



# 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

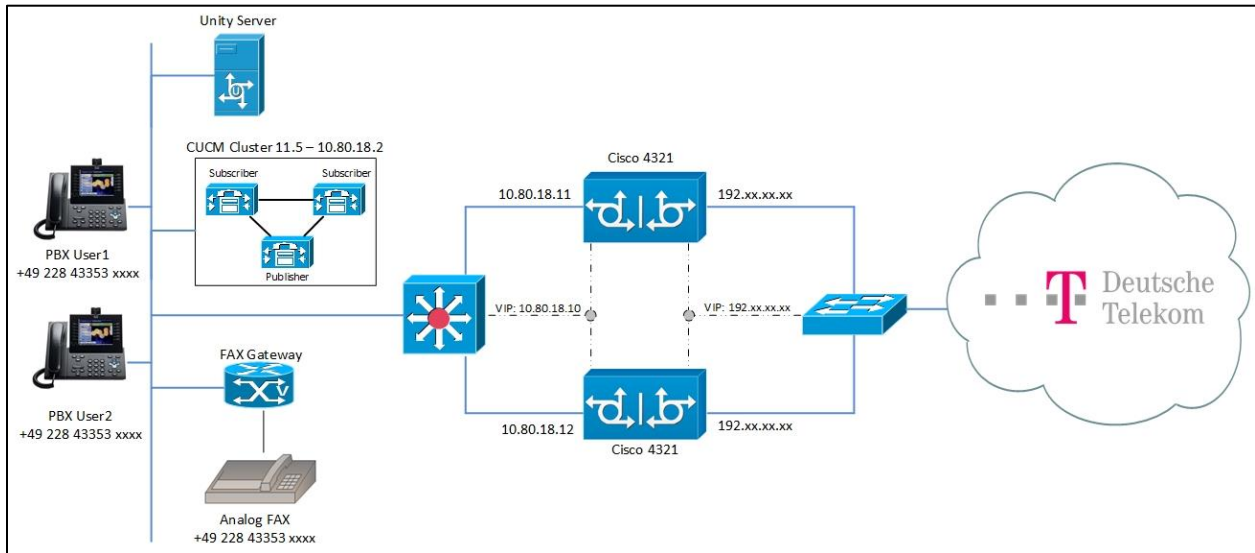


Figure 1 Network Topology

- 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.

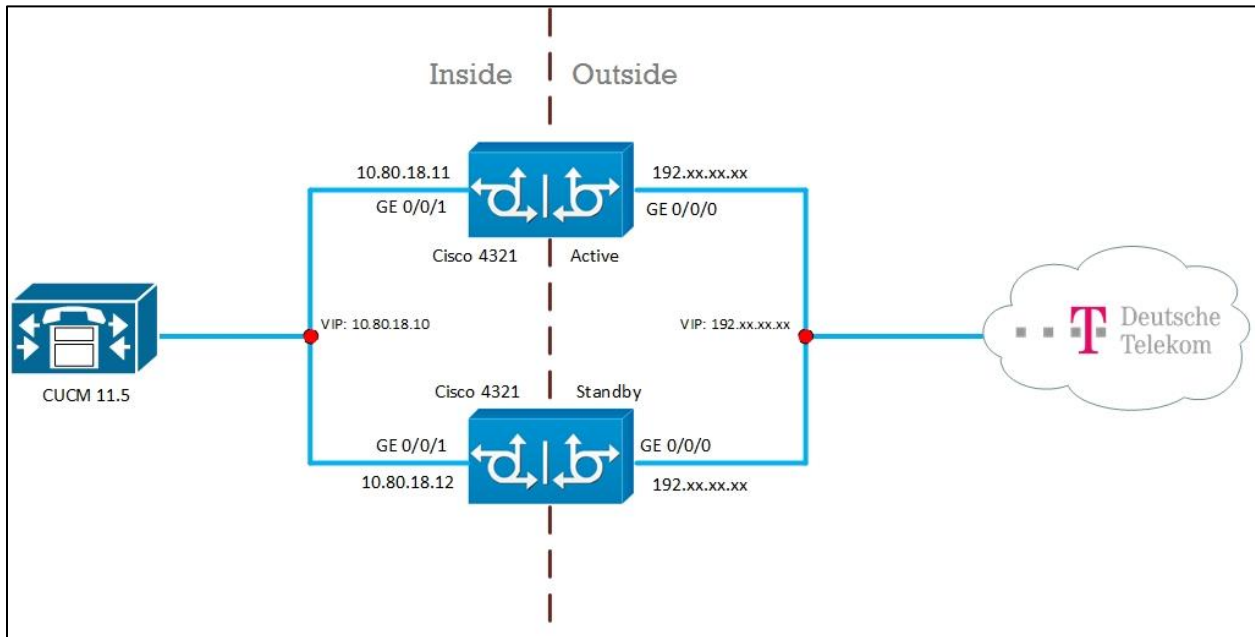


Figure 2 High Availability topology





### 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 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 FD055550MQ8
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
```

---

<sup>1</sup> 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 FD019220MW3  
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
```

