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SIP Trunk Configuration Guide for Swisscom Enterprise SIP Standard / WAN

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To	IP PBX and Communication System Integrators
For information	

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Review

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1.5	06.09.2021	Daniele Varone	New Release V3 R1.x

Release

Version	Date of release	Released by	Comments
1.2		Product Manager	Verification of the Configuration Guide

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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 in order to interoperate with the Enterprise SIP Standard / WAN service which using demarcation with eSBC or IMG at customer premise.

This documents is not applicable for Enterprise SIP Unbundled!

More information about the Swisscom Enterprise SIP is available here:
https://www.swisscom.ch/de/business/enterprise/angebot/wireline-networks-bundles/business-voice/sip-trunking.html

1.2 Target audience, requirements for the reader

IP PBX and Communication System Integrators.

1.3 Terms, abbreviations

Terms	Abbreviations
SIP	Session Initation Protocol
IP	Internet Protocol
PBX	Public Branch Exchange

1.4 Referenced documents

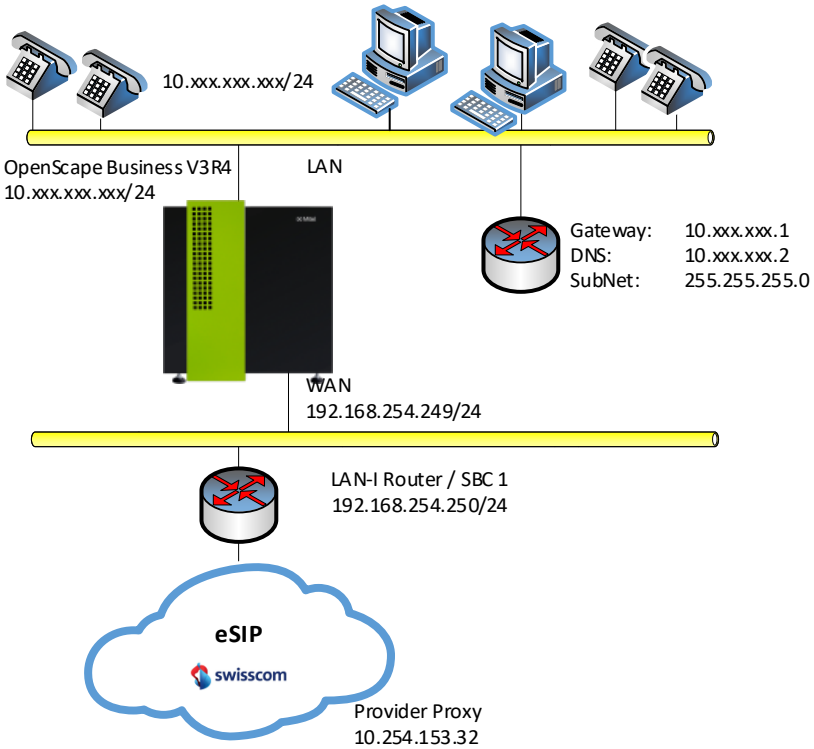
[1] SIP Trunk Technical User Network Interface Guide for Enterprise SIP (UNI Guide)

2 Overview

2.1 SIP Trunk network architecture

Schema PBX

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 / S

2.3 Software requirements

OpenScape Business

- 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, 0900)
- 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
- E112 with geolocation
- Call hold/resume
- Music on hold
- Call forwarding unconditional
- Call forwarding busy
- Call forwarding no answer
- Attended call transfer
- Blind call transfer
- T.38 Fax
- Fax with G.711 pass-through
- 3-party conference
- Call forwarding unconditional *(2nd SIP INVITE or 302) NEW!*
- Call forwarding busy *(2nd SIP INVITE or 302) NEW!*
- Call forwarding no answer *(2nd SIP INVITE or 302) NEW!*

3.2 Caveats and known restrictions

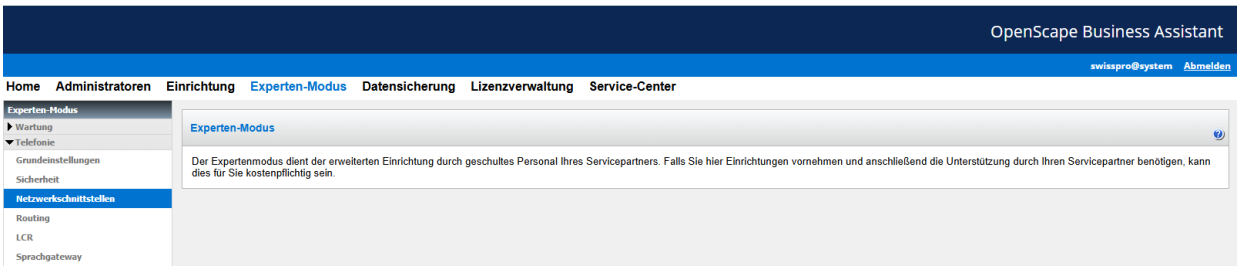
- Modem not tested
- Billing with Special Arrangement: the billing will be done on the trunk main number instead of the user number.

4 SIP Trunk configuration PBX side

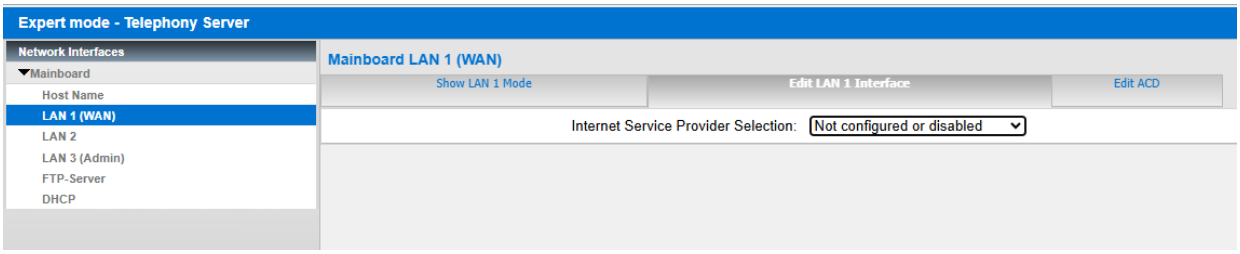
4.1 Manual configuration in expert mode

4.1.1 WAN address configuration

Expert mode > Telephony Server > Network Interfaces



Change Internet Service Provider Selection, chance from “Not configured or disabled” to LAN Connection Type TCP/IP

