# Technical Report 1314

Certification Test of NFON “Nconnect Voice 2.0”

SIP Trunk with OpenScape 4000 and OpenScape SBC



Version 1.0 – 7th March 2022



Customer Solution Lab

# Table of Contents

# 1. Document Overview

## 1.1. Executive Summary

This document describes the test setup and test results between NFON Nconnect Voice 2.0 and OpenScape 4000/OpenScape SBC, carried out at customer Raps GmbH & Co. KG in Germany. The SIP Trunk was provided by the German company Deutsche Telefon Standard (DTS), which belongs to the German telephone company NFON.

The certification tests were passed for the devices specified in section 4.2 Hardware/Software releases.

**This certification is valid as long as no changes are made on the SIP interface of NFON.**

It was not tested:

- Emergency calls
- Fax transmission

## 1.2. Document Control

### 1.2.1. Authors of the Document

| Name | Company - Department |
|---|---|
| Rolf Lang | Unify Communications and Collaboration GmbH & Co. KG – UCaC GER PRA CCS PS C&I 4 |

Only the individual listed above is authorized to make changes to the document.

### 1.2.2. Version / Changes

| Date | Version | Author | Remarks |
|---|---|---|---|
| 10th January 2022 | 0.1 | Rolf Lang | Initial structure |
| 11th January 2022 | 0.2 | Rolf Lang | Draft |
| 7rd February 2022 | 0.9 | Rolf Lang | Ready for review |
| 7rd March 2022 | 1.0 | Rolf Lang | Final |

### 1.2.3. Reference to other Documents

| Title | File Name/Comment |
|---|---|
| NFON<br>Technisches Handbuch<br>Nconnect Voice 2.0 | https://www.nfon.com/media/Service/Documentation/Manuals/Nconnect_Voice_2.0/<br>Handbuch_Nconnect_Voice_2.0.pdf |
| SIP-PBX / Service Provider Interoperability<br><br>"SIPconnect 2.0 Technical Recommendation" | SIPconnect_2.0_FINAL.pdf<br><br>Download link:<br>https://www.sipforum.org/download/sipconnect-technical-recommendation-version-2-0-2/?wpdmdl=2819&refresh=61fae982dba001643833730 |

### 1.2.4. Contact List

| Name | Company – Job Title | Phone | E-Mail |
|---|---|---|---|
| Rolf Lang | Unify Deutschland GmbH & Co KGSolution Integrator | +49 89 7007 20823 | rolf.lang@atos.net |
| Axel Braun | Deutsche Telefon Standard GmbH<br>Sales Consultant | +49 6131 32797-60 | axel.braun@deutsche-telefon.de |
| | | | |

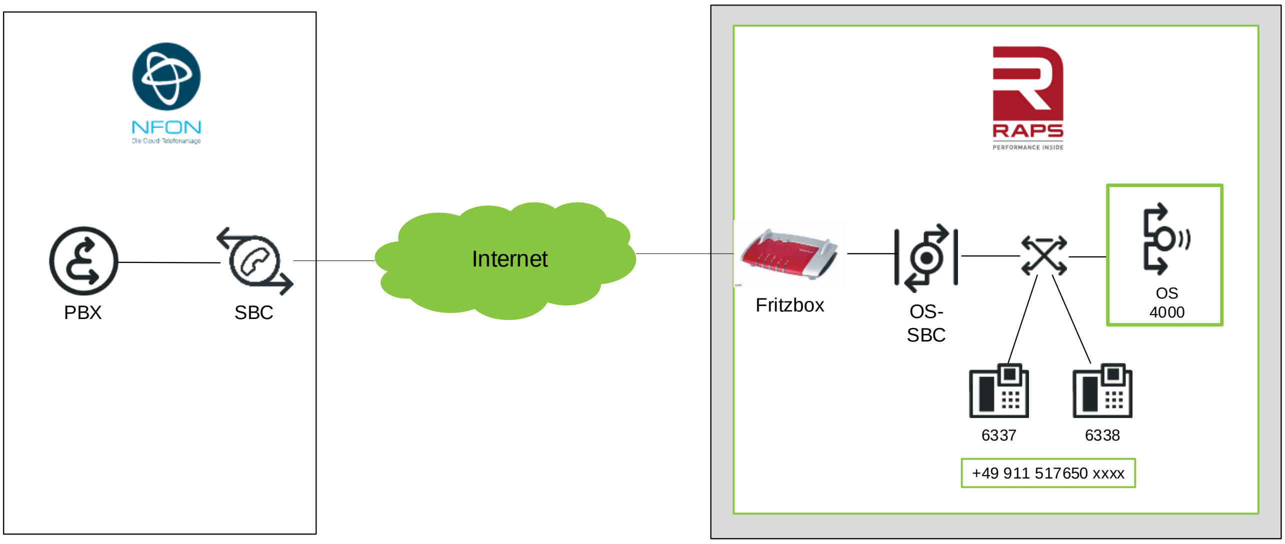
# 2. Issues & Limitations

## 2.1. Issues

Below the list of issues risen during this certification test:

| Description |
|---|
| For an anonymous incoming call from NFON was specified in the Form header in Invite:<br>sip:anonymous@nfon.net<br>According to SIPconnect 2.0 specification NFON must send instead:<br>sip:anonymous@anonymous.invalid |

# 3. General Plan



# 4. Test Setup

## 4.1. Test Numbers

| | |
|---|---|
| NFON test number 1 | +49 (911) 517650 6337 |
| NFON test number 2 | +49 (911) 517650 6338 |
| PSTN landline phone | +49 (89) 7007 20823 |
| PSTN mobile phone | +49 (171) 7600644 |

## 4.2. Hardware/Software releases

| HW / SW | SW / Release |
|---|---|
| OpenScape 4000 | V10 R0.28.6 |
| Unify OpenScape Hosted SBC on Ecoserver | V10 R1.04.05 |
| Unify OpenScape Desk Phone CP400 HFA | V1 R5.5.0 |
| Unify Openstage 40 | V3 R0.48.0 |

## 4.3. Network Integration Data

IP Addresses used:

| IP address | Remark |
|---|---|
| 10.51.110.150 | OpenScape 4000 SIP IP |
| 10.51.110.151 | OpenScape SBC LAN IP |
| 192.168.178.5 | OpenScape SBC WAN IP |
| 109.68.96.123 | NFON IP |

# 5. Configuration

The NFON “Nconnect Voice 2.0” SIP Trunk is based on the the SIPconnect 2.0 Technical Recommendation. The SBC is configured accordingly.

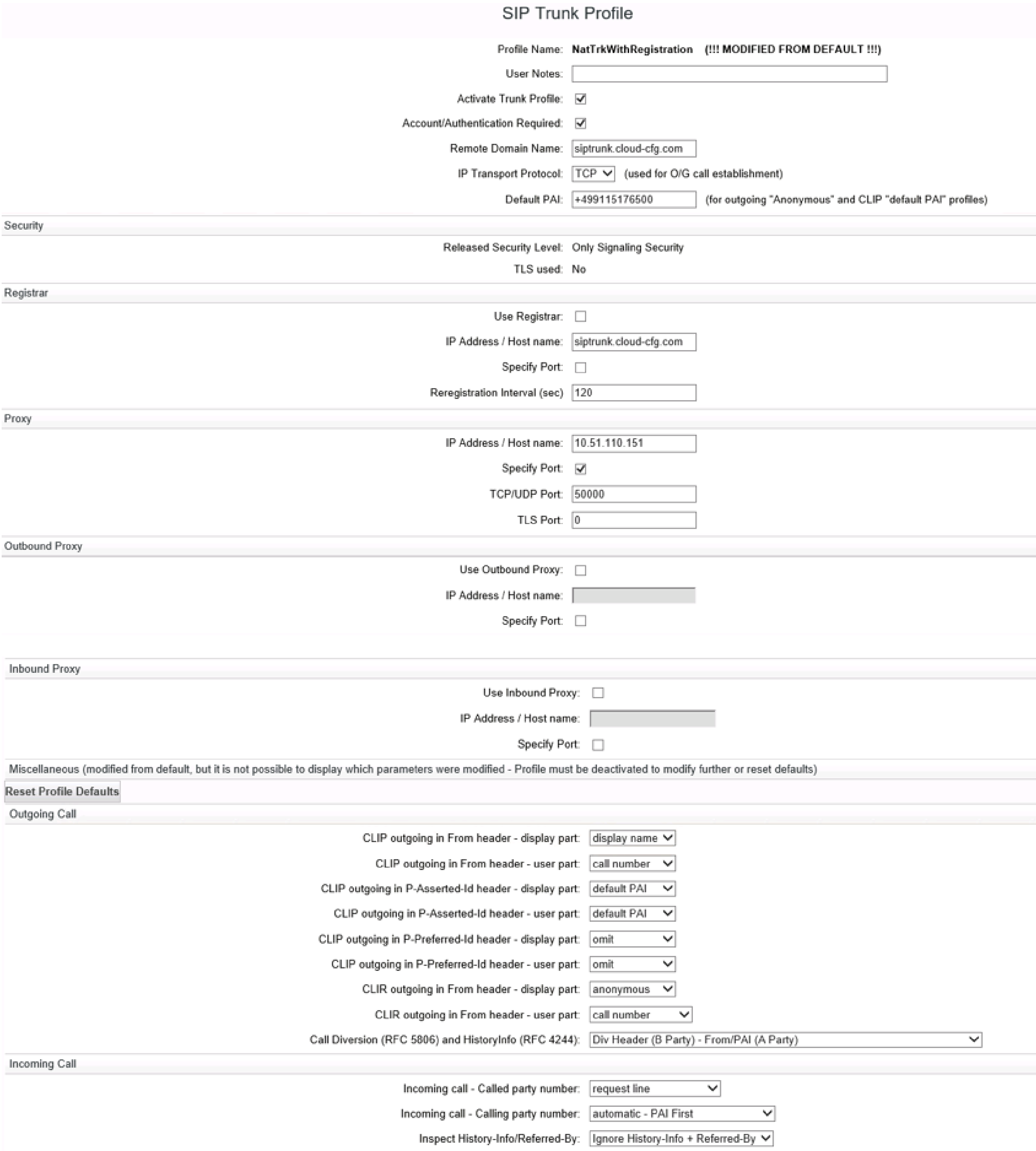
SIP Service Provider NFON

- expect to receive the phone number to be authenticated in the P-Asserted-Identity header
- expect to receive the calling number to be display on the called phone in the FROM header

## 5.1. Configuration of OpenScape 4000

For the test it was configured on OS 4000 the prefix 68 to route outgoing calls via SBC over the NFON SIP Trunk.

In OS 4000 was deactivated the SIP Session Timer according to the NFON recommendation.



