Application Note

Connecting Cisco Unified Communication Manager 11.5.1 to Deutsche Telekom All IP SIP Trunks via Cisco Unified Border Element v11.5.2 [IOS-XE 16.3.3]

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Introduction

Service Providers today, such as Deutsche Telekom, substitute the PSTN Network. The existing PSTN/ISDN network of Telekom Deutschland will be substituted by an IP based Next Generation Network (NGN) using the SIP protocol. 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.

Deutsche Telekom SIP Trunk is a service provider offering that allows connection to the PSTN and may offer the end customer a viable alternative to traditional PSTN connectivity. A demarcation device between these services and customer owned services is recommended. As an intermediary device between Cisco UCM 11.5.1 and Deutsche Telekom Network, Cisco Unified Border Element (Cisco UBE) v11.5.2 can be used. The Cisco Unified Border Element provides demarcation, security and inter-working and session control services for Cisco UCM connected to Deutsche Telekom IP network.

This document assumes the reader is knowledgeable with the terminology and configuration of Cisco UCM. Only configuration settings specifically required for Deutsche Telekom 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 UCM 11.5.1 and Cisco Unified Border Element (Cisco UBE) v11.5.2 for connectivity to Deutsche Telekom SIP Trunking service. The deployment model covered in this application note is Cisco UCM to PSTN via Cisco Unified Border Element v11.5.2 [IOS-XE] 16.3.3.
- Testing was performed in accordance to Cisco generic SIP Trunking test methodology and among features verified were – basic calls, DTMF transport, Music on Hold (MOH), unattended and attended transfers, call forward, conferences and High Availability.
- The Cisco Unified Border Element (Cisco UBE) configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between Deutsche Telekom SIP network and Cisco UCM. 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 Deutsche Telekom SIP Trunking network.

For more information, Refer: 1TR118 Technical Specification of the SIP-Trunking Interface between a SIP-PBX with DDI and the NGN Platform of Telekom Deutschland

- https://www.telekom.de/hilfe/geraete-zubehoer/telefone-und-anlagen/informationen-zu-telefonanlagen/schnittstellenbeschreibungen-fuer-hersteller?samChecked=true

Network Topology

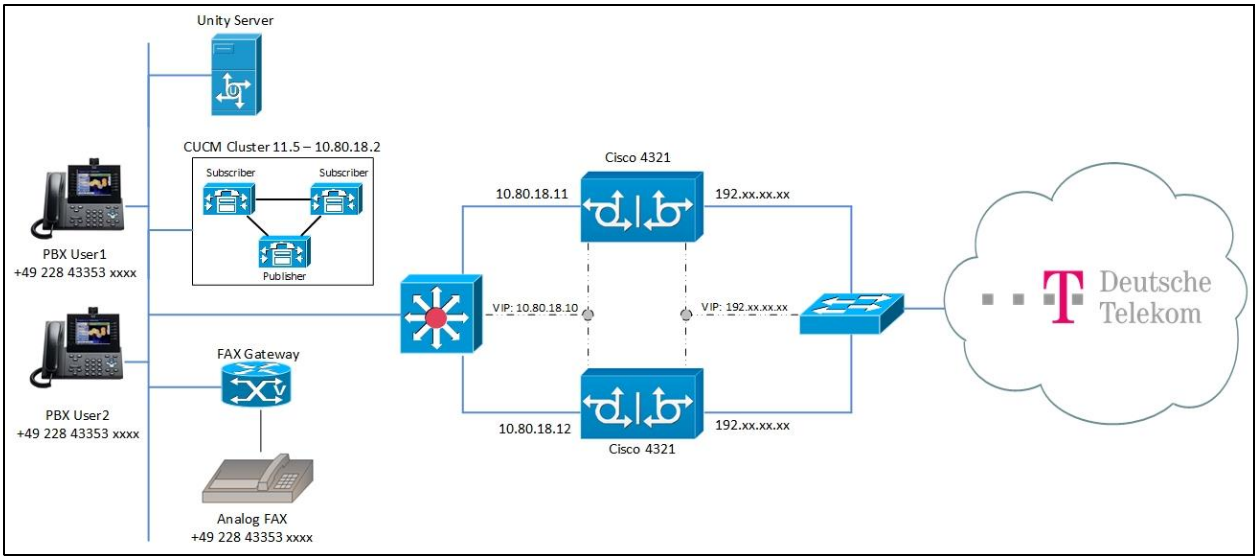

Figure 1 Network Topology

- The network topology includes the Cisco UCM Cluster, Unity Voicemail system, Cisco Fax gateway and 2 Cisco Endpoints. Cisco UCM has a trunk configured to Cisco UBE's Virtual IP Address. Deutsche Telekom was used as the service provider with SIP trunk to the Cisco UBE using the WAN Virtual IP Address.
- 2 Cisco Unified Border Elements are used here for High Availability.
- SIP Trunk transport type used between Cisco Unified Border Element and Cisco UCM is TCP and to Deutsche Telekom is TCP.

Cisco UCM and Cisco UBE Settings:

Setting	Value
Transport from Cisco UBE to Cisco UCM	TCP with RTP
Transport from Cisco UBE to Deutsche Telekom	TCP with RTP
Voice Mail Support	YES
Session Refresh	YES
Early Media support with PRACK	YES
G729 Conference Support	NO

System Components

Hardware Requirements

- Cisco UBE on Cisco ISR 4321 router
- CUCM cluster on UCS, 1 Publisher node and 2 Subscriber nodes
- Cisco 2851 with FXS ports and Analog Fax machine
- For ADSL / VDSL (not tested in this setup, but required if the SIP Trunk is offered via ADSL / VDSL):
    - NIM-VA-B (4300, 4400 Series)
    - EHWIC-VA-DSL-B (2900 and 3900 Series)
- Generic Cisco IP-Phones

Software Requirements

- CUBE-Version: 11.5.2 running IOS-XE 16.3.3
- CUCM UCOS for 1 Publisher and 2 Subscriber
- Cisco IOS v12.4 for the fax gateway

Features

Features Supported

- Incoming and outgoing off-net calls using G711alaw
- International Calls and digit manipulations
- Call Conference with G711alaw support
- CPE Voice Mail support
- Call hold & Resume with and without MoH
- Unattended and Attended Call transfer
- Call forward (all, busy and no answer)
- DTMF (RFC2833)
- Fax Pass-through
- IP-PBX Calling number privacy
- High Availability

Features Not Supported

- Cisco UCM does not support Blind Call transfer
- International Fax using T.38
- G729 voice codec
- In HA Redundancy mode, the Primary Cisco UBE will not take over the Primary/Active role after a reboot/network outage

Caveats

- As of writing this application note, Deutsche Telekom supports G.711 pass-through for faxing on DT SIP Trunk. The NGN supports the transmission of T.38 fax, in a passive, transparent way, if both user entities (caller and callee) are attached to the NGN using SIP-Trunks and they agree to use T.38 fax (offer-answer).
- Caller ID updates are not observed on attended and unattended call transfer scenarios.
- Testing is done with only one IP PBX.
- Workaround is done for SIP header manipulations for Register and P-Asserted ID
- The Cisco UBE HA tested here is layer 2 box to box Cisco UBE redundancy.
- With International Calls, in an Unattended Call Transfer Scenario, once the CPE completes the transfer, call is disconnected between both Off-net Users with DT sending "488 Not Acceptable". Tested with National Germany Number and was successful.
- With International Outbound Faxing, Fax Invite from DT contains: From<sip:anonymous@anonymous.invalid>. Tested with National Germany Numbers and this was not seen. Fax was successful to International and National destinations.

Configuration

Configuring Cisco Unified Border Element

Network Interface

The IP address used are for illustration only, the actual IP address can vary. The Active/Standby pair share the same virtual IP address and continually exchange status messages.

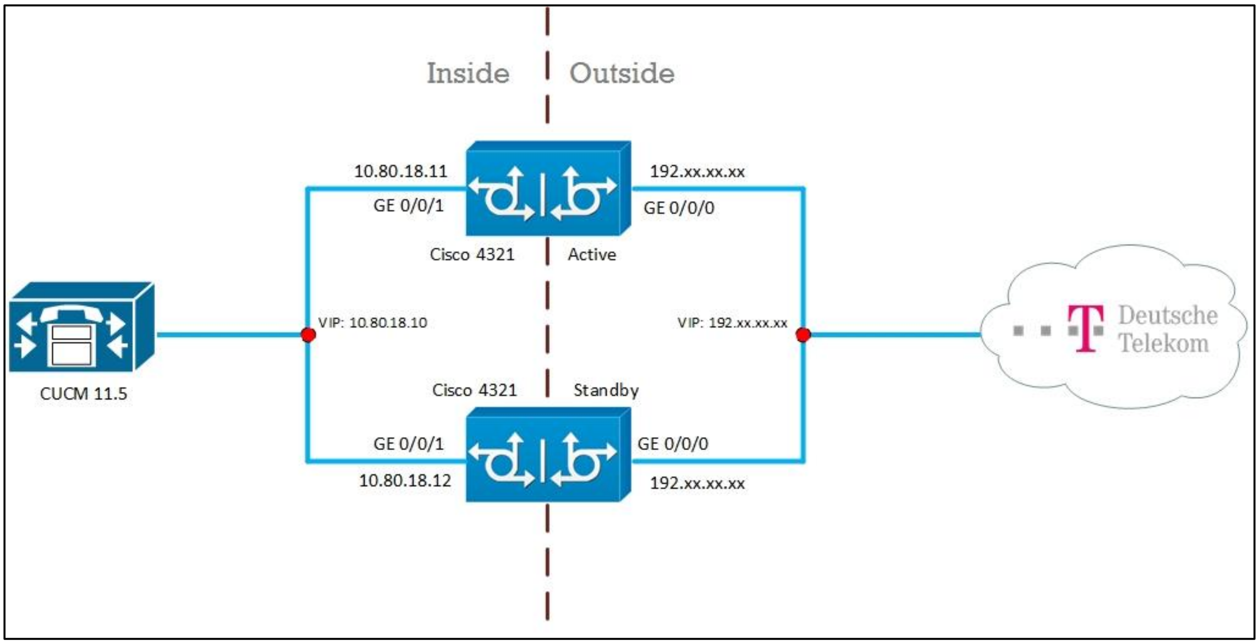

Figure 2 High Availability topology

Cisco UBE 1:

```
interface GigabitEthernet0/0/0
 description WAN Interface
 ip address 192.65.79.141 255.255.255.128
 media-type rj45
 negotiation auto
 redundancy rii 11
 redundancy group 2 ip 192.65.79.155 exclusive
 service-policy output parent
!
interface GigabitEthernet0/0/1
 description LAN Interface
 ip address 10.80.18.11 255.255.255.0
 negotiation auto
 redundancy rii 12
 redundancy group 2 ip 10.80.18.10 exclusive
!
interface GigabitEthernet0/1/0
 description CUBE HA
 ip address 10.89.20.7 255.255.255.0
 negotiation auto
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 negotiation auto
!
```

Cisco UBE 2:

```
interface GigabitEthernet0/0/0
 description WAN Interface
 ip address 192.65.79.140 255.255.255.128
 media-type rj45
 negotiation auto
 redundancy rii 11
 redundancy group 2 ip 192.65.79.155 exclusive
 service-policy output parent
!
interface GigabitEthernet0/0/1
 description LAN Interface
 ip address 10.80.18.12 255.255.255.0
 negotiation auto
 redundancy rii 12
 redundancy group 2 ip 10.80.18.10 exclusive
!
interface GigabitEthernet0/1/0
 description CUBE HA
 ip address 10.89.20.8 255.255.255.0
 negotiation auto
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 negotiation auto
!
```

Global Cisco UBE settings

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

```
voice service voip
 ip address trusted list
  ipv4 217.0.0.0 255.255.0.0
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 2
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-
through g711alaw
 sip
  bind control source-interface GigabitEthernet0/0/0
  bind media source-interface GigabitEthernet0/0/0
  session refresh
  asserted-id pai
  outbound-proxy dns:reg.sip-trunk.telekom.de
  conn-reuse
  privacy-policy passthru
  sip-profiles inbound
  sip-profiles 3000
  audio forced
```

Explanation

Command	Description
allow-connections sip to sip	Allow IP2IP connections between two SIP call legs
redundancy-group 2	Enable High Availability for the VoIP service
fax protocol	Specifies the fax protocol
asserted-id	Specifies the privacy header in the outgoing SIP requests and response messages

Codecs

G711alaw is used primarily towards Deutsche Telekom until specified otherwise.

```
voice class codec 1
 codec preference 1 g711alaw
 codec preference 2 g722-64
```

Dial peer

Outbound Dial-peer to Deutsche Telekom:

```
dial-peer voice 201 voip
 description **SIP-TRUNK.TELEKOM.DE**
 session protocol sipv2
 session target sip-server
 session transport tcp
 destination e164-pattern-map 201
 incoming called-number .T
 voice-class codec 1
 voice-class sip outbound-proxy dns:reg.sip-trunk.telekom.de
 voice-class sip profiles 201
 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 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711alaw
 ip qos dscp cs6 signaling
 clid strip name
 no vad
```