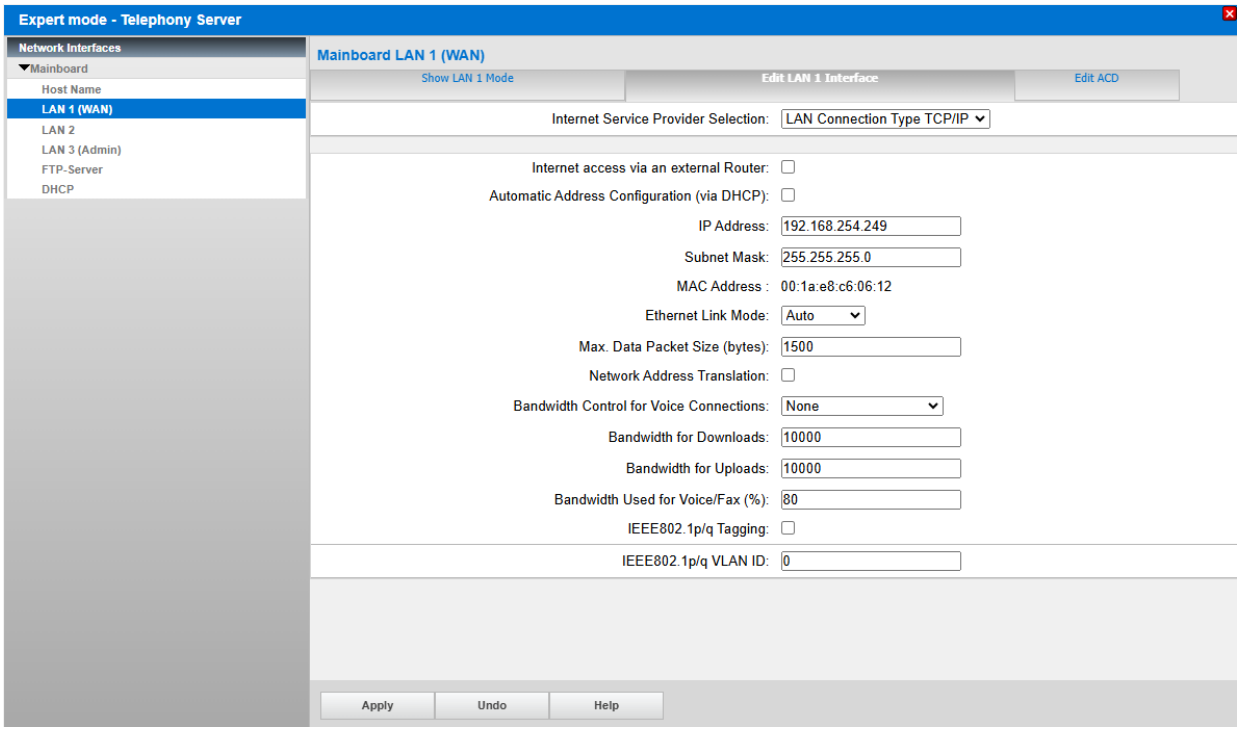


Apply

With this configuration, only SIP data is sent via the WAN interface!

- Internet Service Provider selection, select "LAN connection type TCP/IP"
- Disable Internet access via an external Router
- Disable Automatic Address Configuration (via DHCP)
- Fill the fixed IP address & Subnet Mask
- Disable Network Address Translation

Expert mode > Telephony Server > Network Interfaces > Mainboard > LAN1 (WAN) > edit LAN 1 Interface



Now the WAN Network-Configuration is done.

