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Mitel OpenScape Business V3 R4.x

PBX Configuration Guide for Swisscom Smart Business Connect Internet Trunk

From	UMB		**To**	IP PBX and Communication System Integrators
Date	09.04.2025		**For information**	
Subject	SIP Trunk PBX Configuration Guide for Smart BCon Internet			

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Change history

Version	Review date	Reviewed by	Comments
1.0	24.02.2025	Daniele Varone	New Document for Version V3 R4.x via Internet Trunk
1.1	09.04.2025	Daniele Varone	Small changes

Review

Version	Review date	Reviewed by	Comments
1.0	21.03.2025	Product Manager	Review
1.1	09.04.2025	Product Manager	Review

Contents

1 Introduction

1.1 Objective and purpose

Describes the SIP Trunk configuration of IP PBX or communication systems. The IP PBX or communications systems are homologated using this SIP Trunk configuration to interoperate with Swisscom Smart BCon Internet service which allows direct connection to Swisscom SIP Core (no Enterprise Session Border Controller required).
The Mitel OpenScape Business may reside on customer's premises or in the Google Cloud Platform.
In the case of SBCon SIP-SIP or SIP-Virtual setup (with eSBC on premises or in Swisscom TelcoCloud), please use the relevant document.

1.2 Target audience

IP PBX and Communication System Integrators, who have joined the Swisscom Partner Training for Smart Business Connect Trunk.

1.3 Terms, abbreviations

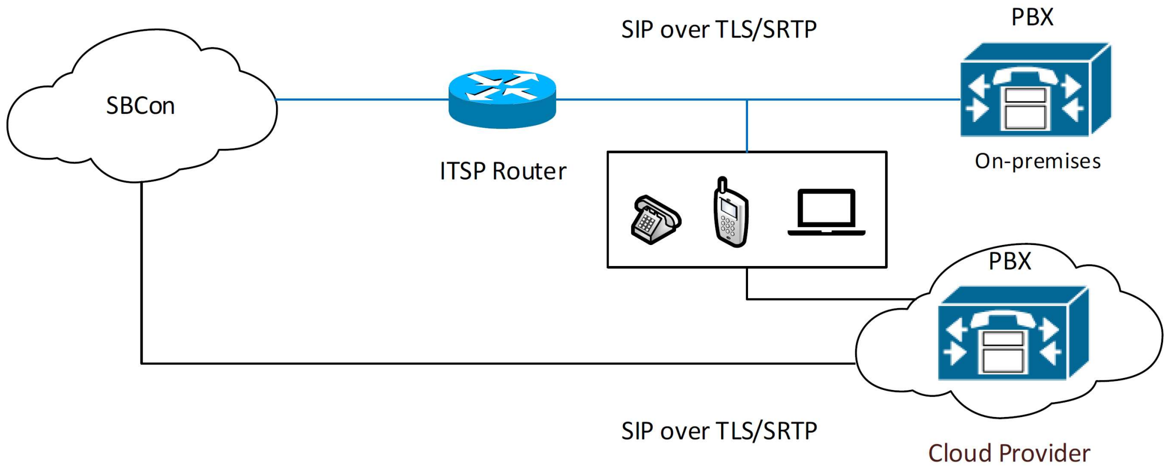
Terms	Abbreviations
SIP	Session Initiation Protocol
IP	Internet Protocol
PBX	Private Branch Exchange
eSBC	Enterprise Session Border Controller
SIP-SIP	eSBC on premises
SIP-Virtual	veSBC in Swisscom Telco Cloud

1.4 Referenced documents

[1] https://wiki.unify.com/images/e/e7/OpenScape_Business_S_Image_for_Google_Cloud.pdf

2 Overview PBX

2.1 SIP Trunk network architecture customer side



2.2 Hardware requirements

If OpenScape Business S is deployed in Google Cloud Platform, there are no specific hardware requirements from the end user's perspective.

On-premise OpenScape Business S and OpenScape Business X V3 Mainboard Family

Note: The connections are only enabled via the LAN interface!

2.3 Software requirements

OpenScape Business, software version V3 R4.x

2.4 Support Contacts

Your local PBX distributor

3 SIP Trunk features

3.1 Features supported and tested

- National calls
- International calls
- National calls with international prefix
- Toll free numbers (0800)
- DTMF (RFC 2833)
- Call cancellation
- Call rejection
- Calls with early media
- Calls to special/short numbers
- Calling line indication presentation (CLIP)
- Calling line indication restriction (CLIR)
- Special Arrangement
- Call hold/resume
- Music on hold
- Call forwarding unconditional *(with 2nd SIP INVITE and SIP 302 methods)*
- Call forwarding busy *(with 2nd SIP INVITE and SIP 302 methods)*
- Call forwarding no answer *(with 2nd SIP INVITE and SIP 302 methods)*
- Attended call transfer
- Blind call transfer
- 3-party conference
- Fax with G711 pass-through (tested with Audiocodes MP-112)

3.2 Caveats and known restrictions

- Due to the mandatory encryption settings on SIP trunk, FAX transmission needs to be done through G.711 exclusively.
- In case of outgoing calls from myPortal@Work to a destination with "early media" (voice announcement before connect) the received audio is audible, however no DTMF digits can be sent, e.g. for navigating in an IVR menu.
- E112 Emergency Calling with PANI header is currently not implemented.
- Billing with Special Arrangement: the billing will be done on the trunk main number instead of the user number.

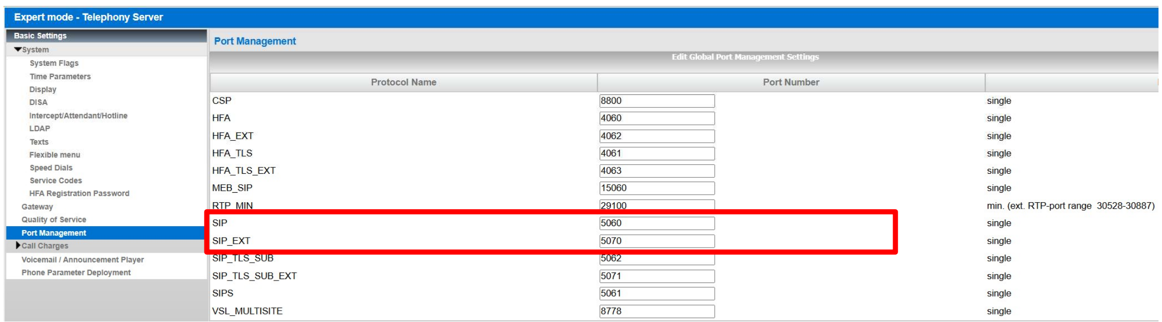
4 SIP Trunk configuration PBX side

4.1 Default SIP-Port

The default SIP Port for ITSPs (SIP_EXT) is configured to "5070".

Under Expert mode> Telephony Server > Basic Settings > Port Management

For Security Reason SIP and SIP_EXT must be set to different Values (see Printscreen).

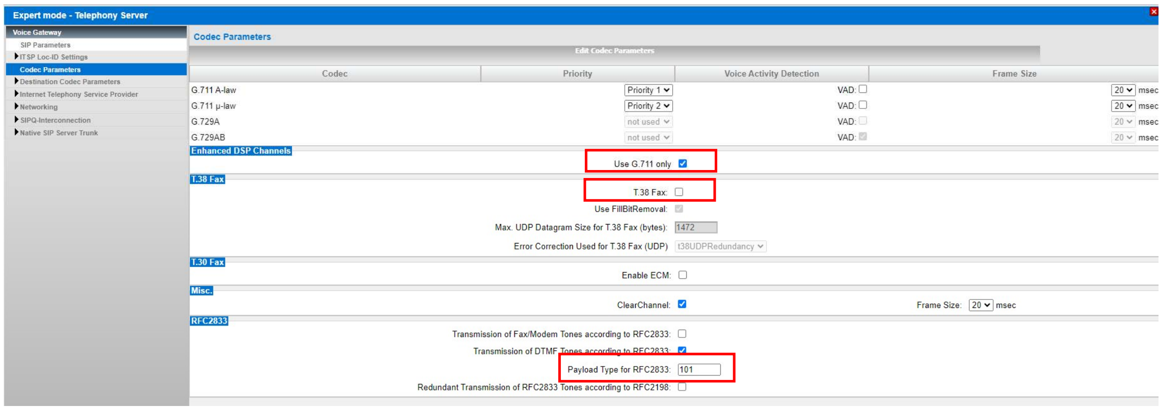


If the Customer is using SIP Clients, they must register with SIP Port 5060!

4.2 Set Codec Parameters and Payload Type

Open "Expert mode > Voice Gateway > Codec Parameters"

Change "Payload Type for RFC2833" from 98 to 101
Deactivate "T.38 FAX"



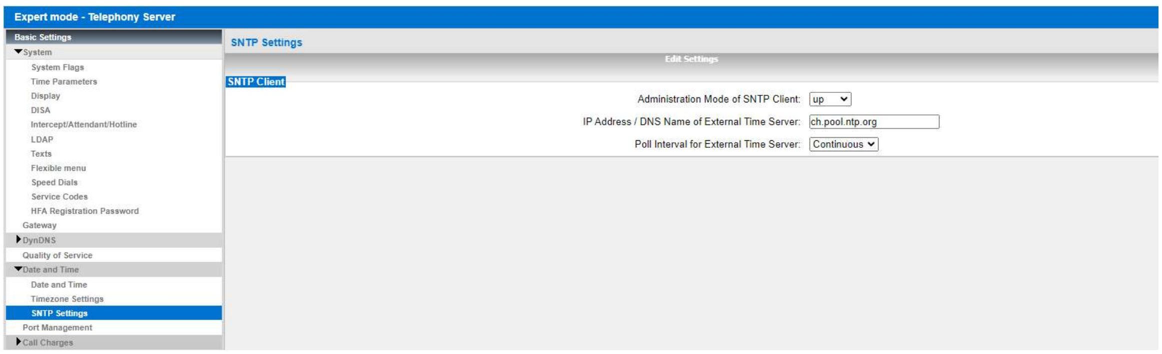
After that the System need a Restart!

