# Telekom SIP Trunk / DE

Dienstag, 22. September 2020 08:58

```
ip name-server 217.0.43.145 217.0.43.129
!
voice call send-alert
no voice call carrier capacity active
voice rtp send-recv
!
voice service voip
 ip address trusted list
  ipv4 217.0.0.0 255.255.0.0
  ipv4 [CUCM Pub IP] 255.255.255.255
  ipv4 [CUCM Sub IP] 255.255.255.255
 rtp-port range 16384 32766
 address-hiding
 mode border-element license capacity [Session count]
 media statistics
 media bulk-stats
 allow-connections sip to sip
 no supplementary-service sip moved-temporarily
 no supplementary-service sip refer
 no supplementary-service sip handle-replaces
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711alaw
 sip
  rel1xx disable
  session refresh
  header-passing
  error-passthru
  asserted-id pai
  conn-reuse
  no update-callerid
  midcall-signaling passthru
  privacy-policy passthru
  call-route p-called-party-id
  pass-thru headers unsupp
  sip-profiles inbound
  audio forced
!
voice class uri 1000 sip
 host ipv4:[CUCM Pub IP]
 host ipv4:[CUCM Sub IP]
!
voice class codec 1000
 codec preference 1 g711alaw
!
voice class sip-profiles 2001
 rule 1 request INVITE sip-header P-Asserted-Identity remove
 rule 2 request REINVITE sip-header P-Asserted-Identity remove
!
voice class sip-profiles 2000
 rule 1 request INVITE peer-header sip P-Asserted-Identity copy "sip:(.*)@" u01
 rule 2 request INVITE peer-header sip Diversion copy "sip:(.*)@" u01
 rule 3 request INVITE sip-header P-Asserted-Identity modify "sip:.*@(.*)" "sip:\u01@\1"
 rule 4 request ANY sip-header Min-SE remove
 rule 5 request ANY sip-header Diversion remove
 rule 6 request ANY sdp-header Connection-Info remove
 rule 7 request ANY sip-header User-Agent remove (optional)
 rule 8 response ANY sip-header User-Agent remove (optional)
 rule 9 request ANY sip-header Cisco-Guid remove (optional)
 rule 10 response ANY sip-header Cisco-Guid remove (optional)
!
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"
 rule 4 request ANY sip-header User-Agent remove (optional)
 rule 5 response ANY sip-header User-Agent remove (optional)
 rule 6 request ANY sip-header Cisco-Guid remove (optional)
 rule 7 response ANY sip-header Cisco-Guid remove (optional)
!
voice class sip-copylist 1000
 sip-header P-Asserted-Identity
 sip-header Diversion
!
voice class e164-pattern-map 1000
  e164 11[68]T
  e164 11[025]
  e164 +T
  e164 0T
```

```
 !
voice class e164-pattern-map 2000
  e164 +49[CLIENT PUBLIC NUMBER]T
 !
voice class server-group 1000
 ipv4 [CUCM PUB IP] preference 2
 ipv4 [CUCM SUB IP] preference 1
 description ### CUCM Server Group ###
!
voice class sip-options-keepalive 1000
 up-interval 30
 retry 3
!
voice class tenant 1000
  no remote-party-id
  timers buffer-invite 5000
  session transport tcp
  session refresh
  header-passing
  error-passthru
  bind control source-interface [LAN INTERFACE]
  bind media source-interface [LAN INTERFACE]
  no pass-thru content custom-sdp
  privacy-policy passthru
!
voice class tenant 2000
 registrar dns:sip-trunk.telekom.de expires 240 tcp auth-realm sip-trunk.telekom.de
 credentials number [TELEKOM REGISTRATION NUMBER] username [TELEKOM REGISTRATION ID] password [TELEKOM REGISTRATION PASSWORD] realm sip-trunk.telekom.de
 authentication username [TELEKOM REGISTRATION ID] password [TELEKOM REGISTRATION PASSWORD]
 no remote-party-id
 timers buffer-invite 5000
 timers dns registrar-cache ttl
 sip-server dns:sip-trunk.telekom.de
 session transport tcp
 session refresh
 no update-callerid
 header-passing
 error-passthru
 bind control source-interface [WAN INTERFACE]
 bind media source-interface [WAN INTERFACE]
 asserted-id pai
 no pass-thru content custom-sdp
 conn-reuse
 sip-profiles 3000
 outbound-proxy dns:reg.sip-trunk.telekom.de
 privacy-policy passthru
!
dial-peer voice 1000 voip
 description ### From / To CUCM ###
 huntstop
 session protocol sipv2
 session server-group 1000
 destination e164-pattern-map 2000
 incoming uri via 1000
 voice-class codec 1000
 voice-class sip tenant 1000
 voice-class sip block 183 sdp present
 voice-class sip options-keepalive profile 1000
 voice-class sip copy-list 1000
 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 2000 voip
 description ### From / To SIP-Provider DTAG ###
 huntstop
 session protocol sipv2
 session target sip-server
 destination e164-pattern-map 1000
 incoming called e164-pattern-map 2000
 voice-class codec 1000
 voice-class sip profiles 2000
 voice-class sip profiles 2001 inbound
 voice-class sip tenant 2000
 voice-class sip block 183 sdp present
 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
```