SIP Trunk Profile

| | |
|---|---|
| Profile Name: | **NatTrkWithRegistration (!!! MODIFIED FROM DEFAULT !!!)** |
| User Notes: | |
| Activate Trunk Profile: | ☑ |
| Account/Authentication Required: | ☑ |
| Remote Domain Name: | siptrunk.cloud-cfg.com |
| IP Transport Protocol: | TCP (used for O/G call establishment) |
| Default PAI: | +499115176500 (for outgoing "Anonymous" and CLIP "default PAI" profiles) |
| **Security** | |
| Released Security Level: | Only Signaling Security |
| TLS used: | No |
| **Registrar** | |
| Use Registrar: | ☐ |
| IP Address / Host name: | siptrunk.cloud-cfg.com |
| Specify Port: | ☐ |
| Reregistration Interval (sec) | 120 |
| **Proxy** | |
| IP Address / Host name: | 10.51.110.151 |
| Specify Port: | ☑ |
| TCP/UDP Port: | 50000 |
| TLS Port: | 0 |
| **Outbound Proxy** | |
| Use Outbound Proxy: | ☐ |
| IP Address / Host name: | |
| Specify Port: | ☐ |
| **Inbound Proxy** | |
| Use Inbound Proxy: | ☐ |
| IP Address / Host name: | |
| Specify Port: | ☐ |

Miscellaneous (modified from default, but it is not possible to display which parameters were modified - Profile must be deactivated to modify further or reset defaults)

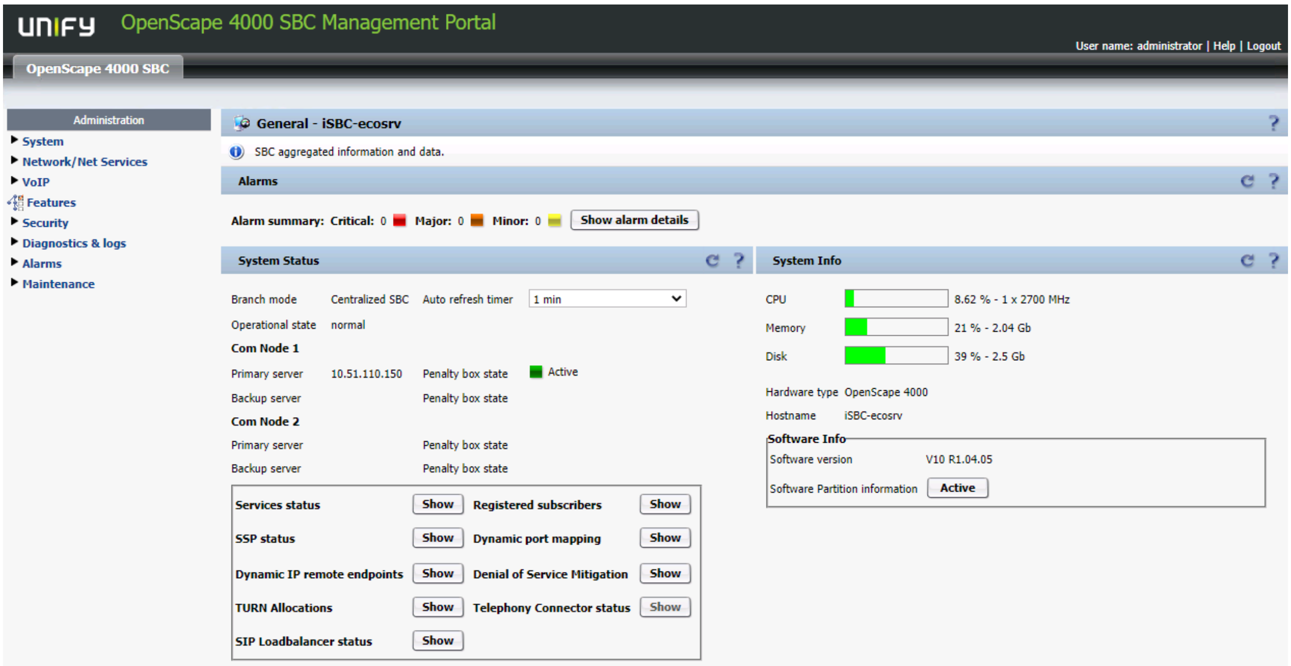
Reset Profile Defaults

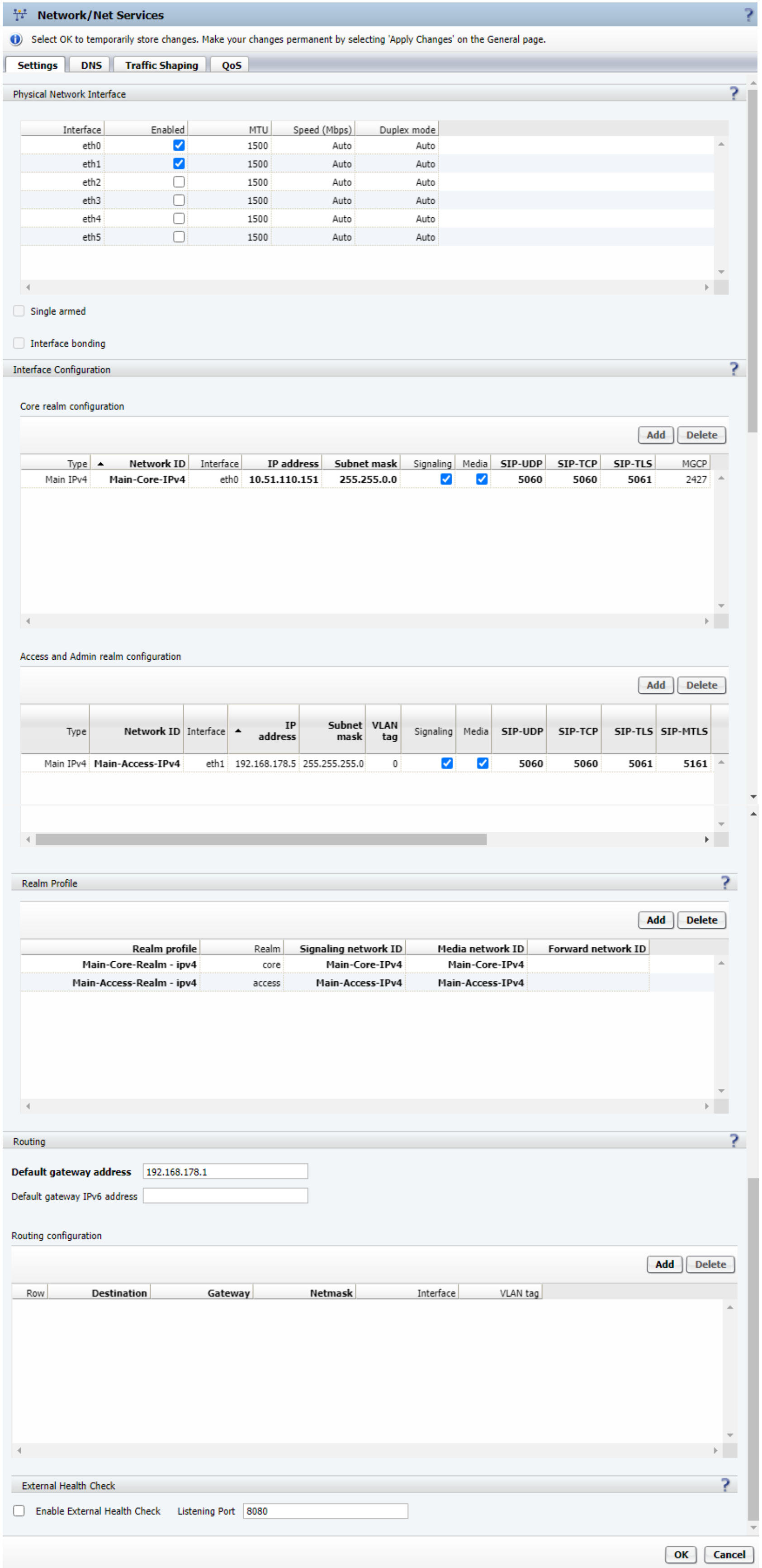
| | |
|---|---|
| **Outgoing Call** | |
| CLIP outgoing in From header - display part: | display name |
| CLIP outgoing in From header - user part: | call number |
| CLIP outgoing in P-Asserted-Id header - display part: | default PAI |
| CLIP outgoing in P-Asserted-Id header - user part: | default PAI |
| CLIP outgoing in P-Preferred-Id header - display part: | omit |
| CLIP outgoing in P-Preferred-Id header - user part: | omit |
| CLIR outgoing in From header - display part: | anonymous |
| CLIR outgoing in From header - user part: | call number |
| Call Diversion (RFC 5806) and HistoryInfo (RFC 4244): | Div Header (B Party) - From/PAI (A Party) |
| **Incoming Call** | |
| Incoming call - Called party number: | request line |
| Incoming call - Calling party number: | automatic - PAI First |
| Inspect History-Info/Referred-By: | Ignore History-Info + Referred-By |