Inbound Dial-peer from Deutsche Telekom:

```
dial-peer voice 101 voip
 description **CUCM/PBX **
 destination-pattern +492284335329T
 session protocol sipv2
 session transport tcp
 session server-group 1
 incoming uri via 101
 voice-class codec 1
 no voice-class sip outbound-proxy
 voice-class sip options-keepalive profile 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 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711alaw
 no vad
```

Configuration example

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

Active Cisco UBE:

```
version 16.3
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
service password-encryption
service internal
service sequence-numbers
no platform punt-keepalive disable-kernel-core
!
hostname CUBE1
!
boot-start-marker
boot system bootflash:isr4300-universalk9.16.03.03.SPA.bin
boot-end-marker
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
no logging queue-limit
logging buffered 999999
no logging rate-limit
```

```
enable secret 5 ********
!
no aaa new-model
ip name-server 8.8.8.8
subscriber templating
multilink bundle-name authenticated
!
crypto pki trustpoint TP-self-signed-1270583006
 enrollment selfsigned
 subject-name cn=IOS-Self-Signed-Certificate-1270583006
 revocation-check none
 rsakeypair TP-self-signed-1270583006
!
!
crypto pki certificate chain TP-self-signed-1270583006
 certificate self-signed 01
  30820330 30820218 A0030201 02020101 300D0609 2A864886 F70D0101 05050030
  31312F30 2D060355 04031326 494F532D 53656C66 2D536967 6E65642D 43657274
  69666963 6174652D 31323730 35383330 3036301E 170D3137 30323130 31343237
  34345A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649
  4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D31 32373035
  38333030 36308201 22300D06 092A8648 86F70D01 01010500 0382010F 00308201
  0A028201 0100A34F 136DD7A1 E1815DA1 B05C5396 E3B88AC9 7DE1A1D7 12F1BEA7
  4985E12B C6858F9D 95E7082B 3BBC56CD AAFDEC4F 7250D4CE 892713BE 509A6DCE
  05FD3768 ED1EF293 B3C2C1CE 4684F8F9 E920AE8F 33F4DFE0 FF04BE27 B75A28C1
  6A2084C5 31BFF5C1 CD07916D 83FD56EF 9023C974 A9835AAF AC1AAF93 C0FB6856
  5CA7B10A AF9EFCE1 5DE8651F D30847FF 024D6EF3 3AADB77D 68519BA9 F21AC1FE
  5A50CA58 A00CDBB5 25C693E8 4D8C639D 6E5A3935 2F050F4D A3A7B2AD 47942BDD
  4D78EFEE 81FDAFE0 F26220A6 6AF1D505 C601A2B3 56B2D2FE 5DD60B95 7B149AC6
```

```
  EB0CACE9 CA5D42CC 4B0CA1DE 2895251A 4C1AFBC0 4FD54872 50BC69B2 445DF62E
  CA556655 9BEB0203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF
  301F0603 551D2304 18301680 140623A5 BE29331A B5FF4081 20569978 FB29842F
  58301D06 03551D0E 04160414 0623A5BE 29331AB5 FF408120 569978FB 29842F58
  300D0609 2A864886 F70D0101 05050003 82010100 9A8F4A49 4CC83788 4EC24211
  CE24EE7A B8552513 F9F34632 04B8119D 612FFA57 370471EF 123E1385 EBC74CBD
  92DD9795 086536A0 F2469390 219B288F 3D9EC787 48A4EE78 5A492BA1 1680D1C9
  F3A8A820 1D065DEB E8F0D00E 37A2A866 F759FB2D E30CD9B8 8900E25E 8171A288
  FB2BB185 B6A6ED29 3BAA4495 31FCC789 0305E830 6EBA491E 211F0B7C FE808066
  4E78E657 D16239B2 6E40B8A0 41631417 40EAB264 D31C2A10 24FEFD2B F0C5A1D9
  693C7384 D1C54B99 BAC2A7AA 5B646CED 6E31FEC1 EB64C663 9F703970 BFA72795
  06252993 E38182F3 7F760357 37556092 A5FE18F4 4FCC6BA3 716886FA 76106709
  8D4EF4C5 14A81EB9 F0A29EB9 DB41CAB7 F98A5D15
        quit
!
voice service voip
 ip address trusted list
  ipv4 217.0.0.0 255.255.0.0
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 2
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711alaw
 sip
  bind control source-interface GigabitEthernet0/0/0
  bind media source-interface GigabitEthernet0/0/0
  session refresh
  asserted-id pai
  outbound-proxy dns:reg.sip-trunk.telekom.de
```

```
  conn-reuse
  privacy-policy passthru
  sip-profiles inbound
  sip-profiles 3000
  audio forced
!
voice class uri 101 sip
 host ipv4:10.80.18.3
voice class codec 1
 codec preference 1 g711alaw
 codec preference 2 g722-64
!
voice class sip-profiles 3000
 rule 1 request REGISTER sip-header Contact modify "<.*:.*@(.*)>"
"<sip:\1;bnc>"
 rule 2 request REGISTER sip-header Proxy-Require add "Proxy-Require: gin"
 rule 3 request REGISTER sip-header Require add "Require: gin"
!
voice class sip-profiles 201
 rule 1 request ANY sip-header P-Asserted-Identity modify "<sip:(.*)>"
"<sip:+4922843353290@sip-trunk.telekom.de>”
 rule 2 request ANY sip-header Min-SE remove
 rule 3 request ANY sip-header Diversion remove
 rule 4 request ANY sdp-header Connection-Info remove
 rule 5 response ANY sdp-header Connection-Info remove
!
voice class e164-pattern-map 201
  e164 11[68]T
  e164 11[025]
  e164 +T
```

```
  e164 OT
 !
!
voice class server-group 1
 ipv4 10.80.18.3
 description **CUCM Server Group**
!
voice class sip-options-keepalive 101
 up-interval 30
 retry 3
 transport tcp
!
voice translation-rule 1004
 rule 1 /^8/ /+/
!
voice translation-profile DT
translate called 1004
!
license udi pid ISR4321/K9 sn FDO55550MQ8
license boot level appxk9
license boot level uck9
!
no diagnostic bootup level
spanning-tree extend system-id
!
username cisco privilege 15 password 7 ********
!
redundancy
 mode none
```

```
 application redundancy
  group 2
   name voice-b2bha
   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
!
track 1 interface GigabitEthernet0/0/0 line-protocol
track 2 interface GigabitEthernet0/0/1 line-protocol
!
class-map match-any Realtime
 match ip dscp cs6
 match ip dscp ef
!
policy-map child
 class Realtime
  priority 5000
 class class-default
  random-detect
policy-map parent
 class class-default
  shape average percent 100
   service-policy child
!
interface GigabitEthernet0/0/0
 description WAN Interface
```

```
 ip address 192.XX.XX.XX 255.255.255.128
 media-type rj45
 negotiation auto
 redundancy rii 11
 redundancy group 2 ip 192.XX.XX.XX exclusive
 service-policy output parent
!
interface GigabitEthernet0/0/1
 description LAN Interface
 ip address 10.80.18.11 255.255.255.0
 negotiation auto
 redundancy rii 12
 redundancy group 2 ip 10.80.18.10 exclusive
!
interface GigabitEthernet0/1/0
 description CUBE HA
 ip address 10.89.20.7 255.255.255.0
 negotiation auto
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 negotiation auto
!
ip forward-protocol nd
no ip http server
no ip http secure-server
ip tftp source-interface GigabitEthernet0
ip dns server
```

```
ip route 0.0.0.0 0.0.0.0 192.XX.XX.XX
ip route 10.64.0.0 255.255.0.0 10.80.18.1
ip route 10.64.0.0 255.255.0.0 10.64.1.1
ip route 10.80.0.0 255.255.0.0 10.80.18.1
ip route 172.16.0.0 255.255.0.0 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 **SIP-TRUNK.TELEKOM.DE**
 session protocol sipv2
 session target sip-server
 session transport tcp
 destination e164-pattern-map 201
 incoming called-number .T
 voice-class codec 1
 voice-class sip outbound-proxy dns:reg.sip-trunk.telekom.de
 voice-class sip profiles 201
 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 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711alaw
 ip qos dscp cs6 signaling
 clid strip name
 no vad
!
dial-peer voice 101 voip
 description **CUCM/PBX **
 destination-pattern +492284335329T
 session protocol sipv2
 session transport tcp
 session server-group 1
 incoming uri via 101
 voice-class codec 1
 no voice-class sip outbound-proxy
 voice-class sip options-keepalive profile 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 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711alaw
 no vad
!
dial-peer voice 901 voip
```

```
 description **LOOPBACK DIAL-PEER**[1]
 translation-profile outgoing DT
 destination-pattern 849T
 session protocol sipv2
 session target sip-server
 session transport tcp
 voice-class codec 1
 voice-class sip outbound-proxy dns:reg.sip-trunk.telekom.de
 voice-class sip profiles 201
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 ip qos dscp cs6 signaling
 clid strip name
 no vad
!
!
sip-ua
 credentials number +4922843353290 username 55************** password 7 ********************* realm sip-trunk.telekom.de
 authentication username 55************** password 7 ********************* realm sip-trunk.telekom.de
 no remote-party-id
 timers expires 900000
 timers register 100
 timers dns registrar-cache ttl
 registrar dns:sip-trunk.telekom.de expires 240 tcp auth-realm sip-trunk.telekom.de
 sip-server dns:sip-trunk.telekom.de
```

[1] This dial-peer is explicitly used for routing IP-PBX to IP-PBX calls out to the SP Network and back

```
 no transport udp
 connection-reuse
!
!
line con 0
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 exec-timeout 0 0
 password 7 ********
 login
line vty 5
 exec-timeout 0 0
 password 7 ********
 login
!
ntp server 0.de.pool.ntp.org
!
end
```