## Access Liste for WAN-Interface

```
ip access-list extended FROM-DT-TO-CUBE
 remark ### Permitted ISP Public address Range ###
 permit udp 217.0.0.0 0.0.255.255 range 1025 65534 any
 permit tcp 217.0.0.0 0.0.255.255 any eq 5060
 permit tcp 217.0.0.0 0.0.255.255 any established
 remark ### Deutsch Telekom DNS ###
 permit udp host 217.0.43.145 eq domain any
 permit udp host 217.0.43.129 eq domain any
 remark ### Permitted PING from inside to outside ###
 permit icmp any any echo-reply
 remark ### Deny all Other traffic ###
 deny ip any any
!
interface [WAN-Interface]
 ip access-group FROM-DT-TO-CUBE in
```

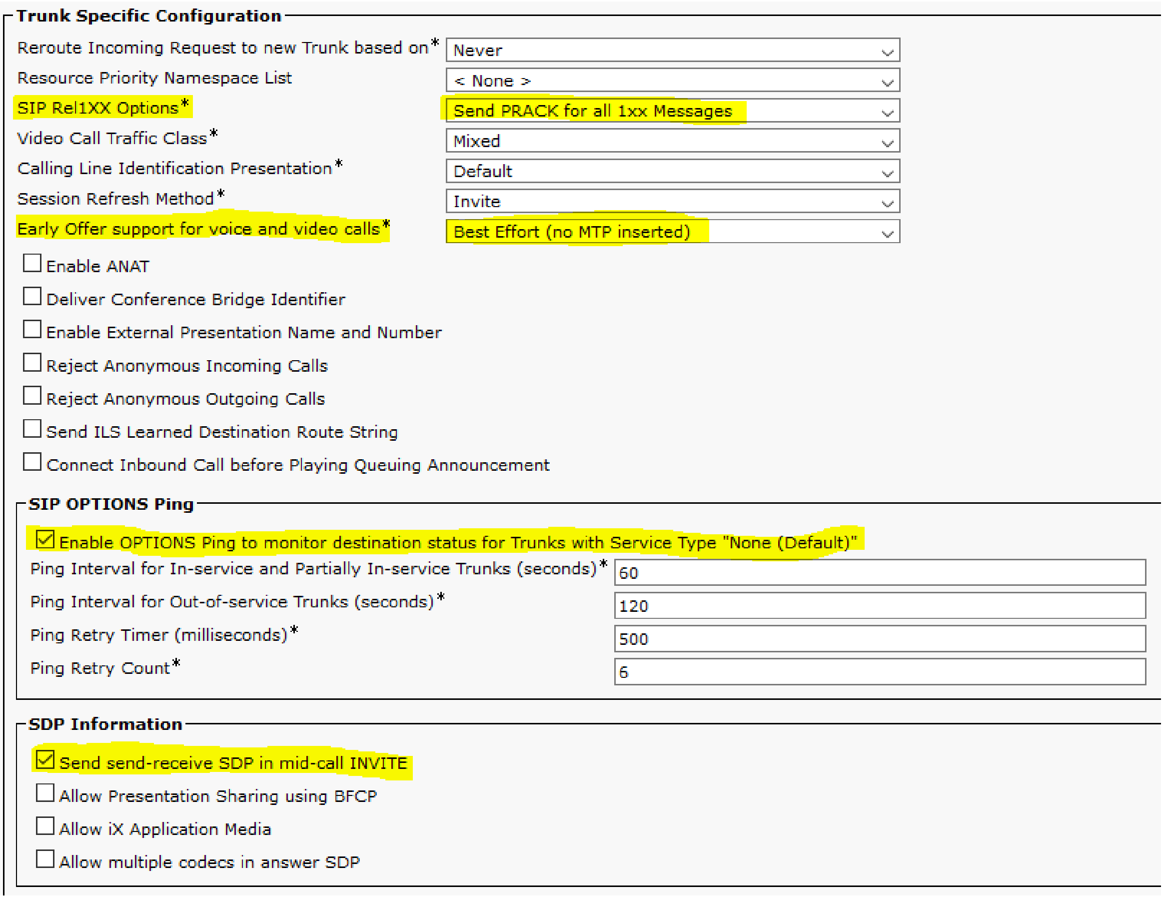
# CUCM Konfiguration

Dienstag, 22. September 2020 09:09

**SIP Profile:**

--> Copy of Standard SIP Profile

- SIP Rel1xx Options: Send PRAK for All 1xx Messages
- Early Offer Support for Voice and Video Calls: Best Effort (No MTP inserted)
- SIP Options Ping: On (optional)
- SDP Information: Send Send-receive SDP in mid-Call Invite




**SIP Trunk Security Profile:**

--> Copy of Non Secure SIP Trunk Profile

**SIP Trunk:**

- [x] PSTN Access
- [x] Run on all active Unified CM Nodes
- [ ] Remote Party id: OFF
- [x] Asserted-Identity: ON
  Asserted-Type: PAI
- [x] Redirecting Diversion Header Delivery - Outbound
  SIP Trunk Security Profile: select the added on from above
  SIP Profile: select the added on from above

☑ PSTN Access
☑ Run On All Active Unified CM Nodes

**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
Trust Received Identity*: Trust All (Default)