---

<sup>2</sup> 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

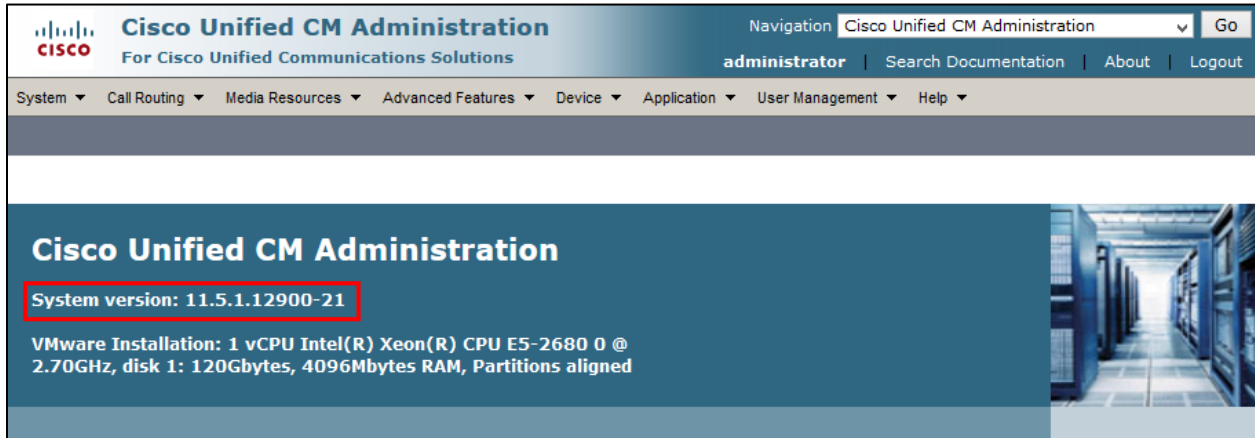


Figure 3: Cisco UCM Version



## 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

**Service Parameter Configuration**

Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Help

Related Links: Parameters for All Servers | Go

Save | Set to Default | Advanced

**Select Server and Service**

Server\* Clus28Sub1--CUCM Voice/Video (Active) ▾  
Service\* Cisco CallManager (Active) ▾

All parameters apply only to the current server except parameters that are in the cluster-wide group(s).

**Cisco CallManager (Active) Parameters on server Clus28Sub1--CUCM Voice/Video (Active)**

Parameter Name	Parameter Value	Suggested Value
<b>Call Throttling</b>		
<a href="#">Code Yellow Entry Latency</a> *	20	20
<a href="#">Code Yellow Exit Latency Calculation</a> *	40	40
<a href="#">Code Yellow Duration</a> *	5 ▾	5
<a href="#">Max Events Allowed</a> *	2000	2000
<a href="#">System Throttle Sample Size</a> *	10	10

Figure 4: Service Parameters



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

The screenshot displays the 'SIP Trunk Security Profile Configuration' page in the Cisco Unified CM Administration interface. The page title is 'SIP Trunk Security Profile Configuration' and it includes a 'Related Links: Back To Find/List' button. The configuration is for a profile named 'DT Non Secure SIP Trunk Profile' with the description 'Non Secure SIP Trunk Profile authenticated by null String'. The 'Device Security Mode' is set to 'Non Secure', 'Incoming Transport Type\*' is 'TCP+UDP', and 'Outgoing Transport Type' is 'TCP'. The 'Incoming Port\*' is 5060. Several checkboxes are checked: 'Accept out-of-dialog refer\*\*', 'Accept unsolicited notification', and 'Accept replaces header'. The 'SIP V.150 Outbound SDP Offer Filtering\*' is set to 'Use Default Filter'. The interface includes a navigation menu at the top, a search bar, and a toolbar with buttons for Save, Delete, Copy, Reset, Apply Config, and Add New.

