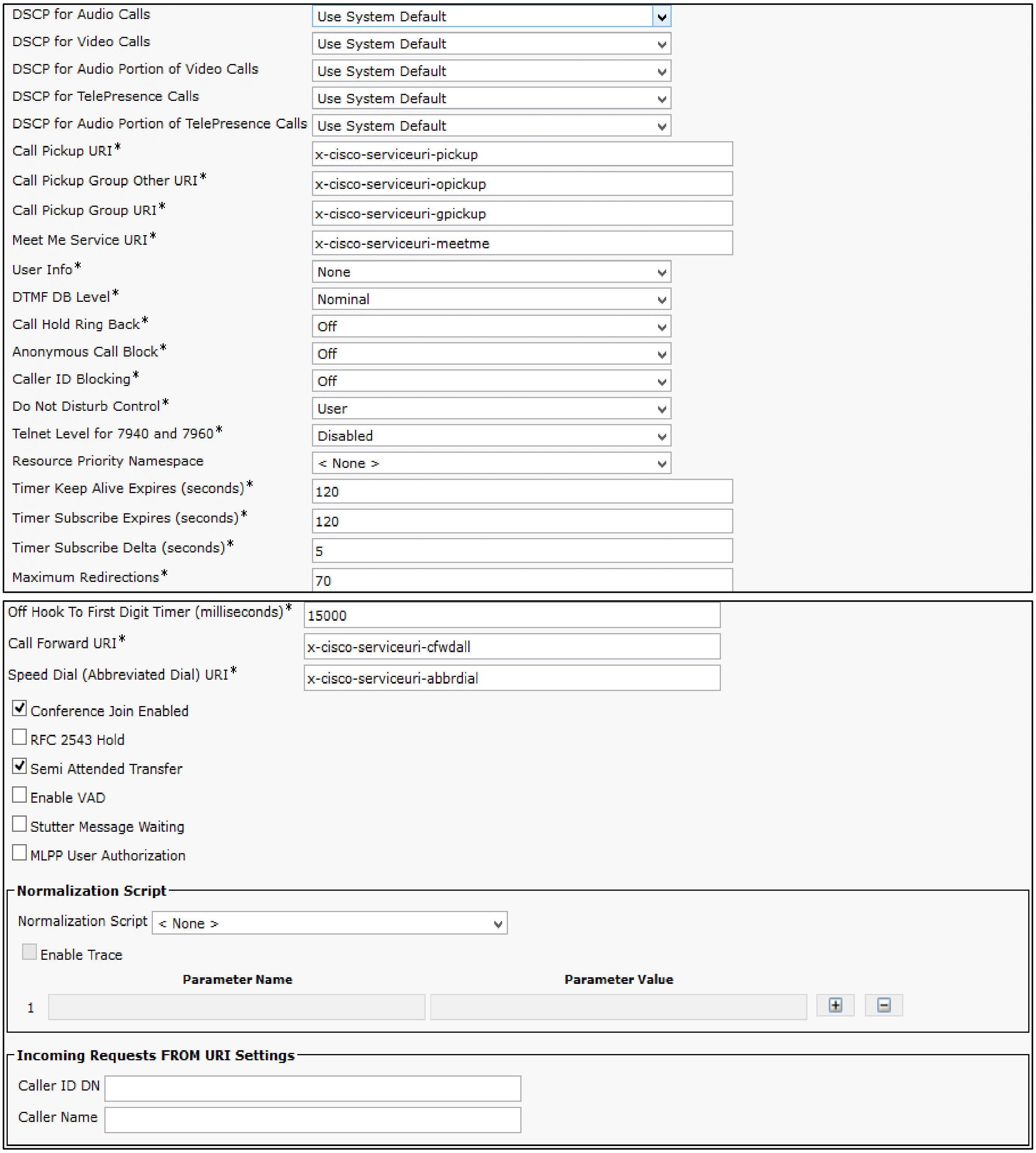


Figure 11: SIP Profile (Cont.)

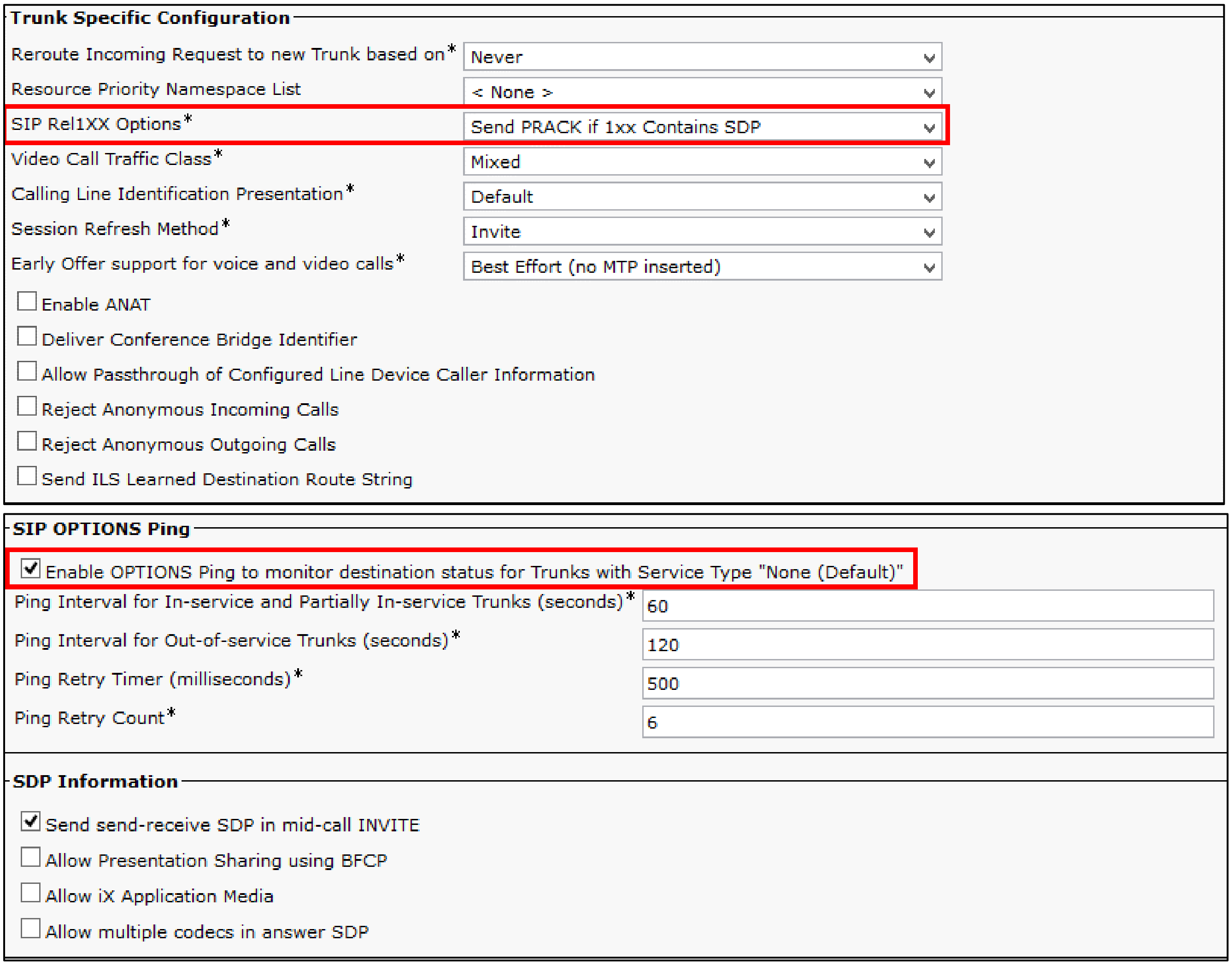


Figure 12: SIP Profile (Cont.)

Explanation

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)	60	OPTIONS message parameters- interval time
Ping Interval for Out-of-service Trunks (seconds)	120	OPTIONS message parameters- interval time

Device Pool Configuration

Device Pool will be later associated with the SIP trunk
Navigation Path: System > Device Pool