Standby Cisco UBE:

```
Current configuration : 9066 bytes
!
! Last configuration change at 06:27:52 UTC Wed Apr 26 2017
! NVRAM config last updated at 06:27:43 UTC Wed Apr 26 2017
!
version 16.3
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
no platform punt-keepalive disable-kernel-core
!
hostname CUBE2
!
boot-start-marker
boot system bootflash:isr4300-universalk9.16.03.03.SPA.bin
boot-end-marker
!
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
no logging queue-limit
no logging buffered
```

```
no logging rate-limit
enable secret 5 ********
!
no aaa new-model
!
ip name-server 8.8.8.8
!
subscriber templating
multilink bundle-name authenticated
!
crypto pki trustpoint TP-self-signed-2548443246
 enrollment selfsigned
 subject-name cn=IOS-Self-Signed-Certificate-2548443246
 revocation-check none
 rsakeypair TP-self-signed-2548443246
!
!
crypto pki certificate chain TP-self-signed-2548443246
 certificate self-signed 01
  30820330 30820218 A0030201 02020101 300D0609 2A864886 F70D0101 05050030
  31312F30 2D060355 04031326 494F532D 53656C66 2D536967 6E65642D 43657274
  69666963 6174652D 32353438 34343332 3436301E 170D3137 30323130 31373132
  33365A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649
  4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D32 35343834
  34333234 36308201 22300D06 092A8648 86F70D01 01010500 0382010F 00308201
  0A028201 0100B73A 8AE876C0 62A381D9 4C331F21 6FBF60E9 20F9420A 6F2C3A5A
  2DA4B74B 1A9B55DB 65BC3A4B 016D4E96 3CB638A7 31C61AA1 A2E8EF3E FE7733F5
  A0035F13 9AE153CE D55D4F64 FBBCA3CE EC8D110A 6490B2CD 44509DEE 14A60E75
  66CF37C5 3DF0BBBE 7B27306D C2ACDBA2 A3497E3D 7EFDCC2B 1902A0A8 038AD01E
```

```
  68FD339C B2620BFF 00E703AB 88DD7796 B7C0351F 27BFF1EC 791ECF53 87B57E81
  166B26BA 1428E6F7 A7484680 ACF2B8F7 BB95977B C3854F78 D4295377 CE568896
  451A72F7 5D117423 6A69CEEE 8E13BDAB 96B61B29 1165C7C4 E7DB2BCB 1A2095F6
  E9C80DDD 9B1DCED7 CE87C3F0 BF726628 6AF272BA 9A3D33B0 4AD1444F 87DF933F
  9BCCF78D 3C6B0203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF
  301F0603 551D2304 18301680 14EADBB3 9378001B 592B6EC4 0CD4C83F 3FA14FA3
  AD301D06 03551D0E 04160414 EADBB393 78001B59 2B6EC40C D4C83F3F A14FA3AD
  300D0609 2A864886 F70D0101 05050003 82010100 A99C7D6F 325E52A8 9F0221FE
  3BC2460A 5383DF63 B1A66575 D8CE62F8 725B14FA F18879B1 38173AF8 3A05D05F
  B72318A1 23B11F12 E14DE814 CF93938D 41F75435 21999BC2 6FFAFBAC 84347F5F
  0B4787B0 2A54C190 8E4A7505 3957998E 8A58C6A1 D6BBD261 11C24F7F 61FA6931
  3A311AC7 E7D50544 7A6DA790 09A5E366 4635FE2F C5824130 AB3FDA56 AE0BEA53
  E674A90C C13EF3B9 2C0D7F9B 3ED7CD5A 3FABB4D6 A0B9E76A 180E8DC1 E1E8024F
  E431C813 64B5F09E A3BB9F8D 669E33F5 2BFB2295 34FDDE83 1290F246 61BA7FEB
  943F10C5 29499BFE 173CE107 6D938477 5A4DB917 B88D2CC5 6A2F77B1 3E605311
  B4A19B46 1A05F455 3E2816E0 59D13336 027F827D
        quit
!
voice service voip
 ip address trusted list
  ipv4 217.0.0.0 255.255.0.0
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 redundancy-group 2
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-
through g711alaw
 sip
  bind control source-interface GigabitEthernet0/0/0
  bind media source-interface GigabitEthernet0/0/0
```

```
  session refresh
  asserted-id pai
  outbound-proxy dns:reg.sip-trunk.telekom.de
  conn-reuse
  privacy-policy passthru
  sip-profiles inbound
  sip-profiles 3000
  audio forced
!
!
voice class uri 101 sip
 host ipv4:10.80.18.3
voice class codec 1
 codec preference 1 g711alaw
 codec preference 2 g722-64
!
voice class sip-profiles 3000
 rule 1 request REGISTER sip-header Contact modify "<.*:.*@(.*)>"
"<sip:\1;bnc>"
 rule 2 request REGISTER sip-header Proxy-Require add "Proxy-Require: gin"
 rule 3 request REGISTER sip-header Require add "Require: gin"
!
voice class sip-profiles 201
 rule 1 request ANY sip-header P-Asserted-Identity modify "<sip:(.*)>"
"<sip:+4922843353290@sip-trunk.telekom.de>"
 rule 2 request ANY sip-header Min-SE remove
 rule 3 request ANY sip-header Diversion remove
 rule 4 request ANY sdp-header Connection-Info remove
 rule 5 response ANY sdp-header Connection-Info remove
!
```

```
!
voice class e164-pattern-map 201
  e164 11[68]T
  e164 11[025]
  e164 +T
  e164 0T
 !
!
voice class server-group 1
 ipv4 10.80.18.3
 description **CUCM Server Group**
!
voice class sip-options-keepalive 101
 up-interval 30
 retry 3
 transport tcp
!
voice translation-rule 1004
 rule 1 /^8/ /+/
!
voice translation-profile DT
translate called 1004
!
license udi pid ISR4321/K9 sn FDO19220MW3
license boot level appxk9
license boot level uck9
!
no diagnostic bootup level
spanning-tree extend system-id
```

```
!
!
username cisco privilege 15 password 7 ********
!
redundancy
 mode none
 application redundancy
  group 2
   name voice-b2bha
   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
!
track 1 interface GigabitEthernet0/0/0 line-protocol
track 2 interface GigabitEthernet0/0/1 line-protocol
!
class-map match-any Realtime
 match ip dscp cs6
 match ip dscp ef
!
policy-map child
 class Realtime
  priority 5000
 class class-default
  random-detect
policy-map parent
```

```
 class class-default
  shape average percent 100
   service-policy child
!
interface GigabitEthernet0/0/0
 description WAN Interface
 ip address 192.XX.XX.XX 255.255.255.128
 media-type rj45
 negotiation auto
 redundancy rii 11
 redundancy group 2 ip 192.XX.XX.XX exclusive
 service-policy output parent
!
interface GigabitEthernet0/0/1
 description LAN Interface
 ip address 10.80.18.12 255.255.255.0
 negotiation auto
 redundancy rii 12
 redundancy group 2 ip 10.80.18.10 exclusive
!
interface GigabitEthernet0/1/0
 description CUBE HA
 ip address 10.89.20.8 255.255.255.0
 negotiation auto
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 negotiation auto
```

```
!
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 10.64.1.1
ip route 0.0.0.0 0.0.0.0 192.XX.XX.XX
ip route 10.64.0.0 255.255.0.0 10.80.18.1
ip route 10.80.0.0 255.255.0.0 10.80.18.1
ip route 172.16.0.0 255.255.0.0 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 **SIP-TRUNK.TELEKOM.DE**
 session protocol sipv2
 session target sip-server
 session transport tcp
 destination e164-pattern-map 201
 incoming called-number .T
 voice-class codec 1
 voice-class sip outbound-proxy dns:reg.sip-trunk.telekom.de
```

```
 voice-class sip profiles 201
 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 14400
 fax nsf 000000
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-
through g711alaw
 ip qos dscp cs6 signaling
 clid strip name
 no vad
!
dial-peer voice 101 voip
 description **CUCM/PBX **
 destination-pattern +492284335329T
 session protocol sipv2
 session transport tcp
 session server-group 1
 incoming uri via 101
 voice-class codec 1
 no voice-class sip outbound-proxy
 voice-class sip options-keepalive profile 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 14400
 fax nsf 000000
```

```
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711alaw
 no vad
!
dial-peer voice 901 voip
 description **LOOPBACK DIAL-PEER**[2]
 translation-profile outgoing DT
 destination-pattern 849T
 session protocol sipv2
 session target sip-server
 session transport tcp
 voice-class codec 1
 voice-class sip outbound-proxy dns:reg.sip-trunk.telekom.de
 voice-class sip profiles 201
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 voice-class sip bind media source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 ip qos dscp cs6 signaling
 clid strip name
 no vad
!
!
sip-ua
 credentials number +4922843353290 username 55************** password 7 ********************* realm sip-trunk.telekom.de
 authentication username 55************** password 7 ********************* realm sip-trunk.telekom.de
 no remote-party-id
 timers expires 900000
```

[2] This dial-peer is explicitly used for routing IP-PBX to IP-PBX calls out to the SP Network and back

```
 timers register 100
 timers dns registrar-cache ttl
 registrar dns:sip-trunk.telekom.de expires 240 tcp auth-realm sip-trunk.telekom.de
 sip-server dns:sip-trunk.telekom.de
 no transport udp
 connection-reuse
!
!
line con 0
 stopbits 1
line aux 0
 stopbits 1
line vty 0 5
 exec-timeout 0 0
 password 7 ********
 login
!
ntp server 0.de.pool.ntp.org
!
!
!
!
!
end
```

Configuring Cisco UCM 11.5 Cluster

Cisco UCM Version

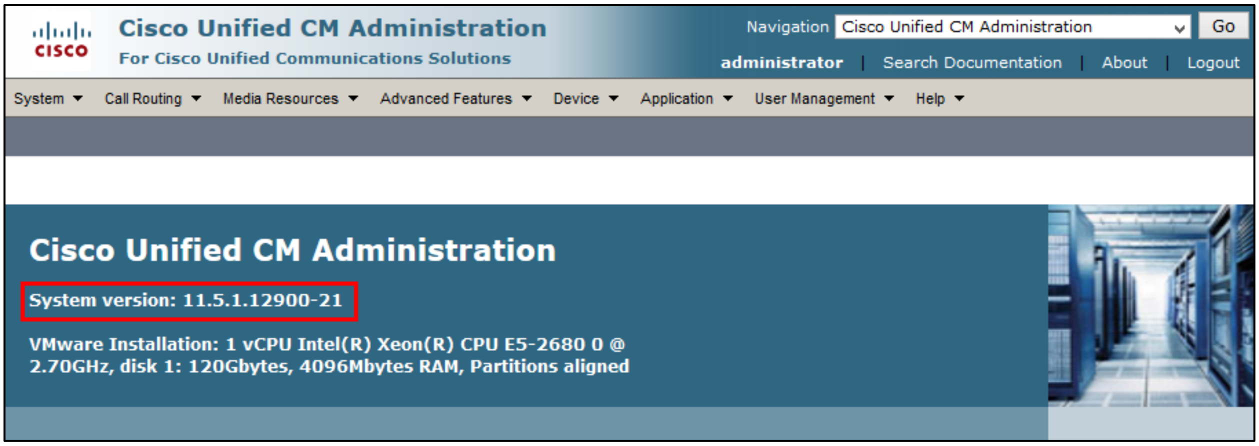

Figure 3: Cisco UCM Version

Cisco Call Manager Service Parameters

Navigation: System → Service Parameters

- Select Server* = Clus28Sub1--CUCM Voice/Video (Active)
- Select Service*= Cisco CallManager (Active)
- Duplex Streaming Enabled* = True
- All other fields are set to default values

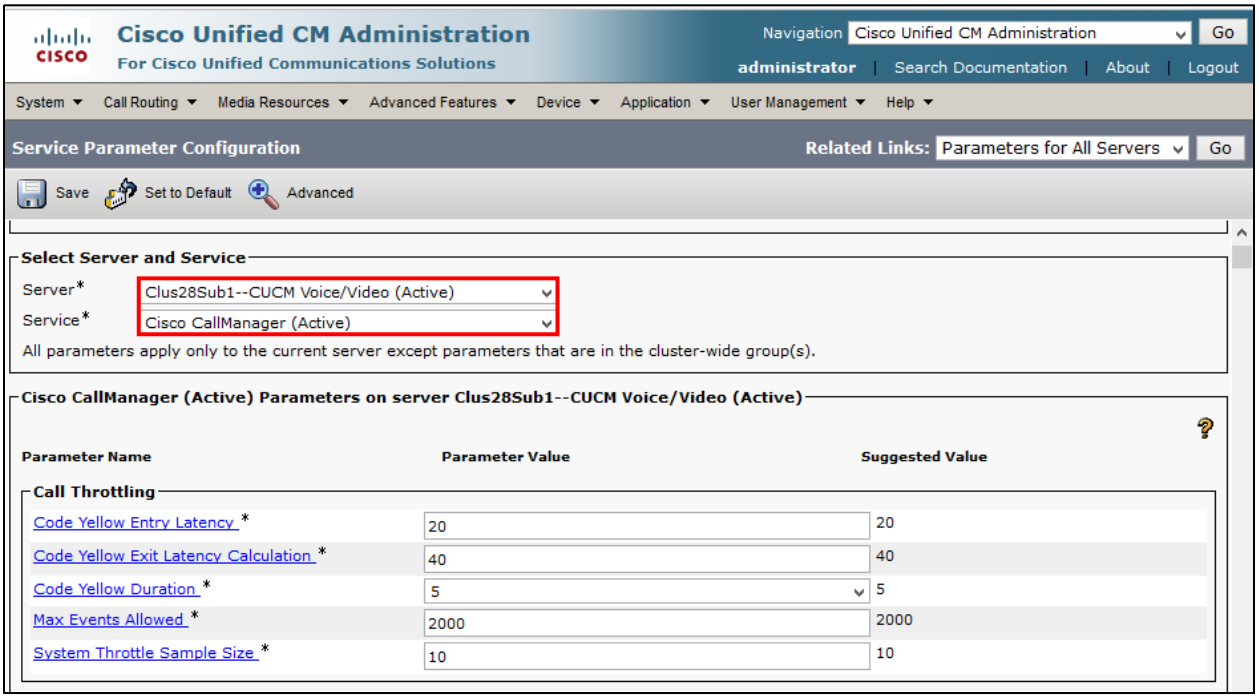

Figure 4: Service Parameters

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 *	True	False
Silence Suppression for Gateways *	True	False
Strip G.729 Annex B (Silence Suppression) from Capabilities *	True	False
Enable Source IP Address Verification for Software Media Devices *	True	True

Figure 5: Service Parameters (Cont.)

