

OpenScape Voice V10  
OpenScape SBC V10  
OpenScape Branch V10



## How to Configure SIP Trunk of Deutsche Telekom Company Flex

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**Tested Scenario:** Provider <-> OS SBC / OS Branch <-> OS Voice

**Provider Connectivity:** TCP or encrypted TLS SIP Trunk

**Provider Registration:** Registration Mode

## Foreword

This 'How to' Guide describes the configuration of different setups in different chapters:

- Unencrypted TCP SIP Trunk between OS SBC / OS Branch and Deutsche Telekom CompanyFlex
- TLS encrypted SIP Trunk between OS SBC / OS Branch and Deutsche Telekom CompanyFlex

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## Table of History

<b>Date</b>	<b>Version</b>	<b>Changes</b>
08.11.2022	0.1	Initial version
16.11.2022	0.9	Ready for review
25.11.2022	1.0	Final release

## Trunk Configuration Data provided by Deutsche Telekom

### Configuration Documents:

- Technical Specification of the SIP-Trunking Interface of CompanyFlex of Deutsche Telekom, 1TR119, <https://www.telekom.de/hilfe/downloads/1tr119.pdf>
- How to Configure SIP Trunk for CompanyFlex (Telekom Deutschland GmbH), CompanyFlex-OSV-OSSBC-Trunk-Configuration.pdf by Deutsche Telekom

The configuration data required to setup the SIP trunk can be seen in the Business Service Portal of Deutsche Telekom as it looks like below:

### SIP-Trunk Übersicht

The screenshot displays the 'SIP-Trunk Übersicht' (SIP Trunk Overview) page. On the left, there is a vertical list of configuration sections, each with an edit icon:

- Name:** SIP-Trunk Name: UNIFY-MCH-CSL
- Kontakt:** Administrator:
- SIP-Trunk Profil:** Zugeordnetes SIP-Trunk Profil: Standard (SIP Connect 1.1)
- Gebuchte Zusatz-Pakete:**
- Verkehrssteuerung SIP-Trunk:** Max. Anzahl: 4, Max. eingehend: 2, Max. abgehend: 2
- Limitierung Parallele Gespräche:** Maximale Anzahl: Keine Limitierung
- TK-Anlagen Rufaufbauüberwachung:** Wert: 10 Sekunden

On the right, the 'Telefonie-Anmeldedaten' (Telephony Registration Data) section is highlighted in yellow. It contains the following information:

- SIP-Domain: tel.t-online.de
- Outbound-Proxy: 55XXXXXXXXXX.primary.companyflex.de, 55XXXXXXXXXX.secondary.companyflex.de
- Registrar: tel.t-online.de
- Registrierungsrufnummer: +4919929600000XXXXXX
- Telefonie-Benutzername: +4919929600000XXXXXX@tel.t-online.de
- Telefonie-Passwort: XXXXXXXX
- Neues Passwort generieren (button)

The advanced parameter 'Rufaufbauüberwachung' should be increased e. g. to 10 seconds. It specifies the time the provider waits for a response from a called PSTN phone. For mobile phones the default value 3 is a too short time span.

## OpenScope Voice Resilient Telco Platform (RTP) Parameter

The OpenScope Voice RTP parameter Srx/Sip/UpdateMethodSessionTimingEnable has to be set to RtpTrue. This cause OSV to send UPDATE SIP messages instead of INVITE for Session Updates.

The screenshot shows the 'Unify OpenScope Common Management Platform' interface. The user is logged in as 'administrator@system'. The navigation menu includes 'Configuration', 'Maintenance', 'User Management', 'Fault Management', 'Performance Management', and 'Accounting'. The main content area is titled '[susi] - RTP Management' and contains a search bar with the text 'Srx/Sip/UpdateMethodSess'. Below the search bar, there is a table with the following data:

Name	Value	Default Value	Is Defa
Srx/Sip/UpdateMethodSessionTimingEnable	RtpTrue	RtpFalse	No

## Configuration of unencrypted SIP Trunk

### OpenScope Voice Configuration

#### OpenScope SBC Endpoint Profile

The screenshot shows the 'Edit Endpoint Profile' configuration page for 'EPP\_SBC\_SSP\_Test'. The 'General' tab is active. The fields are as follows:

- Name: EPP\_SBC\_SSP\_Test
- Remark: (empty text area)
- Numbering Plan: NP\_DeutschlandLAN
- Management Information: (empty fields for Class of Service, Routing Area, Calling Location, Time Zone, SIP Privacy Support, Failed Calls Intercept Treatment, Language, and Impact Level)

The screenshot shows the 'Edit Endpoint Profile' configuration page for 'EPP\_SBC\_SSP\_Test' with the 'Endpoints' tab active. It displays a table of endpoints currently assigned to this profile:

Name	Type	Registered	Primary
EP_SBC_SSP_Test	Static	Yes	192.168.110.58
EP_SBC_DlandLAN	Static	Yes	192.168.110.58
EP_SBC_CompFlex	Static	Yes	sbcv10-ssp2.gcsstg.net

The screenshot shows the 'Edit Endpoint Profile' configuration page for 'EPP\_SBC\_SSP\_Test' with the 'Services' tab active. The services are configured as follows:

- Message Waiting: No
- Call Transfer: Yes
- Call Forward Invalid Destination: No
- Toll and Call Restrictions: No
- Park to Server: No
- CSTA Network Interface Device: No

## OpenScope SBC Signaling Endpoint

### General Tab

- **Profile:** Use the Endpoint Profile configured before
- **Default Home DN:** Specify one number of the DID range, best use the first number from this range

The screenshot shows the 'General' tab of the 'Edit Endpoint' configuration window. The endpoint name is 'EP\_SBC\_CompFlex'. The 'Registered' checkbox is checked. The profile is set to 'EPP\_SBC\_SSP\_Test'. The default home DN is '49711388800'. The last update timestamp is '2022-10-06 09:51:29.0'. The associated subscriber DN is also '49711388800'. There are 'Save' and 'Cancel' buttons at the bottom right.

### SIP Tab

- **Endpoint Address:** OS SBC LAN IP or FQDN
- **Port:** OS SBC Remote Endpoint Core Port
- **Transport protocol:** TCP
- **Trusted:** OS SBC Remote Endpoint Core Port

The screenshot shows the 'SIP' tab of the 'Edit Endpoint' configuration window. The 'SIP Trunking' radio button is selected. The signaling address type is 'IP Address or FQDN' and the endpoint address is 'sbcv10-ssp2.gcsstg.net'. The port is '50005' and the transport protocol is 'TCP'. The 'SRTP media mode' is set to 'Enabled'. There are 'Save' and 'Cancel' buttons at the bottom right.

The screenshot shows the 'Attributes' tab of the 'Edit Endpoint' configuration window. The 'ANAT Support', 'ICE Support', and 'DTLS Support' are all set to 'Enabled'. 'SIP UA Forking Support' is set to 'None'. There are checkboxes for 'Use Proxy/SBC Best-Effort SRTP settings for calls to subscribers: AS-SIP Interface', 'Red Sky E911 Manager node', and 'Treat endpoint as secure'. The 'Trusted' checkbox is checked, and the ports are '50005-50005'. There are 'Save' and 'Cancel' buttons at the bottom right.

## SIP Security – Trusted on SIP Tab

- **Signaling Primary:** OS SBC LAN IP
- **Port Range:** OS SBC Remote Endpoint Core Port
- **Local Realm:** tel.t-online.de

### SIP-Trunk Übersicht

<b>Name</b>	SIP-Trunk Name: UNIFYMCHCSL
<b>Kontakt</b>	Administrator
<b>SIP-Trunk Profil</b>	Zugeordnetes SIP-Trunk Profil: Standard (SIP Connect 1.1)
<b>Gebuchte Zusatz-Pakete</b>	
<b>Verkehrssteuerung SIP-Trunk</b>	Max. Anzahl: 4 Max. eingehend: 2 Max. abgehend: 2
<b>Limitierung Parallele Gespräche</b>	Maximale Anzahl: Keine Limitierung
<b>Tk-Anlagen Rufaufbauüberwachung</b>	Wert: 10 Sekunden

<b>Telefonie-Anmeldedaten</b>
SIP-Domain: tel.t-online.de
Outbound-Proxy: 550000000000.primary.complex.flex.de
550000000000.secondary.complex.flex.de
Registrar: tel.t-online.de
Registrierungsnummer: +491992960000000000000000
Telefonie-Benutzername: +491992960000000000000000@tel.t-online.de
Telefonie-Passwort: *****
<a href="#">Neues Passwort generieren</a>

### [osvv10] - SIP Configuration

In this section you can configure Realm attributes, Port(s) e.g. 4713-4717, REALM, User and Password.

**Security**

Signaling Primary: sbcv10-ssp2.gcsstg.net

Signaling Port:

Trusted entity:

All Ports  
 Port Range

Port Range: 50005-50005

Local Realm: tel.t-online.de

Local User Name: +49 \*\*\*\*\* 60€

Local Password: \*\*\*\*\*

Confirm Local Password: \*\*\*\*\*

Remote Realm:

Remote User Name:

Remote Password:

Confirm Remote Password:

## Attributes Tab

Enabled SIP Attributes:

- **Public/Offnet Traffic**
- **Send International Numbers in Global Number Format (GNF)**
- **Send Authentication Number in P-Asserted-Identity header**
- **Use Subscriber Home DN as Authentication Number**
- **Enable Session Timer**

## Aliases Tab

- OS SBC LAN IP or FQDN, optional additional OS SBC Remote Endpoint Core Port

### [osvv10] - [BG\_V10] - [Main Office] - Edit Endpoint : EP\_SBC\_CompFlex

General SIP Attributes Aliases Routes Accounting

Aliases

You can associate here aliases with a SIP Endpoint.

[Add...](#) [Delete](#)

Set: 0 | Items/Page: 200 | All: 2

<input type="checkbox"/>	Name
<input type="checkbox"/>	192.168.110.58:50005
<input type="checkbox"/>	sbcv10-ssp2.gcsstg.net



## OpenScope Branch Endpoint

### SIP Security – Trusted on SIP Tab

- **Signaling Primary:** OS Branch LAN IP
- **Port Range:** All Ports
- **Local Realm:** tel.t-online.de

#### SIP-Trunk Übersicht

Name	SIP-Trunk Name: UNIFYMCH CSL
Kontakt	Administrator:
SIP-Trunk Profil	Zugeordnetes SIP-Trunk Profil: Standard (SIP Connect 1.1)
Gebuchte Zusatz-Pakete	
Verkehrssteuerung SIP-Trunk	Max. Anzahl: 4 Max. eingehend: 2 Max. abgehend: 2
Limitierung Parallele Gespräche	Maximale Anzahl: Keine Limitierung
TK-Anlagen Rufaufbauüberwachung	Wert: 10 Sekunden

Telefonie-Anmeldedaten
SIP-Domain: tel.t-online.de
Outbound-Proxy: 550000000000,primary,unifyflex.de 550000000000,secondary,complextel.de
Registrar: tel.t-online.de
Registrierungsnummer: +491902960000000000000000
Telefonie-Benutzername: +491902960000000000000000@tel.t-online.de
Telefonie-Passwort: xxxxxxxxxx <a href="#">Neues Passwort generieren</a>

### [osvv10] - SIP Configuration

In this section you can configure Realm attributes, Port(s) e.g. 4713-4717, REALM, User and Password.