Name*= DIDforsale_Codec G711_devicepool

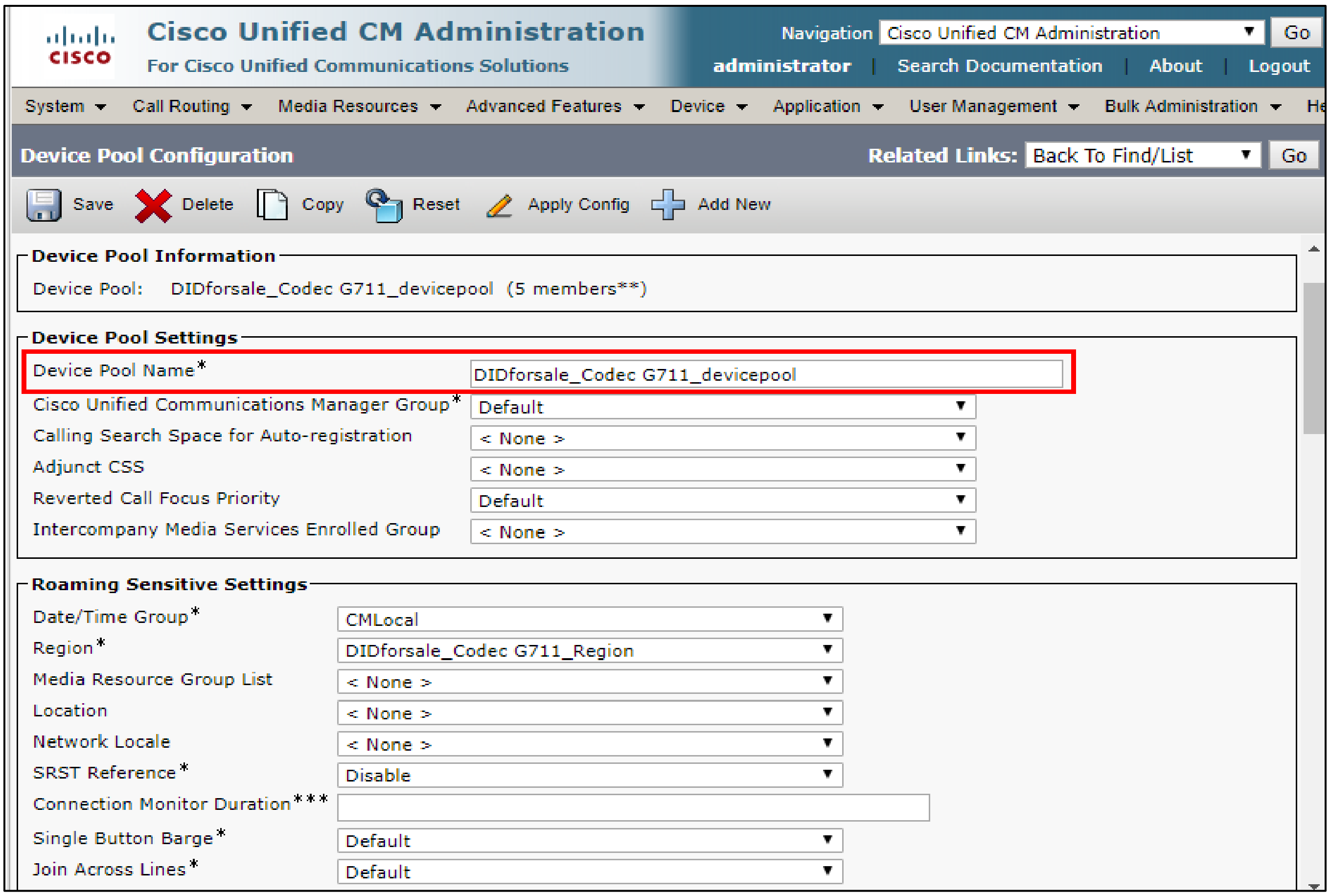


Figure 13: Device Pool

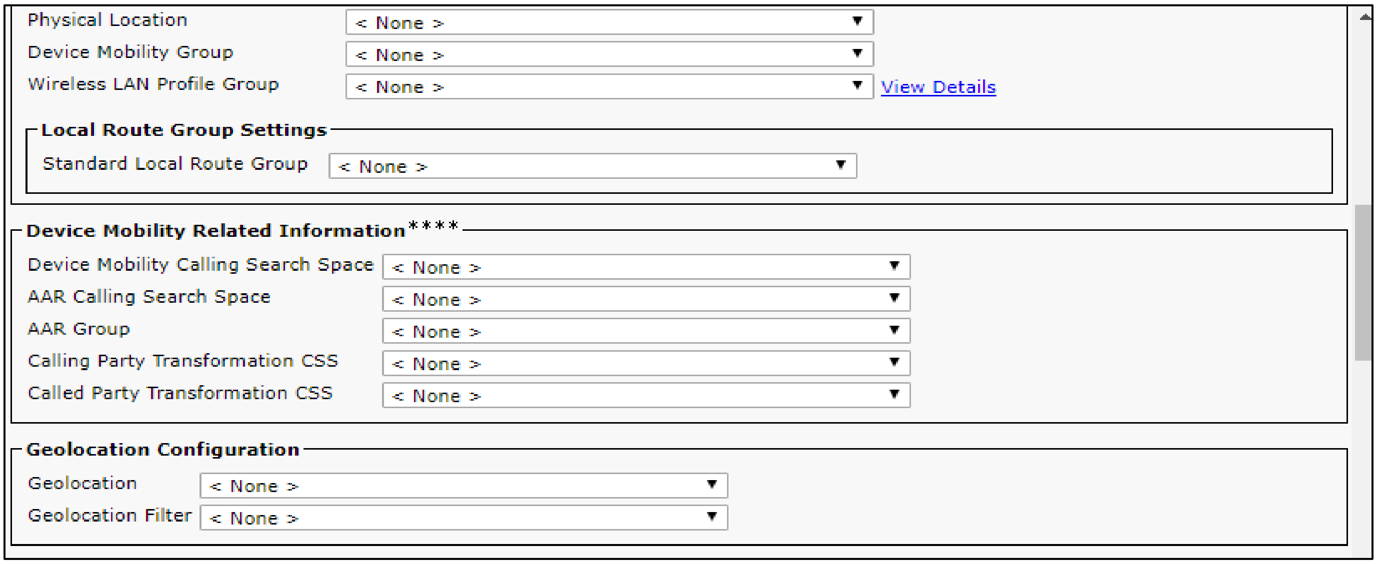
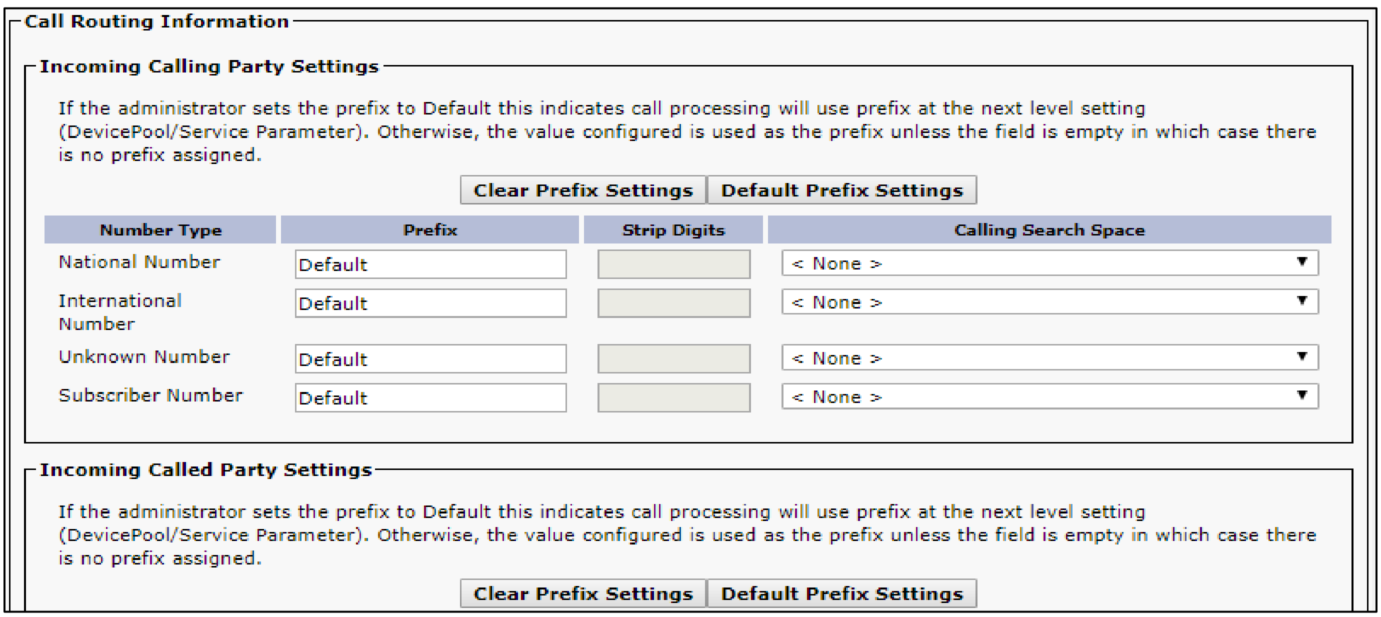
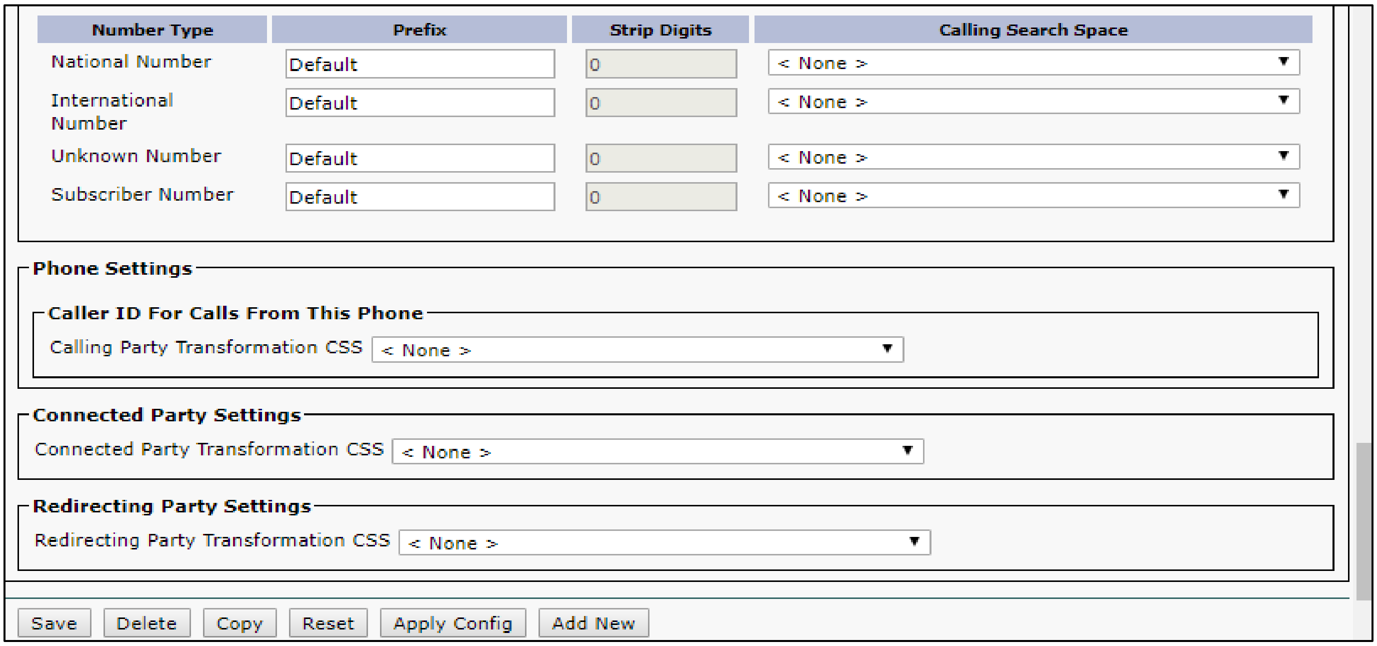


Figure 13: Device Pool (cont.)