Off-net Calls via Deutsche Telekom SIP Trunk

SIP Trunk Security Profile

Navigation: System → Security → SIP Trunk Security Profile

- Name*= **DT Non Secure SIP Trunk Profile** is used as an example
- Description = **Non Secure SIP Trunk Profile authenticated by null String** is used as an example
- Device Security Mode = **Non Secure**
- Incoming Transport Type* = **TCP + UDP**
- Outgoing Transport Type = **TCP**

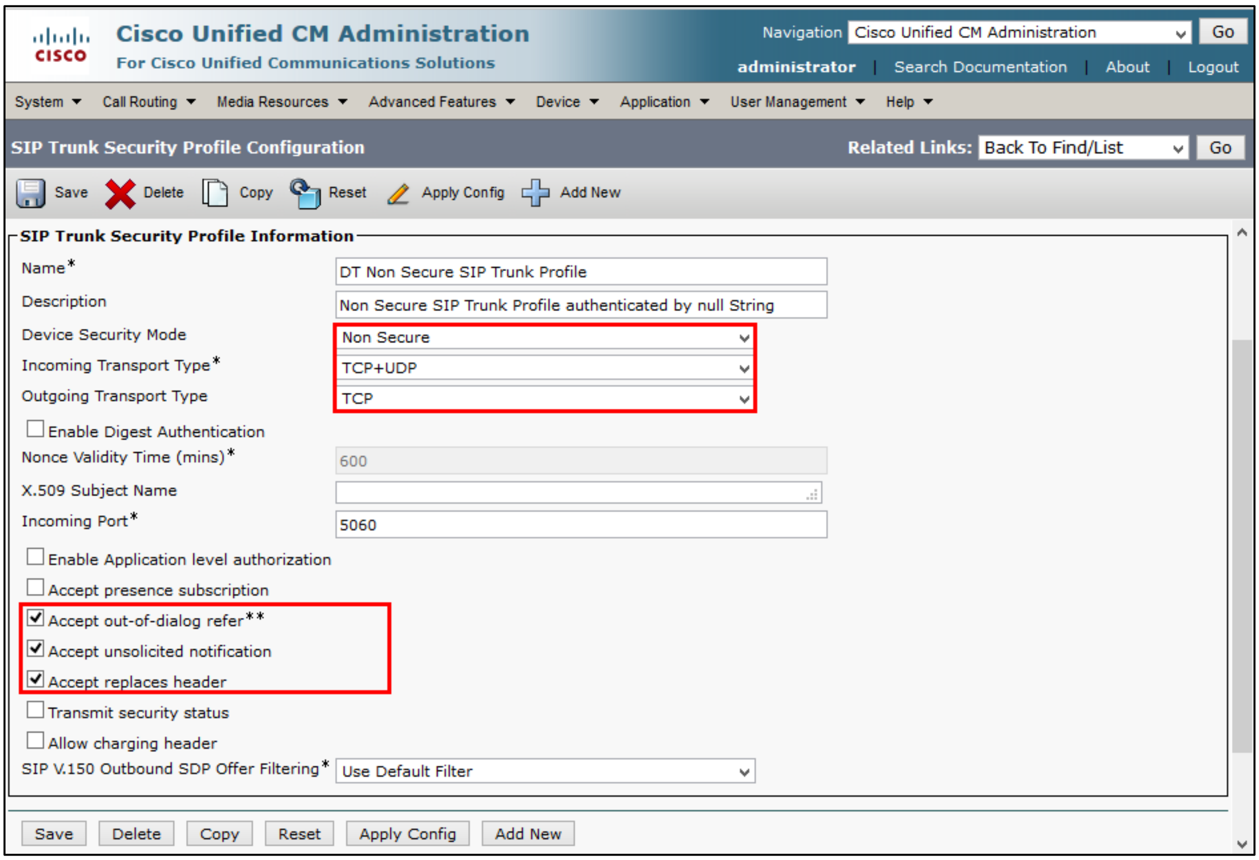

Figure 6: SIP Trunk Security Profile

SIP Profile

Navigation: Device → Device Settings → SIP Profile

- Name*= **Deutsche Telekom Standard SIP Profile** is used as an example
- Description = **Default SIP Profile** is used as an example

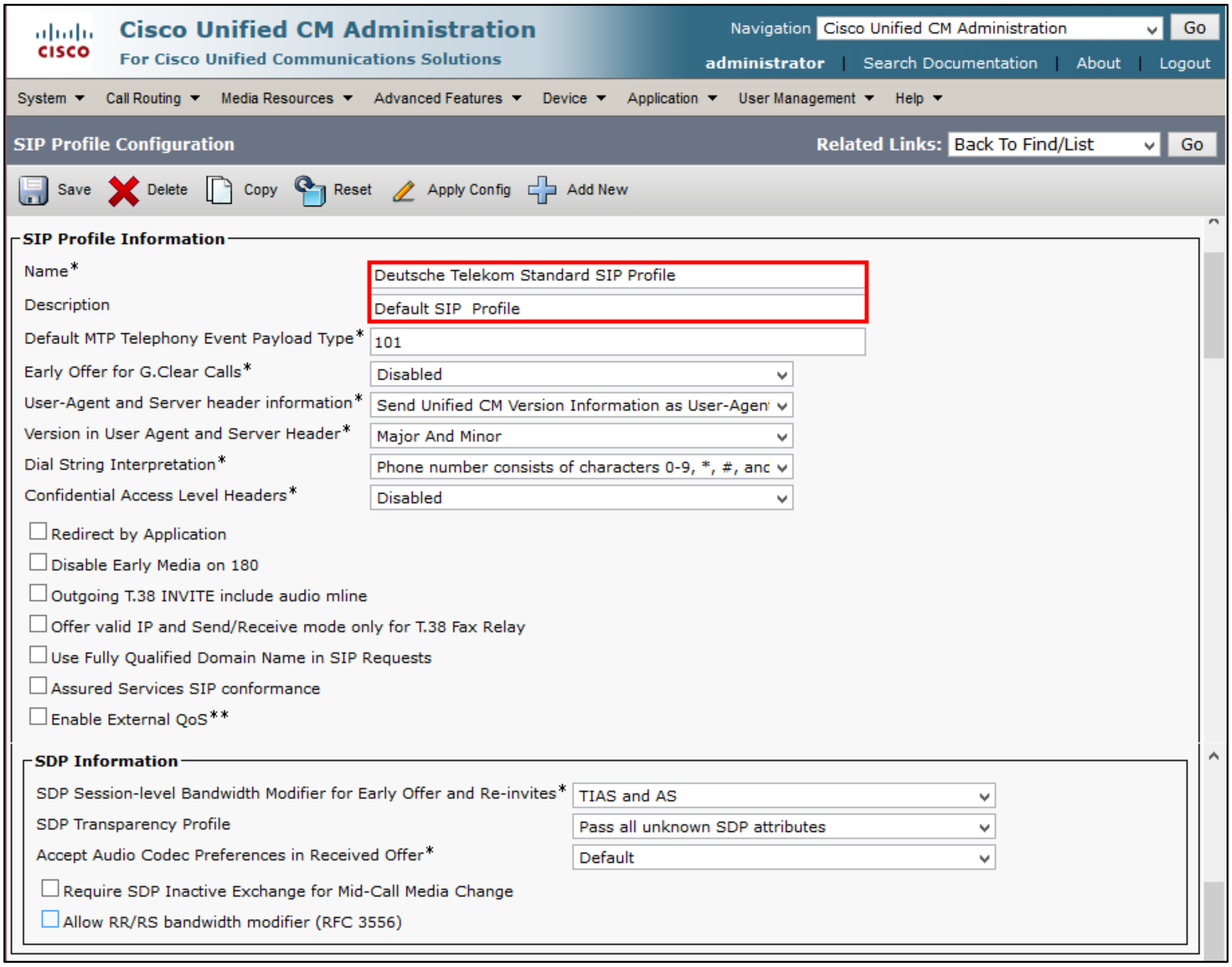

Figure 7: SIP Profile

Parameters used in Phone

Parameter	Value
Timer Invite Expires (seconds)*	180
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10
Media Port Ranges	(●) Common Port Range for Audio and Video ( ) Separate Port Ranges for Audio and Video
Start Media Port*	16384
Stop Media Port*	32766
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

Figure 8: SIP Profile (Cont.)

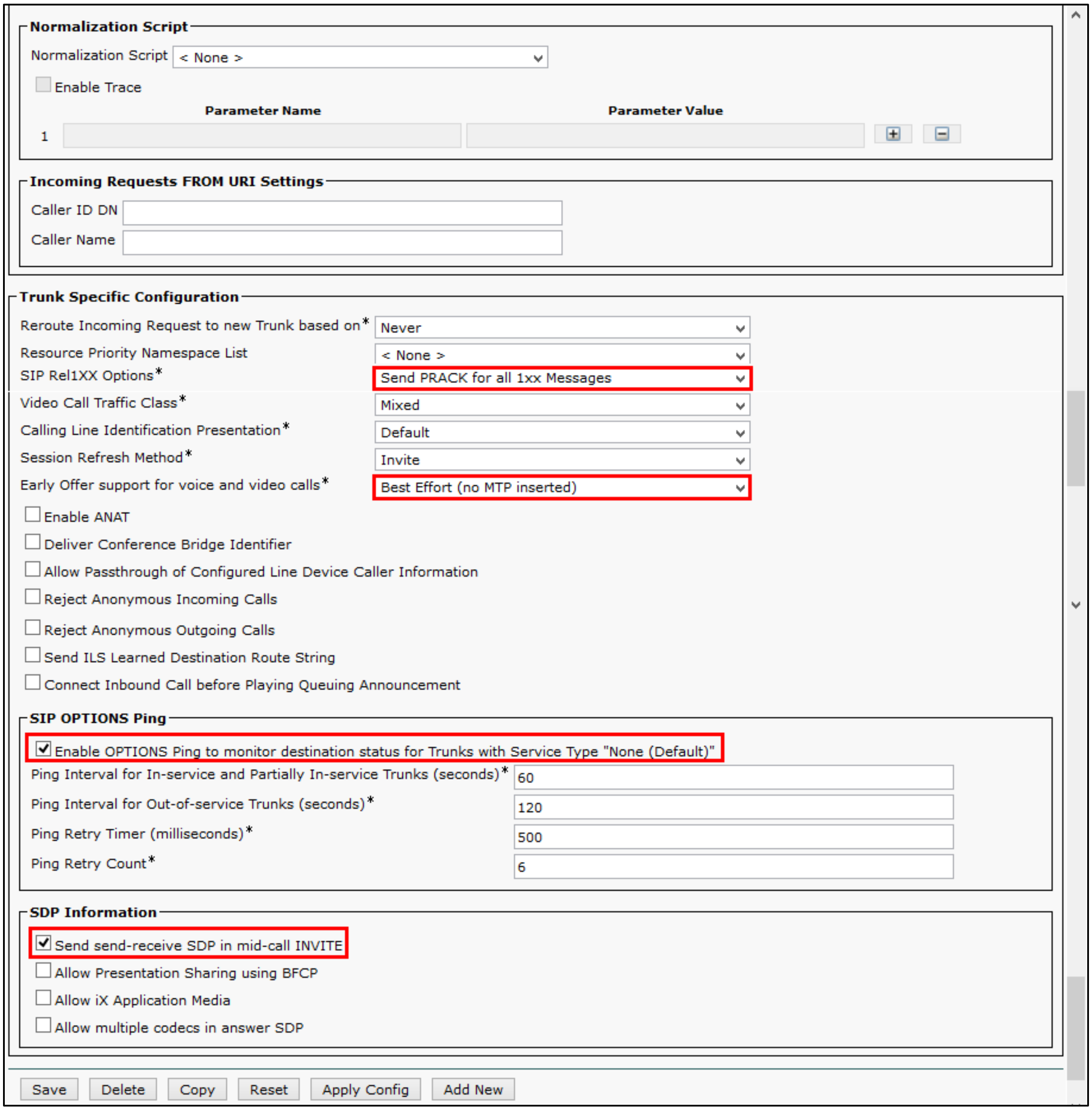

Figure 9: SIP Profile (Cont.)

Trunk configuration

Trunk configuration from Cisco UCM to the LAN side of the Cisco UBE:

Navigation: Device → Trunk → Add New

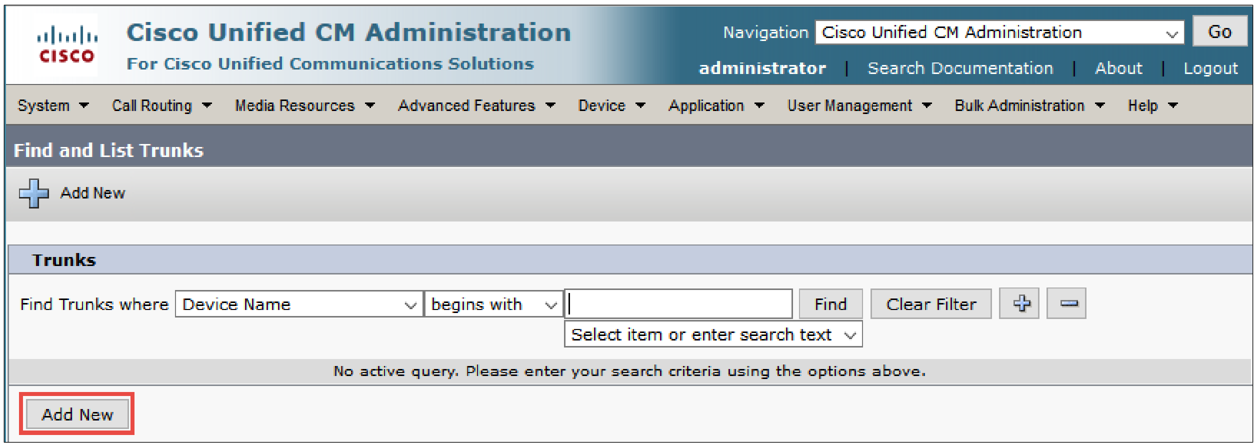

Figure 10: Add New Trunk to Cisco UBE

- Select 'Trunk Type' as SIP Trunk and 'Device Protocol' as SIP and select 'Next' as shown below.

