

Unify – OpenScape Business – V3 R3.x

SIP Trunk PBX Configuration Guide for Swisscom
Smart Business Connect

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		For	
		information	

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1 Introduction

1.1 Objective and purpose

This Guide describes the SIP Trunk configuration for the IP PBX / communication system, in order to interoperate with the Smart Business Connect Service. This Guide is based on a standard homologated Configuration with an eSBC or IMG, which is the demarcation to the Swisscom Network.

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

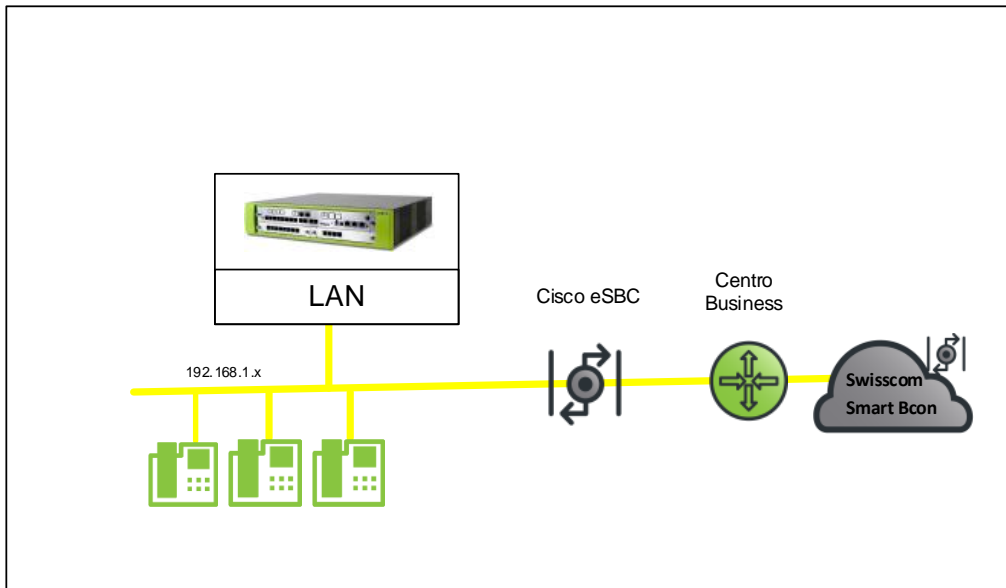
Terms	Abbreviations
SIP	Session Initiation Protocol
IP	Internet Protocol
PBX	Private Branch Exchange
eSBC	Enterprise Session Border Controller
IMG	ISDN Media Gateway

2 Overview PBX

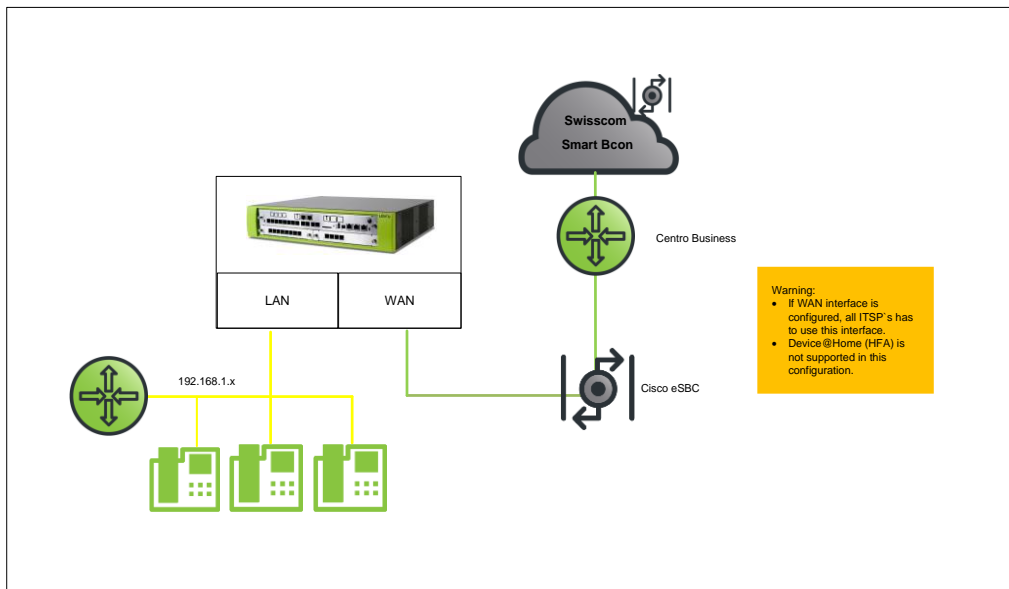
2.1 SIP Trunk network architecture customer side