#### Security

Signaling Primary:	192.168.111.213
Signaling Port:	
Trusted entity:	<input checked="" type="checkbox"/>
	<input checked="" type="radio"/> All Ports <input type="radio"/> Port Range
Port Range:	
Local Realm:	tel.t-online.de
Local User Name:	+49: 160€
Local Password:	••••••••
Confirm Local Password:	••••••••
Remote Realm:	
Remote User Name:	
Remote Password:	
Confirm Remote Password:	

# OpenScope SBC Configuration

**Unify OpenScope Session Border Control Management Portal** | User name: administrator | Atos

Atos Unify OpenScope SBC

**Administration**

- System
- Network/Net Services
- VoIP
- Features
- Security
- Diagnostics & logs
- Alarms
- Maintenance

**General - SBCSSP**

SBC aggregated information and data.

**Alarms**

Alarm summary: Critical: 0 Major: 0 Minor: 0 [Show alarm details](#)

**System Status**

Branch mode: Centralized SBC | Auto refresh timer: 1 min

Operational state: normal

**Com Node 1**

Primary server: 192.168.115.102 | Penalty box state: Active

Backup server: | Penalty box state:

**Com Node 2**

Primary server: 192.168.115.103 | Penalty box state: Active

Backup server: | Penalty box state:

**Services status** [Show](#) | **Registered subscribers** [Show](#)

**SSP status** [Show](#) | **Dynamic port mapping** [Show](#)

**Dynamic IP remote endpoints** [Show](#) | **Denial of Service Mitigation** [Show](#)

**TURN Allocations** [Show](#) | **Telephony Connector status** [Show](#)

**SIP Loadbalancer status** [Show](#)

**System Info**

CPU: 3.58 % - 2 x 2400 MHz

Memory: 30.23 % - 2 Gb

Disk: 10.92 % - 42 Gb

System uptime: 12 days 19:02

Hardware type: Virtual OSS 250

Hostname: SBCSSP

**Software Info**

Software version: V10 R2.05.00

Software Partition information: [Active](#) [Backup](#)

## VOIP – Sip Server Settings Tab

- **Primary server:** OSV SIP TCP IP
- **Transport:** TCP
- **Port:** 5060

**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: Active-Standby

Allow Register from SERVER

[Other trusted servers](#)

**Node 1**

Target type: Binding

**Primary server:** 192.168.115.102 | Transport: TCP | Port: 5060

Backup server: | Transport: TCP | Port:

SRV record: | Transport: TCP

**Node 2**

Target type: Binding

**Primary server:** 192.168.115.103 | Transport: TCP | Port: 5060

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):** 50

## VOIP – Port and Signaling Settings Tab

**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:  Port max:  Time to live (sec):

Enable Media Specific Ports

Audio Port min:  Audio Port max:

Video Port min:  Video Port max:

Subscribers dynamic SIP ports

Port min:  Port max:

Remote Endpoints Static SIP Ports

Port min:  Port max:  Number of reserved SIP ports:

TCP/BFCP ports

Port min:  Port max:

---

**Signaling and Transport Settings** ?

TCP connect timeout (sec):  TCP send timeout (sec):

TCP connection lifetime (sec):   TCP keep alive

BFCP connection timer (min):

Maximal call session time (hr):

---

**Miscellaneous** ?

SIP SSL single context

## VOIP – Media Profile Tab

- **Core Media Protocol:** RTP only
- **Codecs Access Side:** At least G.711A and one additional codec

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	<input checked="" type="checkbox"/>
WE_Phone_default		Best Effort SRTP	mikey + sdes	
SSP_Media	G711A,G711U,G722,G729	RTP only	none	
Unify_Phone_default		SRTP only	dtls	

**Cloud Support**

- Support OpenScope 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: SSP\_Media

Media protocol: RTP only  Direct Media Support

- Support ICE
- Support NGTC Trickle ICE
- Enable NGTC WebRTC Compatibility
- 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**

RTCP Mode: 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	Auto
2	G711U 8 kHz - 64 kbps	Auto
3	G722 8 kHz - 64 kbps	Auto
4	G729 8 kHz - 8 kbps	Auto

## SIP Service Provider Profile

### - Default SSP profile: DTAG/CompanyFlex

**SIP Service Provider Profile**

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

**General**

Name: CompanyFlex Default SSP profile: DTAG/CompanyFlex

**SIP Service Address**

Use SIP Service Address for identity headers  
**SIP service address** tel.t-online.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)** 480

**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 displ.

**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  
 1TR118

## Remote Endpoint Configuration

- **Profile: SIP Service:** Provider Profile configured before
- **Signaling address type:** DNS SRV

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: CompanyFlex

Type: SSP

Profile: CompanyFlex

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:

Enable access control

Signaling address type: DNS SRV

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	tel.t-online.de	0	TCP		SSP_Media	Server authenticat	OSV Solution	<input type="checkbox"/>	120	10	360000	3000	92	.primary.company	0	tel.t-online.de	0

Remote Location Identification/Routing

Core FQDN: sbcv10-ssp2.gcsdn.net

Core realm port: 50005

Default core realm location domain name:

Default home DN: +49.....

## SIP-Trunk Übersicht

Name: SIP-Trunk Name: UNIFYMCH-CSL

Kontakt: Administrator:

SIP-Trunk Profil: Zugeordnetes SIP-Trunk Profil: Standard (SIP Connect 1.1)

Gebuchte Zusatz-Pakete:

Verkehrssteuerung SIP-Trunk: Max. Anzahl: 4, Max. eingehend: 2, Max. abgehend: 2

Limitierung Parallele Gespräche: Maximale Anzahl: Keine Limitierung

TK-Anlagen Rufaufbauüberwachung: Wert: 10 Sekunden

Telefonie-Anmeldedaten

SIP-Domain: tel.t-online.de

Outbound-Proxy: 55XXXXXX.XXXXX,primary.companyflex.de  
55XXXXXX.XXXXX,secondary.companyflex.de

Registrar: tel.t-online.de

Registrierungsnummer: +4919929600000XXXXXX

Telefonie-Benutzername: +4919929600000XXXXXX@tel.t-online.de

Telefonie-Passwort: XXXXXXXX

## Remote Location Domain

- **Remote URL:** tel.t-online.de
- **Media profile:** Media profile for access side configured before
- **Registrar Server:** tel.t-online.de

**Remote Location Domain** ?

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

**General** ?

Remote URL:   Shared domain  
 Remote port:   
 Remote transport: TCP

**Signaling** ?

INVITE No Answer timeout (msec):   
 INVITE No Reply timeout (msec):

**TLS** ?

TLS mode: Server authentication  
 Certificate profile: OSV Solution  
 TLS keep-alive  
 Keep-alive interval (seconds):   
 Keep-Alive timeout (sec):

**Media Configuration** ?

Media profile: SSP\_Media  
 Media realm subnet IP address:

**Outbound Proxy Configuration** ?

Outbound Proxy:   
 Outbound Proxy Port:

**Registrar Server Configuration** ?

Registrar Server:   
 Registrar Server Port:

### SIP-Trunk Übersicht

Name ✎  
SIP-Trunk Name: UNIFY-MCH-CSL

Kontakt ✎  
Administrator:

SIP-Trunk Profil ✎  
Zugeordnetes SIP-Trunk Profil: Standard (SIP Connect 1.1)

Gebuchte Zusatz-Pakete ✎

Verkehrssteuerung SIP-Trunk ✎  
 Max. Anzahl: 4  
 Max. eingehend: 2  
 Max. abgehend: 2

Limitierung Parallele Gespräche ✎  
 Maximale Anzahl: Keine Limitierung

TK-Anlagen Rufaufbauüberwachung ✎  
 Wert: 10 Sekunden

Telefonie-Anmeldedaten ⊞

SIP-Domain: tel.t-online.de  
 Outbound-Proxy 55XXXXXXX.primary.companyflex.de  
55XXXXXXXXX.secondary.companyflex.de  
 Registrar: tel.t-online.de  
 Registrierungsnummer: +491992960000000000000000  
 Telefonie-Benutzername: +491992960000000000000000@tel.t-online.de  
 Telefonie-Passwort: XXXXXXXX Neues Passwort generieren

## Remote Endpoint Configuration (cont'd)

- **Send RTP dummy packets: Enabled**

The screenshot displays the 'Remote endpoint configuration' interface. The 'Digest Authentication' section is active, showing the following fields:

- Digest authentication supported:**
- Digest authentication realm:** tel.t-online.de
- Digest authentication user ID:** +49 @tel.t-onlin
- Digest authentication password:** [Redacted]

The 'Emergency call routing' section is also visible, featuring an 'Emergency call routing' button. The 'MSRP Data Configuration' section includes options for 'Enable MSRP Relay Support' (noted as 'not licensed'), 'use IP address in MSRP-path', 'use FQDN in MSRP-', 'Authentication required', and 'Access side only'. The 'Miscellaneous' section has 'Open external firewall pinhole' and 'Send RTP dummy packets' checked.

Two overlays are present on the right side:

- SIP-Trunk Übersicht:** A summary panel showing details for the SIP-Trunk, including Name, Kontakt, SIP-Trunk Profil, and various traffic and parallel call limits.
- Telefonie-Anmeldeinformationen:** A yellow overlay showing registration details for the SIP-Trunk, including SIP-Domain, Contact-Proxy, Registrar, and Telephone-Password. A 'Neues Passwort generieren' button is visible.

Red arrows point from the 'Digest authentication user ID' and 'Digest authentication password' fields in the main configuration to their respective fields in the 'Telefonie-Anmeldeinformationen' overlay.



## OpenScope Branch Configuration

OpenScope Branch is configured to run in SBC-Proxy mode.