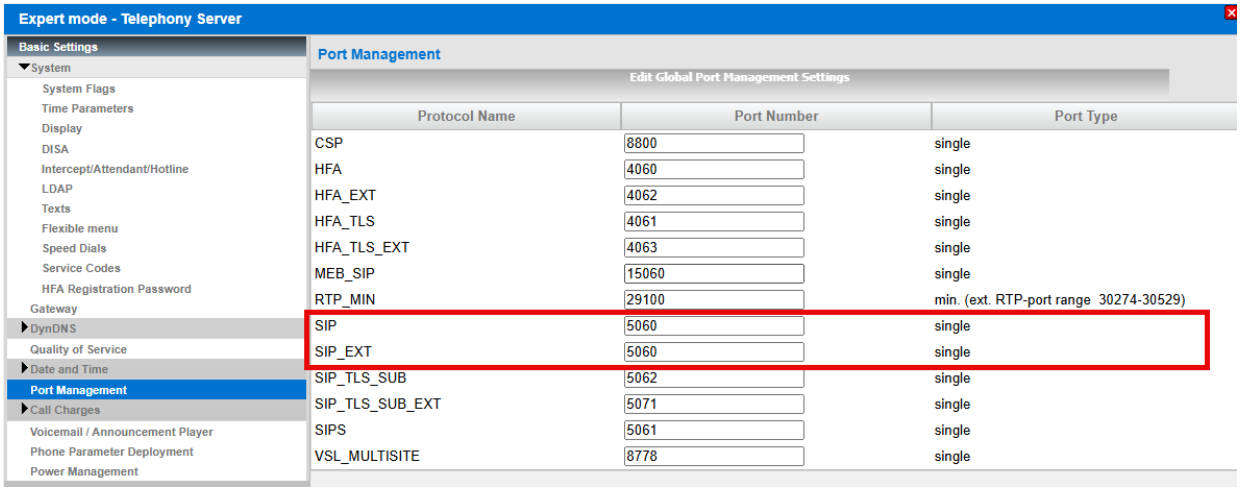
4.1.2 Default SIP-Port

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

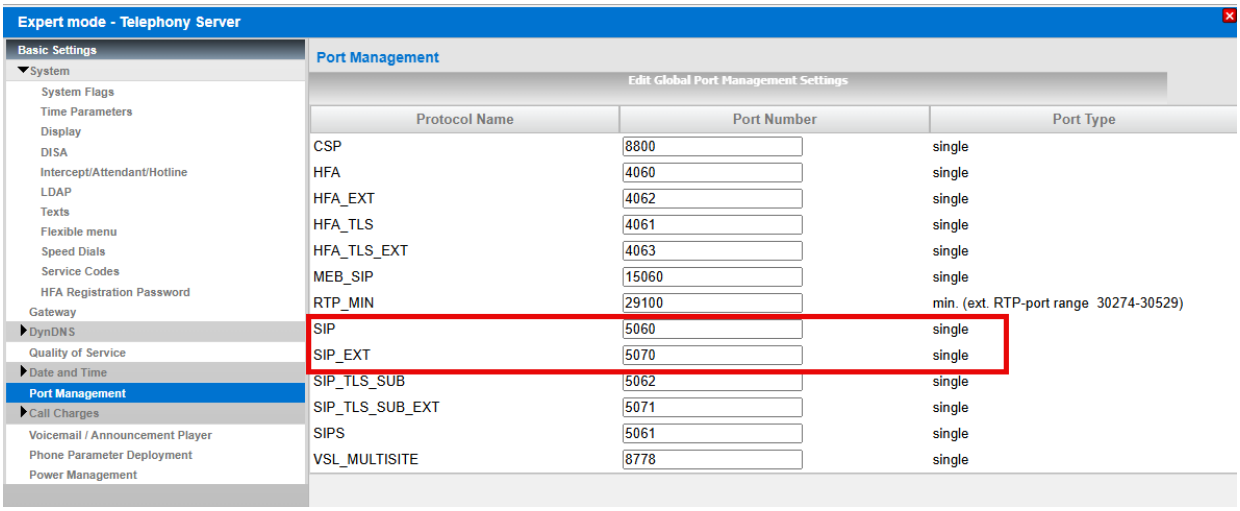
The SIP_EXT Port in WBM must be set to 5060 for Swisscom Enterprise SIP.
Reboot of the system is needed.

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

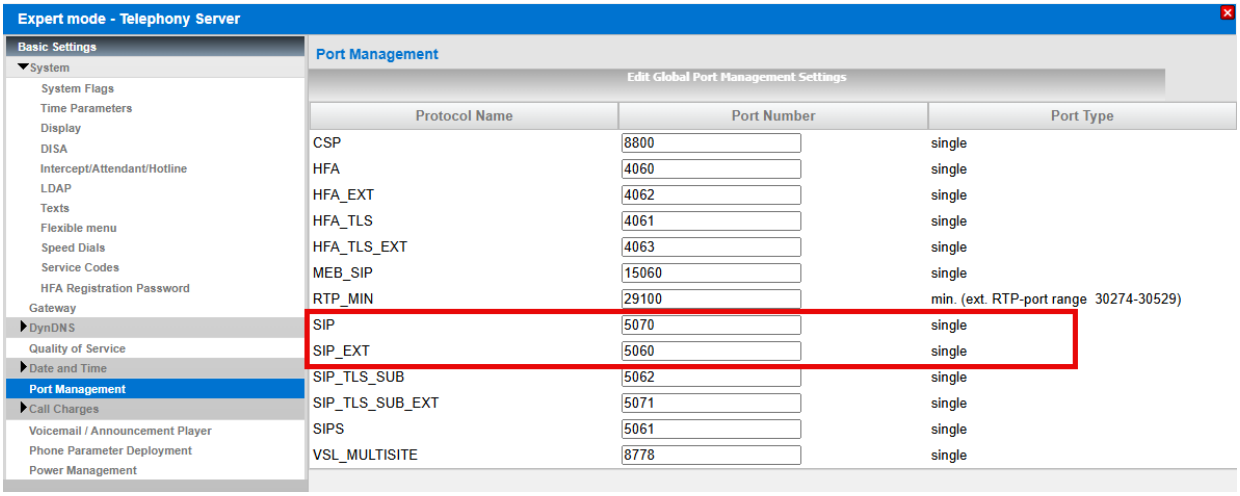
After an update



New installed system from V2R3



Please change the SIP Port and the SIP_EXT



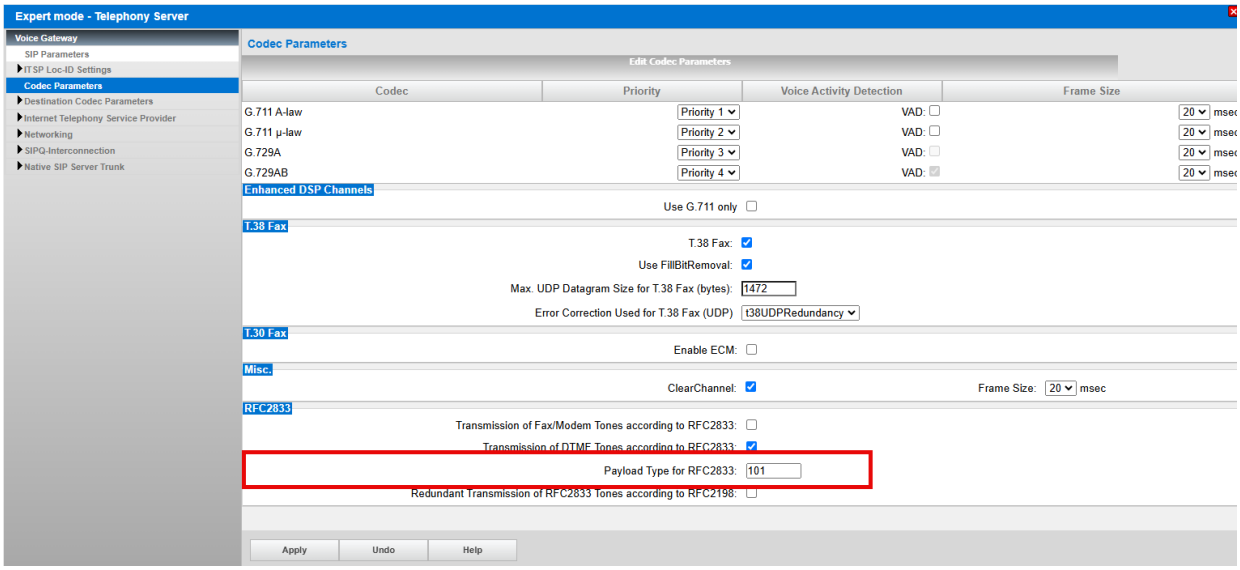
If the Customer is using SIP Clients, they must register with SIP Port 5070!