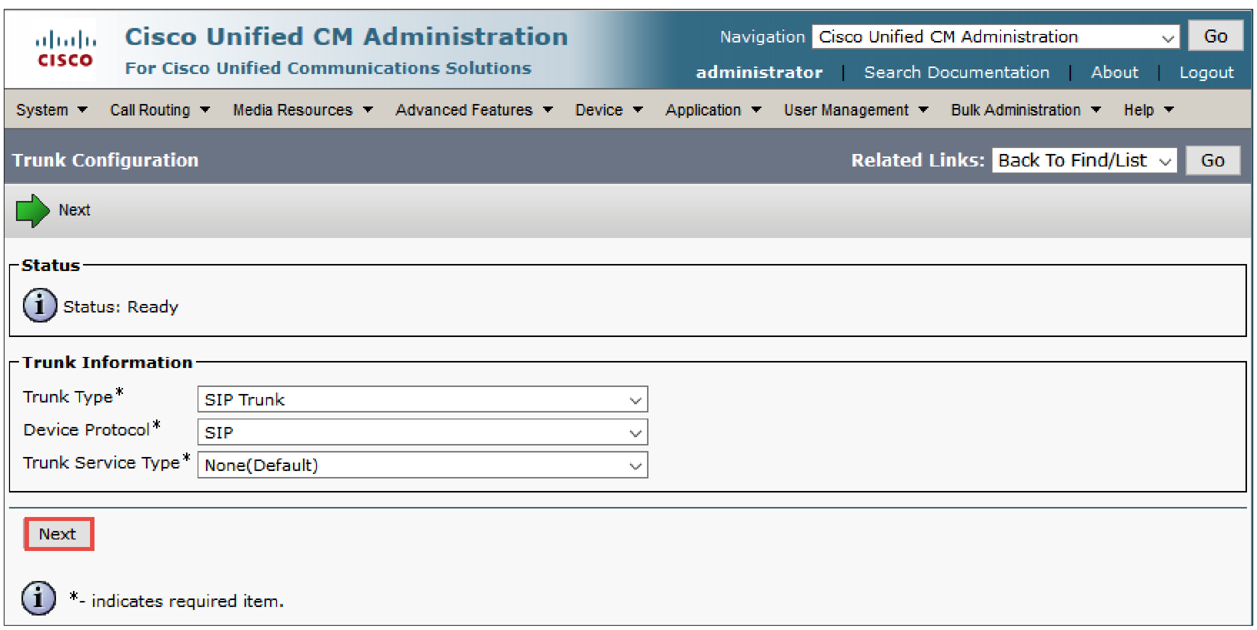

Figure 11: Add SIP Trunk Type

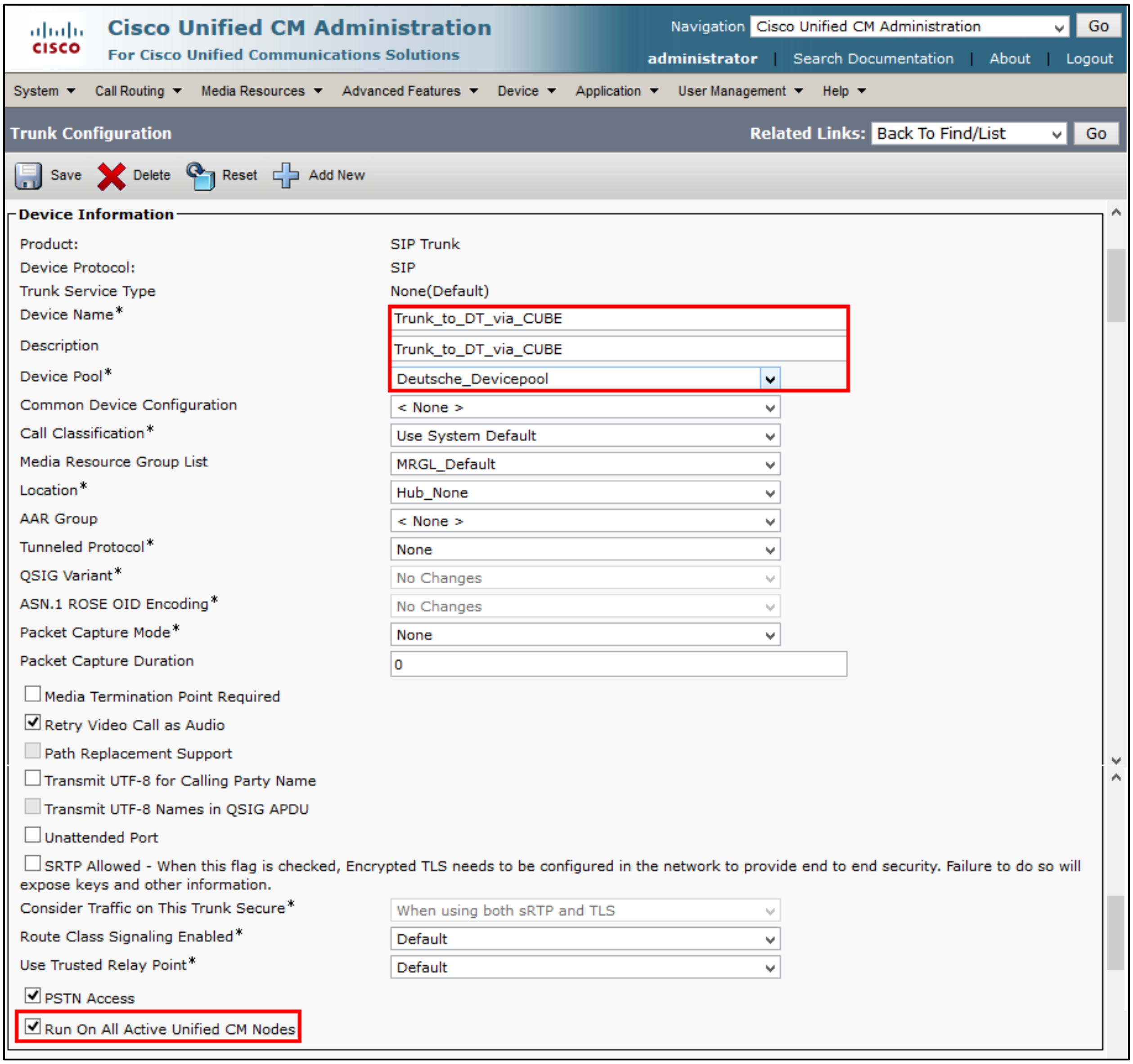

Figure 12: SIP Trunk to Cisco UBE

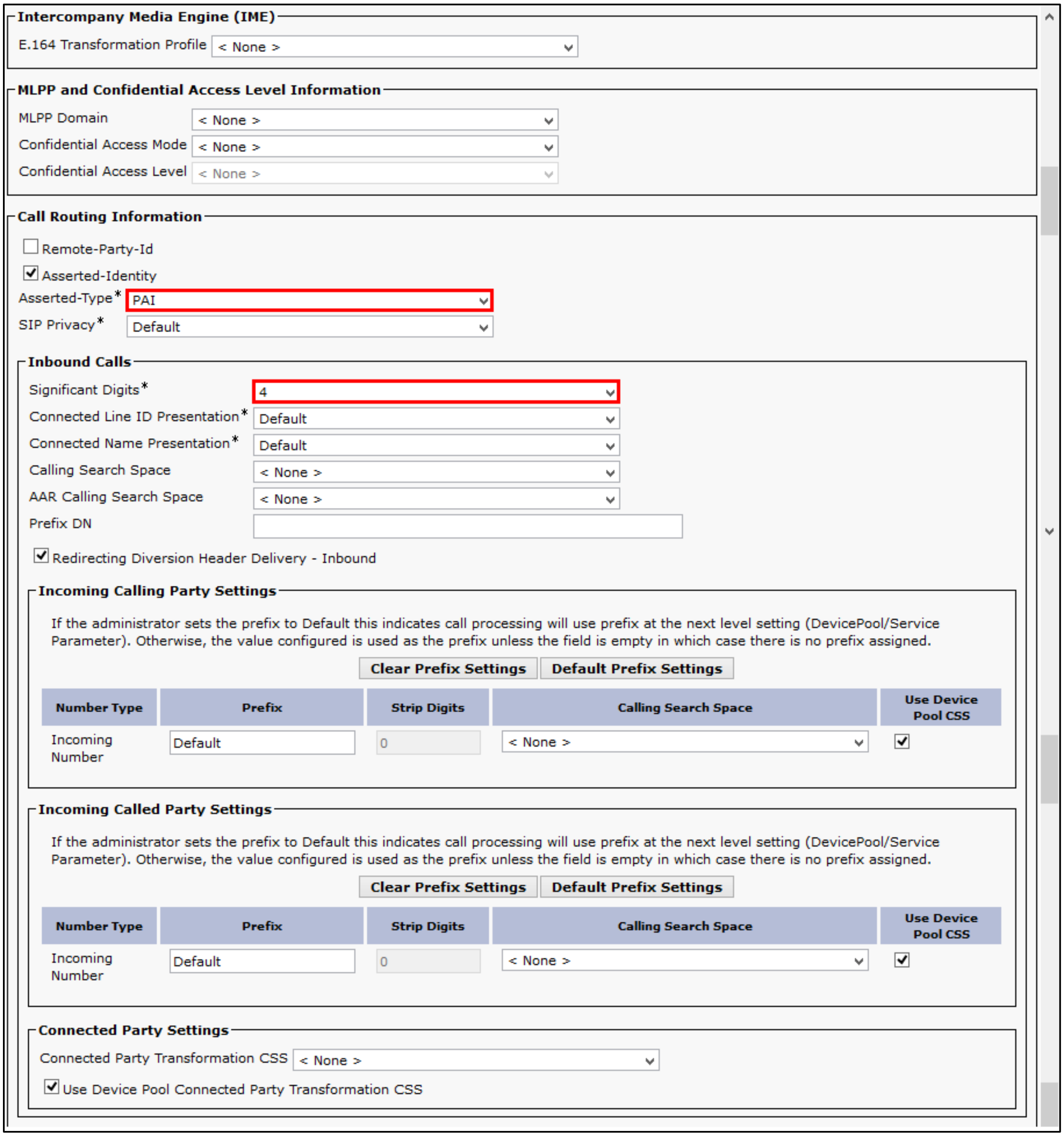

Figure 13: SIP Trunk to Cisco UBE (Cont.)

- Configure the Virtual LAN IP address of the Cisco UBE and the Destination Port
- Configure the SIP Trunk Security Profile and SIP Profile as shown below

- The rest of the configuration is all default
- Click Save and Reset after completion

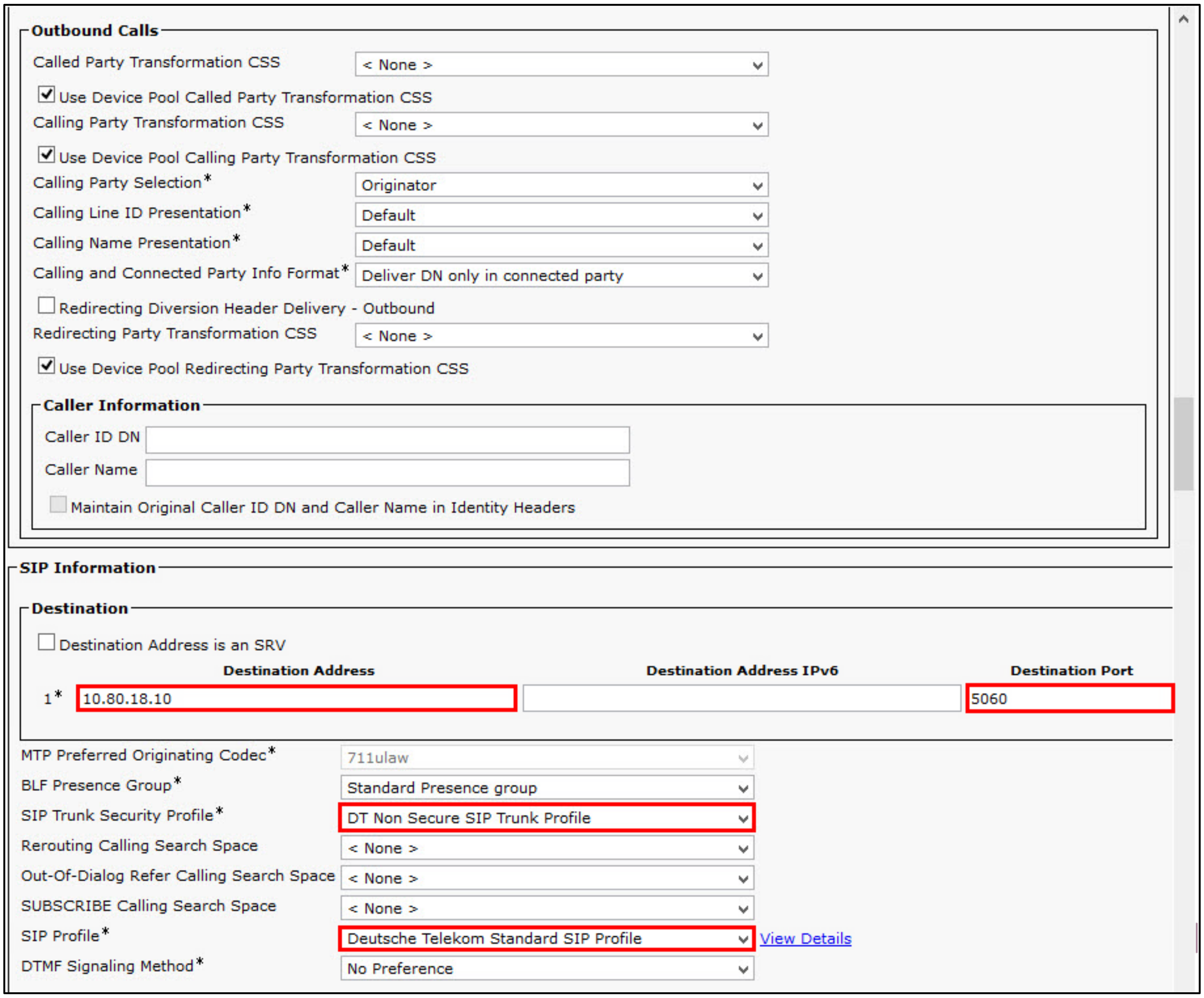

Figure 14: SIP Trunk to Cisco UBE (Cont.)

