

OpenScape 4000 V10
OpenScape SBC V10



Konfigurationsanleitung der OS 4000 und des OS SBC für den M-net SIP-Trunk

Getestete Szenarien: OS 4000<->OS SBC<->M-net und OS 4000<->M-net

Provider-Anbindung: UDP/TCP/TLS

Provider-Registration: Dynamische Registrierung

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Trunk-Zugangsdaten aus dem M-net Kundenportal

Konfigurationsdokument:

- [M-net: Technische Grundlage für die Anschaltung SIP-Trunk](#)

Die Zugangsdaten für das Einrichten des SIP-Trunks sind im M-net Kundenportal ersichtlich und sehen beispielhaft wie unten gezeigt aus:

The screenshot shows a navigation bar with three items: 'Vertragsübersicht', 'Tarif-Upgrade', and 'SIM-Karten aktivieren'. Below the navigation bar is a breadcrumb trail: 'Verträge > Vertragsübersicht > Zugangsdaten'. The main heading is 'Zugangsdaten'. Below the heading is a note: 'Hier finden Sie alle notwendigen Daten für Ihren Anschluss 089 18XXXXX 00 bis 29.' Below this note is a table with four rows of data.

SIP-Benutzername	+498918XXXXX0
SIP-Passwort	XXXXXXXXXXXXXXXXXX
SIP-Registrar	business.mnet-voip.de
Gültig ab	19.12.2022

Konfiguration der OpenScape 4000

Bei der Anbindung von M-net über den OpenScape SBC wird in der OS 4000 das Default-Profil "NatTrkWithoutRegistration" ohne Anpassungen verwendet.

Bei der direkten Anbindung der OpenScape 4000 mit M-net wird das Profil M-net verwendet.

Konfiguration des OpenScape SBC

The screenshot shows the 'General - oss2' configuration page in the Unify OpenScape Session Border Control Management Portal. The user is logged in as 'administrator'. The page is divided into several sections:

- Alarms:** Alarm summary shows 0 Critical, 0 Major, and 0 Minor alarms. A 'Show alarm details' button is available.
- System Status:**
 - Branch mode: Centralized SBC
 - Operational state: normal
 - Auto refresh timer: 1 min
 - Com Node 1:** Primary server: 172.29.179.30, Penalty box state: Active
 - Com Node 2:** Primary server: [empty], Penalty box state: [empty]
- System Info:**
 - CPU: 2.35 % - 2 x 2600 MHz
 - Memory: 20.68 % - 4 Gb
 - Disk: 11.02 % - 42 Gb
 - System uptime: 6 days 21:49
 - Hardware type: Virtual OSS 250
 - Hostname: oss2
- Software Info:**
 - Software version: V10 R2.05.00
 - Software Partition information: Active, Backup
- Services status:** Registered subscribers (Show)
- SSP status:** Dynamic port mapping (Show)
- Dynamic IP remote endpoints:** Denial of Service Mitigation (Show)
- TURM Allocations:** Telephony Connector status (Show)
- SIP Loadbalancer status:** (Show)

VOIP – Sip Server Settings Tab

- Primary server: OS 4000 SIP IP
- Transport: TCP
- Port: 5060

The screenshot shows the 'Sip Server Settings' configuration page under the 'VOIP' section. The page includes a warning message: 'Select OK to temporarily store changes. Make your changes permanent by selecting 'Apply Changes' on the General page.' The configuration is organized into several sections:

- General:**
 - Comm System Type: Simplex
 - Allow Register from SERVER:
 - Other trusted servers: [button]
- Node 1:**
 - Target type: Binding
 - Primary server: 172.29.179.30, Transport: TCP, Port: 5060
 - Backup server: [empty], Transport: TCP, Port: [empty]
 - SRV record: [empty], Transport: TCP
- Node 2:**
 - Target type: Binding
 - Primary server: [empty], Transport: TCP, Port: [empty]
 - Backup server: [empty], Transport: TCP, Port: [empty]
 - SRV record: [empty], 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): 50

VOIP – Port and Signaling Settings Tab

VOIP ?

i 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): 4000 TCP keep alive

BFCP connection timer (min): 720

Maximal call session time (hr): 12

Miscellaneous ?

