Application Note

TalkTalk SIP Trunking:

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

March 7, 2016

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Introduction

Service Providers today, such as TalkTalk, 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.

TalkTalk 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 TalkTalk network, Cisco Unified Border Element (Cisco UBE) ISR 4321/K9 running IOS-XE 15.6(1)S/3.17 can be used. The Cisco Unified Border Element 15.6(1)S provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 11.0.1 connected to TalkTalk 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 TalkTalk 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 - 15.6(1)S, 3.17] for connectivity to TalkTalk SIP Trunking service. The deployment model covered in this application note is CPE (Cisco UCM 11.0.1) to PSTN (TalkTalk).
- Testing was performed in accordance to TalkTalk 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 TalkTalk 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 TalkTalk 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

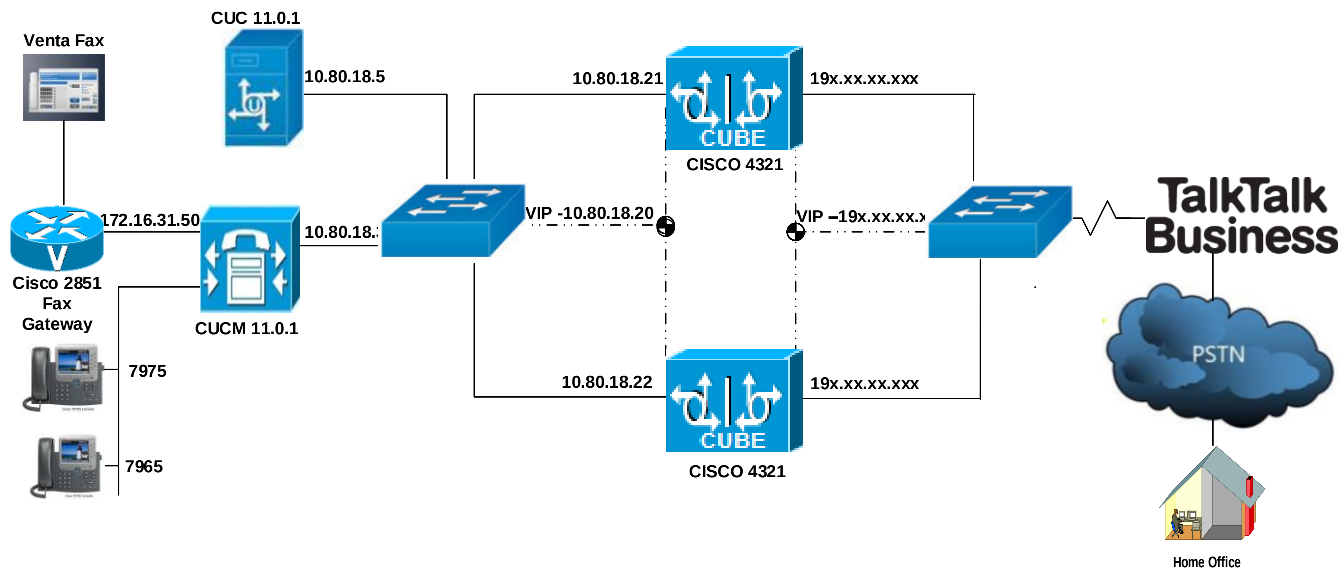


Figure 1: Network Topology

- Cisco IP Phones 7975 and 7965 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 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 7965 (SIP) and 7975 (SCCP)

Software Requirements

- Cisco Unified Communications Manager 11.0.1
- Cisco Unity Connection 11.0.1
- IOS 15.6(1)S for ISR 4321/K9 Cisco Unified Border Element
- IOS 15.1(4)M5 for Cisco 2851 Fax Gateway

Features

Features Supported

- Incoming and outgoing off-net calls using G711ALaw
- 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)

Features Not Supported

- Cisco IP phones used in this test do not support blind transfer
- Fax (T.38) is not supported by Service Provider

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 modify PAI/PPI header and forward to network in the tested release. CISCO BUG ID: CSCuv04539.
- Call Forward Unconditional to OffNet PSTN scenarios was executed by configuring Calling Party Selection* to “First Redirect number (External)” under Trunk Settings

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
 ip address 10.80.18.21 255.255.255.0
 media-type rj45
 negotiation auto
 redundancy rii 1
 redundancy group 1 ip 10.80.18.20 exclusive
!
interface GigabitEthernet0/0/1
 ip address 192.65.79.140 255.255.255.128
 negotiation auto
 redundancy rii 2
 redundancy group 1 ip 192.65.79.149 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
  ipv4 91.146.112.10
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 1
 no supplementary-service sip handle-replaces
 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
  privacy-policy send-always
  g729 annexb-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

G711alaw is used as the preferred codec for this testing

```
voice class codec 1
        codec preference 1 g711alaw
        codec preference 2 g711ulaw
```

Dial Peer

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

```
dial-peer voice 200 voip
 description Outbound-from IP PBX to PSTN - WAN facing
 huntstop
 destination-pattern .T
 session protocol sipv2
 session target sip-server
 voice-class codec 1
 voice-class sip asserted-id pai
 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 nsf 000000
 fax protocol pass-through g711alaw
 no vad
!
dial-peer voice 100 voip
 description Inbound-from PSTN to IP PBX - LAN facing
 huntstop
 destination-pattern 0203.......
 session protocol sipv2
 session target ipv4:10.80.18.3:5060
 voice-class codec 1
 voice-class sip asserted-id pai
 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 nsf 000000
 fax protocol pass-through g711alaw
 no vad
!
dial-peer voice 420 voip
 description PBX to PBX dialing
 translation-profile outgoing TALK-TALK
 huntstop
 destination-pattern 61..
 session protocol sipv2
 session target sip-server
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 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 500 voip
 description cube dp
 huntstop
 session protocol sipv2
 session target sip-server
 incoming called-number 0203.......
```

```
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip profiles 101
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax nsf 000000
 fax protocol pass-through g711alaw
 no vad
!
dial-peer voice 210 voip
 description outgoing call to TalkTalk - LAN facing
 huntstop
 session protocol sipv2
 incoming called-number .T
 voice-class codec 1
 voice-class sip asserted-id pai
 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 nsf 000000
 fax protocol pass-through g711alaw
 no vad
!
dial-peer voice 998 voip
 description "Emergency and Operator call-LAN side"
 session protocol sipv2
 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
 no vad
!
dial-peer voice 999 voip
 description "Emergency and Operator call-WAN side"
 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
 no vad
!
dial-peer voice 117 voip
 description "information service call-LAN side"
 session protocol sipv2
 session target sip-server
 incoming called-number 118500
 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
 no vad
!
dial-peer voice 118 voip
 description "information service call-WAN side"
```

```
destination-pattern 118500
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
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 a “9” 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 “9”. A “9.001@” 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 TalkTalk, Caller dial 9 prefix followed by the target four-digits extension number, 9 was stripped and the four-digits extension number was send to Cisco UBE, Cisco UBE translate the 4 digits extension number to its full ten-digits DID under Dial Peer 420 and send to TalkTalk network which will direct back to Cisco UBE and handled same as normal incoming PSTN call.

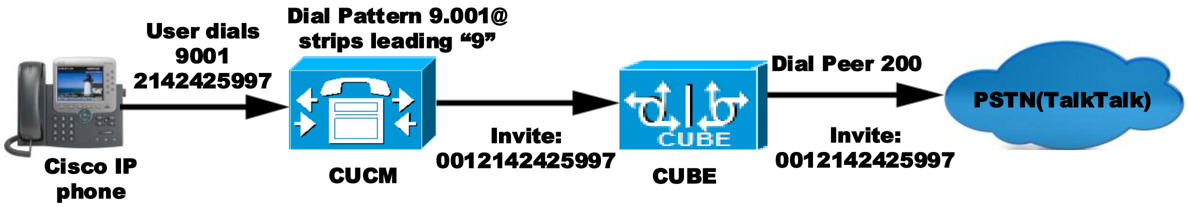


Figure 2: Outbound Voice Call

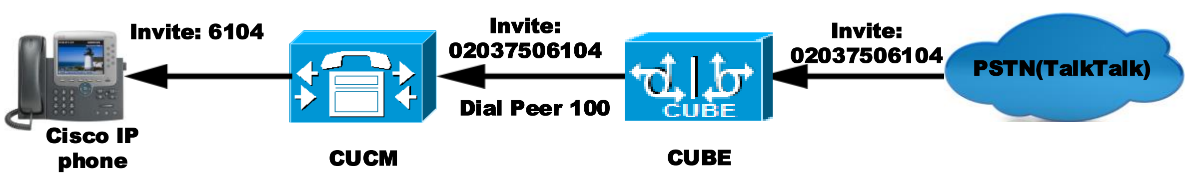


Figure 3: Inbound Voice Call

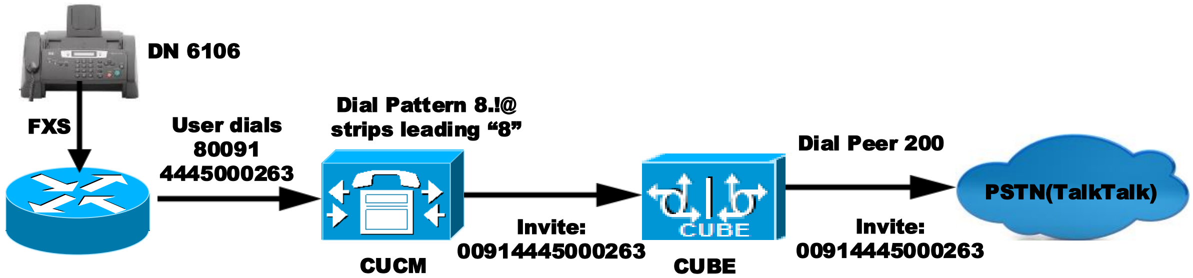


Figure 4: Outbound Fax Call

Figure 5: Inbound Fax Call

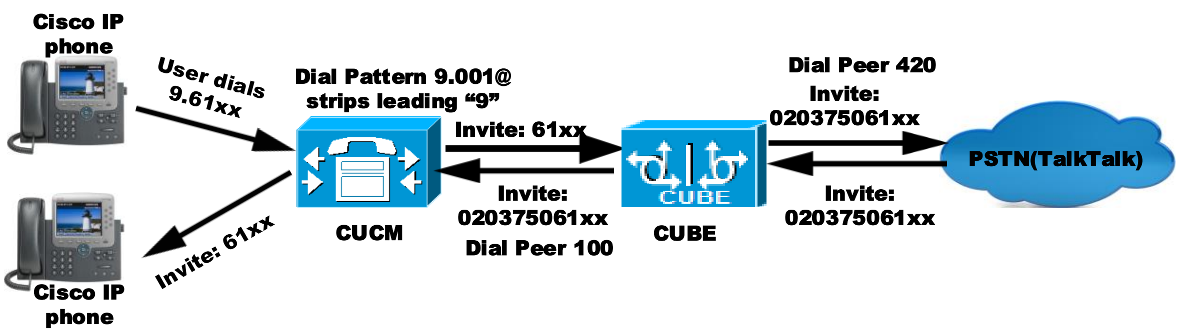


Figure 6: PBX to PBX via TalkTalk Call

Configuration Example

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

Active Cisco UBE

```
TalkTalk_CUBE1#sh running-config
Building configuration...

version 15.6
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname TalkTalk_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
!
no aaa new-model
!
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
  ipv4 91.146.112.10
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 1
 no supplementary-service sip handle-replaces
 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
  privacy-policy send-always
  g729 annexb-all
!
voice class codec 1
 codec preference 1 g711alaw
 codec preference 2 g711ulaw
!
```

```
!
voice class sip-profiles 101
 response ANY sip-header Allow-Header modify "UPDATE," ""
 request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:0203750\1@\2
>"
!
voice translation-rule 2
 rule 1 /^.*\(61..\)/ /0203750\1/
!
voice translation-profile TALK-TALK
translate called 2
!
license udi pid ISR4321/K9 sn FDO19220MSQ
!
spanning-tree extend system-id
!
redundancy
 mode none
 application redundancy
  group 1
   name voice-b2bhaTalkTalk
   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
 ip address 10.80.18.21 255.255.255.0
 media-type rj45
 negotiation auto
 redundancy rii 1
 redundancy group 1 ip 10.80.18.20 exclusive
!
interface GigabitEthernet0/0/1
 ip address 192.65.79.140 255.255.255.128
 negotiation auto
 redundancy rii 2
 redundancy group 1 ip 192.65.79.149 exclusive
!
interface GigabitEthernet0/1/0
 description CUBE HA MS5 3/0/37
 ip address 10.89.20.9 255.255.255.0
 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 route 0.0.0.0 0.0.0.0 192.65.79.129
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 200 voip
 description Outbound-from IP PBX to PSTN - WAN facing
 huntstop
 destination-pattern .T
 session protocol sipv2
 session target sip-server
 voice-class codec 1
 voice-class sip asserted-id pai
 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 nsf 000000
 fax protocol pass-through g711alaw
 no vad
!
dial-peer voice 100 voip
 description Inbound-from PSTN to IP PBX - LAN facing
 huntstop
 destination-pattern 0203.......
 session protocol sipv2
 session target ipv4:10.80.18.3:5060
 voice-class codec 1
 voice-class sip asserted-id pai
 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 nsf 000000
 fax protocol pass-through g711alaw
 no vad
!
dial-peer voice 420 voip
 description PBX to PBX dialing
 translation-profile outgoing TALK-TALK
 huntstop
 destination-pattern 61..
 session protocol sipv2
 session target sip-server
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
```

```
 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 500 voip
 description cube dp
 huntstop
 session protocol sipv2
 session target sip-server
 incoming called-number 0203.......
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip profiles 101
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax nsf 000000
 fax protocol pass-through g711alaw
 no vad
!
dial-peer voice 210 voip
 description outgoing call to TalkTalk - LAN facing
 huntstop
 session protocol sipv2
 incoming called-number .T
 voice-class codec 1
```

```
 voice-class sip asserted-id pai
 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 nsf 000000
 fax protocol pass-through g711alaw
 no vad
!
dial-peer voice 998 voip
 description "Emergency and Operator call-LAN side"
 session protocol sipv2
 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
 no vad
!
dial-peer voice 999 voip
 description "Emergency and Operator call-WAN side"
 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
```

```
 no vad
!
dial-peer voice 117 voip
 description "information service call-LAN side"
 session protocol sipv2
 session target sip-server
 incoming called-number 118500
 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
 no vad
!
dial-peer voice 118 voip
 description "information service call-WAN side"
 destination-pattern 118500
 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
 no vad
!
!
sip-ua
 keepalive target ipv4:91.146.112.10:5060
 disable-early-media 180
 timers keepalive active 180
```