4.3 SNTP configuration

Due to the AllIP Migration, there is no more longer time delivered to the PBX, from public Network. Therefore, it is needed to configure a SNTP Server.

Open "Expert mode > Basic Settings > Date and Time > SNTP Settings" :

Administration Mode of SNTP Client = up
IP Address / DNS Name of External Time Server = IP Address / DNS name of costumer Time Server (alternative = ch.pool.ntp.org)
Poll Interval for External Time Server = Continuous



4.4 Gateway Location (just to check)

Open the Expert mode > Telephony Server > Basic Settings > System > Gateway

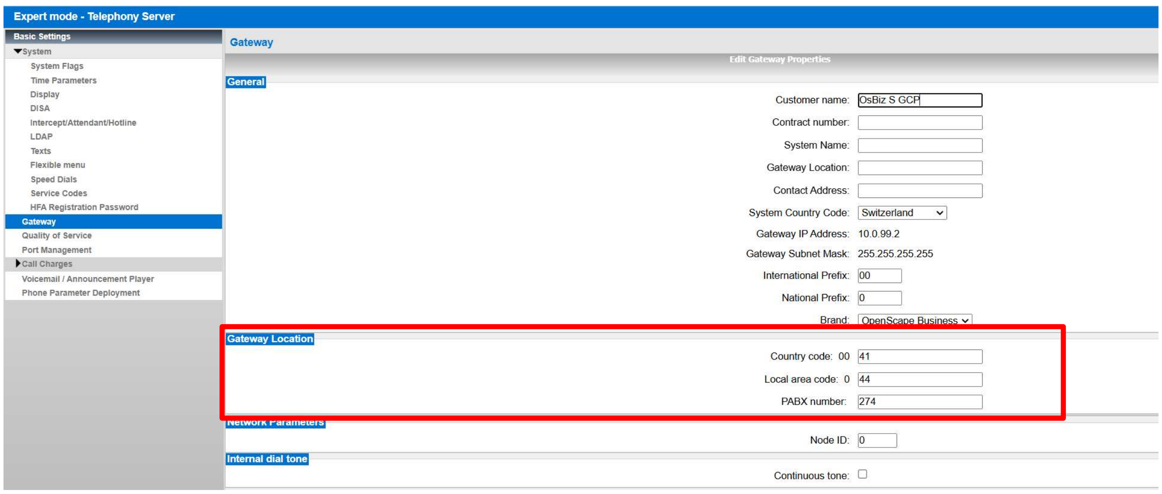
Open Gateway-Location, and enter the below Parameters

Country Code = 41

Loca area code = 44

PABX Number = 2747 (Systemnumber without DDI Range)

Without this Setting, it could be, that some Number ranges are not dialable!



4.5 Seizure code setting

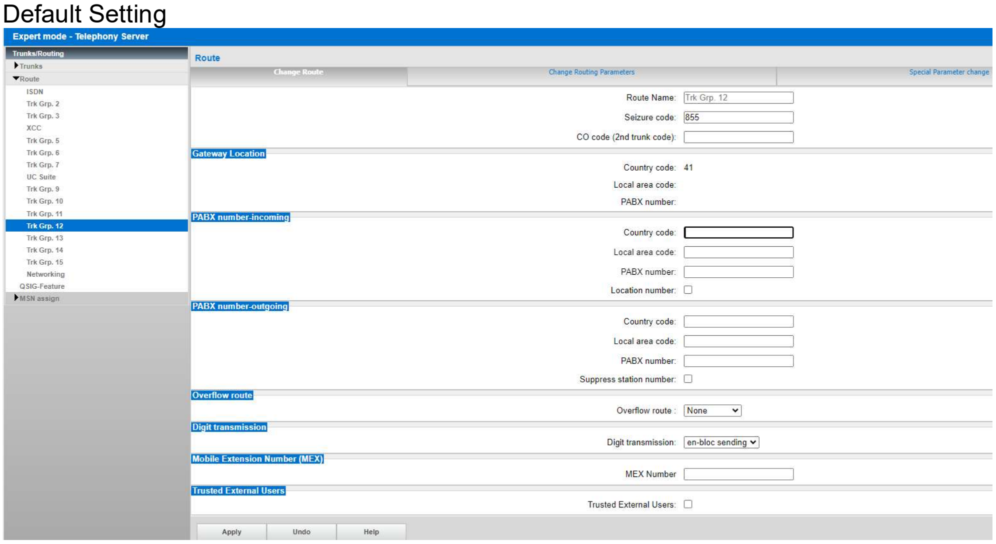
Open the Expert mode > Telephony Server > Trunks/Routing > Route Trk. (Default 1.ITSP = Trk.12)

Per default in Trk.Grp. 1, Seizure Code „0“ is configured, change this with another Seizure Code eg. “855”, after that, you can add the Seizure Code “0” in Trk Grp. 12.

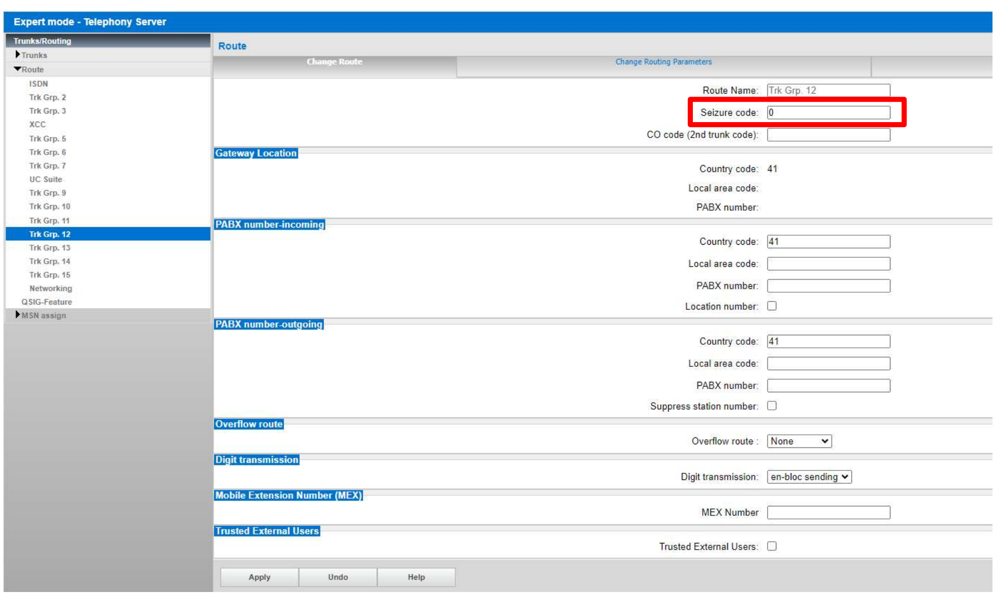
This is important for the correct presenting of the number in the display like:

- Number incoming / outgoing
- Caller list
- Redial

Default Setting



Changed Setting



4.6 Station creating

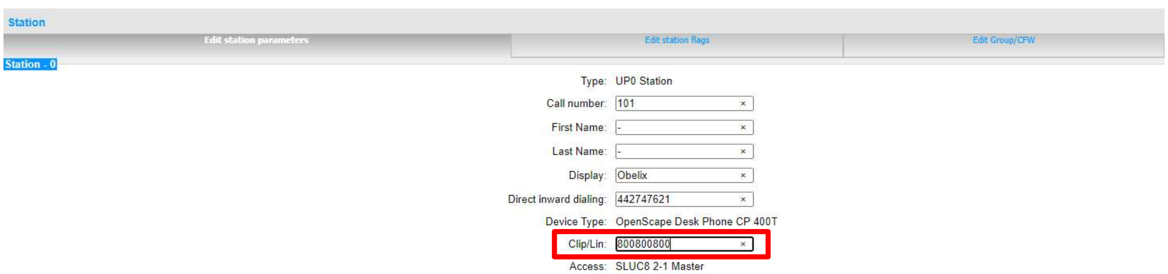
Open the Expert mode > Telephony Server > Station

Call number	=	internal Number of the station
DID	=	Public Number over which the user can be reached from Public Network

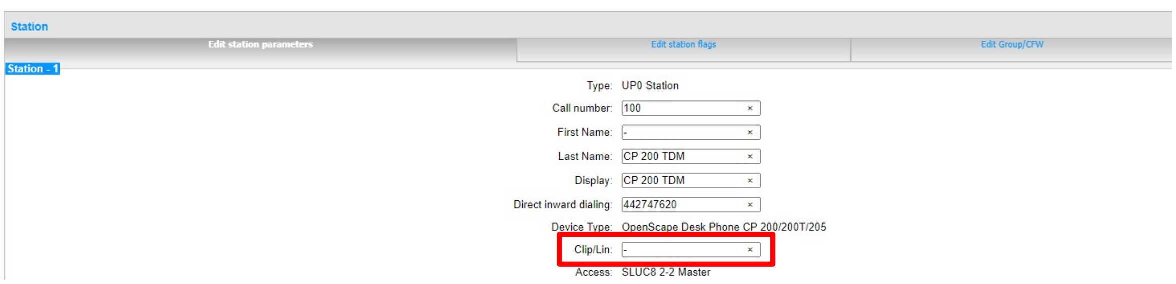
It's recommended to add DID with 9 digits.