Schema PBX eSBC Centro Business

Connection over LAN



Connection over WAN



Different configurations are possible, follow the hints in the linked Configuration guides below:
http://wiki.unify.com/index.php/Collaboration_with_VoIP_Providers#General_Configuration_guides

2.2 Hardware requirements

OpenScape Business X / OpenScape Business S

2.3 Software requirements

OpenScape Business

- This document is valid for current OpenScape Business V3 R3

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 (*2nd SIP INVITE or 302*)
- Call forwarding busy (*2nd SIP INVITE or 302*)
- Call forwarding no answer (*2nd SIP INVITE or 302*)
- Attended call transfer
- Blind call transfer
- 3-party conference

3.2 Caveats and known restrictions

- For optimum FAX transmission on the PBX a/b port, at least software version V3 R3.1.1_008 (HF5) must be used and T.38 must be deactivated.
- Billing with Special Arrangement: the billing will be done on the trunk main number instead of the user number.
- Modem not tested.

4 SIP Trunk configuration PBX side

Manual configuration in expert mode

4.1 Configuration of the WAN Address

No static route is needed!

Open "Expert mode> Telephony Server > Network Interfaces > Mainboard > LAN 1 (WAN)" => Edit LAN 1 Interfaces

Open "Expert mode> Telephony Server > Network Interfaces > Mainboard > LAN 1 (WAN)" edit "LAN 1-Interface"

Configure the WAN/LAN Interface with the same IP Address which is configured in the Swisscom Smart BCon Portal.

- „Internet Service Provider-Selection“, choose „LAN-Connection Type TCP/IP“
- Deactivate „Internet access via an external Router“
- Deactivate „Automatic Address Configuration (via DHCP)“
- Fill in the fix IP Address & Subnet Mask
- Deactivate "NAT" (internal firewall is switched off)

After this change open "Experte mode > Telephony Server > Mainboard > LAN 1 (WAN)" > Show LAN1 Mode, "The WAN is Use as LAN Connection Type TCP/IP" Should be displayed.

Expert mode - Telephony Server

Network Interfaces

- Mainboard
 - Host Name
 - LAN 1 (WAN)**
 - LAN 2
 - LAN 3 (Admin)
 - FTP-Server
 - DHCP
- Application Board
 - Host Name
 - LAN 1
 - LAN 2

Mainboard LAN 1 (WAN) [Show LAN 1 Mode](#) [Edit LAN 1 Interface](#)

Internet Service Provider Selection: LAN Connection Type TCP/IP ▼

Internet access via an external Router:

Automatic Address Configuration (via DHCP):

IP Address: 192.168.1.3

Subnet Mask: 255.255.255.0

MAC Address: 00:1a:e0:ac:69:ff

Ethernet Link Mode: Auto ▼

Max. Data Packet Size (bytes): 1500

Network Address Translation:

Bandwidth Control for Voice Connections: None ▼

Bandwidth for Downloads: 10000

Bandwidth for Uploads: 10000

Bandwidth Used for Voice/Fax (%): 80

IEEE802.1p/q Tagging:

IEEE802.1p/q VLAN ID: 0

Expert mode - Telephony Server

Network Interfaces

- Mainboard
 - Host Name
 - LAN 1 (WAN)**
 - LAN 2
 - LAN 3 (Admin)

Mainboard LAN 1 (WAN) [Show LAN 1 Mode](#) [Edit LAN 1 Interface](#)

The WAN is Used as
LAN Connection Type TCP/IP

Now the WAN Network-Configuration is done.

