

## Mitel OpenScape Business V3 R4.x

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PBX Configuration Guide for Swisscom Smart Business  
Connect Internet Trunk

**From** UMB  
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**To** IP PBX and  
Communication  
System Integrators  
**For  
information**

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**Review**

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## 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 SBC on 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

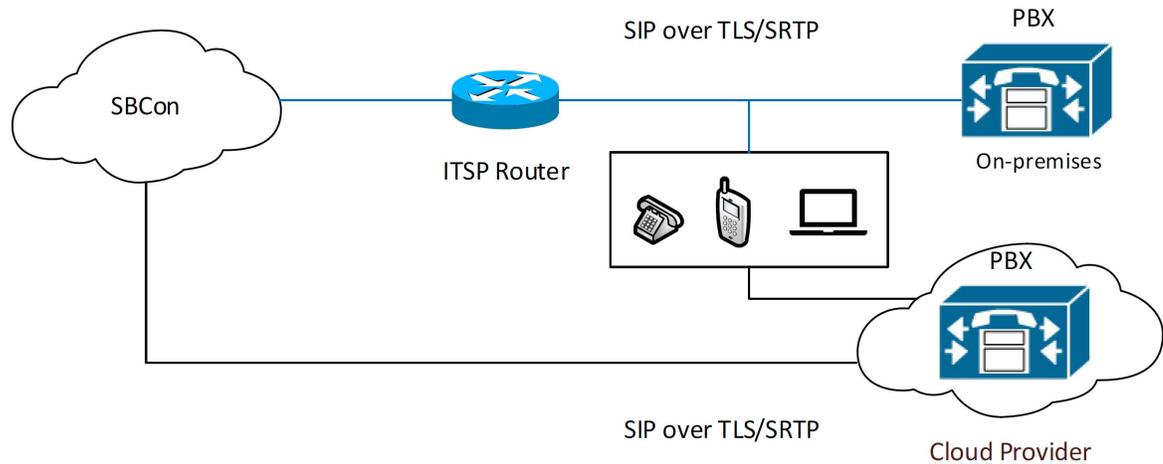
Terms	Abbreviations
SIP	Session Initiation Protocol
IP	Internet Protocol
PBX	Private Branch Exchange
eSBC	Enterprise Session Border Controller
SIP-SIP	eSBC on premises
SIP-Virtual	veSBC in Swisscom Telco Cloud

### 1.4 Referenced documents

[1] [https://wiki.unify.com/images/e/e7/OpenScape\\_Business\\_S\\_Image\\_for\\_Google\\_Cloud.pdf](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).

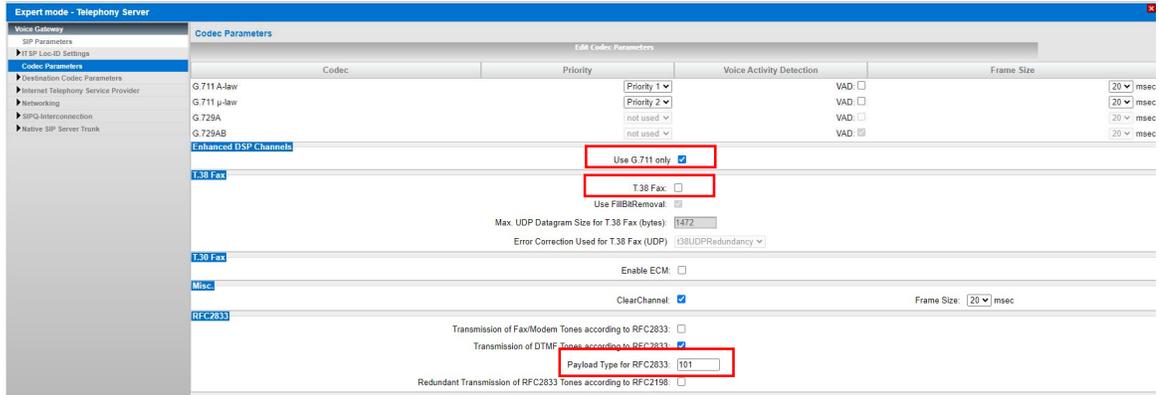
Expert mode - Telephony Server			
Basic Settings		Port Management	
Edit Global Port Management Settings			
Protocol Name	Port Number		
CSP	8900		single
HFA	4060		single
HFA_EXT	4062		single
HFA_TLS	4061		single
HFA_TLS_EXT	4063		single
MEB_SIP	15060		single
RTP_MIN	29100		min (ext. RTP-port range 30528-30887)
SIP	5060		single
SIP_EXT	5070		single
SIP_TLS_SUB	5062		single
SIP_TLS_SUB_EXT	5071		single
SIPS	5061		single
VSL_MULTISITE	8778		single