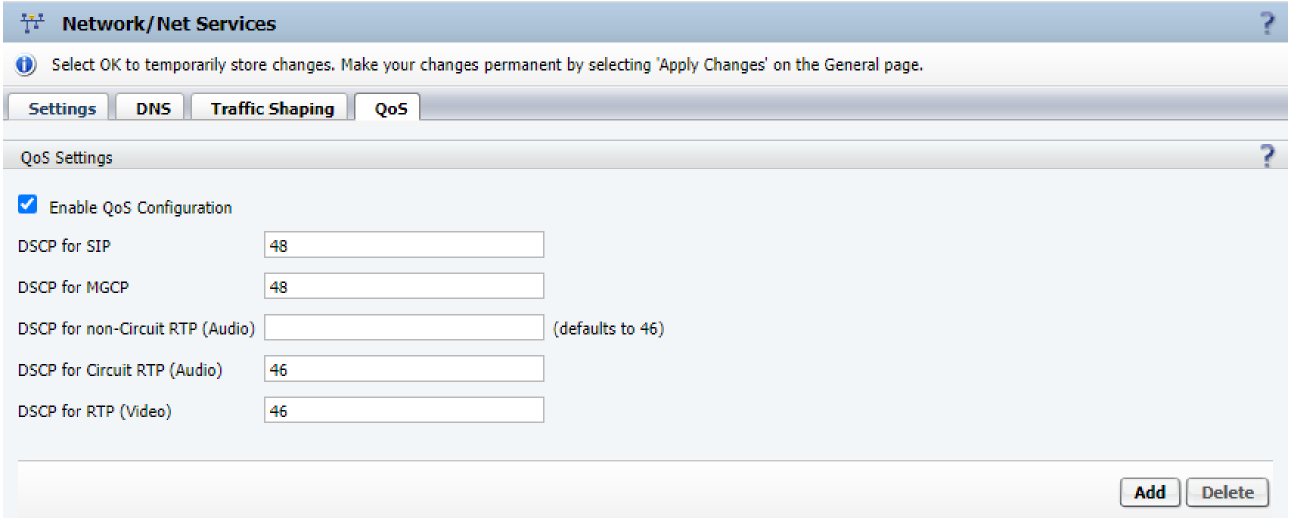
2021_10_21_ALL.SAMTXT

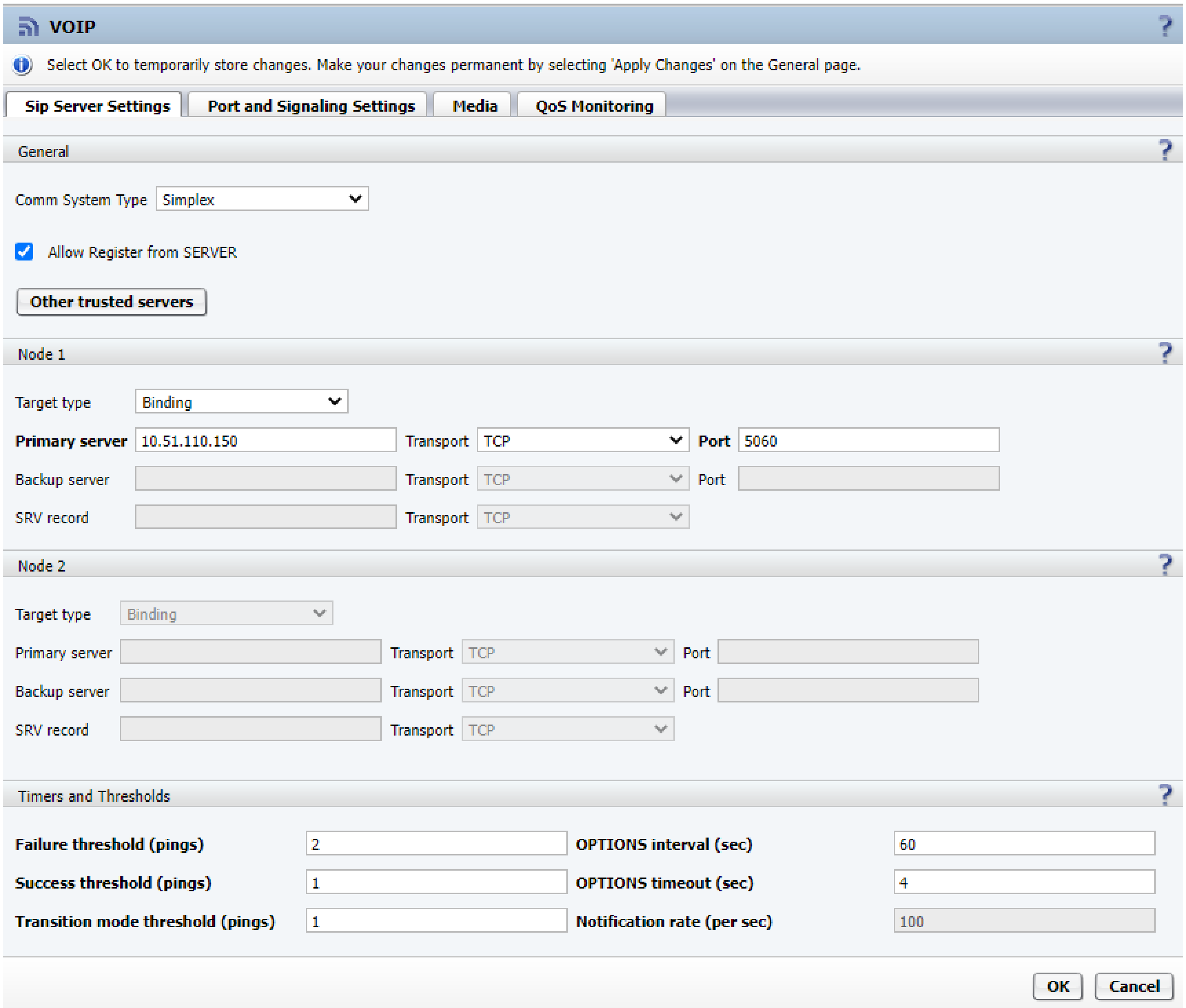
## 5.2. Configuration of OpenScape SBC

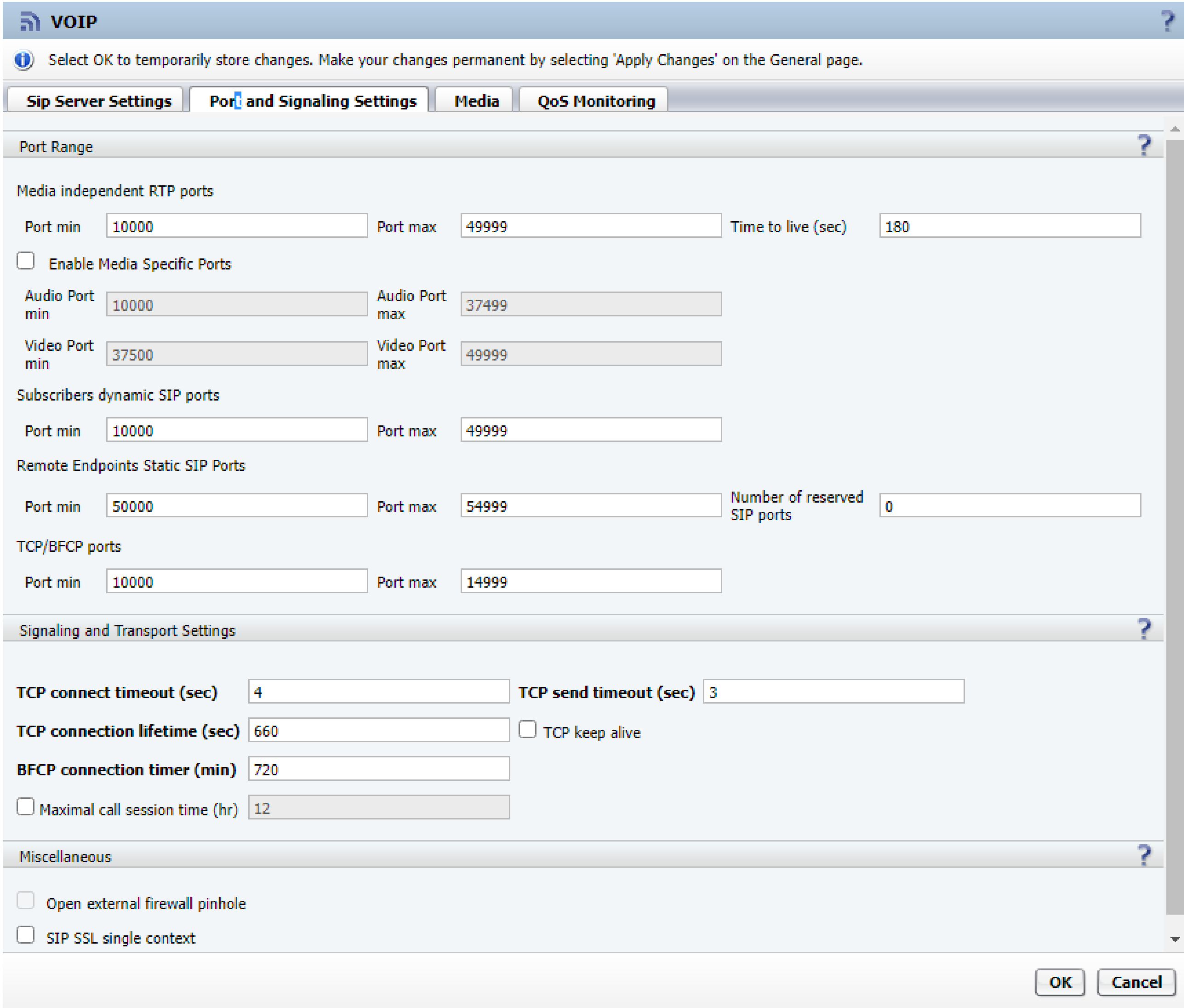
- NFON expect to receive registration requests every 120 seconds