4.2 Default SIP-Port

From V2R3 the default SIP Port for ITSPs (SIP_EXT) is configured to "5070".

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

The SIP_EXT Port in WBM must be set to 5060 for Swisscom Smart BCon. Reboot of the system is needed.

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

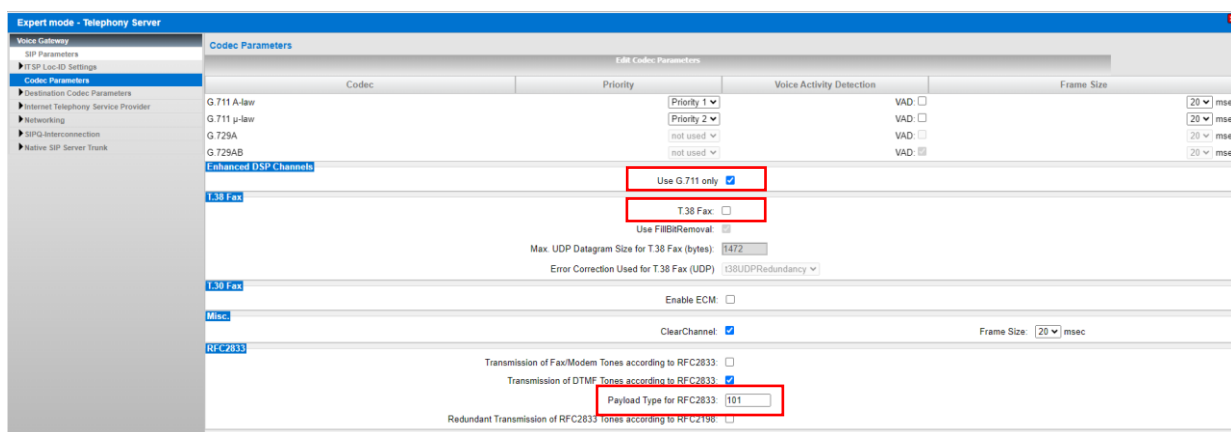
Protocol Name	Port Number	Port
CSP	8800	single
HFA	4060	single
HFA_EXT	4062	single
HFA_TLS	4061	single
HFA_TLS_EXT	4063	single
MEB_SIP	15060	single
RTP_MIN	29100	min. (ext. RTP-port range 30274-30529)
SIP	5070	single
SIP_EXT	5060	single
SIP_TLS_SUB	5062	single
SIP_TLS_SUB_EXT	5071	single
SIPS	5061	single
VSL_MULTISITE	8778	single

If the Customer is using SIP Clients, they must register with SIP Port 5070!

4.3 Set Codec Parameters and Payload Type

Open "Expert mode > Voice Gateway > Codec Parameters"

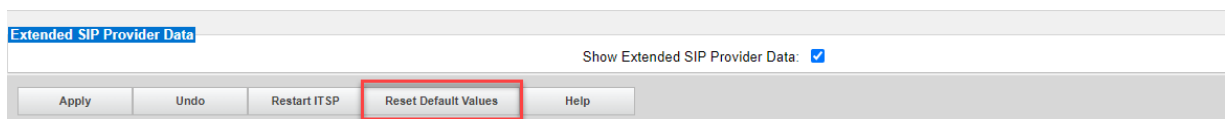
Change "Payload Type for RFC2833" from 98 to **101**
Deactivate "T.38 FAX"



After that the System need a Restart!

4.3.1 «Prack» Support

Since OpenScape Business V2 R7 Prack Support is enabled in the Profile «Swisscom Smart Business Communication» PRACK (Provisional Response Acknowledgement). If your PBX is configured with this Profile already, this change will not be active automatically. In this Case following Procedure "Reset Default Values" should be applied in Expertmode.