**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"



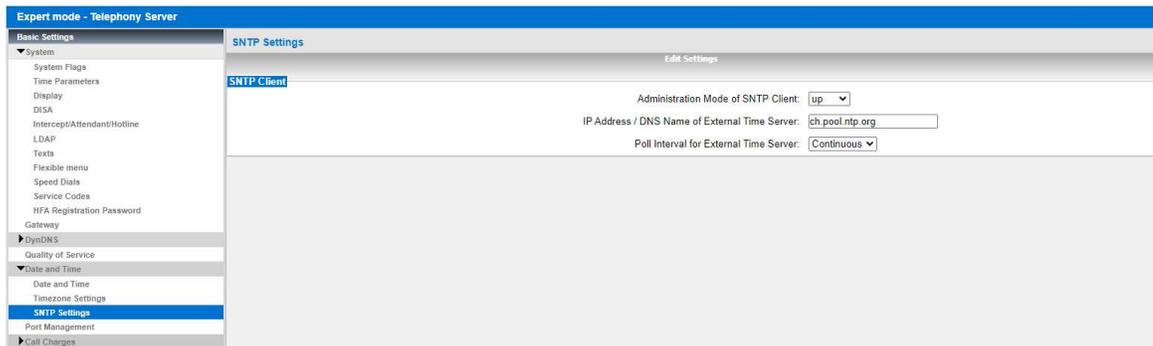
After that the System need a Restart!

## 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](http://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!

The screenshot displays the 'Expert mode - Telephony Server' interface. On the left, a navigation menu lists various settings categories: Basic Settings, System, Gateway, Quality of Service, Port Management, Call Charges, and Voicemail / Announcement Player. The 'Gateway' section is selected. The main area shows the 'Edit Gateway Properties' form. The 'General' tab is active, displaying fields for Customer name (OsBiz S GCF), Contract number, System Name, Gateway Location, Contact Address, System Country Code (Switzerland), Gateway IP Address (10.0.99.2), Gateway Subnet Mask (255.255.255.255), International Prefix (00), National Prefix (0), and Brand (OpenScape Business). The 'Gateway Location' section is highlighted with a red box and contains the following fields: Country code (00 41), Local area code (0 44), and PABX number (274). Below this, the 'Network Parameters' section shows Node ID (0) and a checkbox for Continuous tone.

## 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

The screenshot shows the 'Route' configuration page for 'Trk Grp. 12'. The 'Seizure code' field is set to '855'. Other fields include 'Route Name: Trk Grp. 12', 'CO code (2nd trunk code):', 'Gateway Location' (Country code: 41, Local area code, PABX number), 'PABX number incoming' (Country code, Local area code, PABX number, Location number), 'PABX number outgoing' (Country code, Local area code, PABX number, Suppress station number), 'Overflow route' (None), 'Digit transmission' (en-bloc sending), 'Mobile Extension Number (MEX)', and 'Trusted External Users'.

### Changed Setting

The screenshot shows the 'Route' configuration page for 'Trk Grp. 12'. The 'Seizure code' field is highlighted in red and set to '0'. Other fields include 'Route Name: Trk Grp. 12', 'CO code (2nd trunk code):', 'Gateway Location' (Country code: 41, Local area code, PABX number), 'PABX number incoming' (Country code: 41, Local area code, PABX number, Location number), 'PABX number outgoing' (Country code: 41, Local area code, PABX number, Suppress station number), 'Overflow route' (None), 'Digit transmission' (en-bloc sending), 'Mobile Extension Number (MEX)', and 'Trusted External Users'.