The screenshot displays the OpenScope Branch configuration interface for a device with ID OSB50I-50-DlandLAN-8205899. The interface is divided into several sections:

- General:** Shows "Branch aggregated information and data."
- Alarms:** Displays an alarm summary with 0 Critical (red), 0 Major (orange), and 0 Minor (yellow) alarms. A "Show alarm details" button is present.
- System Status:** Shows the branch mode as "SBC-Proxy" and an auto refresh timer set to "20 seconds". It lists operational state as "normal" and details for two Com Nodes (Com Node 1 and Com Node 2), each with primary and backup servers and their respective penalty box states (Active).
- System Info:** Provides hardware and software details, including CPU usage (9.97%), Memory usage (49.31% - 2 Gb), Disk usage (44.06% - 8 Gb), system uptime (3 days 5:14), hardware type (Advantech 50i (BRI - FXS)), and hostname (OSB50I-50-DlandLAN-8205899).
- Software Info:** Shows the software version as V10 R2.05.00 and software partition information as Active, with a Backup button.
- Services status:** A grid of buttons to show various service statuses: Registered subscribers, SSP status, Link Status, Dynamic port mapping, Denial of Service Mitigation, and Subscriber data.

## VOIP – SIP Server Settings Tab

- **Primary server:** OSV SIP TCP IP

- **Transport:** TCP

- **Port:** 5060

Sip Server Settings	Port and Signaling Settings	Manipulation and Routing	Error Codes	Media	
<b>General</b>					
Comm System Type	Collocated				
<b>OPTIONS source port</b>	5060				
IP Version Towards SIP Server	IPv4				
<input type="checkbox"/> Enable path tagging					
<input type="checkbox"/> Branch behind SBC					
<input type="checkbox"/> Branch behind NAT					
<input type="checkbox"/> Synch subscriber data					
<input type="checkbox"/> Disable notification in survivable mode					
<input type="checkbox"/> Enforce minimum Subscriber TransportType Security					
<input checked="" type="checkbox"/> Disable Basic Security Check for LAN					
<input type="checkbox"/> Send CANCEL on Unreplied Branches					
<input type="checkbox"/> SDP ANAT Bypass					
<a href="#">Other trusted servers</a> <a href="#">Load Balance Mapping Table</a>					
<b>Node 1</b>					
Target type	Binding				
<b>Primary server</b>	192.168.115.102	Transport	TCP	Port	5060
Backup server		Transport	TCP	Port	
SRV record		Transport	TCP		
<b>Node 2</b>					
Target type	Binding				
<b>Primary server</b>	192.168.115.103	Transport	TCP	Port	5060
Backup server		Transport	TCP	Port	
SRV record		Transport	TCP		
<b>Timers and Thresholds</b>					
<b>Failure threshold (pings)</b>	2	<b>OPTIONS interval (sec)</b>	60		

## VOIP – Port and Signaling Settings Tab

**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** | Manipulation and Routing | Error Codes | Media

---

**Port Range** ?

Media independent RTP ports  
Port min:  Port max:  Time to live (sec):

Subscribers dynamic SIP ports  
Port min:  Port max:

Gateways/trunks static SIP ports  
Port min:  Port max:

TCP/BFCP ports  
Port min:  Port max:

---

**Signaling and Transport Settings** ?

**INVITE No Answer timeout - Normal Mode (ms)**  **INVITE No Answer timeout - Survival Mode (ms)**

Disable answer supervision for emergency calls line

**TCP connect timeout (sec)**  **TCP send timeout (sec)**

**TCP connection lifetime (sec)**   TCP keep alive

**BFCP connection timer (min)**

---

**Miscellaneous** ?

SIP SSL single context

## VOIP – Media Tab

- **Media protocol:** For Media Profile igw\_features\_lan RTP only

- **Codec List:** At least G.711A on core and G.711A on access side

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 | Manipulation and Routing | Error Codes | **Media**

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

LAN/WAN Media Configuration

Media profile: igw\_features\_lan

Media Profiles

Add Edit Delete

Name	Codecs	Media protocol	SRTP crypto context negotiation	Mark SRTP Call-leg as Secure
default		Strict Pass-Thru	none	
igw_features_lan	G711A,G711U,G729	RTP only	none	
TelecomDLan	G711A,G711U	RTP only	none	

Media Profile

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

General

Name: igw\_features\_lan

Media protocol: RTP only

SRTP configuration

SRTP crypto context negotiation:  HKEY  SCES  SCES both

Mark SRTP Call-leg as Secure

RTCP configuration

RTCP Mode: 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/3000

Codec:  Add

Priority	Codec	Packetization interval
1	G711A 8 kHz - 64 kbps	20
2	G711U 8 kHz - 64 kbps	20
3	G729 8 kHz - 8 kbps	20

Media Profile

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

General

Name: TelecomDLan

Media protocol: RTP only

SRTP Compatibility Mode

SRTP configuration

SRTP crypto context negotiation:  HKEY  SCES  SCES AES-128 only

Mark SRTP Call-leg as Secure

RTCP configuration

RTCP Mode: 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/3000

Codec: G729 8 kHz - 8 kbps Add

Priority	Codec	Packetization interval
1	G711A 8 kHz - 64 kbps	20
2	G711U 8 kHz - 64 kbps	20

## Features

**Features**

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

**Features Available in Normal Mode and Survivability Mode**

Enable gateways/trunks Configure

Enable integrated gateway Configure

Sip Service Provider profiles Configure

Enable auto attendant **(not licensed)** Configure

Enable Voice Mail Service **(not licensed)** Configure

Enable phone software management Configure

Enable Media Server / Streaming Configure

Enable LAN-WAN media interwork Configure

Enable Codec Support for transcoding Configure

Emergency calling Configure

**Features Available in Survivability Mode Only**

Multi-line Hunt Groups Configure

Call Forwarding Configure

Enable Call Detail Records Configure

Enable Music On Hold For Gateways ▼

Use PAI/PPI as ISDN Calling Party Number

System calling number suppression access code:

## SIP Service Provider Profile

**SIP Service Provider Profile**

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

**General**

Name:  Default SSP profile:

Allow sending of insecure Referred-By header  Send authentication number in Diversion header

Send P-Preferred-Identity rather than P-Asserted-Identity  Send authentication number in P-Asserted-Identity header

Do not send Diversion header  Send authentication number in From header

Send URI in telephone-subscriber format  Include restricted numbers in From header

**SIP Service Address**

Use SIP Service Address for identity headers

SIP service address:  Privacy support:

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

**Home DN**

Mandatory default home DN  Mandatory default home DN - Normal Mode

Default home DN:

**SIP User Agent**

SIP User Agent towards SSP:  SIP User Agent:

**Registration**

Registration required

Registration interval (sec):

**Flags**

FQDN in TO header to SSP

REFER supported by SSP

FQDN in Request-URI to SSP

Use To DN to populate the RURI

Send Default Home DN in Contact for Call messages

Request-URI user in TO header to SSP

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 local MOH in Survivable Mode

Force direction attribute to sendrcv

**TLS**

TLS Signaling:

**Sip Connect**

Use tel URI

Send user=phone in SIP URI

Registration mode

## Gateways/Trunks

Row	Signaling address type	Remote URL	Port	Interface	Transport	Mapped port	Routing prefix	Gateway / Trunk type	Functional type	Trunk profile	Output digit strip	Output digit add	Priority
1	DNS SRV	tel.t-online.de	0	WAN	TCP	21000,21001,;	%	SIP Trunk	All Modes Egress/Ingress	CompanyFlex	0		1

## Gateway Configuration

- **Signaling address type:** DNS SRV
- **Remote URL:** tel.t-online.de
- **Transport:** TCP
- **Mapped port:** 10 ports in a row

### Gateway Configuration

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

DNS dynamic refresh interval (min)

Route to R-URI domain

**General**

**Signaling address type**

**Remote URL**

**Port:**

**Interface**

**Transport**

**Mapped port**

**Routing prefix**

**Gateway / Trunk type**

**Functional type**

**Trunk profile**

**Output digit strip**

**Output digit add**

**Priority**

Operational Mode in OPTIONS Response

**Signaling**

**INVITE no answer timeout - Normal Mode (sec)**

**INVITE no answer timeout - Survivable Mode (sec)**

**INVITE no reply timeout - Normal Mode (sec)**

**INVITE no reply timeout - Survivable Mode (sec)**

**Digest Authentication**

Digest authentication supported

**Digest authentication realm**

**Digest authentication user ID**

**Digest authentication password**

### SIP-Trunk Übersicht

Name

Kontakt

SIP-Trunk Profil

Gebühren Zusatzpakete

Verkehrskategorie SIP-Trunk

Min. Anzahl: 4  
Max. eingehend: 1  
Max. ausgehend: 2

Limitierung Parallele Gespräche

Maximale Anzahl: Keine Limitierung

TK-Anlagen Rufaufbauüberwachung

Wert: 10 Sekunden

**Telefonie-Anmeldeinformationen**

SIP-Domäne: tel.t-online.de  
Outbound-Proxy URI: 55XXXXXXXXXX.primary.companyflex.de  
55XXXXXXXXXX.secondary.companyflex.de  
Registrar: tel.t-online.de  
Registrierungsnummer: +491992960000000000000000  
Telefonie-Schlüsselnummer: +491992960000000000000000@tel.t-online.de  
TK-ANMELDENUMMER: VXXXXXXXXX

## Gateway Configuration (cont'd)

**Media Configuration**

Media profile: TelekomDLan

Media realm subnet IP address:

Media realm subnet mask:

Anchoring media: Forced

Force media anchoring on transcoding

Record calls from this Gateway/Trunk

Allow Asymmetric RTP

---

**Outbound Proxy Configuration**

Outbound Proxy: 9  primary.companyflex.de

Outbound Proxy Port:

---

**Registrar Server Configuration**

Registrar Server:

Registrar Server Port:

---

**Miscellaneous**

Open external firewall pinhole

Send RTP dummy packets

### SIP-Trunk Übersicht

Name: SIP-Trunk Name: UNIFYMCH-CSL

Kontakt: Administrator

SIP-Trunk Profil: Zugeordnetes SIP-Trunk Profil: Standard (SIP Connect 1.1)

Gebuchte Zusatz-Pakete

Verkehrssteuerung SIP-Trunk: Max. Anzahl: 4  
Max. eingehend: 2  
Max. abgehend: 2

Limitierung Parallele Gespräche: Maximale Anzahl: Keine Limitierung

TK-Anlagen Rufaufbauüberwachung: Wert: 10 Sekunden

Telefonie-Anmeldedaten

SIP-Domain: tel.t-online.de

Outbound-Proxy: 55XXXXXXXXXX.primary.companyflex.de  
55XXXXXXXXXX.secondary.companyflex.de

Registrar: tel.t-online.de

Registrierungsrufnummer: +491992960000000000000000

Telefonie-Benutzername: +491992960000000000000000@tel.t-online.de

Telefonie-Passwort: XXXXXXXXXX

Neues Passwort generieren

# Configuration of TLS encrypted SIP Trunk

## OpenScape Voice Configuration

OpenScape SBC Endpoint Profile

### General Tab

- **SIP Privacy Support: Full Send**

The screenshot shows the 'General' tab of the 'Edit Endpoint Profile' configuration page. The title bar reads '[susi] - [DTAG] - Edit Endpoint Profile : EPP\_SBC\_DTAG'. Below the title bar, there are three tabs: 'General', 'Endpoints', and 'Services'. The 'General' tab is active. The page contains several sections: 'Endpoint Profile' with a message 'Please enter the profile data.' and a sub-message 'Please enter a unique name to identify this profile.'; 'Name:' field with the value 'EPP\_SBC\_DTAG'; 'Remark:' field (empty); 'Numbering Plan:' dropdown menu with the value 'NP\_DTAG\_SBC'; 'Management Information' section with a message 'Please enter the data for the following fields in the corresponding scr...'. Below this, there are several fields: 'Class of Service:', 'Routing Area:', 'Calling Location:', 'Time Zone:' (set to 'LOCAL'), 'SIP Privacy Support:' (set to 'Full Send', highlighted with a red box), and 'Failed Calls Intercept Treatment:' (set to 'Disabled').