Field	Value
Name*	DT Non Secure SIP Trunk Profile
Description	Non Secure SIP Trunk Profile authenticated by null String
Device Security Mode	Non Secure
Incoming Transport Type*	TCP+UDP
Outgoing Transport Type	TCP
Enable Digest Authentication	<input type="checkbox"/>
Nonce Validity Time (mins)*	600
X.509 Subject Name	
Incoming Port*	5060
Enable Application level authorization	<input type="checkbox"/>
Accept presence subscription	<input type="checkbox"/>
Accept out-of-dialog refer**	<input checked="" type="checkbox"/>
Accept unsolicited notification	<input checked="" type="checkbox"/>
Accept replaces header	<input checked="" type="checkbox"/>
Transmit security status	<input type="checkbox"/>
Allow charging header	<input type="checkbox"/>
SIP V.150 Outbound SDP Offer Filtering*	Use Default Filter

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

The screenshot displays the 'SIP Profile Configuration' page in the Cisco Unified CM Administration interface. The page is titled 'SIP Profile Configuration' and includes a navigation breadcrumb: 'System > Call Routing > Media Resources > Advanced Features > Device > Application > User Management > Help'. The user is logged in as 'administrator'. The page contains several sections:

- SIP Profile Information:**
  - Name\*: Deutsche Telekom Standard SIP Profile (highlighted with a red box)
  - Description: Default SIP Profile
  - Default MTP Telephony Event Payload Type\*: 101
  - Early Offer for G.Clear Calls\*: Disabled
  - User-Agent and Server header information\*: Send Unified CM Version Information as User-Agent
  - Version in User Agent and Server Header\*: Major And Minor
  - Dial String Interpretation\*: Phone number consists of characters 0-9, \*, #, and
  - Confidential Access Level Headers\*: Disabled
  - Redirect by Application:
  - Disable Early Media on 180:
  - Outgoing T.38 INVITE include audio mline:
  - Offer valid IP and Send/Receive mode only for T.38 Fax Relay:
  - Use Fully Qualified Domain Name in SIP Requests:
  - Assured Services SIP conformance:
  - Enable External QoS\*\*:
- SDP Information:**
  - SDP Session-level Bandwidth Modifier for Early Offer and Re-invites\*: TIAS and AS
  - SDP Transparency Profile: Pass all unknown SDP attributes
  - Accept Audio Codec Preferences in Received Offer\*: Default
  - Require SDP Inactive Exchange for Mid-Call Media Change:
  - Allow RR/RS bandwidth modifier (RFC 3556):

Figure 7: SIP Profile





Parameters used in Phone	
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	<input checked="" type="radio"/> Common Port Range for Audio and Video <input type="radio"/> 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
<input checked="" type="checkbox"/> Conference Join Enabled	
<input type="checkbox"/> RFC 2543 Hold	
<input checked="" type="checkbox"/> Semi Attended Transfer	
<input type="checkbox"/> Enable VAD	
<input type="checkbox"/> Stutter Message Waiting	
<input type="checkbox"/> MLPP User Authorization	

Figure 8: SIP Profile (Cont.)

**Normalization Script**

Normalization Script < None >

Enable Trace

	Parameter Name	Parameter Value	
1			<input type="button" value="+"/> <input type="button" value="-"/>

**Incoming Requests FROM URI Settings**

Caller ID DN

Caller Name

**Trunk Specific Configuration**

Reroute Incoming Request to new Trunk based on\* Never

Resource Priority Namespace List < None >

SIP Rel1XX Options\* Send PRACK for all 1xx Messages

Video Call Traffic Class\* Mixed

Calling Line Identification Presentation\* Default

Session Refresh Method\* Invite

Early Offer support for voice and video calls\* Best Effort (no MTP inserted)

Enable ANAT

Deliver Conference Bridge Identifier

Allow Passthrough of Configured Line Device Caller Information

Reject Anonymous Incoming Calls

Reject Anonymous Outgoing Calls

Send ILS Learned Destination Route String

Connect Inbound Call before Playing Queuing Announcement

**SIP OPTIONS Ping**

Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"