Number Type
Prefix
Strip Digits
Calling Search Space
National Number
Default
0
< None >
International Number
Default
0
< None >
Unknown Number
Default
0
< None >
Subscriber Number
Default
0
< None >
Phone Settings
Caller ID For Calls From This Phone
Calling Party Transformation CSS
< None >
Connected Party Settings
Connected Party Transformation CSS
< None >
Redirecting Party Settings
Redirecting Party Transformation CSS
< None >
Save
Delete
Copy
Reset
Apply Config
Add New

SIP Trunk Configuration

Create SIP trunks to Cisco UBE

Navigation Path: Device > Trunk

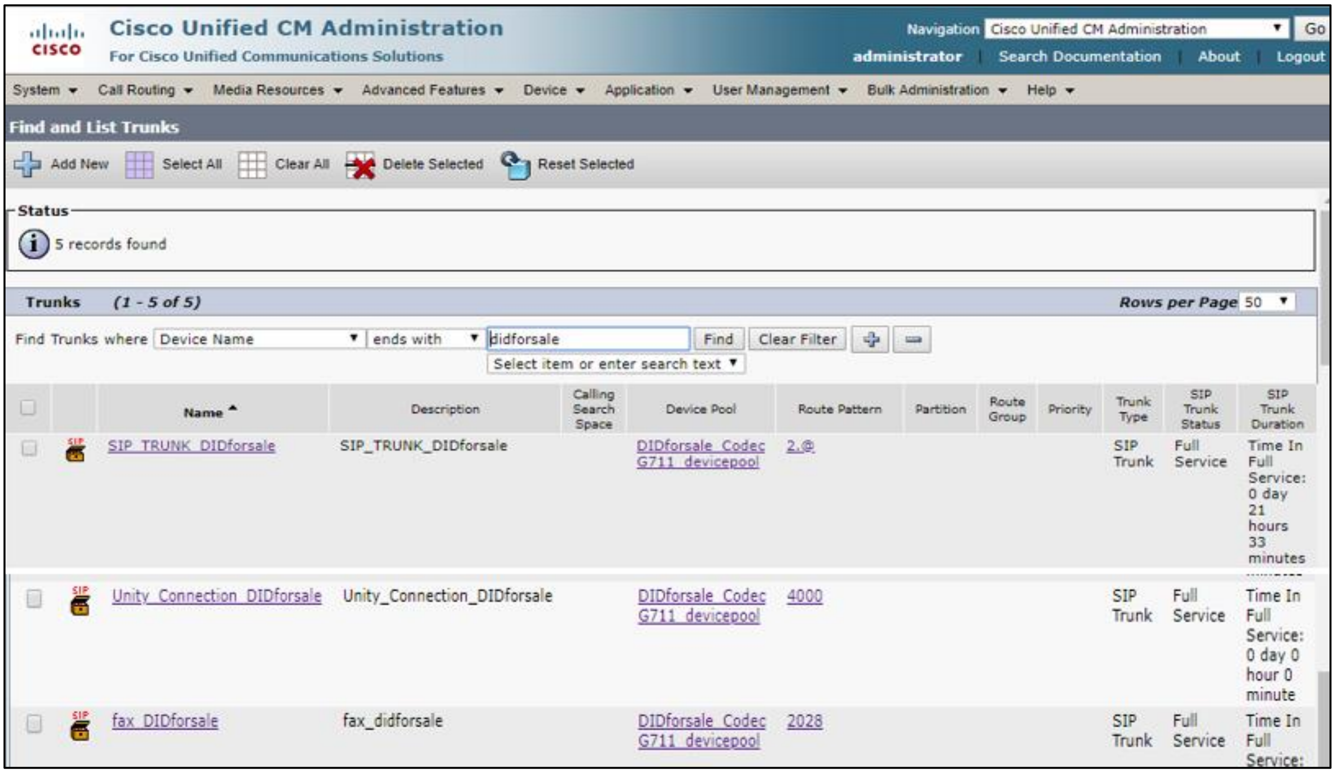


Figure 14: SIP Trunks List

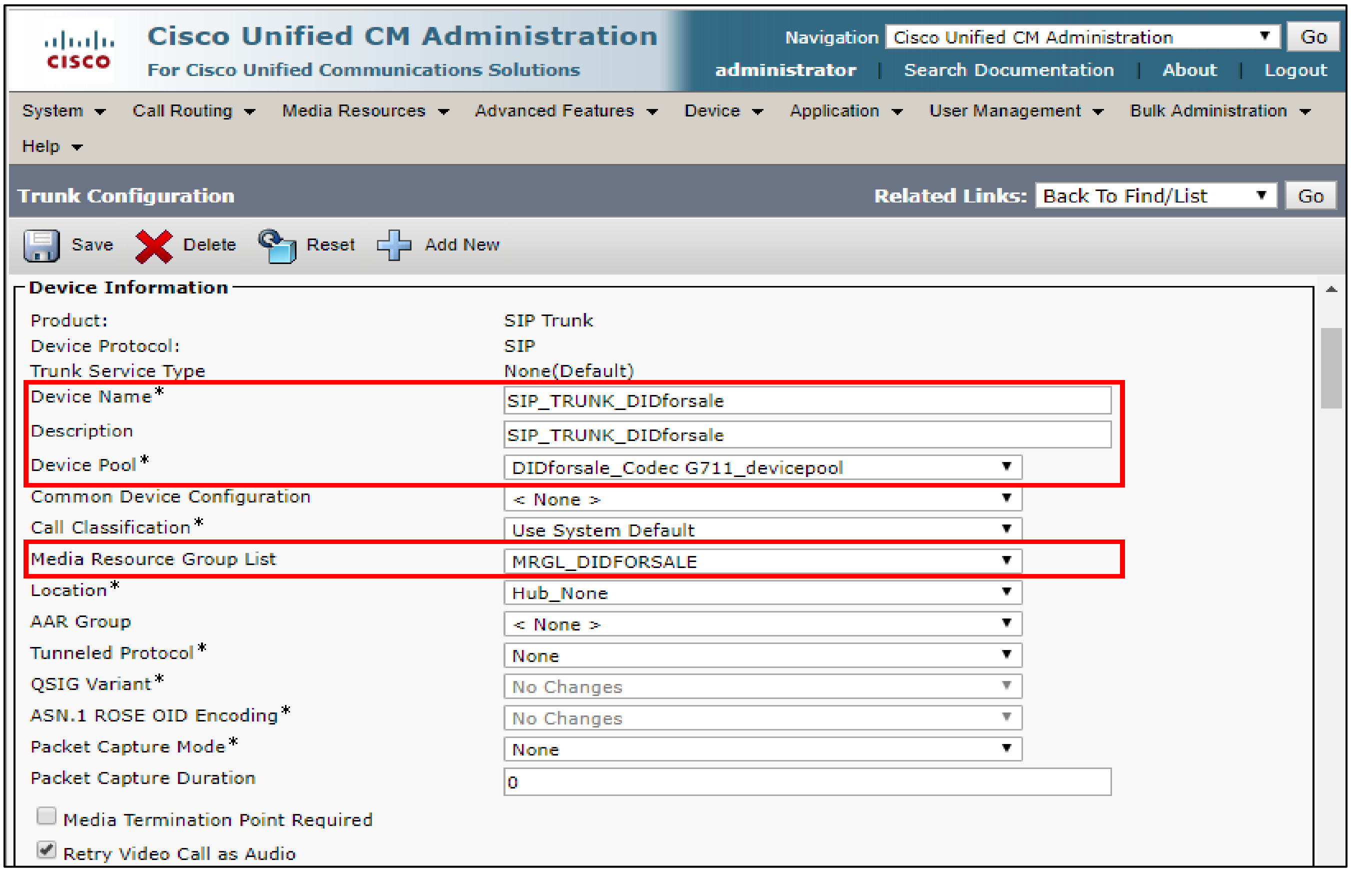


Figure 13: SIP Trunk to Cisco UBE

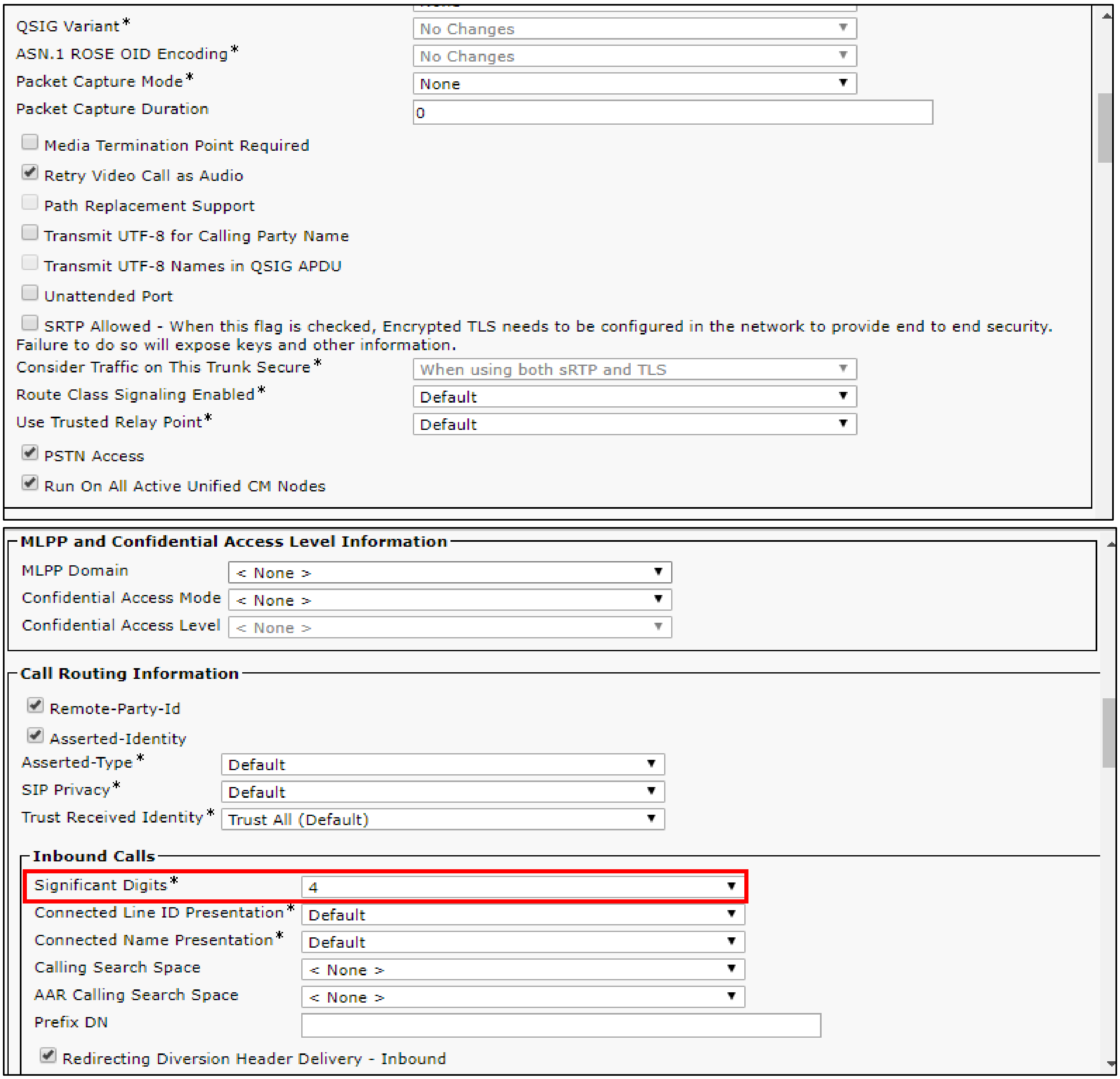
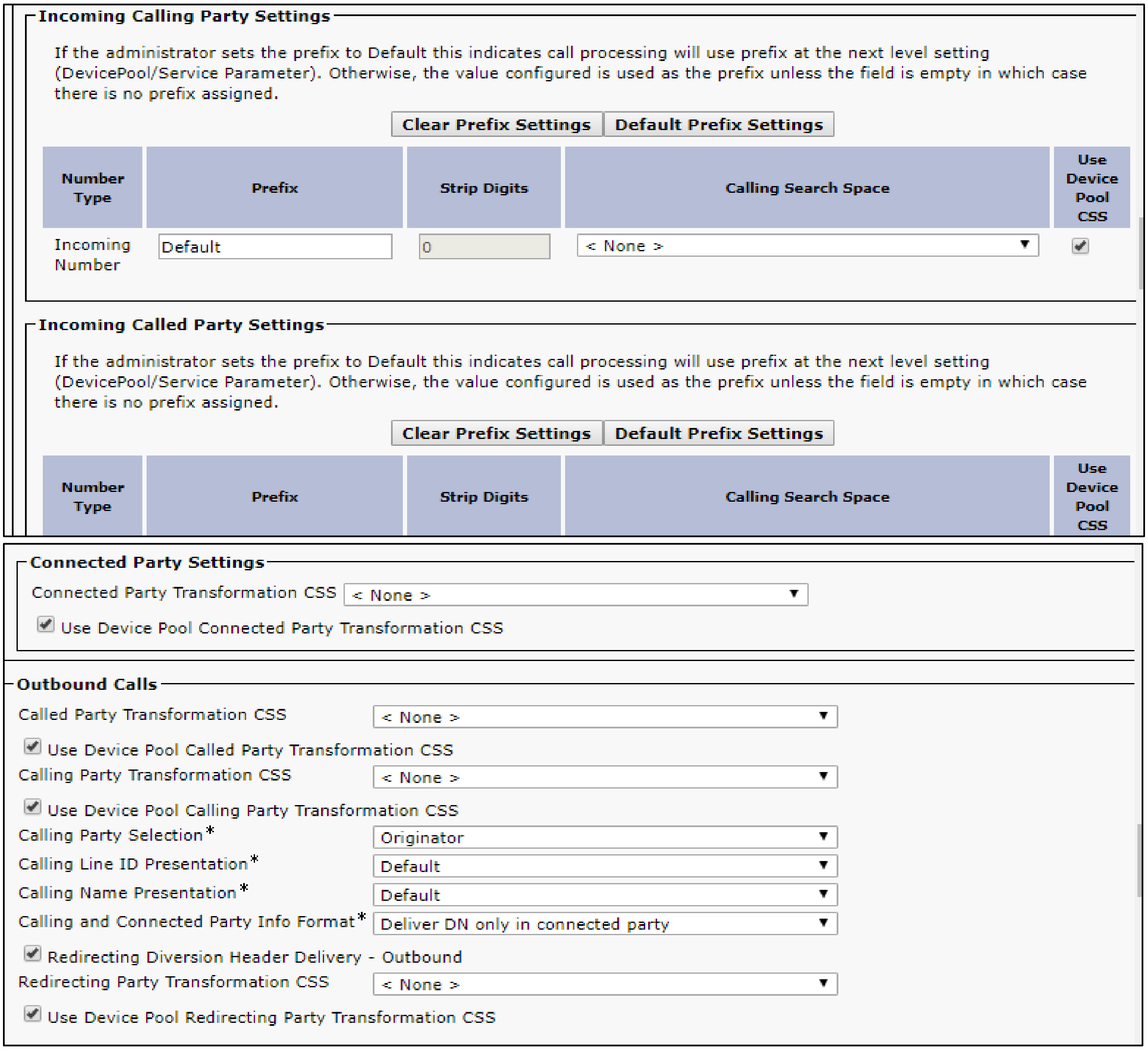