```
 sip-server ipv4:91.146.112.10:5060
 connection-reuse
!
!
line con 0
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 exec-timeout 0 0
 password tekV1z10n
 login local
!
!
end
```

Standby Cisco UBE

```
TalkTalk_CUBE2#sh running-config
Building configuration...

version 15.6
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname TalkTalk_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
!
no aaa new-model
!
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
  ipv4 91.146.112.10
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 1
 no supplementary-service sip handle-replaces
 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
  privacy-policy send-always
  g729 annexb-all
!
voice class codec 1
 codec preference 1 g711alaw
 codec preference 2 g711ulaw
!
```

```
voice class sip-profiles 101
 response ANY sip-header Allow-Header modify "UPDATE," ""
 request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:0203750\1@\2
>"
!
voice translation-rule 2
 rule 1 /^.*\(61..\)/ /0203750\1/
!
voice translation-profile TALK-TALK
translate called 2
!
license udi pid ISR4321/K9 sn FDO19220MQ9
!
spanning-tree extend system-id
!
redundancy
 mode none
 application redundancy
  group 1
   name voice-b2bhaTalkTalk
   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
 ip address 10.80.18.22 255.255.255.0
 media-type rj45
 negotiation auto
 redundancy rii 1
 redundancy group 1 ip 10.80.18.20 exclusive
!
interface GigabitEthernet0/0/1
 ip address 192.65.79.141 255.255.255.128
 negotiation auto
 redundancy rii 2
 redundancy group 1 ip 192.65.79.149 exclusive
!
interface GigabitEthernet0/1/0
 description CUBE HA MS5 3/0/38
 ip address 10.89.20.10 255.255.255.0
 negotiation auto
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 shutdown
 negotiation auto
!
interface Vlan1
 no ip address
 shutdown
!
ip forward-protocol nd
no ip http server
```

```
no ip http secure-server
ip tftp source-interface GigabitEthernet0
ip route 0.0.0.0 0.0.0.0 192.65.79.129
ip route 10.64.0.0 255.255.0.0 10.80.18.1
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 200 voip
 description Outbound-from IP PBX to PSTN - WAN facing
 huntstop
 destination-pattern .T
 session protocol sipv2
 session target sip-server
 voice-class codec 1
 voice-class sip asserted-id pai
 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 nsf 000000
 fax protocol pass-through g711alaw
```

```
 no vad
!
dial-peer voice 100 voip
 description Inbound-from PSTN to IP PBX - LAN facing
 huntstop
 destination-pattern 0203.......
 session protocol sipv2
 session target ipv4:10.80.18.3:5060
 voice-class codec 1
 voice-class sip asserted-id pai
 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 nsf 000000
 fax protocol pass-through g711alaw
 no vad
!
dial-peer voice 420 voip
 description PBX to PBX dialing
 translation-profile outgoing TALK-TALK
 huntstop
 destination-pattern 61..
 session protocol sipv2
 session target sip-server
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 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 500 voip
 description cube dp
 huntstop
 session protocol sipv2
 session target sip-server
 incoming called-number 0203.......
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip profiles 101
 dtmf-relay rtp-nte
 fax-relay ecm disable
 fax rate disable
 fax nsf 000000
 fax protocol pass-through g711alaw
 no vad
!
dial-peer voice 210 voip
 description outgoing call to TalkTalk - LAN facing
 huntstop
 session protocol sipv2
 session target sip-server
 incoming called-number .T
 voice-class codec 1
 voice-class sip asserted-id pai
```

```
 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 nsf 000000
 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
 no vad
!
dial-peer voice 999 voip
 description "Emergency and Operator call-WAN side"
 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
```

```
 no vad
!
dial-peer voice 117 voip
 description "information service call-LAN side"
 session protocol sipv2
 session target sip-server
 incoming called-number 118500
 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
 no vad
!
dial-peer voice 118 voip
 description "information service call-WAN side"
 destination-pattern 118500
 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
 no vad
!
!
sip-ua
 keepalive target ipv4:91.146.112.10:5060
 disable-early-media 180
 timers keepalive active 180
```