Ping Interval for In-service and Partially In-service Trunks (seconds)\* 60

Ping Interval for Out-of-service Trunks (seconds)\* 120

Ping Retry Timer (milliseconds)\* 500

Ping Retry Count\* 6

**SDP Information**

Send send-receive SDP in mid-call INVITE

Allow Presentation Sharing using BFCP

Allow iX Application Media

Allow multiple codecs in answer SDP

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

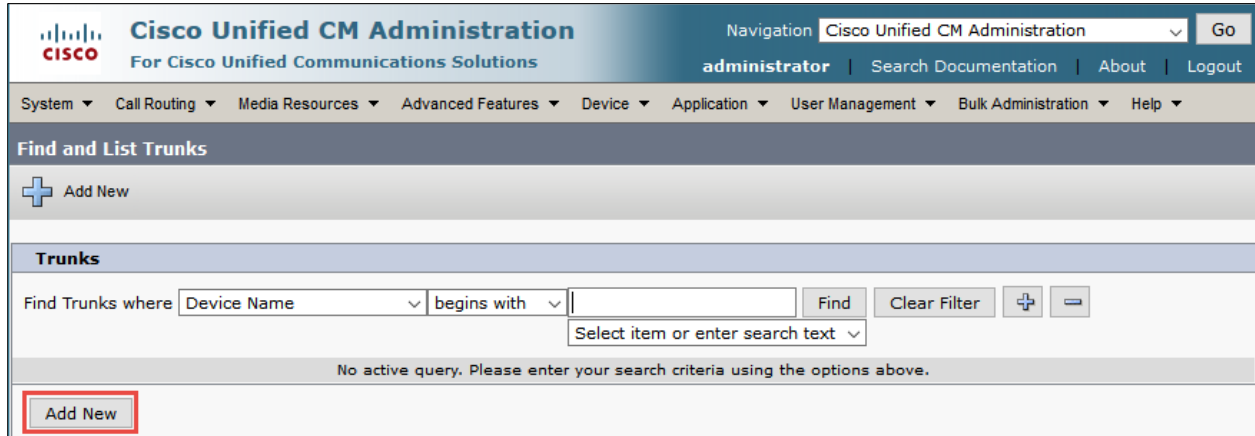


Figure 10: Add New Trunk to Cisco UBE

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

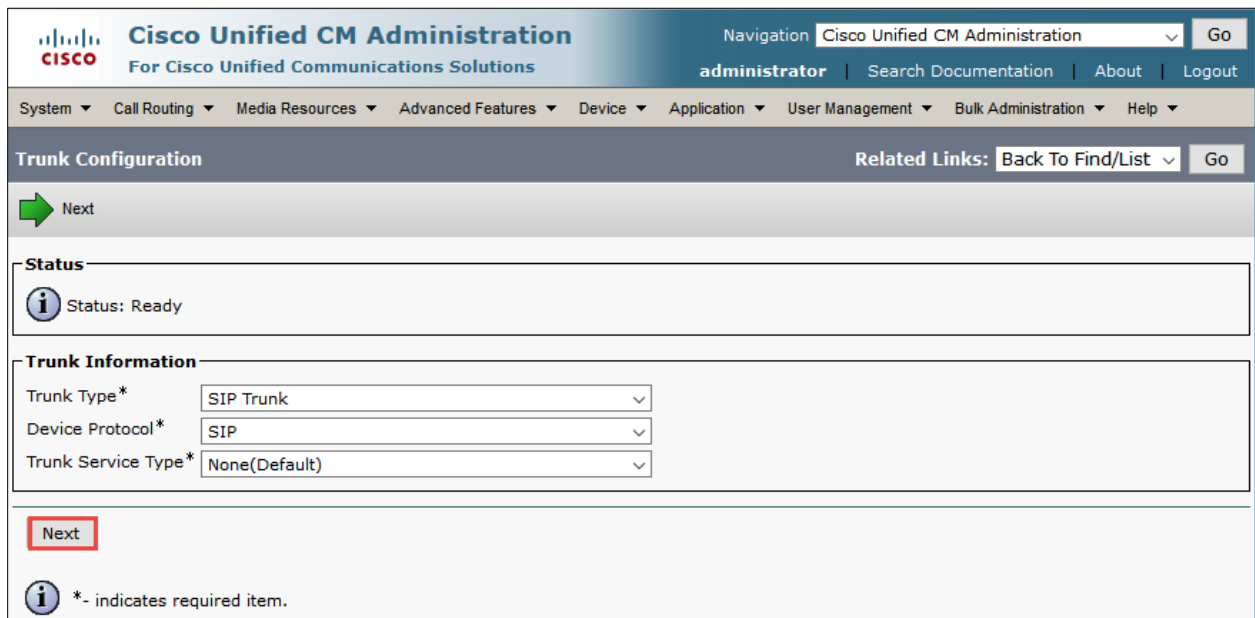


Figure 11: Add SIP Trunk Type

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | Go

administrator | Search Documentation | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Help

