

OpenScape 4000



## How to Configure SIP Trunk for enviaTEL Germany

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## Table of History

| Date       | Version | Changes          |
|------------|---------|------------------|
| 17.11.2021 | 1.0     | Initial Creation |
|            |         |                  |
|            |         |                  |

## Information

The enviaTEL SIP-Trunk will be released for the first time with OpenScope 4000 V8R2 / V10R0.

OS4K version: V10 R0.28  
Certified LW version: A9.035  
Certified SBC version: V10 R1.04.02

Release of Profile: A9.109

## Certification

### Tested Scenarios:

Provider <=> OpenScope 4000  
Provider <=> OpenScope SBC <=> OpenScope 4000

**Provider Topology:** Registered Trunk

## Trunk Configuration Data provided by envia-TEL

**Trunk Name:** enviaTEL enVoice IP-range (bis 2021)

### Used platform:

Cirpack MultiNodeB v4.2j17  
Cirpack Access-SBC v4.3

**Trunk connection:** Registered Trunk

**Transport Protocol:** UDP

**Signaling IP:** 193.98.115.6 (ngn-pbx.enviatel.net); **Signaling Port:** 5060

**Media IP:** 193.98.115.240/28; **Media Ports:** 30000 – 38192

**Supported Connectivity:** SBC with public IP, SBC with private IP behind NAT (static public internet IP is required)

**Documentation:** Technische Richtlinie IP-Anlagenanschluss

The trunk data will be provided via email similar like this:

Details für die Einrichtung des Telefoniedienstes (SIP)<sup>2</sup>:

|               |                      |
|---------------|----------------------|
| Benutzername: | 0342[REDACTED]       |
| Passwort:     | [REDACTED]           |
| Registrar:    | ngn-pbx.enviatel.net |
| Ortskennzahl: | 034298               |

### Note:

EnviaTEL planes to release a new platform beginning of 2022, which requires re-certification

# OpenScope 4000 Configuration

See Service Documentation OpenScope 4000 – Document: IP Solutions for details.

## AMOs (Certification Example)

Note: Certification specific example – for general description see Service-Dokumentation.

```
M2 1 EINR-BUEND:211,"SIP AMT",30,0,* ,1,ON,0,0,NEUTRAL;
M2 1 EINRICHTEN-COP:201,IMEX,FBKW,FBKW;
M2 1 AENDERN-COP:201,COPZU,,,,UNAB,"SIP AMT ÜBER SBC";
M2 1 EINRICHTEN-
COT:210,MVLT&UELM&SAAO&BLOC&LOKN&OLCR&TSCS&VKNN&NOFT&ANZR&SUPN&NOCI&NOCT
&RMVN&KTON;
M2 1 EINR-COSSU:,300,,,,,;
M2 1 AEND-COSSU:COS,300,FBKW&QVKW&GESP&GEZ&AULEXT&MSN;
M2 1 AEND-COSSU:COS,300,AULERU;
M2 1 AEND-COSSU:COS,300,,FBKW&QVKW&GRUBE&MSN&MULTRA;
M2 1 AEND-COSSU:COS,300,,,FBKW&QVKW&GRUBE&MSN&MULTRA;
M2 1 AENDERN-COT:210,COTZU,,UNAB,"210 - SIP AMT TEST";
M2 2 EINR-TDCSU:NEU,1-90-001-0,210,201,1,0,300,9,9,"
",0,ECMAV2,8,,KEINE,,,,GDTR,N,AMT,KEINE,N,0,,00,0,,,,10,VIELE,1-1-
211,2,1,1,LEER,210,10,N,,,,,16,8,1,10,2,EC&G711&G729AOPT,,211,AB,J,TRADITIO,0,60,60,HG3550CO
,1&&30,N,1,,0,0,0,0,N,1-1-130,KEINE;
M2 1 EINR-RICHT:LRTGNEU,2100,ALLE,"IP NOTRUF   ",211,1-1-
210,NEIN,NEIN,FIX,,,PP300,,,,,JA,JA,,1-1-211,NEUTRAL,NEIN,NEIN,NEIN,NEIN,NEIN,NEIN,NEIN,NEIN;
M2 1 EINR-RICHT:LRTGNEU,2101,ALLE,"IP ORT   ",211&&213,1-1-
211,NEIN,,FIX,,,PP300,,,,,JA,NEIN,,1-1-211,NEUTRAL,NEIN,NEIN,NEIN,NEIN,NEIN,NEIN,NEIN,NEIN;
M2 1 EINR-RICHT:LRTGNEU,2102,ALLE,"IP NATIONAL  ",211&&213,1-1-
211,NEIN,,FIX,,,PP300,,,,,JA,NEIN,,1-1-211,NEUTRAL,NEIN,NEIN,NEIN,NEIN,NEIN,NEIN,NEIN,NEIN;
M2 1 EINR-RICHT:LRTGNEU,2103,ALLE,"IP INTERNAT  ",211&&213,1-1-
213,NEIN,,FIX,,,PP300,,,,,JA,NEIN,,1-1-211,NEUTRAL,NEIN,NEIN,NEIN,NEIN,NEIN,NEIN,NEIN,NEIN;
M2 1 EINRICHTEN-LODR:2100,,,,ECHOFFELD,3;
M2 1 EINRICHTEN-LODR:2100,,,,ECHOFFELD,4;
M2 1 EINRICHTEN-LODR:2100,,,,NPI,ISDN,UNKNOWN;
M2 1 EINRICHTEN-LODR:2100,,,,ENDE;
M2 1 EINRICHTEN-LODR:2101,,,,ZIFFSEND,034298;
M2 1 EINRICHTEN-LODR:2101,,,,ECHOFFELD,3;
M2 1 EINRICHTEN-LODR:2101,,,,ECHOFFELD,4;
M2 1 EINRICHTEN-LODR:2101,,,,NPI,ISDN,UNKNOWN;
M2 1 EINRICHTEN-LODR:2101,,,,ENDE;
M2 1 EINRICHTEN-LODR:2102,,,,ZIFFSEND,0;
M2 1 EINRICHTEN-LODR:2102,,,,ECHOFFELD,4;
M2 1 EINRICHTEN-LODR:2102,,,,ECHOFFELD,5;
M2 1 EINRICHTEN-LODR:2102,,,,NPI,ISDN,UNKNOWN;
M2 1 EINRICHTEN-LODR:2102,,,,ENDE;
M2 1 EINRICHTEN-LODR:2103,,,,ZIFFSEND,00;
M2 1 EINRICHTEN-LODR:2103,,,,ECHOFFELD,4;
M2 1 EINRICHTEN-LODR:2103,,,,ECHOFFELD,5;
M2 1 EINRICHTEN-LODR:2103,,,,NPI,ISDN,UNKNOWN;
```