### Services Tab

The screenshot shows the 'Services' tab of the 'Edit Endpoint Profile' configuration page. The title bar reads '[susi] - [DTAG] - Edit Endpoint Profile : EPP\_SBC\_DTAG'. Below the title bar, there are three tabs: 'General', 'Endpoints', and 'Services'. The 'Services' tab is active. The page contains a list of services with their respective status and dropdown menus: 'Message Waiting:' (checked, 'Yes'), 'Call Transfer:' (checked, 'Yes'), 'Call Forward Invalid Destination:' (unchecked, 'No'), 'Toll and Call Restrictions:' (unchecked, 'No'), 'Park to Server:' (unchecked, 'No'), and 'CSTA Network Interface Device:' (unchecked, 'No').



## OpenScope SBC Signaling Endpoint

### General Tab

- **Profile:** Endpoint Profile configured before
- **Default Home DN:** A number from the DID range, most suitable use the first number from the DID range

The screenshot shows the 'General' tab of the configuration interface. The fields are as follows:

Name:	EP_SBC_DTAG
Remark:	
Registered:	<input checked="" type="checkbox"/>
Profile:	EPP_SBC_DTAG
Branch Office:	
Associated Endpoint:	
Default Home DN:	49228422709100
Location Domain:	
Endpoint Template:	
Endpoint Type:	Central SBC

### SIP Tab

- **Endpoint Address:** OS SBC LAN IP
- **Port:** SBC Remote Endpoint Core Port
- **Transport protocol:** TCP for unencrypted or MTLS for encrypted connection to OS SBC
- **SRTP media mode:** Must be enabled for MTLS

The screenshot shows the 'SIP' tab of the configuration interface. The fields are as follows:

Endpoint Type	
SIP Private Networking:	<input type="radio"/>
SIP Trunking:	<input checked="" type="radio"/>
SIP-Q Signaling:	<input type="radio"/>
<b>SIP Signaling</b>	
For the static Endpoints the address of the SIP signaling interface format. Note that the address of the signaling interface cannot be modified if the address has first been removed.	
Type:	Static
Signaling Address Type:	IP Address or FQDN
Endpoint Address:	192.168.57.10
Port:	50000
Transport protocol:	MTLS
Endpoint does not accept incoming TLS connections:	<input type="checkbox"/>
SRTP media mode:	Enabled



## OpenScope Branch Endpoint

### General Tab

- **Profile:** Endpoint Profile configured before
- **Default Home DN:** One number from the DID range, most suitable use the first number from the DID range

The screenshot shows the 'General' tab of the OpenScope configuration interface for an endpoint. The 'Name' field is set to 'EP\_OSB\_DTAG'. The 'Registered' checkbox is checked. The 'Profile' is set to 'EPP\_SBC\_DTAG'. The 'Default Home DN' is set to '49228422709100'. The 'Endpoint Type' is set to 'OpenScope Branch Proxy - ACD'.

Field	Value
Name	EP_OSB_DTAG
Remark	
Registered	<input checked="" type="checkbox"/>
Profile	EPP_SBC_DTAG
Branch Office	
Associated Endpoint	
Default Home DN	49228422709100
Location Domain	
Endpoint Template	
Endpoint Type	OpenScope Branch Proxy - ACD

### SIP Tab

- **Endpoint Address:** OS Branch LAN IP
- **Port:** 5060 for TCP or 5061 for TLS
- **Transport protocol:** TCP for unencrypted or TLS for encrypted connection to OS Branch
- **SRTP media mode:** Must be enabled for TLS

The screenshot shows the 'SIP' tab of the OpenScope configuration interface for an endpoint. The 'SIP Trunking' radio button is selected. The 'SIP Signaling' section is expanded, showing 'Type' set to 'Static', 'Signaling Address Type' set to 'IP Address or FQDN', 'Endpoint Address' set to '192.168.57.11', 'Port' set to '5061', and 'Transport protocol' set to 'TLS'. The 'SRTP media mode' is set to 'Enabled'.

Field	Value
SIP Private Networking	<input type="radio"/>
SIP Trunking	<input checked="" type="radio"/>
SIP-Q Signaling	<input type="radio"/>
Type	Static
Signaling Address Type	IP Address or FQDN
Endpoint Address	192.168.57.11
Port	5061
Transport protocol	TLS
Endpoint does not accept incoming TLS connections	<input type="checkbox"/>
SRTP media mode	Enabled

## SIP Security – Trusted on SIP Tab

- **Signaling Primary:** OS Branch LAN IP
- **Port Range:** 5060 for TCP or 5061 for TLS or All Ports
- **Local Realm:** tel.t-online.de

### SIP-Trunk Übersicht

<b>Name</b> SIP-Trunk Name: UNIFYMCHCSL	[edit]
<b>Kontakt</b> Administrator:	[edit]
<b>SIP-Trunk Profil</b> Zugeordnetes SIP-Trunk Profil: Standard (SIP Connect 1.1)	[edit]
<b>Gebuchte Zusatz-Pakete</b>	[edit]
<b>Verkehrsteuerung SIP-Trunk</b> Max. Anzahl: 4 Max. eingehend: 2 Max. abgehend: 2	[edit]
<b>Limitierung Parallele Gespräche</b> Maximale Anzahl: Keine Limitierung	[edit]
<b>TK-Anlagen Rufaufbauüberwachung</b> Wert: 10 Sekunden	[edit]

**Telefonie-Anmeldedaten**

SIP-Domain: tel.t-online.de

Outbound-Proxy: 55000000000000000000.primary.companyflex.de  
55000000000000000000.secondary.companyflex.de

Registrar: tel.t-online.de

Registrierungsnummer: +491992960000000000000000

Telefonie-Benutzername: +491992960000000000000000@tel.t-online.de

Telefonie-Passwort: xxxxxxxx

[Neues Passwort generieren]

**[susi] - SIP Configuration**

In this section you can configure Realm attributes, Port(4713-4717, REALM, User and Password.

---

**Security**

Signaling Primary: 192.168.57.11

Signaling Port: [ ]

Trusted entity:

All Ports  
 Port Range

Port Range: 5060-5061

Local Realm: tel.t-online.de

Local User Name: +49 [ ] [€]

Local Password: [ ]

Confirm Local Password: [ ]

## Attributes Tab

Enabled SIP Attributes:

- **Survivable Endpoint**
- **SIP Proxy**
- **Central SBC**
- **Route via Proxy**
- **Public/Offnet Traffic**
- **Do not send Diversion header**
- **Send International Numbers in Global Number Format (GNF)**
- **Send Authentication Number in P-Asserted-Identity header**
- **Use Subscriber Home DN as Authentication Number**
- **Enable Session Timer**

## Aliases Tab

<SBC LAN IP>

# OpenScope SBC Configuration

The screenshot displays the Unify OpenScope Session Border Control Management Portal. The top navigation bar shows the user name as 'administrator'. The main content area is divided into several sections:

- Administration:** A sidebar menu with options like System, Network/Net Services, VoIP, Features, Security, Diagnostics & logs, Alarms, and Maintenance.
- General - oss:** The main configuration area, starting with a note: "SBC aggregated information and data." Below this is an "Alarms" section with a summary: "Alarm summary: Critical: 0 Major: 0 Minor: 0" and a "Show alarm details" button.
- System Status:** A section with a refresh icon and a help icon. It contains:
  - Branch mode: Centralized SBC
  - Operational state: normal
  - Auto refresh timer: 10 seconds
  - Com Node 1:** Primary server (192.168.163.24) with Penalty box state Active; Backup server with Penalty box state.
  - Com Node 2:** Primary server (192.168.163.25) with Penalty box state Active; Backup server with Penalty box state.
  - Services status, SSP status, Dynamic IP remote endpoints, TURN Allocations, and SIP Loadbalancer status, each with a "Show" button.
- System Info:** A section showing hardware and software details:
  - CPU: 4.15 % - 2 x 2600 MHz
  - Memory: 14.11 % - 4 Gb
  - Disk: 13.54 % - 42 Gb
  - System uptime: 8 days 20 min
  - Hardware type: Virtual OSS 6000
  - Hostname: oss
  - Software Info: Software version V10 R2.05.00
  - Software Partition information: Active (Backup button)

## VOIP – Sip Server Settings Tab

- **Primary server:** OSV SIP TCP IP for TCP or OSV Mutual TLS IP for TLS
- **Transport:** TCP or TLS
- **Port:** 5060 for TCP or 5161 for TLS

The screenshot shows the "Sip Server Settings" configuration page. The "Sip Server Settings" tab is selected. The configuration is organized into several sections:

- General:**
  - Comm System Type: Active-Standby
  - Allow Register from SERVER:
  - Other trusted servers: [button]
- Node 1:**
  - Target type: Binding
  - Primary server: 192.168.163.24, Transport: TLS, Port: 5161
  - Backup server: [input], Transport: TCP, Port: [input]
  - SRV record: [input], Transport: TCP
- Node 2:**
  - Target type: Binding
  - Primary server: 192.168.163.25, Transport: TLS, Port: 5161
  - Backup server: [input], Transport: TCP, Port: [input]
  - SRV record: [input], 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

## VOIP – Profile Media Tab

- **Media protocol:** RTP only for TCP or SRTP only for TLS
- **SRTP crypto context negotiation:** SDES for SRTP usage
- **Codec List:** At least G.711A