**Trunk Configuration** | Related Links: Back To Find/List | Go

Save | Delete | Reset | Add New

**Device Information**

Product: SIP Trunk  
 Device Protocol: SIP  
 Trunk Service Type: None(Default)  
 Device Name\*: **Trunk\_to\_DT\_via\_CUBE**  
 Description: Trunk\_to\_DT\_via\_CUBE  
 Device Pool\*: **Deutsche\_Devicepool**  
 Common Device Configuration: < None >  
 Call Classification\*: Use System Default  
 Media Resource Group List: MRGL\_Default  
 Location\*: Hub\_None  
 AAR Group: < None >  
 Tunneled Protocol\*: None  
 QSIG Variant\*: No Changes  
 ASN.1 ROSE OID Encoding\*: No Changes  
 Packet Capture Mode\*: None  
 Packet Capture Duration: 0

Media Termination Point Required  
 Retry Video Call as Audio  
 Path Replacement Support  
 Transmit UTF-8 for Calling Party Name  
 Transmit UTF-8 Names in QSIG APDU  
 Unattended Port  
 SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.  
 Consider Traffic on This Trunk Secure\*: When using both sRTP and TLS  
 Route Class Signaling Enabled\*: Default  
 Use Trusted Relay Point\*: Default  
 PSTN Access  
 Run On All Active Unified CM Nodes

Figure 12: SIP Trunk to Cisco UBE

**Intercompany Media Engine (IME)**

E.164 Transformation Profile < None >

---

**MLPP and Confidential Access Level Information**

MLPP Domain < None >

Confidential Access Mode < None >

Confidential Access Level < None >

---

**Call Routing Information**

Remote-Party-Id

Asserted-Identity

Asserted-Type\* PAI

SIP Privacy\* Default

---

**Inbound Calls**

Significant Digits\* 4

Connected Line ID Presentation\* Default

Connected Name Presentation\* Default

Calling Search Space < None >

AAR Calling Search Space < None >

Prefix DN

Redirecting Diversion Header Delivery - Inbound

---

**Incoming Calling Party Settings**

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

---

**Incoming Called Party Settings**

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

---

**Connected Party Settings**

Connected Party Transformation CSS < None >

Use Device Pool Connected Party Transformation CSS

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

**Outbound Calls**

Called Party Transformation CSS: < None >

Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS: < None >

Use Device Pool Calling Party Transformation CSS

Calling Party Selection\*: Originator

Calling Line ID Presentation\*: Default

Calling Name Presentation\*: Default

Calling and Connected Party Info Format\*: Deliver DN only in connected party

Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS: < None >

Use Device Pool Redirecting Party Transformation CSS

**Caller Information**

Caller ID DN:

Caller Name:

Maintain Original Caller ID DN and Caller Name in Identity Headers

**SIP Information**

Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1 *	10.80.18.10		5060

MTP Preferred Originating Codec\*: 711ulaw

BLF Presence Group\*: Standard Presence group

SIP Trunk Security Profile\*: DT Non Secure SIP Trunk Profile

Rerouting Calling Search Space: < None >

Out-Of-Dialog Refer Calling Search Space: < None >

SUBSCRIBE Calling Search Space: < None >

SIP Profile\*: Deutsche Telekom Standard SIP Profile [View Details](#)

DTMF Signaling Method\*: No Preference

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

Trunk configuration from Cisco UCM to Fax Gateway:

**Navigation:** Devices → Trunk → Add New

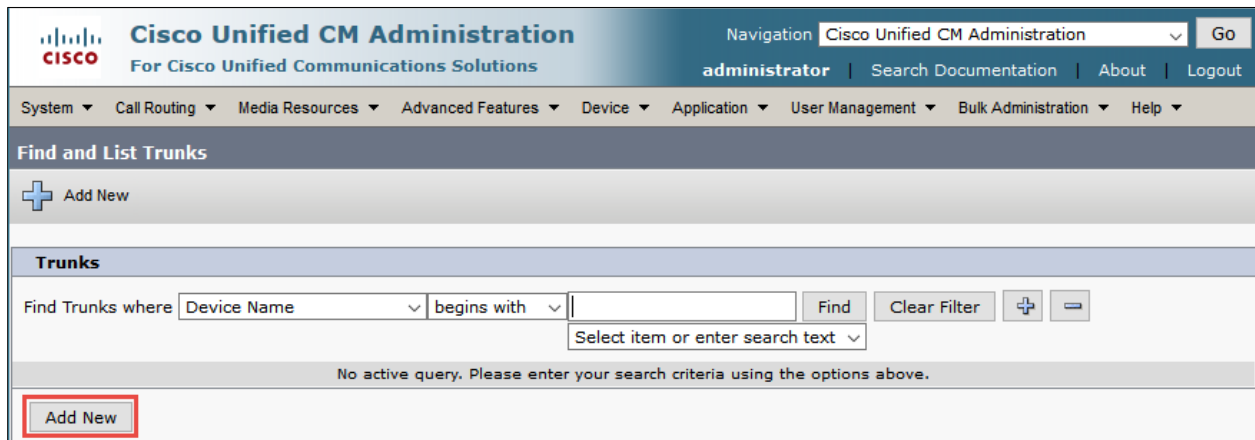


Figure 15: Add New Trunk to Fax Gateway

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

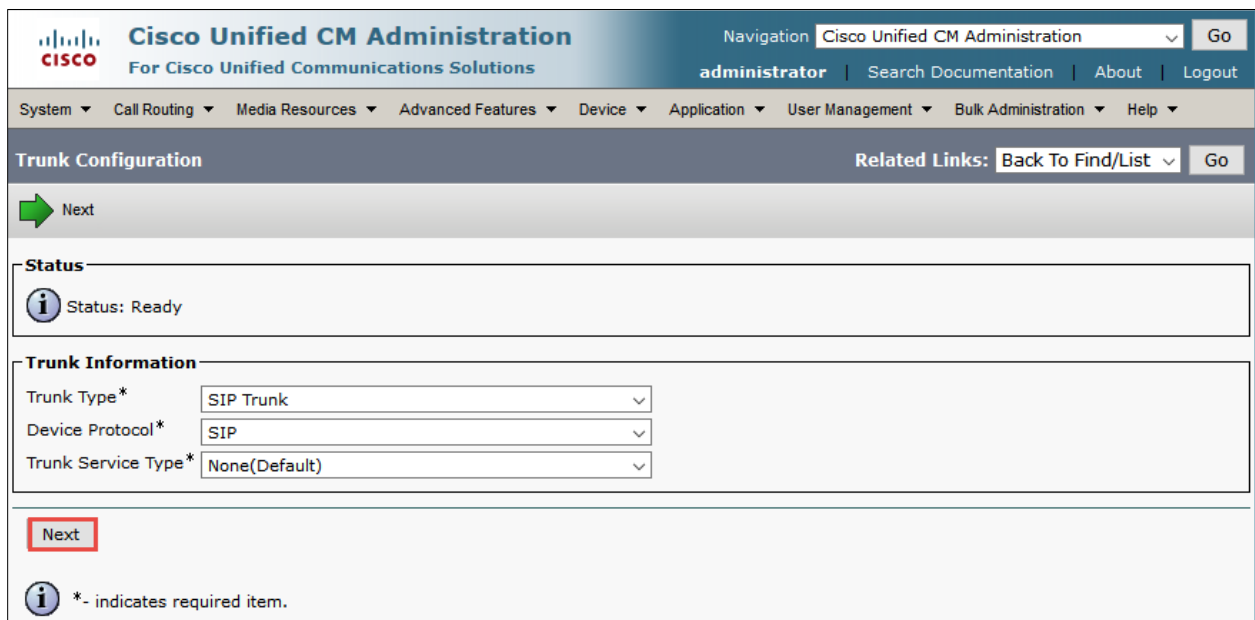


Figure 16: Add SIP Trunk Type

**Cisco Unified CM Administration**  
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Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Help

**Trunk Configuration** | Related Links: Back To Find/List | Go

Save | Delete | Reset | Add New

**Device Information**

Product: SIP Trunk  
 Device Protocol: SIP  
 Trunk Service Type: None(Default)  
 Device Name\*: Trunk\_to\_FAX\_GW\_DT  
 Description: Trunk\_to\_FAX\_GW\_DT  
 Device Pool\*: Deutsche\_Devicepool  
 Common Device Configuration: < None >  
 Call Classification\*: Use System Default  
 Media Resource Group List: MRGL\_Default  
 Location\*: Hub\_None  
 AAR Group: < None >  
 Tunneled Protocol\*: None  
 QSIG Variant\*: No Changes  
 ASN.1 ROSE OID Encoding\*: No Changes  
 Packet Capture Mode\*: None  
 Packet Capture Duration: 0

Media Termination Point Required  
 Retry Video Call as Audio  
 Path Replacement Support  
 Transmit UTF-8 for Calling Party Name  
 Transmit UTF-8 Names in QSIG APDU  
 Unattended Port  
 SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.