SIP SSL single context

Konfigurationsanleitung der OS 4000 und des OS SBC für den M-net SIP Trunk

VOIP – Media Profile Tab

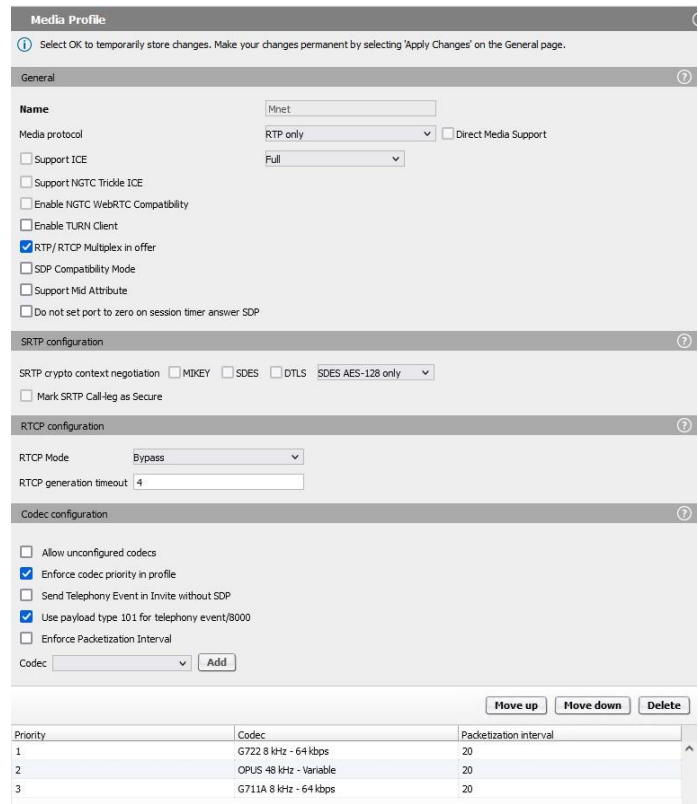
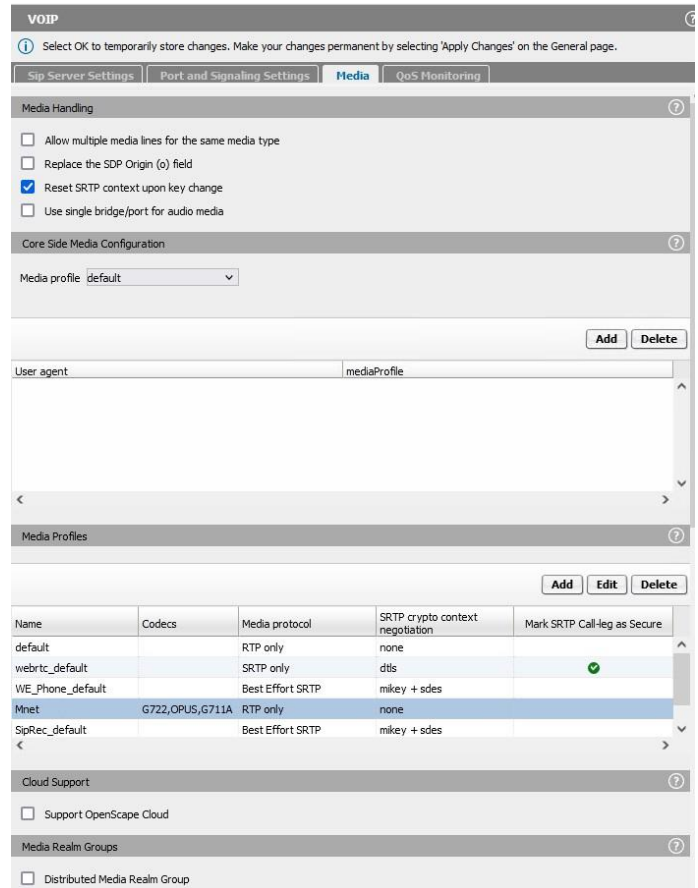
- **Access Media Protocol:** *RTP only* für UDP oder TCP,