Clip/Lin:
Here you can set a 9 digit DID for each station. If no Number is entered, the configured incoming DID will be sent in outgoing direction.

Example for a User with Clip no Screening => Clip Setting



Example for a User without Clip no Screening => Clip Setting

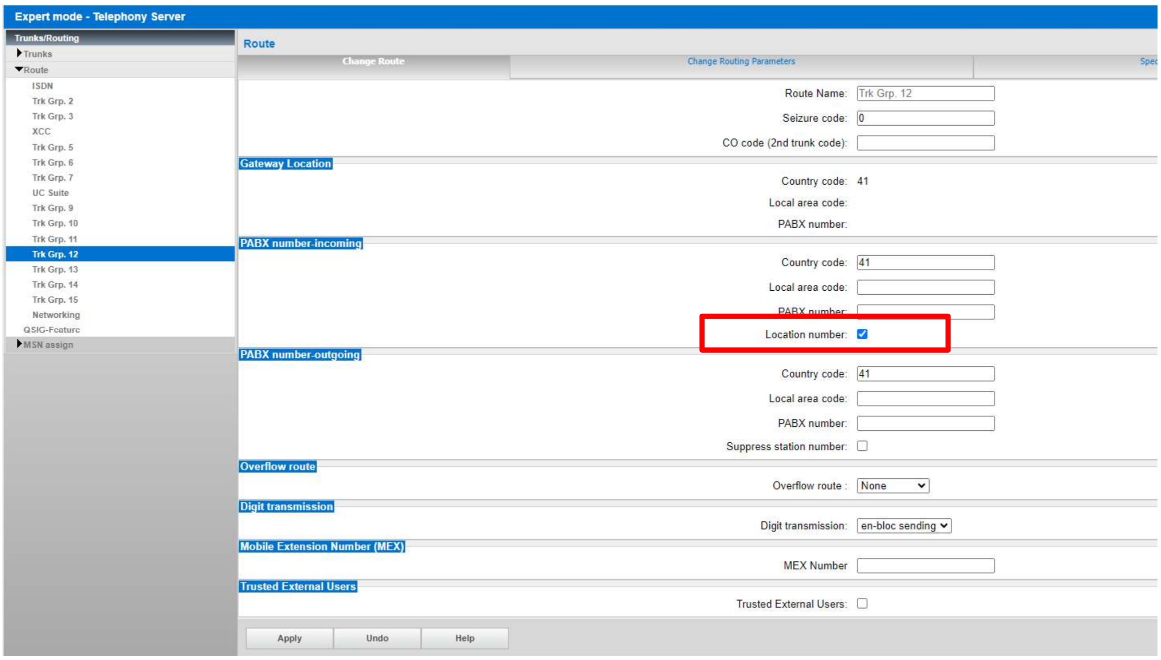


4.7 PABX Number

Open the Expert mode –Telephony Server – Trunks/Routing – Route

We recommend to keeping "Local area code" and "PABX number" empty. (Incoming and outgoing)

Activate only Location number



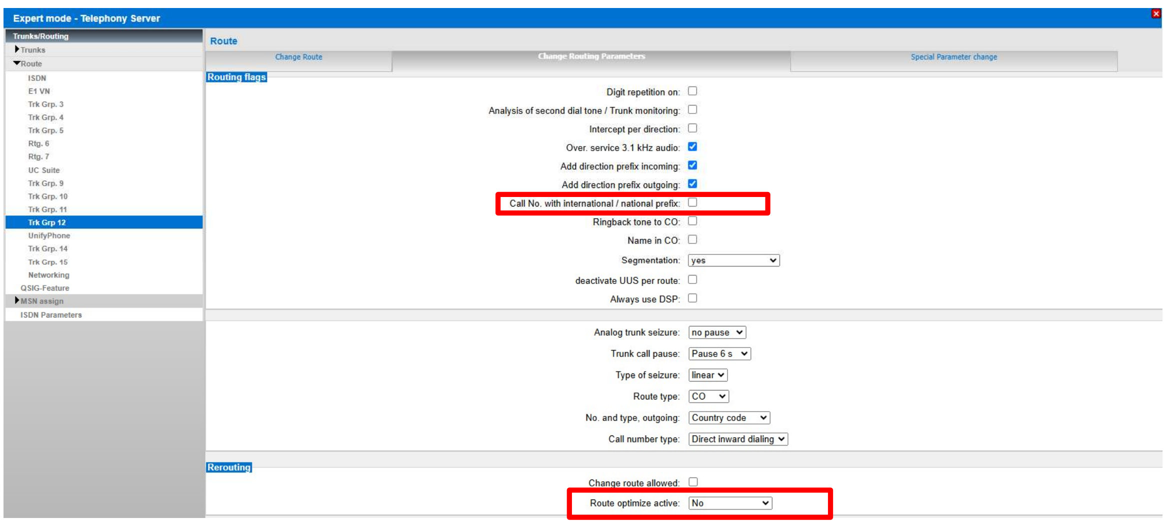
4.8 Routings Parameters (just to check after Wizard is completed)

Open Expert mode –Telephony Server – Trunks/Routing – Change Routing Parameters

Routing flags:

Over. service 3.1KHz audio	= activated
Add direction prefix incoming	= activated
Add direction prefix outgoing	= activated
Call No. with international /national prefix	= deactivated
Segmentation	= yes
No. and type, outgoing	= Country code

Rerouting > "Route optimize active" allows you to activate "Call deflection" for Call forwarding / Rerouting



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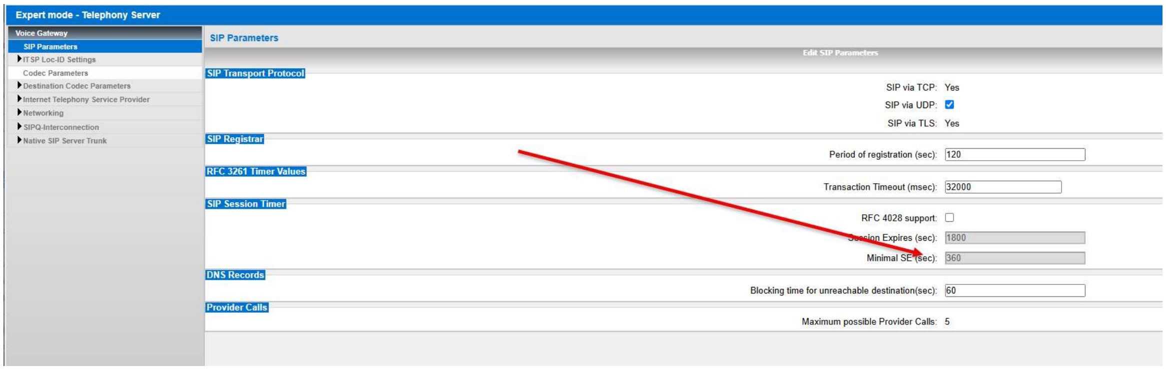
4.9 SIP Parameters

Change the values under SIP Sessions Timer

Enable RFC 4028 support and adjust the value for minimal SE to “360” and disable again RFC4028 afterwards.



4.10 Installation of the certificate for TLS

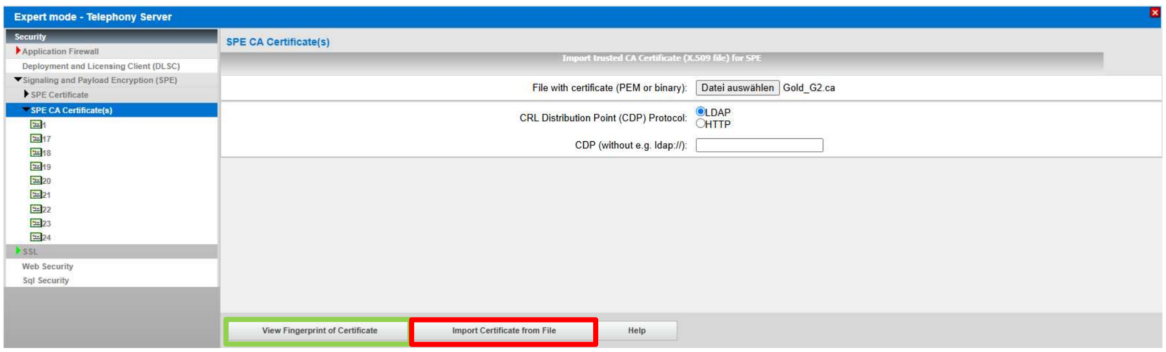
To successfully establish a TLS connection to Swisscom Smart Business Connect Internet trunk, the SwissSign Gold CA - G2 is required as a trusted certificate.