M2 1 EINRICHTEN-LODR:2103,,,,ENDE;  
M2 1 EINRICHTEN-LDAT:2100,ALLE,1,,211,2100,1,,1,LEER,,4,,,,,,,,,,,,;  
M2 1 EINRICHTEN-LDAT:2101,ALLE,1,,211,2101,1,,1,LEER,GEBAKT&PUBNUM,,4,,,,,,,,,,,,;  
M2 1 EINRICHTEN-LDAT:2102,ALLE,1,,211,2102,1,,1,LEER,GEBAKT&PUBNUM,,4,,,,,,,,,,,,;  
M2 1 EINRICHTEN-LDAT:2103,ALLE,1,,211,2103,1,,1,LEER,GEBAKT&PUBNUM,,4,,,,,,,,,,,,;  
M2 1 EINR-LPROF:"NOTRUF",1,2100,100;  
M2 1 EINR-LPROF:"AMT ORT",1,2101,101;  
M2 1 EINR-LPROF:"AMT ISDN NATIONAL",1,2102,102;  
M2 1 EINR-LPROF:"AMT ISDN INTERNAT",1,2103,103;  
M2 1 EINRICHTEN-LDPLN:LCRMUST,0,0-W-00-0-Z,0&1,,103,,7,,,,NEIN;  
M2 1 EINRICHTEN-LDPLN:LCRMUST,0,0-W-00-1-Z,0&1,,103,,7,,,,NEIN;  
M2 1 EINRICHTEN-LDPLN:LCRMUST,0,0-W-00-2-Z,0&1,,103,,7,,,,NEIN;  
M2 1 EINRICHTEN-LDPLN:LCRMUST,0,0-W-00-3-Z,0&1,,103,,7,,,,NEIN;  
M2 1 EINRICHTEN-LDPLN:LCRMUST,0,0-W-00-4-Z,0&1,,103,,7,,,,NEIN;  
M2 1 EINRICHTEN-LDPLN:LCRMUST,0,0-W-00-5-Z,0&1,,103,,7,,,,NEIN;  
M2 1 EINRICHTEN-LDPLN:LCRMUST,0,0-W-00-6-Z,0&1,,103,,7,,,,NEIN;  
M2 1 EINRICHTEN-LDPLN:LCRMUST,0,0-W-00-7-Z,0&1,,103,,7,,,,NEIN;  
M2 1 EINRICHTEN-LDPLN:LCRMUST,0,0-W-00-8-Z,0&1,,103,,7,,,,NEIN;  
M2 1 EINRICHTEN-LDPLN:LCRMUST,0,0-W-00-9-Z,0&1,,103,,7,,,,NEIN;  
M2 1 EINRICHTEN-LDPLN:LCRMUST,0,0-W-0-1-Z,0&1,,102,,4,,,,NEIN;  
M2 1 EINRICHTEN-LDPLN:LCRMUST,0,0-W-0-2-Z,0&1,,102,,4,,,,NEIN;  
M2 1 EINRICHTEN-LDPLN:LCRMUST,0,0-W-0-3-Z,0&1,,102,,4,,,,NEIN;  
M2 1 EINRICHTEN-LDPLN:LCRMUST,0,0-W-0-4-Z,0&1,,102,,4,,,,NEIN;  
M2 1 EINRICHTEN-LDPLN:LCRMUST,0,0-W-0-5-Z,0&1,,102,,4,,,,NEIN;  
M2 1 EINRICHTEN-LDPLN:LCRMUST,0,0-W-0-6-Z,0&1,,102,,4,,,,NEIN;  
M2 1 EINRICHTEN-LDPLN:LCRMUST,0,0-W-0-7-Z,0&1,,102,,4,,,,NEIN;  
M2 1 EINRICHTEN-LDPLN:LCRMUST,0,0-W-0-8-Z,0&1,,102,,4,,,,NEIN;  
M2 1 EINRICHTEN-LDPLN:LCRMUST,0,0-W-0-9-Z,0&1,,102,,4,,,,NEIN;  
M2 1 EINRICHTEN-LDPLN:LCRMUST,0,0-W-1-10,0&1,,100,,1,,,,NEIN;  
M2 1 EINRICHTEN-LDPLN:LCRMUST,0,0-W-1-12,0&1,,100,,1,,,,NEIN;  
M2 1 EINRICHTEN-LDPLN:LCRMUST,0,0-W-1-Z,0&1,,101,,2,,,,NEIN;  
M2 1 EINRICHTEN-LDPLN:LCRMUST,0,0-W-2-Z,0&1,,101,,2,,,,NEIN;  
M2 1 EINRICHTEN-LDPLN:LCRMUST,0,0-W-3-Z,0&1,,101,,2,,,,NEIN;  
M2 1 EINRICHTEN-LDPLN:LCRMUST,0,0-W-4-Z,0&1,,101,,2,,,,NEIN;  
M2 1 EINRICHTEN-LDPLN:LCRMUST,0,0-W-5-Z,0&1,,101,,2,,,,NEIN;  
M2 1 EINRICHTEN-LDPLN:LCRMUST,0,0-W-6-Z,0&1,,101,,2,,,,NEIN;  
M2 1 EINRICHTEN-LDPLN:LCRMUST,0,0-W-7-Z,0&1,,101,,2,,,,NEIN;  
M2 1 EINRICHTEN-LDPLN:LCRMUST,0,0-W-8-Z,0&1,,101,,2,,,,NEIN;  
M2 1 EINRICHTEN-LDPLN:LCRMUST,0,0-W-9-Z,0&1,,101,,2,,,,NEIN;