```
 sip-server ipv4:91.146.112.10:5060
 connection-reuse
!
!
line con 0
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 password tekV1z10n
 login local
!
!
end
```

Configuring Cisco Unified Communications Manager

Cisco UCM Version

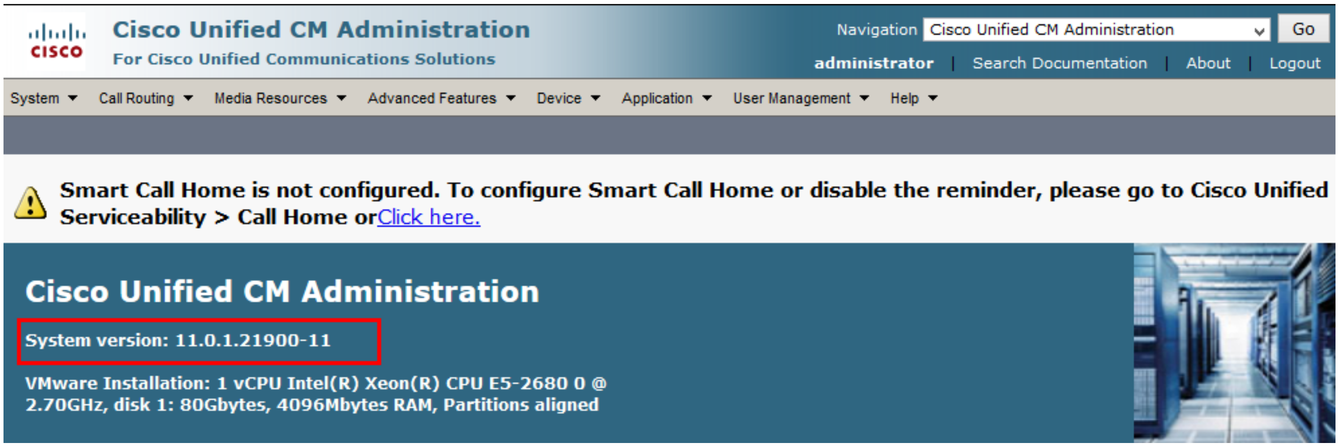


Figure 7: Cisco UCM Version

Cisco Call Manager Service Parameters

Navigation path: System > Service Parameters

Select Server* = Clus28Sub1--CUCM Voice/Video (Active)

Select Service*= Cisco CallManager (Active)

All other fields are set to default values

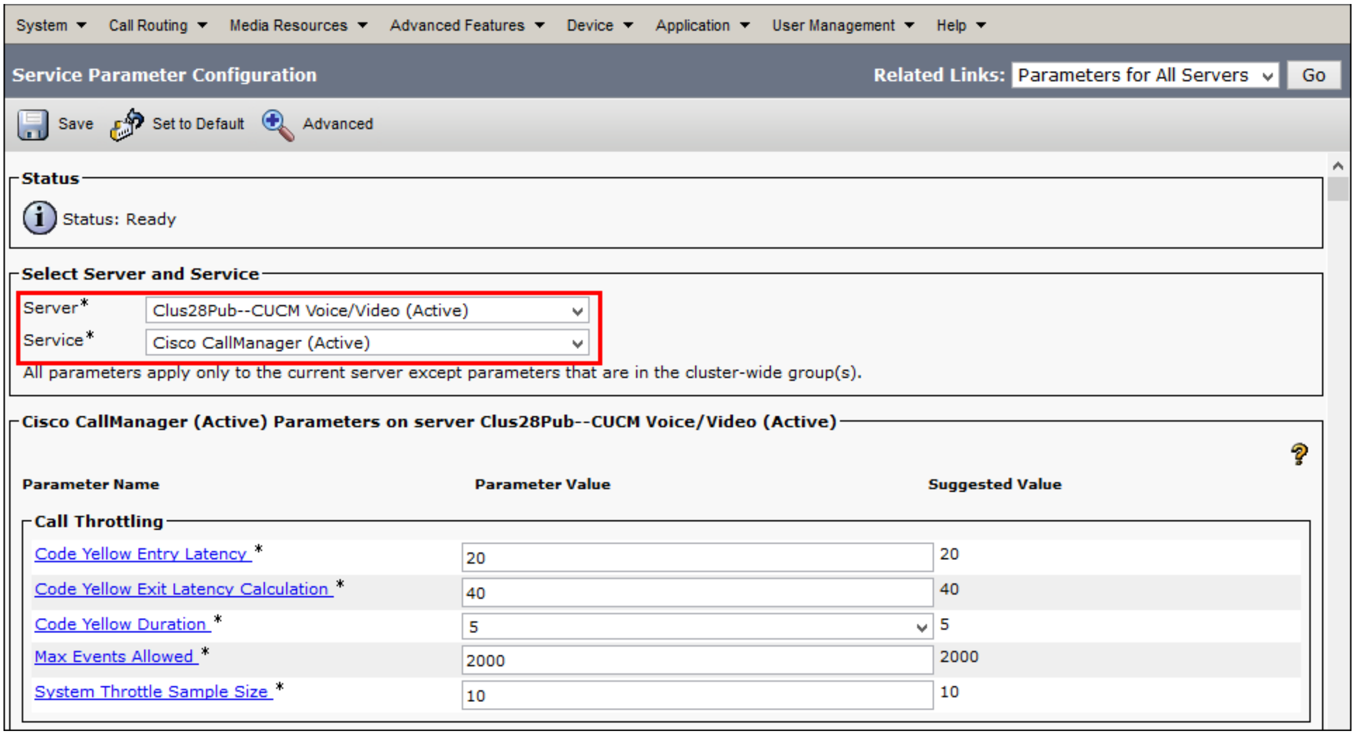
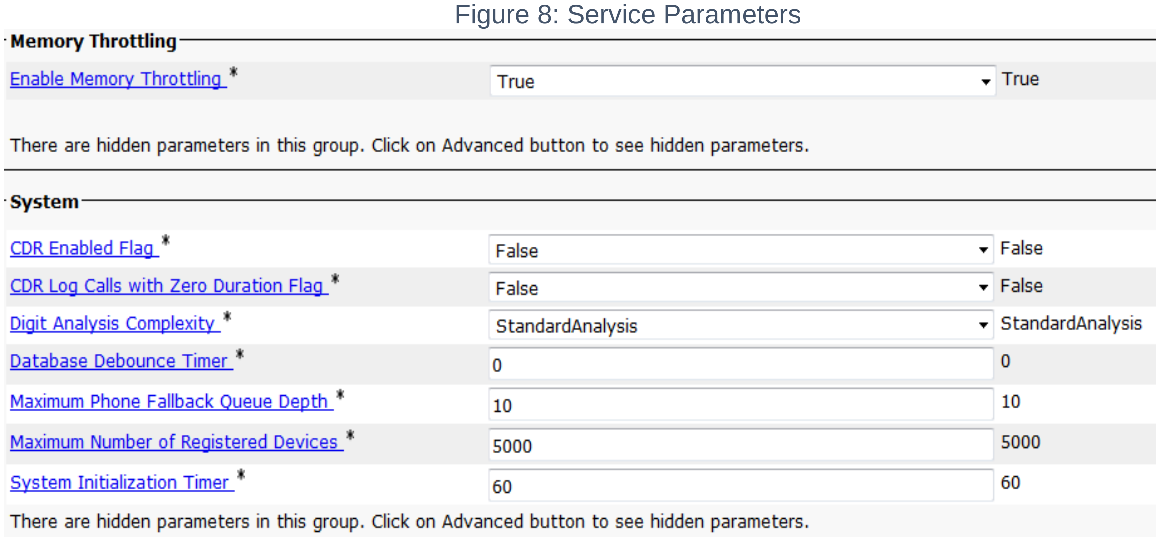


Figure 8: Service Parameters



SDL Trace

SDL Trace Data Flags *	0x00000111	0x00000111
SDL Trace Flush Immediately *	False	False
SDL Trace Data Size *	0	0
SDL Trace Flag *	True	True
SDL TraceType Flags *	0x8000EB15	0x8000EB15

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

Clusterwide Parameters (Device - General)

Call Diagnostics Enabled *	Disabled	Disabled
Show Line Group Member DN in finalCalledPartyNumber CDR Field *	False	False
Show Line Group Member Non Masked DN in finalCalledPartyNumber CDR Field *	False	False
CTI New Call Accept Timer *	4	4
CTI Generate Digits Interval *	250	250
CTI Dial Digits Interval *	250	250
CTI Await Further Digits *	False	False
CTI Use Wildcard Pattern as calledPartyDN *	False	False
CTI Report Ringback on SIP 183 with SDP *	True	True
Retain Media on Disconnect with PI for Active Call *	False	False
Station and Backup Server KeepAlive Interval *	60	60
Station KeepAlive Interval *	30	30
Status Enquiry Poll Flag *	False	False
Strip # Sign from Called Party Number *	True	True

Figure 9: Service Parameters (Cont.)

Session Handoff Alerting Timer *	10	10
T301 Timer *	180000	180000
T302 Timer *	15000	15000
T303 Timer *	4000	4000
T304 Timer *	30000	30000
T305 Timer *	30000	30000
T306 Timer *	30000	30000
T308 Timer *	4000	4000
T309 Timer *	90000	90000
T310 Timer *	60000	60000
T313 Timer *	4000	4000
T316 Timer *	120000	120000
T317 Timer *	100000	100000
T321 Timer *	30000	30000
T322 Timer *	4000	4000
Tone on Hold Timer *	10	10

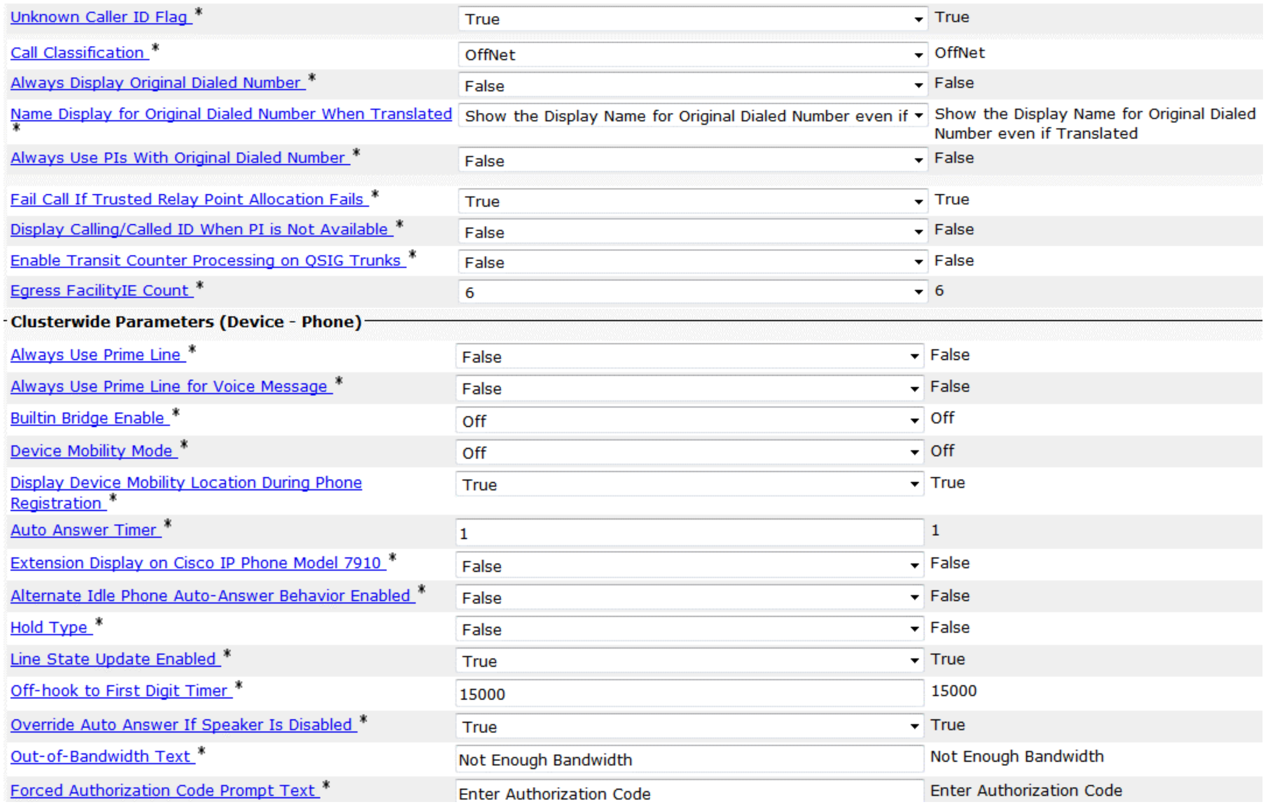
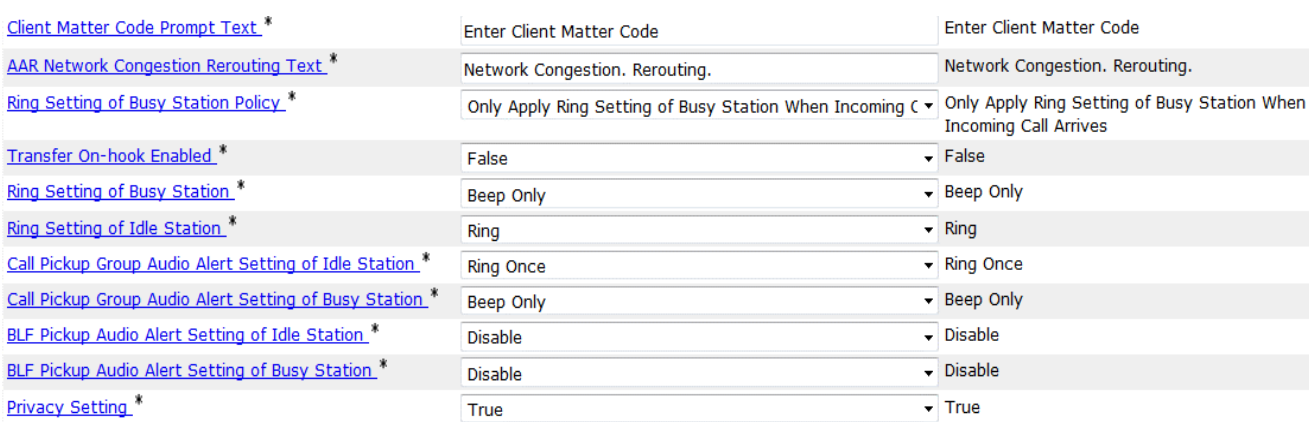


Figure 10: Service Parameters (Cont.)



Enforce Privacy Setting on Held Calls *	False	False
SIP Station KeepAlive Interval *	120	120
SIP Station Realm *	ccmsipline	ccmsipline
Hunt Group Logoff Notification *	None	None
Speed Dial Await Further Digits *	False	False
Display CTI Route Point Name or DN *	False	False
Display Original Calling Number on Transfer from Cisco Unity *	False	False
URI Dialing Display Preference *	DN	DN
Insert Hyphens in 12-Digit Numbers *	False	False
Allow Call Waiting During an In-Progress Outbound Analog Call *	True	True

Clusterwide Parameters (Device - PRI and MGCP Gateway)

Calling Party Number Screening Indicator *	CallManager sets the screening indicator value - Default	CallManager sets the screening indicator value - Default setting
Enable Outbound NetworkTrunk CallingParty Restriction *	False	False
Clear Calls Flag When Datalink Is Down *	True	True
Device Status Poll Interval *	3000	3000
Disable Alerting Progress Indicator *	False	False
Discard Non Inband Progress in Overlap Sending *	False	False
Disable Resume from Shared-line MGCP FXS Port *	True	True
DTMF Silence Tone Flag *	False	False
Enable Display IE in Codeset 6 *	False	False
Enable Sending PRI NI2 Service Message *	False	False

Figure 11: Service Parameters (Cont.)

Flash Hook Duration *	500	500
Gateway Poll Timer *	10	10
Location In PRI Progress Indicator IE (User Side Only) *	Use the Network Side PRI progress indicator IE	Use the Network Side PRI progress indicator IE
Matching Calling Party with Attendant Flag *	False	False
MGCP Database Query Delay Timer *	1000	1000
MGCP FXS On-Hook Pending Timer *	3	3
MGCP Response Timer *	30	30
MGCP Timer *	3	3
Numbering Plan Info *	1	1
Overlap Receiving Flag for PRI *	True	True
Outgoing Media Connect Time for PRI *	Connect ASAP	Connect ASAP
Port Release Timer *	0	0
SMDI Call Delay Timer *	0	0
Stable in State 4 Flag *	False	False
Optimize MGCP Registration *	True	True
Suppress Out-of-Channels Alarms *	True	True
I-Frame Timer *	2000	2000
User-to-User IE Status *	False	False
Convert European Progress Message to Alerting *	False	False
Enable DMS PRI Notify Message from User to Network *	True	True
Audit OOS Channels Interval *	10	10
Digital and Analog Ports Enabled *	True	True

Clusterwide Parameters (Device - H323)

Accept Unknown TCP Connection *	False	False
BRQ Enabled *	False	False
Call Present Disconnect Flag *	False	False
Check Progress Indicator Before Establishing Media *	False	False