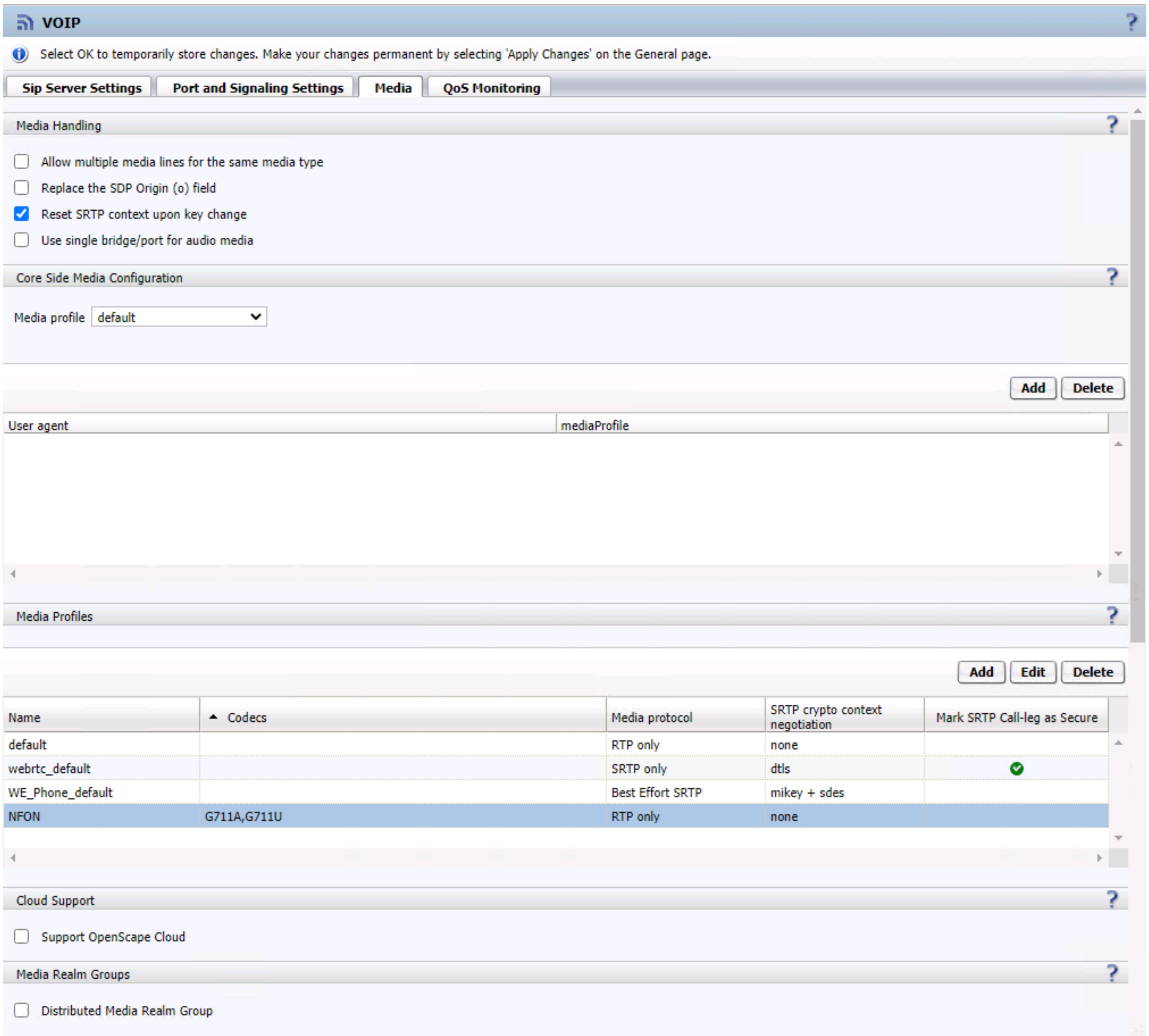


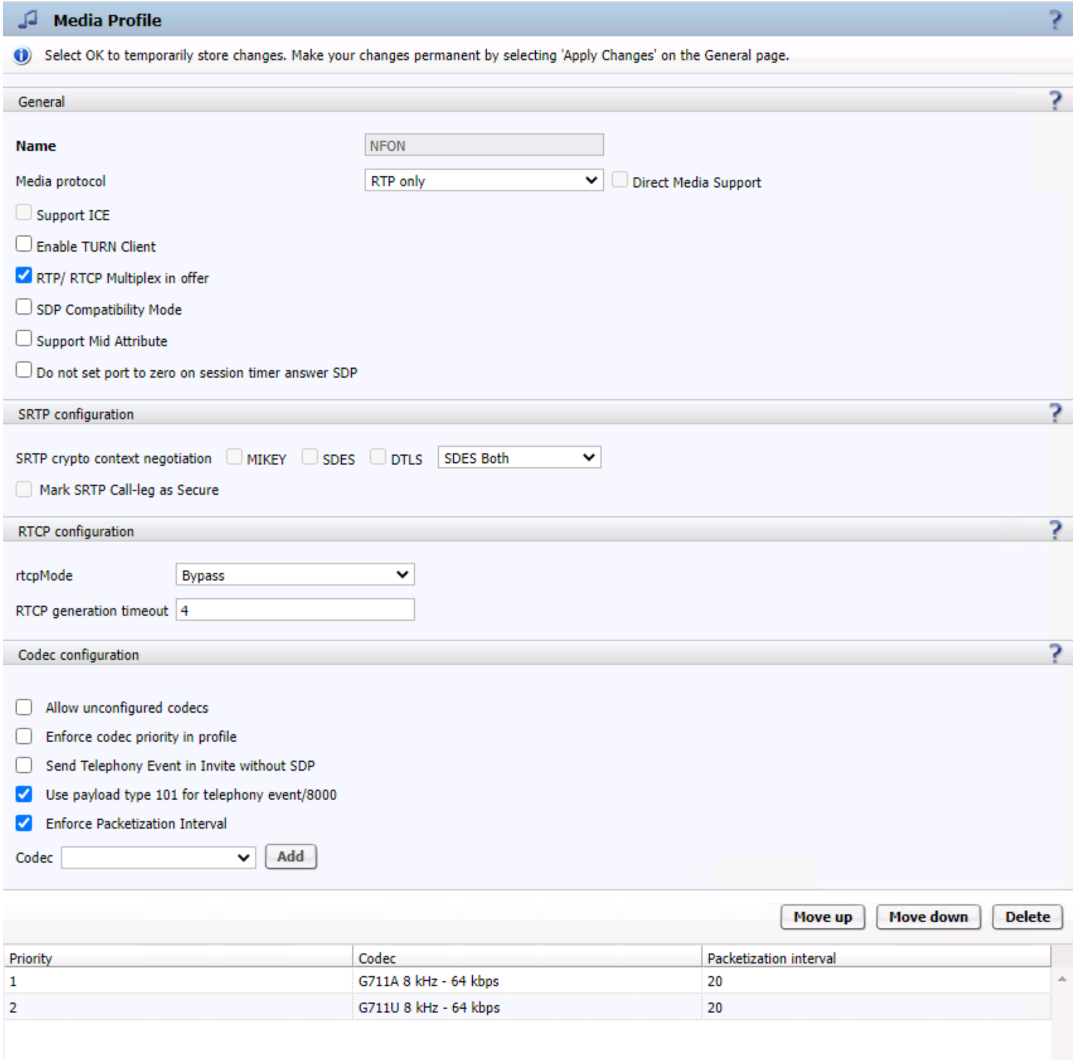

Network/Net Services
Select OK to temporarily store changes. Make your changes permanent by selecting 'Apply Changes' on the General page.
Settings
DNS
Traffic Shaping
QoS
Physical Network Interface
Interface Enabled MTU Speed (Mbps) Duplex mode
eth0 1500 Auto Auto
eth1 1500 Auto Auto
eth2 1500 Auto Auto
eth3 1500 Auto Auto
eth4 1500 Auto Auto
eth5 1500 Auto Auto
Single armed
Interface bonding
Interface Configuration
Core realm configuration
Add
Delete
Type Network ID Interface IP address Subnet mask Signaling Media SIP-UDP SIP-TCP SIP-TLS MGCP
Main IPv4 Main-Core-IPv4 eth0 10.51.110.151 255.255.0.0 5060 5060 5061 2427
Access and Admin realm configuration
Add
Delete
Type Network ID Interface IP address Subnet mask VLAN tag Signaling Media SIP-UDP SIP-TCP SIP-TLS SIP-MTLS
Main IPv4 Main-Access-IPv4 eth1 192.168.178.5 255.255.255.0 0 5060 5060 5061 5161
Realm Profile
Add
Delete
Realm profile Realm Signaling network ID Media network ID Forward network ID
Main-Core-Realm - ipv4 core Main-Core-IPv4 Main-Core-IPv4
Main-Access-Realm - ipv4 access Main-Access-IPv4 Main-Access-IPv4
Routing
Default gateway address 192.168.178.1
Default gateway IPv6 address
Routing configuration
Add
Delete
Row Destination Gateway Netmask Interface VLAN tag
External Health Check
Enable External Health Check
Listening Port 8080
OK
Cancel

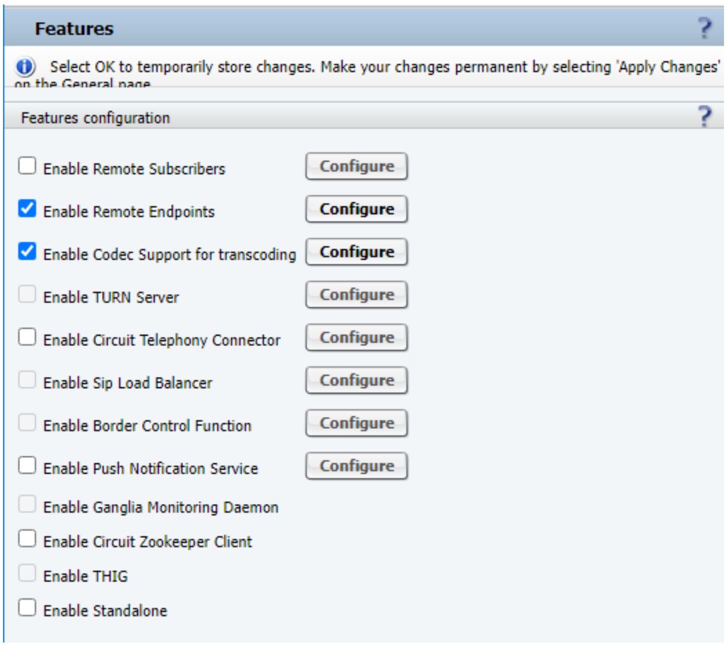

Network/Net Services
Select OK to temporarily store changes. Make your changes permanent by selecting 'Apply Changes' on the General page.
Settings
DNS
Traffic Shaping
QoS
QoS Settings
Enable QoS Configuration
DSCP for SIP
48
DSCP for MGCP
48
DSCP for non-Circuit RTP (Audio)
(defaults to 46)
DSCP for Circuit RTP (Audio)
46
DSCP for RTP (Video)
46
Add
Delete

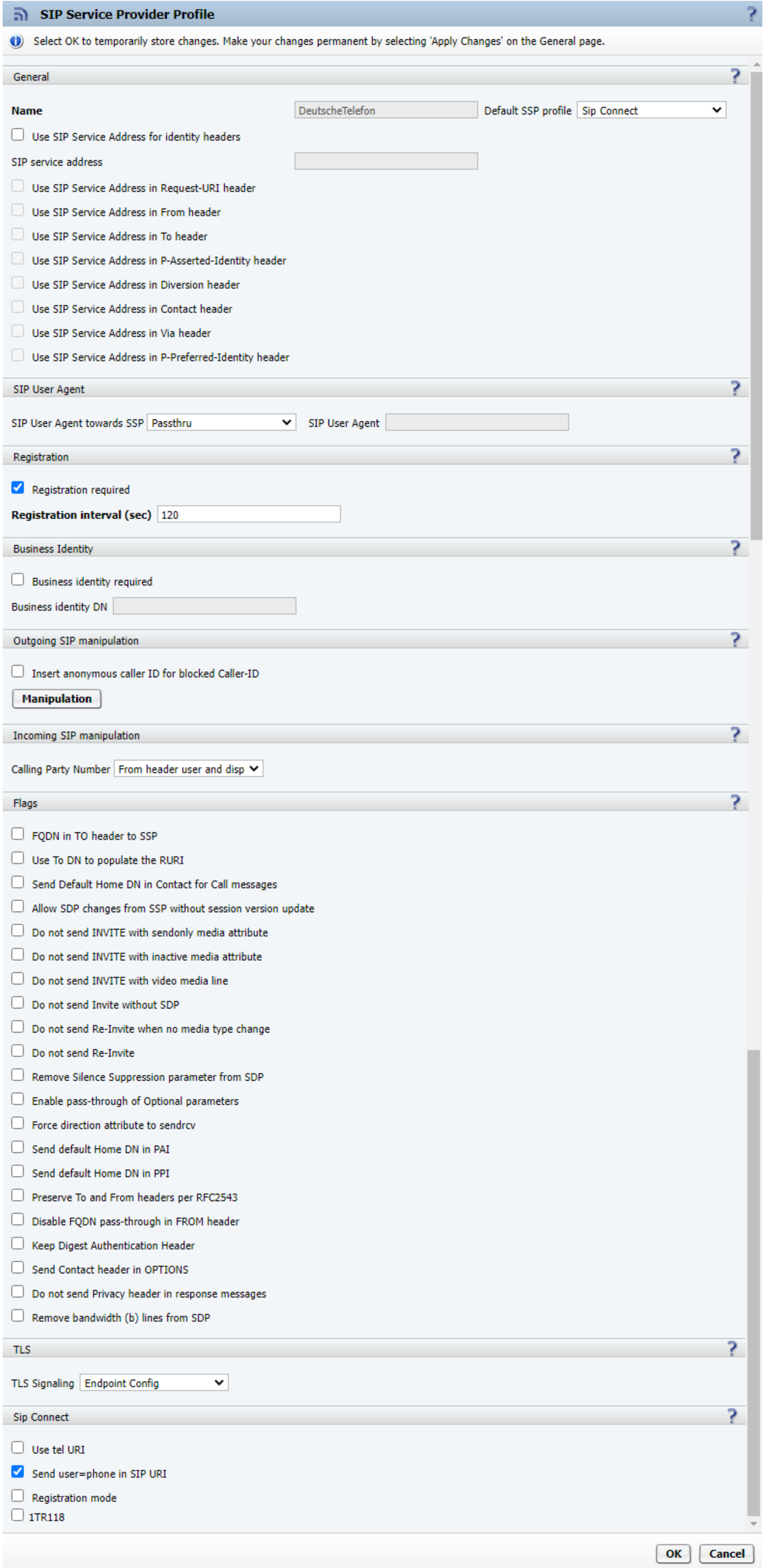

VOIP
Select OK to temporarily store changes. Make your changes permanent by selecting 'Apply Changes' on the General page.
Sip Server Settings
Port and Signaling Settings
Media
QoS Monitoring
General
Comm System Type
Simplex
Allow Register from SERVER
Other trusted servers
Node 1
Target type
Binding
Primary server
10.51.110.150
Transport
TCP
Port
5060
Backup server
Transport
TCP
Port
SRV record
Transport
TCP
Node 2
Target type
Binding
Primary server
Transport
TCP
Port
Backup server
Transport
TCP
Port
SRV record
Transport
TCP
Timers and Thresholds
Failure threshold (pings)
2
OPTIONS interval (sec)
60
Success threshold (pings)
1
OPTIONS timeout (sec)
4
Transition mode threshold (pings)
1
Notification rate (per sec)
100
OK
Cancel