Trunk configuration from Cisco UCM to Fax Gateway:

Navigation: Devices → Trunk → Add New

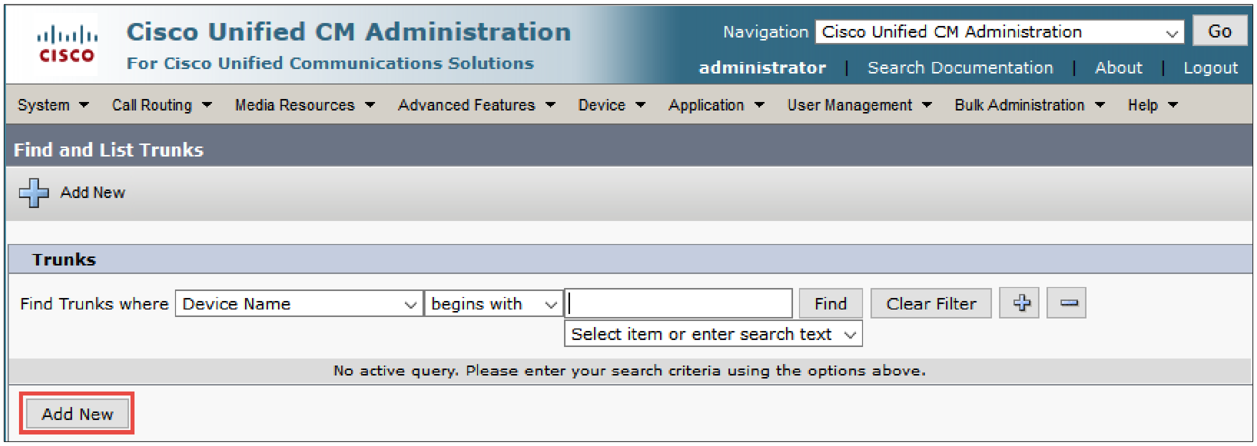

Figure 15: Add New Trunk to Fax Gateway

- Select 'Trunk Type' as SIP Trunk and 'Device Protocol' as SIP and select 'Next' as shown below.

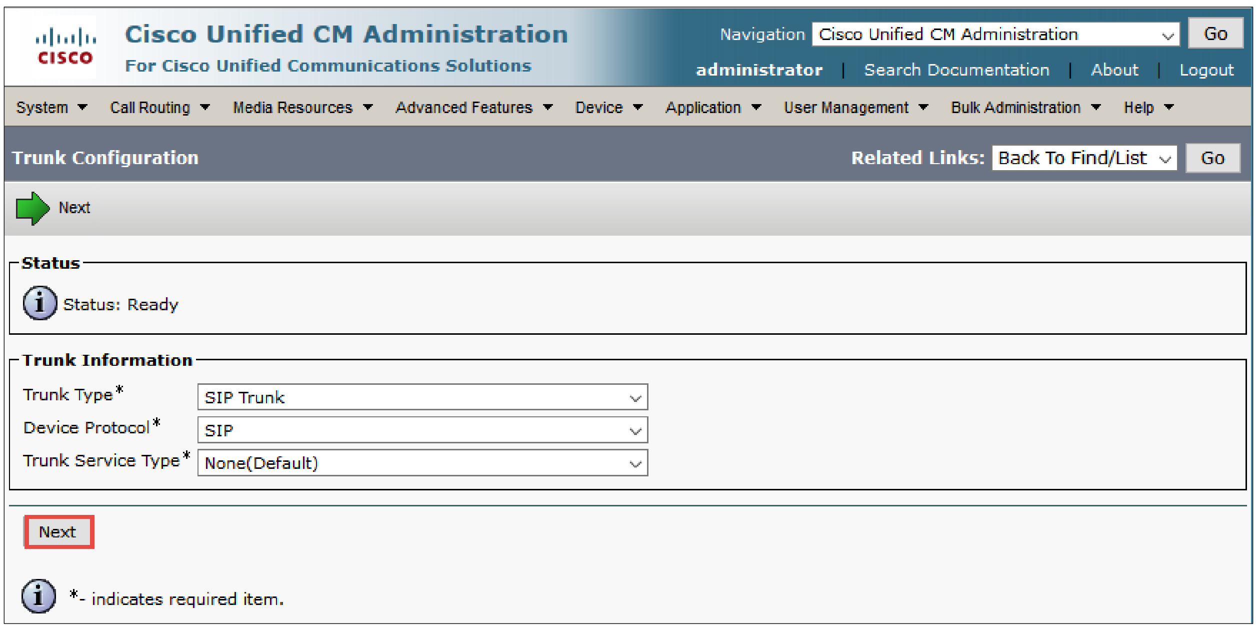

Figure 16: Add SIP Trunk Type

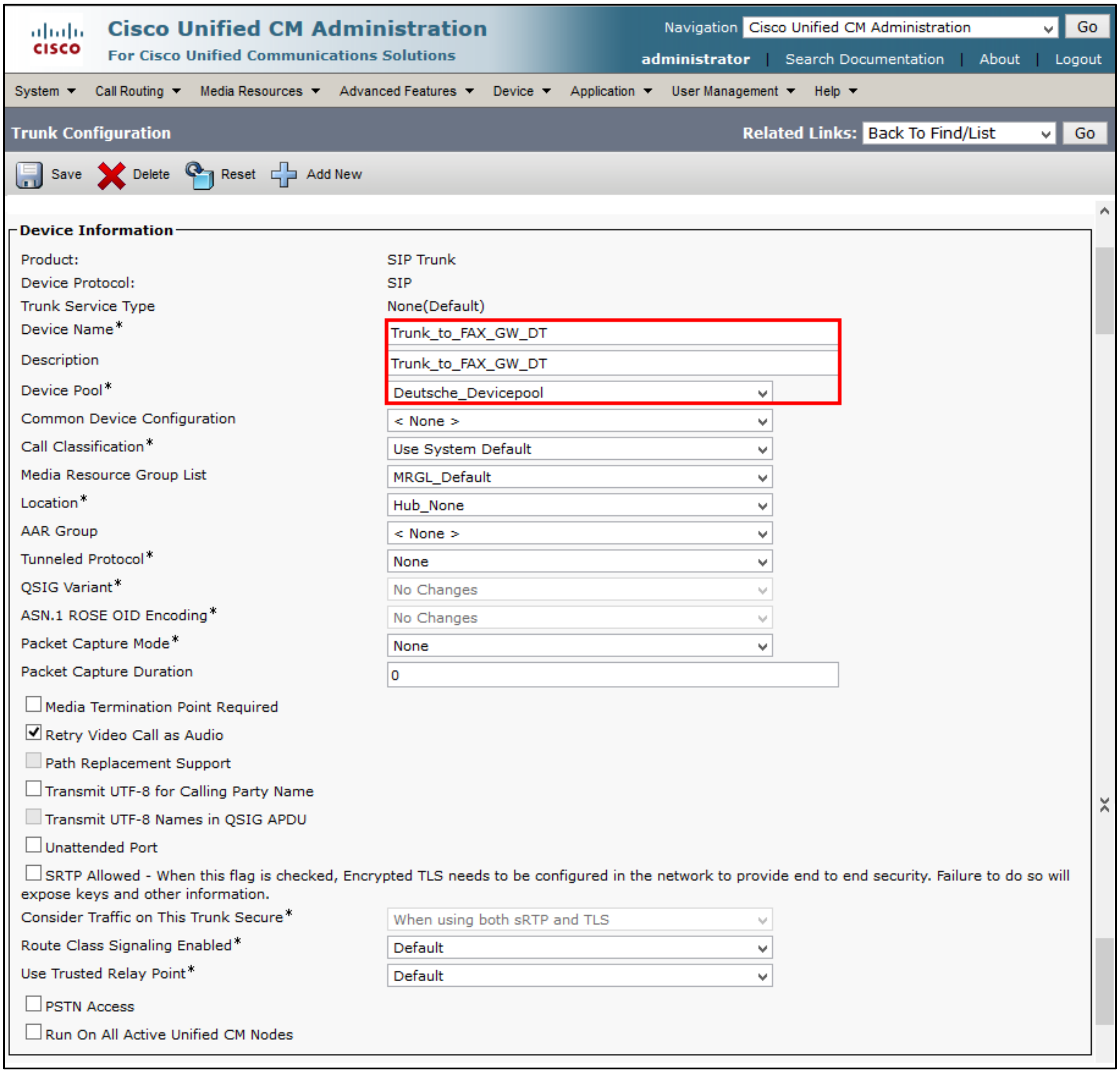

Figure 17: SIP Trunk to FAX Gateway

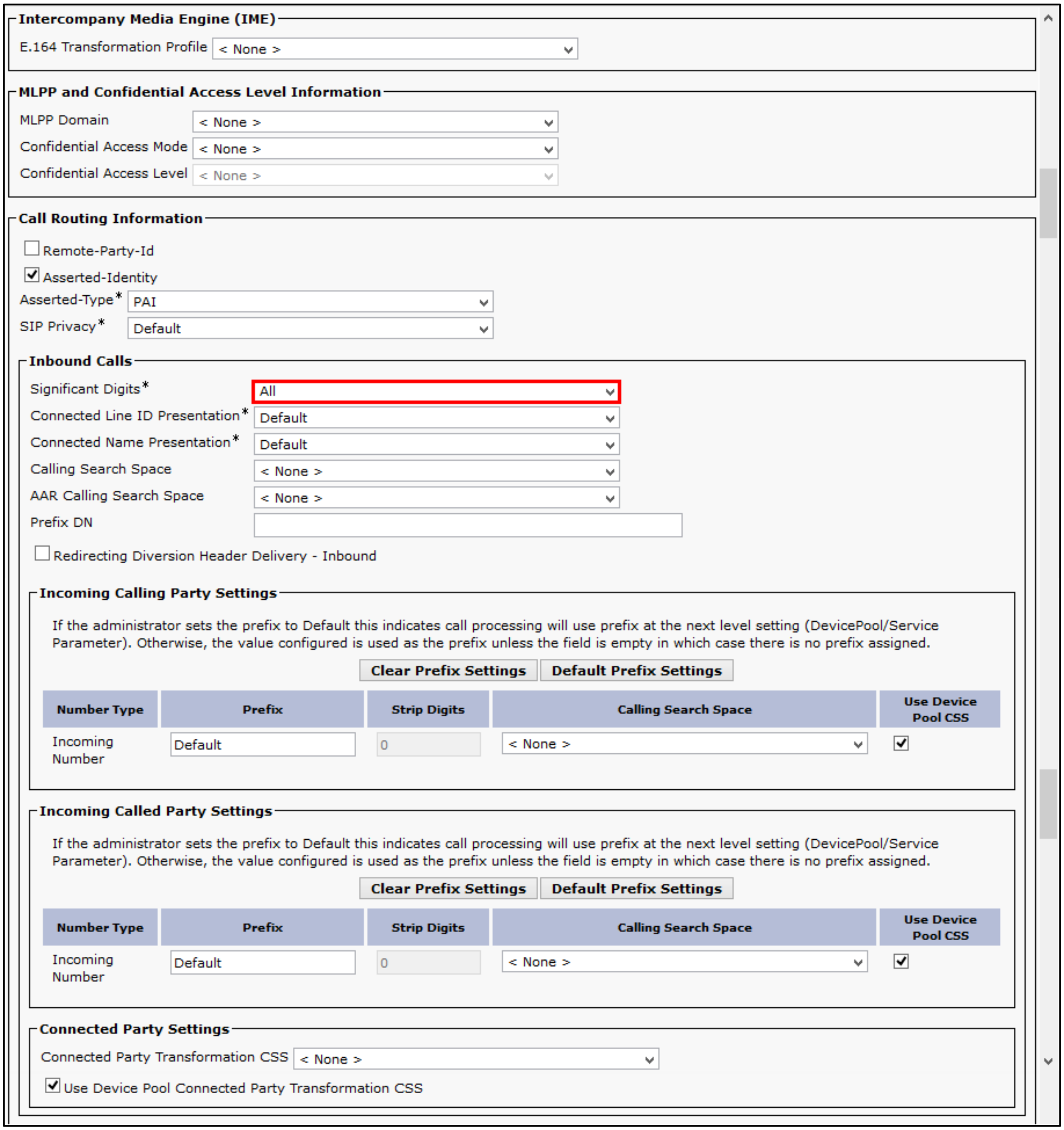

Figure 18: SIP Trunk to FAX Gateway (Cont.)