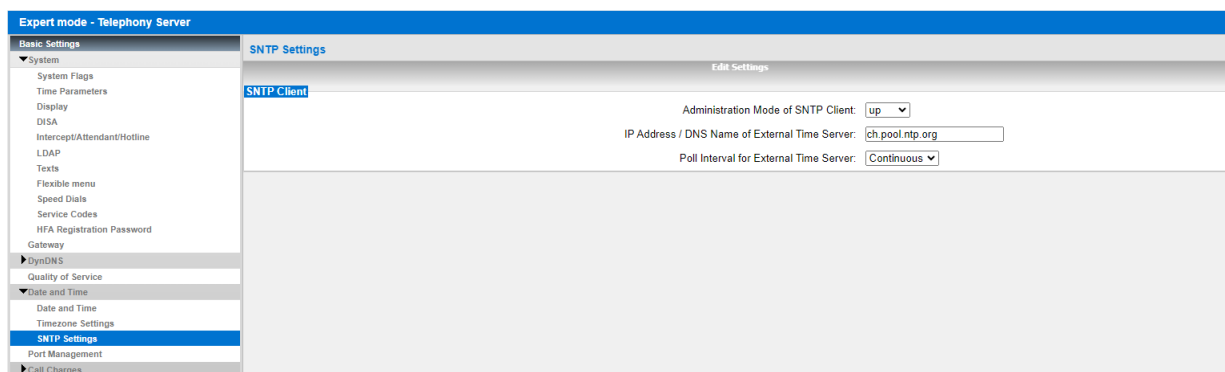


4.4 SNTP configuration

Due to the All IP 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



4.5 Gateway Location (just to check)

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

Open Gateway-Location, and enter the follow Parameters

Country Code = 41

Loca area code = 44

PABX Number = 27476 (Systemnumber without DDI Range)

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

The screenshot shows the 'Expert mode - Telephony Server' configuration window. The left sidebar lists various settings categories, with 'Gateway' selected. The main area is titled 'Edit Gateway Properties' and contains several sections:

- General:** Includes fields for Customer name, Contract number, System Name, Gateway Location (set to 'OsBiz_X'), Contact Address, System Country Code (set to 'Switzerland'), Gateway IP Address, Gateway Subnet Mask (255.255.255.0), International Prefix (00), National Prefix (0), and Brand (OpenScape Business).
- Gateway Location:** This section is highlighted with a red box and contains three input fields:
 - Country code: 00 41
 - Local area code: 0 44
 - PABX number: 27476
- Network Parameters:** Includes a Node ID field set to '1'.
- Internal dial tone:** Includes a checked checkbox for 'Continuous tone'.

4.6 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 configuration page for 'Route Trk. 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), 'PABX number-incoming' (Country code: , Local area code: , PABX number: , Location number:), 'PABX number-outgoing' (Country code: , Local area code: , PABX number: , Location number: , Suppress station number:), 'Overflow route' (None), 'Digit transmission' (en-bloc sending), 'Mobile Extension Number (MEX)' (MEX Number:), and 'Trusted External Users' (Trusted External Users:).

Changed Setting

The screenshot shows the configuration page for 'Route Trk. 12' after the seizure code has been changed. The 'Seizure code' field is now set to '0'. All other configuration parameters remain the same as in the default setting screenshot.

4.7 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.

User with Clip No Screening => Clip Setting

Station	Edit station parameters	Edit station flags	Edit Group/CFW
Station - 0	<p>Type: UP0 Station</p> <p>Call number: <input type="text" value="101"/></p> <p>First Name: <input type="text" value="-"/></p> <p>Last Name: <input type="text" value="-"/></p> <p>Display: <input type="text" value="Obelix"/></p> <p>Direct inward dialing: <input type="text" value="442747621"/></p> <p>Device Type: OpenScape Desk Phone CP 400T</p> <p>Clip/Lin: <input type="text" value="00000000"/></p> <p>Access: SLUC8 2-1 Master</p>		