Figure 14: SIP Trunk to Cisco UBE (cont.)


Incoming Calling Party Settings
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.
Clear Prefix Settings
Default Prefix Settings
Number Type
Prefix
Strip Digits
Calling Search Space
Use Device Pool CSS
Incoming Number
Default
0
< None >
Incoming Called Party Settings
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.
Clear Prefix Settings
Default Prefix Settings
Number Type
Prefix
Strip Digits
Calling Search Space
Use Device Pool CSS
Connected Party Settings
Connected Party Transformation CSS
< None >
Use Device Pool Connected Party Transformation CSS
Outbound Calls
Called Party Transformation CSS
< None >
Use Device Pool Called Party Transformation CSS
Calling Party Transformation CSS
< None >
Use Device Pool Calling Party Transformation CSS
Calling Party Selection*
Originator
Calling Line ID Presentation*
Default
Calling Name Presentation*
Default
Calling and Connected Party Info Format*
Deliver DN only in connected party
Redirecting Diversion Header Delivery - Outbound
Redirecting Party Transformation CSS
< None >
Use Device Pool Redirecting Party Transformation CSS

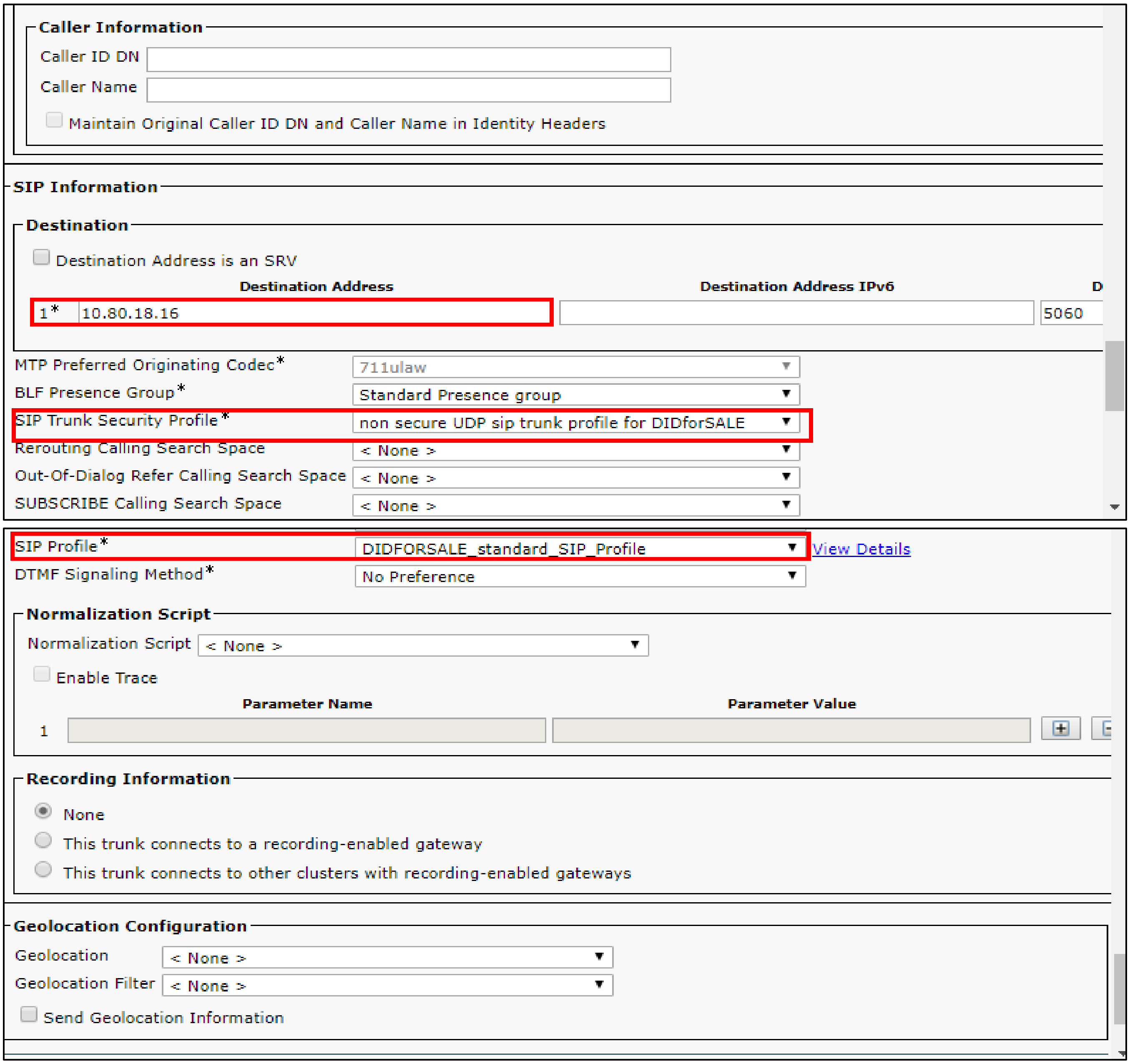


Figure 15: SIP Trunk to Cisco UBE (Cont.)

Explanation

Parameter	Value	Description
Device Name	SIP_TRUNK_DIDforsale	Name for the trunk
Device Pool	DIDforsale_codecg711_devicepool	Default Device Pool is used for this trunk
Media Resource Group List	MRGL_DIDFORSALE	MRG with resources: ANN, CFB, MOH and MTP
Significant Digits	4	4 digits Extension for all CPE phones
Destination Address	10.80.18.16	IP address of the Cisco UBE Virtual LAN
SIP Trunk Security Profile	non Secure UDP sip trunk profile for DIDforSale	SIP Trunk Security Profile configured earlier
SIP Profile	DIDforSale_standard_SIP_Profile	SIP Profile configured earlier