- Configure the IP address of Fax Gateway and the Destination Port
- Configure the SIP Trunk Security Profile and SIP Profile as shown below
- The rest of the configuration is all default

- Click Save and Reset after completion

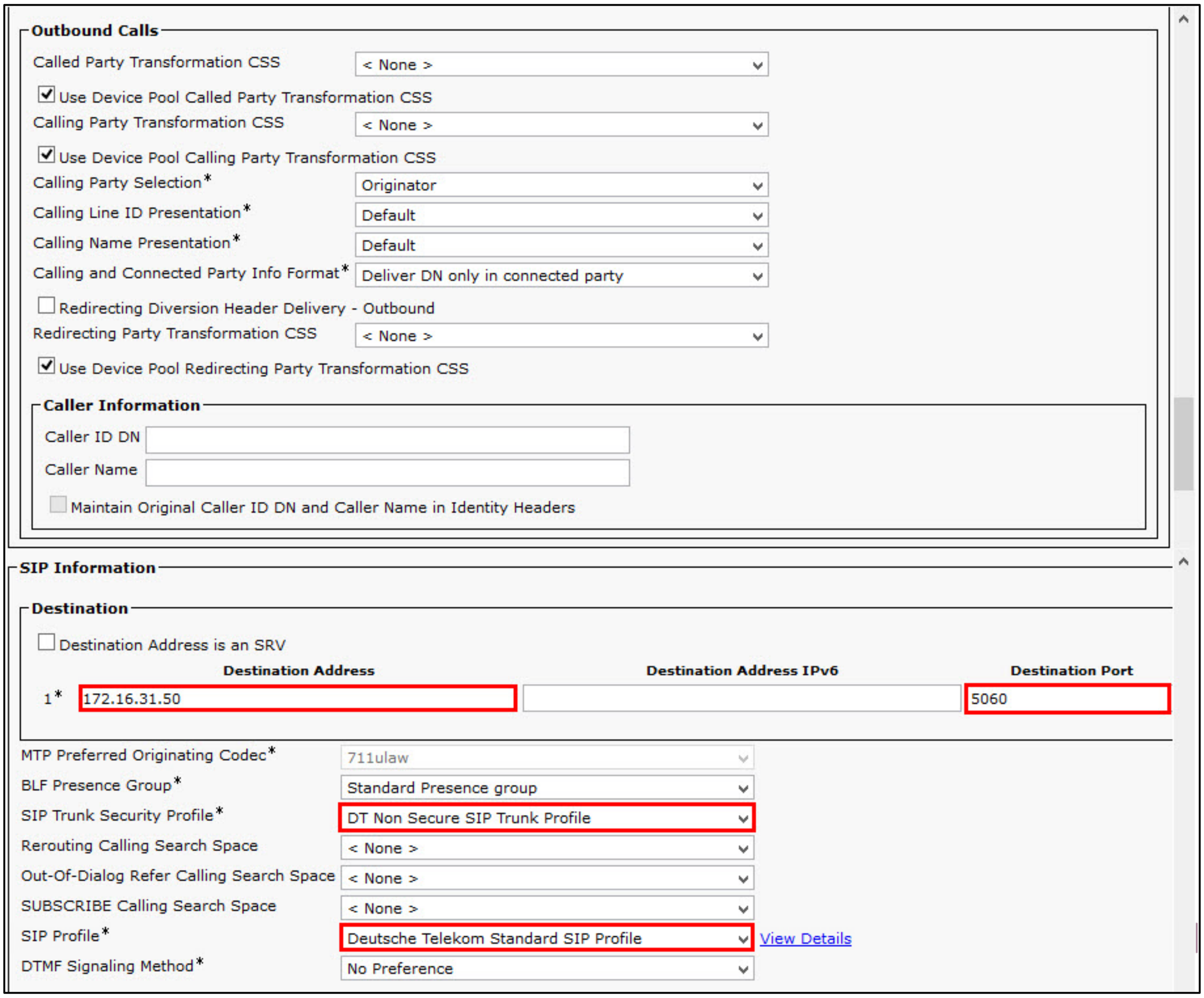

Figure 19: SIP Trunk to FAX Gateway (Cont.)

Routing configuration

Route Pattern for Cisco UBE:

Navigation: Call Routing → Route/Hunt → Route Pattern → Add New

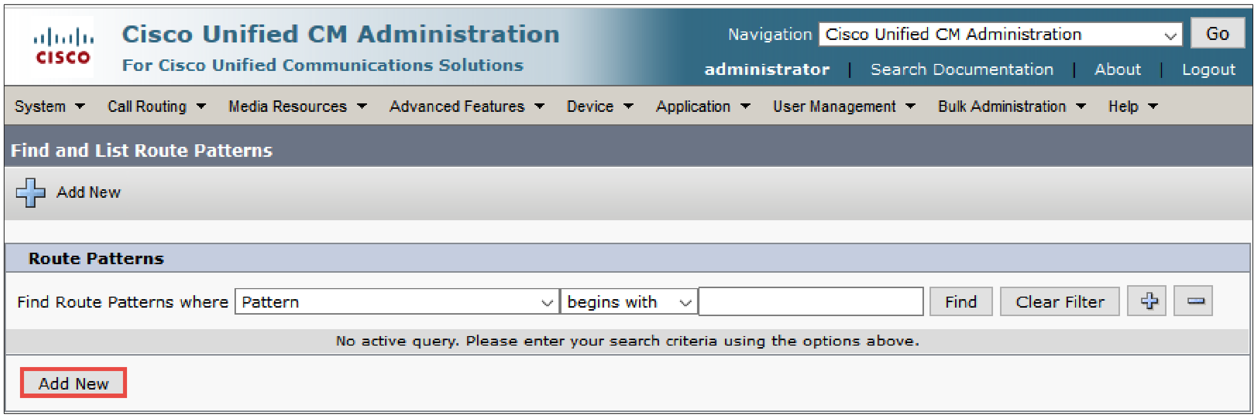

Figure 20: Add New Route Pattern for Cisco UBE

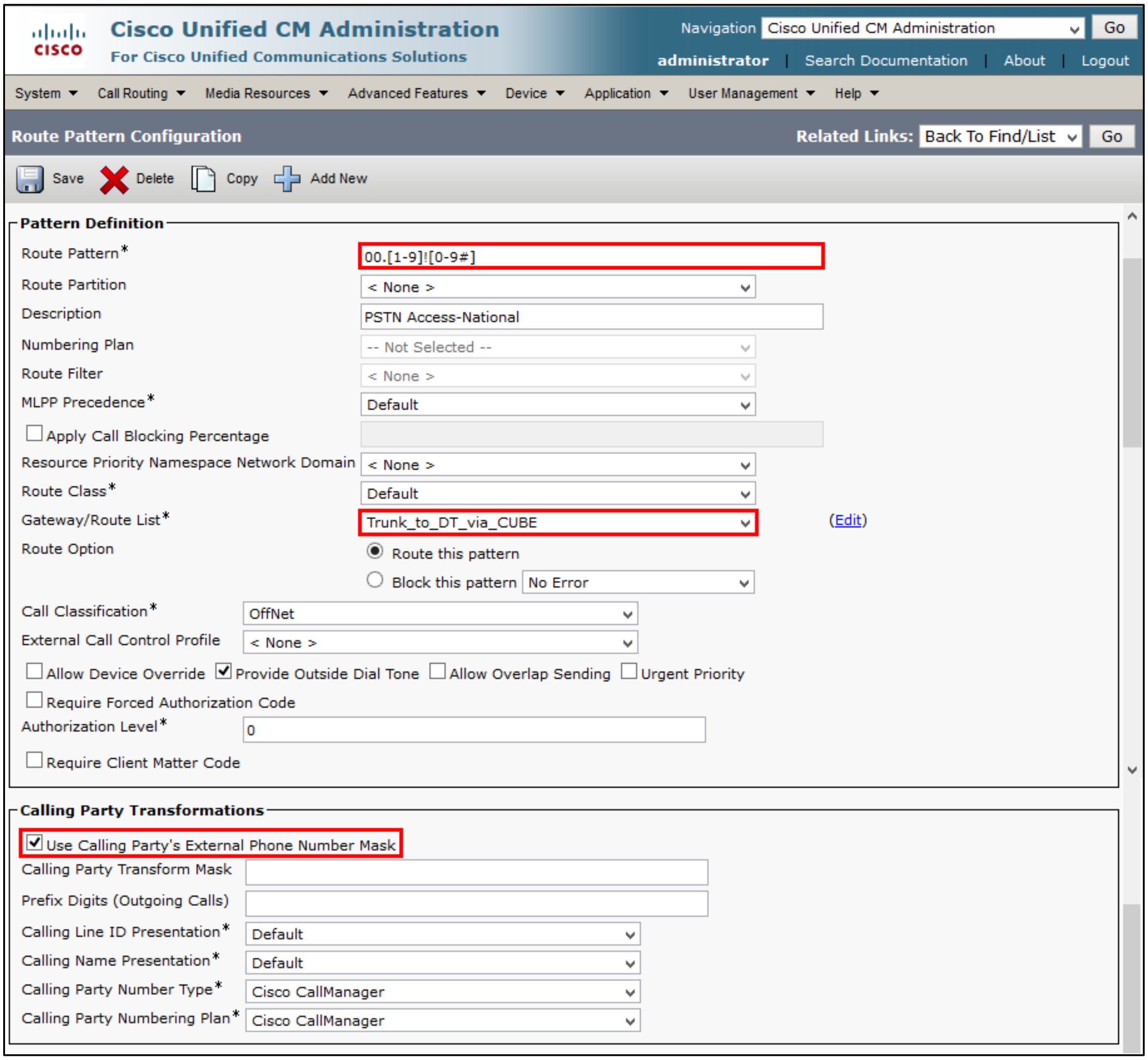

Figure 21: Route Pattern Configuration for Cisco UBE-PSTN Access-National

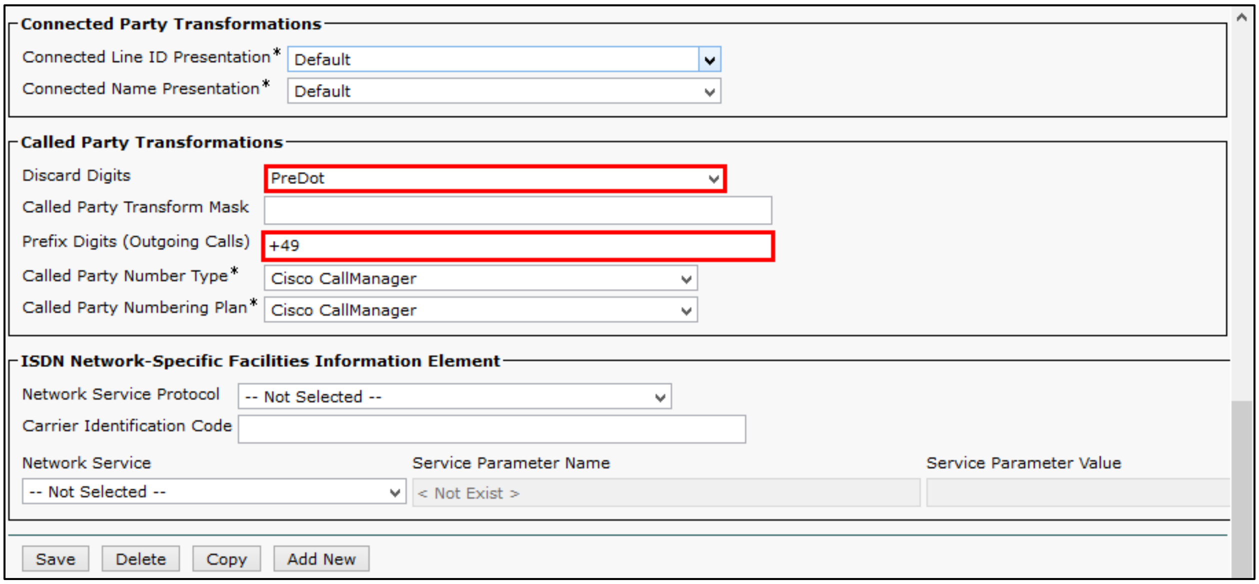

Figure 22: Route Pattern Configuration for Cisco UBE-PSTN Access-National (Cont.)