**Media Profile**

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

**General**

**Name**

Media protocol   Direct Media Support

Support ICE

Support NGTC Trickle ICE

Enable NGTC WebRTC Compatibility

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

Mark SRTP Call-leg as Secure

**RTCP configuration**

RTCP Mode

RTCP generation timeout

**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

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

## Codec Support for transcoding

Enable at least G711A

**Codecs**

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

Enable	Codecs
<input checked="" type="checkbox"/>	G711A 8 kHz - 64 kbps
<input checked="" type="checkbox"/>	G711U 8 kHz - 64 kbps
<input checked="" type="checkbox"/>	G722 8 kHz - 64 kbps
<input type="checkbox"/>	G7221 16 kHz - 24Kbps
<input type="checkbox"/>	G7221 16 kHz - 32Kbps
<input type="checkbox"/>	G7221C 32 kHz - 24Kbps
<input type="checkbox"/>	G7221C 32 kHz - 32Kbps
<input checked="" type="checkbox"/>	G729 8 kHz - 8 kbps
<input type="checkbox"/>	OPUS 48 kHz - Variable
<input type="checkbox"/>	ILBC 8 kHz - Variable
<input type="checkbox"/>	ISAC 16 kHz - Variable

## SIP Service Provider Profile

- **Default SSP profile:** DTAG/CompanyFlex

SIP Service Provider Profile	
<p><i>i</i> Select OK to temporarily store changes. Make your changes permanent by selecting 'Apply Changes' on the General page.</p>	
<b>General</b>	
Name	CompanyFlex <span style="border: 1px solid red; padding: 2px;">Default SSP profile: DTAG/CompanyFlex</span>
<b>SIP Service Address</b>	
<input checked="" type="checkbox"/> Use SIP Service Address for identity headers	
<b>SIP service address</b>	tel.t-online.de
<input checked="" type="checkbox"/> Use SIP Service Address in Request-URI header	<input checked="" type="checkbox"/> Use SIP Service Address in From header
<input checked="" type="checkbox"/> Use SIP Service Address in To header	<input checked="" type="checkbox"/> Use SIP Service Address in P-Asserted-Identity header
<input checked="" 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
<b>SIP User Agent</b>	
SIP User Agent towards SSP	Passthru
SIP User Agent	
<b>Registration</b>	
<input checked="" type="checkbox"/> Registration required	
<b>Registration interval (sec)</b>	480
<b>Flags</b>	
<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 checked="" type="checkbox"/> Do not send INVITE with inactive media attribute	
<input checked="" 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	
<input checked="" type="checkbox"/> Remove Silence Suppression parameter from SDP	
<input type="checkbox"/> Enable pass-through of Optional parameters	
<input 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 checked="" 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 checked="" type="checkbox"/> Remove bandwidth (b) lines from SDP	
<input type="checkbox"/> Keep P-Asserted-Identity from access side	
<input type="checkbox"/> Avoid sending 183 messages	
<input type="checkbox"/> Avoid sending 180 message (for 60s)	
<b>TLS</b>	
TLS Signaling	Endpoint Config
<b>Sip Connect</b>	
<input type="checkbox"/> Use tel URI	
<input checked="" type="checkbox"/> Send user=phone in SIP URI	
<input type="checkbox"/> Registration mode	
<input checked="" type="checkbox"/> ITR118	

# Remote Endpoint Configuration

**Remote endpoint configuration**

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

**Remote Endpoint Settings**

**Name**  **Edit**

Type: SSP

Profile: CompanyFlex

Access realm profile: Main-Access-Realm - ipv4

Core realm profile: Main-Core-Realm - ipv4

Associated Endpoint:

Enable Call Limits

Maximum Permitted Calls:

Reserved Calls:

**Remote Location Information**

Support Peer Domains

Support Foreign Peer Domains **White list**

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
1	tel.t-online.de	0	TLS		CompanyFlex	Server authentication	DTAG

**Remote Location Identification/Routing**

Core FQDN:

**Core realm port**: 50000

Default core realm location domain name:

Default home DN: +4919929600000490810

Enable routing based on domain

FQDN:

Incoming Routing prefix:

**Digest Authentication**

Digest authentication supported

**Digest authentication realm**: tel.t-online.de

**Digest authentication user ID**: +49 @tel.t-onlin

**Digest authentication password**: ●●●●●●

**Remote Location Domain**

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

**General**

Remote URL:

Remote port:

Remote transport: TLS

**Signaling**

INVITE No Answer timeout (msec): 360000

INVITE No Reply timeout (msec): 3000

**TLS**

TLS mode: Server authentication

Certificate profile:

TLS keep-alive

Keep-alive interval (seconds): 120

Keep-Alive timeout (sec): 10

**Media Configuration**

Media profile:

Media realm subnet IP address:

**Outbound Proxy Configuration**

Outbound Proxy:

Outbound Proxy Port:

**Registrar Server Configuration**

Registrar Server:

Registrar Server Port:

Telefonie-Anmeldedaten

SIP-Domain: tel.t-online.de

Outbound-Proxy: 55XXXXXXXXX.primary.companyflex.de  
55XXXXXXXXXX.secondary.companyflex.de

Registrar: tel.t-online.de

Registrierungsrufrummer: +4919929600000XXXXXX

Telefonie-Benutzernummer: +4919929600000XXXXXX@tel.t-online.de

Telefonie-Passwort: XXXXXXXX

**Neues Passwort generieren**

**Miscellaneous**

Open external firewall pinhole

**Send RTP dummy packets**



## Security - Certificate Profile

- **Local CA file:** Specify the installed Telekom *Root-CA-Certificate T-TeleSec GlobalRoot Class 2* downloaded from URL [https://www.telesec.de/assets/downloads/PKI-Repository/T-TeleSec\\_GlobalRoot\\_Class\\_2.cer](https://www.telesec.de/assets/downloads/PKI-Repository/T-TeleSec_GlobalRoot_Class_2.cer)

- **Minimum TLS version:** Only TLS version 1.2 is supported by Deutsche Telekom CompanyFlex

**Certificate Profile**

Select OK to temporarily store changes. Make your changes perm

Certificate Profile configuration

**Certificate profile name** DTAG

**Certificate service** SIP-TLS

Local client certificate file

Local server certificate file client-cert.pem

Local CA file T-TeleSec\_GlobalRoot\_Class

Remote CA file

Local key file clientserver-key.pem

EC param secp256r1

Attach to Config file

Validation

Certificate Verification None

Revocation status

Identity Check

Renegotiation

Enforce TLS session renegotiation

TLS session renegotiation interval (minutes) 60

TLS version

Minimum TLS version TLS V1.2

This Certificate Profile is specified in the Remote Location Domain.

## OpenScape Branch Configuration

OpenScape Branch is configured to run in SBC-Proxy mode.

The screenshot displays the 'Unify OpenScape Branch Management Portal' interface. The main content area is titled 'General - osb' and includes an information icon and the text 'Branch aggregated information and data.' Below this is an 'Alarms' section with an 'Alarm summary' showing 0 Critical, 1 Major, and 0 Minor alarms, along with a 'Show alarm details' button. The 'System Status' section shows 'Branch mode' as 'SBC-Proxy' and 'Operational state' as 'normal'. It lists 'Com Node 1' and 'Com Node 2', each with 'Primary server' and 'Backup server' details and 'Penalty box state' indicators. A 'Services status' section contains several 'Show' buttons for 'Registered subscribers', 'SSP status', 'Link Status', 'Dynamic port mapping', 'Denial of Service Mitigation', and 'Subscriber data'. The 'System Info' section on the right provides hardware and software details: CPU (1.43% - 4 x 2600 MHz), Memory (26.19% - 4 Gb), Disk (15.1% - 42 Gb), System uptime (8 days 1:00), Hardware type (Virtual OSB 250), and Hostname (osb). The 'Software Info' section shows 'Software version' as 'V10 R2.04.00' and 'Software Partition information' with 'Active' and 'Backup' buttons.

It must be verified that two configuration parameters on OpenScape Branch are set correctly. This can't be verified/done via GUI interface, but via export of the configuration file. To do so select in the GUI Maintenance > Import/Export and click on the Export button in the Export section. Save and open the exported configuration file in a text editor. Search for

```
<removeRRfromWAN>0</removeRRfromWAN>
```

```
<removeRRfromLAN>0</removeRRfromLAN>
```

and ensure that both parameters are set to 0. In case of a parameter change save the configuration file and import it in the GUI via Maintenance > Browse and Import in the Import section, and click on Apply afterwards.

## VOIP – Sip Server Settings Tab

- **Primary server:** OSV SIP TCP/TLS IP

- **Transport:** TCP or TLS

- **Port** 5060 for TCP or 5061 for TLS

**Sip Server Settings** | Port and Signaling Settings | Manipulation and Routing | Error Codes | Media

**General**

Comm System Type: Collocated

**OPTIONS source port:** 5060

IP Version Towards SIP Server: IPv4

Enable path tagging

Branch behind SBC

Branch behind NAT

Synch subscriber data

Disable notification in survivable mode

Enforce minimum Subscriber TransportType Security

Disable Basic Security Check for LAN

Send CANCEL on Unreplied Branches

SDP ANAT Bypass

**Other trusted servers** | **Load Balance Mapping Table**

**Node 1**

Target type: Binding

**Primary server:** 192.168.163.22 | Transport: TCP | **Port:** 5060

**Backup server:** | Transport: TCP | **Port:**

**SRV record:** | Transport: TCP

**Node 2**

Target type: Binding

**Primary server:** 192.168.163.23 | Transport: TCP | **Port:** 5060

**Backup server:** | Transport: TCP | **Port:**

**SRV record:** | Transport: TCP

## SIP Service Provider Profile

### - Default SSP profile: DTAG/CompanyFlex

**SIP Service Provider Profile**

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

**General**

Name:  Default SSP profile:

Allow sending of insecure Referred-By header  Send authentication number in Diversion header

Send P-Preferred-Identity rather than P-Asserted-Identity  Send authentication number in P-Asserted-Identity header

Do not send Diversion header  Send authentication number in From header

Send URI in telephone-subscriber format  Include restricted numbers in From header

**SIP Service Address**

Use SIP Service Address for identity headers

SIP service address:  Privacy support:

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

**Home DN**

Mandatory default home DN  Mandatory default home DN - Normal Mode

Default home DN:

**SIP User Agent**

SIP User Agent towards SSP:  SIP User Agent:

**Registration**

Registration required

Registration interval (sec):

**Business Identity**

Business identity required

Business identity DN:

**Outgoing SIP manipulation**

Insert anonymous caller ID for blocked Caller-ID

Use single via header

**Flags**

FQDN in TO header to SSP

REFER supported by SSP

FQDN in Request-URI to SSP

Use To DN to populate the RURI

Send Default Home DN in Contact for Call messages

Request-URI user in TO header to SSP

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 local MOH in Survivable Mode

Force direction attribute to sendrcv

## Remote Endpoint Configuration

### - Signaling address type: DNS SRV

**Remote endpoint configuration**

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

**Remote Endpoint Settings**

**Name**  **Edit**

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
1	tel.t-online.de	0	TLS		CompanyFlex	Server authentication	DTAG

**Remote Location Domain**

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

**General**

Remote URL

Remote port

Remote transport

**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

Media realm subnet IP address

**Outbound Proxy Configuration**

Outbound Proxy

Outbound Proxy Port

**Registrar Server Configuration**

Registrar Server

Registrar Server Port

### Gateway Configuration

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

**General**

Signaling address type: DNS SRV

Remote URL: tel.t-online.de

Port:

Interface: WAN

Transport: TLS

Mapped port: 21000,21001,21002,21003,21004,21005

Routing prefix: %

Gateway / Trunk type: SIP Trunk

Functional type: All Modes Egress/Ingress

Trunk profile: CompanyFlex

Output digit strip: 0

Output digit add:

Priority: 1

Operational Mode in OPTIONS Response

**Signaling**

INVITE no answer timeout - Normal Mode (sec): 360

INVITE no answer timeout - Survivable Mode (sec): 180

INVITE no reply timeout - Normal Mode (sec): 3

INVITE no reply timeout - Survivable Mode (sec): 3

**Digest Authentication**

Digest authentication supported

Digest authentication realm: tel.t-online.de

Digest authentication user ID: +49...@tel.t-onlin

Digest authentication password: .....