Dial Plan

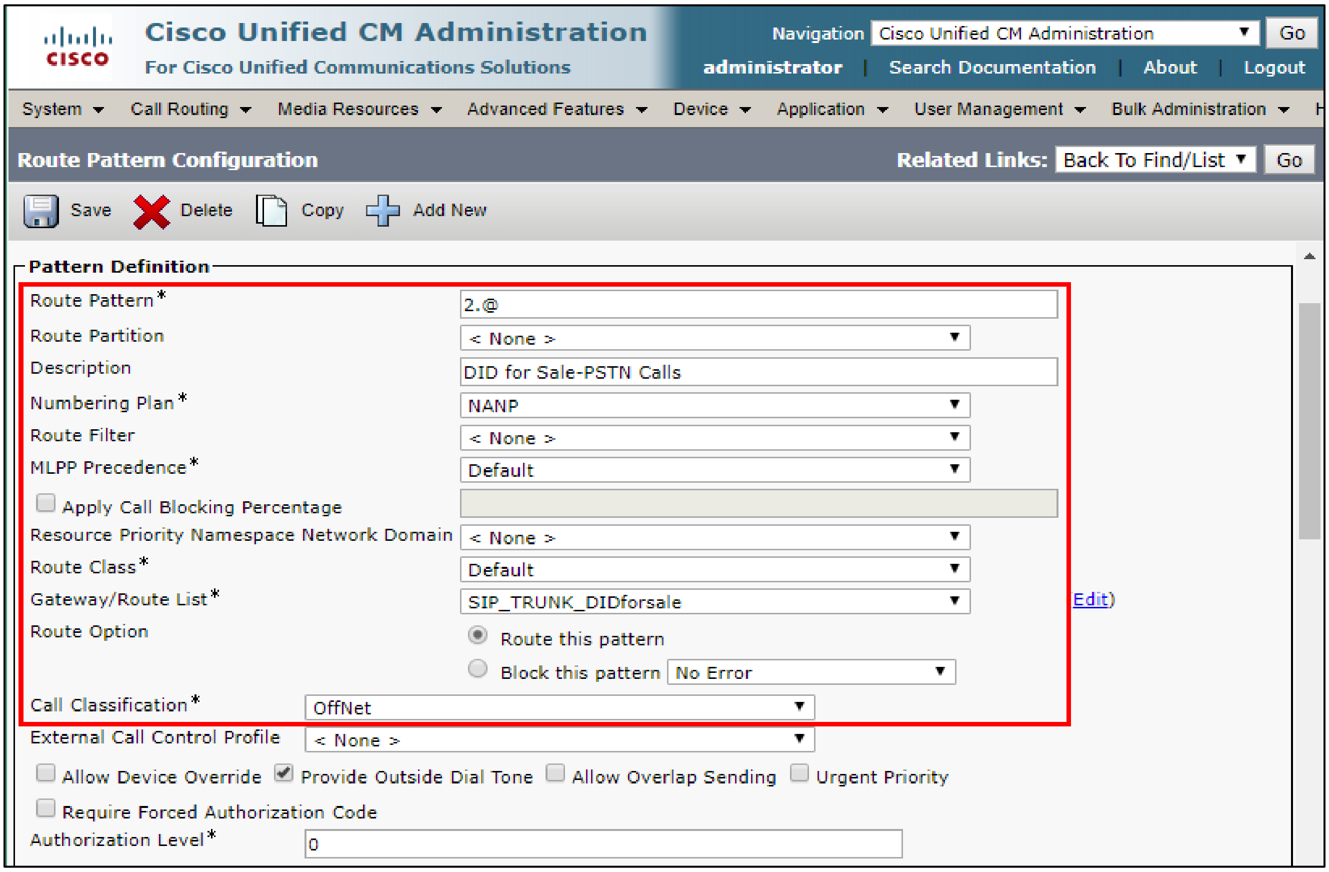
Route Pattern Configuration

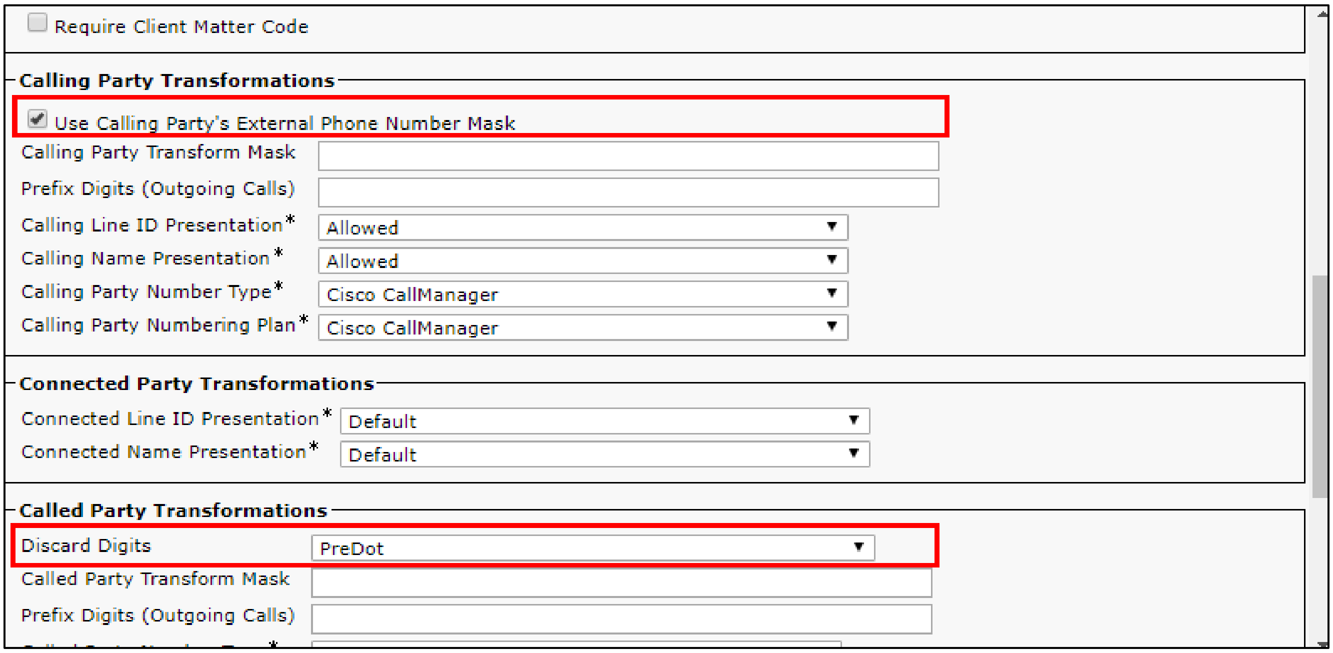
Navigation: Call Routing > Route/Hunt > Route Pattern

Route patterns are configured as below:

- Cisco IP phone dial “2”+”1”+10 digits number to access PSTN via Cisco UBE
    - “2” is removed before sending to Cisco UBE
- For FAX call, Access Code “2”+10 digits number is used at Cisco Fax gateway
    - “2” is removed at Cisco UCM
    - The rest of the number is sent to Cisco UBE to DIDforSale network
- Incoming fax call to 2028 will be sent to Cisco ATA
- Cisco IP phones dial X11 for emergency call and will send all digits to Cisco UBE to DIDforSale network

Figure 16: Route Patterns List


Cisco Unified CM Administration
For Cisco Unified Communications Solutions
Navigation Cisco Unified CM Administration Go
administrator | Search Documentation | About | Logout
System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration
Route Pattern Configuration
Related Links: Back To Find/List Go
Save Delete Copy Add New
Pattern Definition
Route Pattern* 2.@
Route Partition < None >
Description DID for Sale-PSTN Calls
Numbering Plan* NANP
Route Filter < None >
MLPP Precedence* Default
Apply Call Blocking Percentage
Resource Priority Namespace Network Domain < None >
Route Class* Default
Gateway/Route List* SIP_TRUNK_DIDforsale
Edit)
Route Option
Route this pattern
Block this pattern No Error
Call Classification* OffNet
External Call Control Profile < None >
Allow Device Override
Provide Outside Dial Tone
Allow Overlap Sending
Urgent Priority
Require Forced Authorization Code
Authorization Level* 0



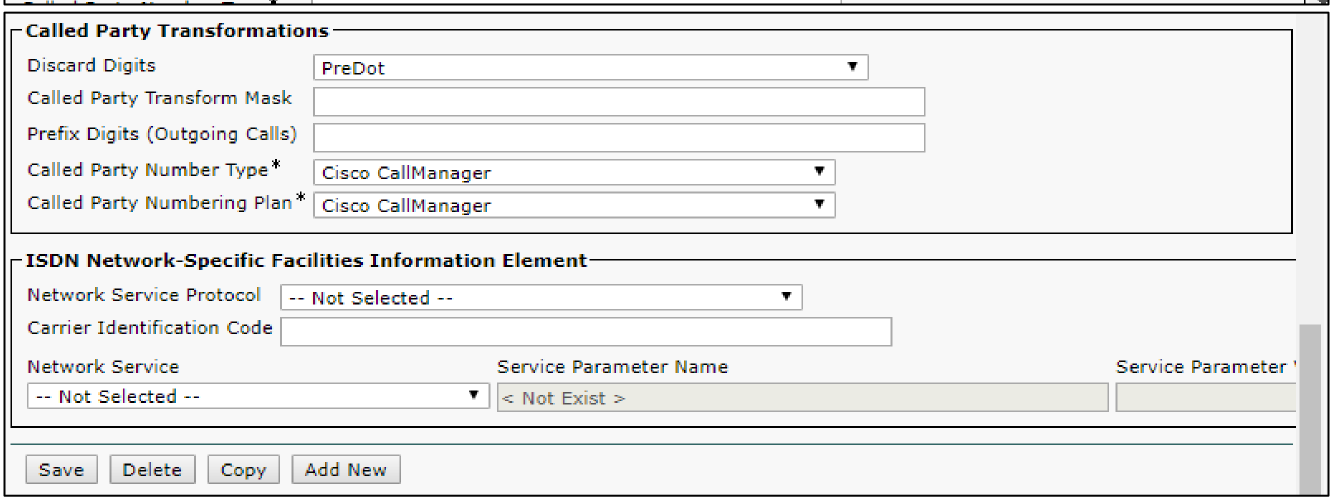
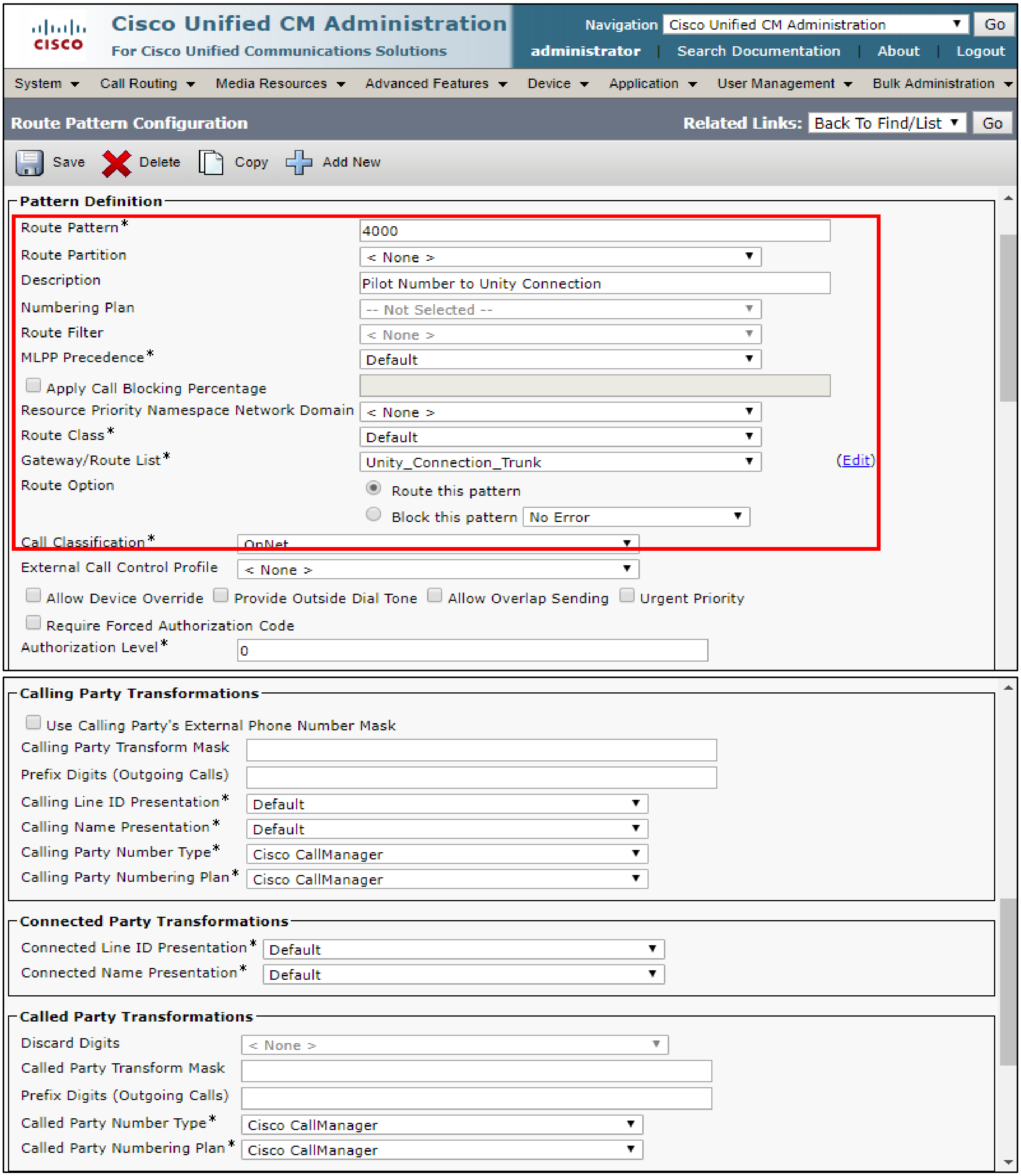