H225 Block Setup Destination *	False	False
H225 DB Retry Timer *	0	0
H225 Device Connect Timer *	0	0
H225 DTMF Duration *	100	100
H225 TspReq Retry *	2	2
H225 Intercluster Call Throttle Timer *	30	30
H225 T301 Timer *	180000	180000
H225 T302 Timer *	15000	15000
H225 T303 Timer *	4000	4000

Figure 12: Service Parameters (Cont.)

H225 T304 Timer *	30000	30000
H225 T305 Timer *	30000	30000
H225 T310 Timer *	60000	60000
H225 TCP Timer *	5	5
H245 TCS Timeout *	10	10
H323 Calling Party Number Screening Indicator *	Calling number screened and passed	Calling number screened and passed
Apply External Phone Number Mask for H.323 Calls *	False	False
Tone on Connect *	False	False
Wait Time for SDP with SR/RO Mode *	3	3
RAS ARQ Timer *	3	3
RAS BRQ Timer *	3	3
RAS DRQ Timer *	3	3
RAS RRQ Timer *	3	3
Ras URQ Timer *	3	3
Retry Count for ARQ *	2	2
Retry Count for BRQ *	2	2
Retry Count for DRQ *	2	2
Retry Count for RRQ *	2	2
Retry Count for URQ *	1	1
Send Product ID and Version ID *	False	False
Send Unified CM Version as Version ID in H225Setup *	False	False
Send Progress Timer *	3000	3000
Send H225 User Info Message *	User Info for Call Progress Tone	User Info for Call Progress Tone
Status Enquiry Poll Timer *	10000	10000
Device Name of GK-controlled Trunk That Will Use Port 1720 *	None	None
Host Name/IP Address of GK That Will Use RAS UDP Port 1719 *	None	None
Fail Call If MTP Allocation Fails *	False	False
Overlap Receiving Flag for H323 *	False	False
Allocate Transcoder for H.323 on Early Offer SIP Trunk for Calls with Early Media *	False	False

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

Clusterwide Parameters (Device - SIP)

SIP Interoperability Enabled *	True	True
Retry Count for SIP Bye *	10	10
Retry Count for SIP Cancel *	10	10
Retry Count for SIP Invite *	6	6
Retry Count for SIP PRACK *	6	6
Retry Count for SIP Rel1XX *	10	10
Retry Count for SIP Publish *	6	6
Retry Count for SIP Response *	6	6
SIP Connect Timer *	500	500
SIP Disconnect Timer *	500	500
SIP Expires Timer *	180000	180000
SIP PRACK Timer *	500	500

Figure 13: Service Parameters (Cont.)

SIP Rel1XX Timer *	500	500
SIP Trying Timer *	500	500
SIP Publish Timer *	500	500
SIP Min-SE Value *	1800	1800
SIPS URI Handling *	Reject	Reject
SIP statistics Periodic update Timer *	2	2
SIP Session Expires Timer *	1800	1800
SIP Trunk TspReq Retry *	2	2
SIP TCP Unused Connection Timer *	14	14
SIP TCP Timer *	5	5
SIP Station TCP Port Throttle Threshold *	100	100
SIP Trunk TCP Port Throttle Threshold *	500	500
SIP V.150 Outbound SDP Offer Filtering *	No Filtering	No Filtering
Send SIP Multicast TTL in SDP *	False	False
Default PUBLISH Expiration Timer *	3600	3600
Minimum PUBLISH Expiration Timer *	60	60
IM and Presence Publish Trunk	< None >	
Send 181 Call Is Being Forwarded *	False	False
Delay Sending 181 until 180/183 message is received *	True	True
Fail Call Over SIP Trunk if MTP Allocation Fails *	False	False
Log Call-Related REFER/NOTIFY/SUBSCRIBE SIP Messages for Session Trace *	True	True
Port Received Timer for Outbound Call Setup *	2	2

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

Clusterwide Parameters (Feature - General)

Call Park Display Timer *	10	10
Caller ID Display Priority Enabled *	True	True
Call Park Reversion Timer *	60	60
Park Monitoring Reversion Timer *	60	60
Park Monitoring Periodic Reversion Timer *	30	30
Park Monitoring Forward No Retrieve Timer *	300	300
Preserve globalCallId for Parked Calls *	True	True
Maximum Call Duration Timer *	720	720
Maximum Hold Duration Timer *	360	360

Figure 14: Service Parameters (Cont.)

Party Entrance Tone *	True	True
Message Waiting Lamp Policy *	Primary Line - Light and Prompt	Primary Line - Light and Prompt
Audible Message Waiting Indication Policy *	OFF	OFF
Message Waiting Indicator Inbound Calling Search Space	< None >	
Multiple Tenant MWI Modes *	False	False
MWI Non Message Center Signaling Call Duration *	0	0
Message Waiting Indicator APDU Digit Translation CSS	< None >	
Block OffNet To OffNet Transfer *	False	False
Use Original Call Classification for Transferred Calls *	False	False
Use Restriction attribute of ID/Name Presentation of Transferring Party *	True	True
Local route group for redirected calls *	Local route group of calling party	Local route group of calling party
Block Unencrypted Calls *	False	False

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

Clusterwide Parameters (Feature - Conference)

Suppress MOH to Conference Bridge *	True	True
Drop Ad Hoc Conference *	Never	Never
Maximum Ad Hoc Conference *	4	4
Maximum MeetMe Conference Unicast *	4	4
Advanced Ad Hoc Conference Enabled *	False	False
