# Deutsche Telekom DeutschlandLAN

Configuration of the Unify OpenScape Enterprise Express Servers

 $\ensuremath{\mathbb{C}}$  Unify Communications and Collaboration GmbH & Co. KG

Customer Solution Lab Munich

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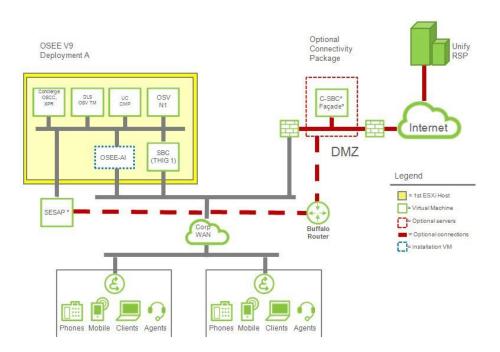
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# 1. Document Overview

### 1.1. Executive Summary

This document describes the configuration of the OpenScape Enterprise Express V9R2 servers to connect to Deutsche Telekom DeutschlandLAN via SIP trunk as they were configured during a certification test in the Deutsche Telekom certification lab. Deutsche Telekom is hereinafter referred to as Telekom.



#### 1.2. Document Control

#### 1.2.1. Authors of the Document

Name	Company - Department
Rolf Lang	Unify Communications and Collaboration GmbH & Co. KG – IDM CCS PS SP

Only the individuals listed above are authorized to make changes to the document.

#### 1.2.2. Version / Changes

Date	Version	Author	Remarks		
12 <sup>th</sup> of March 2018	0.1	Rolf Lang	Initial structure		
15 <sup>th</sup> of May 2018	1.0	Rolf Lang	Released		

# 2. OpenScape SBC Configuration

For the certification test was used SBC version 09.03.25.01-1 on the Central SBC and THIG SBC. The configuration data has be taken out from the letter from Telekom:



# 2.1. <u>DNS</u>

It must be configured a DNS server which can resolve the Telekom DNS records configured in the Remote Endpoints:

12 Network/Net Services
() Select OK to temporarily store changes. Make your changes permanent by selecting 'Apply Changes' on the General page.
Settings DNS NTP Traffic Shaping QoS
Client
Refresh DNS
DNS server IP address Add Alias Add
Division     Delete     Down.telekom.de       192.168.1.1     Delete
Server
Enable DNS server     DNS configuration     Enable customization     Administer custom files

### 2.2. Quality of Service (QoS)

Telekom has specified in their 1TR114 document QoS requirements which must be applied on the SBC:

#### 8.4.2 Traffic Classes in Layer 3

The UE uses the following traffic classes at Layer 3 (according to the Architecture of T-Home)

- Voice Control Class 6 (DSCP 110 000)
- Voice Bearer Class 5 (DSCP 101 110)

#### 1 Network/Net Services

Select OK to temporarily st	tore ch	nanges. Make your cha	anges perr
Settings DNS NTF		Traffic Shaping	Qo5
QoS Settings			
Enable QoS Configuration			
DSCP for SIP	48		
DSCP for MGCP	48		
DSCP for RTP (Audio)	46		
DSCP for RTP (Video)	46		
Row Protoc	ol	In interface	Ou

# 2.3. Media Profile

Depending on whether the SIP trunk is encrypted via TLS or not it were prepared two Media Profiles:

Sip Server Settings	Port and Signaling Settings	Media QoS Monitoring			
C					
Media Profiles					
				A	Add Edit Dek
			SRTP crypto context		
Name	Codecs	Media protocol	negotiation	Mark SRTP Call-leg as Secure	Single m-line SRTP
-	Codecs	Media protocol Best Effort SRTP		Mark SRTP Call-leg as Secure	Single m-line SRTP
 WE_Phone_default	Codecs		negotiation	Mark SRTP Call-leg as Secure	Single m-line SRTP
Name WE_Phone_default SRTP RTP	Codecs	Best Effort SRTP	negotiation mikey + sdes	Mark SRTP Call-leg as Secure	-

For an unencrypted SIP trunk the Media Protocol **RTP** was used:

🎜 🛛 Media Profile			
Select OK to temporarily store	ore changes. Make your change	s permane	nt by selecting 'Apply Cha
General			
Name	RTP		
Media protocol	RTP only	~	Direct Media Support
Suppress ICE Candidates		-	
Support Trickle ICE			
RTP/RTCP Multiplex in offer			
SRTP configuration			
SRTP crypto context negotiation		LS	
Codec configuration			
Allow unconfigured codecs     Enforce codec priority in pr     Send Telephony Event in In     Use payload type 101 for t     Enforce Packetization Inter     Codec	ivite without SDP elephony event/8000		

For an encrypted SIP trunk the Media Protocol **SRTP** and **SDES** to negotiate the cryptographic parameters was used. MIKEY may not be enabled because it's not supported by Telekom.

🎜 🛛 Media Profile		
Select OK to temporarily store	re changes. Make your changes perr	manent by selecting 'Apply Cha
General		
Name	SRTP	
Media protocol	SRTP only	Direct Media Support
Suppress ICE Candidates		
Support Trickle ICE		
RTP/RTCP Multiplex in offer		
SRTP configuration		
SRTP crypto context negotiation	□mikey ⊠sdes □dtls	
Mark SRTP Call-leg as Secure	:	
Codec configuration		
<ul> <li>Allow unconfigured codecs</li> <li>Enforce codec priority in pro</li> <li>Send Telephony Event in In</li> <li>Use payload type 101 for te</li> <li>Enforce Packetization Intern</li> </ul>	vite without SDP elephony event/8000	
Codec	Add	

The application of the Media Profile used for the SIP trunk is described in the section Remote Endpoint below.