## 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

The screenshot shows the configuration page for Station 0. The 'Clip/Lin' field is highlighted with a red box and contains the value '800800800'. Other fields include: Type: UP0 Station, Call number: 101, First Name: -, Last Name: -, Display: Obelix, Direct inward dialing: 442747621, Device Type: OpenScape Desk Phone CP 400T, and Access: SLUC8 2-1 Master.

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

The screenshot shows the configuration page for Station 1. The 'Clip/Lin' field is highlighted with a red box and is empty. Other fields include: Type: UP0 Station, Call number: 100, First Name: -, Last Name: CP 200 TDM, Display: CP 200 TDM, Direct inward dialing: 442747620, Device Type: OpenScape Desk Phone CP 200/200T/205, and Access: SLUC8 2-2 Master.

## 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

The screenshot displays the 'Expert mode - Telephony Server' interface, specifically the 'Route' configuration page. The left sidebar shows a tree view of 'Trunks/Routing' with 'Trk Grp. 12' selected. The main area is titled 'Route' and contains several sections for configuration:

- Route Name:** Trk Grp. 12
- Seizure code:** 0
- CO code (2nd trunk code):** (empty)
- Gateway Location:**
  - Country code: 41
  - Local area code: (empty)
  - PABX number: (empty)
- PABX number-incoming:**
  - Country code: 41
  - Local area code: (empty)
  - PABX number: (empty)
  - Location number:**  (highlighted with a red box)
- PABX number-outgoing:**
  - Country code: 41
  - Local area code: (empty)
  - PABX number: (empty)
  - Suppress station number:
- Overflow route:** Overflow route: None
- Digit transmission:** Digit transmission: en-bloc sending
- Mobile Extension Number (MEX):** MEX Number: (empty)
- Trusted External Users:** Trusted External Users:

At the bottom, there are 'Apply', 'Undo', and 'Help' buttons.

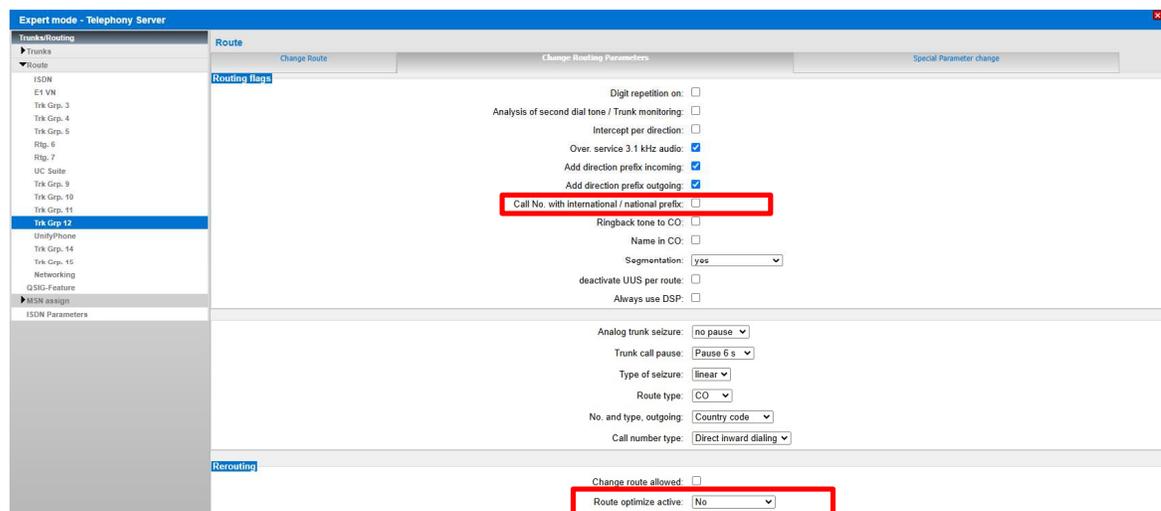
#### 4.8 Routings Parameters (just to check after Wizard is completed)