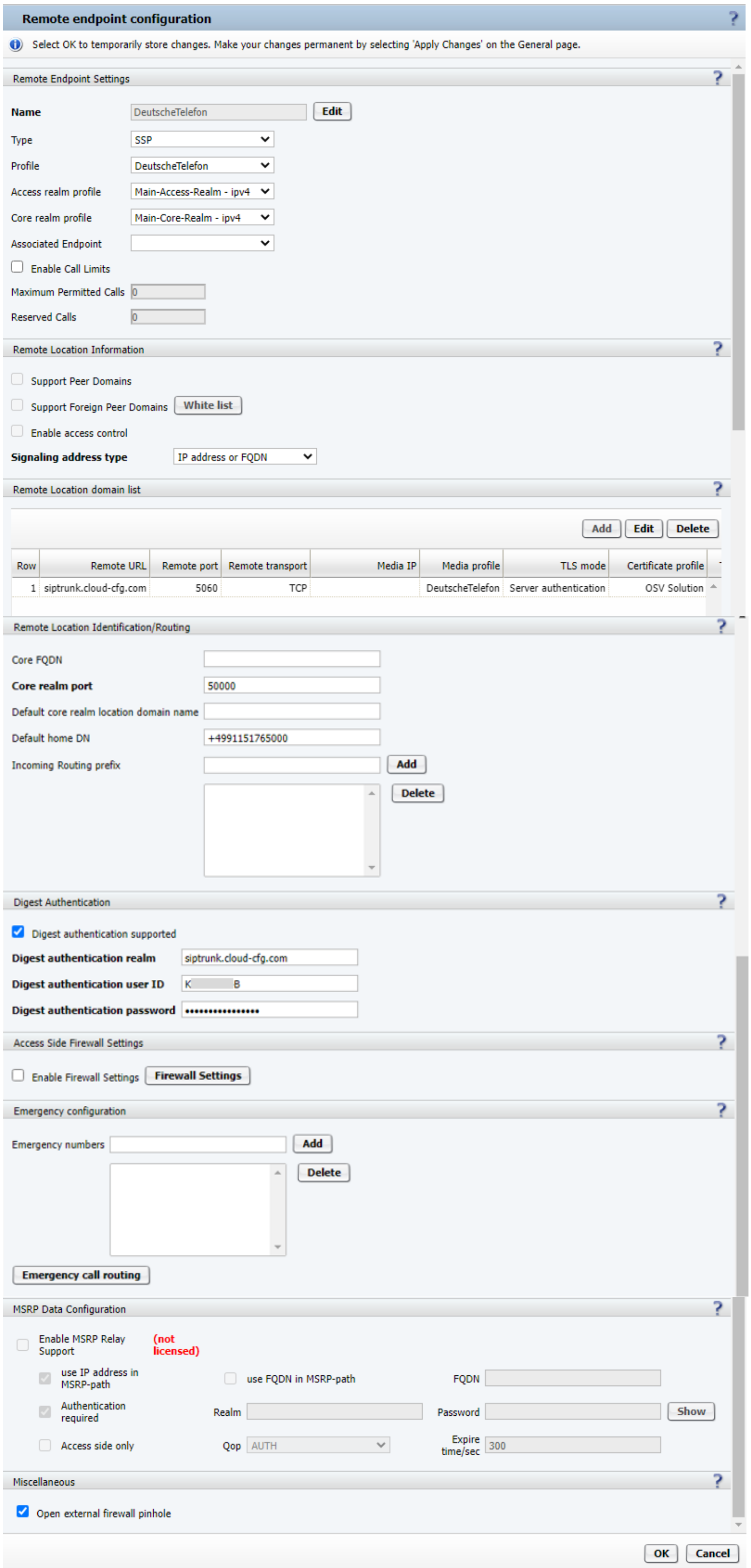

VOIP
Select OK to temporarily store changes. Make your changes permanent by selecting 'Apply Changes' on the General page.
Sip Server Settings
Port and Signaling Settings
Media
QoS Monitoring
Port Range
Media independent RTP ports
Port min 10000
Port max 49999
Time to live (sec) 180
Enable Media Specific Ports
Audio Port min 10000
Audio Port max 37499
Video Port min 37500
Video Port max 49999
Subscribers dynamic SIP ports
Port min 10000
Port max 49999
Remote Endpoints Static SIP Ports
Port min 50000
Port max 54999
Number of reserved SIP ports 0
TCP/BFCP ports
Port min 10000
Port max 14999
Signaling and Transport Settings
TCP connect timeout (sec) 4
TCP send timeout (sec) 3
TCP connection lifetime (sec) 660
TCP keep alive
BFCP connection timer (min) 720
Maximal call session time (hr) 12
Miscellaneous
Open external firewall pinhole
SIP SSL single context
OK
Cancel

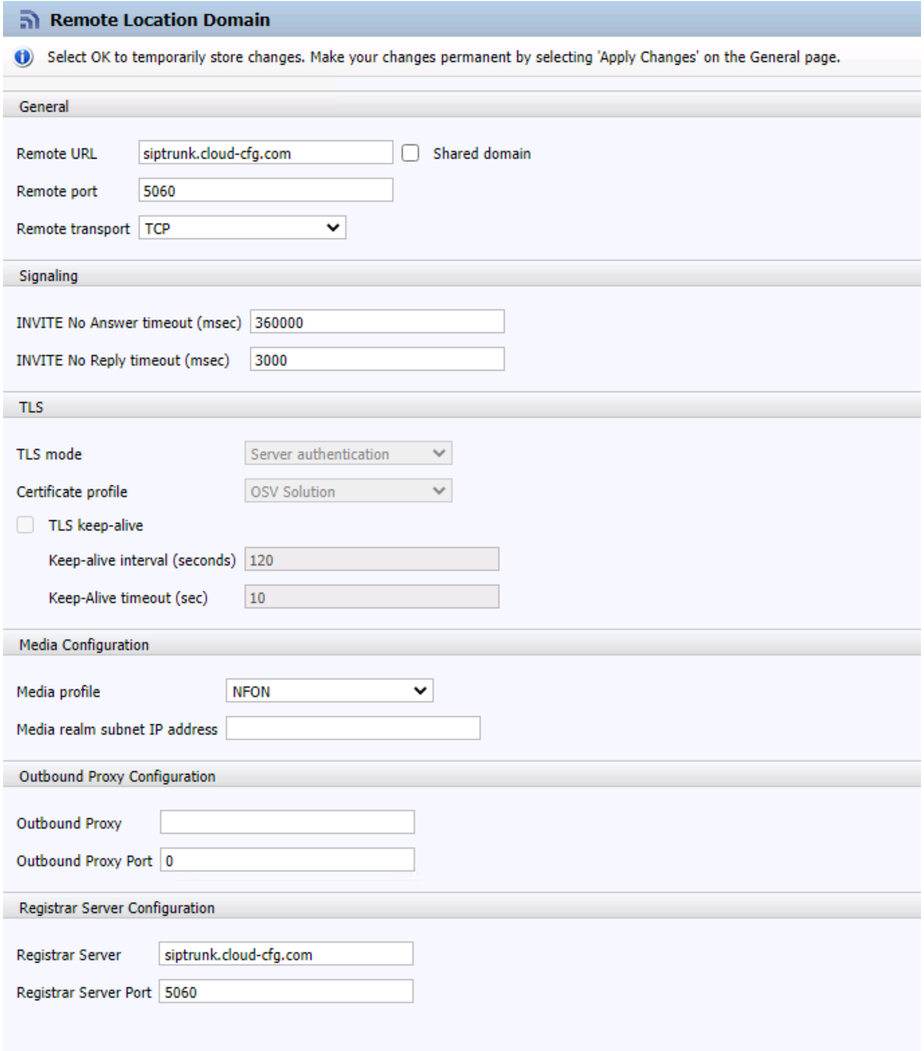

VOIP
Select OK to temporarily store changes. Make your changes permanent by selecting 'Apply Changes' on the General page.
Sip Server Settings
Port and Signaling Settings
Media
QoS Monitoring
Media Handling
Allow multiple media lines for the same media type
Replace the SDP Origin (o) field
Reset SRTP context upon key change
Use single bridge/port for audio media
Core Side Media Configuration
Media profile
default
Add
Delete
User agent
mediaProfile
Media Profiles
Add
Edit
Delete
Name
Codecs
Media protocol
SRTP crypto context negotiation
Mark SRTP Call-leg as Secure
default
RTP only
none
webrtc_default
SRTP only
dtls
WE_Phone_default
Best Effort SRTP
mikey + sdes
NFON
G711A,G711U
RTP only
none
Cloud Support
Support OpenScape Cloud
Media Realm Groups
Distributed Media Realm Group

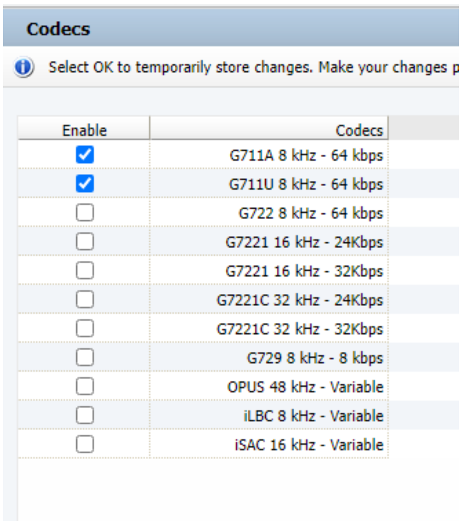

Media Profile
Select OK to temporarily store changes. Make your changes permanent by selecting 'Apply Changes' on the General page.
General
Name
NFON
Media protocol
RTP only
Direct Media Support
Support ICE
Enable TURN Client
RTP/ RTCP Multiplex in offer
SDP Compatibility Mode
Support Mid Attribute
Do not set port to zero on session timer answer SDP
SRTP configuration
SRTP crypto context negotiation
MIKEY
SDES
DTLS
SDES Both
Mark SRTP Call-leg as Secure
RTCP configuration
rtcpMode
Bypass
RTCP generation timeout
4
Codec configuration
Allow unconfigured codecs
Enforce codec priority in profile
Send Telephony Event in Invite without SDP
Use payload type 101 for telephony event/8000
Enforce Packetization Interval
Codec
Add
Move up
Move down
Delete
Priority
Codec
Packetization interval
1
G711A 8 kHz - 64 kbps
20
2
G711U 8 kHz - 64 kbps
20

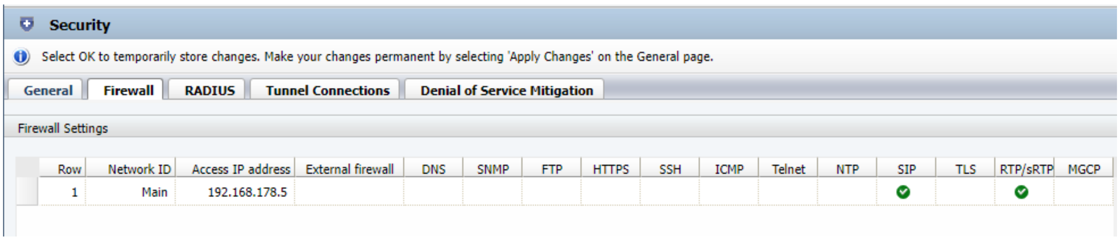

Features
Select OK to temporarily store changes. Make your changes permanent by selecting 'Apply Changes' on the General page
Features configuration
Enable Remote Subscribers
Configure
Enable Remote Endpoints
Configure
Enable Codec Support for transcoding
Configure
Enable TURN Server
Configure
Enable Circuit Telephony Connector
Configure
Enable Sip Load Balancer
Configure
Enable Border Control Function
Configure
Enable Push Notification Service
Configure
Enable Ganglia Monitoring Daemon
Enable Circuit Zookeeper Client
Enable THIG
Enable Standalone