Choose Encrypted Audio Conference Instead Of Video Conference *	True	True
Minimum Video Capable Participants To Allocate Video Conference *	2	2
Enable Click-to-Conference for Third-Party Applications *	False	False
IMS Conference Factory URI *	cucm-conference-factory@cucm1.company.com	cucm-conference-factory@cucm1.company.com
Cluster Conferencing Prefix Identifier		

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

Clusterwide Parameters (Feature - Call Secure Status Policy)

Secure Call Icon Display Policy *	All media except BFCP and iX transports must be encrypt	All media except BFCP and iX transports must be encrypted

Clusterwide Parameters (Feature - Forward)

Forward Maximum Hop Count *	12	12
Forward No Answer Timer *	12	12
Max Forward Hops to DN *	12	12
Retain Forward Information *	False	False
Forward By Reroute Enabled *	False	False
Transform Forward by Reroute Destination *	True	True
Always Forward Switch Voice Mail Calls *	True	True
Forward By Reroute T1 Timer *	10	10
Include Original Called Info for Q.SIG Call Diversions *	Only after the first diversion	Only after the first diversion
Set Private Numbering Plan for Call Forward *	False	False
Set Type of Number for Call Forward *	Level1RegionalNumber	Level1RegionalNumber
Max Forward UnRegistered Hops to DN *	0	0

Figure 15: Service Parameters (Cont.)

CFA CSS Activation Policy *	With Configured CSS	With Configured CSS
Cause Code When Maximum Forward Hop Count is Triggered *	Normal Unspecified	Normal Unspecified

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

Clusterwide Parameters (Feature - Hold Reversion)

Hold Reversion Duration *	0	0
Hold Reversion Notification Interval *	30	30
CFA Destination Override *	False	False

Clusterwide Parameters (Feature - Call Pickup)

Auto Call Pickup Enabled *	False	False
Call Pickup Locating Timer *	1	1
Call Pickup No Answer Timer *	12	12

Clusterwide Parameters (Feature - Refer)

Validate Refer-to URI *	Validate Except for Anonymous Users	Validate Except for Anonymous Users

Clusterwide Parameters (Feature - Replaces)

Block OffNet To OffNet Replaces *	False	False

Clusterwide Parameters (Feature - Redirection [3xx])

Redirection Ring No Answer Reversion Timer *	24	24
Maximum Redirection Count *	70	70

Clusterwide Parameters (Feature - Multilevel Precedence and Preemption)

Locations-based MLPP Enable *	False	False
Executive Override Call Preemptable *	False	False
Location-based Maximum Bandwidth Enforcement Level for MLPP Calls *	Lenient	Lenient
Non-Preemption Pattern CSS	< None >	
MLPP Exception Level *	Executive Override	Executive Override

Clusterwide Parameters (Feature - Path Replacement)

Path Replacement Enabled *	False	False
Path Replacement on Tromboned Calls *	True	True
Start Path Replacement Minimum Delay Time *	0	0
Start Path Replacement Maximum Delay Time *	0	0
Path Replacement T1 Timer *	30	30
Path Replacement T2 Timer *	15	15
Path Replacement PINX ID		
Path Replacement Calling Search Space	< None >	

Clusterwide Parameters (Feature - Call Back)

Call Back Enabled Flag *	True	True
Call Back Notification Audio File Name *	CallBack.raw	CallBack.raw
Connection Proposal Type *	Connection Retention	Connection Retention
Connection Response Type *	Default to Connection Retention	Default to Connection Retention

Figure 16: Service Parameters (Cont.)

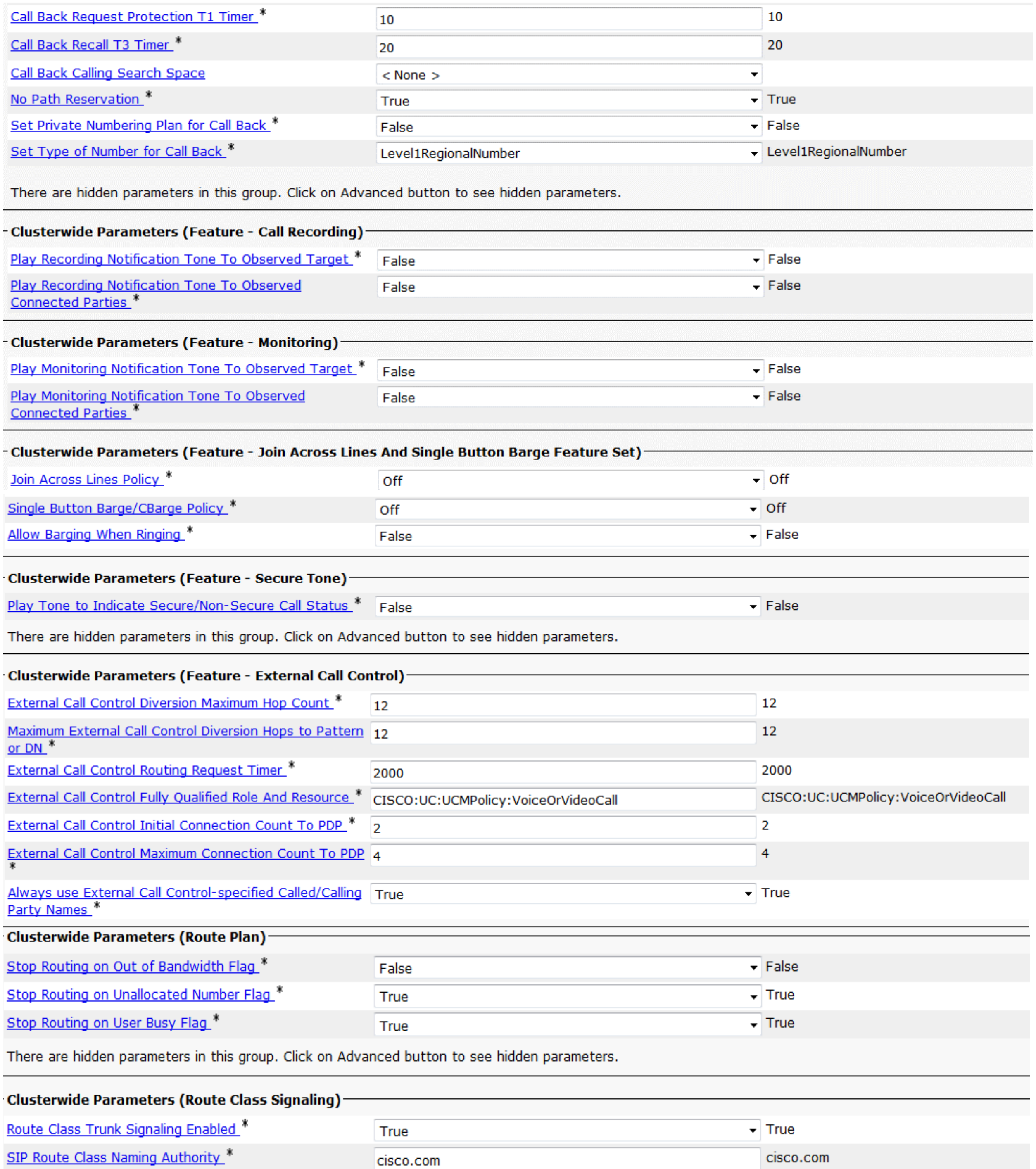


Figure 17: Service Parameters (Cont.)

Clusterwide Parameters (Hunt List)

Stop Hunting on Out of Bandwidth Flag *	False	False
Use Pickup Group Of Line Group Member DN *	False	False

Clusterwide Parameters (External QoS)

External QoS Enabled *	False	False

Clusterwide Parameters (Service)

Default Network Hold MOH Audio Source ID *	1	1
Default User Hold MOH Audio Source ID *	1	1
Duplex Streaming Enabled *	True	False
Media Exchange Interface Capability Timer *	8	8
Send Multicast MOH in H.245 OLC Message *	True	True
Media Exchange Timer *	12	12
Media Exchange Stop Streaming Timer *	8	8
Open Video Channel Response Timer for SIP Interop *	500	500
Port Received Timer After Call Connection *	500	500
Media Resource Allocation Timer *	12	12
MTP and Transcoder Resource Throttling Percentage *	95	95
Intercluster Capabilities Mismatch Timer *	1000	1000
Silence Suppression *	False	False
Silence Suppression for Gateways *	False	False
Strip G.729 Annex B (Silence Suppression) from Capabilities *	False	False
Enable Source IP Address Verification for Software Media Devices *	True	True

Clusterwide Parameters (System - General)

Always Use Dial Tone Setting *	Default	Default
Restart Cisco CallManager on Initialization Exception *	True	True
Digit Analysis Timer *	6	6
Statistics Enabled *	True	True