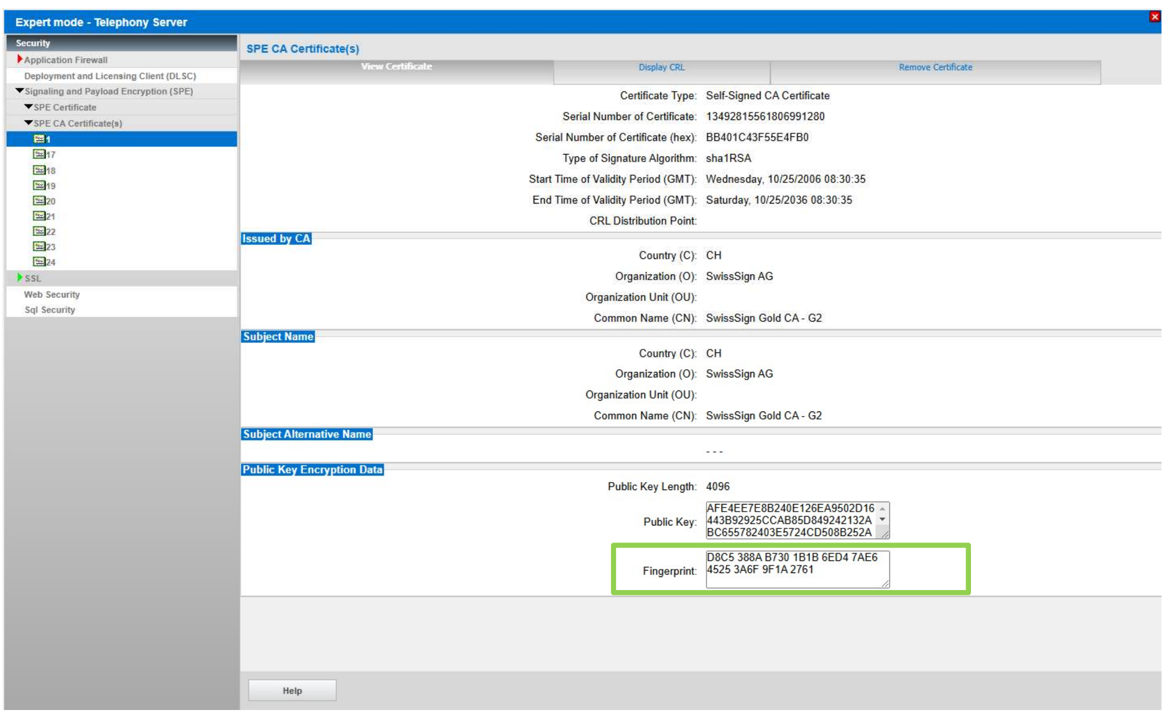
Download: https://www.swisssign.com/support/ca-prod
The Gold G2 certificate in the PEM version is required.

Go to Expert mode > Telephony Server > Signaling and Payload Encryption (SPE) > SPE CA Certificate

Import the downloaded certificate by entering the path, using “View Fingerprint of Certificate” if you can see the fingerprint, then click on Import and the certificate will be imported into the system! The ITSP trunk is only encrypted with a valid certificate

Once the file has been selected, check the fingerprint. This can be checked after the import



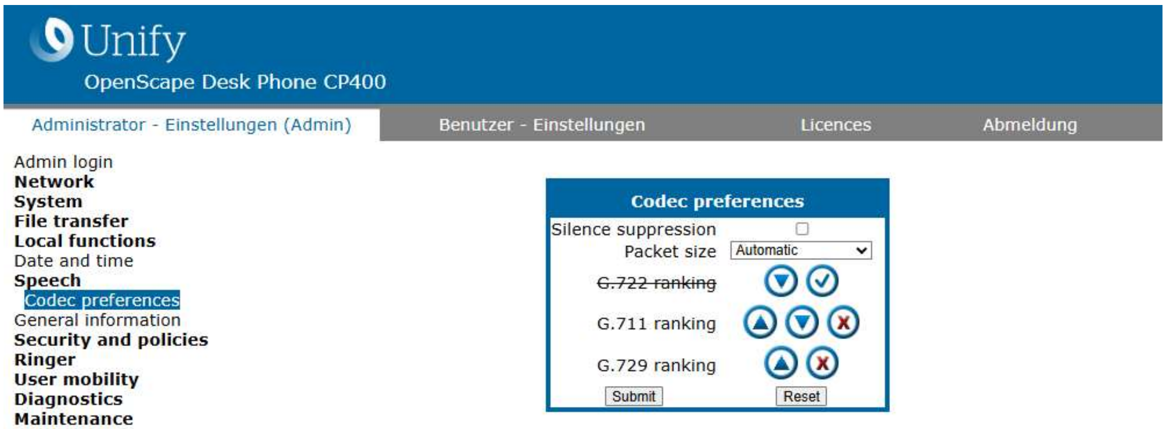


4.11 Settings on the IP end device Codec setting

Go to the admin settings of the IP end device (https://ipadressdevice)

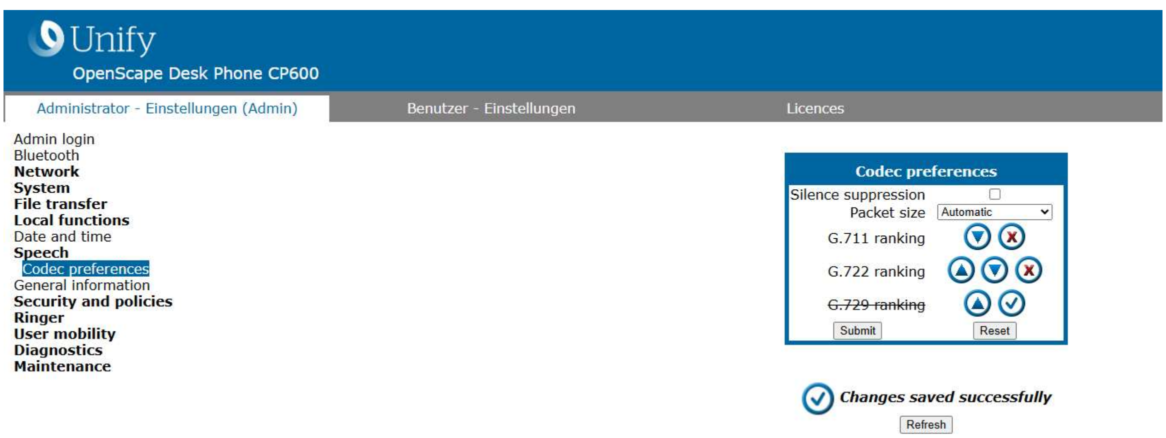
Speech > Codec preferences

Default settings of the device



Activate the customisations on all IP end devices

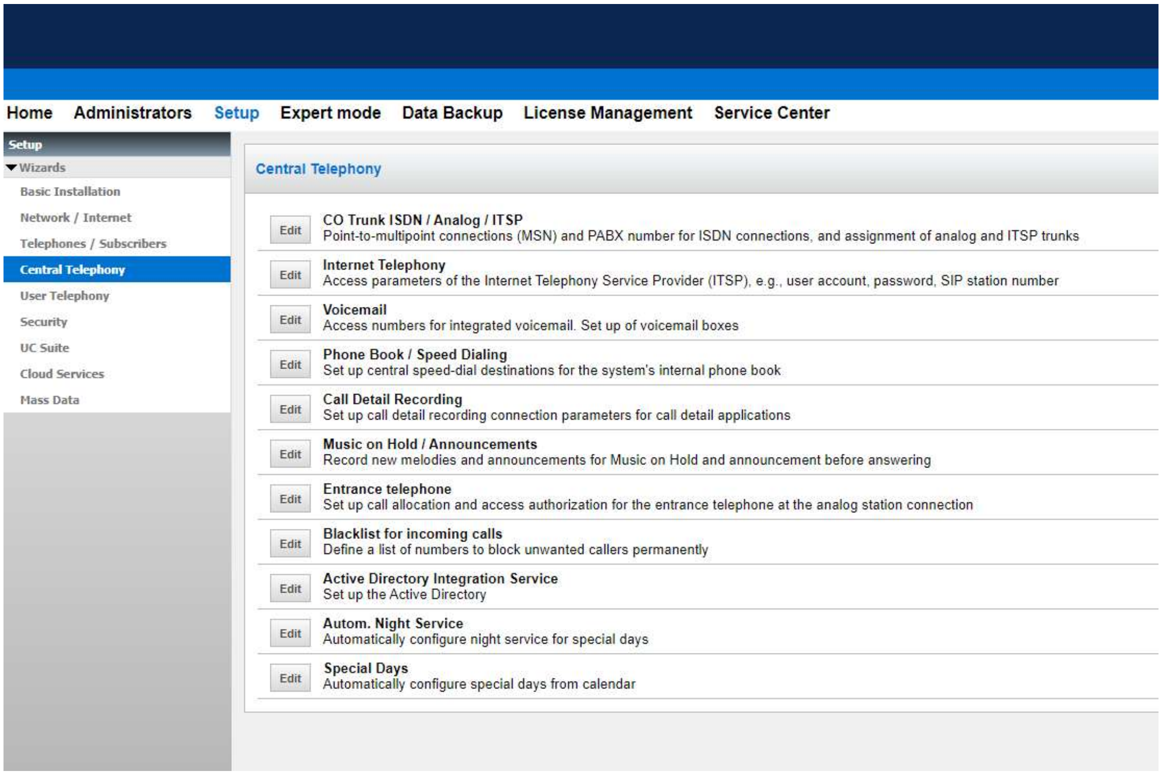
- Deactivate the codec G.729
- Activate codec G.722
- Move Codec G.722 to the priority 2



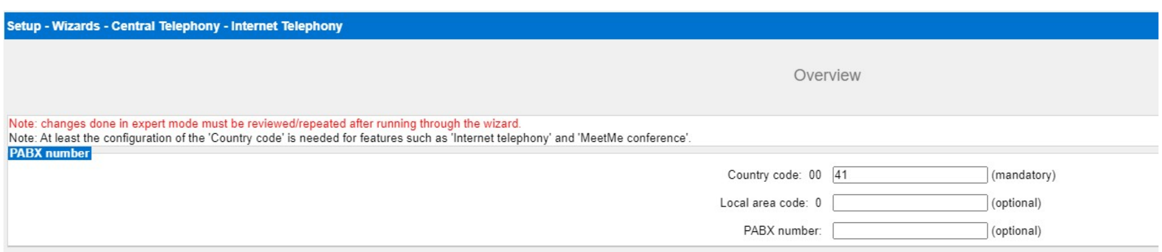
Important: The OpenScape Desk Phone CP 100 terminal does not support codec G.722!