### 2.4. Remote Endpoint

On the Central SBC must be activated Enable Remote Endpoints:

Features	
Select OK to temporarily store change	s. Make your changes permanent b
Features configuration	
Enable Remote Subscribers	Configure
Enable Remote Endpoints	Configure
Enable Codec Support for transcoding	Configure
Enable TURN Server	Configure
Enable Circuit Telephony Connector	Configure
Enable Sip Load Balancer	Configure
Enable Border Control Function	Configure (not licensed)
Enable Push Notification Service	Configure
Enable Ganglia Monitoring Daemon	
Enable Circuit Zookeeper Client	
Enable THIG	

When opening the *Remote Endpoints* window the *SIP Service Provider Profile* and the *Remote Endpoint* has to be configured:

Remo	te Endpoints	;												1	
Select	OK to temporar	ily store changes. Ma	ke your change	s permanent by se	electing 'Apply Changes	s' on the G	eneral page	e.							
SIP Service	e Provider Profile	e												?	
Hostname															
Remote dir	ectory														
User name		trator													
Password	•••••	•••													
		Download New Pro	file List												
												A	dd Edit De	elete	
▲ Row		Name	Registration required	Regis	tration interval (sec)										
1		DTAGTSystems			480									^	
														~	
<														>	
Remote er	ndpoint configura	ation												?	
													dd Edit De	elete	
						.1						-		acte	
▲ Row	Name	Access realm	profile Type	Profile / Circuit ID	Remote IP address / Logical-Endpoint-ID / Circuit UPL	Remote	Remote transport	Associated Endpoint	Core realm profile	Core FQDN	Core realm port	Routing prefix	Default home DN	4	
1	DTAGTSystems	Main-Access-Realm	-ipv4 SSP	DTAGTSystems	sip-trunk.telekom.de	0	τιs		Main-Core- Realm - ipv4		50000		+49	^	

In the *SIP Service Provider Profile* window must be selected as default SSP profile *DTAG/NGN Registration Mode*. The registration interval has to be set to 480 seconds:

SIP Service Provider Profile	?
Select OK to temporarily store changes. Make your changes permanent by selecting 'Apply Changes' on the General page.	
General	?
Name DTAGTSystems Default SSP profile DTAG/NGN Registration Moc v	
Use SIP Service Address for all identity headers	
SIP service address sip-trunk.telekom.de	
	~
SIP User Agent	?
SIP User Agent towards SSP Passthru 🗸 SIP User Agent	
Registration	?
Registration required	
Registration interval (sec) 480	
Business Identity	2
_	
Business identity required	
Business identity DN	
Outgoing SIP manipulation	?
Insert anonymous caller ID for blocked Caller-ID	
Manipulation	
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 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/PPI	
Preserve To and From headers per RFC2543	
Allow single SSP with different home DN prefix based handling	
Ignore last digit in Default home DN for incoming calls from SIP trunk	
TLS	?
TLS Signaling Endpoint Config	
	2
Sip Connect	5
✓ Send user =phone in SIP URI	
✓ Registration mode ✓ 1TR118	

n the Remote Endpoint Configura	ion window the SIP Service	Provider Profile shown abov	e has to be selected:
---------------------------------	----------------------------	-----------------------------	-----------------------

Remote endpoint configuration	1
Select OK to temporarily store changes. Make your changes permanent by selecting 'Apply Changes'	inges' on the General page.
Remote Endpoint Settings	?
Name DTAGTSystems Edit	
Type SSP V	
Profile DTAGTSystems	
Access realm profile Main-Access-Realm - ipv4 V	
Core realm profile Main-Core-Realm - ipv4	
Associated Endpoint	
Enable Call Limits	
Maximum Permitted Calls 0	
Reserved Calls D	
Remote Location Information	?
URI based routing	
Enable access control	
Signaling address type DNS SRV	
Remote Location domain list	?
	Add Edit Delete
Row Remote URL SIP/MGCP transport Media IP profile TLS mode Certificate T port	LS keep-alive (keep-Alive keep-alive) keep-alive (keep-Alive keep-alive) timeout timeo
1 sip- trunk.telekom.de 0 TLS SRTP Mutual authentication Telekom	Control         Control         (msec)         (msec) <th(msec)< th=""> <th(msec)< th=""> <th(msec)< <="" td=""></th(msec)<></th(msec)<></th(msec)<>
Remote Location Identification/Routing         Core FQDN         Core realm port         S0000         Default core realm location domain name         Routing prefix         Default home DN         ±49         Digest Authentication         Digest Authentication supported         Digest authentication nealm         sip-trunk. telekom.de         Digest authentication password         Access Side Firewall Settings	ZUGANGSDATEN         Vertraulich, bitte aufbewahren!         Datum       01. März 2018         Ortsvorwahl       0228         Durchwahlnr.       12345         Abfragestelle       0         Registrierungsrufnummer       +49 228 123450         Rufnummernblock       ?         von 000 bis 299       ?         Image: State Control of the state Contrelevel t
_	Telefonie-Passwort: Yni2Fi84 ?
Enable Firewall Settings	Outbound-Proxy: reg.sip-trunk.telekom.de
Emergency configuration Emergency numbers Add Delete	Registrar: sip-trunk.telekom.de ?
Emergency call routing	
MSRP Data Configuration	?
Enable MSRP Relay Support (not licensed)	
use IP address in MSRP-path use FQDN in MSRP-path	FQDN
Authentication required Realm	Password Show
Access side only Qop AUTH	Expire time/sec 300
Miscellaneous	?
🗹 Open external firewall pinhole	
	OK Cancel

If a NAT router is in between SBC and SIP Trunk *Open external firewall pinhole* must be enabled so the SBC will open the RTP port on the NAT router by sending UDP packets to let the NAT router pass RTP packets from a PSTN phone.