4.1.3 Set Codec Parameters and Payload Typ

Please check setting

Change "Payload Type for RFC2833" from 98 to 101

Expert mode > Telephony Server > Voice Gateway > Codec Parameters



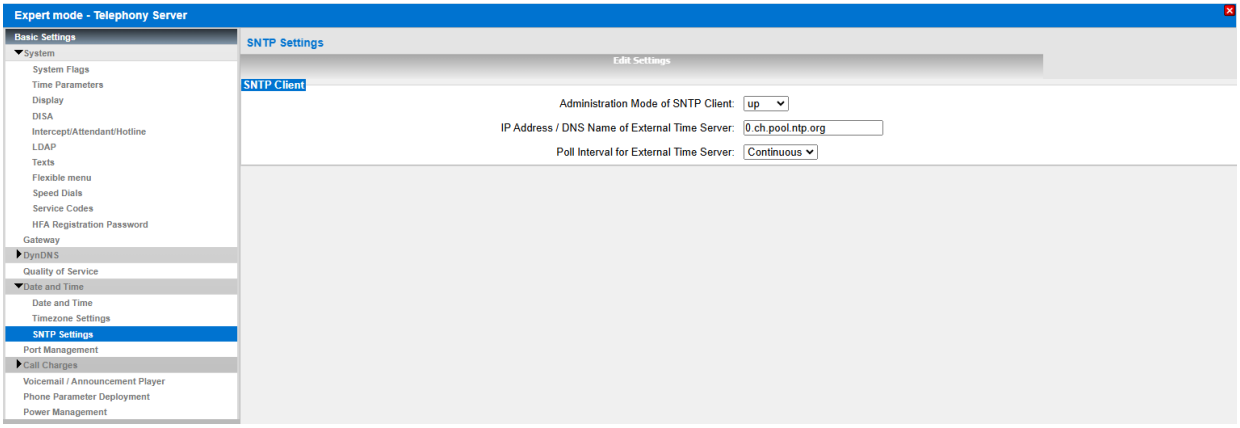
After this Setting PBX needs to be restarted!

4.1.4 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.

Administration Mode of SNTP Client	= up
IP Address / DNS name of External Time Server	= IP Address / DNS name of the Costumer Time Server
Poll Interval for External Time Server	= Continuous

Expert mode > Basic Settings > Date and Time > SNTP Settings



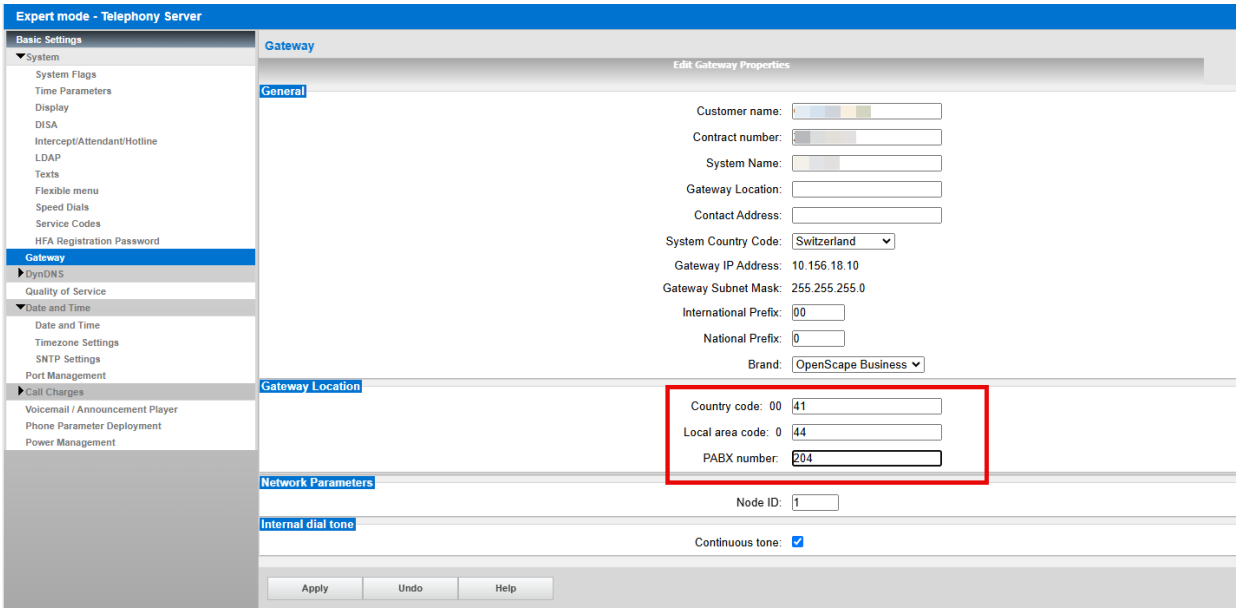
4.1.5 Gateway Location (just to check)

Open Gateway-Location, and enter he follow Parameters

Country Code = 41
Loca area code = 44
PABX Number = 204 (System number without DDI Range)