User without Clip No Screening => Clip Setting

Station	Edit station parameters	Edit station flags	Edit Group/CFW
Station - 1	<p>Type: UP0 Station</p> <p>Call number: <input type="text" value="100"/></p> <p>First Name: <input type="text" value="-"/></p> <p>Last Name: <input type="text" value="CP 200 TDM"/></p> <p>Display: <input type="text" value="CP 200 TDM"/></p> <p>Direct inward dialing: <input type="text" value="442747620"/></p> <p>Device Type: OpenScape Desk Phone CP 200/200T/205</p> <p>Clip/Lin: <input type="text" value="-"/></p> <p>Access: SLUC8 2-2 Master</p>		

4.8 PABX Number

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

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

Activate only Location number

The screenshot shows the configuration page for a route in the Expert mode - Telephony Server. The left sidebar lists various trunk groups, 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)
- PABX number-outgoing:**
 - Country code: 41
 - Local area code: (empty)
 - PABX number: (empty)
 - Location number: (highlighted with a red box)
 - Suppress station number:
- Overflow route:** None
- Digit transmission:** en-bloc sending
- Mobile Extension Number (MEX):** (empty)
- Trusted External Users:**

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

4.9 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	= activated
Segmentation	= yes
No. and type, outgoing	= Country code

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

The screenshot shows the 'Expert mode - Telephony Server' interface. On the left, a tree view shows 'Trunks/Routing' expanded to 'Trk Grp. 42'. The main area is titled 'Route' and contains several sections:

- Routing flags:**
 - Digit repetition on:
 - Analysis of second dial tone / Trunk monitoring:
 - Intercept per direction:
 - Over. service 3.1 kHz audio:
 - Add direction prefix incoming:
 - Add direction prefix outgoing:
 - Call No. with international / national prefix:
 - Ringback tone to CO:
 - Name in CO:
 - Segmentation:
 - deactivate UUS per route:
 - Always use DSP:
- Trunk parameters:**
 - Analog trunk seizure:
 - Trunk call pause:
 - Type of seizure:
 - Route type:
 - No. and type, outgoing:
 - Call number type:
- Rerouting:**
 - Change route allowed:
 - Route optimize active: (highlighted with a red box)

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

*For details see page 21

5 Establishment of the ITSP Smart Business Communication Trunk

Setup (Wizards) > Central Telephony > Internet Telephony

The screenshot shows the UMB configuration interface. At the top, there is a navigation bar with links: Home, Administrators, Setup, Expert mode, Data Backup, License Management, and Service Center. On the left, a sidebar menu is open to 'Setup', with 'Wizards' expanded to show 'Central Telephony' selected. The main content area is titled 'Central Telephony' and lists several configuration options, each with an 'Edit' button:

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

Internet Telephony «Edit»

The screenshot shows the 'Internet Telephony' configuration page. At the top, the breadcrumb is 'Setup - Wizards - Central Telephony - Internet Telephony'. Below the breadcrumb is a tab labeled 'Overview'. A red note states: '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'.' Below the note, there is a section for 'PABX number' configuration:

- Country code: 00 (mandatory)
- Local area code: 0 (optional)
- PABX number: (optional)

Deactivate the Flag «No call via Internet»

The screenshot shows the 'Internet Telephony' configuration page. At the top, the breadcrumb is 'Setup - Wizards - Central Telephony - Internet Telephony'. Below the breadcrumb is a tab labeled 'Provider configuration and activation for Internet Telephony'. A red note states: 'Note: changes done in expert mode must be reviewed/repeated after running through the wizard.' Below the note, there is a checkbox labeled 'No call via Internet:' which is checked.

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:

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

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

Configure the Profil «Swisscom Smart Business Communication»

<input type="button" value="Edit"/>	<input checked="" type="checkbox"/>	Swisscom Smart Business Communication
-------------------------------------	-------------------------------------	---------------------------------------

5.1 Internet Telephony Service Provider

Enter the Data from Swisscom Smart BCon Portal.