SRTP only für TLS

- **SRTP crypto context negotiation:** *SDES* für TLS

- **Codecs Access Side:** Zumindest G.722, OPUS und G.711A

- **Packetization Interval:** 20 ms



Features - SIP Service Provider Profile

- **Default SSP profile:** *Sip Connect*
- **SIP service address:** *business.mnet-voip.de*

Konfigurationsanleitung der OS 4000 und des OS SBC für den M-net SIP Trunk

SIP Service Provider Profile

Select OK to temporarily store changes. Make your changes permanent by selecting 'Apply Changes' on the General page.

General

Name: Mnet Default SSP profile: Sip Connect

SIP Service Address

Use SIP Service Address for identity headers

SIP service address: business.mnet-voip.de

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): 3600

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 displa

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

Keep P-Asserted-Identity from access side

Avoid sending 183 messages

Avoid sending 180 message (for 60s)

TLS

TLS Signaling: Endpoint Config

Sip Connect

Use tel URI

Send user=phone in SIP URI

Registration mode

1TR.118

Features - Remote Endpoint Configuration

- **Name:** Das zuvor erstellte SIP Service Provider-Profil
- **Signaling address type:** DNS SRV
- **Remote URL und Registrar Server:** *business.mnet-voip.de*
- **Remote Port:** 0
- **Remote transport:** UDP, TLS oder TLS. M-net empfiehlt UDP
- **Media profile:** Das zuvor erstellte VoIP Media-Profil
- **Certificate profile:** Bei TLS das unter Certificate Management erstellte Profil

Konfigurationsanleitung der OS 4000 und des OS SBC für den M-net SIP Trunk

- Registrar Server Port: 0

Remote endpoint configuration

Select OK to temporarily store changes. Make your changes permanent by selecting 'Apply Changes' on the General page.

Name

Type

Profile

Access realm profile

Core realm profile

Associated Endpoint

Enable Call Limits

Maximum Permitted Calls

Reserved Calls

Remote Location Information

Support Peer Domains

Support Foreign Peer Domains

Enable access control

Signaling address type

Remote Location domain list

Row	Remote URL	Remote port	Remote transport	Media IP	Media profile	TLS mode	Certificate profile	TLS keep-alive	Keep-alive interval (seconds)	Keep-Alive timeout (sec)	INVITE No Answer timeout (msec)	INVITE No Reply timeout (msec)	Outbound Proxy	Outbound Proxy Port	Registrar Server	Registrar Server Port
1	business.mnet-voip.de	0	UDP		Mnet	Server authentication	Mnet	<input type="checkbox"/>	120	10	360000	3000		0	business.mnet-voip.de	0

Remote Location Identification/Routing

Core FQDN

Core realm port

Default core realm location domain name

Default home DN

Vertragsübersicht | Tarif-Upgrade | SIM-Karten aktivieren

Verträge > Vertragsübersicht > Zugangsdaten

Zugangsdaten

Hier finden Sie alle notwendigen Daten für Ihren Anschluss 089 18XXXXX 00 bis 29.

SIP-Benutzername	+498918XXXXX0
SIP-Passwort	XXXXXXXXXXXXXXXXXX
SIP-Registrar	business.mnet-voip.de
Gültig ab	19.12.2022

Features - Remote Endpoint Configuration(Fortsetzung)