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

Open the Expert mode > Basic Settings > System > Gateway



4.1.6 Seizure code setting

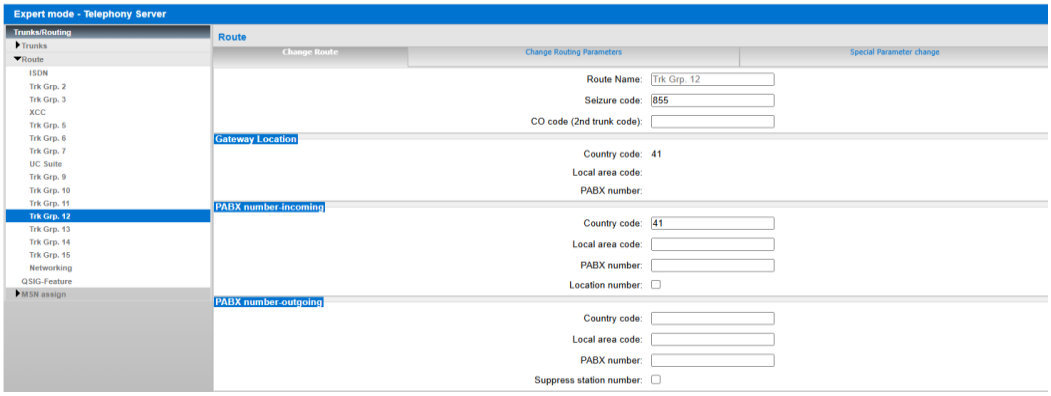
Per default in Trk.Grp. 1, Seizure Code „0“ is configured, change this with another Seizure Code eg. “855”, after that, you can add the Seizure Code “0” in Trk.Grp 12.

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

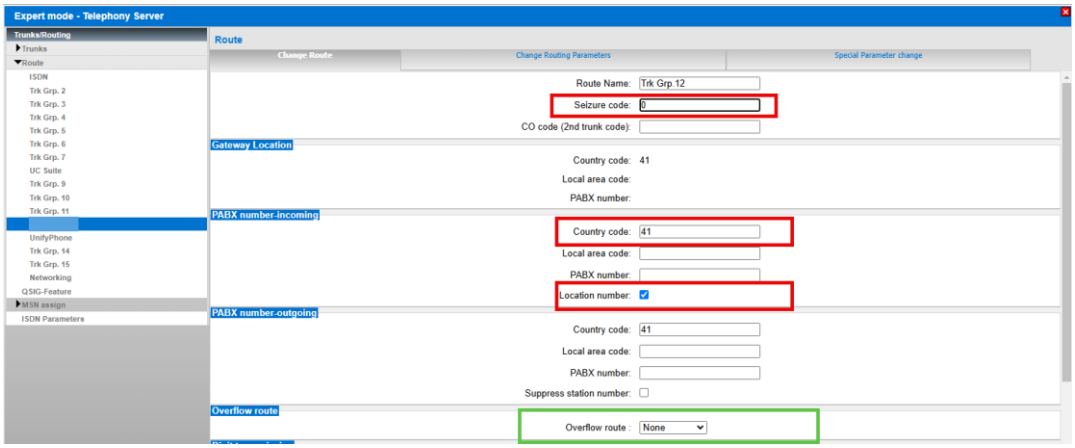
- Number incoming/outgoing
- Caller list
- Redial

Default Setting

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



Route Name, Seizure Code and Location number



Changed Setting

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.

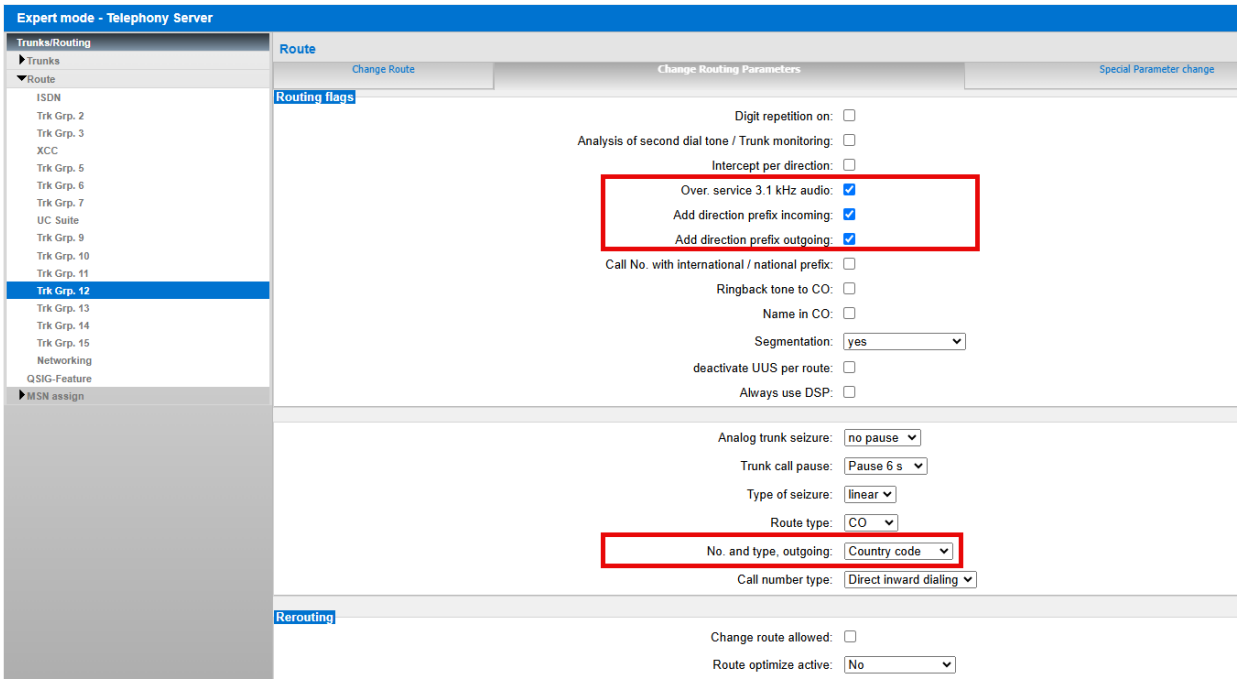
"Rerouting active" activated -> in case of call forwarding, rerouting is performed in the exchange. The system loses further control of the call.