Consider Traffic on This Trunk Secure\*: When using both sRTP and TLS  
 Route Class Signaling Enabled\*: Default  
 Use Trusted Relay Point\*: Default

PSTN Access  
 Run On All Active Unified CM Nodes

Figure 17: SIP Trunk to FAX Gateway



**Intercompany Media Engine (IME)**  
 E.164 Transformation Profile < None >

**MLPP and Confidential Access Level Information**  
 MLPP Domain < None >  
 Confidential Access Mode < None >  
 Confidential Access Level < None >

**Call Routing Information**  
 Remote-Party-Id  
 Asserted-Identity  
 Asserted-Type\* PAI  
 SIP Privacy\* Default

**Inbound Calls**  
 Significant Digits\* All  
 Connected Line ID Presentation\* Default  
 Connected Name Presentation\* Default  
 Calling Search Space < None >  
 AAR Calling Search Space < None >  
 Prefix DN  
 Redirecting Diversion Header Delivery - Inbound

**Incoming Calling Party Settings**  
 If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.  
 Clear Prefix Settings Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

**Incoming Called Party Settings**  
 If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.  
 Clear Prefix Settings Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

**Connected Party Settings**  
 Connected Party Transformation CSS < None >  
 Use Device Pool Connected Party Transformation CSS

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

**Outbound Calls**

Called Party Transformation CSS

Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS

Use Device Pool Calling Party Transformation CSS

Calling Party Selection\*

Calling Line ID Presentation\*

Calling Name Presentation\*

Calling and Connected Party Info Format\*

Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS

Use Device Pool Redirecting Party Transformation CSS

**Caller Information**

Caller ID DN

Caller Name

Maintain Original Caller ID DN and Caller Name in Identity Headers

**SIP Information**

**Destination**

Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1 *	<input type="text" value=" 172.16.31.50"/>	<input type="text"/>	<input type="text" value=" 5060"/>

MTP Preferred Originating Codec\*

BLF Presence Group\*

SIP Trunk Security Profile\*

Rerouting Calling Search Space

Out-Of-Dialog Refer Calling Search Space

SUBSCRIBE Calling Search Space

SIP Profile\*  [View Details](#)

DTMF Signaling Method\*

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



## Routing configuration

Route Pattern for Cisco UBE:

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

The screenshot shows the Cisco Unified CM Administration web interface. At the top, there is a navigation bar with the Cisco logo and the text "Cisco Unified CM Administration For Cisco Unified Communications Solutions". The user is logged in as "administrator". Below the navigation bar, there is a menu with options: System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled "Find and List Route Patterns" and contains a "+ Add New" button. Below this, there is a search section titled "Route Patterns" with a search bar containing "Pattern" and a dropdown menu set to "begins with". There are "Find", "Clear Filter", and "Add New" buttons. A message below the search bar reads: "No active query. Please enter your search criteria using the options above." The "Add New" button is highlighted with a red box.

Figure 20: Add New Route Pattern for Cisco UBE



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### Route Pattern Configuration

Related Links: Back To Find/List | Go

Save | Delete | Copy | Add New

#### Pattern Definition

Route Pattern*	00.[1-9]! [0-9#]
Route Partition	< None >
Description	PSTN Access-National
Numbering Plan	-- Not Selected --
Route Filter	< None >
MLPP Precedence*	Default
<input type="checkbox"/> Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	Trunk_to_DT_via_CUBE (Edit)
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error
Call Classification*	OffNet
External Call Control Profile	< None >
<input type="checkbox"/> Allow Device Override <input checked="" type="checkbox"/> Provide Outside Dial Tone <input type="checkbox"/> Allow Overlap Sending <input type="checkbox"/> Urgent Priority	
<input type="checkbox"/> Require Forced Authorization Code	
Authorization Level*	0
<input type="checkbox"/> Require Client Matter Code	

#### Calling Party Transformations

<input checked="" type="checkbox"/> Use Calling Party's External Phone Number Mask	
Calling Party Transform Mask	
Prefix Digits (Outgoing Calls)	
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Calling Party Number Type*	Cisco CallManager
Calling Party Numbering Plan*	Cisco CallManager