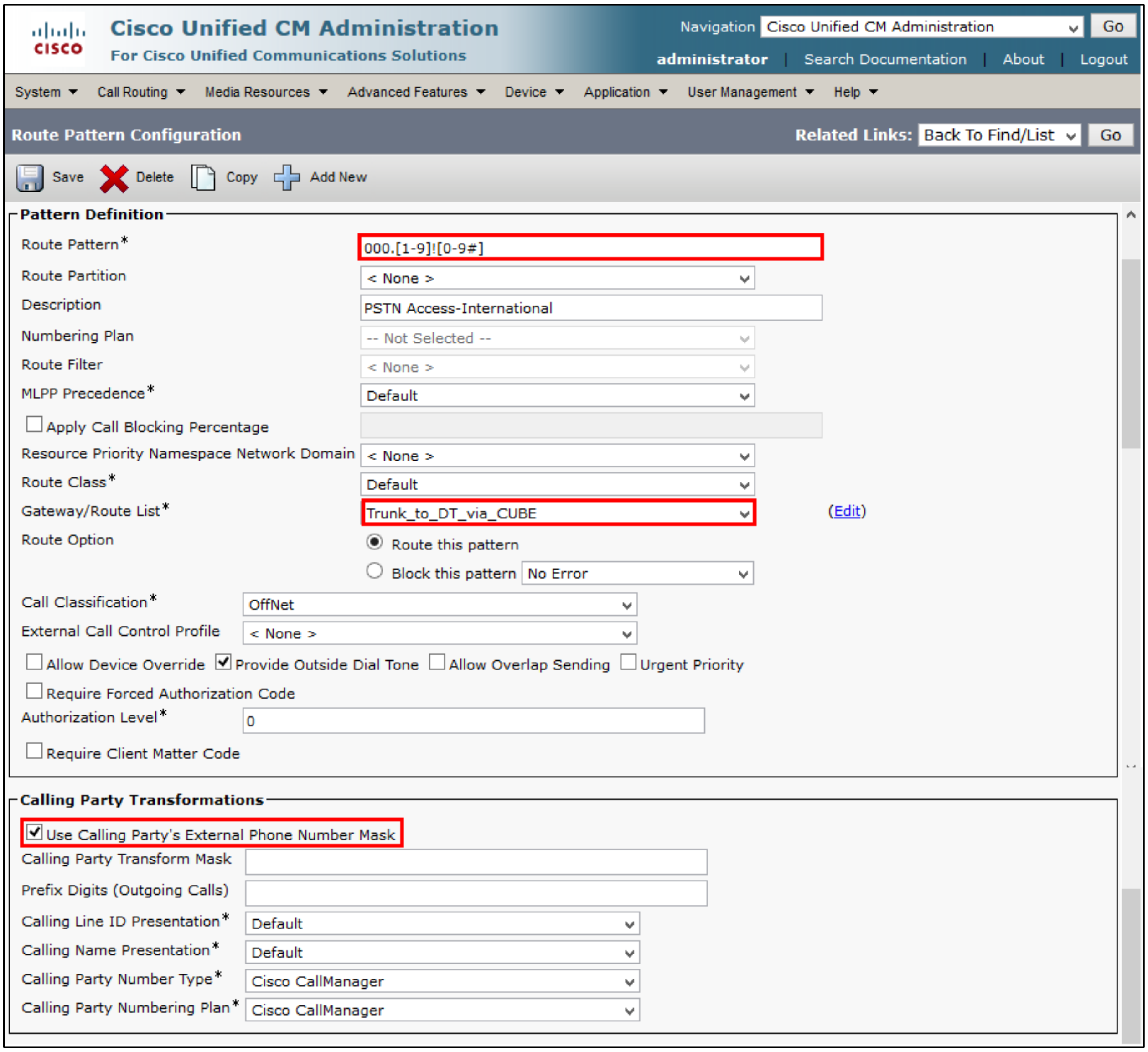

Figure 23: Route Pattern Configuration for Cisco UBE-PSTN Access-International

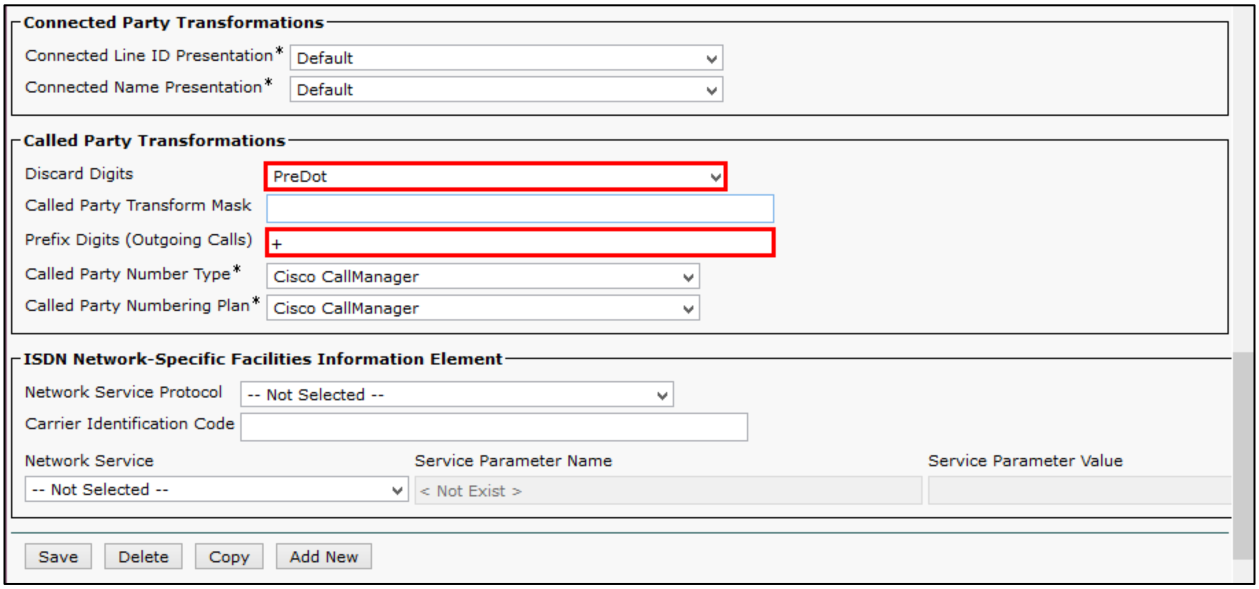

Figure 24: Route Pattern Configuration for Cisco UBE-PSTN Access-International (Cont.)

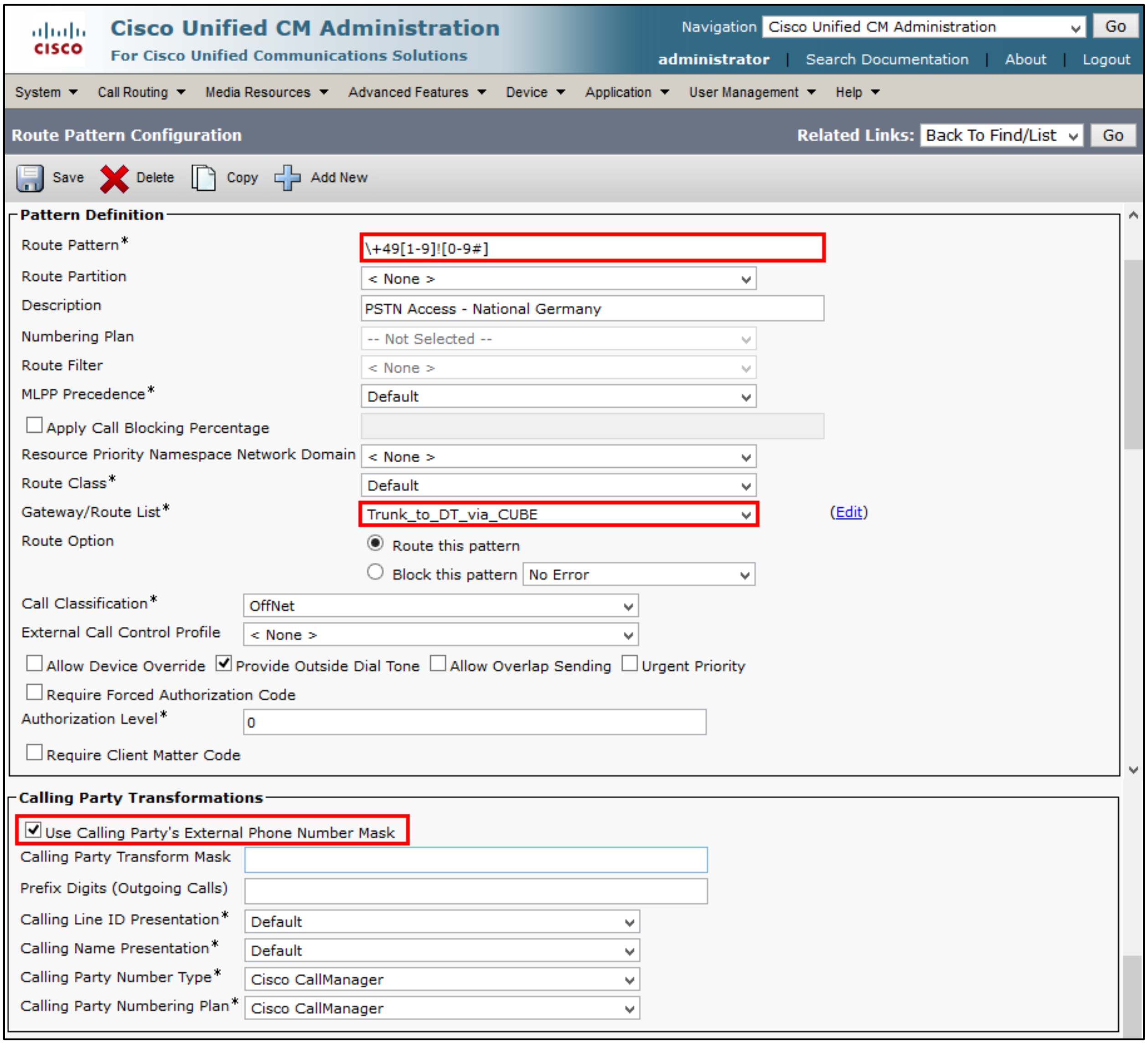

Figure 25: Route Pattern Configuration for Cisco UBE-PSTN Access-National

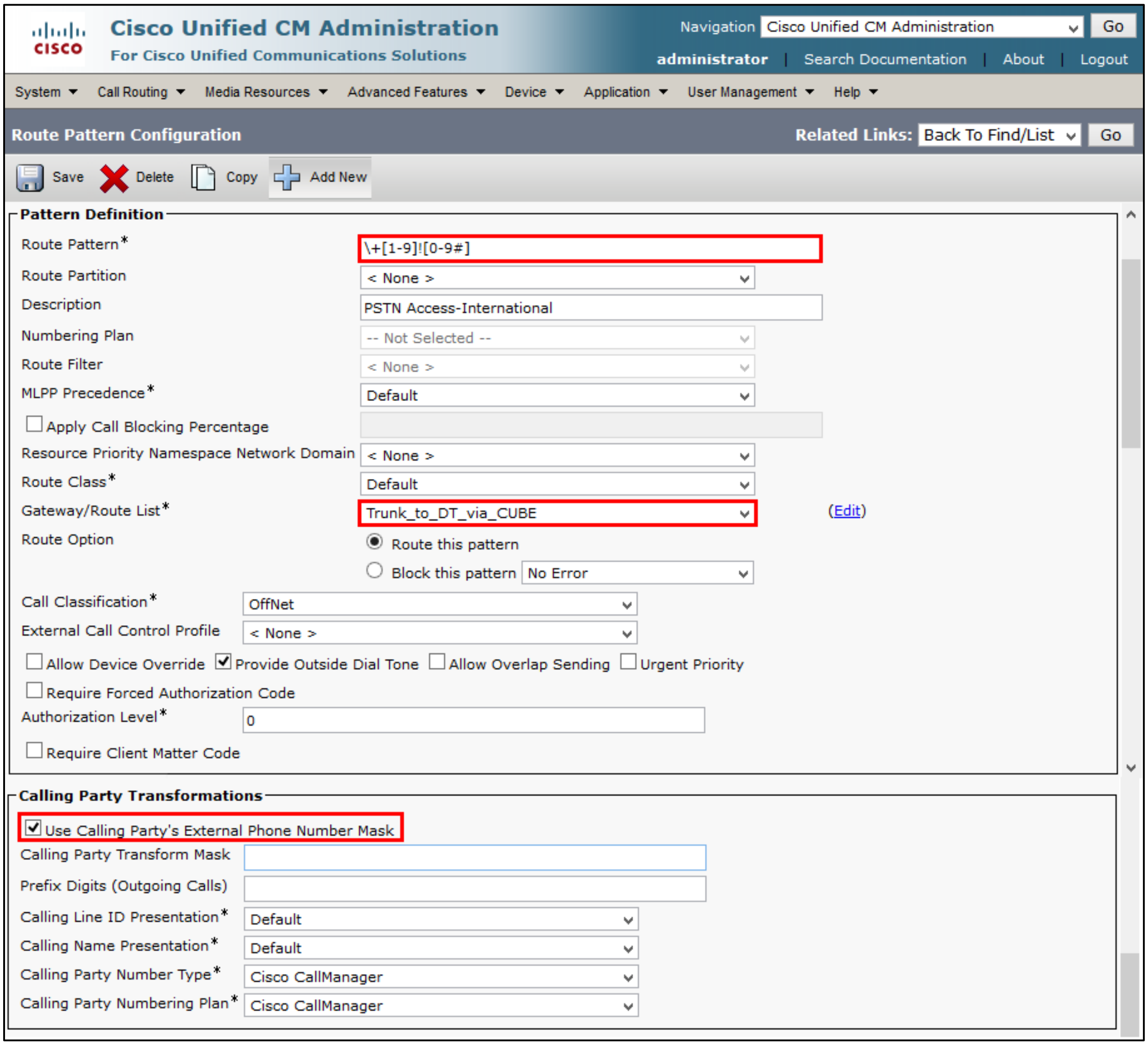

Figure 26: Route Pattern Configuration for Cisco UBE-PSTN Access-International

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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