Routings Parameters (just to check after Wizard is completed)

Change Routing Parameters:

Over. service 3.1KHz audio: = activated
Add direction prefix incoming: = activated
Add direction prefix outgoing: = activated
Segmentation: = yes
No. and type, outgoing: = Country code
Route optimize = No

Expert mode > Telephony Server > Trunks/Routing > Change Routing Parameters

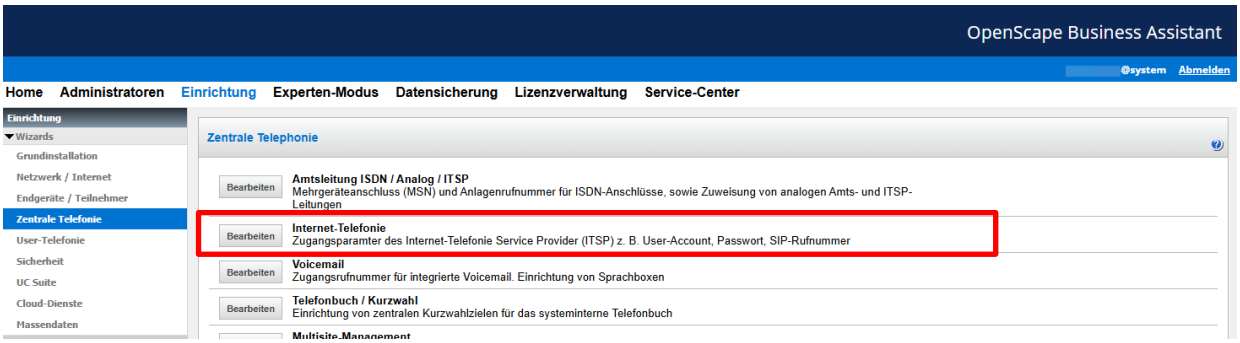


5 Configure the Swisscom Enterprise-SIP Trunk

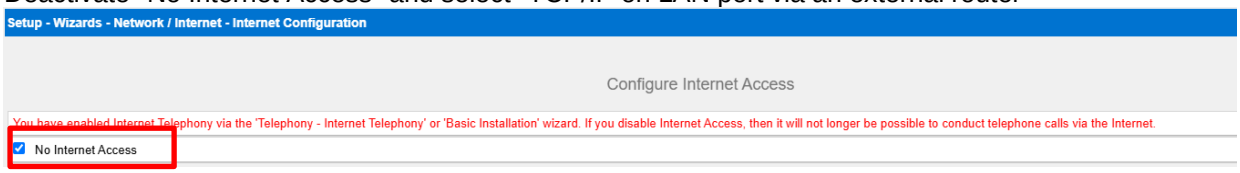
5.1 Internet-configuration

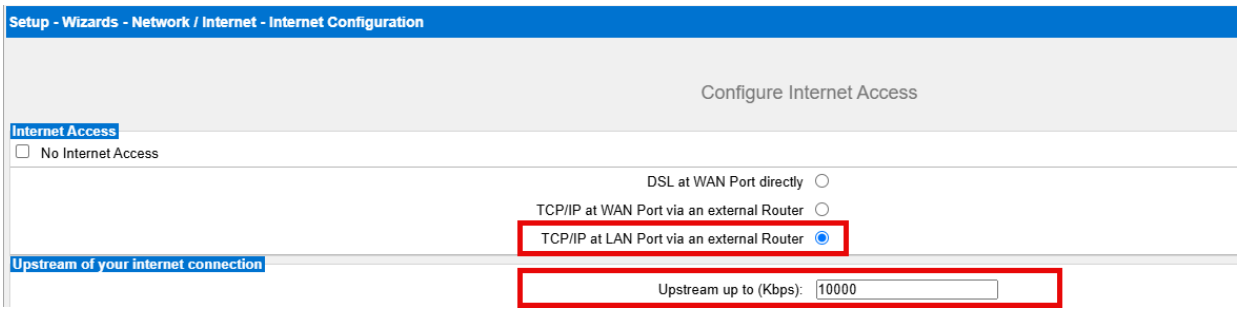
Configure Internet access for your telephone system at

Setup > Network / Internet > Internet Configuration



Deactivate "No Internet Access" and select "TCP/IP on LAN port via an external router"

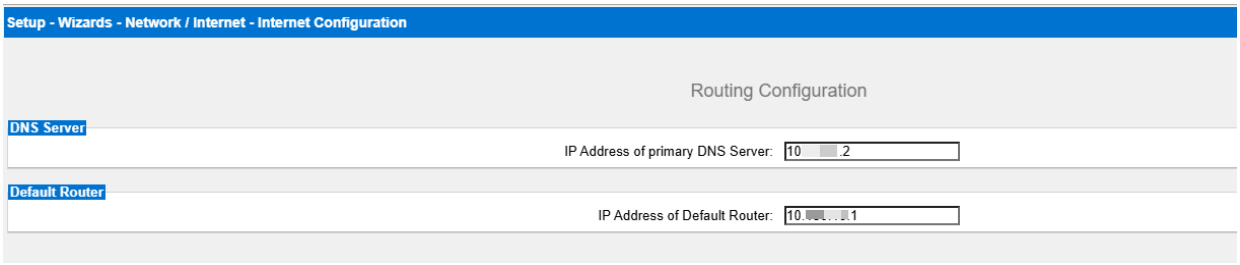


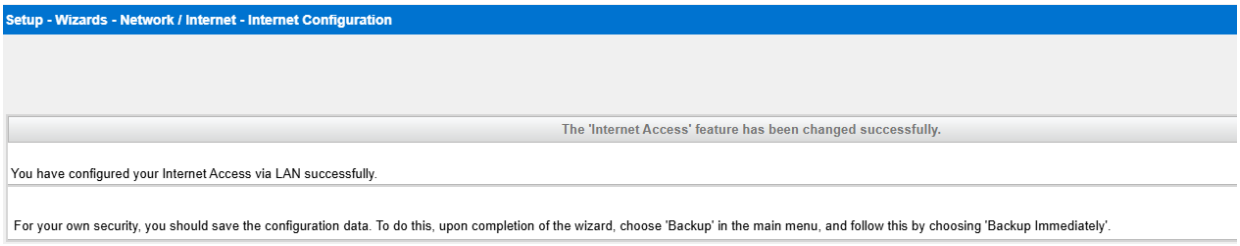


5.2 Enter customer data under Routing Configuration

IP Address of the preferred DNS-Server: DNS entry of the customer network
IP Address default Router: IP address Default Router from customer