5 Establishment of the ITSP Smart Business Communication Trunk

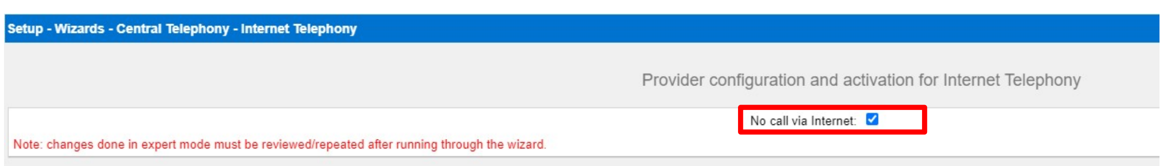
Setup (Wizards) > Central Telephony > Internet Telephony



Internet Telephony «Edit»

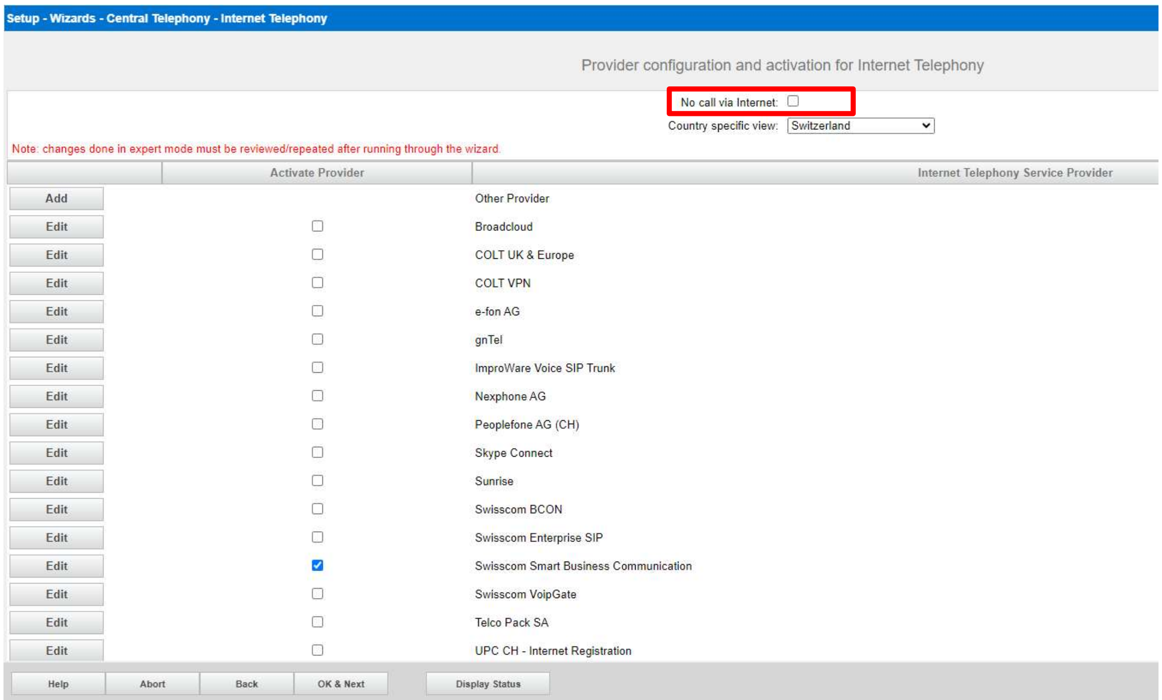


Deactivate the Flag «No call via Internet»



Until a separate template for the Smart Business Connect Internet trunk is available in the PBX, you can modify the existing template "Swisscom Smart Business Communication" like described below.

Activate the Provider «Swisscom Smart Business Communication»



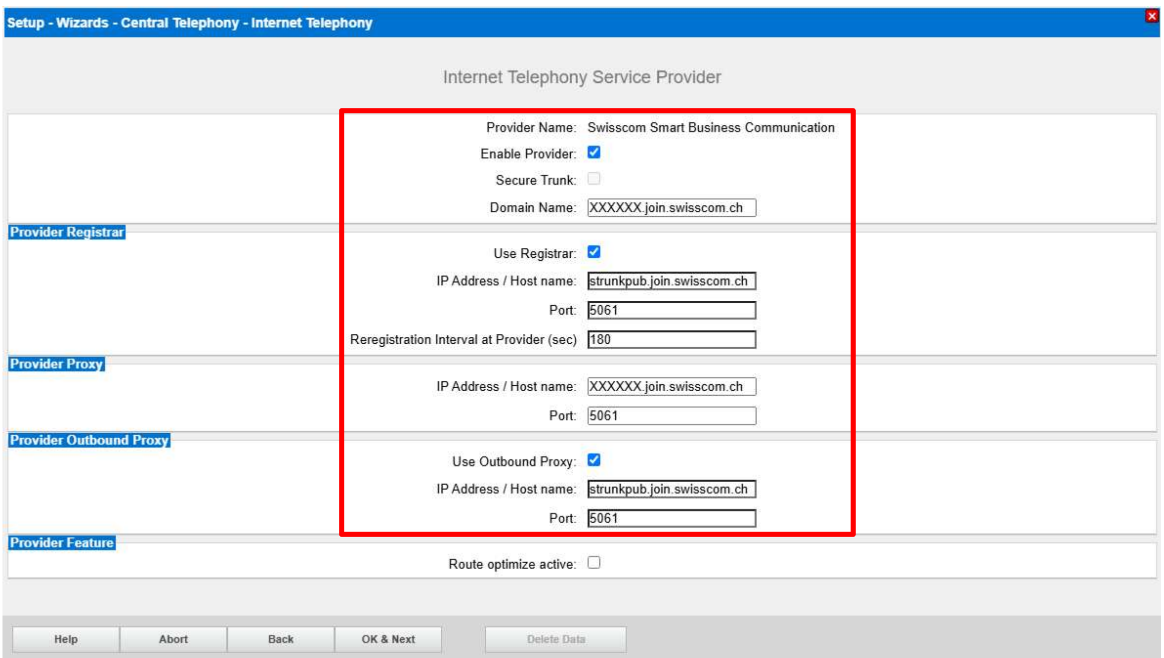
Configure the profile «Swisscom Smart Business Communication»

5.1 Internet Telephony Service Provider

Enter the Data from Swisscom Smart BCon Portal.

Enable Provider	= YES
Domain Name	**= XXXXXX.join.swisscom.ch**
Provider Registrar IP Address / Host name	**= strunkpub.join.swisscom.ch**
Port	= SIP Port (5061)
Reregistration-Interval at Provider (sec)	= 180
Provider Proxy IP Address / Host name	**= XXXXXX.join.swisscom.ch**
Port	= SIP Port (5061)
Provider Outbound Proxy	**= strunkpub.join.swisscom.ch**
Port	= SIP Port (5061)

Secure Trunk (Media security) must be activated in expert mode!