Konfigurationsanleitung der OS 4000 und des OS SBC für den M-net SIP Trunk

The image shows two screenshots from a configuration interface. The left screenshot displays the 'Digest Authentication' settings, including a checked box for 'Digest authentication supported', a realm of 'business.mnet-voip.de', and fields for 'user ID' and 'password'. Below this are sections for 'Access Side Firewall Settings', 'Emergency configuration', 'MSRP Data Configuration', and 'Miscellaneous'. The right screenshot shows the 'Zugangsdaten' (Access Data) page, which lists SIP credentials: 'SIP-Benutzername' (+498918XXXXX), 'SIP-Passwort' (XXXXXXXXXXXXXXXXXX), and 'SIP-Registrar' (business.mnet-voip.de). Three colored arrows (blue, green, red) point from the right screenshot to the corresponding fields in the left screenshot: a blue arrow from the registrar to the user ID field, a green arrow from the username to the password field, and a red arrow from the password to the password field.

Security - Certificate Profile

Bei der Anbindung an M-net über TLS muss das M-net Root-Zertifikat von URL https://m-net.de/fileadmin/2_Geschaefskunden/GK.2_Telefonie/SIP_Trunk/M-net-Root-X1.zip heruntergeladen und als *Local CA file* im OS SBC installiert werden. Zusätzlich muss die Einstellung *Certificate Verification* auf *Full* gesetzt werden.

Konfigurationsanleitung der OS 4000 und des OS SBC für den M-net SIP Trunk

Certificate Profile

① Select OK to temporarily store changes. Make your changes permanent by select

Certificate Profile configuration

Certificate profile name Mnet

Certificate service SIP-TLS

Local client certificate file **Show**

Local server certificate file servercert.pem **Show**

Local CA file M-Net-Root-X1.pem **Show**

Remote CA file **Show**

Local key file serverkey.pem

EC param secp256r1

Attach to Config file

Validation

Certificate Verification Full

Revocation status

Identity Check

Renegotiation

Enforce TLS session renegotiation

TLS session renegotiation interval (minutes) 60

TLS version

Minimum TLS version TLS V1.2

DTLS version

Minimum DTLS version DTLS V1.0

Cipher Suites

Perfect Forward Secrecy Preferred PFS

Encryption Preferred AES-128

Mode of Operation Preferred GCM

Hinweis:

T.38 FAX wurde im Rahmen der Zertifizierung nicht getestet.

Anrufbeispiele

Registration des OpenScape SBC an M-net

```
1 0.00 88.217.251.183 62.216.220.1 SIP 510 Request: REGISTER sip:business.mnet-voip.de;transport=udp (1 binding) |
2 0.01 62.216.220.1 88.217.251.183 SIP 497 Status: 401 Unauthorized |
3 0.10 88.217.251.183 62.216.220.1 SIP 784 Request: REGISTER sip:business.mnet-voip.de;transport=udp (1 binding) |
4 0.10 62.216.220.1 88.217.251.183 SIP 461 Status: 200 OK (REGISTER) (1 binding) |

Frame 3: 784 bytes on wire (6272 bits), 784 bytes captured (6272 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: 88.217.251.183, Dst: 62.216.220.1
User Datagram Protocol, Src Port: 5060, Dst Port: 5060
Session Initiation Protocol (REGISTER)
> Request-Line: REGISTER sip:business.mnet-voip.de;transport=udp SIP/2.0
< Message Header
> Via: SIP/2.0/UDP 88.217.251.183:5060;branch=z9hG4bK4d38.084ca31f6798b8f33f43ee9531a0482f.0;i=1
Expires: 3600
Call-ID: fc3e1f10
[Generated Call-ID: fc3e1f10]
> From: <sip:+498918_0@business.mnet-voip.de>;tag=d11dbc88
> CSeq: 3579 REGISTER
Max-Forwards: 70
> To: <sip:+498918_0@business.mnet-voip.de>
User-Agent: OpenScape Branch
> Contact: <sip:+498918_0@88.217.251.183:5060;transport=udp>;expires=3600
Content-Length: 0
> [truncated]Authorization: Digest username="+498918_0", realm="business.mnet-voip.de", nonce="14e5375e82a3", uri="sip:business.mnet-voip.de;transport=udp",
```