# Gateway Configuration

## SIP Parameter

| SIP Parameters   |  |
|--|--|
| SIP User Agent   | <b>"SIP User Agent" settings ignored due to usage of SIP trunk profiles</b>            |
| Use SIP Registrar:   | No   |
| SIP Registrar IP Address:                                      | 0.0.0.0  |
| SIP Registrar TLS Port Number:                                 | 5061   |
| SIP Registrar TCP/UDP Port Number:                             | 5060   |
| Alternative SIP Registrar IP Address:                          | 0.0.0.0  |
| Alternative SIP Registrar TLS Port Number:                     | 5061   |
| Alternative SIP Registrar TCP/UDP Port Number:                 | 5060   |
| Period of Registration (sec):                                  | 300  |
| SIP Server (Registrar / Redirect)                              |  |
| SIP Server IP Address:   | 172.29.93.180  |
| SIP Server TCP/UDP Port Number:                                | 5060   |
| SIP Server TLS Port Number:                                    | 5061   |
| Default Registration Period (sec):                             | <input type="text" value="600"/> (used when no 'Expires' value received)               |
| Range used for Randomized Registration (%):                    | <input type="text" value="25"/> 0 means: don't use Randomization                       |
| RFC 3261 Timer Values  |  |
| Transaction Timeout (msec):                                    | <input type="text" value="32000"/> (Should only be changed for DNS failover scenarios) |
| SIP Transport Protocol   |  |
| SIP via TCP:   | Yes  |
| SIP via UDP:   | <input checked="" type="checkbox"/>  |
| SIP via TLS:   | Yes  |
| SIP Session Timer  |  |
| RFC 4028 Support:  | <input checked="" type="checkbox"/>  |
| Session Expires (sec):   | <input type="text" value="1800"/>  |
| Minimal SE (sec):  | <input type="text" value="90"/>  |
| DNS-SRV Records / SIP Flooding Defense                         |  |
| Blocking time for unreachable destination/flood defense (sec): | <input type="text" value="60"/>  |
| Trunking Parameters  |  |
| SIP OPTIONS ping interval (sec, 0=deactivate):                 | <input type="text" value="0"/>   |
| SIP loop call  |  |
| SIP loop call From number:                                     | <input type="text"/>   |
| SIP loop call To number:                                       | <input type="text"/>   |
| SIP loop call frequency (sec, 0=deactivate):                   | <input type="text" value="0"/>   |
| SIP loop call Out of service threshold:                        | <input type="text" value="1"/>   |
| Call Supervision   |  |
| MakeCallReq Timeout (sec):                                     | <input type="text" value="3"/>   |
| SIP Connect Timeout (sec):                                     | <input type="text" value="300"/>   |

## Codec Parameter

### Codec Parameters

| Codec       | Priority     | Voice Activity Detection                 | Frame Size |
|-------------|--------------|--|------------|
| G.711 A-law | Priority 1 ▾ | VAD: <input type="checkbox"/>            | 20 ▾ msec  |
| G.711 μ-law | not used ▾   | VAD: <input type="checkbox"/>            | 20 ▾ msec  |
| G.729       | not used ▾   | VAD: <input type="checkbox"/>            | 20 ▾ msec  |
| G.729A      | not used ▾   | VAD: <input type="checkbox"/>            | 20 ▾ msec  |
| G.729B      | not used ▾   | VAD: <input checked="" type="checkbox"/> | 20 ▾ msec  |
| G.729AB     | not used ▾   | VAD: <input checked="" type="checkbox"/> | 20 ▾ msec  |
| G.722       | not used ▾   | VAD: <input type="checkbox"/>            | 20 ▾ msec  |
| Opus        | not used ▾   | VAD: <input type="checkbox"/>            | 20 ▾ msec  |

#### Opus-Parameter

Use Inband Forward Error Correction (FEC):

Use Constant Bitrate:

Low Delay:

Payload Type for Opus:

Max. Playback Sample Rate (Hz):

Complexity:

#### T.38 Fax

T.38 Fax:

Max. UDP Datagram Size for T.38 Fax (bytes):

Error Correction Used for T.38 Fax (UDP):

Time Range for Immediate Switch to T.38 Fax (s):  0 means: No Immediate Switching

#### Misc.

ClearMode (ClearChannelData):