Setup > Network / Internet > Internet Configuration





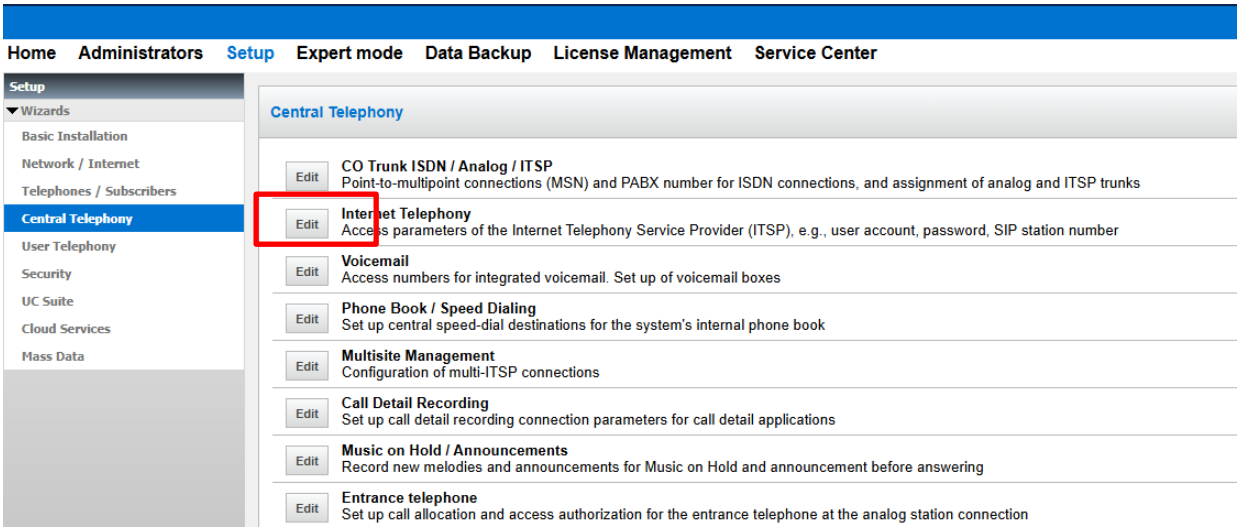
Finish

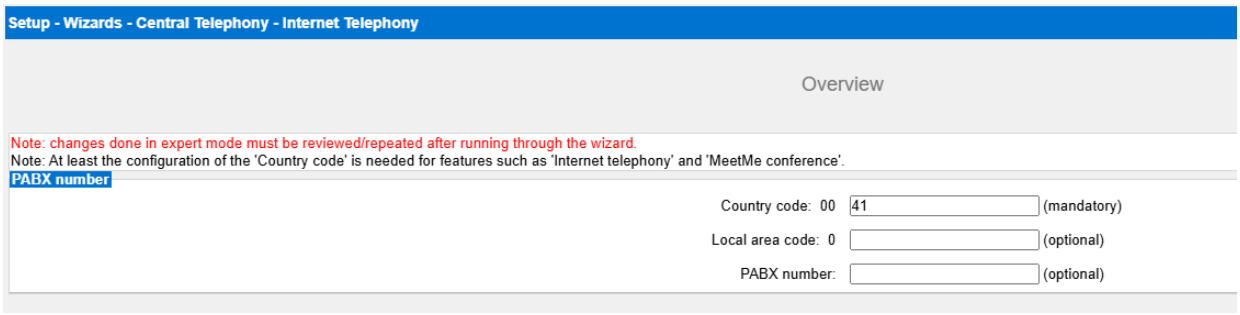
5.3 Benefits of this configuration

- SSDP / RSP access for Unify
- Software update of the system is possible via internet

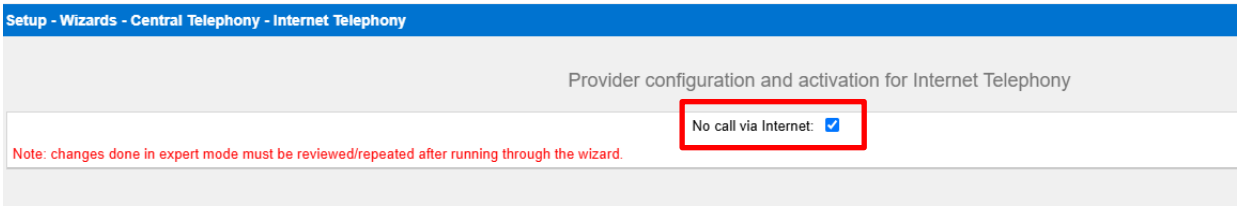
6 Add the ITSP Provider

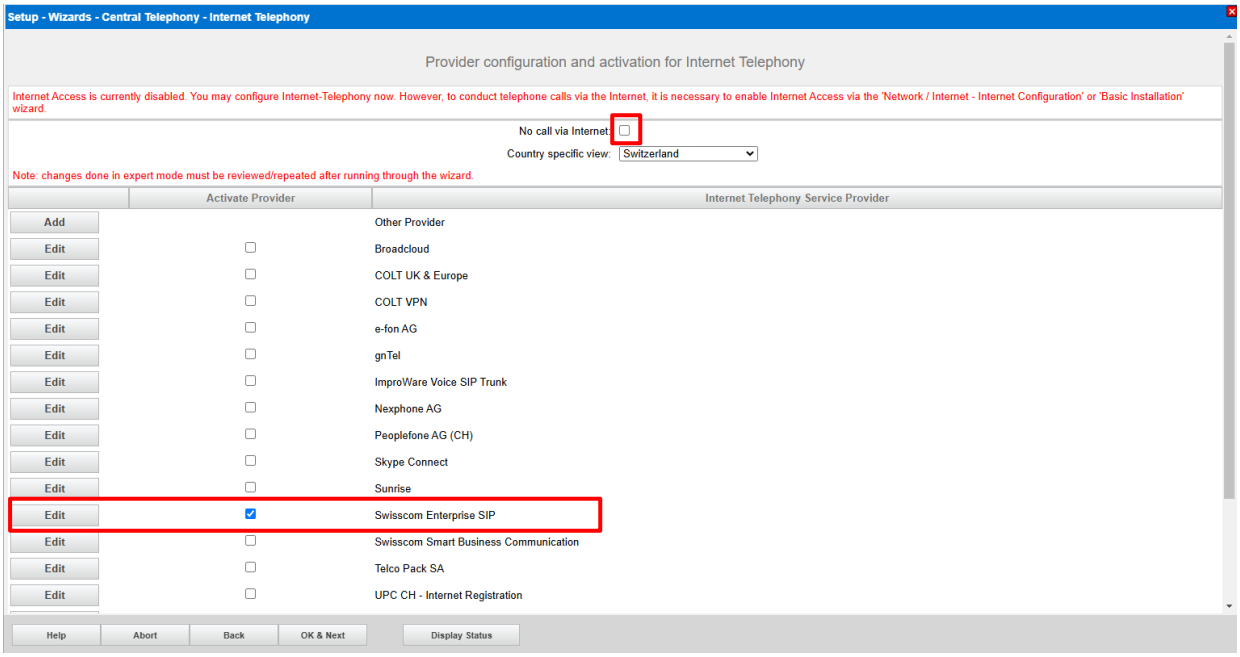
Setup > Central Telephony > Internet Telephony





