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

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:

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Transport from Cisco UBE to Cisco UCM</td>
<td>TCP with RTP</td>
</tr>
<tr>
<td>Transport from Cisco UBE to Deutsche Telekom</td>
<td>TCP with RTP</td>
</tr>
<tr>
<td>Voice Mail Support</td>
<td>YES</td>
</tr>
<tr>
<td>Session Refresh</td>
<td>YES</td>
</tr>
<tr>
<td>Early Media support with PRACK</td>
<td>YES</td>
</tr>
<tr>
<td>G729 Conference Support</td>
<td>NO</td>
</tr>
</tbody>
</table>
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**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>allow-connections sip to sip</td>
<td>Allow IP2IP connections between two SIP call legs</td>
</tr>
<tr>
<td>redundancy-group 2</td>
<td>Enable High Availability for the VoIP service</td>
</tr>
<tr>
<td>fax protocol</td>
<td>Specifies the fax protocol</td>
</tr>
<tr>
<td>asserted-id</td>
<td>Specifies the privacy header in the outgoing SIP requests and response messages</td>
</tr>
</tbody>
</table>
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
    3435A17 0D323030 31303330 305A3031 312F302D 06035504 03132649
    4F532D53 656C662D 5369676E 65642D43 65727469 66696361 38333030
    36308201 22300D06 092A8648 86F70D01 01010500 0382010F 00308201
    0A028201 010034F 136DD7A1 E1815DA1 B05C5396 E3B88AC9 7DE1A1D7 12F1BEA7
    4985E12B C685F9D 95E7082B 3BBC56CD AAFDEC4F 7250D4CE 892713BE 509A6DCE
    05FD3768 ED1EF293 B3C2C1CE 4684F8F9 E920AE8F 33F4DFE0 FF04BE27 B75A28C1
    6A2084C5 31BFF5C1 CD07916D 83FD56EF 9023C974 A9835AA1 C0FB6856
    5CA7B10A AF9EFC1 5DE8651F D30847FF 02D46EFE 3AADB77D 68519BA9 F21AC1FE
    5A50CA58 A00CDBB5 25C693E8 4D8C639D 6E5A3935 2F05F04D A3A7B2AD 47942BDD
    4D78EFE8 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 03551DOE 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 E8FD00E0 37A2A866 F759FB2D E30CD988 8900E25E 8171A288
FB2BB185 B6A6ED29 3BAA4495 31FCC789 0305E830 6EBA491E 211F0B7C FE808066
693C7384 D1C54B99 BAC2A7AA 5B646C6D 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 FDO5555OMQ8
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-nie
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**

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:reg.sip-trunk.telekom.de expires 240 tcp auth-realm sip-trunk.telekom.de

  sip-server dns:reg.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
collection-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 43657469
    69666966 6174652D 32333332 3436301E 170D3137 30323130 31373132
    33365A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649
    4F532D53 656C662D 5365642D 43657469 69666966 6174652D 32353438 34333234
    34333234 36308201 22300D06 092A8648 86F70D01 01010500 0382010F 00308201
    0A028201 0100873A 8AE876C0 62A38109 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 7EFDDC2B 1902A0A8 038AD01E
voice service voip

ip address trusted list

ipv4 217.0.0.0 255.255.0.0

disable-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**
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

![Cisco Unified CM Administration](image)

**Cisco Unified CM Administration**

**System version: 11.5.1.12900-21**

VMware Installation: 1 vCPU Intel(R) Xeon(R) CPU E5-2680 0 @ 2.70GHz, disk 1: 120Gbytes, 4096Mbytes RAM, Partitions aligned

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**
<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Notes</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default Network Hold MOH Audio Source ID.</td>
<td>1</td>
<td></td>
</tr>
<tr>
<td>Default User Hold MOH Audio Source ID.</td>
<td>1</td>
<td></td>
</tr>
<tr>
<td>Max Recursion Depth for Conditional Call Forward</td>
<td>32</td>
<td></td>
</tr>
<tr>
<td>[Note: This parameter is not shown in the image.]</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Media Exchange Interface Capability Timer</td>
<td>10</td>
<td></td>
</tr>
<tr>
<td>Send Multicast MOH in H.245 OLC Message.</td>
<td>True</td>
<td></td>
</tr>
<tr>
<td>Media Exchange Timer</td>
<td>12</td>
<td></td>
</tr>
<tr>
<td>Media Exchange Stop Streaming Timer</td>
<td>12</td>
<td></td>
</tr>
<tr>
<td>Open Video Channel Response Timer for SIP Interop.</td>
<td>500</td>
<td></td>
</tr>
<tr>
<td>Part Received Timer After Call Connection.</td>
<td>300</td>
<td></td>
</tr>
<tr>
<td>Media Resource Allocation Timer.</td>
<td>12</td>
<td></td>
</tr>
<tr>
<td>MTP and Transcoder Resource Throttling Percentage.</td>
<td>95</td>
<td></td>
</tr>
<tr>
<td>Intercluster Capabilities Mismatch Timer.</td>
<td>1000</td>
<td></td>
</tr>
<tr>
<td>Silence Suppression</td>
<td>True</td>
<td></td>
</tr>
<tr>
<td>Silence Suppression for Gateways</td>
<td>True</td>
<td></td>
</tr>
<tr>
<td>Strip G.729 Annex B (Silence Suppression) from Capabilities.</td>
<td>True</td>
<td></td>
</tr>
<tr>
<td>Enable Source IP Address Verification for Software Media Devices.</td>
<td>True</td>
<td></td>
</tr>
</tbody>
</table>