Frame Size:  msec

#### RFC2833

Transmission of Fax/Modem Tones according to RFC2833:

Transmission of DTMF Tones according to RFC2833:

Payload Type for ClearChannel:

Payload Type for RFC2833:

Payload Type for RFC2198:  (= 'Payload Type for RFC2833' + 1)

Redundant Transmission of RFC2833 Tones according to RFC2198:

Payload Type for RFC4733 WideBand:  (= 'Payload Type for RFC2833' + 2)





## SIP trunking Profile – direct connectivity

|                                  |   |
|----------------------------------|---|
| Profile Name:                    | envia TEL   |
| User Notes:                      | <input type="text"/>  |
| Activate Trunk Profile:          | <input type="checkbox"/>  |
| Account/Authentication Required: | <input checked="" type="checkbox"/>   |
| Remote Domain Name:              | <input type="text" value="ngn-pbx.envia.tel.net"/>  |
| IP Transport Protocol:           | UDP (used for O/G call establishment)   |
| Default PAI:                     | <input type="text"/> (for outgoing "Anonymous" and CLIP "default PAI" profiles)   |
| Security                         | Released Security Level: No Security  |
| Registrar                        | Use Registrar: <input checked="" type="checkbox"/><br>IP Address / Host name: <input type="text" value="ngn-pbx.envia.tel.net"/><br>Specify Port: <input type="checkbox"/><br>Reregistration Interval (sec): <input type="text" value="480"/> |
| Proxy                            | IP Address / Host name: <input type="text" value="ngn-pbx.envia.tel.net"/><br>Specify Port: <input type="checkbox"/>  |
| Outbound Proxy                   | Use Outbound Proxy: <input type="checkbox"/><br>IP Address / Host name: <input type="text"/><br>Specify Port: <input type="checkbox"/><br>Port: <input type="text" value="0"/>  |
| Inbound Proxy                    | Use Inbound Proxy: <input type="checkbox"/><br>IP Address / Host name: <input type="text"/><br>Specify Port: <input type="checkbox"/>   |

## SIP trunking Profile – connectivity via OS – SBC

|                |  |
|----------------|--|
| Registrar      | Use Registrar: <input type="checkbox"/><br>IP Address / Host name: <input type="text"/><br>Specify Port: <input type="checkbox"/><br>Reregistration Interval (sec): <input type="text" value="480"/>   |
| Proxy          | IP Address / Host name: <input type="text" value="&lt;SBC ip-address&gt;"/><br>Specify Port: <input checked="" type="checkbox"/><br>TCP/UDP Port: <input type="text" value="&lt;SBC core-port&gt;"/><br>TLS Port: <input type="text" value="0"/> |
| Outbound Proxy | Use Outbound Proxy: <input type="checkbox"/><br>IP Address / Host name: <input type="text"/><br>Specify Port: <input type="checkbox"/>   |

## SIP trunking profile – parameter (both deployments):

Miscellaneous

[Reset Profile Defaults](#)

Outgoing Call

CLIP outgoing in From header - display part:

CLIP outgoing in From header - user part:

CLIP outgoing in P-Asserted-Id header - display part:

CLIP outgoing in P-Asserted-Id header - user part:

CLIP outgoing in P-Preferred-Id header - display part:

CLIP outgoing in P-Preferred-Id header - user part:

CLIR outgoing in From header - display part:

CLIR outgoing in From header - user part:

Call Diversion (RFC 5806) and HistoryInfo (RFC 4244):

Incoming Call

Incoming call - Called party number:

Incoming call - Calling party number:

Inspect History-Info/Referred-By:

Other

Emergency Call Behavior:

Emergency Call Geolocation (RFC 6442):

Hold Mode for SDP outgoing:

Ignore 100 Rel (RFC 3262):

ContactUriContains:

UPDATE Allowed (RFC 3311):

REFER Allowed (RFC 3515):

Direct Blind Transfer using Referred-By header (RFC 3892):

Registration for Multiple Phone Numbers (RFC 6140):

Silence Suppression Support:

Enable P-Early-Media (RFC 5009):

Send response to OPTIONS request:

TCP Connection Reuse:

[Apply](#) [Undo](#) [Delete](#)

## Account-Name

Account Name: Benutzername  
 Authorization Name: Benutzername  
 Password: Passwort

Account Name

Account Name:

Authorization name:

Provider Name:

New Password:

Confirm Password:

[Apply](#) [Undo](#) [Delete](#)

Details für die Einrichtung des Telefoniedienstes (SIP):

Benutzername: 0342...

Passwort: .....

Registrar: ngn-pbx.enviatel.net

Ortskennzahl: 034298

## CLIPNoScreening

If the customer has not ordered the provider feature “CLIPNoScreening”, then the parameter “Call Diversion” needs to be changed via WBM GUI to the following value:

Call Diversion (RFC 5806) and HistoryInfo (RFC 4244):

# OpenScape SBC Configuration

## Media Profile

- Media protocol: RTP only
- RTP/ RTCP Multiplex in offer: enable
- Codec List: G.711A

**Media Profile**

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

**General**

Name: RTPonly

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: G711U 8 kHz - 64 kbps

| Priority | Codec                 | Packetization interval |
|----------|-----------------------|------------------------|
| 1        | G711A 8 kHz - 64 kbps | Auto                   |