The figures below show the *Remote Location Domain* window for an unencrypted SIP trunk using TCP and RTP on the left and for an encrypted SIP trunk using TLS and SRTP on the right:

a Remote Location Domain	a Remote Location Domain		
Select OK to temporarily store changes. Make your changes permanent by	Select OK to temporarily store changes. Make your changes permanent by selecting '		
General	General		
Remote URL     sip-trunk.telekom.de       Remote SIP/MGCP port     0       Remote transport     TCP	Remote URL     sip-trunk.telekom.de       Remote SIP/MGCP port     0       Remote transport     TLS		
Signaling	Signaling		
INVITE No Answer timeout (msec)     360000       INVITE No Reply timeout (msec)     3000	INVITE No Answer timeout (msec)     360000       INVITE No Reply timeout (msec)     3000		
TLS	TLS		
TLS mode     Mutual authentication       Certificate profile     Telekom       TLS keep-alive     Image: Certificate profile       Keep-alive interval (seconds)     60       Keep-Alive timeout (sec)     10	TLS mode     Mutual authentication       Certificate profile     Telekom       ✓     TLS keep-alive       Keep-alive interval (seconds)     60       Keep-Alive timeout (sec)     10		
Media Configuration	Media Configuration		
Media profile     RTP       Media realm subnet IP address	Media profile     SRTP       Media realm subnet IP address		
Outbound Proxy Configuration	Outbound Proxy Configuration		
Outbound Proxy     reg.sip-trunk.telekom.de       Outbound Proxy Port     0       Registrar Server Configuration     1	Outbound Proxy     reg.sip-trunk.telekom.de       Outbound Proxy Port     0       Registrar Server Configuration     0		
Registrar Server Registrar Server Port 0	Registrar Server Registrar Server Port 0		

#### 2.7. Preparing and Installing TLS Certificates

For using TLS and SRTP over the SIP trunk uploading and configuration of the TLS certificates on the SBC is required:

- Download the Telekom *Deutsche Telekom Root CA 1* certificate dt-root-ca-1.cer from URL
   <u>https://www.telesec.de/de/public-key-infrastruktur/support/root-zertifikate/category/57-deutsche-telekom-root-ca-1</u>
- Download the Telekom ,*Shared Business CA*<sup>+</sup> certificate named Shared\_Business\_CA4.der from URL <u>https://www.telesec.de/de/sbca/support/ca-zertifikate/category/96-shared-business-ca-4</u>

Because the OpenScape SBC supports only certificates in pem format the Telekom , *Shared Business CA*<sup>+</sup> certificate Shared\_Business\_CA4.der has to be converted

- via Linux shell e.g. on the OpenScape SBC via command openssl x509 -inform der -in Shared\_Business\_CA4.der -out Shared\_Business\_CA4.pem
- or via e.g. online converter https://www.sslshopper.com/ssl-converter.html

Certificate Conversion Options	
Certificate File to Convert           Durchsuchen         Shared_Business_CA4.der	
Type of Current Certificate	
DER/Binary ~	
Detected type from file extension	
Type To Convert To	
Standard PEM ~	
Convert Certificate	

Click on Convert Certificate and save the converted certificate with file extension .pem.

Create in the next step a chained certificate based on the certificates *Deutsche Telekom Root CA 1* and *Shared Business CA* named e.g. dt-chain-ca.pem and copy the content of this certificate files into it in the following order:

```
-----BEGIN CERTIFICATE-----
<dt-root-ca-2.cer>
-----END CERTIFICATE-----
-----BEGIN CERTIFICATE-----
<Shared_business_CA4.cer>
-----END CERTIFICATE-----
```

Then the chained certificate should look like as below:

```
----BEGIN CERTIFICATE-----
MIIGITCCBXGgAwIBAgIIMBWLWM1WMfUwDQYJKoZIhvcNAQELBQAwcTELMAkGA1UE
...
nfKouiXc6eG1ojopwckO/uEu0JVEnyMOzGoIPU2/PhFvG6aAPsB4tvv/AHzR
-----END CERTIFICATE-----
-----BEGIN CERTIFICATE-----
MIIDnzCCAoegAwIBAgIBJjANBgkqhkiG9w0BAQUFADBxMQswCQYDVQQGEwJERTEc
...
Cm26OWMohpLzGITY+9HPBVZkVw==
-----END CERTIFICATE-----
```

Then upload this certificate via GUI at Security -> General -> Certificate Management into OpenScape SBC in the in section *CA Certificates* by selecting this certificate and clicking on *Upload*. Then the certificate appears in the CA certificates list:

Certificate Man	agement		
Select OK to temporar	ily store changes.	Make your changes permar	nent by selecting 'A
CA Certificates			
Upload CA certificate file	Durchsuchen	Keine Datei ausgewählt.	Upload
CA certificates			
dt-chain-ca.pem dt-root-ca-2.pem		^ Delete	
ossbcCA.pem			
serverCA.pem			
		×	