Eingehender Anruf über den OpenScape SBC

```
1 0.00 62.216.220.1 88.217.251.183 SIP/SDP 845 Request: INVITE sip:+498918 0@88.217.251.183:5060;transport=udp
2 0.01 88.217.251.183 62.216.220.1 SIP 442 Status: 100 Trying |
```

Frame 1: 845 bytes on wire (6760 bits), 845 bytes captured (6760 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: 62.216.220.1, Dst: 88.217.251.183
User Datagram Protocol, Src Port: 5060, Dst Port: 5060
Session Initiation Protocol (INVITE)

- > Request-Line: INVITE sip:+498918 0@88.217.251.183:5060;transport=udp SIP/2.0
- ▼ Message Header
 - > Via: SIP/2.0/UDP 62.216.220.1:5060;branch=z9hG4bK+f1db7e15430e2234668c7515546532221+sip+3+ad0e4ccf
 - > From: <sip:+4989 3@business.mnet-voip.de>;tag=62.216.220.1+3+2d530548+5d790c14
 - > To: <sip:+4989 0@business.mnet-voip.de>
 - > CSeq: 133444356 INVITE
 - Expires: 180
 - Content-Length: 171
 - Supported: resource-priority, histinfo
 - > Contact: <sip:3876ad968164c43639fb2ea63b2e1ef4@62.216.220.1:5060>
 - Content-Type: application/sdp
 - Call-ID: 29a77e3a74d32e35016c5e68c3dbcfb3@62.216.220.1
 - [Generated Call-ID: 29a77e3a74d32e35016c5e68c3dbcfb3@62.216.220.1]
 - Max-Forwards: 56
 - Accept: application/sdp, application/dtmf-relay
- ▼ Message Body
 - ▼ Session Description Protocol
 - Session Description Protocol Version (v): 0
 - > Owner/Creator, Session Id (o): - 49883440453345 49883440453345 IN IP4 62.216.222.1
 - Session Name (s): -
 - > Connection Information (c): IN IP4 62.216.222.1
 - > Time Description, active time (t): 0 0
 - > Media Description, name and address (m): audio 54274 RTP/AVP 8 100
 - > Media Attribute (a): rtpmap:100 telephone-event/8000
 - > Media Attribute (a):ptime:20
 - [Generated Call-ID: 29a77e3a74d32e35016c5e68c3dbcfb3@62.216.220.1]

Ausgehender Anruf über den OpenScape SBC

```

8 0.028118 88.217.251.183 62.216.220.1 SIP/SDP 1500 Request: INVITE sip:+4989
9 0.030115 62.216.220.1 88.217.251.183 SIP 470 Status: 100 Trying |

Frame 8: 1500 bytes on wire (12000 bits), 1500 bytes captured (12000 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: 88.217.251.183, Dst: 62.216.220.1
User Datagram Protocol, Src Port: 5060, Dst Port: 5060
Session Initiation Protocol (INVITE)
> Request-Line: INVITE sip:+4989
3@business.mnet-voip.de;transport=udp;user=phone SIP/2.0
  Message Header
    > Via: SIP/2.0/UDP 88.217.251.183:5060;branch=z9hG4bK1e81.fbe6eced8b3fba468403723bcfb5181.0;i=2678
      Max-Forwards: 69
    > Contact: <sip:+4989
0@88.217.251.183:5060;transport=udp;user=phone>
    > To: <sip:+4989
3@business.mnet-voip.de:5060;transport=udp;user=phone>
    > From: "
1"<sip:+4989
0@88.217.251.183;transport=udp;user=phone>;tag=1073f0e543
      Call-ID: bda74442c26840d4
      [Generated Call-ID: bda74442c26840d4]
    > CSeq: 604321986 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 - SoftGate
    > P-Asserted-Identity: "
1" <sip:+4989
0@88.217.251.183;user=phone>
      Content-Length: 570
    > X-Siemens-OSS: OpenScape SBC V10 R2.05.00-2
  Message Body
    Session Description Protocol
      Session Description Protocol Version (v): 0
      > Owner/Creator, Session Id (o): MxSIP 131122 769120005 IN IP4 172.29.179.30
      Session Name (s): SIP Call
      > Connection Information (c): IN IP4 88.217.251.183
      > Time Description, active time (t): 0 0
      > Media Description, name and address (m): audio 12318 RTP/AVP 8 18 9 124 99 100 101
      > Media Attribute (a): rtpmap:8 PCMA/8000
      > Media Attribute (a): rtpmap:18 G729/8000
      > Media Attribute (a): fmp:18 annex=no
      > Media Attribute (a): rtpmap:9 G722/8000
      > Media Attribute (a): rtpmap:124 opus/48000/2
      > Media Attribute (a): fmp:124 maxaveragebitrate=48000;maxplaybackrate=16000;stereo=0;cb:0;useinbandfec=1;usedtx=0;sprop-maxcapture=48000
      > Media Attribute (a): rtpmap:99 red/8000
      > Media Attribute (a): fmp:99 98
      > Media Attribute (a): rtpmap:100 telephone-event/48000
      > Media Attribute (a): fmp:100 0-15
      > Media Attribute (a): rtpmap:101 telephone-event/8000
      > Media Attribute (a): silenceSupp:off - - -
      Media Attribute (a): sendrecv
      Media Attribute (a): rtcp-mux
      [Generated Call-ID: bda74442c26840d4]
  