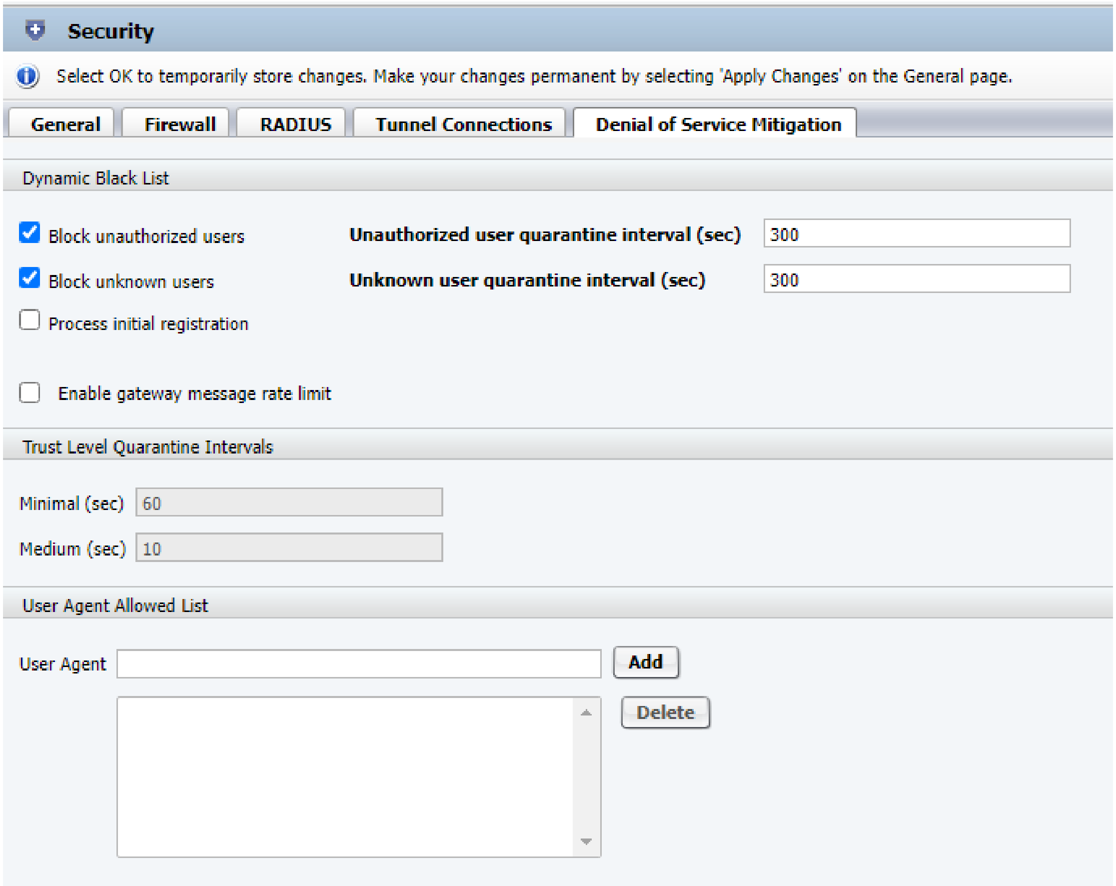

SIP Service Provider Profile
Select OK to temporarily store changes. Make your changes permanent by selecting 'Apply Changes' on the General page.
General
Name
DeutscheTelefon
Default SSP profile
Sip Connect
Use SIP Service Address for identity headers
SIP service address
Use SIP Service Address in Request-URI header
Use SIP Service Address in From header
Use SIP Service Address in To header
Use SIP Service Address in P-Asserted-Identity header
Use SIP Service Address in Diversion header
Use SIP Service Address in Contact header
Use SIP Service Address in Via header
Use SIP Service Address in P-Preferred-Identity header
SIP User Agent
SIP User Agent towards SSP
Passthru
SIP User Agent
Registration
Registration required
Registration interval (sec)
120
Business Identity
Business identity required
Business identity DN
Outgoing SIP manipulation
Insert anonymous caller ID for blocked Caller-ID
Manipulation
Incoming SIP manipulation
Calling Party Number
From header user and disp
Flags
FQDN in TO header to SSP
Use To DN to populate the RURI
Send Default Home DN in Contact for Call messages
Allow SDP changes from SSP without session version update
Do not send INVITE with sendonly media attribute
Do not send INVITE with inactive media attribute
Do not send INVITE with video media line
Do not send Invite without SDP
Do not send Re-Invite when no media type change
Do not send Re-Invite
Remove Silence Suppression parameter from SDP
Enable pass-through of Optional parameters
Force direction attribute to sendrcv
Send default Home DN in PAI
Send default Home DN in PPI
Preserve To and From headers per RFC2543
Disable FQDN pass-through in FROM header
Keep Digest Authentication Header
Send Contact header in OPTIONS
Do not send Privacy header in response messages
Remove bandwidth (b) lines from SDP
TLS
TLS Signaling
Endpoint Config
Sip Connect
Use tel URI
Send user=phone in SIP URI
Registration mode
1TR118
OK
Cancel


Remote endpoint configuration
Select OK to temporarily store changes. Make your changes permanent by selecting 'Apply Changes' on the General page.
Remote Endpoint Settings
Name
DeutscheTelefon
Edit
Type
SSP
Profile
DeutscheTelefon
Access realm profile
Main-Access-Realm - ipv4
Core realm profile
Main-Core-Realm - ipv4
Associated Endpoint
Enable Call Limits
Maximum Permitted Calls
0
Reserved Calls
0
Remote Location Information
Support Peer Domains
Support Foreign Peer Domains
White list
Enable access control
Signaling address type
IP address or FQDN
Remote Location domain list
Add
Edit
Delete
Row
Remote URL
Remote port
Remote transport
Media IP
Media profile
TLS mode
Certificate profile
1
siptrunk.cloud-cfg.com
5060
TCP
DeutscheTelefon
Server authentication
OSV Solution
Remote Location Identification/Routing
Core FQDN
Core realm port
50000
Default core realm location domain name
Default home DN
+4991151765000
Incoming Routing prefix
Add
Delete
Digest Authentication
Digest authentication supported
Digest authentication realm
siptrunk.cloud-cfg.com
Digest authentication user ID
K B
Digest authentication password
Access Side Firewall Settings
Enable Firewall Settings
Firewall Settings
Emergency configuration
Emergency numbers
Add
Delete
Emergency call routing
MSRP Data Configuration
Enable MSRP Relay Support
(not licensed)
use IP address in MSRP-path
use FQDN in MSRP-path
FQDN
Authentication required
Realm
Password
Show
Access side only
Qop
AUTH
Expire time/sec
300
Miscellaneous
Open external firewall pinhole
OK
Cancel


Remote Location Domain
Select OK to temporarily store changes. Make your changes permanent by selecting 'Apply Changes' on the General page.
General
Remote URL
siptrunk.cloud-cfg.com
Shared domain
Remote port
5060
Remote transport
TCP
Signaling
INVITE No Answer timeout (msec)
360000
INVITE No Reply timeout (msec)
3000
TLS
TLS mode
Server authentication
Certificate profile
OSV Solution
TLS keep-alive
Keep-alive interval (seconds)
120
Keep-Alive timeout (sec)
10
Media Configuration
Media profile
NFON
Media realm subnet IP address
Outbound Proxy Configuration
Outbound Proxy
Outbound Proxy Port
0
Registrar Server Configuration
Registrar Server
siptrunk.cloud-cfg.com
Registrar Server Port
5060


Codecs
Select OK to temporarily store changes. Make your changes p
Enable
Codecs
G711A 8 kHz - 64 kbps
G711U 8 kHz - 64 kbps
G722 8 kHz - 64 kbps
G7221 16 kHz - 24Kbps
G7221 16 kHz - 32Kbps
G7221C 32 kHz - 24Kbps
G7221C 32 kHz - 32Kbps
G729 8 kHz - 8 kbps
OPUS 48 kHz - Variable
iLBC 8 kHz - Variable
iSAC 16 kHz - Variable


Security
Select OK to temporarily store changes. Make your changes permanent by selecting 'Apply Changes' on the General page.
General
Firewall
RADIUS
Tunnel Connections
Denial of Service Mitigation
Firewall Settings
Row
Network ID
Access IP address
External firewall
DNS
SNMP
FTP
HTTPS
SSH
ICMP
Telnet
NTP
SIP
TLS
RTP/sRTP
MGCP
1
Main
192.168.178.5


Security
Select OK to temporarily store changes. Make your changes permanent by selecting 'Apply Changes' on the General page.
General
Firewall
RADIUS
Tunnel Connections
Denial of Service Mitigation
Dynamic Black List
Block unauthorized users
Unauthorized user quarantine interval (sec)
300
Block unknown users
Unknown user quarantine interval (sec)
300
Process initial registration
Enable gateway message rate limit
Trust Level Quarantine Intervals
Minimal (sec)
60
Medium (sec)
10
User Agent Allowed List
User Agent
Add
Delete

# 6. Test List

| **Test Result List v2.4** | | | |
|---|---|---|---|
| date: | 01.02.2022 | | |
| provider and country of account: | NFON, Germany | | |
| user name of account: | KCJILR7A4B | | |
| supported transport protocol(s): | UDP and TCP | | |
| supported security mode: | no security | | |
| account number and extensions: | +49 911 517650 xxxx | | |
| OpenScape 4000 version: | V10 R0.28.6 | | |
| phone types used and its versions: | OpenScape Desk Phone HFA CP400 V1 R5.5.0, Unify OpenStage 40 HFA V3 R0.48.0 | | |
| other comments: | - | | |
| Test Numbers: | | | |
| A1: +49 911 517650 6338 | | | |
| A2: +49 911 517650 6337 | | | |
| B1: +49 89 7007 20823 | | | |
| B2: +49 171 1600644 | | | |
| | | | |
| **Test Case** | **Result** | **Display**<br>**1) connected state A-B**<br>**2) connected state A-C**<br>**3) hold/held state** | **Remarks** |
| 1-1 REG SBC SP | ok | - | |
| | | | |
| 2-1 National Call 4K SP | ok | A1: 68089700720823<br>B1: 0911 5176506338 | A1 calls B1 |
| 2-2 National Call SP 4K | ok | A1: 0089700720823<br>B1: 0911 5176506338 | B1 calls A1 |
| 2-3 Emergency Call | n. r. | | No tested, not configured in OS 4000 |
| | | | |
| 3-1 SP 4K No Answer | ok | A1: 0089700720823 | B1 calls A1 |
| | | B1: 0911 5176506338 | |
| 3-2 4K SP No Answer | ok | A1: 68089700720823<br>B1: 0911 5176506338 | A1 calls B1 |
| 3-3 4K SP Busy | ok | A1: 6801717600644 | A1 calls B2 |
| 3-4 SP 4K Busy | ok | B1: 0911 5176506338 | B1 calls A1 |
| 3-5 4K SP Reject | ok | A1: 6801717600644<br>B1: +49 911 5176506338 | A1 calls B2<br>Busy Here received from NFON |
| 3-6 SP 4K Reject | ok | OK | 03684640616 calls -6338, 6338 rejects the call |
| 3-7 4K SP Invalid Number | ok | OK | A1 calls invalid number<br>404 Not Found received from NFON |
| | | | |
| 4-1 4K SP CLIR | ok | OK | A2 calls B2 |
| 4-2 SP 4K CLIR | ok | OK | B2 calls A1 |