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



**Connected Party Transformations**

Connected Line ID Presentation\*

Connected Name Presentation\*

**Called Party Transformations**

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type\*

Called Party Numbering Plan\*

**ISDN Network-Specific Facilities Information Element**

Network Service Protocol

Carrier Identification Code

Network Service	Service Parameter Name	Service Parameter Value
<input type="text" value="-- Not Selected --"/>	<input type="text" value="&lt; Not Exist &gt;"/>	<input type="text"/>

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

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Help

### Route Pattern Configuration

Related Links: Back To Find/List | Go

Save | Delete | Copy | Add New

**Pattern Definition**

Route Pattern\* **000.[1-9]!([0-9#]**

Route Partition: < None >

Description: PSTN Access-International

Numbering Plan: -- Not Selected --

Route Filter: < None >

MLPP Precedence\*: Default

Apply Call Blocking Percentage

Resource Priority Namespace Network Domain: < None >

Route Class\*: Default

Gateway/Route List\* **Trunk\_to\_DT\_via\_CUBE** (Edit)

Route Option  
 Route this pattern  
 Block this pattern: No Error

Call Classification\*: OffNet

External Call Control Profile: < None >

Allow Device Override  Provide Outside Dial Tone  Allow Overlap Sending  Urgent Priority

Require Forced Authorization Code

Authorization Level\*: 0

Require Client Matter Code

**Calling Party Transformations**

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask:

Prefix Digits (Outgoing Calls):

Calling Line ID Presentation\*: Default

Calling Name Presentation\*: Default

Calling Party Number Type\*: Cisco CallManager

Calling Party Numbering Plan\*: Cisco CallManager

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



<b>Connected Party Transformations</b>		
Connected Line ID Presentation*	Default	
Connected Name Presentation*	Default	
<b>Called Party Transformations</b>		
Discard Digits	PreDot	
Called Party Transform Mask		
Prefix Digits (Outgoing Calls)	+	
Called Party Number Type*	Cisco CallManager	
Called Party Numbering Plan*	Cisco CallManager	
<b>ISDN Network-Specific Facilities Information Element</b>		
Network Service Protocol	-- Not Selected --	
Carrier Identification Code		
Network Service	Service Parameter Name	Service Parameter Value
-- Not Selected --	< Not Exist >	
Save Delete Copy Add New		

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



**Cisco Unified CM Administration** Navigation Cisco Unified CM Administration Go  
administrator | Search Documentation | About | Logout

System Call Routing Media Resources Advanced Features Device Application User Management Help

### Route Pattern Configuration

 Related Links: Back To Find/List Go

Save Delete Copy Add New

#### Pattern Definition

Route Pattern*	<input type="text" value="\+49[1-9]![0-9#]"/>
Route Partition	< None >
Description	PSTN Access - National Germany
Numbering Plan	-- Not Selected --
Route Filter	< None >
MLPP Precedence*	Default
<input type="checkbox"/> Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	Trunk_to_DT_via_CUBE <a href="#">(Edit)</a>
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern <input type="text" value="No Error"/>
Call Classification*	OffNet
External Call Control Profile	< None >
<input type="checkbox"/> Allow Device Override <input checked="" type="checkbox"/> Provide Outside Dial Tone <input type="checkbox"/> Allow Overlap Sending <input type="checkbox"/> Urgent Priority	
<input type="checkbox"/> Require Forced Authorization Code	
Authorization Level*	0
<input type="checkbox"/> Require Client Matter Code	

#### Calling Party Transformations

<input checked="" type="checkbox"/> Use Calling Party's External Phone Number Mask	
Calling Party Transform Mask	<input type="text"/>
Prefix Digits (Outgoing Calls)	<input type="text"/>
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Calling Party Number Type*	Cisco CallManager
Calling Party Numbering Plan*	Cisco CallManager

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





**Cisco Unified CM Administration** Navigation Cisco Unified CM Administration Go  
administrator Search Documentation About Logout

System Call Routing Media Resources Advanced Features Device Application User Management Help

### Route Pattern Configuration

 Related Links: Back To Find/List Go

Save Delete Copy Add New

#### Pattern Definition

Route Pattern\*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence\*

Apply Call Blocking Percentage

Resource Priority Namespace Network Domain

Route Class\*

Gateway/Route List\*  (Edit)

Route Option  
 Route this pattern  
 Block this pattern

Call Classification\*

External Call Control Profile

Allow Device Override  Provide Outside Dial Tone  Allow Overlap Sending  Urgent Priority

Require Forced Authorization Code

Authorization Level\*

Require Client Matter Code

#### Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation\*

Calling Name Presentation\*

Calling Party Number Type\*

Calling Party Numbering Plan\*

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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