Figure 17: Route Pattern for Voice


Cisco Unified CM Administration
For Cisco Unified Communications Solutions
Navigation Cisco Unified CM Administration Go
administrator | Search Documentation | About | Logout
System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration
Route Pattern Configuration
Related Links: Back To Find/List Go
Save Delete Copy Add New
Pattern Definition
Route Pattern* 4000
Route Partition < None >
Description Pilot Number to Unity Connection
Numbering Plan -- Not Selected --
Route Filter < None >
MLPP Precedence* Default
Apply Call Blocking Percentage
Resource Priority Namespace Network Domain < None >
Route Class* Default
Gateway/Route List* Unity_Connection_Trunk (Edit)
Route Option
Route this pattern
Block this pattern No Error
Call Classification* OnNet
External Call Control Profile < None >
Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority
Require Forced Authorization Code
Authorization Level* 0
Calling Party Transformations
Use Calling Party's External Phone Number Mask
Calling Party Transform Mask
Prefix Digits (Outgoing Calls)
Calling Line ID Presentation* Default
Calling Name Presentation* Default
Calling Party Number Type* Cisco CallManager
Calling Party Numbering Plan* Cisco CallManager
Connected Party Transformations
Connected Line ID Presentation* Default
Connected Name Presentation* Default
Called Party Transformations
Discard Digits < None >
Called Party Transform Mask
Prefix Digits (Outgoing Calls)
Called Party Number Type* Cisco CallManager
Called Party Numbering Plan* Cisco CallManager

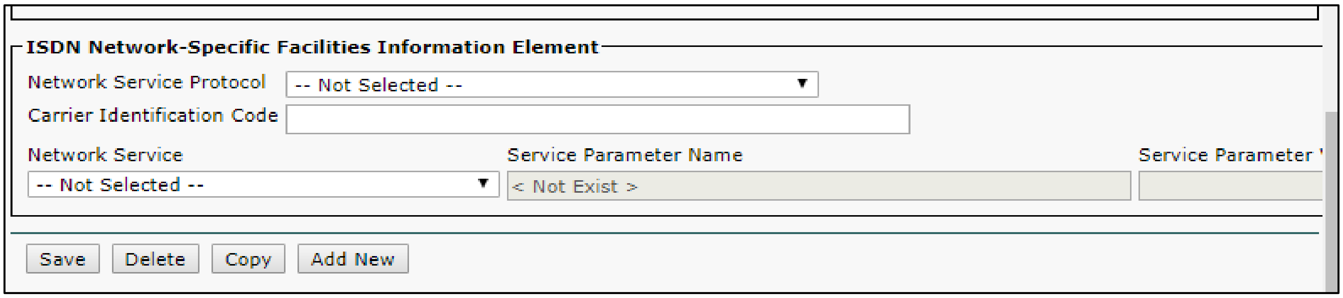
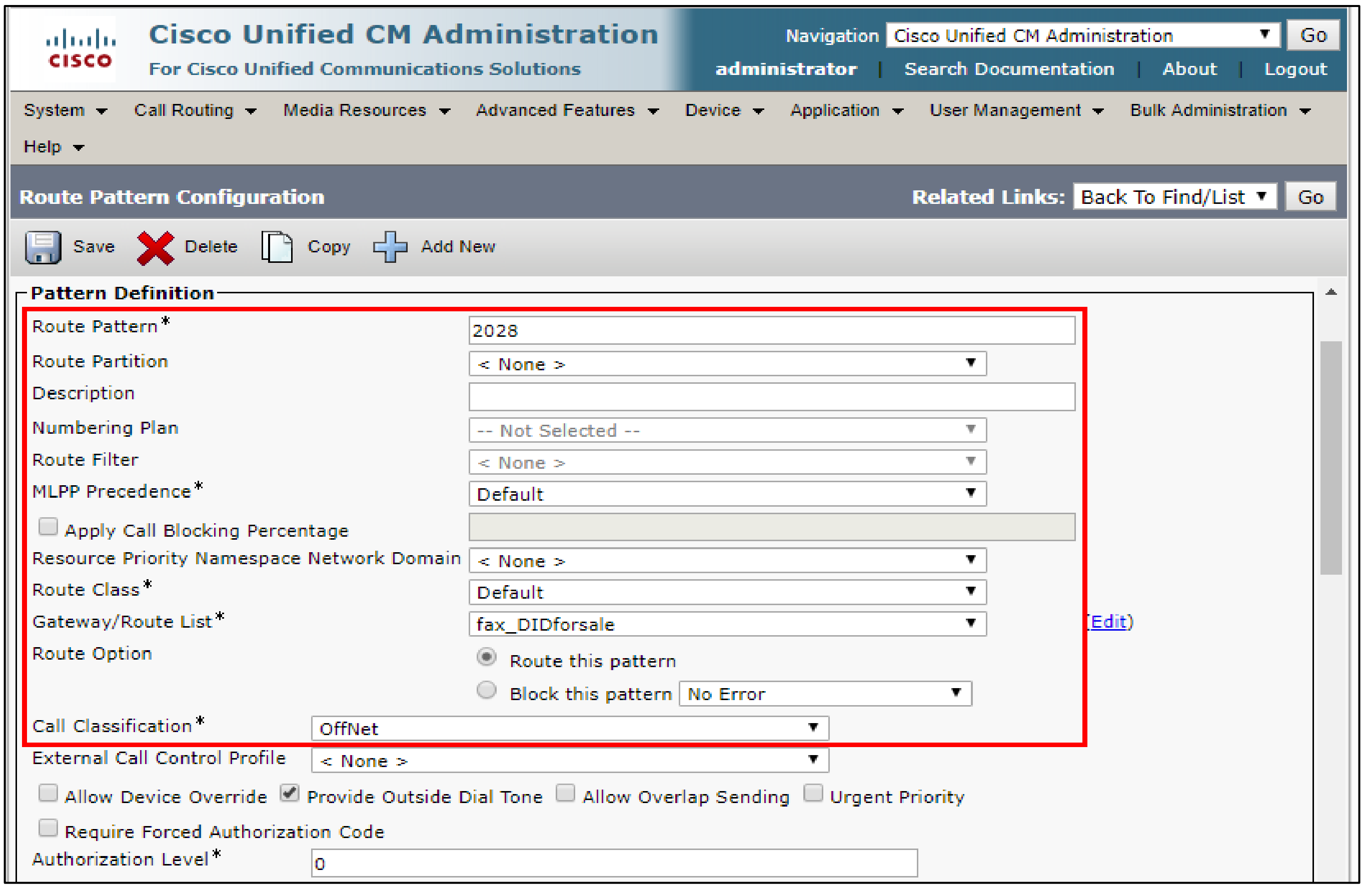


Figure 18: Route Pattern for Voice (Cont.)