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



## 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.

The screenshot displays the 'SIP Parameters' configuration page. The left sidebar shows a tree view with 'SIP Parameters' selected. The main content area is titled 'SIP Parameters' and includes a sub-header 'Edit SIP Parameters'. The configuration is organized into several sections:

- SIP Transport Protocol:** SIP via TCP: Yes, SIP via UDP: , SIP via TLS: Yes.
- SIP Registrar:** Period of registration (sec): 120.
- RFC 3261 Timer Values:** Transaction Timeout (msec): 32000.
- SIP Session Timer:** RFC 4028 support: , Session Expires (sec): 1800, Minimal SE (sec): 360.
- DNS Records:** Blocking time for unreachable destination(sec): 60.
- Provider Calls:** Maximum possible Provider Calls: 5.

A red arrow points from the 'RFC 4028 support' checkbox to the 'Minimal SE (sec)' field, indicating the sequence of operations described in the text: enabling RFC 4028 support, adjusting the Minimal SE value, and then disabling RFC 4028 support.

#### 4.10 Installation of the certificate for TLS

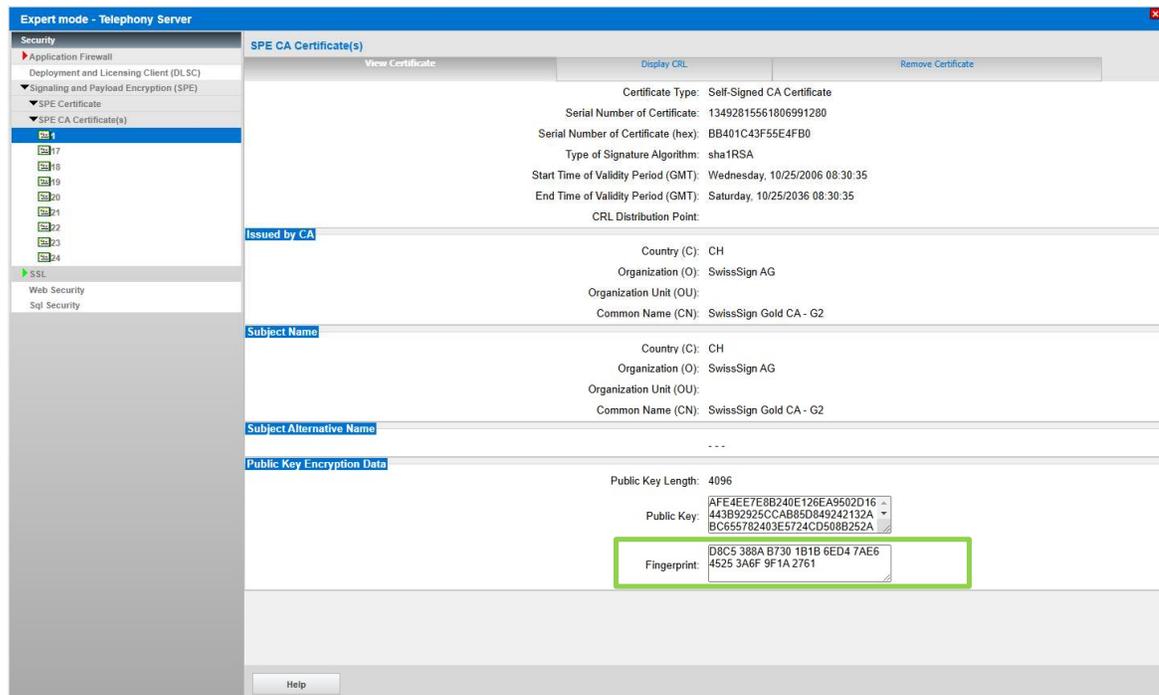
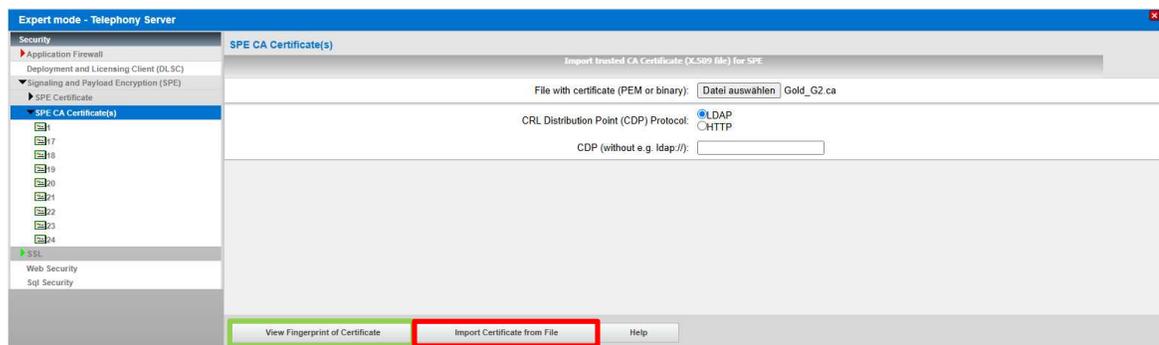
To successfully establish a TLS connection to Swisscom Smart Business Connect Internet trunk, the SwissSign Gold CA - G2 is required as a trusted certificate.

