

OpenScape 4000



How to Configure SIP Trunk for enviaTEL Germany

Table of Contents

Information.....	4
Certification	4
Trunk Configuration Data provided by envia-TEL	4
OpenScape 4000 Configuration.....	5
AMOs (Example).....	5
Gateway Configuration	7
SIP Parameter.....	7
Codec Parameter	9
SIP trunking Profile	11
Account-Name.....	11
OpenScape SBC Configuration	13
Media Profile	13
Enable Codec Support for transcoding.....	Fehler! Textmarke nicht definiert.
Remote Endpoint Configuration	15
Known limitations and restrictions:	16
Appendix.....	17
Supported Numbering Formats.....	17
Call examples:.....	18

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 OpenScape 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 <=> OpenScape 4000

Provider <=> OpenScape SBC <=> OpenScape 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)²:

Benutzername:	034200550050
Passwort:	[REDACTED]
Registrar:	ngn-pbx.enviatel.net
Ortskennzahl:	034298

Note:

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

OpenScape 4000 Configuration

See Service Documentation OpenScape 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,,0,0,0,0,0,N,1-1-130,KEINE;
M2 1 EINR-RICHT:LRTGNEU,2100,ALLE,"IP NOTRUF   ",211,1-1-
210,NEIN,NEIN,NEIN,NEIN,PP300,,,,,JA,JA,,1-1-211,NEUTRAL,NEIN,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,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,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,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	"SIP User Agent" settings ignored due to usage of SIP trunk profiles
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"/>
<input type="button" value="Apply"/> <input type="button" value="Undo"/>	

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:	<input type="checkbox"/>
Max. UDP Datagram Size for T.38 Fax (bytes):	<input type="text" value="375"/>
Error Correction Used for T.38 Fax (UDP)	t38UDPRedundancy ▼
Time Range for Immediate Switch to T.38 Fax (s):	<input type="text" value="0"/> 0 means: No Immediate Switching
Misc.	
ClearMode (ClearChannelData):	<input checked="" type="checkbox"/> Frame Size: <input type="text" value="20"/> msec
RFC2833	
Transmission of Fax/Modem Tones according to RFC2833:	<input checked="" type="checkbox"/>
Transmission of DTMF Tones according to RFC2833:	<input checked="" type="checkbox"/>
Payload Type for ClearChannel:	<input type="text" value="96"/>
Payload Type for RFC2833:	<input type="text" value="100"/>
Payload Type for RFC2198:	<input type="text" value="101"/> (= 'Payload Type for RFC2833' + 1)
Redundant Transmission of RFC2833 Tones according to RFC2198:	<input checked="" type="checkbox"/>
Payload Type for RFC4733 WideBand:	<input type="text" value="102"/> (= 'Payload Type for RFC2833' + 2)
<input type="button" value="Apply"/> <input type="button" value="Undo"/>	

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.enviatel.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"/>	
IP Address / Host name: <input type="text" value="ngn-pbx.enviatel.net"/>	
Specify Port: <input type="checkbox"/>	
Reregistration Interval (sec): <input type="text" value="480"/>	
Proxy	
IP Address / Host name: <input type="text" value="ngn-pbx.enviatel.net"/>	
Specify Port: <input type="checkbox"/>	
Outbound Proxy	
Use Outbound Proxy: <input type="checkbox"/>	
IP Address / Host name: <input type="text"/>	
Specify Port: <input type="checkbox"/>	
Port: <input type="text" value="0"/>	
Inbound Proxy	
Use Inbound Proxy: <input type="checkbox"/>	
IP Address / Host name: <input type="text"/>	
Specify Port: <input type="checkbox"/>	

SIP trunking Profile – connectivity via OS – SBC

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

Account-Name

Account Name: Benutzername

Authorization Name: Benutzername

Password: Passwort

Account Name

Account Name:

Authorization name:

Provider Name:

New Password:

Confirm Password:

Details für die Einrichtung des Telefoniedienstes (SIP):

Benutzername:

Passwort:

Registrar:

Ortskennzahl:

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 **Add**

Move up **Move down** **Delete**

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

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:

Type: SSP

Profile: EnviaTel

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: 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):

INVITE No Reply timeout (msec):

TLS

TLS mode:

Certificate profile:

☐ TLS keep-alive

Keep-alive interval (seconds):

Keep-Alive timeout (sec):

Media Configuration

Media profile: RTPonly

Media realm subnet IP address:

Outbound Proxy Configuration

Outbound Proxy:

Outbound Proxy Port:

Registrar Server Configuration

Registrar Server:

Registrar Server Port:

Remote Location Identification/Routing

Core FQDN:

Core realm port: 53180

Default core realm location domain name:

Default home DN: 0342985050

Incoming Routing prefix:

Digest Authentication

☒ Digest authentication supported

Digest authentication realm:

Digest authentication user ID: 0342985050

Digest authentication password:

Access Side Firewall Settings

☐ Enable Firewall Settings

Emergency configuration

Emergency numbers:

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:

Qop:

Expire time/sec:

☒ Open external firewall pinhole

Details für die Einrichtung des Telefoniedienstes (SIP):

Benutzername: 0342985050

Passwort: [REDACTED]

Registrar: ngn-pbx.enviatel.net

Ortskennzahl: 034298

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): ptm: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: <cid:10@172.29.93.180>
    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
✓ 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=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
✓ Session Initiation Protocol (401)
  > Status-Line: SIP/2.0 401 Unauthorized
  ✓ Message Header
    Call-ID: ff4251c4
    [Generated Call-ID: ff4251c4]
    > CSeq: 1 REGISTER
    ✓ From: <sip:03429855853@ngn-pbx.enviatel.net>;tag=fe4f67e
      > SIP from address: sip:03429855853@ngn-pbx.enviatel.net
      SIP from tag: fe4f67e
    ✓ 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
    ✓ Via: SIP/2.0/UDP 172.18.216.68:65060;received=172.18.216.68;rport=65060;branch=z9hG4bK3ddf.906e9b466b67875a7374358f706285e0.0;i=3
      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
    ✓ 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
```