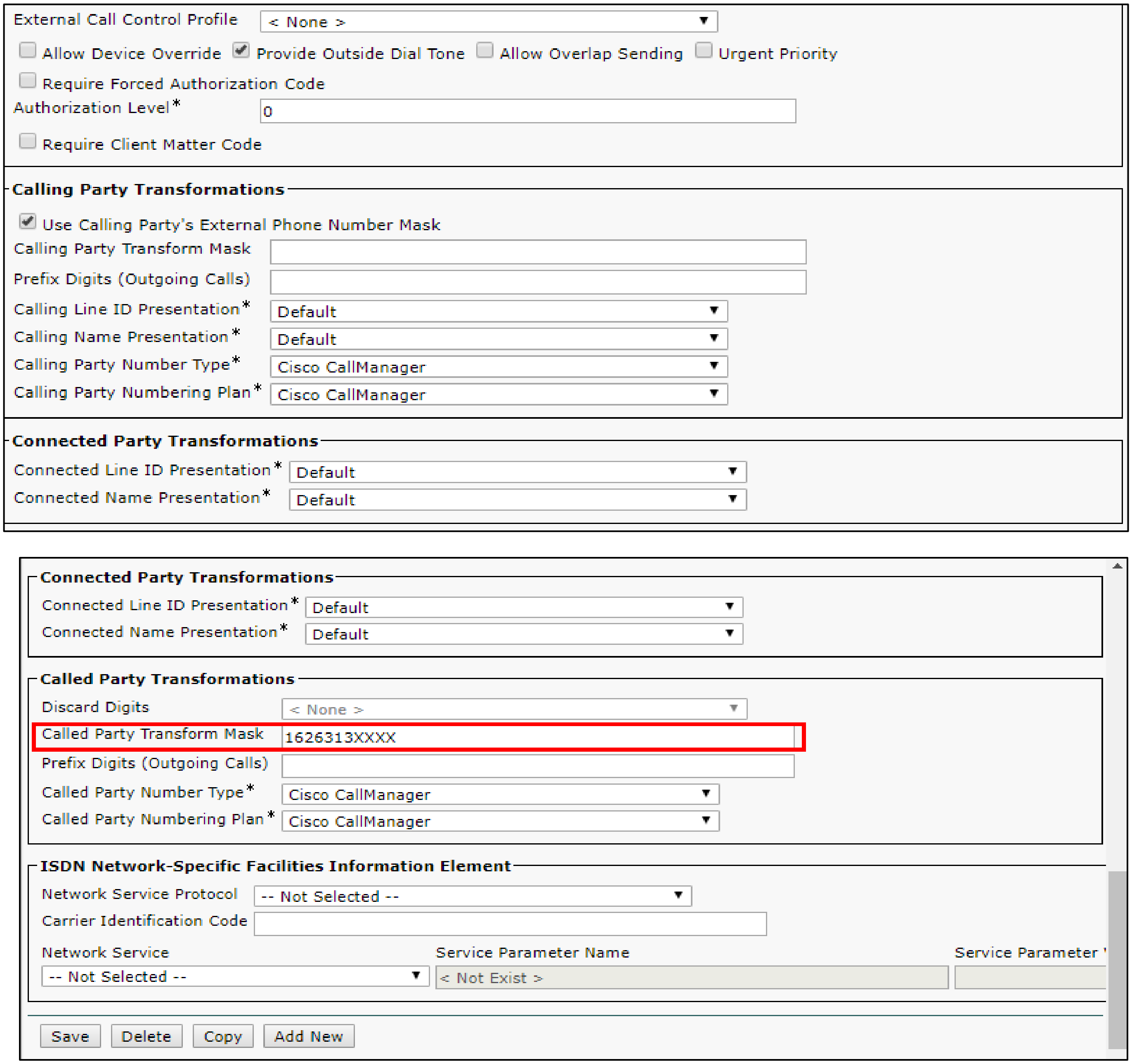


Figure 19: Route Pattern for Fax

Explanation

Setting	Value	Description
Route Pattern	2.@ for Voice & International Calls, and 2028 for fax call and 4000 for Unity connection	Specify appropriate Route Pattern
Gateway/Route List	To_DIDforSale route Pattern 9.@, 2028 for SIP Trunk to FAX Gateway and 4000 for Unity Connection	SIP Trunk name configured earlier
Numbering Plan	NANP for Route Pattern 2.@	North American Numbering Plan
Call Classification	Offnet for Route Pattern 2.@, 2028 and 4000	Restrict the transferring of an external call to an external device
Discard Digits	PreDot for Route Pattern 2.@	Specifies how to modify digit before they are sending to DIDforSale network

FAX ATA

FAX Configuration

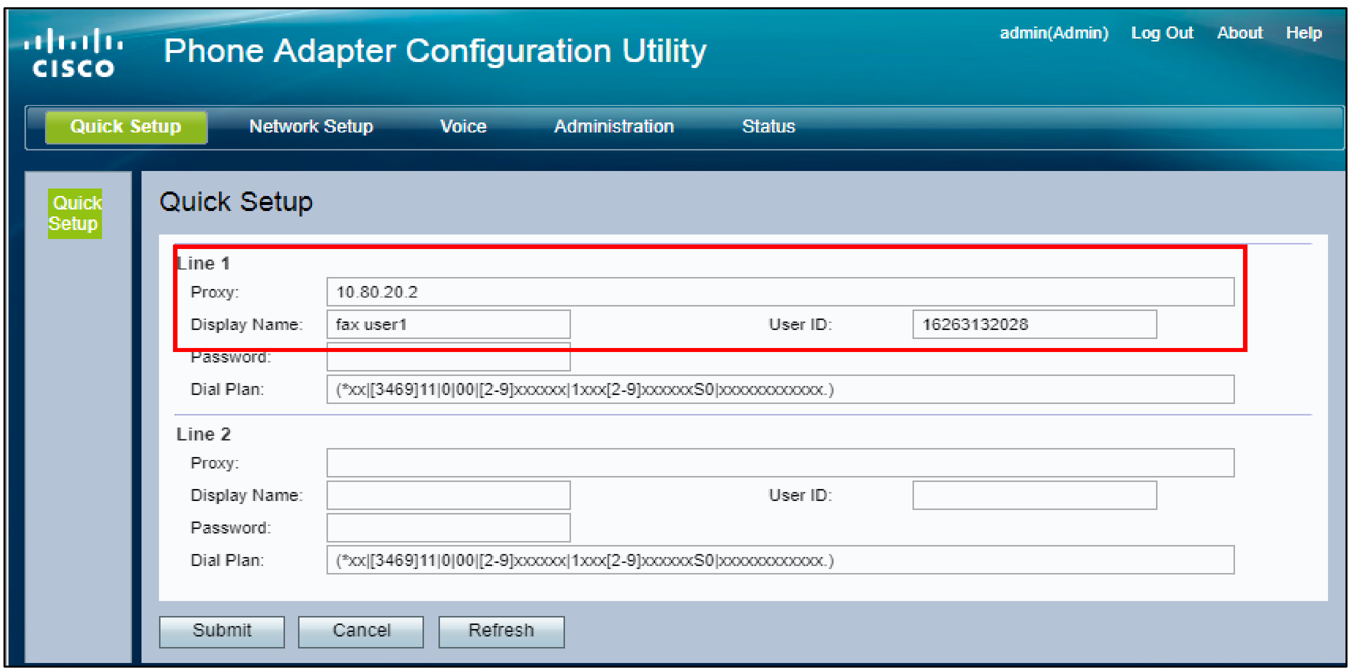


Figure 21: FAX ATA Configuration

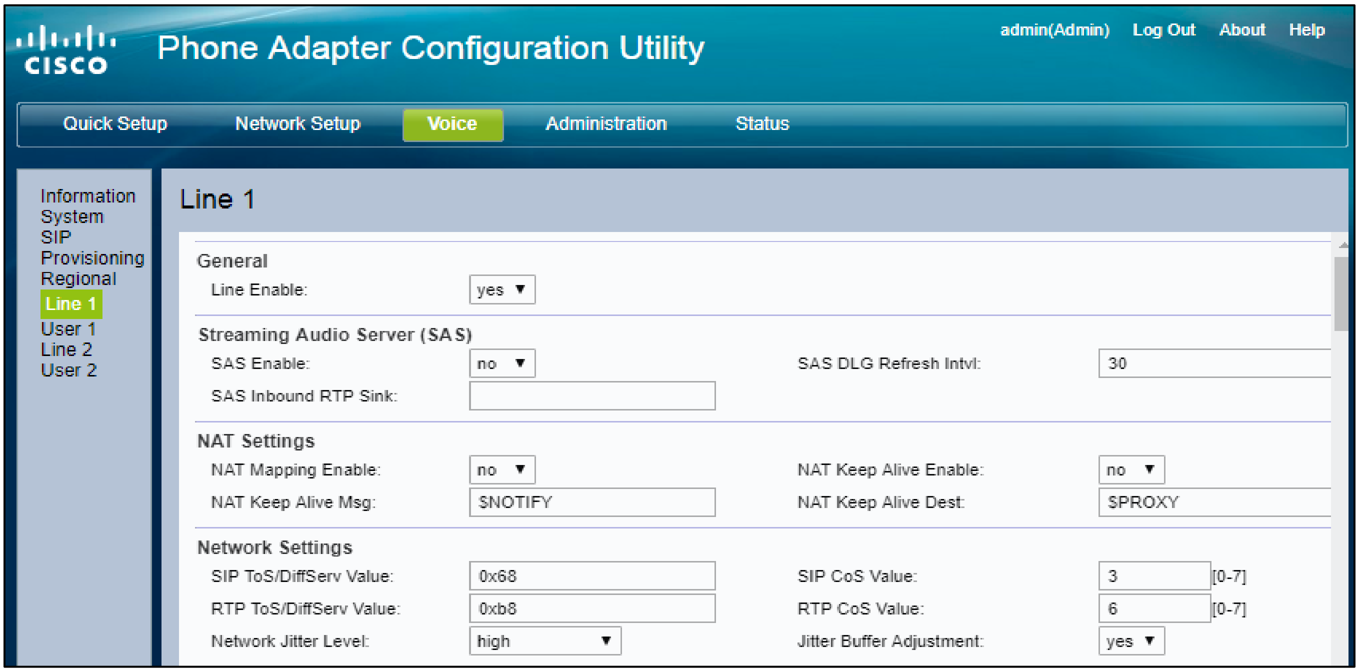
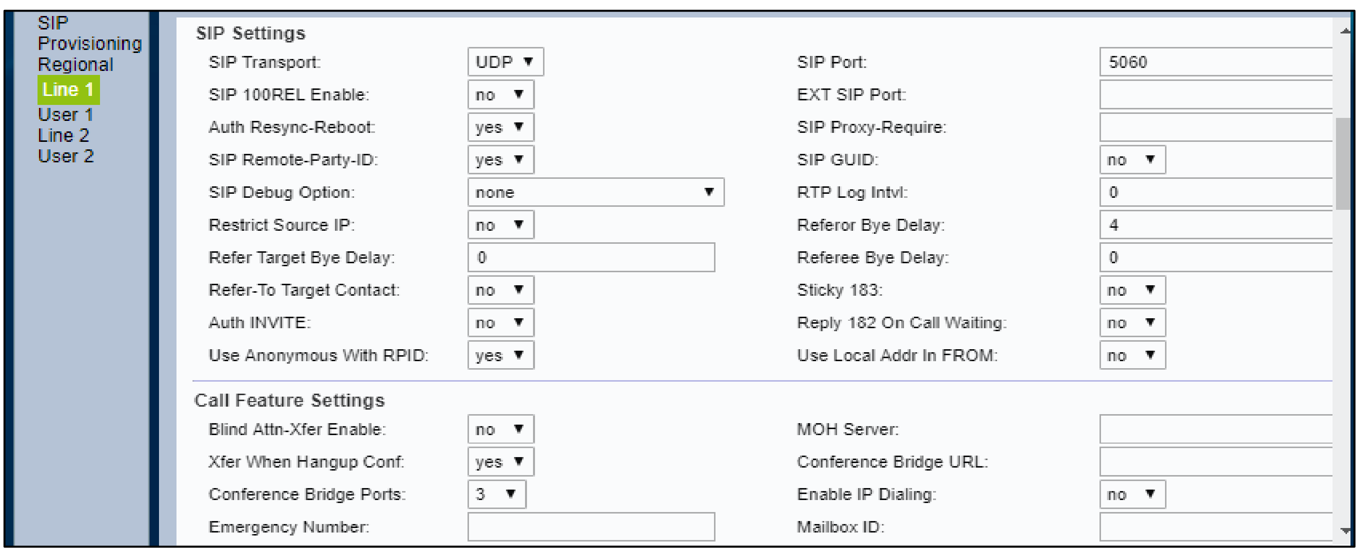
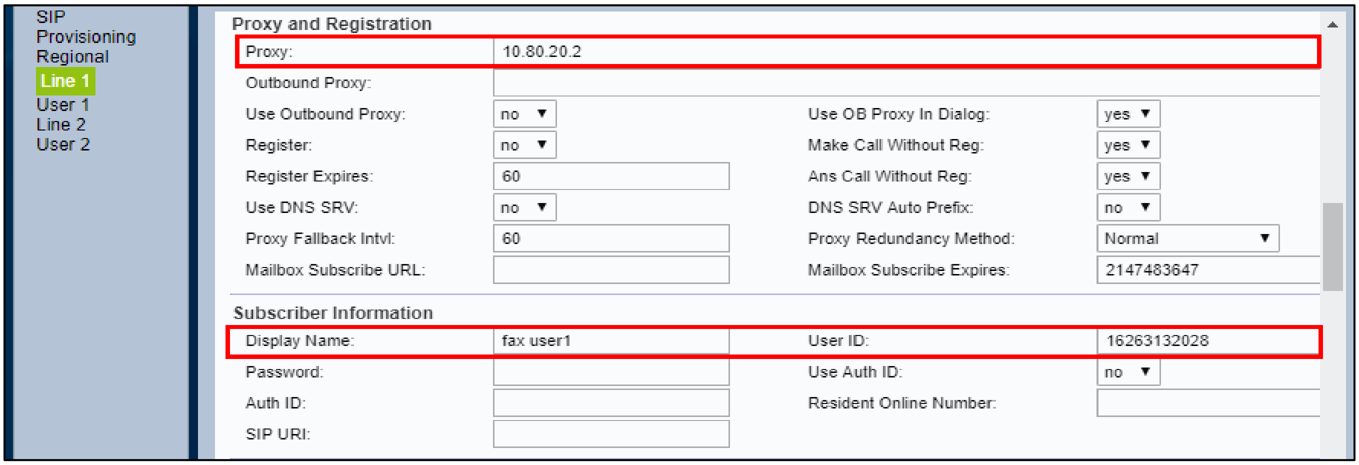
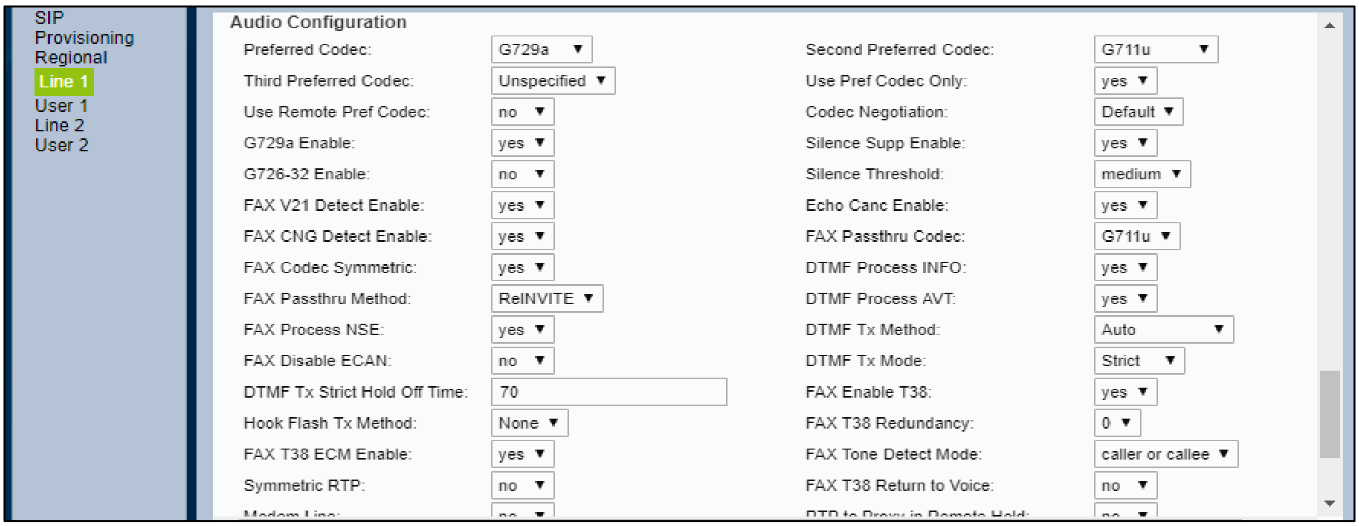
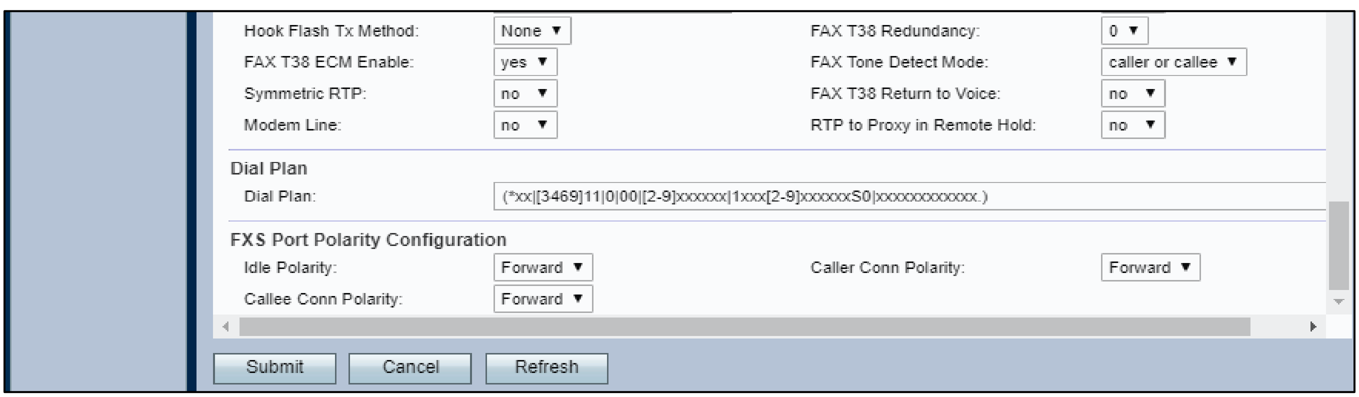