```

Notruf

```
10 10.279157 88.217.251.183 62.216.220.1 SIP/SDP 1276 Request: INVITE sip:112@business.mnet-voip.de;transport=udp
Internet Protocol Version 4, Src: 88.217.251.183, Dst: 62.216.220.1
User Datagram Protocol, Src Port: 5060, Dst Port: 5060
Session Initiation Protocol (INVITE)
> Request-Line: INVITE sip:112@business.mnet-voip.de;transport=udp SIP/2.0
v Message Header
> Via: SIP/2.0/UDP 88.217.251.183:5060;branch=z9hG4bKdbe6.de7e04285805ecb11277a71da48d1df3.0;i=1499
Max-Forwards: 69
> Contact: <sip:+498918[REDACTED]1@88.217.251.183:5060;transport=udp;user=phone>
> To: <sip:112@business.mnet-voip.de;transport=udp>
> From: "[REDACTED] 2" <sip:+498918[REDACTED]1@business.mnet-voip.de;transport=udp;user=phone>;tag=5bf5942b2c
Call-ID: 132b64d4b7cea625
[Generated Call-ID: 132b64d4b7cea625]
> CSeq: 1949655109 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 - SoftGate
> P-Asserted-Identity: "[REDACTED] 2" <sip:+498918[REDACTED]1@business.mnet-voip.de;user=phone>
Content-Length: 380
> X-Siemens-OSS: OpenScape SBC V10 R2.05.00-2
v Message Body
v Session Description Protocol
  Session Description Protocol Version (v): 0
  > Owner/Creator, Session Id (o): MxSIP 131338 464961134 IN IP4 172.29.179.30
  Session Name (s): SIP Call
  > Connection Information (c): IN IP4 88.217.251.183
  > Time Description, active time (t): 0 0
  > Media Description, name and address (m): audio 38622 RTP/AVP 8 96 9 100 101
  > Media Attribute (a): rtpmap:8 PCMA/8000
  > Media Attribute (a): rtpmap:96 OPUS/48000/2
  > Media Attribute (a): fmtp:96 bitrate=32000
  > Media Attribute (a): rtpmap:9 G722/8000
  > Media Attribute (a): rtpmap:100 telephone-event/48000
  > Media Attribute (a): fmtp:100 0-15
  > Media Attribute (a): rtpmap:101 telephone-event/8000
  > Media Attribute (a): silenceSupp:off - - - -
  Media Attribute (a): sendrcv
  Media Attribute (a): rtcp-mux
  > Media Attribute (a):ptime:20
  [Generated Call-ID: 132b64d4b7cea625]
```