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

Clusterwide Parameters (System - QOS)

Priority Class *	Normal Priority	Normal Priority
DSCP for Audio Calls *	46 (101110)	46 (101110)
DSCP for Video Calls *	34 (100010)	34 (100010)
DSCP for Audio Portion of Video Calls *	34 (100010)	34 (100010)
DSCP for TelePresence Calls *	32 (100000)	32 (100000)
DSCP for Audio Portion of TelePresence Calls *	32 (100000)	32 (100000)
DSCP for Priority Audio Calls *	45 (101101)	45 (101101)
DSCP for Immediate Audio Calls *	44 (101100)	44 (101100)
DSCP for Flash Audio Calls *	41 (101001)	41 (101001)

Figure 18: Service Parameters (Cont.)

DSCP for Flash Override Audio Calls *	42 (101010)	42 (101010)
DSCP for Executive Override Audio Calls *	42 (101010)	42 (101010)
DSCP for Priority Video Calls *	39 (100111)	39 (100111)
DSCP for Immediate Video Calls *	37 (100101)	37 (100101)
DSCP for Flash Video Calls *	35 (100011)	35 (100011)
DSCP for Flash Override Video Calls *	33 (100001)	33 (100001)
DSCP for Executive Override Video Calls *	33 (100001)	33 (100001)
DSCP for G.Clear Calls *	46 (101110)	46 (101110)
DSCP for Priority G.Clear Calls *	45 (101101)	45 (101101)
DSCP for Immediate G.Clear Calls *	44 (101100)	44 (101100)
DSCP for Flash G.Clear Calls *	41 (101001)	41 (101001)
DSCP for Flash Override G.Clear Calls *	42 (101010)	42 (101010)
DSCP for Executive Override G.Clear Calls *	42 (101010)	42 (101010)
DSCP for Audio Calls when RSVP Fails *	0 (000000)	0 (000000)
DSCP for Video Calls when RSVP Fails *	0 (000000)	0 (000000)
DSCP for ICCP Protocol Links *	24 (011000)	24 (011000)
Clusterwide Parameters (System - SDL)		
SDL Listening Port Number *	8002	8002
SDL Max Router Latency *	20	20
Suppress Debug Info for Router Death *	0	0
Asynchronous SDL Logging Enabled *	False	False
Clusterwide Parameters (System - Location and Region)		
Enforce Millisecond Packet Size *	True	True
Locations Trace Details Enabled *	False	False
Preferred G.711 Millisecond Packet Size *	20	20
Preferred G.722 Millisecond Packet Size *	20	20
Preferred G.723.1 Millisecond Packet Size *	30	30
Preferred G.729 Millisecond Packet Size *	20	20
Always Use Preferred G.729 Packet Size For SIP Trunk Answers *	False	False
Preferred GSM EFR Bytes Packet Size *	31	31
G.711 A-law Codec Enabled *	Enabled for All Devices	Enabled for All Devices
G.711 mu-law Codec Enabled *	Enabled for All Devices	Enabled for All Devices
G.722 Codec Enabled *	Enabled for All Devices	Enabled for All Devices
iLBC Codec Enabled *	Enabled for All Devices	Enabled for All Devices
iSAC Codec Enabled *	Enabled for All Devices	Enabled for All Devices
Opus Codec Enabled *	Enabled for All Devices	Enabled for All Devices
Default Intraregion Max Audio Bit Rate *	64 kbps (G.722, G.711)	64 kbps (G.722, G.711)
Default Interregion Max Audio Bit Rate *	8 kbps (G.729)	8 kbps (G.729)
Default Intraregion Max Video Call Bit Rate (Includes Audio) *	384	384
Default Interregion Max Video Call Bit Rate (Includes Audio) *	384	384
Default Intraregion Max Immersive Video Call Bit Rate (Includes Audio) *	2000000000	2000000000
Default Interregion Max Immersive Video Call Bit Rate (Includes Audio) *	2000000000	2000000000

Figure 19: Service Parameters (Cont.)

Default Interregion Max Immersive Video Call Bit Rate (Includes Audio) *	2000000000	2000000000
Use Video BandwidthPool for Immersive Video Calls *	True	True
Default Intraregion and Interregion Link Loss Type *	Low Loss	Low Loss
Default Audio Codec List between Regions *	Factory Default low loss	Factory Default low loss
Default Audio Codec List within Region *	Factory Default low loss	Factory Default low loss
Accept Audio Codec Preferences in Received Offer *	Off	Off
G.Clear Bandwidth Override *	False	False

Clusterwide Parameters (System - CCM Automated Alternate Routing)

Automated Alternate Routing Enable *	False	False

Clusterwide Parameters (System - RSVP)

Default inter-location RSVP Policy *	No Reservation	No Reservation
RSVP Retry Timer *	60	60
Mandatory RSVP Mid-call Retry Counter *	1	1
Mandatory RSVP mid call error handle option *	Call becomes best effort	Call becomes best effort
RSVP Video Tspec Burst Size Factor *	5	5
MLPP EXECUTIVE_OVERRIDE To RSVP Priority Mapping *	65535	65535
MLPP FLASH_OVERRIDE To RSVP Priority Mapping *	65534	65534
MLPP FLASH To RSVP Priority Mapping *	65533	65533
MLPP IMMEDIATE To RSVP Priority Mapping *	65532	65532
MLPP PL_PRIORITY To RSVP Priority Mapping *	65531	65531
MLPP PL_ROUTINE To RSVP Priority Mapping *	65530	65530
RSVP Audio Application ID *	AudioStream	AudioStream
RSVP Video Application ID *	VideoStream	VideoStream
RSVP Response Timer *	2	2

TLS Packet Capture Configurations