## SIP Service Provider Profile

- **Default SSP profile:** empty
- **Registration required:** enable
- **Registration interval (sec):** 480
- **Incoming SIP manipulation - Calling Party Number:** From header user and display name part
- **Flag - Send Default Home DN in Contact for Call messages:** enable
- **Flag – Remove Silence Suppression parameter from SDP:** enable
- **Flag – Force direction attribute to sendrcv:** enable
- **Flag – Send user=phone in SIP URI:** enable

| SIP Service Provider Profile   |  | Flags  |  |
|--|--|--|--|
| <p>Select OK to temporarily store changes. Make your changes permanent by selecting 'Apply Changes' on the General page.</p>   |  |  |  |
| <p><b>General</b></p> <p>Name: EnviaTel <span style="border: 1px solid red; padding: 2px;">Default SSP profile</span></p> <p><input type="checkbox"/> Use SIP Service Address for identity headers</p> <p>SIP service address: <input type="text"/></p> <p><input type="checkbox"/> Use SIP Service Address in Request-URI header</p> <p><input type="checkbox"/> Use SIP Service Address in From header</p> <p><input type="checkbox"/> Use SIP Service Address in To header</p> <p><input type="checkbox"/> Use SIP Service Address in P-Asserted-Identity header</p> <p><input type="checkbox"/> Use SIP Service Address in Diversion header</p> <p><input type="checkbox"/> Use SIP Service Address in Contact header</p> <p><input type="checkbox"/> Use SIP Service Address in Via header</p> <p><input type="checkbox"/> Use SIP Service Address in P-Preferred-Identity header</p> |  | <p><input type="checkbox"/> FQDN in TO header to SSP</p> <p><input type="checkbox"/> Use To DN to populate the RURI</p> <p><input checked="" type="checkbox"/> <span style="border: 1px solid red; padding: 2px;">Send Default Home DN in Contact for Call messages</span></p> <p><input type="checkbox"/> Allow SDP changes from SSP without session version update</p> <p><input type="checkbox"/> Do not send INVITE with sendonly media attribute</p> <p><input type="checkbox"/> Do not send INVITE with inactive media attribute</p> <p><input type="checkbox"/> Do not send INVITE with video media line</p> <p><input type="checkbox"/> Do not send Invite without SDP</p> <p><input type="checkbox"/> Do not send Re-Invite when no media type change</p> <p><input type="checkbox"/> Do not send Re-Invite</p>   |  |
| <p><b>SIP User Agent</b></p> <p>SIP User Agent towards SSP: Passthru <input type="text"/> SIP User Agent: <input type="text"/></p>   |  | <p><input checked="" type="checkbox"/> <span style="border: 1px solid red; padding: 2px;">Remove Silence Suppression parameter from SDP</span></p> <p><input type="checkbox"/> Enable pass-through of Optional parameters</p> <p><input checked="" type="checkbox"/> <span style="border: 1px solid red; padding: 2px;">Force direction attribute to sendrcv</span></p> <p><input type="checkbox"/> Send default Home DN in PAI</p> <p><input type="checkbox"/> Send default Home DN in PPI</p> <p><input type="checkbox"/> Preserve To and From headers per RFC2543</p> <p><input type="checkbox"/> Disable FQDN pass-through in FROM header</p> <p><input type="checkbox"/> Keep Digest Authentication Header</p> <p><input type="checkbox"/> Send Contact header in OPTIONS</p> <p><input type="checkbox"/> Do not send Privacy header in response messages</p> <p><input type="checkbox"/> Remove bandwidth (b) lines from SDP</p> |  |
| <p><b>Registration</b></p> <p><input checked="" type="checkbox"/> <span style="border: 1px solid red; padding: 2px;">Registration required</span></p> <p>Registration interval (sec): <span style="border: 1px solid red; padding: 2px;">480</span></p>  |  |  |  |
| <p><b>Business Identity</b></p> <p><input type="checkbox"/> Business identity required</p> <p>Business identity DN: <input type="text"/></p>   |  |  |  |
| <p><b>Outgoing SIP manipulation</b></p> <p><input type="checkbox"/> Insert anonymous caller ID for blocked Caller-ID</p> <p><b>Manipulation</b></p>  |  |  |  |
| <p><b>Incoming SIP manipulation</b></p> <p>Calling Party Number: <span style="border: 1px solid red; padding: 2px;">From header user and disp</span></p>   |  |  |  |
|  |  | <p><b>TLS</b></p> <p>TLS Signaling: Pass-Thru <input type="text"/></p>   |  |
|  |  | <p><b>Sip Connect</b></p> <p><input type="checkbox"/> Use tel URI</p> <p><input checked="" type="checkbox"/> <span style="border: 1px solid red; padding: 2px;">Send user=phone in SIP URI</span></p> <p><input type="checkbox"/> Registration mode</p> <p><input type="checkbox"/> 1TR118</p>   |  |

# Remote Endpoint Configuration

Fill all red marked fields.

### Remote endpoint configuration

Select OK to temporarily store changes. Make your changes permanent by selecting OK.

Remote Endpoint Settings

Name: TestTrunk1\_Tauchha Edit

Type: **SSP**

Profile: **EnviaTel**

Access realm profile: Main-Access-Realm - ipv4

Core realm profile: Main-Core-Realm - ipv4

Associated Endpoint: **TestTrunk1\_Tauchha**

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

| Row | Remote URL           | Remote port | Remote transport |
|-----|----------------------|-------------|------------------|
| 1   | ngn-pbx.enviatel.net | 5060        | UDP              |

### Remote Location Domain

Select OK to temporarily store changes. Make your changes permanent by selecting OK.