**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

**Presentation Information**

☐ Anonymous Presentation
Presentation Number:
Presentation Name:
☐ Send Presentation Name and Number only in the FROM header and not in the other identity headers

**SIP Information**

**Destination**

☐ Destination Address is an SRV

| | Destination Address | Destination Address IPv6 | Destination Port |
|---|---|---|---|
| 1* | | | 5060 |

MTP Preferred Originating Codec*: 711ulaw
BLF Presence Group*: Standard Presence group
SIP Trunk Security Profile*: -- Not Selected --
Rerouting Calling Search Space: < None >
Out-Of-Dialog Refer Calling Search Space: < None >
SUBSCRIBE Calling Search Space: < None >
SIP Profile*: -- Not Selected -- View Details
DTMF Signaling Method*: No Preference

**Service Parameter:**

--> CUCM-Node --> Service "Cisco Callmanager":

- Duplex Streaming Enabled => True

**Clusterwide Parameters (Service)**

| | |
|---|---|
| Default Network Hold MOH Audio Source ID * | 1 |
| Default User Hold MOH Audio Source ID * | 1 |
| Duplex Streaming Enabled * | True |
| Media Exchange Interface Capability Timer * | |

--> CUCM-Node --> Service "Cisco IP Voice Media Streaming App":

- Supported MOH Codecs: 711 alaw should be marked

**Clusterwide Parameters (Parameters that apply to all servers)**

| | |
|---|---|
| Supported MOH Codecs * | 711 mulaw<br>711 alaw<br>729 Annex A |

# CUC Konfiguration

Donnerstag, 14. Januar 2021 09:48

**<u>Telephony Integrations:</u>**

--> Check Port Group, if in "G.711a-law" is select in "Codec Advertising"