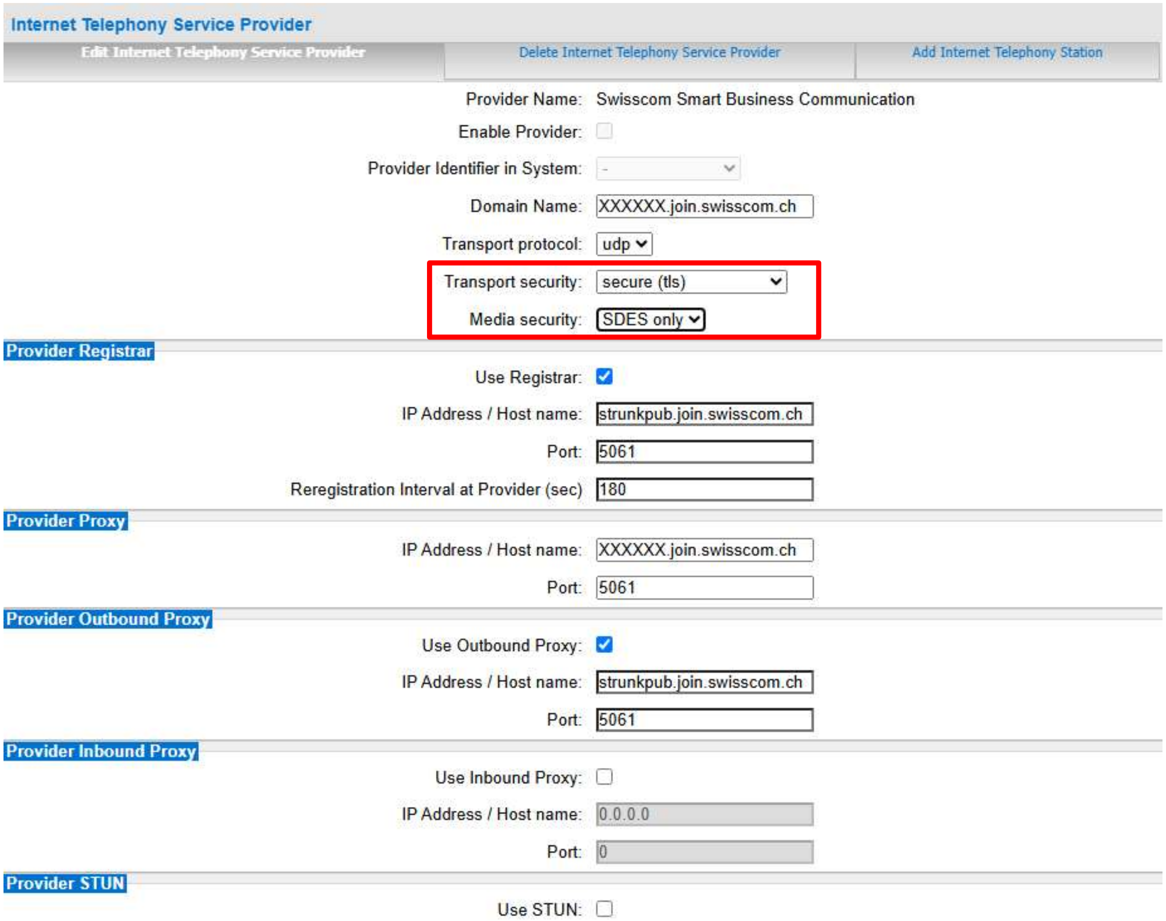
Call forwarding/redirection by means of "Call deflection" can be activated here!
(If call forwarding is activated, incoming calls are triggered with a SIP 302 response and the call forwarding is carried out by the provider).

Call forwarding via rerouting

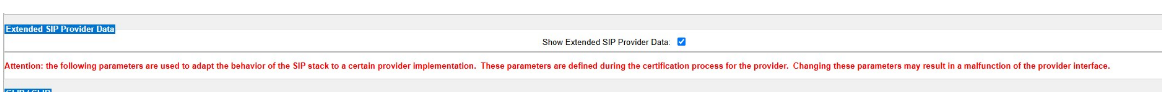
"Rerouting active" deactivated (default) -> in case of call forwarding a second connection is established and the control of the call remains in the system.

"Rerouting active" activated -> in case of call forwarding, rerouting is performed in the Smart BCon Network. The system loses further control of the call.

Secure Trunk (Media security) activation after wizard execution in expert mode!



Now edit the other data under Show Extended SIP Provider Data



Change the red marked settings of Template Swisscom Smart Business Communication

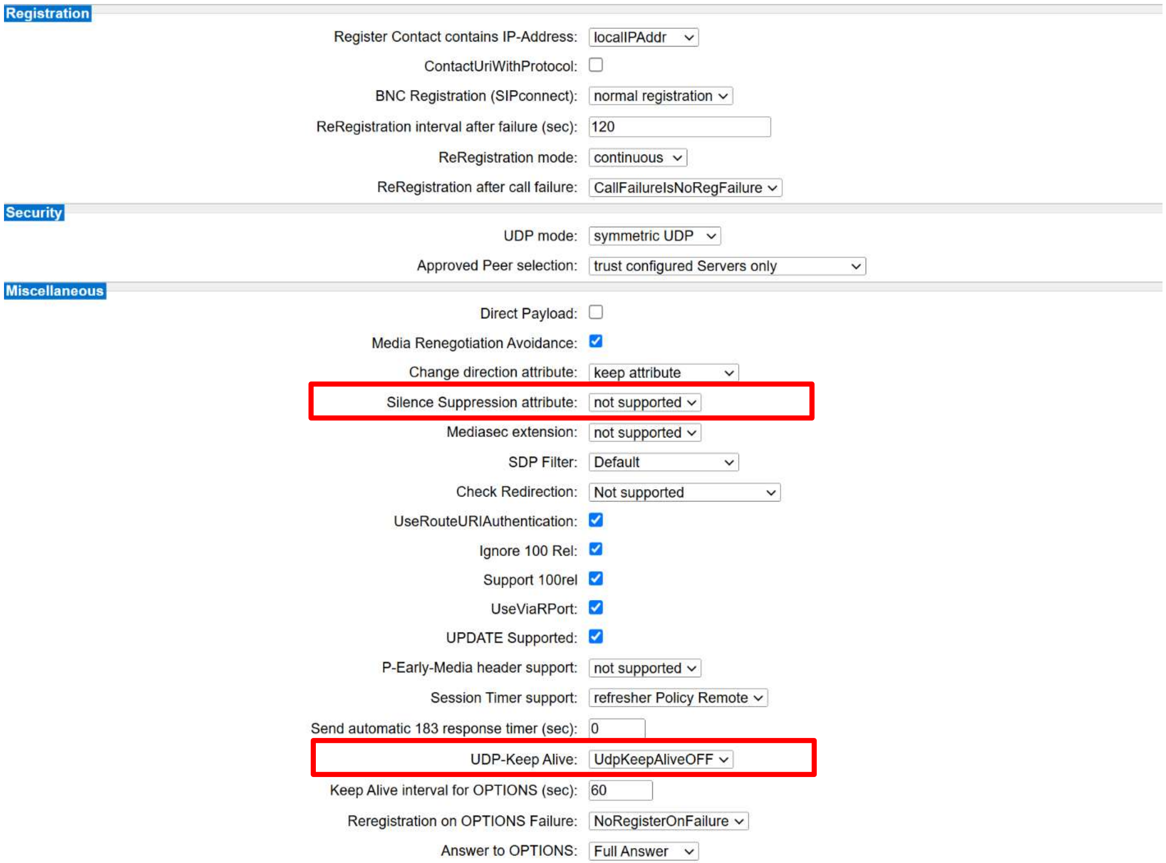
Attention: the following parameters are used to adapt the behavior of the SIP stack to a certain provider implementation. These parameters are defined during the certification process for the provider. Changing these parameters may result in a malfunction of the provider interface.

CLIP / CLIR

CLIP outgoing in From header - display part:	omit
CLIP outgoing in From header - user part:	call number
Outgoing From Header - domain/host part:	domainName
Diversion: From contains original CallingPartyNumber:	☑
Diversion: PAI contains original CallingPartyNumber:	☑
CLIP outgoing in P-Asserted-Id header - display part:	omit
CLIP outgoing in P-Asserted-Id header - user part:	omit
CLIP outgoing in P-Preferred-Id header - display part:	omit
CLIP outgoing in P-Preferred-Id header - user part:	account
CLIP outgoing in Diversion header - display part:	omit
CLIP outgoing in Diversion header - user part:	call number
CLIP outgoing in History-Info header - user part:	omit
CLIR outgoing in From header - display part:	anonymous
CLIR outgoing in From header - user part:	fully anonymous
CLIR outgoing Privacy header:	id
COLP / TIP supported for outgoing calls:	COLP supported

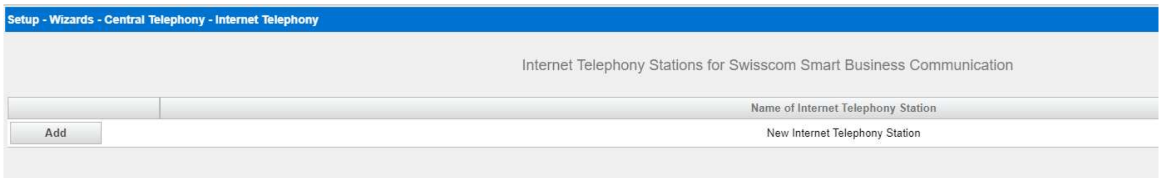
Call number formatting

Incoming call - Called party number:	To header user part
Incoming call - Calling party number:	automatic
Incoming call - Type of number (calling):	automatic
Incoming call - Type of number (called):	automatic
Outgoing call - Type of number (calling):	automatic
Outgoing call - Type of number (called):	automatic
Mapping of provider number:	off
CLIP no Screening support:	not supported
Call No. with international/national prefix:	no
Called number in E164 format:	yes
Route optimization:	allowed
MEX supported:	no
Contact URI contains:	call number:
TCP port used in Contact URI:	ephem. src-port

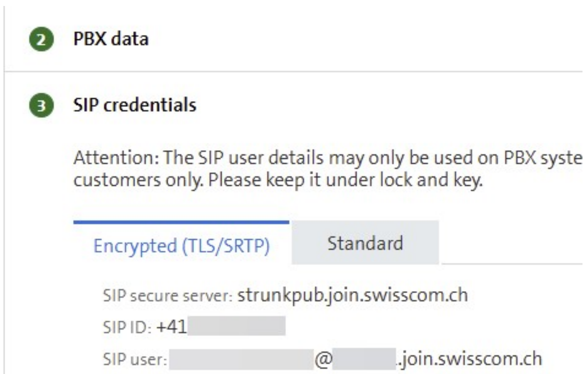


Click «OK & Next»