General

Remote URL: **ngn-pbx.enviatel.net**  Shared domain

Remote port: **5060**

Remote transport: **UDP**

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: **RTPonly**

Media realm subnet IP address: **10.10.10.10**

Outbound Proxy Configuration

Outbound Proxy: **10.10.10.10**

Outbound Proxy Port: 0

Registrar Server Configuration

Registrar Server: **10.10.10.10**

Registrar Server Port: 0

### Remote Location Identification/Routing

Core FQDN: **ngn-pbx.enviatel.net**

Core realm port: **53180**

Default core realm location domain name: **ngn-pbx.enviatel.net**

Default home DN: **034298**

Incoming Routing prefix: **034298**

Digest Authentication

Digest authentication supported

Digest authentication realm: **NM1-PBX.enviatel.net**

Digest authentication user ID: **034298**

Digest authentication password: **\*\*\*\*\***

Access Side Firewall Settings

Enable Firewall Settings Firewall Settings

Emergency configuration

Emergency numbers: **112** Add

Delete

**Emergency call routing**

Details für die Einrichtung des Telefoniedienstes (SIP)?:

Benutzername: **034298**

Passwort: **\*\*\*\*\***

Registrar: **ngn-pbx.enviatel.net**

Ortskennzahl: **034298**

### MSRP Data Configuration

Enable MSRP Relay Support (not licensed)

use IP address in MSRP-path  use FQDN in MSRP-path

Authentication required

Access side only

Realms: **enviatel.net** **enviatel.net** **enviatel.net**

Qop: **AUTH**

Expire time/sec: **300**

Miscellaneous

Open external firewall pinhole

## Known limitations and restrictions:

- Provider does not analyze the Diversion header. That means the forwarding-to party gets only the number of the caller and not the forwarders number additionally.
- In rarely cases the provider presents a connected number in PAI for outgoing calls. This number is in national format and is shown on the OS4K phone. This number missed the PNAC (0). OS4K is not able to manipulate this number and add the PNAC in front.  
Therefore the Connected Party update must be disabled with the COT Parameter ANZR
- Provider does not support Fax via T.38
- Fax is transparent transmitted via G.711. Fax was not tested in the certification.

# Appendix

## Supported Numbering Formats

**Incoming: Called Party (TO):** National prefixed + national number (0...)

**Incoming: Calling Party (FROM):** International prefixed number (0049...)for international calls  
National prefixed number (0...)for national and local calls

**Incoming: Account (REQUEST):** Trunk Account number (in national format)

**Outgoing: Called Party (REQUEST, TO):** International prefixed number (0049...)for international calls  
National prefixed number (0...)for national and local calls

**Outgoing: Calling Party (PPI):** National prefixed number (0...)

**Outgoing: Account (FROM, Contact):** Trunk Account number (in national format)

**Outgoing Emergency Call - Called Party (REQUEST, TO):** <emergency number>

**Outgoing Emergency Call - Calling Party (PPI):** National prefixed number (0...)

**Outgoing Emergency Call - Account (FROM, Contact):** Trunk Account number (in national format)

**Outgoing OPTIONS:** Provider doesn't support receiving OPTIONS

**Incoming OPTIONS:** Provider doesn't monitor the trunk via OPTION



## Call examples:

### International Incoming Call:

```
> User Datagram Protocol, Src Port: 5060, Dst Port: 65060
v Session Initiation Protocol (INVITE)
  > Request-Line: INVITE sip:03429855850@172.18.216.68:65060;transport=udp SIP/2.0
  v Message Header
    Call-ID: 00938-LO-89025488-58821fb04@NM1-PBX.enviatel.net
    [Generated Call-ID: 00938-LO-89025488-58821fb04@NM1-PBX.enviatel.net]
    > Contact: <sip:193.98.115.6:5060>
    Content-Type: application/sdp
    > CSeq: 127295897 INVITE
    > From: "003232070880" <sip:003232070880@NM1-PBX.enviatel.net;user=phone>;tag=00938-NB-89025489-5a0277a75
    Max-Forwards: 58
    > Record-Route: <sip:193.98.115.6:5060;lr>;session=1172
    > To: <sip:034298558510@ngn-pbx.enviatel.net;user=phone>
    > Via: SIP/2.0/UDP 193.98.115.6:5060;branch=z9hG4bK-BAOC-1b354370-7936a4e5
    Allow: ACK,BYE,CANCEL,INFO,INVITE,REFER,REGISTER,UPDATE
    User-Agent: TAUTAS/01 (NBE)
    Content-Length: 246
  v Message Body
    v Session Description Protocol
      Session Description Protocol Version (v): 0
      > Owner/Creator, Session Id (o): anonymous 163540317600 163540317600 IN IP4 193.98.115.6
      Session Name (s): SIP Call
      > Connection Information (c): IN IP4 193.98.115.251
      > Time Description, active time (t): 0 0
      > Media Description, name and address (m): audio 36950 RTP/AVP 8 101
      > Bandwidth Information (b): AS:82
      > Media Attribute (a): rtpmap:8 PCMA/8000/1
      > Media Attribute (a): rtpmap:101 telephone-event/8000
      > Media Attribute (a): fmtp:101 0-15
      > Media Attribute (a):ptime:20
      Media Attribute (a): sendrecv
      [Generated Call-ID: 00938-LO-89025488-58821fb04@NM1-PBX.enviatel.net]
```