**TLS**

TLS mode: Server authentication

Certificate profile: DTAG

### Telefonie-Anmeldedaten

SIP-Domain: tel.t-online.de

Outbound-Proxy: 551000000000.primary.companyflex.de  
551000000000.secondary.companyflex.de

Registrar: tel.t-online.de

Registrierungsrufrnummer: +491992960000000000000000

Telefonie-Benutzername: +491992960000000000000000@tel.t-online.de

Telefonie-Passwort: XXXXX

Neues Passwort generieren

### Media Configuration

Media profile: SRTP

Media realm subnet IP address:

Media realm subnet mask:

Anchoring media: Forced

Force media anchoring on transcoding

Record calls from this Gateway/Trunk

Allow Asymmetric RTP

**Outbound Proxy Configuration**

Outbound Proxy: 92697453.primary.companyflex.de

Outbound Proxy Port:

### Miscellaneous

Open external firewall pinhole

Send RTP dummy packets

## Security - Certificate Profile

- **Local CA file:** Specify the installed Telekom *Root-CA-Certificate T-TeleSec GlobalRoot Class 2* downloaded from URL [https://www.telesec.de/assets/downloads/PKI-Repository/T-TeleSec\\_GlobalRoot\\_Class\\_2.cer](https://www.telesec.de/assets/downloads/PKI-Repository/T-TeleSec_GlobalRoot_Class_2.cer)

- **Minimum TLS version:** Only TLS version 1.2 is supported by Deutsche Telekom

**Certificate Profile**

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

**Certificate Profile configuration**

**Certificate profile name** DTAG

**Certificate service** SIP-TLS

Local client certificate file  **Show**

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

Local CA file T-TeleSec\_GlobalRoot\_Class **Show**

Remote CA file  **Show**

Local key file key.pem

EC param secp256r1

**Validation**

Certificate Verification None

Revocation status

Identity Check

**Renegotiation**

Enforce TLS session renegotiation

TLS session renegotiation interval (minutes) 60

**TLS version**

Minimum TLS version TLS V1.2

## Appendix

### Call examples for unencrypted SIP Trunk

#### Incoming Call

```
▼ Session Initiation Protocol (INVITE)
  > Request-Line: INVITE sip:+49615135996868821@192.168.10.202:5060;transport=tcp SIP/2.0
  ▼ Message Header
    > Via: SIP/2.0/TCP 217.0.149.48:5060;branch=z9hG4bKmvodi-0-264-e9d-5-100000-110c0000-5e9b2064cc9aa-a84-ffffffffffffffff-16-90c40000-5e9b206466336-3281320023-4374
    Max-Forwards: 68
    > From: <sip:+497111399014800@sip-trunk.telekom.de;user=phone>;tag=2115586986-1665041842852-
    > To: <sip:+49615135996868821@tel.t-online.de;user=phone>;csf
    Call-ID: BW0937228520610221497534635@62.156.74.66
    [Generated Call-ID: BW0937228520610221497534635@62.156.74.66]
    > CSeq: 371007827 INVITE
    Min-SE: 900
    Session-Expires: 1800
    Supported: 100rel
    Supported: timer
    > Contact: <sip:mvodi-0-266-4e0-5-ffffffff1-cd790000-5e9b2064cc644-a84-ffffffffffffffff-@217.0.149.48:5060>
    Accept: application/dtmf-relay
    Accept: application/emergencycall.providerinfo+xml
    Accept: application/EmergencyCallData.ProviderInfo+xml
    Accept: application/media_control+xml
    Accept: application/pidf+xml
    Accept: application/sdp
    P-Early-Media: supported
    Allow: ACK, BYE, CANCEL, INFO, INVITE, OPTIONS, PRACK, REFER, NOTIFY, UPDATE
    Privacy: none
    Accept-Contact: *;description="<sip:+4919929600000005160@tel.t-online.de>;explicit;require
    > Feature-Caps: *;+g.3pp.trf="<unused.invalid>"
    > Session-ID: a2a2e5d0d0d02768d0f2817cbedc540; remote=00000000000000000000000000000000
    Recv-Info: x-broadworks-client-session-info
    > Record-Route: <sip:mvodi-0-266-4e0-5-ffffffff-cd790000-5e9b2064cc644-a84-ffffffffffffffff-mavsiyadi-0-26c-16-5-90c40000-5e9b206466336-a84@217.0.149.48:5060;trans
    > P-Asserted-Identity: <sip:+497111399010@sip-trunk.telekom.de;user=phone>
    Content-Type: application/sdp
    Content-Length: 304
  ▼ Message Body
    ▼ Session Description Protocol
      Session Description Protocol Version (v): 0
      > Owner/Creator, Session Id (o): ccs-0-615-5 061251380231846 1830855368 IN IP4 217.0.172.142
      Session Name (s): -
      > Connection Information (c): IN IP4 217.0.172.142
      > Time Description, active time (t): 0 0
      > Media Description, name and address (m): audio 56850 RTP/AVP 8 0 101
      > Bandwidth Information (b): AS:80
      > Bandwidth Information (b): RS:1000
      > Bandwidth Information (b): RR:3000
      Media Attribute (a): sendrecv
      > Media Attribute (a): ptime:20
      > Media Attribute (a): maxptime:20
      > Media Attribute (a): rtpmap:8 PCMA/8000
      > Media Attribute (a): rtpmap:0 PCMU/8000
      > Media Attribute (a): rtpmap:101 telephone-event/8000
      > Media Attribute (a): fmtp:101 0-15,36
      [Generated Call-ID: BW094008073061022-1136912692@62.156.74.66]
```

#### Registration OpenScope Branch -> Provider

```
▼ Session Initiation Protocol (REGISTER)
  > Request-Line: REGISTER sip:tel.t-online.de;transport=tcp SIP/2.0
  ▼ Message Header
    > Route: <sip:92697453.primary.companyflex.de;transport=tcp;lr>
    > Via: SIP/2.0/TCP 192.168.10.213;branch=z9hG4bK9ba1.86759abda64490d8ccaf14363f8cdc11.0;i=1
    > Via: SIP/2.0/TCP 192.168.10.213:41797;branch=z9hG4bK78d377e6
    Expires: 300
    Call-ID: 328fbcae
    [Generated Call-ID: 328fbcae]
    > From: <sip:+4919929600000005160@tel.t-online.de;user=phone>;tag=8597ac8d
    > CSeq: 4 REGISTER
    Max-Forwards: 70
    > To: <sip:+4919929600000005160@tel.t-online.de;user=phone>
    User-Agent: OpenScope Branch
    > Contact: <sip:+4919929600000005160@192.168.10.213:5060;transport=tcp;user=phone>
    Content-Length: 0
    > [truncated]Authorization: Digest username="+4919929600000005160@tel.t-online.de", realm="tel.t-online.de"
```



## Outgoing Call via OpenScope SBC

```

v Session Initiation Protocol (INVITE)
  > Request-Line: INVITE sip:+497111399014800@tel.t-online.de;transport=tcp;user=phone SIP/2.0
  v Message Header
    > Via: SIP/2.0/TCP 192.168.10.202:5060;branch=z9hG4bKd536.b222a321dd9afc180ab716c7fe66566d.0;i=11
      Max-Forwards: 68
    > Route: <sip:92697453.primary.companyflex.de;transport=tcp;lr>
    > Contact: <sip:+49199296000000005160@192.168.10.202:5060;transport=tcp;user=phone>
    > To: <sip:+497111399014800@tel.t-online.de;transport=tcp>
    > From: "Stefan" <sip:+49615135996868821@tel.t-online.de;transport=tcp;user=phone>;tag=sn1_09Lx1Au3sg
      Call-ID: SEC11-1e70a8c0-1f70a8c0-1-217519c0dNG
      [Generated Call-ID: SEC11-1e70a8c0-1f70a8c0-1-217519c0dNG]
    > CSeq: 1236 INVITE
      Accept-Language: en;q=0.0
      Allow: REGISTER, INVITE, ACK, BYE, CANCEL, NOTIFY, INFO, REFER
      Content-Type: application/sdp
      Date: Thu, 06 Oct 2022 07:33:48 GMT
    > [truncated]Proxy-Authorization: Digest username="+49199296000000005160@tel.t-online.de", realm="tel.t-online.de",
      Supported: histinfo, resource-priority, replaces
      Privacy: none
    > P-Asserted-Identity: <sip:+49615135996868821@tel.t-online.de;user=phone>
    > X-Siemens-Call-Type: Org-NWid-external-public
      Content-Length: 342
    > X-Siemens-OSS: OpenScope SBC V10 R2.05.00-2
  v Message Body
    v Session Description Protocol
      Session Description Protocol Version (v): 0
      > Owner/Creator, Session Id (o): OpenStage-Line_0-0_mline 1856032541 339306010 IN IP4 192.168.100.31
      Session Name (s): SIP Call
      > Connection Information (c): IN IP4 192.168.10.202
      > Time Description, active time (t): 0 0
      > Media Description, name and address (m): audio 17756 RTP/AVP 8 0 18 101 9
      > Media Attribute (a): rtpmap:8 PCMA/8000
      > Media Attribute (a): rtpmap:0 PCMU/8000
      > Media Attribute (a): rtpmap:18 G729/8000
      > Media Attribute (a): fmp:18 annex=no
      > Media Attribute (a): rtpmap:101 telephone-event/8000
      > Media Attribute (a): fmp:101 0-15
      > Media Attribute (a): rtpmap:9 G722/8000
      Media Attribute (a): sendrecv
      Media Attribute (a): rtcp-mux
      [Generated Call-ID: SEC11-1e70a8c0-1f70a8c0-1-JGAFz9BFq5v6]