Download: <https://www.swissign.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

The screenshot shows the Unify OpenScope Desk Phone CP400 admin interface. The left sidebar contains a menu with categories: Admin login, Network, System, File transfer, Local functions, Date and time, Speech, Security and policies, Ringer, User mobility, Diagnostics, and Maintenance. The 'Speech' category is expanded, and 'Codec preferences' is selected. The main content area displays the 'Codec preferences' settings for the device. The settings include: Silence suppression (checkbox), Packet size (Automatic), G.722 ranking (priority 2, active), G.711 ranking (priority 3, inactive), and G.729 ranking (priority 1, inactive). There are 'Submit' and 'Reset' buttons at the bottom of the settings panel.

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

The screenshot shows the Unify OpenScope Desk Phone CP600 admin interface. The left sidebar contains a menu with categories: Admin login, Bluetooth, Network, System, File transfer, Local functions, Date and time, Speech, Security and policies, Ringer, User mobility, Diagnostics, and Maintenance. The 'Speech' category is expanded, and 'Codec preferences' is selected. The main content area displays the 'Codec preferences' settings for the device. The settings include: Silence suppression (checkbox), Packet size (Automatic), G.711 ranking (priority 1, inactive), G.722 ranking (priority 2, active), and G.729 ranking (priority 3, inactive). There are 'Submit' and 'Reset' buttons at the bottom of the settings panel. Below the settings panel, a green checkmark icon and the text 'Changes saved successfully' are displayed, along with a 'Refresh' button.

Important: The OpenScope 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

Home Administrators Setup Expert mode Data Backup License Management Service Center

Setup

- Wizards
  - Basic Installation
  - Network / Internet
  - Telephones / Subscribers
  - Central Telephony**
  - User Telephony
  - Security
  - UC Suite
  - Cloud Services
  - Mass Data

**Central Telephony**

- Edit** **CO Trunk ISDN / Analog / ITSP**  
Point-to-multipoint connections (MSN) and PABX number for ISDN connections, and assignment of analog and ITSP trunks
- Edit** **Internet Telephony**  
Access parameters of the Internet Telephony Service Provider (ITSP), e.g., user account, password, SIP station number
- Edit** **Voicemail**  
Access numbers for integrated voicemail. Set up of voicemail boxes
- Edit** **Phone Book / Speed Dialing**  
Set up central speed-dial destinations for the system's internal phone book
- Edit** **Call Detail Recording**  
Set up call detail recording connection parameters for call detail applications
- Edit** **Music on Hold / Announcements**  
Record new melodies and announcements for Music on Hold and announcement before answering
- Edit** **Entrance telephone**  
Set up call allocation and access authorization for the entrance telephone at the analog station connection
- Edit** **Blacklist for incoming calls**  
Define a list of numbers to block unwanted callers permanently
- Edit** **Active Directory Integration Service**  
Set up the Active Directory
- Edit** **Autom. Night Service**  
Automatically configure night service for special days
- Edit** **Special Days**  
Automatically configure special days from calendar