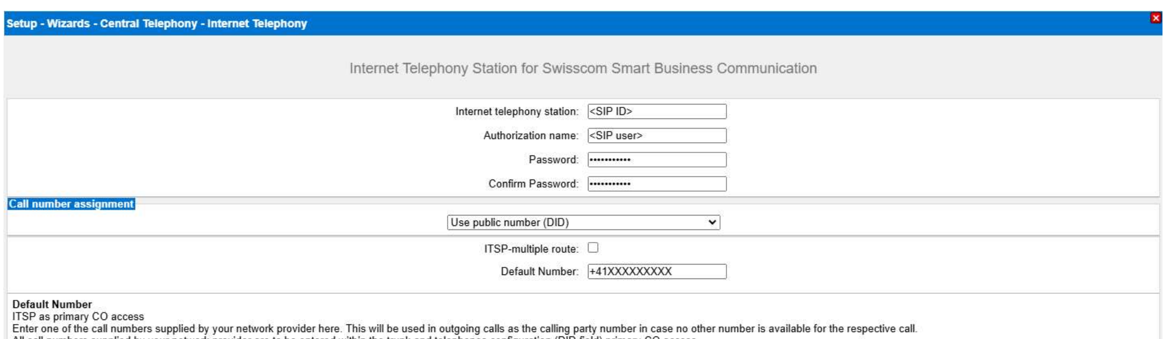
Internet Telephony Station for Swisscom Smart Business Communication



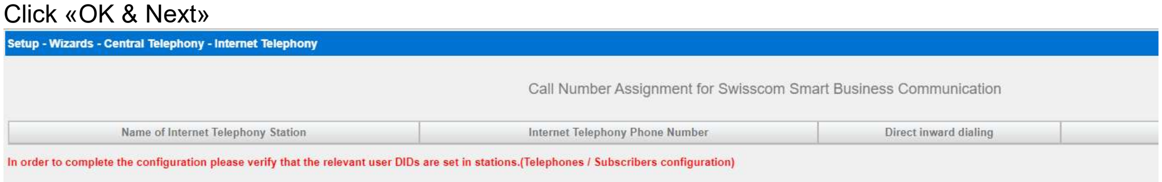
Click the Button "Add"
Enter the Registration Data from Swisscom Smart BCon Portal.



Internet telephony station	= +41XXXXXXX (SIP ID)
Authorization name	= SIP user
Password	= SIP password
Call number assignment	= Use public number (DID)
Default Number	= Main number of the customer in international Format

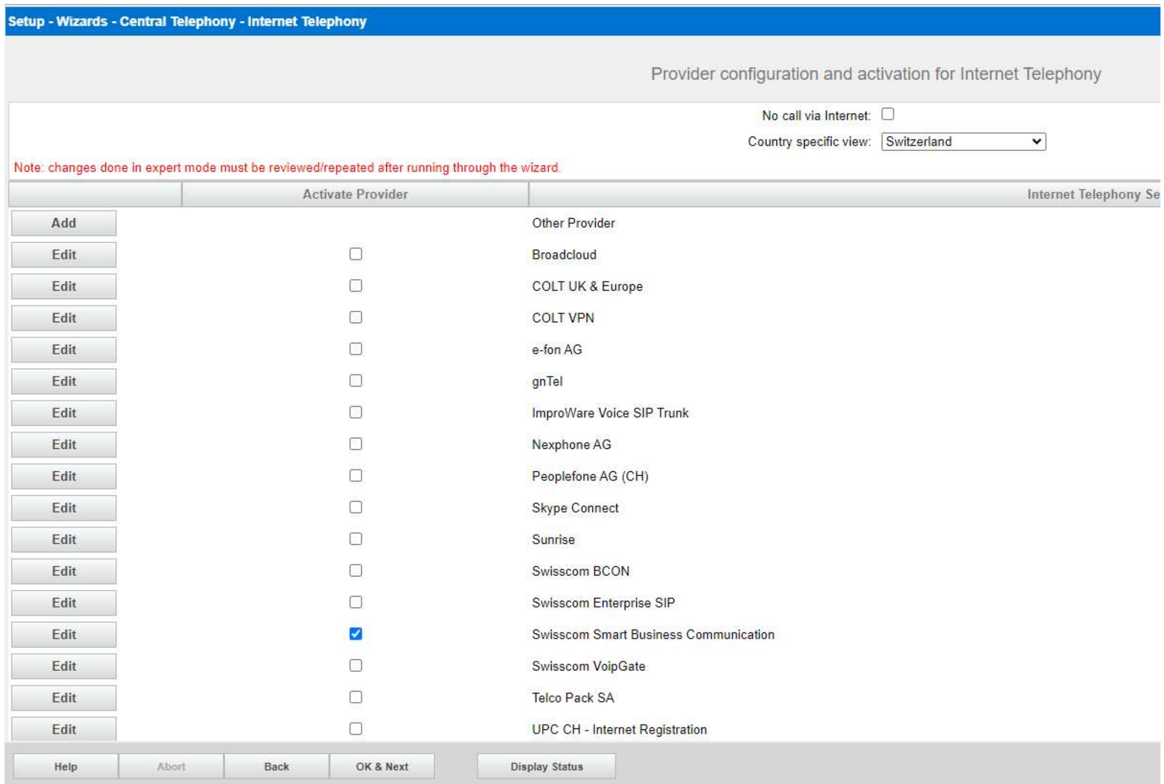


Click «OK & Next»



Click «OK & Next»

Now Wizard is finished and you are back on page Internet Telephony



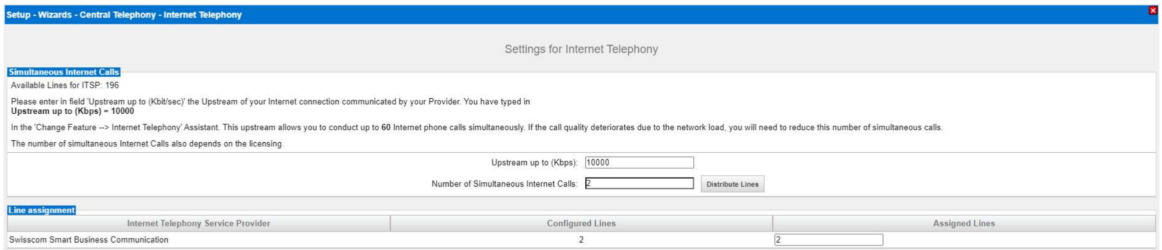
Click «OK & Next»

6 Settings for Internet Telephony (simultaneous Calls to the Provider)

In the next step you can set the number of simultaneous Calls to the Provider.

6.1 Connection over LAN

Here you can define the Upstream, enter the Number of Calls.

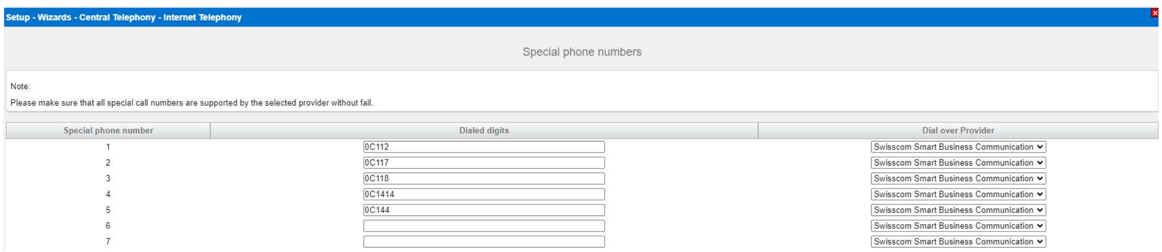


6.2 Special phone numbers

Enter the Special phone numbers. Choose the Provider for outgoing Calls. If Special phone numbers are not routed over ITSP-Trunk change it to dial over Provider ISDN.

Important:
- Service and emergency numbers are not dialed in E164 format!
- Once the configuration is complete, emergency numbers must be tested.

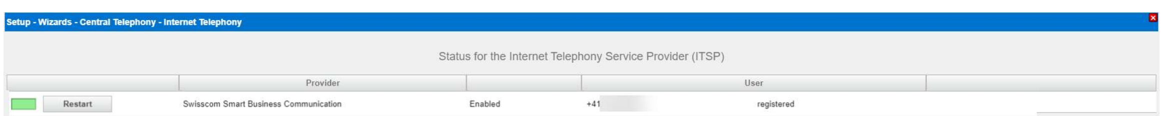
- Europe-wide emergency services
- Police
- Fire Department
- Rega
- Ambulance services
- etc.



Click "OK & Next"

6.3 Status for the Internet Telephony Service Provider (ITSP)

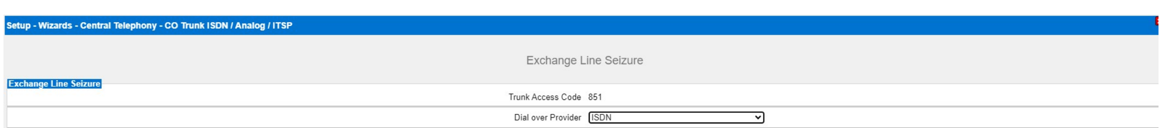
Open Service Center > Diagnostic Status > ITSP Status



If the Provider is not active (registered) you can start first analyzing by click on the Button "Diagnose". (Summary of the config and status messages are shown)

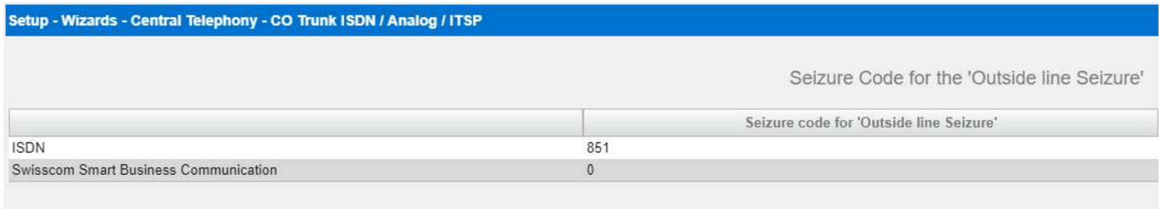
Click "OK & Next"

Following configuration define direction of Public Network.