Enable Provider	= YES
Domain Name	= IP Address Cisco eSBC (Cisco 88x)
Provider Registrar IP Adresse /Host Name	= IP Address Cisco eSBC (Cisco 88x)
Port	= SIP Port (5060)
Reregistrtrtion-interval am Provider (s)	= 120
Provider Proxy	= IP Address Cisco eSBC (Cisco 88x)
Port	= SIP Port (5060)

Setup - Wizards - Central Telephony - Internet Telephony

Internet Telephony Service Provider

Provider Name: Swisscom Smart Business Communication

Enable Provider:

Secure Trunk:

Domain Name:

Provider Registrar

Use Registrar:

IP Address / Host name:

Port:

Reregistration Interval at Provider (sec)

Provider Proxy

IP Address / Host name:

Port:

Provider Outbound Proxy

Use Outbound Proxy:

IP Address / Host name:

Port:

Provider Feature

Route optimize active:

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.

5.2 Internet Telephony Station for Swisscom Smart Business Communication

Setup - Wizards - Central Telephony - Internet Telephony

Internet Telephony Stations for Swisscom Smart Business Communication

	Name of Internet Telephony Station
Add	New Internet Telephony Station

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

PBX Daten (lokal) ⓘ

Hersteller*

Typ*

Version*

IP-Adresse (PBX)* ⓘ

SIP-Credentials ausblenden

SIP Server	gsmw.swisscom.ch
SIP URI	sip:00123456789012345678901234567890@gsmw.swisscom.ch
SIP-Benutzer	LVCV
SIP-Passwort	1234567890

Internet -telephony station = +4144XXXXXXX
 Authorization name = LVCV
 Password = SIP-Password
 Call number assignment = Use public number (DID)
 Default Number = Main number of the customer in international Format

Setup - Wizards - Central Telephony - Internet Telephony

Internet Telephony Station for Swisscom Smart Business Communication

Internet telephony station:

Authorization name:

Password:

Confirm Password:

Call number assignment

ITSP-multiple route:

Default Number:

Default Number
 ITSP as primary CO access
 Enter one of the call numbers supplied by your network provider here. This will be used in outgoing calls as the calling party number in case no other number is available for the respective call.
 All call numbers supplied by your network provider are to be entered within the trunk and telephones configuration (DID field) primary CO access.

Click «OK & Next»

Setup - Wizards - Central Telephony - Internet Telephony

Call Number Assignment for Swisscom Smart Business Communication

Name of Internet Telephony Station	Internet Telephony Phone Number	Direct inward dialing
In order to complete the configuration please verify that the relevant user DIDs are set in stations.(Telephones / Subscribers configuration)		

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"/>	Peopelfone 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.

6.1.1 Connection over LAN:

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

Setup - Wizards - Central Telephony - Internet Telephony

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"/>

6.1.2 Connection over WAN:

In this case Upstream is configured in the WAN Interface. (See Printscreen: WAN Interface)

Setup - Wizards - Central Telephony - Internet Telephony

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.

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"/>

Click "OK & Next"

Now the numbers of simultaneous Calls will be assigned to 1st ITSP for (Direction Trunk Grp. 12)

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.

- international emergency number
- Police
- Fire Department
- Rega
- emergency number
- 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 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 Dialplan afterwards:

Expert mode – Telephony Server > LCR > Dialplan

Emergency number must be marked as Emergency

(If a number that was configured as an emergency number (emergency column checkbox selected) is dialed, and no free line is available, then a line that is being used for a non-emergency number (emergency column checkbox cleared) is disconnected and then made available automatically for the emergency number.)

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 Dialplan have to be changed from Routing table 5 to Routing table 4. Otherwise no Shortnumber can be dialed.

Dial Plan	Name	Dialed digits	Routing Table	Acc. code	Classes of service	Emergency
19	Local	81CNZ	1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>
20	International	81C00-Z	1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>
21	Swisscom Smart B	0CZ	4	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>
22	Swisscom Smart B	0C0-Z	4	<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	4	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>
26	Standard	850CZ	6	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>

Dialplan 4

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

Dialrule «SIP»

Rule Name	Dial rule format	Network access	Type
2 SIP	A	Main network supplie	Unknown

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 systems 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](#)