To replace the OpenScape SBC default certificates provided by installation execute the following steps:

The in *Certificate Creation* section enter e.g. ossbc in the *Name field* an click on Create leaving *Self signed* as *CA file* unchanged:

Certificate Creation		
Create New TLS Certificates		
Name ossbc	CA file Self signed	✓ Create

In the CA certificates, X.509 Certificates and Key files windows appears now the new certificates:

CA Certificates	X.509 Certificates	Key Files
Upload CA certificate file Browse No file selected.	Upload X.509 certificate file Durchsuchen Keine Datei ausgewählt.	Upload key file Browse No file selected.
CA certificates dt-chain-ca.pem ossbcCA.pem serverCA.pem	X.509 certificates Ossbccert.pem Servercert.pem	Key files ossbokey.pem serverkey.pem

In the Certificate Profiles section click on Add:

🖸 Certificate Mar	agement				
Select OK to temporar	ily store changes. Make		by selecting 'Apply Changes' on	the General page.	
Media DTLS certificate pro	fie	×			
Certificate Profiles					?
				Add	Edit Delete
Name	Certificate service	Client certificate file	Server certificate file	Local CA file	Remote CA f
OSV Solution	SIP-TLS		servercert.pem	serverCA.pem	^
HTTPS System Default	HTTPS		server.crt		

Create a new Certificate Profile selecting the certificates created before:

🖆 Certificate Profi	ie					
Select OK to temporarily store changes. Make your changes permanent by selecting the Converting of						
Certificate profile name Telekom						
Certificate service	SIP-TLS 🗸					
Local client certificate file	Show					
Local server certificate file	ossbccert.pem V					
Local CA file	ossbcCA.pem V					
Remote CA file	dt-chain-ca.pem 🗸 Show					
Local key file	ossbckey.pem 🗸					
EC param	secp256r1					
Attach to Config file						
Validation						
Certificate Verification Ful						
Revocation status						
Identity Check						
Renegotiation						
Enforce TLS session re	negotiation					
_	ion interval (minutes) 60					
TLS version						
Minimum TLS version TLS V	/1.2 🗸					
DTLS version						
Minimum DTLS version DTLS V1.0						
Cipher Suites						
Perfect Forward Secrecy	Preferred PFS					
Encryption F	Preferred AES-128					
Mode of Operation	Preferred GCM					

Finally the created Certificate Profile has to be configured in the *Remote Location Domain* window:

Remote Locat	ion Dom	ain	
Select OK to tempo	rarily store	chan	ges. Make your changes permanent by sele
Canada			
General			
Remote URL	sip-trunk.	teleko	om.de
Remote SIP/MGCP port	0		
Remote transport	TLS		$\checkmark$
Signaling			
INVITE No Answer timed	out (msec)	3600	000
INVITE No Reply timeou	t (msec)	3000	)
TLS			
TLS mode			Mutual authentication
Certificate profile			Telekom 🗸
TLS keep-alive			
Keep-alive inter	val (seco	nds)	60
Keep-Alive time	out (sec)		10
Media Configuration			
Media profile	S	RTP	~
Media realm subnet IP a	ddress		
Outbound Proxy Config	uration		
Outbound Proxy reg.sip-trunk.tele			kom.de
Outbound Proxy Port	D		
Registrar Server Config	uration		
Registrar Server			
Registrar Server			
Registrar Server Port (	0		

# 4. OpenScape Voice Configuration

The following configurations are done via Voice Assistant.

#### 4.1. Central SBC Endpoint

The following figure shows the general settings:

📲 " [simpltelekom] - [te	lekom] - [M	ain Office	] - Edit Endp	ooint : SBC_B	onnSP1
General SIP Att	ributes	Aliases	Routes	Accounting	
Endpoint					
Define the connection data	of an endpoint	:, e.g. you ma	ay use this to a	dd a gateway to	a switch.
Name:	SBC_BonnSi	P1		]	
Remark:			Telekom SIP Tr Gsbc from 5000		
Registered;					
Profile:	EPP_cSbc			14 17 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1	NGSDATEN
Branch Office:				Datum Ortsvorwahl	01. März 2018
Associated Endpoint:				Durchwahlm Abfragestell	
Default Home DN	49	•	-	+49 228 123	
Location Domain				Rufnummerr von 000 bis 2	
Endpoint Template:					
Endpoint Type:	Central SBC	:		]	
Max number of users:		]			

Telekom specifies in INVITEs received over the SIP trunk in the P-Asserted-Identity SIP header the SIP trunk ID which has not to be displayed on the called phone. Therefore *SIP Privacy Support* has to be set to **Full Send** in the SBC Endpoint Profile. This causes the P-Asserted-Identity header to be ignored for incoming calls but supported for outgoing calls.