| | | | |
|---|---|---|---|
| 4-3 4K SP Alternative CID | ok | OK | B2 calls B1, CID is an alternative number |
| | | | |
| 5-1 SP 4K Session Timer | ok | OK | Session Refresh Interval of 15 Minuten by Provider |
| | | | |
| 6-1 4K SP Hold | ok | OK | A1 calls B1 and set the call on hold<br>B1 hears Hold Music |
| 6-2 SP 4K Hold | ok | OK | B1 calls A1 and set the call on hold |
| | | | A1 hears Hold Music |
| 6-3 4K Alternate | ok | OK | B2 calls A1, A1 calls B1 and alternates<br>Held party hears Hold Music |
| 6-4 SP Alternate | ok | OK | B2 calls A1, B2 calls B1 and alternates<br>Held party hears Hold Tone |
| | | | |
| 7-1 CFU 4K SP SP | ok | OK | |
| 7-2 CFU SP 4K SP | ok | OK | |
| 7-3 CFB SP 4K SP | ok | OK | -6338 is busy. 03684640616 calls -6338 and will be forwarded on busy to 01719341897 |
| 7-4 CFB 4K SP 4K | n. r. | | Not tested |
| 7-5 CFNR 4K SP 4K | n. r. | | Not tested |
| 7-6 CFNR SP 4K SP | n. r. | | Not tested |
| | | | |
| 8-1 Attended Call Transfer 4K SP 4K | n. r. | | Not tested |
| 8-2 Attended Call Transfer SP 4K SP | ok | OK | |
| 8-3 Semi Attended Call Transfer SP 4K SP | ok | OK | |
| 8-4 Semi Attended Call Transfer SP 4K SP | n. r. | | Not tested |
| 8-5 Blind Call Transfer 4K SP 4K | ok | OK | '-6337 rcalls 03684640616 and blind transfers the call to -6338 |
| 8-6 Blind Call Transfer SP 4K SP | n. r. | | Not tested |
| | | | |
| 9-1 Conference Call | ok | OK | A1 calls B1, A1 starts a second call to B2 and initates a conference call |
| | | | |
| 10-1 DTMF 4K SP | ok | OK | |
| 10-2 DTMF SP 4K | ok | OK | - |
| 10-3 Pickup Group | ok | OK | - |
| 10-4 Multi-Line Hunt Group | ok | OK | - |
| 10-5 One Number Service | ok | OK | - |

# 7. Traces Examples

## 7.1. Registration Request

```
    6 08:34:49,7 192.168.178.5     109.68.96.123      SIP      756 Request: REGISTER sip:siptrunk.cloud-cfg.com:5060;transport=tcp  (1 binding) |

Frame 6: 756 bytes on wire (6048 bits), 756 bytes captured (6048 bits)
Ethernet II, Src: 00:00:00_00:00:00 (00:00:00:00:00:00), Dst: 00:00:00_00:00:00 (00:00:00:00:00:00)
Internet Protocol Version 4, Src: 192.168.178.5, Dst: 109.68.96.123
User Datagram Protocol, Src Port: 34331, Dst Port: 5060
Session Initiation Protocol (REGISTER)
 > Request-Line: REGISTER sip:siptrunk.cloud-cfg.com:5060;transport=tcp SIP/2.0
 v Message Header
   > Via: SIP/2.0/TCP 192.168.178.5:5060;branch=z9hG4bK51b3.6d7b1e7598e54be884079d9ee056140d.0;i=1
     Expires: 120
     User-Agent: SIP alive check
     Call-ID: 195b00a5
     [Generated Call-ID: 195b00a5]
   > From: <sip:+4991151765000@siptrunk.cloud-cfg.com>;tag=8afeb538
   > CSeq: 6699 REGISTER
     Max-Forwards: 70
   > To: <sip:+4991151765000@siptrunk.cloud-cfg.com>
   > Contact: <sip:+4991151765000@192.168.178.5:5060;transport=tcp>;expires=120
     Content-Length: 0
   > Authorization: Digest username="KCJILR7A4B", realm="siptrunk.cloud-cfg.com", nonce="61f8e2b615bd6b0359ea424cbe638b090327aee2", uri="sip:siptrunk.cloud-cfg.com:5060;transport=tcp",
```

## 7.2. Inbound Call INVITE Request

```
   3 2022-02-01 07:56:00,8 109.68.96.123 192.168.178.5 SIP/SDP  1511 Request: INVITE sip:+499115176506338@192.168.178.5:5060;transport=tcp |
   4 2022-02-01 07:56:00,8 192.168.178.5 109.68.96.123 SIP       493 Status: 100 Trying |
  20 2022-02-01 07:56:01,2 192.168.178.5 109.68.96.123 SIP/SDP  1242 Status: 180 Ringing |
1564 2022-02-01 07:56:16,5 192.168.178.5 109.68.96.123 SIP/SDP  1326 Status: 200 OK |
1573 2022-02-01 07:56:16,6 109.68.96.123 192.168.178.5 SIP       817 Request: ACK sip:+499115176506338@92.60.222.66:34331;transport=tcp |
6488 2022-02-01 07:56:41,1 192.168.178.5 109.68.96.123 SIP       803 Request: BYE sip:3f91ff8c39d8e18d55102f9dcac76d9e@10.111.222.2:5160;transport=udp |
6489 2022-02-01 07:56:41,2 109.68.96.123 192.168.178.5 SIP       538 Status: 200 OK |

Frame 3: 1511 bytes on wire (12088 bits), 1511 bytes captured (12088 bits)
Ethernet II, Src: 00:00:00_00:00:00 (00:00:00:00:00:00), Dst: 00:00:00_00:00:00 (00:00:00:00:00:00)
Internet Protocol Version 4, Src: 109.68.96.123, Dst: 192.168.178.5
User Datagram Protocol, Src Port: 5060, Dst Port: 34331
Session Initiation Protocol (INVITE)
 > Request-Line: INVITE sip:+499115176506338@192.168.178.5:5060;transport=tcp SIP/2.0
 v Message Header
   > Record-Route: <sip:109.68.96.123;transport=tcp;r2=on;lr;ftag=10.111.222.2+4+b8982c43+3927c0c;did=fd5.c8ae1e1>
   > Record-Route: <sip:10.111.222.219;r2=on;lr;ftag=10.111.222.2+4+b8982c43+3927c0c;did=fd5.c8ae1e1>
   > Via: SIP/2.0/TCP 109.68.96.123:5060;branch=z9hG4bK616.acd64374.0
   > Via: SIP/2.0/UDP 10.111.222.2:5160;rport=5160;received=10.111.222.2;branch=z9hG4bK+c519e43c4dd561502d3a081739b496c21+sip+4+ab7c2fca
   > From: +4989700720823 <sip:+4989700720823@nfon.net>;tag=10.111.222.2+4+b8982c43+3927c0c
   > To: <sip:+499115176506338@nfon.net>
   > CSeq: 1 INVITE
     Expires: 180
     Content-Length: 267
     Supported: timer
   > Contact: <sip:3f91ff8c39d8e18d55102f9dcac76d9e@10.111.222.2:5160;transport=udp>
     Content-Type: application/sdp
     Call-ID: 64639459bf163b71538c9a7d01ec3364@10.111.222.2
     [Generated Call-ID: 64639459bf163b71538c9a7d01ec3364@10.111.222.2]
     Accept: application/sdp,application/isup,multipart/mixed,application/vnd.siemens.key-event,application/vnd.siemens.surpass,application/dtmf-relay
     Diversion: <sip:00498970070@10.185.251.21;user=phone>;reason="unknown";privacy="off"
     MIME-Version: 1.0
     Max-Forwards: 50
     Session-Expires: 1800
     Allow: ACK,INFO,BYE,CANCEL,INVITE,OPTIONS,NOTIFY,PRACK,UPDATE
 v Message Body
   v Session Description Protocol
       Session Description Protocol Version (v): 0
     > Owner/Creator, Session Id (o): hiQ9200 81888280196842 81888280196842 IN IP4 109.68.99.202
       Session Name (s): -
     > Connection Information (c): IN IP4 109.68.99.202
     > Time Description, active time (t): 0 0
     > Media Description, name and address (m): audio 41032 RTP/AVP 8 101
       Media Attribute (a): sendrecv
     > Media Attribute (a): rtpmap:8 PCMA/8000
     > Media Attribute (a): rtpmap:101 telephone-event/8000
     > Media Attribute (a): fmtp:101 0-15
     > Media Attribute (a): sqn: 0
     > Media Attribute (a): cdsc: 1 image udptl t38
     > Media Attribute (a): ptime:20
       [Generated Call-ID: 64639459bf163b71538c9a7d01ec3364@10.111.222.2]
```

## 7.3. Inbound Call 200 OK Reply

```
   3 2022-02-01 07:56:00,8 109.68.96.123 192.168.178.5 SIP/SDP   1511 Request: INVITE sip:+499115176506338@192.168.178.5:5060;transport=tcp |
   4 2022-02-01 07:56:00,8 192.168.178.5 109.68.96.123 SIP        493 Status: 100 Trying |
  20 2022-02-01 07:56:01,2 192.168.178.5 109.68.96.123 SIP/SDP   1242 Status: 180 Ringing |
1564 2022-02-01 07:56:16,5 192.168.178.5 109.68.96.123 SIP/SDP   1326 Status: 200 OK |
1573 2022-02-01 07:56:16,6 109.68.96.123 192.168.178.5 SIP        817 Request: ACK sip:+499115176506338@92.60.222.66:34331;transport=tcp |
6488 2022-02-01 07:56:41,1 192.168.178.5 109.68.96.123 SIP        803 Request: BYE sip:3f91ff8c39d8e18d55102f9dcac76d9e@10.111.222.2:5160;transport=udp |
6489 2022-02-01 07:56:41,2 109.68.96.123 192.168.178.5 SIP        538 Status: 200 OK |
```

```
Frame 1564: 1326 bytes on wire (10608 bits), 1326 bytes captured (10608 bits)
Ethernet II, Src: 00:00:00_00:00:00 (00:00:00:00:00:00), Dst: 00:00:00_00:00:00 (00:00:00:00:00:00)
Internet Protocol Version 4, Src: 192.168.178.5, Dst: 109.68.96.123
User Datagram Protocol, Src Port: 34331, Dst Port: 5060
Session Initiation Protocol (200)
 > Status-Line: SIP/2.0 200 OK
 v Message Header
   > Via: SIP/2.0/TCP 109.68.96.123:5060;branch=z9hG4bK616.acd64374.0
   > Via: SIP/2.0/UDP 10.111.222.2:5160;rport=5160;received=10.111.222.2;branch=z9hG4bK+c519e43c4dd561502d3a081739b496c21+sip+4+ab7c2fca
   > Record-Route: <sip:109.68.96.123;transport=tcp;r2=on;lr;ftag=10.111.222.2+4+b8982c43+3927c0c;did=fd5.c8ae1e1>
   > Record-Route: <sip:10.111.222.219;r2=on;lr;ftag=10.111.222.2+4+b8982c43+3927c0c;did=fd5.c8ae1e1>
     Require: timer
   > Contact: <sip:+499115176506338@192.168.178.5:5060;transport=tcp>
   > To: <sip:+499115176506338@nfon.net>;tag=582939153
   > From: +4989700720823 <sip:+4989700720823@nfon.net>;tag=10.111.222.2+4+b8982c43+3927c0c
     Call-ID: 64639459bf163b71538c9a7d01ec3364@10.111.222.2
     [Generated Call-ID: 64639459bf163b71538c9a7d01ec3364@10.111.222.2]
   > CSeq: 1 INVITE
     Session-Expires: 1800;refresher=uac
     Allow: INVITE, ACK, OPTIONS, BYE, CANCEL, REGISTER, INFO, PRACK, UPDATE
     Content-Type: application/sdp
     Server: OpenScape 4000 -  Common Gateway
     Supported: replaces, timer
   > P-Asserted-Identity: "Test-SBC" <sip:+499115176506338@192.168.178.5;user=phone>
     Content-Length: 248
 v Message Body
   v Session Description Protocol
       Session Description Protocol Version (v): 0
     > Owner/Creator, Session Id (o): MxSIP 68894 124506574 IN IP4 10.51.110.150
       Session Name (s): SIP Call
     > Connection Information (c): IN IP4 192.168.178.5
     > Time Description, active time (t): 0 0
     > Media Description, name and address (m): audio 25998 RTP/AVP 8 101
     > Media Attribute (a): rtpmap:8 PCMA/8000
     > Media Attribute (a): rtpmap:101 telephone-event/8000
     > Media Attribute (a): fmtp:101 0-15
     > Media Attribute (a): silenceSupp:off - - - -
     > Media Attribute (a): ptime:20
       Media Attribute (a): sendrecv
       [Generated Call-ID: 64639459bf163b71538c9a7d01ec3364@10.111.222.2]
```