### National Incoming Call:

```
> User Datagram Protocol, Src Port: 5060, Dst Port: 65060
v Session Initiation Protocol (INVITE)
  > Request-Line: INVITE sip:03429855850@172.18.216.68:65060;transport=udp SIP/2.0
  v Message Header
    Call-ID: 30991-SW-868442fc-68d77ab12@NM1-PBX.enviatel.net
    [Generated Call-ID: 30991-SW-868442fc-68d77ab12@NM1-PBX.enviatel.net]
    > Contact: <sip:193.98.115.6:5060>
    Content-Type: application/sdp
    > CSeq: 85990614 INVITE
    > From: "089700720104" <sip:089700720104@NM1-PBX.enviatel.net;user=phone>;tag=30991-EZ-868442fd-1b0dd2873
    Max-Forwards: 56
    > Record-Route: <sip:193.98.115.6:5060;lr>;session=8037
    > To: <sip:034298558510@ngn-pbx.enviatel.net;user=phone>
    > Via: SIP/2.0/UDP 193.98.115.6:5060;branch=z9hG4bK-LBFN-16d5916b-02af52ea
    Allow: ACK,BYE,CANCEL,INFO,INVITE,REFER,REGISTER,UPDATE
    User-Agent: TAUTAS/01 (NBE)
    Content-Length: 246
  > Message Body
```

## National Outgoing Call:

```
> User Datagram Protocol, Src Port: 65060, Dst Port: 5060
√ Session Initiation Protocol (INVITE)
  > Request-Line: INVITE sip:089700720104@ngn-pbx.enviatel.net:5060;transport=udp SIP/2.0
  √ Message Header
    > Via: SIP/2.0/UDP 172.18.216.68:65060;branch=z9hG4bK00b8.b0301aba8c882e35f89d875f055445ef.0;i=16110
    Max-Forwards: 69
    > Contact: <sip:03429855850@172.18.216.68:65060;transport=udp>
    > To: <sip:089700720104@ngn-pbx.enviatel.net:5060;transport=udp>
    > From: <sip:03429855850@ngn-pbx.enviatel.net:65060;transport=udp>;tag=f718a1fe62
    Call-ID: b3837eb64a91d679
    [Generated Call-ID: b3837eb64a91d679]
    > CSeq: 319523306 INVITE
    Session-Expires: 1800
    Min-SE: 90
    Allow: INVITE, ACK, OPTIONS, BYE, CANCEL, REGISTER, REFER, INFO, PRACK, UPDATE
    Content-Type: application/sdp
    > [truncated]Proxy-Authorization: Digest username="03429855850",realm="NM1-PBX.enviatel.net",nonce="00853
    Supported: 100rel, timer
    User-Agent: OpenScape 4000 - SoftGate
    > P-Preferred-Identity: <sip:034298558510@172.18.216.68>
    Content-Length: 299
    > X-Siemens-OSS: OpenScape SBC V10 R1.04.02-1
  √ Message Body
    √ Session Description Protocol
      Session Description Protocol Version (v): 0
      > Owner/Creator, Session Id (o): MxSIP 131114 1741549248 IN IP4 172.29.93.180
      Session Name (s): SIP Call
      > Connection Information (c): IN IP4 172.18.216.68
      > Time Description, active time (t): 0 0
      > Media Description, name and address (m): audio 38940 RTP/AVP 8 100 101 0
      > Media Attribute (a): rtpmap:8 PCMA/8000
      > Media Attribute (a): rtpmap:100 telephone-event/8000
      > Media Attribute (a): fmp:100 0-15,32-36,49
      > Media Attribute (a): rtpmap:101 red/8000
      > Media Attribute (a): fmp:101 100
      > Media Attribute (a): rtpmap:0 PCMU/8000
      Media Attribute (a): sendrecv
      Media Attribute (a): rtcp-mux
      [Generated Call-ID: b3837eb64a91d679]
```

## Emergency Call:

```
> User Datagram Protocol, Src Port: 65060, Dst Port: 5060
< Session Initiation Protocol (INVITE)
  > Request-Line: INVITE sip:112@ngn-pbx.enviatel.net:5060;transport=udp SIP/2.0
  < Message Header
    > Via: SIP/2.0/UDP 172.18.216.68:65060;branch=z9hG4bK696b.12e7225cc762c0b5df59cd9255011ed3.0;i=8443
      Max-Forwards: 69
    > Contact: <sip:03429855850@172.18.216.68:65060;transport=udp>
    > To: <sip:112@ngn-pbx.enviatel.net:5060;transport=udp>
    > From: <sip:034298558510@ngn-pbx.enviatel.net:65060;transport=udp>;tag=235e5c3ff1
      Call-ID: 34d97cfa9a61b080
      [Generated Call-ID: 34d97cfa9a61b080]
    > CSeq: 708282156 INVITE
      Session-Expires: 1800
      Min-SE: 90
      Allow: INVITE, ACK, OPTIONS, BYE, CANCEL, REGISTER, REFER, INFO, PRACK, UPDATE
      Content-Type: multipart/mixed;boundary=level-0
    > [truncated]Proxy-Authorization: Digest username="03429855850",realm="NM1-PBX.enviatel.net",nonce="891050670ad4667c74f1
      Supported: 100rel, timer
      User-Agent: OpenScape 4000 - SoftGate
    > P-Preferred-Identity: <sip:034298558510@172.18.216.68>
      User-to-User: 000000303040f601f2ff4b41524c2d4c4945424b4e45434854205354522e20;encoding=hex;purpose=isdn-network;content-
      Geolocation-Routing: no
      Content-Length: 1076
    > X-Siemens-OSS: OpenScape SBC V10 R1.04.02-1
  > Message Body
```