	🥮 [simpltelekom] - [telekom] - Edit Endpoint Profile : EPP_cSbc							
	<ol> <li>Please enter the profile data.</li> </ol>							
	General	Endpoints	Services					
	Name:			EPP_cSbc				
	Remark:							
	Numbering Plan: NP_Bonn							
I	Management Ir	nformation						
	🕕 Please e	nter the data for th	e following field	s in the corresponding screens.				
	Class of Serv	ice:						
	Routing Area	:		RA_Bonn				
	Calling Locati	on:						
	Time Zone:			Europe/Berlin				
	SIP Privacy S	upport:		Full Send	$\sim$			
	Failed Calls In	ntercept Treatment	:	Disabled	$\sim$			
	Language:			SoftSwitch Default (German)	$\sim$			

In the SBC endpoint on tab *SIP* in section *Security* has be configured the telephony authentication credentials to enable Voice to reply to Digest Authentication challenges from Telekom:

🚳 (simpltelekom) -	[ Simplitelekom] - SIP C	Configuration - Mozilla Fir	- 0	×		×	inagem
🛈 🌶 🔒 https;	// 🛈 🔏 https://172.17	.0.4:446/management/;	··· 🛛 🕁	=	0 1	≡	
. [simpltelekon	n] [simpltelekom] - SI	P Configuration		?		?	
General SIP Management Address	DEALM Liner and Daress	configure Realm attributes, Por ord.	t(s) e.g. 4713-4717	60.	and the second		sd
	Security						Test A
Red Sky E911 Manage	Trusted entity:	Ø					
Outgoing Call Supervis Timer(ms):		All Ports					Primary
Proxy Bypass Supervie Timer (ms):	ee.	O Port Range					
Treat endpoint as sec	In Port Range:						10.82.5
eanty	Local Realm:	sip-trunk.telekom.de	Telefonie (	inricht	en		
	Local User Name:		Telefonie-Benu	1111111	000	95012	2345
<ol> <li>Set the Realm, Us</li> </ol>	28		Telefonie-Pass		Yni2Fil		
	Local Password:		Outbound-Prox			_	lekom.de
Sel:1 Items/Page:	2 Confirm Local Password:		Registrar:		sip-truni	toleka	om de
Trustee							
0	Remote Realm:						
	Remote User Name:						
	Remote Password:						
	Confirm Remote Password:						
							8
					5		
			OK	ancel	E Car	icel	

The following figures show the SBC endpoint attributes used:

📲 [simpltelekom] - [telekom] - [Main Office] - Edit E	ndpoint : SBC_BonnSP1		
General SIP Attributes Aliases Routes	Accounting		
Attributes available for this SIP endpoint		Use Subscriber Home DN as Authentication Number	
Supports SIP UPDATE Method for Display Updates		Set NPI/TON to Unknown	
JPDATE for Confirmed Dialogs Supported		Include Restricted Numbers in From Header	
Survivable Endpoint		SIPQ Truncated MIME	
IP Proxy		Enable Session Timer	
ientral SBC		Ignore Answer for Announcement	
Route via Proxy		Enable TLS RFC5626 Ping	
Now Proxy Bypass		Enable TLS Dual Path Method	
ublic/Offnet Traffic		Ignore Receipt of 181 Call is Being Forwarded	
Accept Billing Number		Use extended max, count for loop prevention	
lse Billing Number for Display Purposes		Do Not Audit Endpoint	
llow Sending of Insecure Referred-By Header		Use Proxy/SBC ANAT settings for calls to subscribers	
override IRM Codec Restriction		Support for Callback Path Reservation	
ransfer HandOff iend P-Preferred-Identity rather than P-Asserted-Identity		Send Progress to Stop Call Proceeding Supervision Timer	
end domain name in From and P-Preferred-Identity headers		Limited PRACK Support	
		Support Media Redirection	_
Send Redirect Number instead of calling number for redirected calls		Voice Mail Server	
Do not send Diversion header		Disable Long Call Audit	
Do not Send Invite without SDP		Send/Receive Impact Level	
end International Numbers in Global Number Format (GNF)		Do not send alphanumeric SIP URI	
Rerouting Direct Incoming Calls		Send alphanumeric SIP URI when available	
Rerouting Forwarded Calls		Support Peer Domains	
inhanced Subscriber Rerouting		Reserve 6	
Automatic Collect Call Blocking supported		Allow endpoint to Unregister Stale Registrations	
end Authentication Number in P-Asserted-Identity header	$\checkmark$	Enable Media Termination Point (MTP) Flow	
end Authentication Number in Diversion Header		Video call allowed	
Send Authentication Number in From Header		Trusted Subscriber	
Use SIP Endpoint Default Home DN as Authentication Number		Enable Fast Connect	

#### Configuration of Unify OpenScape Enterprise Express Servers for Deutsche Telekom DeutschlandLAN

Circuit Connector Appliance	
Add Route Header:	
Disable SRTP	
Include OSV SIP User-Agent header field	
Do Not Allow URNs in R-URI/TO Header for NG911 Calls	
Reserve 8	
Accept x-channel header	
Suppress SPE in SIPQ	
Reserve 9	

# 4.2. Avoiding Re-Invite during Session Refresh

When in a long duration call Voice usually send regularly INVITEs to refresh the session. If the SDP o-line version info is different between INVITE and the related 200 OK then Voice detects a change for the session so Voice needs to inform the peer by sending a re-INVITE. To avoid this Voice will send SIP UDPATE messages instead of re-INVITEs by setting the parameter Srx/Sip/UpdateMethodSessionTimingEnable to RtpTrue, as shown below:

Configuration Maintenance	e U	Jser M	anagement   Fault Management   Performance	e Management	Accounting	2 12	9
OpenScape Voice OpenSca	ape Bra	nch	OpenScape SBC R68700 Unified Commun	lications	CMP Device Manager	ment	
simpltelekom 🔽	0	[simplt	elekom] - RTP Management				?
A 😳 🖃 🔍	0	On this p	age you can manage RTP Parameters				
Administration	5	earch fo	in Name V Search	Show All			
General Settings							
Accounting Management						View	/Edit
Endpoint Templates	Sel	:0   Item	s/Page: 200 v   All:217   (+ + 1 v + H)				
Routing Gateways RG2700 CDR		V	Srx/Sip/enable_security_notification_3rd_party_devices	RtpFalse	RtpFalse	Yes	
SOAP/XML Client		0	Srx/Sip/WildcardedResponsibleDomains			Yes	
Operation Mode		0	Srx/Sip/UpdateMethodSessionTimingEnable	RtpTrue	Update_Session_Timing_Enable	No	
i cui		0	Srx/Sip/Trunk_Context			Yes	
Database		0	Srx/Sip/SuppressDTLS	RtpFalse	RtpFalse	Yes	
💰 0SB synchronization		0	Srx/Sip/SipQMaxTransitCount	5	5	Yes	
Report		~	Srx/Sip/Session Timer	YES	YES	Yes	
🔍 RTP		-	Srx/Sip/ResponseCodeForCauseValue 18	408	408	Yes	
Packet Filter Rules		9	or x/oip/response couer or cause value 18	400	400		
EZIP		6	Srx/Sip/ReleaseAdkTimer	5000	5000	Yes	

# 4.3. Disabling to send Diversion SIP Header

Because Telekom doesn't support the Diversion header in SIP messages, sending this header should be prevented in the SIP attributes of the SBC endpoint by enabling *Do not send Diversion header*.

Configuration Maintenance		Jser I	lanagement Fault	t Management Pe	🛈 🐔 https://172.17.0.4:446/management/portal/Appli	cations/Operation/OSV/
OpenScape Voice OpenScap	e Bra	nch	OpenScape SBC	RG8700 Unific	🖳 [simpltelekom] - [telekom] - [Main Office] - Edit En	dpoint : SBC_BonnSP1
🙈 simpltelekom 🖂	Ξ.	[simp	ltelekom] - [telekom] ·	- [Main Office] - Endpo	General SIP Attributes Aliases Routes	Accounting
OpenScape Voice OpenScape Branch OpenScape SBC RG8700 Unific 🗐 [simpltelekom] - [Main Office] - Edit Endpoint : SBC_BonnSP1						
Business Group						
	OpenScape Branch       OpenScape SBC       R68700       Unifi       #_ [simpltelekom] - [telekom] - [Main Office] - Edit Endpoint : S         Image: Simple Branch       Image: Simple Branch       Image: Simple Branch       Simple Branch       Simple Branch       Image: Simple Branch       Simple Branch       Image: Simple Branch       Simple Branch       Simple Branch       Image: Simple Branch       Simple Branch					
OpenScape Voice       OpenScape Branch       OpenScape SBC       R68700       Unify       * [simpltelekom] - [telekom] - [						
telekom	Se	l:0   Ite	ems/Page: 200 🧹   All:8			
,			Name 🔺	Numbering Plan	Rerouting Direct Incoming Calls	
		Ξ_	CPS	NP_Bonn		
► Teams		Ξ.	D_EP_Bonn	NP_Bonn	Rerouting Forwarded Calls	
<ul> <li>Statistics</li> </ul>		Ξ_	EP_MediaSrv1	NP_Bonn	Enhanced Subscriber Rerouting	
<ul> <li>Display Number Modification</li> </ul>		Ξ.	EP_Xpressions	NP_Bonn	-	
		Ξ_	Fallback	NP Common	Automatic Collect Call Blocking supported	
Hain Office		Ξ.	SBC_BonnSP1	NP_Bonn		
		isimpletekom] - [tekkom] - [Main Office] - Endpoints represent Network to Network Interface connections.       General SIP Attributes Aliases Routes Accounts         Do not send Diversion header       Do not send Diversion header         Search for       Do not Send Invite without SDP         Sel0   Items/Page: 200 / [Al:8       Rerouting Direct Incoming Cals         Name A       Numbering Plan         P. DEP_MediaSrv1       NP_Bonn         P. PAthack       NP_Bonn         Enhanced Subscriber Rerouting       Send Authentication Number in P-Asserted-Identity header         Sec_Bonn_OP       NP_Bonn         Sect String1       NP_Bonn         Send Authentication Number in Diversion Header         Use SIP Endpoint Default Home DN as Authentication Number         Use SIP Endpoint Default Home DN as Authentication Number         Use SIP Endpoint Default Home DN as Authentication Number         Use SIP Endpoint Default Home DN as Authentication Number         Set NPI/TON to Unknown				
		Ξ.	SbcThig1	NP_Bonn	Send Authentication Number in Diversion Header	
100						
					Send Authentication Number in From Header	
Translation					Lice STD Endopint Default kinne DN as Authentication Number	
Destinations and Routes					Use sir Endpoint Derault nome on as Authenocadon number	
					Use Subscriber Home DN as Authentication Number	
					Set NPI/TON to Unknown	
					Include Restricted Numbers in From Header	

#### 4.4. Sending External Numbers with leading +

In Voice Assistant can be configured any phone number for a subscriber in the *External Caller ID* field to be displayed on a called phone using feature CLIP:

	[simpltelekom] - [telekom] - [BO_Bonn] -	Edit Subscriber	: 49228422	20436101		
0	Subscriber Description					
Ge	neral Displays Routing Connectio	n Security	Keyset	Groups	Features	Applications
Extens	ion					
0	This is the default extension number which is displayed f	or internal calls to o	from this subs	criber in case th	e Display Number	Modification tables a
	Displayed Extension Number:	101				
Special	Identities					
0	The External Caller ID, if provisioned, is the subscriber's	identity which is use	d for all extern	al calls.		
	External Caller ID	4980012345678	•			