### Internet Telephony «Edit»

Setup - Wizards - Central Telephony - Internet Telephony

Overview

Note: changes done in expert mode must be reviewed/repeated after running through the wizard.  
 Note: At least the configuration of the 'Country code' is needed for features such as 'Internet telephony' and 'MeetMe conference'.

**PABX number**

Country code: 00  (mandatory)

Local area code: 0  (optional)

PABX number:  (optional)

### Deactivate the Flag «No call via Internet»

Setup - Wizards - Central Telephony - Internet Telephony

Provider configuration and activation for Internet Telephony

**No call via Internet:**

Note: changes done in expert mode must be reviewed/repeated after running through the wizard.

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»

Setup - Wizards - Central Telephony - Internet Telephony

Provider configuration and activation for Internet Telephony

No call via Internet:

Country specific view: Switzerland

Note: changes done in expert mode must be reviewed/repeated after running through the wizard.

	Activate Provider	Internet Telephony Service Provider
Add		Other Provider
Edit	<input type="checkbox"/>	Broadcloud
Edit	<input type="checkbox"/>	COLT UK & Europe
Edit	<input type="checkbox"/>	COLT VPN
Edit	<input type="checkbox"/>	e-fon AG
Edit	<input type="checkbox"/>	gnTel
Edit	<input type="checkbox"/>	ImproWare Voice SIP Trunk
Edit	<input type="checkbox"/>	Nexphone AG
Edit	<input type="checkbox"/>	Peoelfone AG (CH)
Edit	<input type="checkbox"/>	Skype Connect
Edit	<input type="checkbox"/>	Sunrise
Edit	<input type="checkbox"/>	Swisscom BCON
Edit	<input type="checkbox"/>	Swisscom Enterprise SIP
Edit	<input checked="" type="checkbox"/>	Swisscom Smart Business Communication
Edit	<input type="checkbox"/>	Swisscom VoipGate
Edit	<input type="checkbox"/>	Telco Pack SA
Edit	<input type="checkbox"/>	UPC CH - Internet Registration

Help Abort Back OK & Next Display Status

Configure the profile «Swisscom Smart Business Communication»

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!

Setup - Wizards - Central Telephony - Internet Telephony

Internet Telephony Service Provider

Provider Name: Swisscom Smart Business Communication  
Enable Provider:   
Secure Trunk:   
Domain Name: XXXXXX.join.swisscom.ch

Provider Registrar

Use Registrar:   
IP Address / Host name: strunkpub.join.swisscom.ch  
Port: 5061  
Reregistration Interval at Provider (sec): 180

Provider Proxy

IP Address / Host name: XXXXXX.join.swisscom.ch  
Port: 5061

Provider Outbound Proxy

Use Outbound Proxy:   
IP Address / Host name: strunkpub.join.swisscom.ch  
Port: 5061

Provider Feature

Route optimize active:

Help Abort Back OK & Next Delete Data

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!

Internet Telephony Service Provider		
<a href="#">Edit Internet Telephony Service Provider</a>	<a href="#">Delete Internet Telephony Service Provider</a>	<a href="#">Add Internet Telephony Station</a>
Provider Name: Swisscom Smart Business Communication		
Enable Provider: <input type="checkbox"/>		
Provider Identifier in System: -		
Domain Name: XXXXXX.join.swisscom.ch		
Transport protocol: udp		
Transport security: secure (tls)		
Media security: SDES only		
<b>Provider Registrar</b>		
Use Registrar: <input checked="" type="checkbox"/>		
IP Address / Host name: strunkpub.join.swisscom.ch		
Port: 5061		
Reregistration Interval at Provider (sec): 180		
<b>Provider Proxy</b>		
IP Address / Host name: XXXXXX.join.swisscom.ch		
Port: 5061		
<b>Provider Outbound Proxy</b>		
Use Outbound Proxy: <input checked="" type="checkbox"/>		
IP Address / Host name: strunkpub.join.swisscom.ch		
Port: 5061		
<b>Provider Inbound Proxy</b>		
Use Inbound Proxy: <input type="checkbox"/>		
IP Address / Host name: 0.0.0.0		
Port: 0		
<b>Provider STUN</b>		
Use STUN: <input type="checkbox"/>		