## Registration:

### SBC -> Provider (Register)

```
> User Datagram Protocol, Src Port: 65060, Dst Port: 5060
v Session Initiation Protocol (REGISTER)
  > Request-Line: REGISTER sip:ngn-pbx.enviatel.net:5060;transport=udp SIP/2.0
  v Message Header
    > Via: SIP/2.0/UDP 172.18.216.68:65060;branch=z9hG4bK3ddf.906e9b466b67875a7374358f706285e0.0;i=3
      Expires: 480
      User-Agent: SIP alive check
      Call-ID: ff4251c4
      [Generated Call-ID: ff4251c4]
    > From: <sip:03429855853@ngn-pbx.enviatel.net>;tag=fe4f67e
    > CSeq: 1 REGISTER
      Max-Forwards: 70
    > To: <sip:03429855853@ngn-pbx.enviatel.net>
    > Contact: <sip:03429855853@172.18.216.68:65060;transport=udp>;expires=480
      Content-Length: 0
```

### Provider -> SBC (401 Unauthorized)

```
> User Datagram Protocol, Src Port: 5060, Dst Port: 65060
v Session Initiation Protocol (401)
  > Status-Line: SIP/2.0 401 Unauthorized
  v Message Header
    Call-ID: ff4251c4
    [Generated Call-ID: ff4251c4]
  > CSeq: 1 REGISTER
  v From: <sip:03429855853@ngn-pbx.enviatel.net>;tag=fe4f67e
    > SIP from address: sip:03429855853@ngn-pbx.enviatel.net
      SIP from tag: fe4f67e
  v To: <sip:03429855853@ngn-pbx.enviatel.net>;tag=00-07812-867e55ae-1ba8aead0
    > SIP to address: sip:03429855853@ngn-pbx.enviatel.net
      SIP to tag: 00-07812-867e55ae-1ba8aead0
  v Via: SIP/2.0/UDP 172.18.216.68:65060;received=172.18.216.68;rport=65060;branch=z9hG4bK3ddf.906e9b466b67875a7374358f706285e0.0
    Transport: UDP
    Sent-by Address: 172.18.216.68
    Sent-by port: 65060
    Received: 172.18.216.68
    RPort: 65060
    Branch: z9hG4bK3ddf.906e9b466b67875a7374358f706285e0.0
    i=3
  v WWW-Authenticate: Digest realm="NM1-PBX.enviatel.net",nonce="867e536f23fd566c41b4542336ea24d2"
    Authentication Scheme: Digest
    Realm: "NM1-PBX.enviatel.net"
    Nonce Value: "867e536f23fd566c41b4542336ea24d2"
    Opaque Value: "867dcea12c6d707"
    Stale Flag: false
    Algorithm: MD5
    Server: TAUTAS/01 (NBE)
    Content-Length: 0
```

### SBC -> Provider (Register)

```
> User Datagram Protocol, Src Port: 65060, Dst Port: 5060
< Session Initiation Protocol (REGISTER)
  > Request-Line: REGISTER sip:ngn-pbx.enviatel.net:5060;transport=udp SIP/2.0
  < Message Header
    > Via: SIP/2.0/UDP 172.18.216.68:65060;branch=z9hG4bK054d.4f409225d9e426ca5cba16e69ea34bdb.0;i=3
      Expires: 480
      User-Agent: SIP alive check
      Call-ID: 126901b6
      [Generated Call-ID: 126901b6]
    > From: <sip:03429855850@ngn-pbx.enviatel.net>;tag=8772f002
    > CSeq: 2 REGISTER
      Max-Forwards: 70
    > To: <sip:03429855850@ngn-pbx.enviatel.net>
    > Contact: <sip:03429855850@172.18.216.68:65060;transport=udp>;expires=480
      Content-Length: 0
  < [truncated]Authorization: Digest username="03429855850", realm="NM1-PBX.enviatel.net", nonce="8
    Authentication Scheme: Digest
      Username: "03429855850"
      Realm: "NM1-PBX.enviatel.net"
      Nonce Value: "867e536f23fd566c41b4542336ea24d2"
      Authentication URI: "sip:ngn-pbx.enviatel.net:5060;transport=udp"
      Digest Authentication Response: "793ebe7a11c7266cc9150924e5818de1"
      Algorithm: MD5
      Opaque Value: "867dcea12c6d707"
```

### Provider -> SBC (OK)

```
> User Datagram Protocol, Src Port: 5060, Dst Port: 65060
< Session Initiation Protocol (200)
  > Status-Line: SIP/2.0 200 OK
  < Message Header
    Call-ID: ff4251c4
    [Generated Call-ID: ff4251c4]
  < Contact: <sip:03429855853@172.18.216.68:65060>;expires=480
    > Contact URI: sip:03429855853@172.18.216.68:65060
      Contact parameter: expires=480
    > CSeq: 2 REGISTER
  < From: <sip:03429855853@ngn-pbx.enviatel.net>;tag=3a0536f9
    > SIP from address: sip:03429855853@ngn-pbx.enviatel.net
      SIP from tag: 3a0536f9
  < To: <sip:03429855853@ngn-pbx.enviatel.net>;tag=00-07560-867e565e-0cb7b6dc1
    > SIP to address: sip:03429855853@ngn-pbx.enviatel.net
      SIP to tag: 00-07560-867e565e-0cb7b6dc1
  < Via: SIP/2.0/UDP 172.18.216.68:65060;received=172.18.216.68;rport=65060;bran
    Transport: UDP
    Sent-by Address: 172.18.216.68
    Sent-by port: 65060
    Received: 172.18.216.68
    RPort: 65060
    Branch: z9hG4bK0ddf.a2adeef57fb1058d2dafd4fd301f6be5.0
    i=3
    Server: TAUTAS/01 (NBE)
    Content-Length: 0
```