To enable Voice to send this number with a leading + via Central SBC to Telekom two preconditions must be met:

1. In the SBC endpoint must be enabled the attribute *Send International Numbers in Global Number Format* (*GNF*):

🖣 [simpltelekom] - [telekom] - [Main Office] - Edit En	dpoint : SBC_BonnSP1	?
General SIP Attributes Aliases Routes	Accounting	
Do not send Diversion header		
Do not Send Invite without SDP		
Send International Numbers in Global Number Format (GNF)		
Rerouting Direct Incoming Calls		

This cause Voice to send a number in international number format with a leading + sign.

2. To recognize an *External Caller ID* as a number in international format the country code used must be configured as a *Display Number Modification Definition*, as shown for the German country code below:

UNIFY Commo	o M	ana	gement Platf	orm Domain: system			(i) 💫 https://172.17.	.0.4:446/management/portal/A
dilles common	TIVI		gement Flati	UTTT Utilian: system			🛃 [simpltelekom]- Di	splay Number Definition
Configuration Maintenanc	e	User	Management	Fault Management	Performance	Mana	Context Settings	
OpenScape Voice OpenSc	ape B	ranch	OpenScape SB	C RG8700 Un	ified Communi	cation	Selected numbering plan	L
simpltelekom 🖂			97 - 989/	n] - Display Number Defi of country code, area code and		o strip de	Business Group:	telekom
Business Group  Quick Tasks		Search	for	in	Numbering Plan	Name	Numbering Plan:	ANY
Business Group List							Number Definition	
telekom	s	el: <b>1  </b> Ite	ems/Page: 200 🗸   All:4				digits to skip in the Local	er, enter country code, area code, loca Office Code to create an extension. To ion that defines the number of digits to
► General			Business Group	Numbering Plan	NPI	Cour		umber of digits in the fully qualified nu
Profiles		4	telekom	NP_Bonn	Public	49		Public 🗸
► Teams		V	telekom	NP_OSMO	Private		Numbering plan indicator	Public 💟
<ul> <li>Statistics</li> </ul>		R	telekom	NP_Bonn	Private		Country/L2 Code:	49
<ul> <li>Display Number Modification</li> </ul>		1	telekom	ANY	Public	49		
Definitions		-				_	Area/L1 Code:	
Prefixes								
Modifications							Local Office/L0 Code:	
Conversions							Number of digits to skip:	0
🐁 Local Toll								
Branch Office List	1						Min. Digits:	·
Main Office	4						Max. Digits:	
<ul> <li>Members</li> <li>Private Numbering Plan List</li> </ul>								
NP Common (Default, Common V	1						Local Toll	
Translation							A Local Toll table may de	fine the format of public network numb
Destination and Routes								
							Local Toll:	

# 4.5. Sending Special Numbers without leading 0

Unless not already fixed in WebCDC in the Destinations D\_xxx\_SP for traffic type *Premium Rate* and D\_xxx\_SE for traffic type *Emergency* in the phone numbering plan in each route must be deleted the leading 0:

OpenScape Voice OpenSc	ape B	ranch	0 Open	sim 😫	pitele	skom] - [tele	kom] - [NP_Bonn]	- Edit Destination: D_Bonn_SE -	Mobilla Firefox					
simpltelekom	4	[simp	pitelekom] -	0.	A https://172.17.0.4:446/management/portal/Applications/Operation/OSV/BusinessGroup/Destination     Section (Completeix) - [IP_Bonn] - Edit Destination: D_Bonn_SE									
A 🗘 🛃 Z	0	Destin	ations are used	🔩 [sin										
Business Group  Ouick Tasks		Search	for	0 Dest	natio	ns are used f	or routing a call to an	i endpoint.						
Business Group List				Genera	a l	Routes	Route Lists	Destination Codes						
telekom 👽	.5	ei0 ( III	ims,Paget 200	Routes										
<ul> <li>General</li> </ul>			Rame 4.	🕘 Muh	ple ro	utes can be	used for prioritizing th	he routes to the gateways.						
<ul> <li>Profiles</li> </ul>		-4	D_Born_J											
▶ Teams		-4	D_Born_N	-										
► Statistics		-2	D_Born_S	-							Add			
<ul> <li>Display Number Hodification</li> </ul>		-4	D_Born_SE	Sel:01	Dette	Piege: 200	01/01							
Branch Office List		4	D_Born_SP		1	D 🔺	Endpoint	Route Type	Delete	Insert	Nature of A			
Main Office	100	-		0 4		0	SBC_BonnSP1	SIP-Endpoint	1		Unknown			
Members				-										
Private Numbering Plan List														
MP_Bonn														
Translation														
▼ Destinations and Routes														
Cestinations														

# 4.6. Caller ID Suppression

To allow subscribers to use the feature *Caller ID Suppression* the subscribers must be assigned the feature *Outgoing CID suppression and Delivery per Call* by, which is activated by using the prefix \*51 by default. This assignment can be done on subscriber level or via Feature Profile.