Now edit the other data under Show Extended SIP Provider Data

Extended SIP Provider Data	
<a href="#">Show Extended SIP Provider Data</a> <input checked="" type="checkbox"/>	
<small>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.</small>	

## 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:	<input checked="" type="checkbox"/>
Diversion: PAI contains original CallingPartyNumber:	<input checked="" type="checkbox"/>
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

**Registration**

Register Contact contains IP-Address:  ▾

ContactUriWithProtocol:

BNC Registration (SIPconnect):  ▾

ReRegistration interval after failure (sec):

ReRegistration mode:  ▾

ReRegistration after call failure:  ▾

---

**Security**

UDP mode:  ▾

Approved Peer selection:  ▾

---

**Miscellaneous**

Direct Payload:

Media Renegotiation Avoidance:

Change direction attribute:  ▾

Silence Suppression attribute:  ▾

Mediasec extension:  ▾

SDP Filter:  ▾

Check Redirection:  ▾

UseRouteURIAuthentication:

Ignore 100 Rel:

Support 100rel:

UseViaRPort:

UPDATE Supported:

P-Early-Media header support:  ▾

Session Timer support:  ▾

Send automatic 183 response timer (sec):

UDP-Keep Alive:  ▾

Keep Alive interval for OPTIONS (sec):

Reregistration on OPTIONS Failure:  ▾

Answer to OPTIONS:  ▾

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

Setup - Wizards - Central Telephony - Internet Telephony

Provider configuration and activation for Internet Telephony

No call via Internet:

Country specific view:

Note: changes done in expert mode must be reviewed/repeated after running through the wizard.

	Activate Provider	Internet Telephony Se
Add		Other Provider
Edit	<input type="checkbox"/>	Broadcloud
Edit	<input type="checkbox"/>	COLT UK & Europe
Edit	<input type="checkbox"/>	COLT VPN
Edit	<input type="checkbox"/>	e-fon AG
Edit	<input type="checkbox"/>	gnTel
Edit	<input type="checkbox"/>	ImproWare Voice SIP Trunk
Edit	<input type="checkbox"/>	Nexphone AG
Edit	<input type="checkbox"/>	Peoplefone AG (CH)
Edit	<input type="checkbox"/>	Skype Connect
Edit	<input type="checkbox"/>	Sunrise
Edit	<input type="checkbox"/>	Swisscom BCON
Edit	<input type="checkbox"/>	Swisscom Enterprise SIP
Edit	<input checked="" type="checkbox"/>	Swisscom Smart Business Communication
Edit	<input type="checkbox"/>	Swisscom VoipGate
Edit	<input type="checkbox"/>	Telco Pack SA
Edit	<input type="checkbox"/>	UPC CH - Internet Registration

Help Abort Back OK & Next Display Status

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.

Settings for Internet Telephony

**Simultaneous Internet Calls**  
Available Lines for ITSP: 196  
Please enter in field 'Upstream up to (Kbit/sec)' the Upstream of your Internet connection communicated by your Provider. You have typed in  
**Upstream up to (Kbps) = 10000**  
In the 'Change Feature -> Internet Telephony' Assistant. This upstream allows you to conduct up to 60 Internet phone calls simultaneously. If the call quality deteriorates due to the network load, you will need to reduce this number of simultaneous calls.  
The number of simultaneous Internet Calls also depends on the licensing

Upstream up to (Kbps):   
Number of Simultaneous Internet Calls:

**Line assignment**

Internet Telephony Service Provider	Configured Lines	Assigned Lines
Swisscom Smart Business Communication	2	<input type="text" value="2"/>

## 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.

Special phone number	Dialed digits	Dial over Provider
1	0C112	Swisscom Smart Business Communication
2	0C117	Swisscom Smart Business Communication
3	0C118	Swisscom Smart Business Communication
4	0C1414	Swisscom Smart Business Communication
5	0C144	Swisscom Smart Business Communication
6		Swisscom Smart Business Communication
7		Swisscom Smart Business Communication

Click "OK & Next"

## 6.3 Status for the Internet Telephony Service Provider (ITSP)

Open Service Center > Diagnostic Status > ITSP Status

Provider	Status	User
Swisscom Smart Business Communication	Enabled	+41 registered

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.

Trunk Access Code: 851  
Dial over Provider: ISDN

Click "OK & Next"

Overview of the Seizure Code for 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"

Setup - Wizards - Central Telephony - CO Trunk ISDN / Analog / ITSP	
The 'Outside Line' Feature has been successfully changed.	
For your own security, you should save the configuration data. To do this, upon completion of the wizard, choose 'Backup' in the main menu, and follow this by choosing 'Backup Immediately'.	

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	Name	Dialed digits	Routing Table	Acc. code	Classes of service	Emergency
1	Emergency call	0C112	4	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>
2	Police	0C117	4	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>
3	Fire brigade	0C118	4	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>
4	Emergency call	0C1414	4	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>
5	Rega	0C144	4	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>
6			4	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>

Hint:

Be careful 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 Servicenummer can be dialed.

Dial Plan	Name	Dialed digits	Routing Table	Acc. code	Classes of service	Emergency
21	Swisscom Smart B	0CZ	4	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>
22	Swisscom Smart B	0C0-Z	28	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>
23	Swisscom Smart B	0C1Z	4	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>
24	Swisscom Smart B	0CNZ	4	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>
25	Swisscom Smart B	0C00-Z	38	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>

Dialplan 4 = Dialrule «SIP»

Index	Dedicated Route	Route	Dial Rule	min. COS	Warning	Dedicated Gateway
1	<input type="checkbox"/>	Swisscom S	SIP	15	None	No
2	<input type="checkbox"/>	None	None	15	None	No

Dialplan 28 = Dialrule «National\_to\_Canonical»

Index	Dedicated Route	Route	Dial Rule	min. COS	Warning	Dedicated Gateway	GW Node ID
1	<input type="checkbox"/>	Swisscom S	National_to_Car	15	None	No	
2	<input type="checkbox"/>	None	None	15	None	No	

Dialplan 38 = Dialrule «International\_to\_Canonical»

Index	Dedicated Route	Route	Dial Rule	min. COS	Warning	Dedicated Gateway	GW Node ID
1	<input type="checkbox"/>	Swisscom S	Internat_to_Car	15	None	No	
2	<input type="checkbox"/>	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 guaranteed by either side.

In case of DTMF transmission the SDP MUST contain the *rtptime* 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

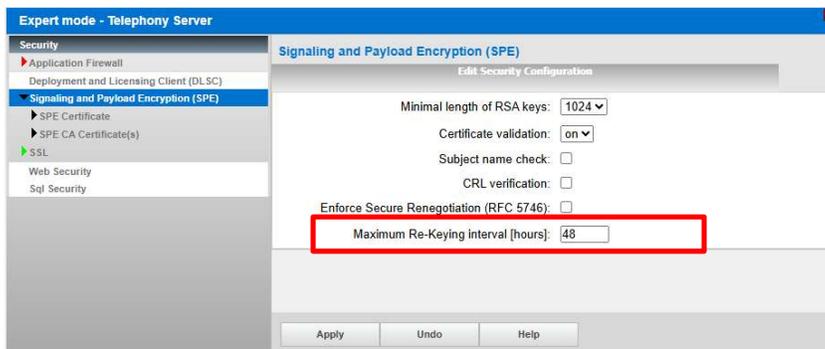
For Fax Transmissions please read our published recommendation and white papers:

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