Packet Capture Enable *	False	False
Packet Capture Max File Size (MB) *	2	2

Clusterwide Parameters(System - Presence)

Presence Subscription Throttling Threshold *	60000	60000
Presence Subscription Resume Threshold *	80	80
Default Inter-Presence Group Subscription *	Disallow Subscription	Disallow Subscription
BLF Status Depicts DND *	False	False

Clusterwide Parameters (System - Mobility)

Enterprise Feature Access Code for Hold *	*81	*81
Enterprise Feature Access Code for Exclusive Hold *	*82	*82
Enterprise Feature Access Code for Resume *	*83	*83
Enterprise Feature Access Code for Transfer *	*84	*84
Enterprise Feature Access Code for Conference *	*85	*85

Figure 20: Service Parameters (Cont.)

Enterprise Feature Access Code for Session Handoff *	*74	*74
Enterprise Feature Access Code for Starting Selective Recording *	*86	*86
Enterprise Feature Access Code for Stopping Selective Recording *	*87	*87
Smart Mobile Phone Interdigit Timer *	500	500
Non-Smart Mobile Phone Interdigit Timer *	2000	2000
Send Call to Mobile Menu Timer *	60	60
SIP Dual Mode Alert Timer *	1500	1500
Call Screening Timer *	4000	4000
Session Resumption Await Timer *	180	180
Inbound Calling Search Space for Remote Destination *	Trunk or Gateway Inbound Calling Search Space	Trunk or Gateway Inbound Calling Search Space
Enable Enterprise Feature Access *	False	False
Dial-via-Office Forward Service Access Number		
Enable Mobile Voice Access *	False	False
Mobile Voice Access Number		
Matching Caller ID with Remote Destination *	Complete Match	Complete Match
Number of Digits for Caller ID Partial Match *	10	10
Enable Use of Called Party Transformed Number for Mobile-terminated Calls *	False	False
Honor Gateway or Trunk Outbound Calling Party Selection for Mobile Connect Calls *	False	False

Clusterwide Parameters (System - Mobility Single Number Reach Voicemail)

Single Number Reach Voicemail Policy *	Timer Control	Timer Control
Dial-via-Office Reverse Voicemail Policy *	Timer Control	Timer Control
User Control Delayed Announcement Timer *	1000	1000
User Control Confirmed Answer Indication Timer *	10000	10000

Clusterwide Parameters (Feature - Reroute Remote Destination Calls to Enterprise Number)

Reroute Remote Destination Calls to Enterprise Number *	False	False
Ring All Shared Lines *	False	False
Ignore Call Forward All on Enterprise DN *	True	True

Clusterwide Parameters (Feature - Immediate Divert)

Use Legacy Immediate Divert *	True	True
Allow QSIG during iDivert *	False	False
Immediate Divert User Response Timer *	5	5

Figure 21: Service Parameters (Cont.)

Clusterwide Parameters (Call Admission Control)

Call Counting CAC Enabled *	False	False
Audio Bandwidth For Call Counting CAC *	102	102
Video Bandwidth For Call Counting CAC *	500	500
UCM to LBM Periodic Reservation Refresh Timer *	5	5
Maximum Bandwidth Deduction Duration *	720	720
Call Treatment When No LBM Available *	Allow Calls	Allow Calls
Locations Media Resource Audio Bit Rate Policy *	Lowest Bit Rate	Lowest Bit Rate
Video Call QoS Marking Policy *	Default	Default
Deduct Audio Bandwidth Portion from Audio Pool for a Video Call *	False	False

Clusterwide Parameters (Emergency Calling for Require Off-premise Location)

Alternate Destination for Emergency Call		
Alternate Calling Search Space for Emergency Call	< None >	

Figure 22: Service Parameters (Cont.)

Offnet Calls via TalkTalk SIP Trunk

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

SIP Trunk Security Profile

Navigation Path: System > Security > SIP Trunk Security Profile

Name*= TalkTalk Non Secure SIP Trunk Profile

Description = non Secure SIP Trunk Profile authenticated by null String

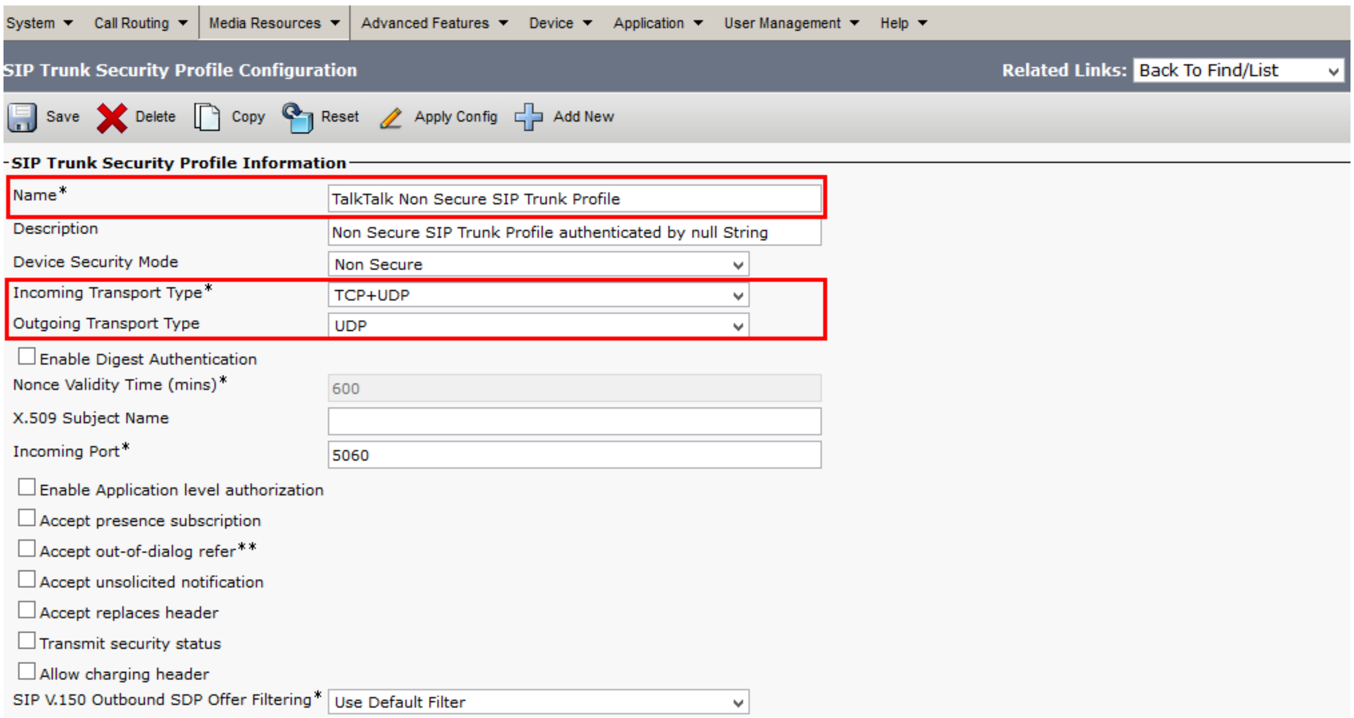


Figure 23: SIP Trunk Security Profile

Explanation

Parameter	Value	Description
Incoming Transport Type	TCP + UDP	
Outgoing Transport Type	UDP	SIP trunks to TalkTalk 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*= TalkTalk Standard SIP Profile

Description = Default SIP Profile

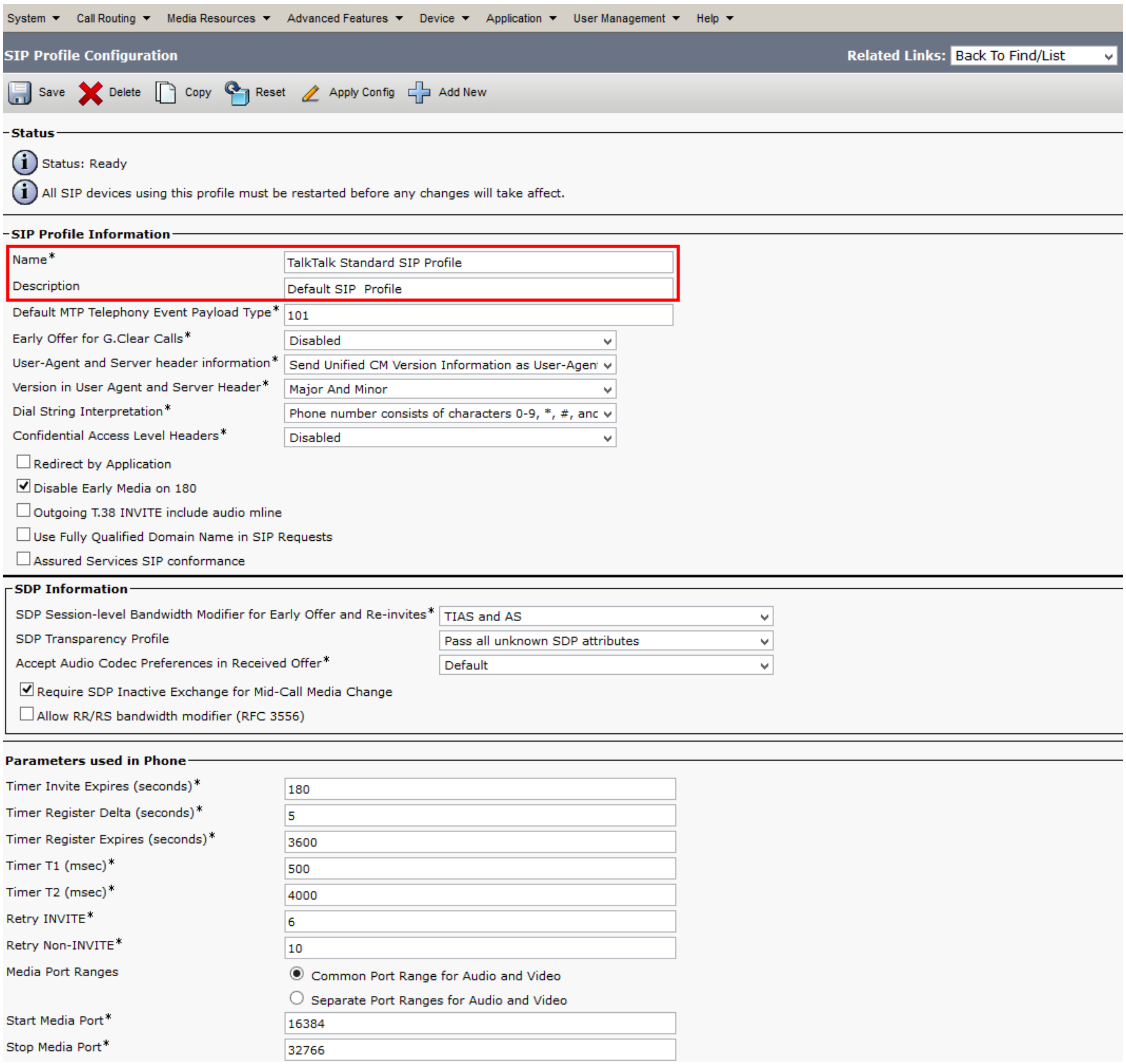


Figure 24: SIP Profile

DSCP for Audio Calls	Use System Default
DSCP for Video Calls	Use System Default
DSCP for Audio Portion of Video Calls	Use System Default
DSCP for TelePresence Calls	Use System Default
DSCP for Audio Portion of TelePresence Calls	Use System Default
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off
Caller ID Blocking*	Off
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
Resource Priority Namespace	< None >
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70
Off Hook To First Digit Timer (milliseconds)*	15000
Call Forward URI*	x-cisco-serviceuri-cfwdall