Click "OK & Next"

Overview of the Seizure Code fort the «Outside line Seizure»

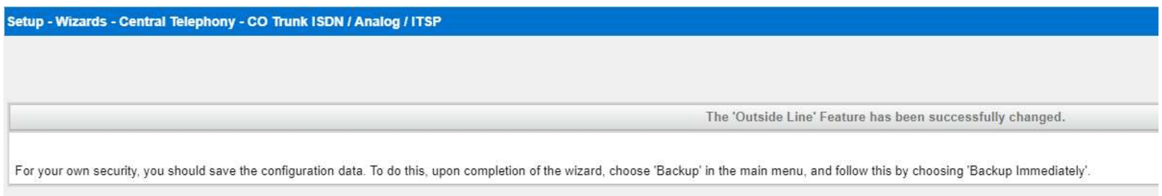


Setup - Wizards - Central Telephony - CO Trunk ISDN / Analog / ITSP

Seizure Code for the 'Outside line Seizure'

	Seizure code for 'Outside line Seizure'
ISDN	851
Swisscom Smart Business Communication	0

Click "OK & Next"



Click "Finish", then the Configuration with Internet Telephony (Wizard) are completed
The following Setting must be done in Expert mode.

7 LCR Least Cost Routing

With the Internet Telephony Wizard LCR is configured as well and can be used for outgoing Dialing immediately.

You can check the Dial Plan afterwards:

Expert mode – Telephony Server > LCR > Dial Plan

Emergency number must be marked as Emergency

Dial Plan

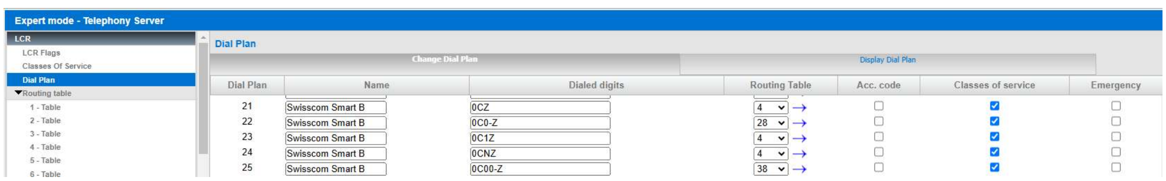
Change Dial Plan | Display Dial Plan

Dial Plan	Name	Dialed digits	Routing Table	Acc. code	Classes of service	Emergency
1	Emergency call	0C112	4	☐	☑	☑
2	Police	0C117	4	☐	☑	☑
3	Fire brigade	0C118	4	☐	☑	☑
4	Emergency call	0C1414	4	☐	☑	☑
5	Rega	0C144	4	☐	☑	☑
6			4	☐	☑	☐

Hint:

Be carefull that Entries with 0Cz, 0C0Z, 0CNZ… can not be marked as Emergency, otherwise no Call Forwarding is possible.

After the Wizard is completed, the Entries 0C1Z and 0CNZ, in the Dial Plan have to be changed from Routing table 5 to Routing table 4. Otherwise no Servicenumber can be dialed.



Dial Plan

Change Dial Plan | Display Dial Plan

Dial Plan	Name	Dialed digits	Routing Table	Acc. code	Classes of service	Emergency
21	Swisscom Smart B	0CZ	4	☐	☑	☐
22	Swisscom Smart B	0C0-Z	28	☐	☑	☐
23	Swisscom Smart B	0C1Z	4	☐	☑	☐
24	Swisscom Smart B	0CNZ	4	☐	☑	☐
25	Swisscom Smart B	0C00-Z	38	☐	☑	☐

Dialplan 4 = Dialrule «SIP»

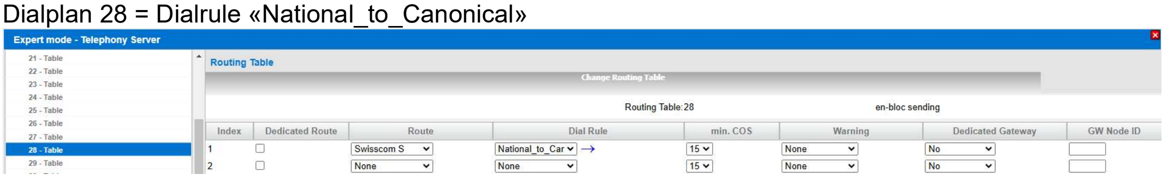
Routing Table

Change Routing Table

Routing Table: 4 en-bloc sending

Index	Dedicated Route	Route	Dial Rule	min. COS	Warning	Dedicated Gateway
1	☐	Swisscom S	SIP	15	None	No
2	☐	None	None	15	None	No

Dialplan 28 = Dialrule «National_to_Canonical»



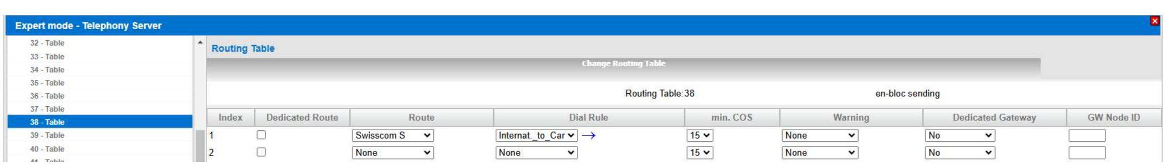
Routing Table

Change Routing Table

Routing Table: 28 en-bloc sending

Index	Dedicated Route	Route	Dial Rule	min. COS	Warning	Dedicated Gateway	GW Node ID
1	☐	Swisscom S	National_to_Car	15	None	No	
2	☐	None	None	15	None	No	

Dialplan 38 = Dialrule «International_to_Canonical»



Routing Table

Change Routing Table

Routing Table: 38 en-bloc sending

Index	Dedicated Route	Route	Dial Rule	min. COS	Warning	Dedicated Gateway	GW Node ID
1	☐	Swisscom S	Internat_to_Car	15	None	No	
2	☐	None	None	15	None	No	

The dialrules are default, no adjustments necessary

8 SIP Trunk recommendations

8.1 DTMF

8.1.1 Sending (to Swisscom)

DTMF Signals SHALL be sent according to the IETF RFC’s 2833/4733. SIP INFO is currently NOT supported. In cases where DTMF Tones are sent in-band in a G.711 RTP Stream, it is transparent to the Network and proper DTMF transmission across the Network can therefore not be cannot guaranteed by either side.

In case of DTMF transmission the SDP MUST contain the *rtpmap* and *fmtp* attributes associated with the DTMF payload.

Swisscom recommendation:

DTMF Signals sent according to the IETF RFC’s 2833/4733 offer the best compatibility with most systems.

8.1.2 Receiving (from Swisscom)

To insure (backward) compatibility with system who do not support/send out-of-band DTMF (RFC 2833), a system MUST be capable to accept both, in-band DTMF (G.711 payload) and out-of-band DTMF (RFC's 2833/4733)

Swisscom recommendation:

In cases the system is depending on DTMF Signals (e.g. Contact Center, Voicemail, etc.) and is not capable to handle both DTMF methods, it is in the responsibility of the solution provider to install appropriate equipment to convert between the two signaling methods.

8.2 Best Practices

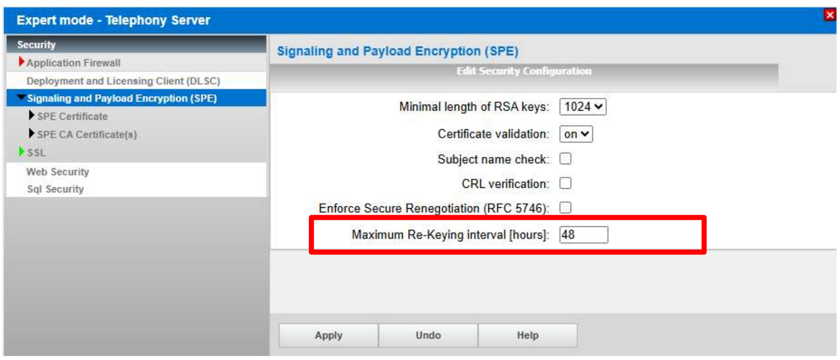
8.2.1 Fax over IP Recommendations & Settings

For Fax Transmissions please read our published recommendation and white papers:

Swisscom Recommendation:
Fax over Smart BCon

8.2.2 TLS v1.2 (RFC5246)

TLS v1.2 suggests an upper limit of 24 hours for session ID lifetimes. This timer can be configured as “Maximum Re-Keying interval [hours]” in Expert mode – Telephony Server – Signaling and Payload Encryption:



Change the value for “Maximum Re-Keying interval [hours]” to 48. This timer is refreshed after system restart or ITSP restart. After expiry of this interval the existing TLS connection will be replaced by a new one which takes around 20 sec. During the 20 sec. no new calls can be established and existing Calls could be terminated.

Note: Current, Mitel is checking for a possibly optimization.

Please adjust this timeout handling via daily or weekly “Garbage collection” for customers who want to have this Re-Keying mechanism out of business hours, e.g.:

Expert mode > Maintenance > Automatic Actions > Garbage Collection