```

# Call examples for TLS encrypted SIP Trunk

## Incoming Call

```

v Session Initiation Protocol (INVITE)
  > Request-Line: INVITE sip:+49228422709102@88.217.251.182:5061;transport=tls SIP/2.0
  v Message Header
    > Via: SIP/2.0/TLS 217.0.138.239:5061;branch=z9hG4bKmavodi-0-264-c98-3-1000000-8Fd70000-5df2cc7c965bb-a9f-ffffffffffffffff-8e-54f40000-5df2cc7c4ed2d-1093163325-7252
    Max-Forwards: 68
    > From: <sip:+4989700720823@ims.colt.net;user=phone>;tag=42063386-1663231473466-
    > To: <sip:+49228422709102@tel.t-online.de;user=phone>;cscf
    Call-ID: BW104433466150922266653898@62.156.74.66
    [Generated Call-ID: BW104433466150922266653898@62.156.74.66]
    > CSeq: 539564958 INVITE
    Min-SE: 900
    Session-Expires: 1800
    Supported: 100rel
    Supported: timer
    > Contact: <sip:mavodi-0-266-15-3-fffffffff1-76c60000-5df2cc7c962ba-a9f-ffffffffffffffff-@217.0.138.239:5061>
    Accept: application/dtmf-relay
    Accept: application/emergencycall.providerinfo+xml
    Accept: application/EmergencyCallData.ProviderInfo+xml
    Accept: application/media_control+xml
    Accept: application/pidf+xml
    Accept: application/sdp
    P-Early-Media: supported
    Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE
    Privacy: none
    Accept-Contact: *;description="<sip:+4919929600000490810@tel.t-online.de>";explicit;require
    > Feature-Caps: *;g.3gpp.trf="<unused.invalid>"
    > Session-ID: 75030690055b6f6ad9bebecd01f8621;remote=00000000000000000000000000000000
    Recv-Info: x-broadworks-client-session-info
    > Record-Route: <sip:mavodi-0-266-15-3-fffffffff-76c60000-5df2cc7c962ba-a9f-ffffffffffffffff-mavsiopodi-0-26c-8e-3-54f40000-5df2cc7c4ed2d-a9f@217.0.138.239:5061;transport=tls
    History-Info: <sip:+498970070@telekom.de>;index=1
    History-Info: <sip:+49228422709102@tel.t-online.de;cause=404>;index=1.1
    > P-Asserted-Identity: <sip:+4989700720823@ims.colt.net:5060;user=phone>
    Content-Type: application/sdp
    Content-Length: 457
  v Message Body
    v Session Description Protocol
      Session Description Protocol Version (v): 0
      > Owner/Creator, Session Id (o): ccs-0-615-3 06123994795318 1710879831 IN IP4 217.0.8.71
      Session Name (s): -
      > Connection Information (c): IN IP4 217.0.8.71
      > Time Description, active time (t): 0 0
      > Media Description, name and address (m): audio 21348 RTP/SAVP 8 101
      > Bandwidth Information (b): AS:80
      > Bandwidth Information (b): RS:1000
      > Bandwidth Information (b): RR:3000
      Media Attribute (a): sendrecv
      > Media Attribute (a):ptime:20
      > Media Attribute (a): 3ge2ae:applied
      > Media Attribute (a): crypto:1 AES_CM_128_HMAC_SHA1_80 inline:uwZr8+0xtuiKtu+cXv6ZjcBE9M9Cu2aYRms4zmB2
      > Media Attribute (a): crypto:2 AES_CM_128_HMAC_SHA1_32 inline:SBt8sw5py8VRVxtA8dk+imb/zlr0EBU1qVug45kM4
      > Media Attribute (a): maxptime:20
      > Media Attribute (a): rtpmap:8 PCMA/8000
      > Media Attribute (a): rtpmap:101 telephone-event/8000
      > Media Attribute (a): fmp:101 0-15

```

## Outgoing Call via OpenScape Branch

```

v Session Initiation Protocol (INVITE)
  > Request-Line: INVITE sip:+497111399014800@tel.t-online.de;transport=tcp;user=phone SIP/2.0
  v Message Header
    > Record-Route: <sip:192.168.10.213:5060;transport=tcp;ftag=sn1_suJInkV0AR;lr;otg=NM;twan>
    > Via: SIP/2.0/TCP 192.168.10.213;branch=z9hG4bKf979.1e5dcd9046033271c741d9598dcc583d.0;i=1422
    > Via: SIP/2.0/TCP 192.168.10.213:5060;branch=z9hG4bKSEC-1e70a8c0-1f70a8c0-1-Bj1cR6GjZr
    Max-Forwards: 69
    > Route: <sip:92697453.primary.companyflex.de;transport=tcp;lr>
    > Contact: <sip:+4919929600000005160@192.168.10.213:5060;user=phone>
    > To: <sip:+497111399014800@tel.t-online.de;user=phone>
    > From: "Stefan"<sip:+49615135996868821@tel.t-online.de;user=phone>;tag=sn1_suJInkV0AR
    Call-ID: SEC11-1e70a8c0-1f70a8c0-1-Id5h4DkTv15P
    [Generated Call-ID: SEC11-1e70a8c0-1f70a8c0-1-Id5h4DkTv15P]
    > CSeq: 1236 INVITE
    Session-Expires: 900;refresher=uac
    Min-SE: 900
    Accept-Language: en;q=0.0
    Allow: REGISTER, INVITE, ACK, BYE, CANCEL, NOTIFY, INFO, REFER, UPDATE
    Content-Type: application/sdp
    Date: Tue, 08 Nov 2022 06:45:23 GMT
    > [truncated]Proxy-Authorization: Digest username="+4919929600000005160@tel.t-online.de", realm="tel.t-online.de"
    Supported: timer, histinfo, resource-priority, replaces
    Privacy: none
    > P-Asserted-Identity: <sip:+49615135996868821@tel.t-online.de;user=phone>
    > X-Siemens-Call-Type: Org-Nwid-external-public
    Content-Length: 269
  v Message Body
    v Session Description Protocol
      Session Description Protocol Version (v): 0
      > Owner/Creator, Session Id (o): OpenStage-Line_0-0_mline 30198770 143224517 IN IP4 192.168.100.31
      Session Name (s): SIP Call
      > Connection Information (c): IN IP4 192.168.10.213
      > Time Description, active time (t): 0 0
      > Media Description, name and address (m): audio 58020 RTP/AVP 8 0 101
      > Media Attribute (a): rtpmap:8 PCMA/8000
      > Media Attribute (a): rtpmap:0 PCMU/8000
      > Media Attribute (a): rtpmap:101 telephone-event/8000
      > Media Attribute (a): fmp:101 0-15
      Media Attribute (a): sendrecv
      > Media Attribute (a):ptime:20
      [Generated Call-ID: SEC11-1e70a8c0-1f70a8c0-1-vzSCICpa3vVi]

```

## Outgoing Call via OpenScope SBC

```
▼ Session Initiation Protocol (INVITE)
  > Request-Line: INVITE sip:+4989700720823@tel.t-online.de;user=phone SIP/2.0
  ▼ Message Header
    > Via: SIP/2.0/TLS 88.217.251.182:5061;branch=z9hG4bK5bee.f731665d668f73dd4dd9d64fcd93e5e.0;i=1274
    Max-Forwards: 68
    > Route: <sip:92697453.primary.companyflex.de;transport=tlslr>
    Proxy-Require: mediasec
    Require: mediasec
    > Contact: <sip:+4919929600000490810@88.217.251.182:5061;user=phone>
    > To: <sip:+4989700720823@tel.t-online.de;user=phone>
    > From: "+ 49(228)422709102"<sip:+49228422709102@tel.t-online.de;transport=tlslslr;user=phone>;tag=snl_Of099uwnNt
    Call-ID: SEC11-aa3a8c0-ba3a8c0-1-TJKsL0p4K50F
    [Generated Call-ID: SEC11-aa3a8c0-ba3a8c0-1-TJKsL0p4K50F]
    > CSeq: 1236 INVITE
    Accept-Language: en;q=0.0
    Allow: REGISTER, INVITE, ACK, BYE, CANCEL, NOTIFY, INFO, REFER
    Content-Type: application/sdp
    Date: Thu, 15 Sep 2022 07:44:10 GMT
    > [truncated]Proxy-Authorization: Digest username="+4919929600000490810@tel.t-online.de", realm="tel.t-online.de", nonce="b286c2d"
    Supported: histinfo, resource-priority, replaces
    > Security-Verify: msrp-tlslslr;mediasec, sdes-srtp;mediasec, dtls-srtp;mediasec
    Privacy: none
    > P-Asserted-Identity: <sip:+49228422709102@tel.t-online.de;user=phone>
    > X-Siemens-Call-Type: ST-insecure
    Content-Length: 559
    > X-Siemens-OSS: OpenScope SBC V10 R2.05.00-2
  ▼ Message Body
    ▼ Session Description Protocol
      Session Description Protocol Version (v): 0
      > Owner/Creator, Session Id (o): OpenStage-Line_0 1049361843 66159503 IN IP4 192.168.140.167
      Session Name (s): SIP Call
      > Connection Information (c): IN IP4 88.217.251.182
      > Time Description, active time (t): 0 0
      > Media Description, name and address (m): audio 44012 RTP/SAVP 9 8 0 18 101
      > Media Attribute (a): rtpmap:9 G722/8000
      > Media Attribute (a): rtpmap:8 PCMA/8000
      > Media Attribute (a): rtpmap:0 PCMU/8000
      > Media Attribute (a): rtpmap:18 G729/8000
      > Media Attribute (a): fmtp:18 annexb=no
      > Media Attribute (a): rtpmap:101 telephone-event/8000
      > Media Attribute (a): fmtp:101 0-15
      Media Attribute (a): sendrecv
      Media Attribute (a): rtcp-mux
      > Media Attribute (a): crypto:1 AES_256_CM_HMAC_SHA1_80 inline:BCNdIU9mRH5taBYeohKIZaLltxxzk71mAJ/3gU8b2nGe0mjSuocBnDhY1Cnow==
      > Media Attribute (a): crypto:2 AES_CM_128_HMAC_SHA1_80 inline:inmw50u5R9sddrcppy+XG1H62Bnx53g8QNZfFXr
      > Media Attribute (a): ptm:20
      > Media Attribute (a): 3ge2ae:requested
      [Generated Call-ID: SEC11-aa3a8c0-ba3a8c0-1-TJKsL0p4K50F]
```