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

![SIP Trunk Security Profile Configuration](image)

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

![SIP Profile Configuration](image)

**Figure 7:** SIP Profile
Table: Parameters used in Phone

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Timer Invite Expires (seconds)</td>
<td>180</td>
</tr>
<tr>
<td>Timer Register Delta (seconds)</td>
<td>5</td>
</tr>
<tr>
<td>Timer Register Expires (seconds)</td>
<td>3600</td>
</tr>
<tr>
<td>Timer T1 (msec)</td>
<td>500</td>
</tr>
<tr>
<td>Timer T2 (msec)</td>
<td>4000</td>
</tr>
<tr>
<td>Retry INVITE</td>
<td>6</td>
</tr>
<tr>
<td>Retry Non-INVITE</td>
<td>10</td>
</tr>
<tr>
<td>Media Port Ranges</td>
<td></td>
</tr>
<tr>
<td>Start Media Port</td>
<td>16384</td>
</tr>
<tr>
<td>Stop Media Port</td>
<td>32766</td>
</tr>
<tr>
<td>DSCP for Audio Calls</td>
<td></td>
</tr>
<tr>
<td>DSCP for Video Calls</td>
<td></td>
</tr>
<tr>
<td>DSCP for Audio Portion of Video Calls</td>
<td></td>
</tr>
<tr>
<td>DSCP for TelePresence Calls</td>
<td></td>
</tr>
<tr>
<td>DSCP for Audio Portion of TelePresence Calls</td>
<td></td>
</tr>
<tr>
<td>Call Pickup URI*</td>
<td></td>
</tr>
<tr>
<td>Call Pickup Group Other URI*</td>
<td></td>
</tr>
<tr>
<td>Call Pickup Group URI*</td>
<td></td>
</tr>
<tr>
<td>Meet Me Service URI*</td>
<td></td>
</tr>
<tr>
<td>User Info*</td>
<td></td>
</tr>
<tr>
<td>DTMF DB Level*</td>
<td></td>
</tr>
<tr>
<td>Call Hold Ring Back</td>
<td></td>
</tr>
<tr>
<td>Anonymous Call Block*</td>
<td></td>
</tr>
<tr>
<td>Caller ID Blocking*</td>
<td></td>
</tr>
<tr>
<td>Do Not Disturb Control*</td>
<td></td>
</tr>
<tr>
<td>Telnet Level for 7940 and 7960*</td>
<td></td>
</tr>
<tr>
<td>Resource Priority Namespace</td>
<td></td>
</tr>
<tr>
<td>Timer Keep Alive Expires (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Expires (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Delta (seconds)</td>
<td>5</td>
</tr>
<tr>
<td>Maximum Redirects*</td>
<td>70</td>
</tr>
<tr>
<td>Off Hook To First Digit Timer (milliseconds)</td>
<td>15000</td>
</tr>
<tr>
<td>Call Forward URI*</td>
<td></td>
</tr>
<tr>
<td>Speed Dial (Abbreviated Dial) URI*</td>
<td></td>
</tr>
<tr>
<td>Conference Join Enabled</td>
<td>✔</td>
</tr>
<tr>
<td>RFC 2543 Hold</td>
<td></td>
</tr>
<tr>
<td>Semi Attended Transfer</td>
<td>✔</td>
</tr>
<tr>
<td>Enable VAD</td>
<td></td>
</tr>
<tr>
<td>Stutter Message Waiting</td>
<td></td>
</tr>
<tr>
<td>MLPP User Authorization</td>
<td></td>
</tr>
</tbody>
</table>

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

![Image of Cisco Unified CM Administration interface showing Trunk configuration]

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

![Image of Cisco Unified CM Administration interface showing SIP Trunk configuration]

**Figure 10:** Add New Trunk to Cisco UBE

**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

![Cisco Unified CM Administration interface](image)

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

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definitions</th>
</tr>
</thead>
<tbody>
<tr>
<td>CPE</td>
<td>Customer Premise Equipment</td>
</tr>
<tr>
<td>Cisco UBE</td>
<td>Cisco Unified Border Element</td>
</tr>
<tr>
<td>Cisco UCM</td>
<td>Cisco Unified Communications Manager</td>
</tr>
<tr>
<td>MTP</td>
<td>Media Termination Point</td>
</tr>
<tr>
<td>POP</td>
<td>Point of Presence</td>
</tr>
<tr>
<td>PSTN</td>
<td>Public Switched Telephone Network</td>
</tr>
<tr>
<td>ESBC</td>
<td>Enterprise Session Border Controller</td>
</tr>
<tr>
<td>SCCP</td>
<td>Skinny Client Control Protocol</td>
</tr>
<tr>
<td>SIP</td>
<td>Session Initiation Protocol</td>
</tr>
</tbody>
</table>
Important Information

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