Figure 22: FAX ATA Configuration (cont.)




SIP
Provisioning
Regional
Line 1
User 1
Line 2
User 2
Proxy and Registration
Proxy: 10.80.20.2
Outbound Proxy:
Use Outbound Proxy: no
Use OB Proxy In Dialog: yes
Register: no
Make Call Without Reg: yes
Register Expires: 60
Ans Call Without Reg: yes
Use DNS SRV: no
DNS SRV Auto Prefix: no
Proxy Fallback Intvl: 60
Proxy Redundancy Method: Normal
Mailbox Subscribe URL:
Mailbox Subscribe Expires: 2147483647
Subscriber Information
Display Name: fax user1
User ID: 16263132028
Password:
Use Auth ID: no
Auth ID:
Resident Online Number:
SIP URI:


SIP
Provisioning
Regional
Line 1
User 1
Line 2
User 2
Audio Configuration
Preferred Codec: G729a
Second Preferred Codec: G711u
Third Preferred Codec: Unspecified
Use Pref Codec Only: yes
Use Remote Pref Codec: no
Codec Negotiation: Default
G729a Enable: yes
Silence Supp Enable: yes
G726-32 Enable: no
Silence Threshold: medium
FAX V21 Detect Enable: yes
Echo Canc Enable: yes
FAX CNG Detect Enable: yes
FAX Passthru Codec: G711u
FAX Codec Symmetric: yes
DTMF Process INFO: yes
FAX Passthru Method: ReINVITE
DTMF Process AVT: yes
FAX Process NSE: yes
DTMF Tx Method: Auto
FAX Disable ECAN: no
DTMF Tx Mode: Strict
DTMF Tx Strict Hold Off Time: 70
FAX Enable T38: yes
Hook Flash Tx Method: None
FAX T38 Redundancy: 0
FAX T38 ECM Enable: yes
FAX Tone Detect Mode: caller or callee
Symmetric RTP: no
FAX T38 Return to Voice: no


Hook Flash Tx Method: None
FAX T38 Redundancy: 0
FAX T38 ECM Enable: yes
FAX Tone Detect Mode: caller or callee
Symmetric RTP: no
FAX T38 Return to Voice: no
Modem Line: no
RTP to Proxy in Remote Hold: no
Dial Plan
Dial Plan: (*xx|[3469]11|0|00|[2-9]xxxxxx|1xxx[2-9]xxxxxxS0|xxxxxxxxxxxx.)
FXS Port Polarity Configuration
Idle Polarity: Forward
Caller Conn Polarity: Forward
Callee Conn Polarity: Forward
Submit
Cancel
Refresh

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

Important Information

THE SPECIFICATIONS AND INFORMATION REGARDING THE PRODUCTS IN THIS MANUAL ARE SUBJECT TO CHANGE WITHOUT NOTICE. ALL STATEMENTS, INFORMATION, AND RECOMMENDATIONS IN THIS MANUAL ARE BELIEVED TO BE ACCURATE BUT ARE PRESENTED WITHOUT WARRANTY OF ANY KIND, EXPRESS OR IMPLIED. USERS MUST TAKE FULL RESPONSIBILITY FOR THEIR APPLICATION OF ANY PRODUCTS. IN NO EVENT SHALL CISCO OR ITS SUPPLIERS BE LIABLE FOR ANY INDIRECT, SPECIAL, CONSEQUENTIAL, OR INCIDENTAL DAMAGES, INCLUDING, WITHOUT LIMITATION, LOST PROFITS OR LOSS OR DAMAGE TO DATA ARISING OUT OF THE USE OR INABILITY TO USE THIS MANUAL, EVEN IF CISCO OR ITS SUPPLIERS HAVE BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGES.

Corporate Headquarters
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-4000
800 553-NETS (6387)
Fax: 408 526-4100

European Headquarters
CiscoSystems
International BV
Haarlerbergpark
Haarlerbergweg 13-19
1101 CH Amsterdam
The Netherlands
www-europe.cisco.com
Tel: 31 0 20 357 1000
Fax: 31 0 20 357 1100

Americas Headquarters
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-7660
Fax: 408 527-0883

AsiaPacific Headquarters
Cisco Systems, Inc.
Capital Tower
168 Robinson Road
#22-01 to #29-01
Singapore 068912
www.cisco.com
Tel: +65 317 7777
Fax: +65 317 7799

Cisco Systems has more than 200 offices in the following countries and regions. Addresses, phone numbers, and fax numbers are listed on the Cisco Web site at http://www.cisco.com/go/offices.

Argentina • Australia • Austria • Belgium • Brazil • Bulgaria • Canada • Chile • China PRC • Colombia • Costa Rica • Croatia • Czech Republic • Denmark • Dubai, UAE • Finland • France • Germany • Greece • Hong Kong SAR • Hungary • India • Indonesia • Ireland • Israel • Italy • Japan • Korea • Luxembourg • Malaysia • Mexico • The Netherlands • New Zealand • Norway • Peru • Philippines • Poland • Portugal • Puerto Rico • Romania • Russia • Saudi Arabia • Scotland • Singapore • Slovakia • Slovenia • South Africa • Spain • Sweden • Switzerland • Taiwan • Thailand • Turkey Ukraine • United Kingdom • United States • Venezuela • Vietnam • Zimbabwe