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial

- [x] Conference Join Enabled
- [ ] RFC 2543 Hold
- [x] Semi Attended Transfer
- [ ] Enable VAD
- [ ] Stutter Message Waiting
- [ ] MLPP User Authorization

Normalization Script

Normalization Script: < None >

- [ ] Enable Trace

	Parameter Name	Parameter Value
1		

Incoming Requests FROM URI Settings

Caller ID DN	
Caller Name	

Figure 25: SIP Profile (Cont.)

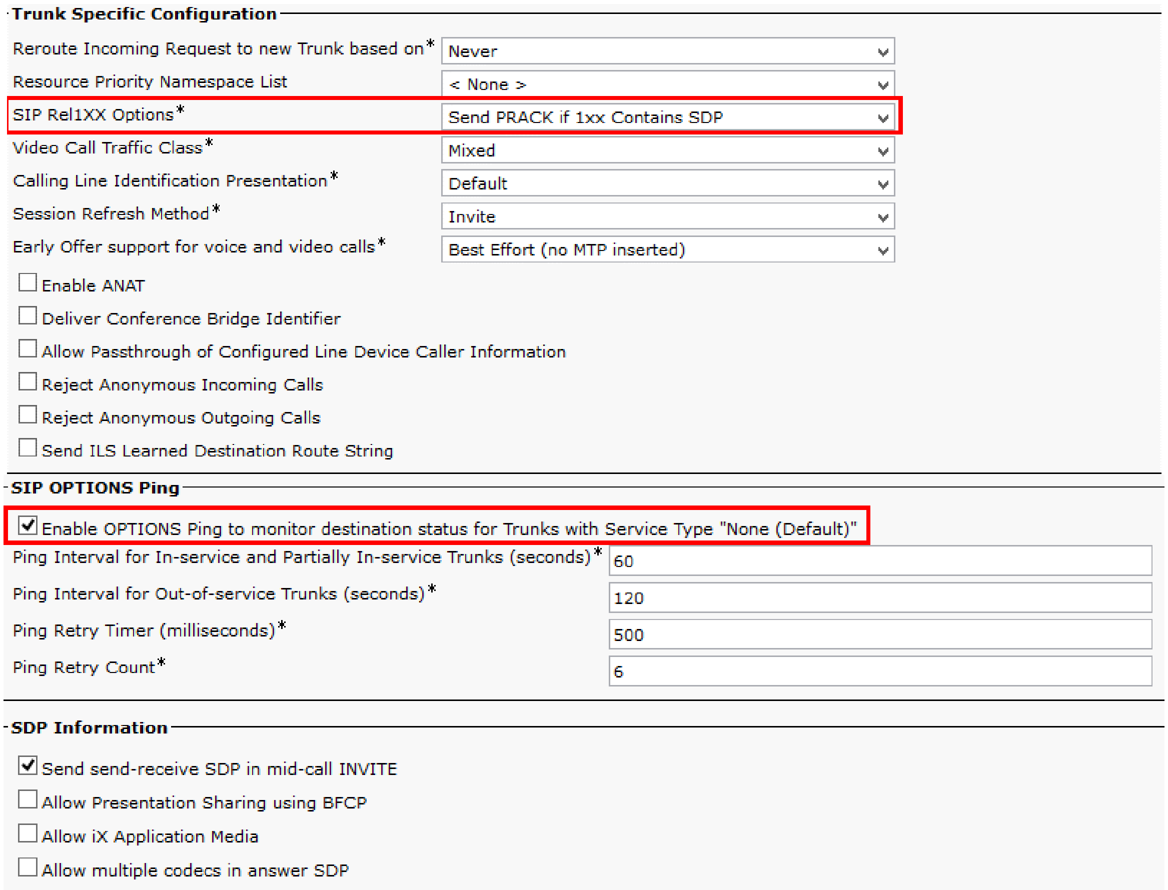


Figure 26: SIP Profile (Cont.)

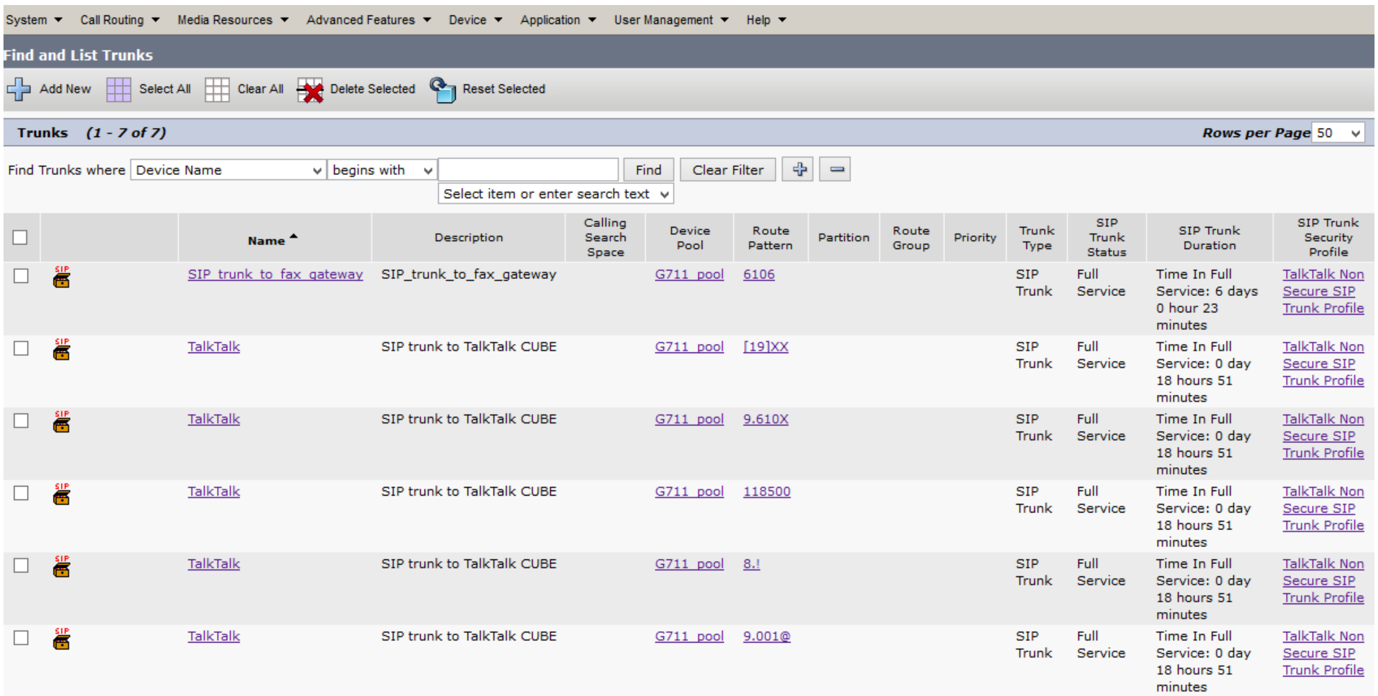
Explanation

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

SIP Trunk Configuration

Create SIP trunks to Cisco UBE

Navigation Path: Device > Trunk



Name	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type	SIP Trunk Status	SIP Trunk Duration	SIP Trunk Security Profile
SIP_trunk_to_fax_gateway	SIP_trunk_to_fax_gateway		G711_pool	6106				SIP Trunk	Full Service	Time In Full Service: 6 days 0 hour 23 minutes	TalkTalk Non Secure SIP Trunk Profile
TalkTalk	SIP trunk to TalkTalk CUBE		G711_pool	[19]XX				SIP Trunk	Full Service	Time In Full Service: 0 day 18 hours 51 minutes	TalkTalk Non Secure SIP Trunk Profile
TalkTalk	SIP trunk to TalkTalk CUBE		G711_pool	9.610X				SIP Trunk	Full Service	Time In Full Service: 0 day 18 hours 51 minutes	TalkTalk Non Secure SIP Trunk Profile
TalkTalk	SIP trunk to TalkTalk CUBE		G711_pool	118500				SIP Trunk	Full Service	Time In Full Service: 0 day 18 hours 51 minutes	TalkTalk Non Secure SIP Trunk Profile
TalkTalk	SIP trunk to TalkTalk CUBE		G711_pool	8.!				SIP Trunk	Full Service	Time In Full Service: 0 day 18 hours 51 minutes	TalkTalk Non Secure SIP Trunk Profile
TalkTalk	SIP trunk to TalkTalk CUBE		G711_pool	9.001@				SIP Trunk	Full Service	Time In Full Service: 0 day 18 hours 51 minutes	TalkTalk Non Secure SIP Trunk Profile

Figure 27: SIP Trunks List

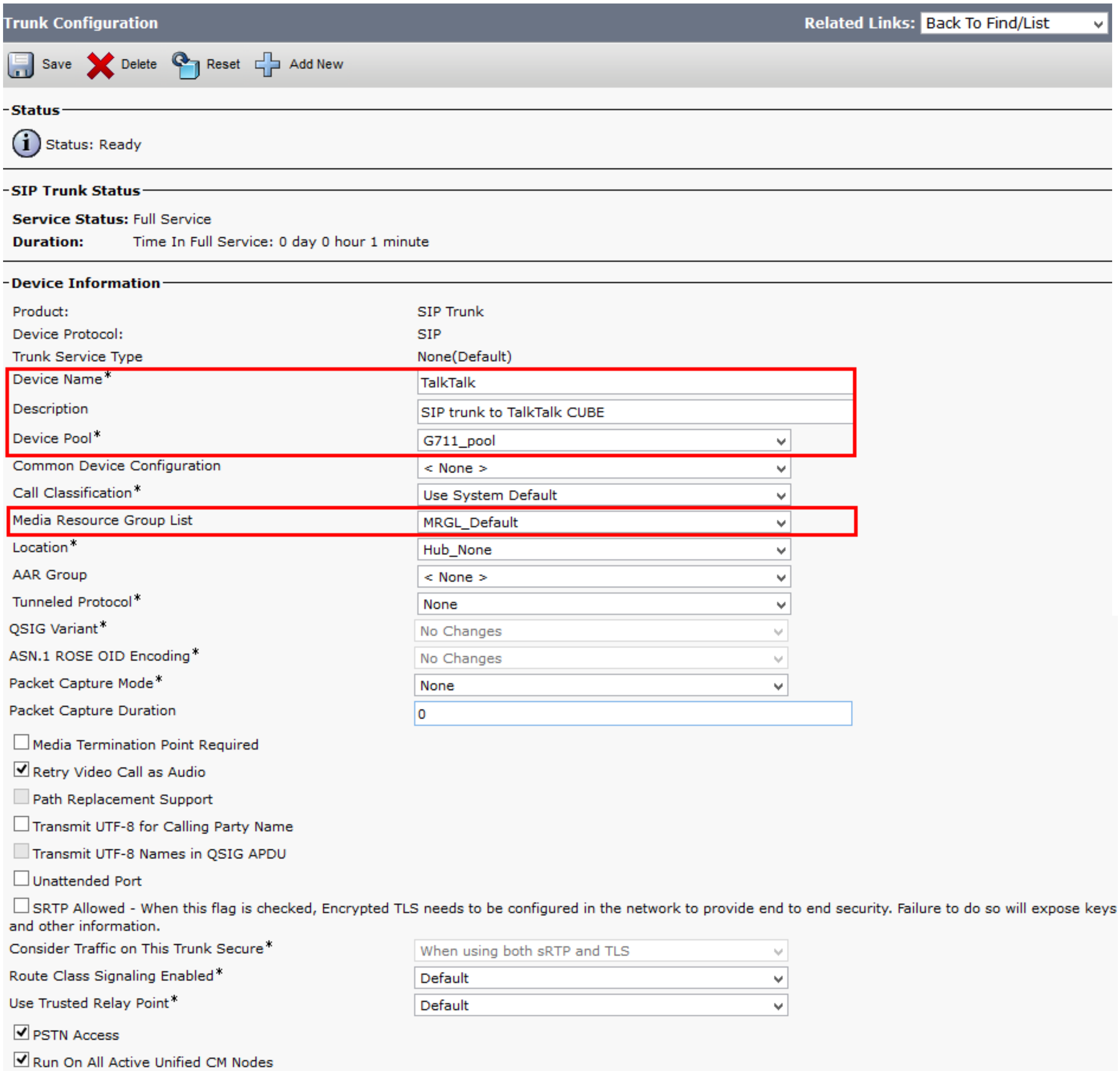


Figure 28: SIP Trunk to Cisco UBE

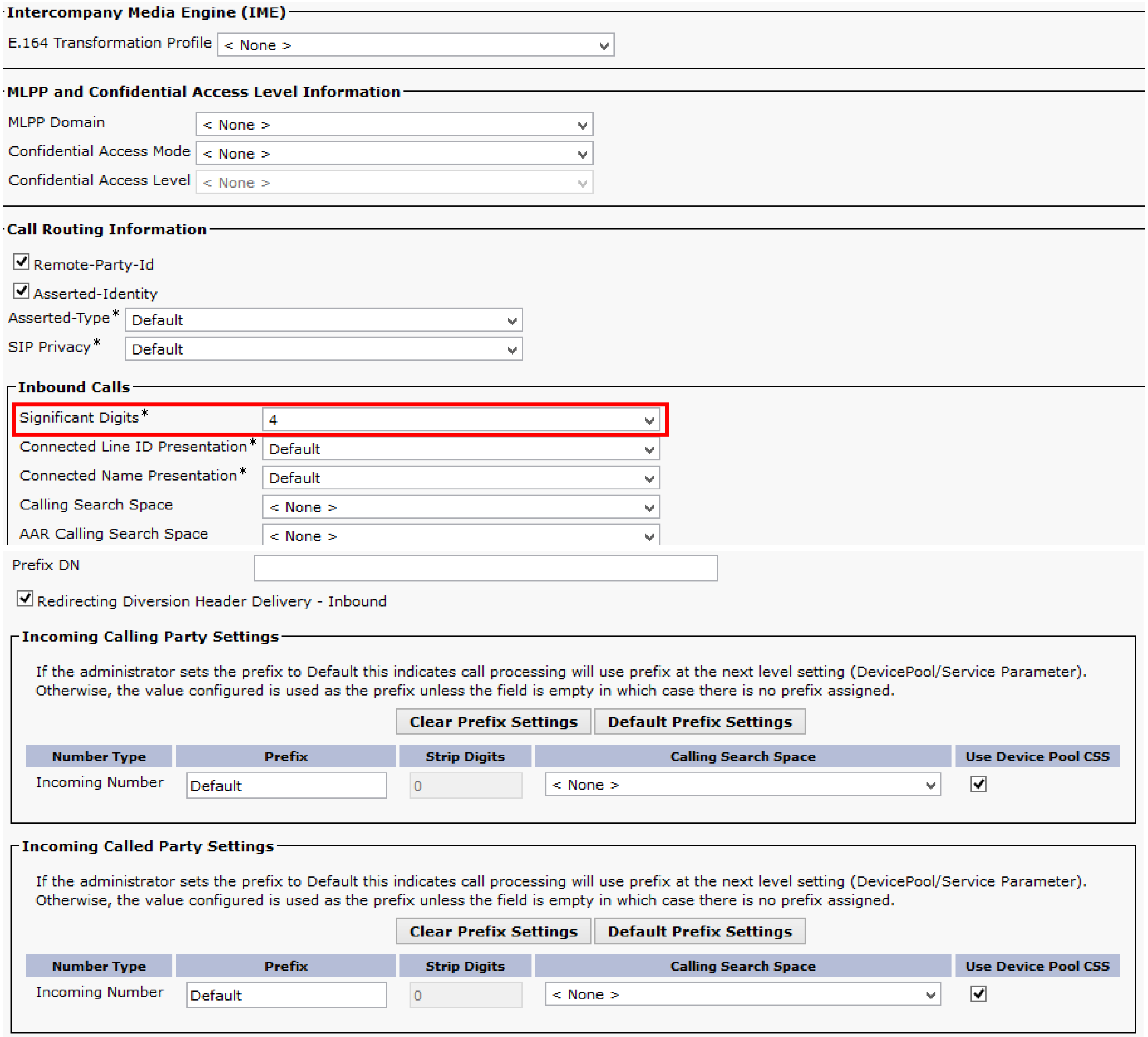


Figure 29: SIP Trunk to Cisco UBE (cont.)

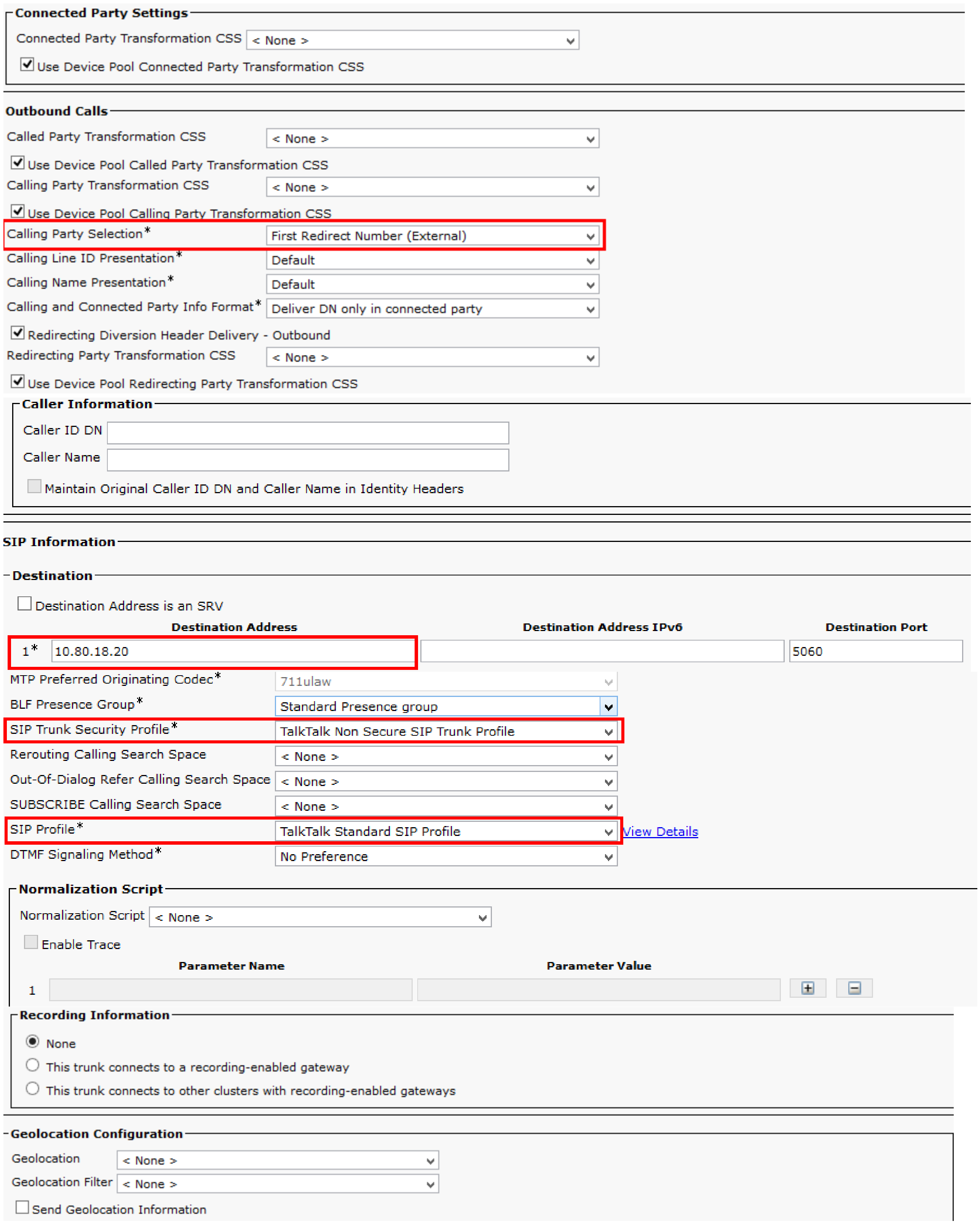


Figure 30: SIP Trunk to Cisco UBE (Cont.)

Explanation

Parameter	Value	Description
Device Name	TalkTalk	Name for the trunk
Device Pool	G711_pool	Default Device Pool is used for this trunk
Media Resource Group List	MRGL_Default	MRG with resources: ANN, CFB, MOH and MTP
Significant Digits	4	4 digits Extension for all CPE phones
Destination Address	10.80.18.20	IP address of the Cisco UBE Virtual LAN
SIP Trunk Security Profile	TalkTalk Non Secure SIP Trunk Profile	SIP Trunk Security Profile configured earlier
SIP Profile	TalkTalk 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 “9”.001+10 digits number to access PSTN via Cisco UBE
  - “9” is removed before sending to Cisco UBE
- For FAX call, Access Code “8”+ 0091+10 digits number is used at Cisco Fax gateway
  - “8” is removed at Cisco UCM
  - The rest of the number is sent to Cisco UBE to TalkTalk network
- Incoming fax call to 6106 will be sent to Cisco Fax gateway
- Cisco IP phones dial 1XX and 9XX for emergency call and will send all digits to Cisco UBE to TalkTalk network

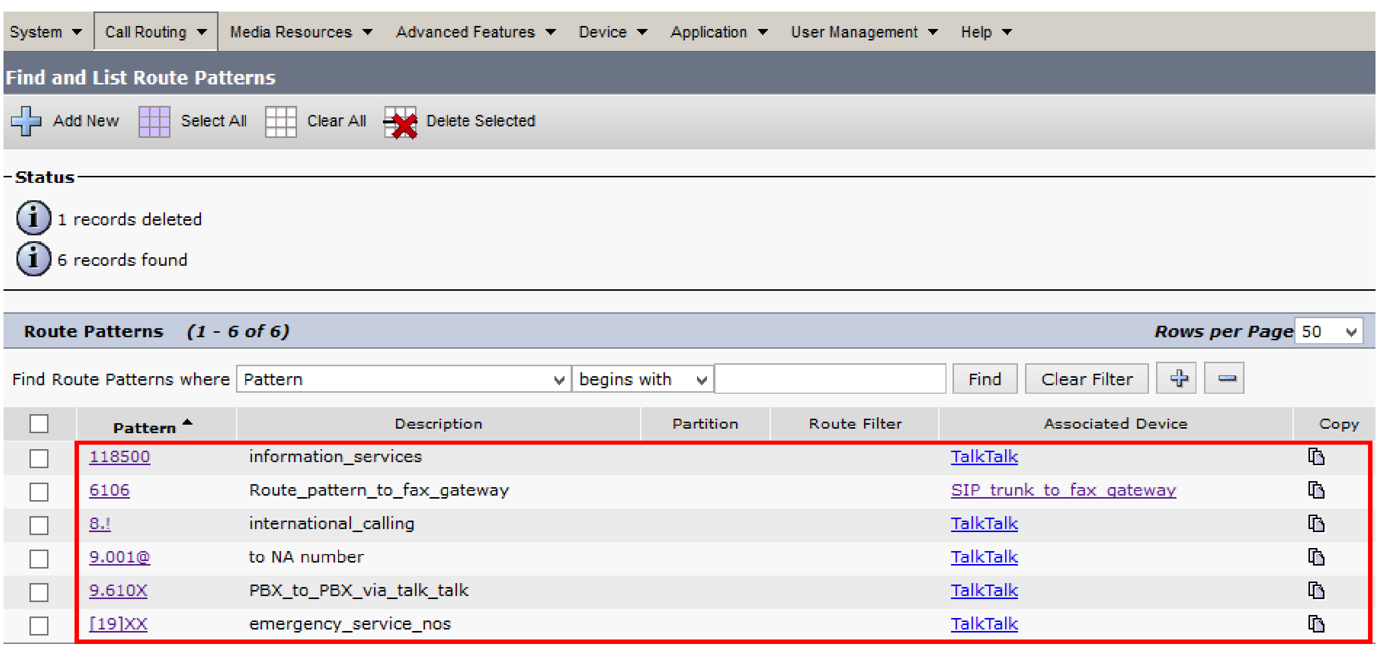
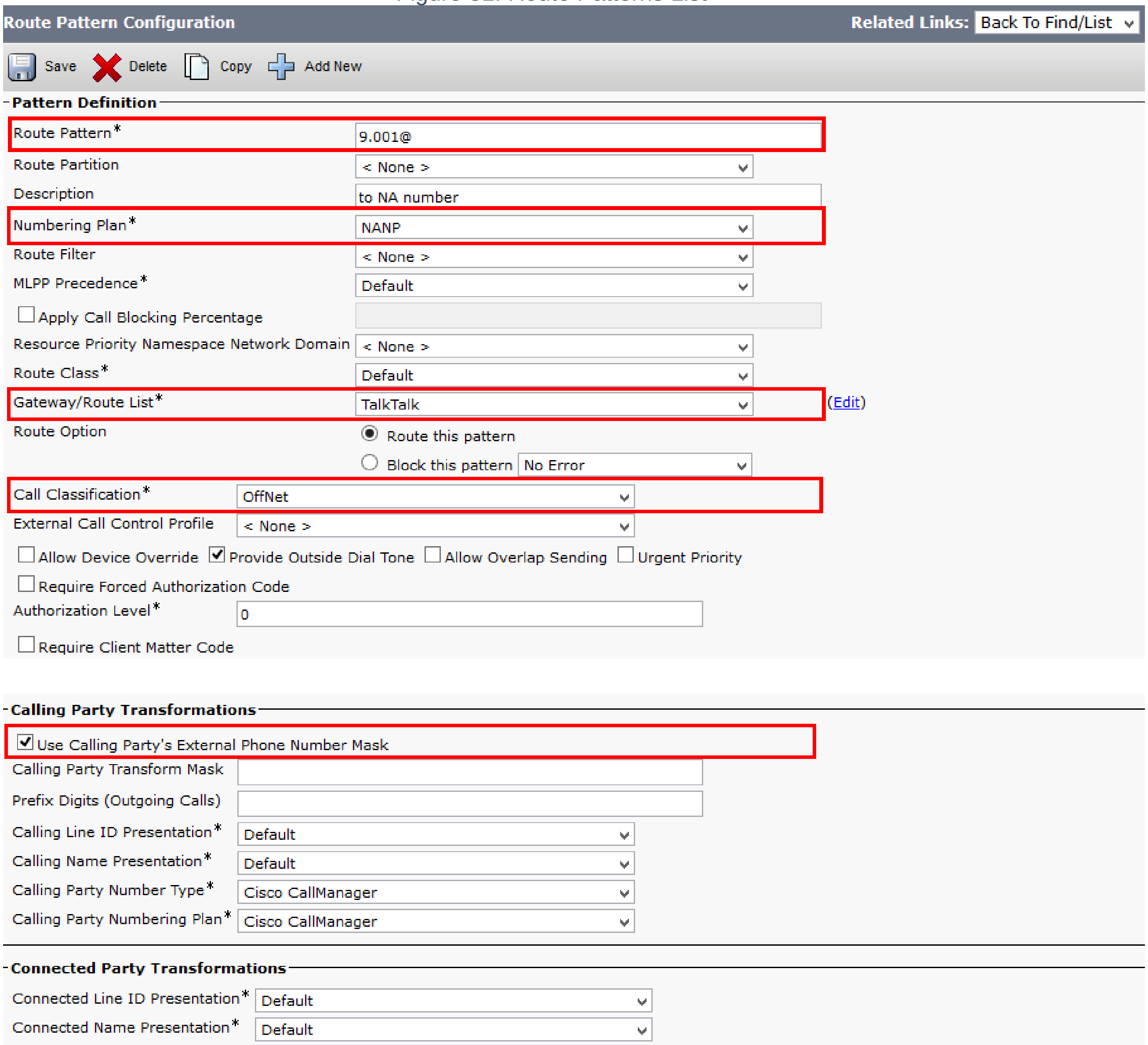


Figure 31: Route Patterns List



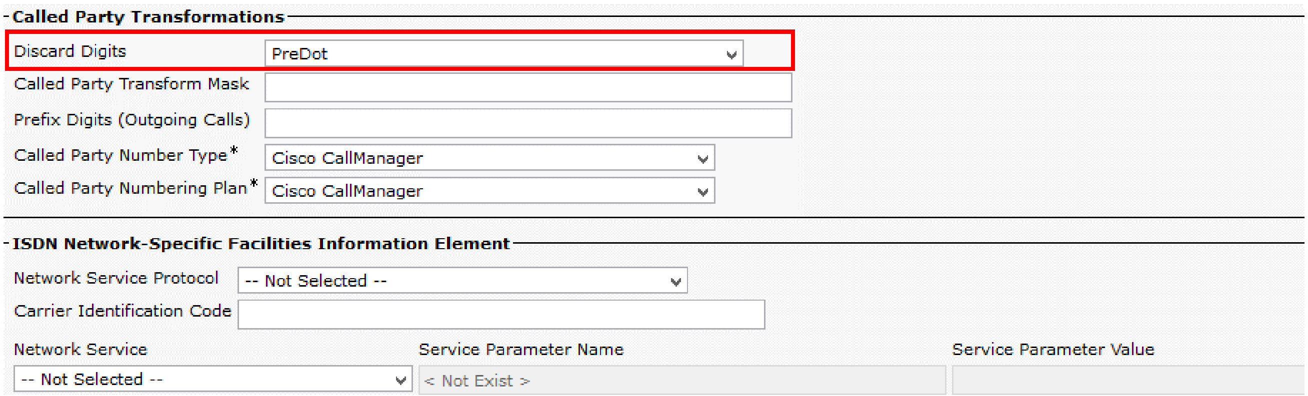


Figure 32: Route Pattern for Voice

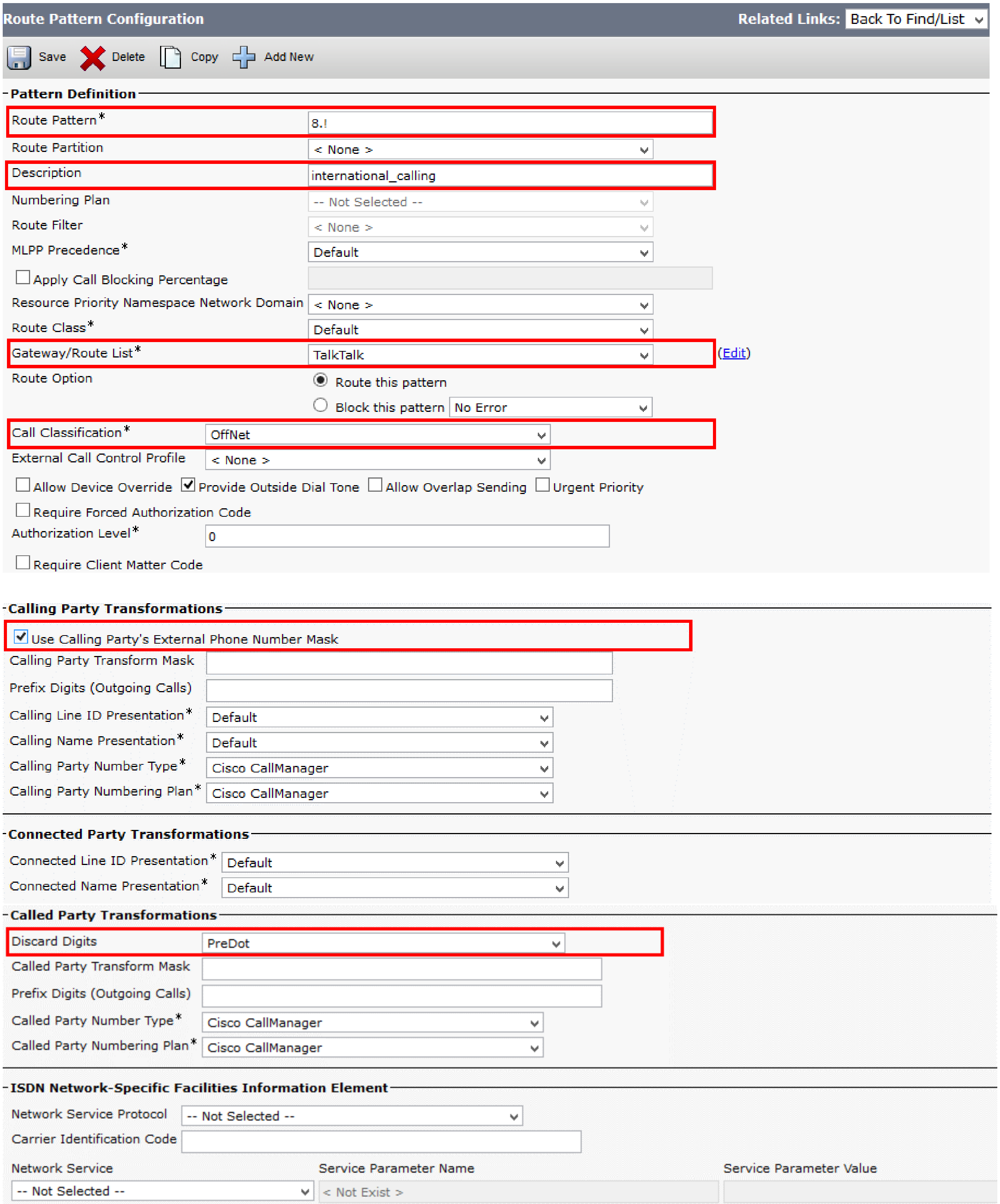


Figure 33: Route Pattern for Voice (Cont.)

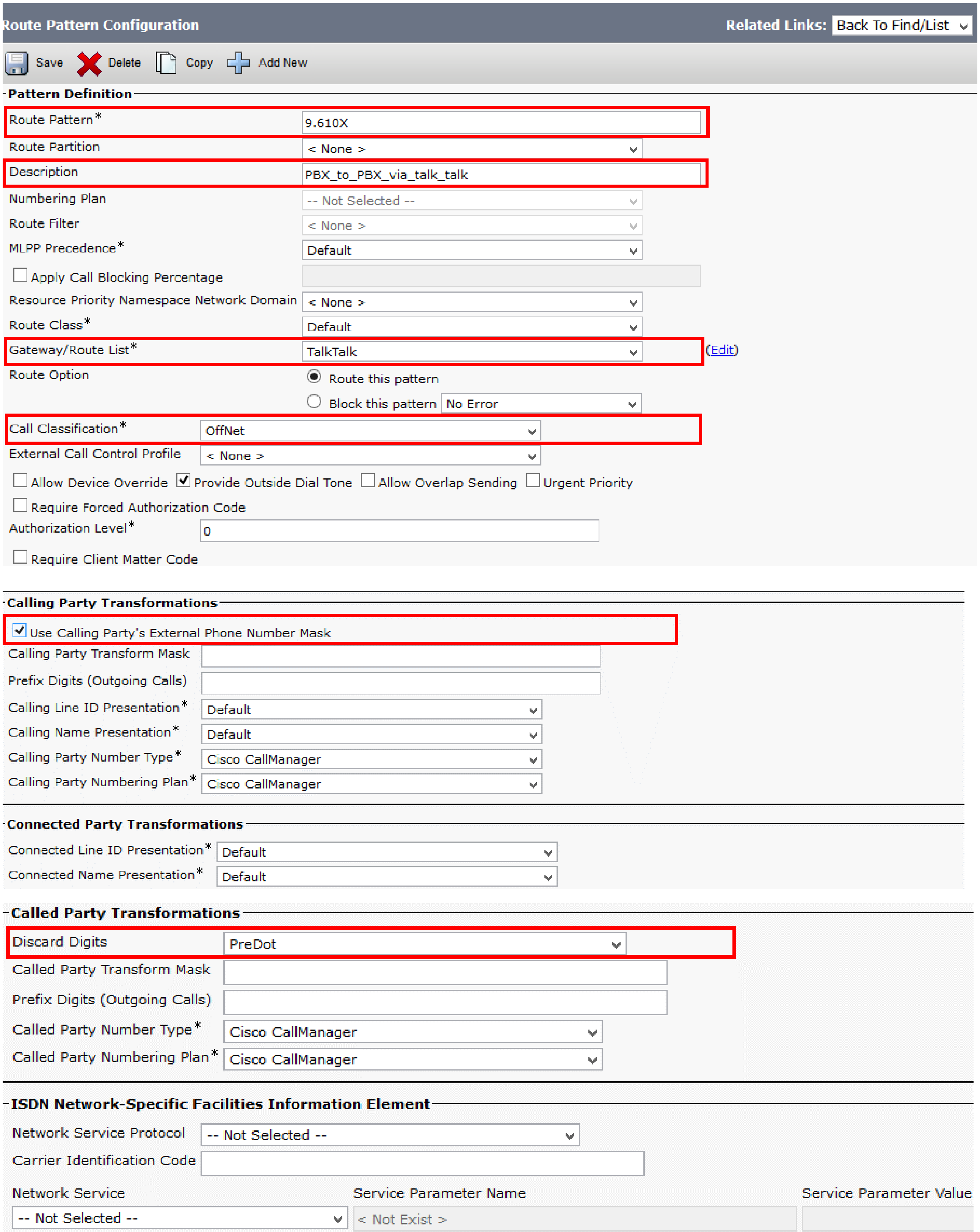


Figure 34: Route Pattern for Voice (Cont.)

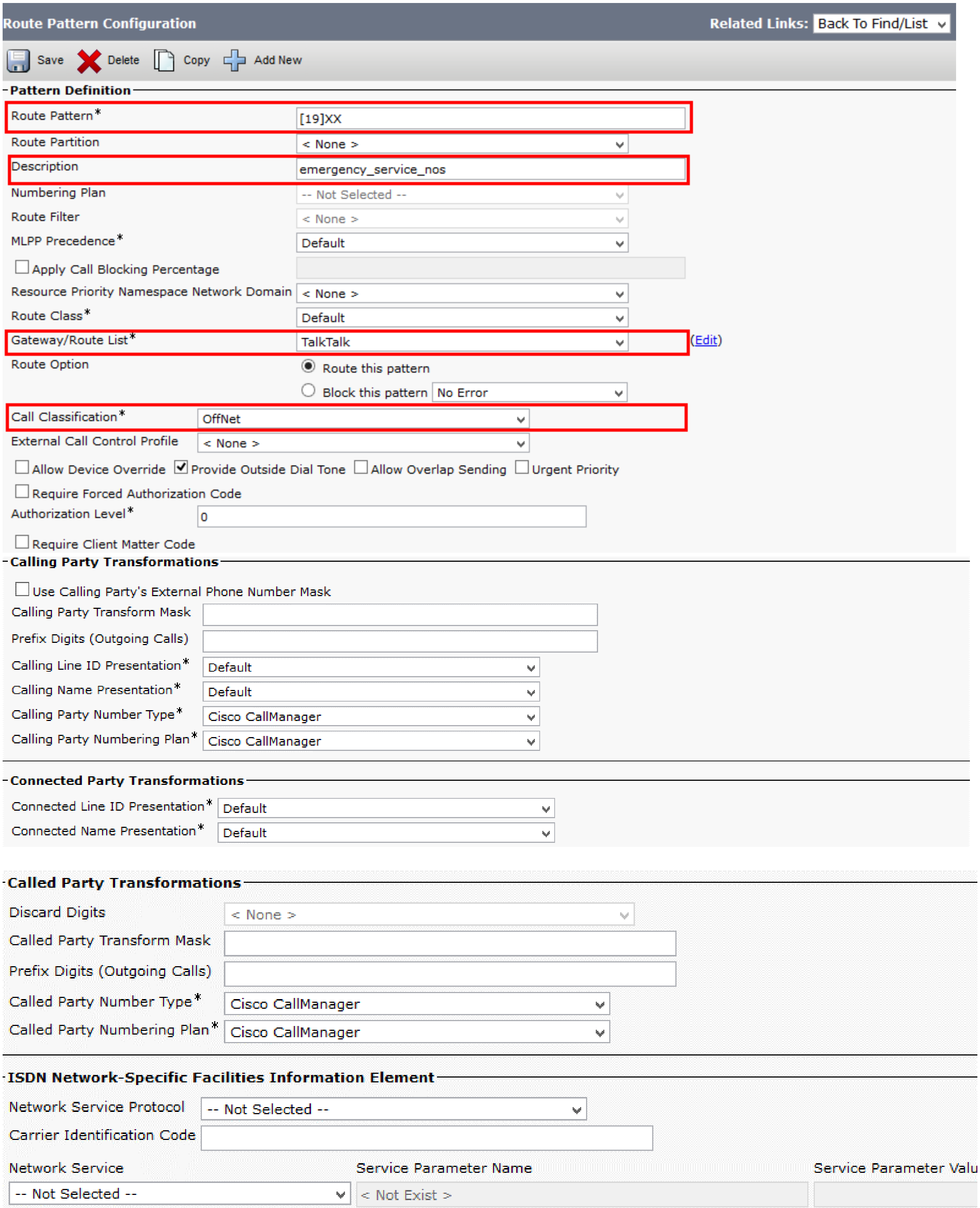


Figure 35: Route Pattern for Voice (Cont.)

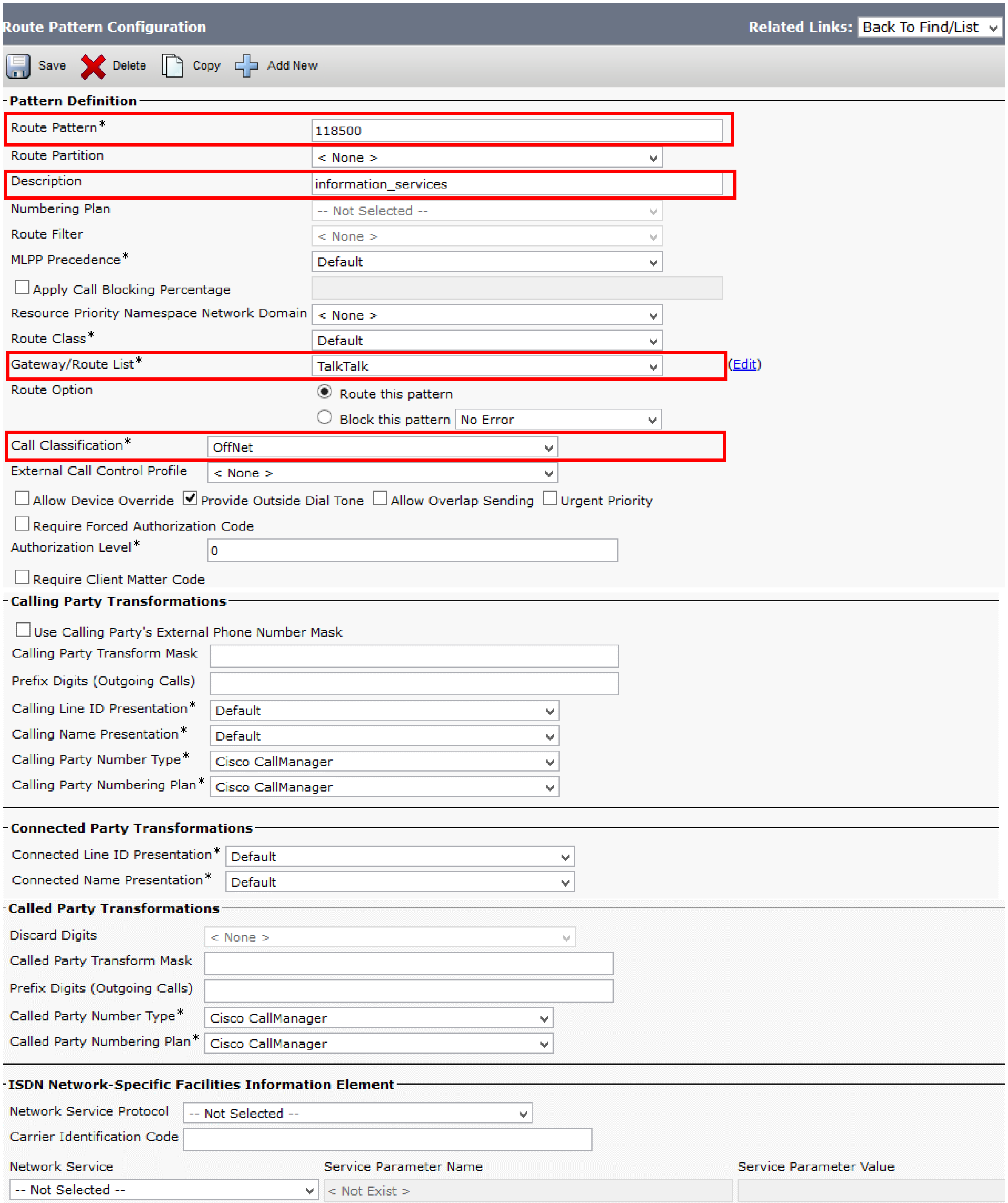


Figure 36: Route Pattern for Voice (Cont.)

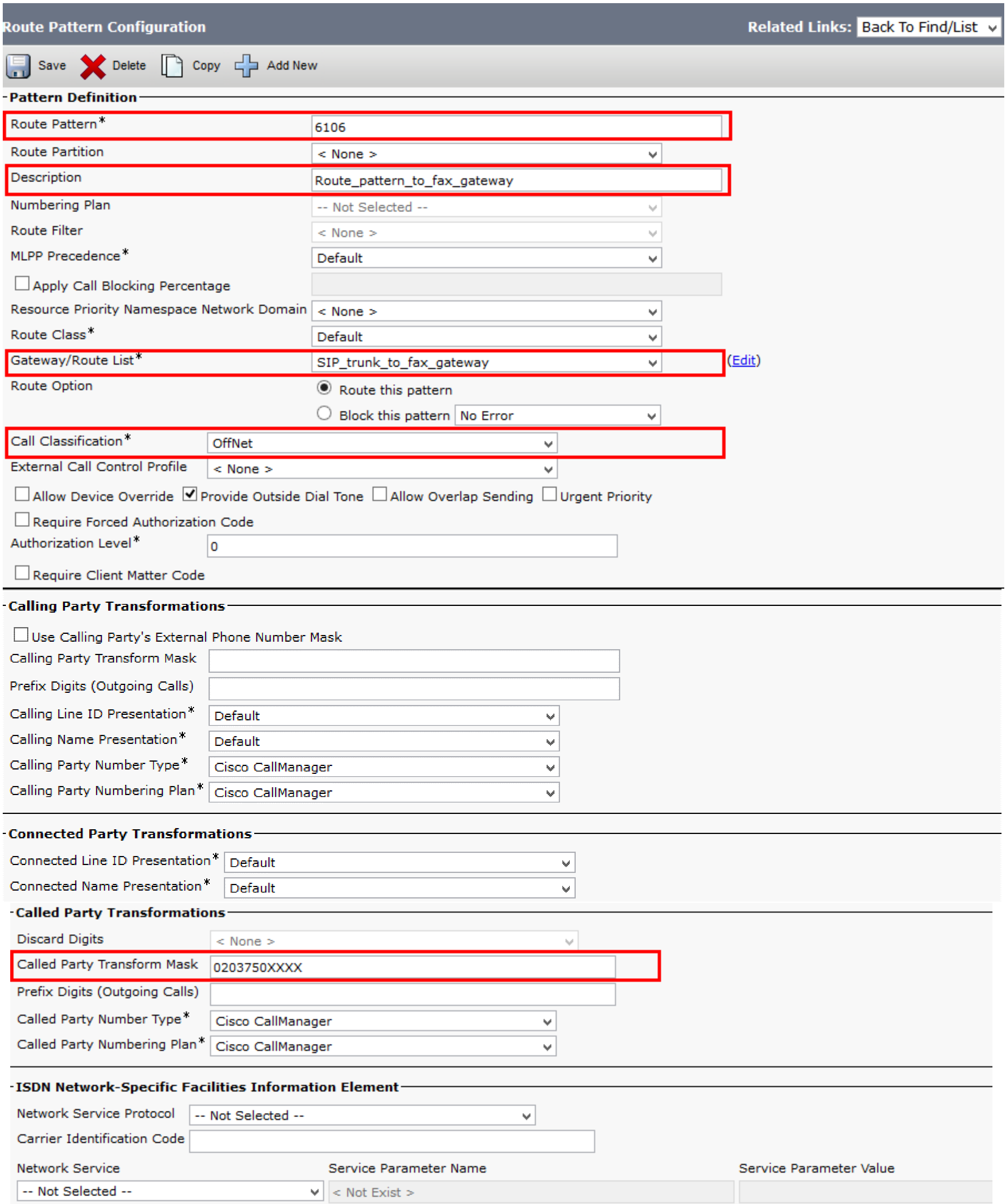


Figure 37: Route Pattern for Fax

Explanation

Setting	Value	Description
Route Pattern	9.001@ for Voice call, 8.! For International Calls, 6106 for fax call & 9.610X for CPE to CPE call via TALKTALK, 118500 for information services, [19]XX for operator call and emergency services	Specify appropriate Route Pattern
Gateway/Route List	TalkTalk for Route Pattern 9.001@, [19]XX ,118500, 9.610X, 8.!, SIP_trunk _to_ fax _ gateway for Route Pattern 6106	SIP Trunk name configured earlier
Numbering Plan	NANP for Route Pattern 9.@	North American Numbering Plan
Call Classification	Offnet for Route Pattern 9.001@, 6106, 118500, 9.610X, 8.! and [19]XX	Restrict the transferring of an external call to an external device
Discard Digits	PreDot for Route Pattern 9.001@, 8.! , 9.610X	Specifies how to modify digit before they are sending to TalkTalk 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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