## Outgoing Call via OpenScope Branch

```
▼ Session Initiation Protocol (INVITE)
  > Request-Line: INVITE sip:+4989700720823@tel.t-online.de;user=phone SIP/2.0
  ▼ Message Header
    > Record-Route: <sip:88.217.251.182:5061;transport=tcpt;ftag=snl_u10qAmkdAR;lr;otg=NM;twan;tlslslr=yes>
    > Via: SIP/2.0/TLS 88.217.251.182:5061;branch=z9hG4bKf977.6df7fb9de7f07afe86f421a5b3fd327f.0;i=287
    > Via: SIP/2.0/TCP 88.217.251.182:5061;branch=z9hG4bKSEC-aa3a8c0-ba3a8c0-1-41efTC3vkZ
    Max-Forwards: 69
    > Route: <sip:92697453.primary.companyflex.de;transport=tlslslr>
    > Contact: <sip:+4919929600000490810@88.217.251.182:5061;user=phone>
    > To: <sip:+4989700720823@tel.t-online.de;user=phone>
    > From: "CompanyFlex 9103"<sip:+49228422709103@tel.t-online.de;user=phone>;tag=snl_u10qAmkdAR
    Call-ID: SEC11-aa3a8c0-ba3a8c0-1-Ru01tbcSF02y
    [Generated Call-ID: SEC11-aa3a8c0-ba3a8c0-1-Ru01tbcSF02y]
    > CSeq: 1236 INVITE
    Accept-Language: en;q=0.0
    Allow: REGISTER, INVITE, ACK, BYE, CANCEL, NOTIFY, INFO, REFER
    Content-Type: application/sdp
    Date: Thu, 22 Sep 2022 13:00:00 GMT
    > [truncated]Proxy-Authorization: Digest username="+4919929600000490810@tel.t-online.de", realm="tel.t-online.de", nonce="2aa85d"
    Supported: histinfo, resource-priority, replaces
    Privacy: none
    > P-Asserted-Identity: <sip:+49228422709103@tel.t-online.de;user=phone>
    > X-Siemens-Call-Type: Org-NWid-external-public, ST-insecure
    Content-Length: 454
  ▼ Message Body
    ▼ Session Description Protocol
      Session Description Protocol Version (v): 0
      > Owner/Creator, Session Id (o): OpenStage-Line_0 1820595761 1113031392 IN IP4 192.168.140.169
      Session Name (s): SIP Call
      > Connection Information (c): IN IP4 88.217.251.182
      > Time Description, active time (t): 0 0
      > Media Description, name and address (m): audio 54420 RTP/SAVP 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): sendrecv
      > Media Attribute (a): crypto:1 AES_256_CM_HMAC_SHA1_80 inline:KdK1SYG1LJYofg98GqmG8vHDC1XDVYeFovvOdnYkz8AmbuzHlRQlGmUwFzHb5Q==
      > Media Attribute (a): crypto:2 AES_CM_128_HMAC_SHA1_80 inline:jvDku3d3kKl1zjEuVRKuEeXRn7TXjnNagBD5UEv9
      > Media Attribute (a): ptm:20
      > Media Attribute (a): 3ge2ae:requested
      [Generated Call-ID: SEC11-aa3a8c0-ba3a8c0-1-Ru01tbcSF02y]
```

## Emergency Call

```

v Session Initiation Protocol (INVITE)
  > Request-Line: INVITE sip:112@tel.t-online.de SIP/2.0
  v Message Header
    > Via: SIP/2.0/TLS 88.217.251.182:65061;branch=z9hG4bK2a77.54f98cc017a196dc531e0225a7d8db46.0;i=93
      Max-Forwards: 68
    > Route: <sip:92697453.primary.companyflex.de;transport=tls;lr>
      Proxy-Require: mediasec
      Require: mediasec
    > Contact: <sip:+491992960000490810@88.217.251.182:65061;user=phone>
    > To: <sip:112@tel.t-online.de>
    > From: "CompanyFlex 9103"<sip:+49228422709103@tel.t-online.de;transport=tls;user=phone>;tag=snl_9235Yys05p
      Call-ID: SEC11-aa3a8c0-ba3a8c0-1-qW4T042giYJU
      [Generated Call-ID: SEC11-aa3a8c0-ba3a8c0-1-qW4T042giYJU]
    > CSeq: 1236 INVITE
      Session-Expires: 1800;refresher=uac
      Min-SE: 1800
      Accept-Language: en;q=0.0
      Allow: REGISTER, INVITE, ACK, BYE, CANCEL, NOTIFY, INFO, REFER, UPDATE
      Content-Type: application/sdp
      Date: Thu, 13 Oct 2022 12:27:24 GMT
    > [truncated]Proxy-Authorization: Digest username="+491992960000490810@tel.t-online.de", realm="tel.t-online.de", nonce="341d53b
      Supported: timer, histinfo, resource-priority, replaces
    > Security-Verify: msrp-tls;mediasec, sdes-srtp;mediasec, dtls-srtp;mediasec
      Privacy: none
    > P-Asserted-Identity: <sip:+49228422709103@tel.t-online.de;user=phone>
    > X-Siemens-Call-Type: Org-Nwid-external-public, ST-insecure
      Content-Length: 466
    > 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): OpenStage-Line_0 1576473781 1760896674 IN IP4 192.168.140.169
      Session Name (s): SIP Call
      > Connection Information (c): IN IP4 88.217.251.182
      > Time Description, active time (t): 0 0
      > Media Description, name and address (m): audio 42012 RTP/SAVP 8 101
      > Media Attribute (a): rtpmap:8 PCMA/8000
      > Media Attribute (a): rtpmap:101 telephone-event/8000
      > Media Attribute (a): fmp:101 0-15
      Media Attribute (a): sendrecv
      Media Attribute (a): rtcp-mux
      > Media Attribute (a): crypto:1 AES_256_CM_HMAC_SHA1_80 inline:hkm/ZIGfSgaHqEHMdGt+PsBXfMep1EPfSk+yi4SieTNvzbJSAYKDF6DpghpjhA==
      > Media Attribute (a): crypto:2 AES_CM_128_HMAC_SHA1_80 inline:h3tgwCKTnn+2qp7f2TuR0ns1RsnW7ybxGAYmYii
      > Media Attribute (a): ptme:20
      > Media Attribute (a): 3ge2ae:requested
      [Generated Call-ID: SEC11-aa3a8c0-ba3a8c0-1-qW4T042giYJU]

```

## Registration OpenScape SBC -> Provider

```

v Session Initiation Protocol (REGISTER)
  > Request-Line: REGISTER sip:tel.t-online.de SIP/2.0
  v Message Header
    > Security-Client: sdes-srtp;mediasec
      Proxy-Require: mediasec
      Require: mediasec
    > Security-Verify: msrp-tls;mediasec
    > Security-Verify: sdes-srtp;mediasec
    > Security-Verify: dtls-srtp;mediasec
    > Via: SIP/2.0/TLS 88.217.251.182:5061;branch=z9hG4bK9ae.010066bbb0a0042cdace7a354bf8e1cb.0;i=62
      Expires: 480
    > Route: <sip:92697453.primary.companyflex.de;transport=tls;lr>
      Call-ID: 5891cba3
      [Generated Call-ID: 5891cba3]
    > From: <sip:+4919929600000490810@tel.t-online.de>;tag=efe7605
    > CSeq: 11 REGISTER
      Max-Forwards: 70
    > To: <sip:+4919929600000490810@tel.t-online.de>
      User-Agent: OpenScape Branch
    > Contact: <sip:+4919929600000490810@88.217.251.182:5061;transport=tls>;expires=480
      Content-Length: 0
  v [truncated]Authorization: Digest username="+4919929600000490810@tel.t-online.de", realm="tel.
      Authentication Scheme: Digest
      Username: "+4919929600000490810@tel.t-online.de"
      Realm: "tel.t-online.de"
      Nonce Value: "7d39b6ffee6fdcd3f6091bf0627696f1"
      Authentication URI: "sip:tel.t-online.de;transport=tls"
      Digest Authentication Response: "b03547982ac8faa0a6831b41ef175a6a"
      Algorithm: MD5
      CNonce Value: "a5aabd1a5515aaf09145957c8038dfc0"
      QOP: auth
      Nonce Count: 0000001

```

## Provider -> SBC (OK)

```

v Session Initiation Protocol (200)
  > Status-Line: SIP/2.0 200 OK
  v Message Header
    > Via: SIP/2.0/TLS 88.217.251.182:5061;received=88.217.251.182;branch=z9hG4bK9ae.010066bbb0a0042cdace7a354bf8e1cb.0;i=62
    v From: <sip:+4919929600000490810@tel.t-online.de>;tag=efe7605
      > SIP from address: sip:+4919929600000490810@tel.t-online.de
        SIP from tag: efe7605
    v To: <sip:+4919929600000490810@tel.t-online.de>;tag=mavodi-0-4b-5-3-fffffc7-9be-fffffffffffffff-489d0000-62424da7-_02D9D4F80C9A
      > SIP to address: sip:+4919929600000490810@tel.t-online.de
        SIP to tag: mavodi-0-4b-5-3-fffffc7-9be-fffffffffffffff-489d0000-62424da7-_02D9D4F80C9A-1897-c2c72700-221578-632459a9-ca70b
        Call-ID: 5891cba3
        [Generated Call-ID: 5891cba3]
    > CSeq: 11 REGISTER
    v Contact: <sip:+4919929600000490810@88.217.251.182:5061;transport=tls>;expires=480;+sip.instance="<urn:uuid:aa11a3c9-470a-c6e5-97
      > Contact URI: sip:+4919929600000490810@88.217.251.182:5061;transport=tls
        Contact parameter: expires=480
        Contact parameter: +sip.instance="<urn:uuid:aa11a3c9-470a-c6e5-97a3-c4a336a8f179>"
        Contact parameter: description="<sip:+4919929600000490810@tel.t-online.de>"\r\n
    v Authentication-Info: nc=00000007,cnonce="a5aabd1a5515aaf09145957c8038dfc0",rspauth="b03547982ac8faa0a6831b41ef175a6a",qop=auth
      Nonce Count: 0000007
      CNonce Value: "a5aabd1a5515aaf09145957c8038dfc0"
      Response auth: "b03547982ac8faa0a6831b41ef175a6a"
      QOP: auth
    > Reason: SIP;cause=200;text="CC_NO_ERROR"
    > Session-ID: 00000000000000000000000000000000; remote=cd30173edc0f0b0d2f53e729655d92c2
    > Security-Server: sdes-srtp;mediasec
      Content-Length: 0

```

## Annotations

- SIP OPTIONS sent by OpenScape Voice to the SSP are sent by the OpenScape SBC to the SSP Registrar Server wrongly and therefore not answered. The OpenScape SBC endpoint in OpenScape Voice changes therefore it's state to *Inaccessible*.
- For the test were used the Deutsche Telekom DNS servers 194.25.0.68, 194.25.0.60 and 194.25.0.52.
- Special numbers like emergency numbers shall be sent as dialled and not in E.164 format