## 7.4. Outbound Call INVITE Request

```
   10 2022-02-01 07:44:57,9  192.168.178.5  109.68.96.123  SIP/SDP  1158 Request: INVITE sip:+4989700720823@siptrunk.cloud-cfg.com:5060;transport=tcp;user=phone |
   11 2022-02-01 07:44:57,9  109.68.96.123  192.168.178.5  SIP       476 Status: 100 Giving a try |
   12 2022-02-01 07:44:58,0  109.68.96.123  192.168.178.5  SIP       837 Status: 401 Unauthorized |
   13 2022-02-01 07:44:58,0  192.168.178.5  109.68.96.123  SIP       541 Request: ACK sip:+4989700720823@siptrunk.cloud-cfg.com:5060;transport=tcp;user=phone |
   25 2022-02-01 07:44:58,1  192.168.178.5  109.68.96.123  SIP/SDP  1397 Request: INVITE sip:+4989700720823@siptrunk.cloud-cfg.com:5060;transport=tcp;user=phone |
   27 2022-02-01 07:44:58,1  109.68.96.123  192.168.178.5  SIP       476 Status: 100 Giving a try |
   28 2022-02-01 07:44:59,2  109.68.96.123  192.168.178.5  SIP       803 Status: 183 Session Progress |
   32 2022-02-01 07:44:59,3  109.68.96.123  192.168.178.5  SIP       794 Status: 180 Ringing |
   34 2022-02-01 07:44:59,3  109.68.96.123  192.168.178.5  SIP/SDP  1323 Status: 200 OK |
   57 2022-02-01 07:44:59,4  192.168.178.5  109.68.96.123  SIP       950 Request: ACK sip:3f91ff8c39d8e18d55102f9dcac76d9e@10.111.222.2:5060;transport=udp |
 7792 2022-02-01 07:45:38,1  109.68.96.123  192.168.178.5  SIP       858 Request: BYE sip:+499115176506338@92.60.222.66:34331;transport=tcp;user=phone |
 7803 2022-02-01 07:45:38,1  192.168.178.5  109.68.96.123  SIP       587 Status: 200 OK |
```

```
Ethernet II, Src: 00:00:00_00:00:00 (00:00:00:00:00:00), Dst: 00:00:00_00:00:00 (00:00:00:00:00:00)
Internet Protocol Version 4, Src: 192.168.178.5, Dst: 109.68.96.123
User Datagram Protocol, Src Port: 34331, Dst Port: 5060
Session Initiation Protocol (INVITE)
> Request-Line: INVITE sip:+4989700720823@siptrunk.cloud-cfg.com:5060;transport=tcp;user=phone SIP/2.0
v Message Header
  > Via: SIP/2.0/TCP 192.168.178.5:5060;branch=z9hG4bK61fd.357b6352176d28ebf7f4b6263d84a2d3.0;i=18695
    Max-Forwards: 69
  > Contact: <sip:+499115176506338@192.168.178.5:5060;transport=tcp;user=phone>
  > To: <sip:+4989700720823@siptrunk.cloud-cfg.com:5060;transport=tcp;user=phone>
  > From: "Test-SBC"<sip:+499115176506338@siptrunk.cloud-cfg.com:5060;transport=tcp;user=phone>;tag=cb2f3d9c96
    Call-ID: ed0dfd320c226cb3
    [Generated Call-ID: ed0dfd320c226cb3]
  > CSeq: 4732 INVITE
    Session-Expires: 1800
    Min-SE: 90
    Allow: INVITE, ACK, OPTIONS, BYE, CANCEL, REGISTER, INFO, PRACK, UPDATE
    Content-Type: application/sdp
    Supported: 100rel, timer
    User-Agent: OpenScape 4000 -  Common Gateway
  > P-Asserted-Identity: +499115176500 <sip:+499115176500@192.168.178.5;user=phone>
    Content-Length: 282
v Message Body
  v Session Description Protocol
      Session Description Protocol Version (v): 0
    > Owner/Creator, Session Id (o): MxSIP 134425 732897234 IN IP4 10.51.110.150
      Session Name (s): SIP Call
    > Connection Information (c): IN IP4 192.168.178.5
    > Time Description, active time (t): 0 0
    > Media Description, name and address (m): audio 41984 RTP/AVP 8 98 0
    > Media Attribute (a): rtpmap:8 PCMA/8000
    > Media Attribute (a): rtpmap:98 telephone-event/8000
    > Media Attribute (a): fmtp:98 0-15
    > Media Attribute (a): rtpmap:0 PCMU/8000
    > Media Attribute (a): silenceSupp:off - - - -
      Media Attribute (a): sendrecv
      Media Attribute (a): rtcp-mux
    > Media Attribute (a): ptime:20
      [Generated Call-ID: ed0dfd320c226cb3]
```

## 7.5. Outbound Call 200 OK Reply

```
  10 2022-02-01 07:44:57,9  192.168.178.5  109.68.96.123  SIP/SDP  1158 Request: INVITE sip:+4989700720823@siptrunk.cloud-cfg.com:5060;transport=tcp;user=phone
  11 2022-02-01 07:44:57,9  109.68.96.123  192.168.178.5  SIP       476 Status: 100 Giving a try |
  12 2022-02-01 07:44:58,0  109.68.96.123  192.168.178.5  SIP       837 Status: 401 Unauthorized |
  13 2022-02-01 07:44:58,0  192.168.178.5  109.68.96.123  SIP       541 Request: ACK sip:+4989700720823@siptrunk.cloud-cfg.com:5060;transport=tcp;user=phone |
  25 2022-02-01 07:44:58,1  192.168.178.5  109.68.96.123  SIP/SDP  1397 Request: INVITE sip:+4989700720823@siptrunk.cloud-cfg.com:5060;transport=tcp;user=phone
  27 2022-02-01 07:44:58,1  109.68.96.123  192.168.178.5  SIP       476 Status: 100 Giving a try |
  28 2022-02-01 07:44:59,2  109.68.96.123  192.168.178.5  SIP       803 Status: 183 Session Progress |
  32 2022-02-01 07:44:59,3  109.68.96.123  192.168.178.5  SIP       794 Status: 180 Ringing |
  34 2022-02-01 07:44:59,3  109.68.96.123  192.168.178.5  SIP/SDP  1323 Status: 200 OK |
  57 2022-02-01 07:44:59,4  192.168.178.5  109.68.96.123  SIP       950 Request: ACK sip:3f91ff8c39d8e18d55102f9dcac76d9e@10.111.222.2:5060;transport=udp |
7792 2022-02-01 07:45:38,1  109.68.96.123  192.168.178.5  SIP       858 Request: BYE sip:+499115176506338@92.60.222.66:34331;transport=tcp;user=phone |
7803 2022-02-01 07:45:38,1  192.168.178.5  109.68.96.123  SIP       587 Status: 200 OK |

Frame 34: 1323 bytes on wire (10584 bits), 1323 bytes captured (10584 bits)
Ethernet II, Src: 00:00:00_00:00:00 (00:00:00:00:00:00), Dst: 00:00:00_00:00:00 (00:00:00:00:00:00)
Internet Protocol Version 4, Src: 109.68.96.123, Dst: 192.168.178.5
User Datagram Protocol, Src Port: 5060, Dst Port: 34331
Session Initiation Protocol (200)
  Status-Line: SIP/2.0 200 OK
  Message Header
      Call-ID: ed0dfd320c226cb3
      [Generated Call-ID: ed0dfd320c226cb3]
      CSeq: 4733 INVITE
      From: "Test-SBC"<sip:+499115176506338@siptrunk.cloud-cfg.com:5060;transport=tcp;user=phone>;tag=cb2f3d9c96
      To: <sip:+4989700720823@siptrunk.cloud-cfg.com:5060;transport=tcp;user=phone>;tag=sip+3+3b95001c+5aee58f9
      Via: SIP/2.0/TCP 192.168.178.5:5060;rport=34331;received=92.60.222.66;branch=z9hG4bK71fd.5e3352ea66f18d10678d6cbf6f96a8b4.0;i=18695
      Server: SIP/2.0
      Record-Route: <sip:10.111.222.219;r2=on;lr;ftag=cb2f3d9c96;did=39d.22e8ffd6>
      Record-Route: <sip:109.68.96.123;transport=tcp;r2=on;lr;ftag=cb2f3d9c96;did=39d.22e8ffd6>
      Content-Length: 264
      Require: timer
      Contact: <sip:3f91ff8c39d8e18d55102f9dcac76d9e@10.111.222.2:5060;transport=udp>
      Content-Type: application/sdp
      Accept: application/sdp,application/isup,multipart/mixed,application/vnd.siemens.key-event,application/vnd.siemens.surpass,application/dtmf-relay
      MIME-Version: 1.0
      Session-Expires: 1800;refresher=uas
      Allow: ACK,INFO,BYE,CANCEL,INVITE,OPTIONS,NOTIFY,PRACK,UPDATE
  Message Body
      Session Description Protocol
        Session Description Protocol Version (v): 0
        Owner/Creator, Session Id (o): hiQ9200 57252347125504 57252347125504 IN IP4 109.68.99.202
        Session Name (s): -
        Connection Information (c): IN IP4 109.68.99.202
        Time Description, active time (t): 0 0
        Media Description, name and address (m): audio 40814 RTP/AVP 8 98
        Media Attribute (a): sendrecv
        Media Attribute (a): rtpmap:8 PCMA/8000
        Media Attribute (a): rtpmap:98 telephone-event/8000
        Media Attribute (a): fmtp:98 0-15
        Media Attribute (a): sqn: 0
        Media Attribute (a): cdsc: 1 image udptl t38
        Media Attribute (a): ptime:20
```

## About Unify

Unify is one of the world's leading communications software and services brands, providing integrated communications and collaboration solutions worldwide. Our customers range in size from 5 employees to 500,000+ employees. Our solutions unify multiple voice, video and data networks, connected devices and applications into one easy-to-use platform that allows teams to collaborate effectively and efficiently – anytime, anywhere. The result is a transformation of how the enterprise communicates and collaborates that amplifies collective effort, energizes the business, improves employee satisfaction and enhances business performance. Unify has a strong heritage of product reliability, innovation, open standards and security. Our OpenScape and Circuit communications solutions provide a seamless and efficient collaboration experience – on any device. Together, the group's global team of UCC experts and service professionals set the standard for a rich communications and collaboration experience that empowers teams to deliver better results. Unify is an Atos company.

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