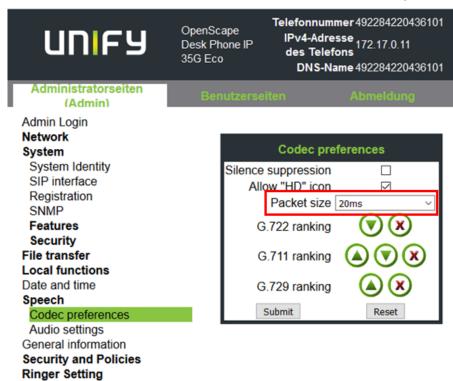
UNIFY Common	i M	lana	gement Pla	6	[simpltelekom] - [telekom] - [BO_Bonn] - Edit Subscril	per: 4922842204	36100 - Mozilla —		×
	_	-		0	🔒 https://172.17.0.4:446/management/porta	al/Applications	/Operation/OSV/ ····	♥ ☆	2 =
Configuration Maintenance	e	User	Management		[simpltelekom] - [telekom] - [BO_Bonn] - Ed	it Subscriber	: 492284220436100		2
OpenScape Voice OpenSca	ape E	Branch	OpenScape		Subscriber Description				
simpltelekom	I	[simp	ltelekom] - [telek	-				-	_
A & R &	0	Click the	e Subscriber ID to edit t		eneral Displays Routing Connection	Security	Keyset Groups	Fea	atures
Business Group	-	click ut			Name	Active	Assignment		
Ouick Tasks		Search	for		Call Completion on No Reply	0	Inherited	~	
Business Group List					Call Completion to Busy Subscriber	0	Inherited	~	
7		-late	ems/Page: 200 -   All:		Call Forwarding Dependable	0	Assigned	~	
telekom	-	sect <b>u  </b> 100	Contraction of the second second second	TH	Call Forwarding Internal/External	0	Inherited	~	
▶ General			Directory Number		Call Forwarding No Reply	0	Inherited	~	
Profiles			492284220436100		Call Forwarding on Busy		Inherited	~	
► Teams		IIII	492284220436101		Call Forwarding to Voice Mail		Inherited	~	
<ul> <li>Statistics</li> </ul>			492284220436102		Call Forwarding Unconditional	0	Inherited with Local Data		
<ul> <li>Display Number Modification</li> </ul>			492284220436103		and an and a state of the second s	0	[	~	
Branch Office List		I	492284220436104		Call Pickup Directed	-	Inherited	~	
BO_Bonn			492284220436105		Call Transfer	0	Inherited	~	
▼ Members			492284220436106	-	CSTA Access	٥	Inherited	~	
Subscribers  Endpoints		100	492284220436107	4	Do Not Disturb	0	Inherited	~	
Media Servers		Im	492284220436108		Enhanced Forwarded-Call Info	0	Inherited	~	
Private Numbering Plan List		IIII	492284220436109		Large Conference	0	Inherited	~	
NP_Common (Default, Common		I	492284220436113		Last Incoming Number Redial	0	Inherited	~	
Translation			492284220436300		Last Outgoing Number Redial	0	Inherited	~	
Destinations and Routes	П	100	492284220436301		Malicious Call Trace	0	Switch-wide	~	
			492284220436302		Music On Hold	0	Inherited	~	
		100	492284220436302		Name Permanent Presentation Status	0	Inherited	~	
			492284220436303		Number Permanent Presentation Status	0	Inherited	~	
		1000			One Number Service	0	Inherited	~	
			492284220436305		Outgoing CID Suppression	0	Inherited		
			492284220436306		Outgoing CID Suppression and Delivery Per Call	٢	Inherited	~	
			492284220436307		Park to Server	0	Inherited	~	
			492284220436308				a descent of the second s		

# 5. SIP Phones

# 5.1. Packet Size

Mobility Diagnostics Maintenance

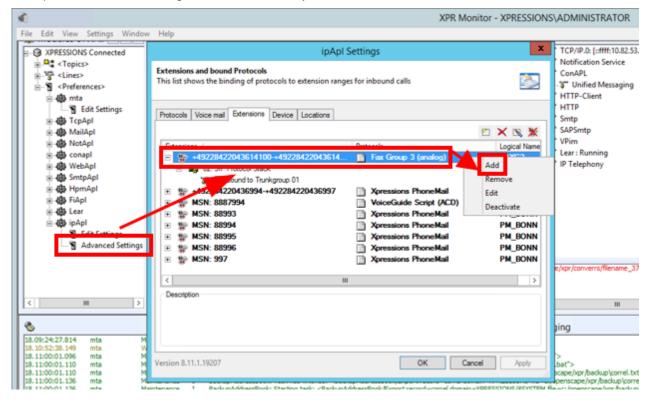
It has to make sure that the Packet Size has to be set to 20 ms on the SIP phones:



# 6. OpenScape Xpressions

### 6.1. Adding Extension Range

The Xpressions Extension Range to be added manually:



Configuration of Unify OpenScape Enterprise Express Servers for Deutsche Telekom DeutschlandLAN

#### **About Unify**

Unify is the Atos brand for communication and collaboration solutions. At the core of the Atos Digital Workplace portfolio, Unify technology enables organizations of all sizes to transform the way they collaborate, creating a more connected and productive workforce which can dramatically improve team performance, individual engagement and business efficiency.

Unify products represent a strong heritage of technology innovation, reliability and flexibility. Their award-winning intuitive user experience can be delivered through almost any device and in any combination of cloud or on-premise deployment. Augmented by Atos' secure digital platforms, vertical solutions and transformation services, they set the global standard for a rich and reliable collaboration experience that empowers teams to deliver extraordinary results.

#### unify.com



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