Disable No call via Internet





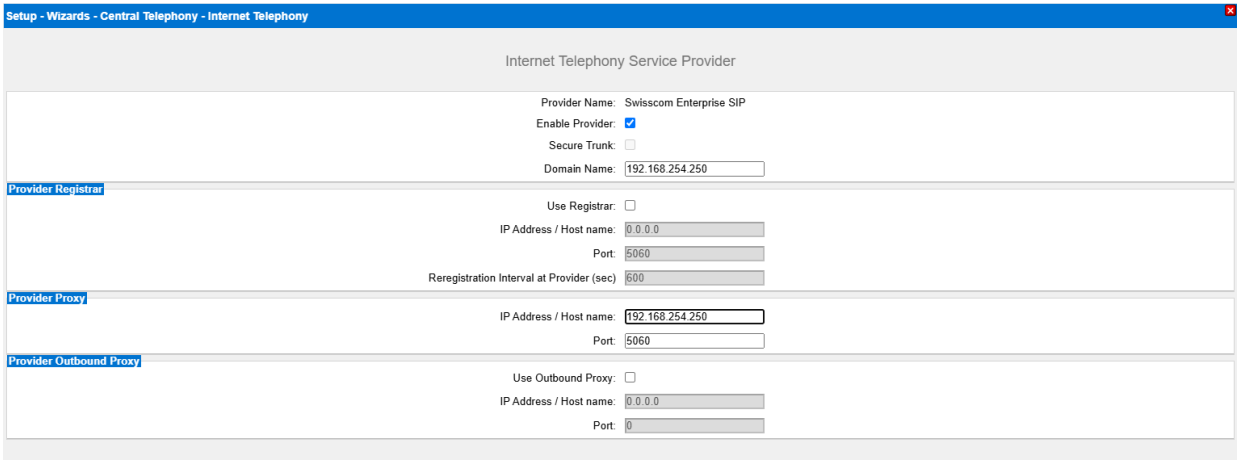
Select the profile Swisscom Enterprise SIP and Edit

7 Activate the Swisscom Enterprise SIP profile

Enter the Data from Swisscom E-SIP Portal.

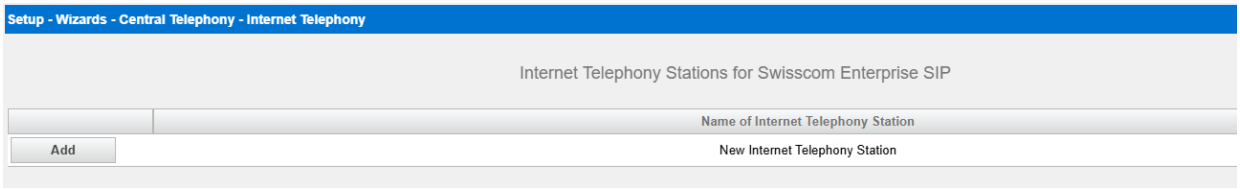
Enable Provider **= YES**
Domain Name **= IP Address SBC**

Provider Proxy **= IP Address SBC**
Port **= SIP Port 5060**



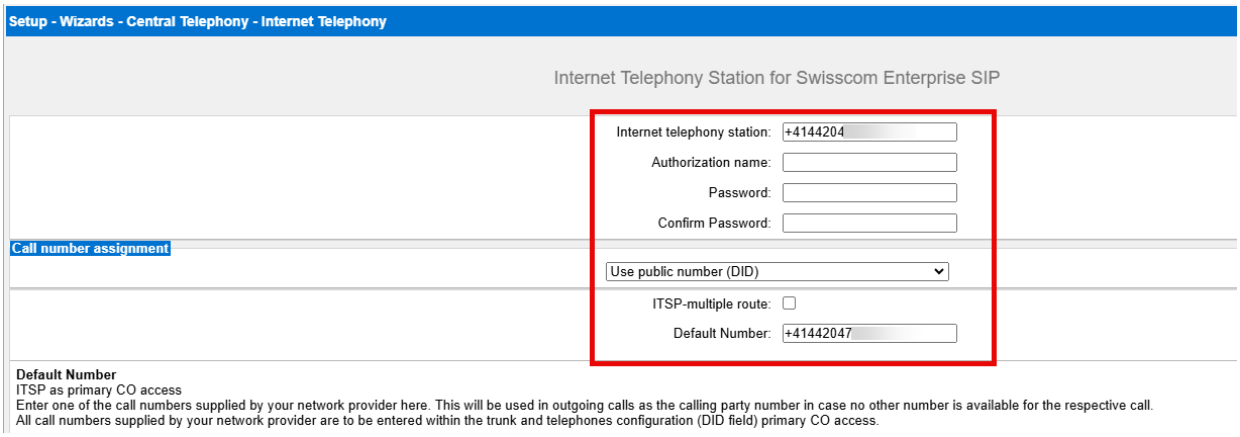
Ok & Next

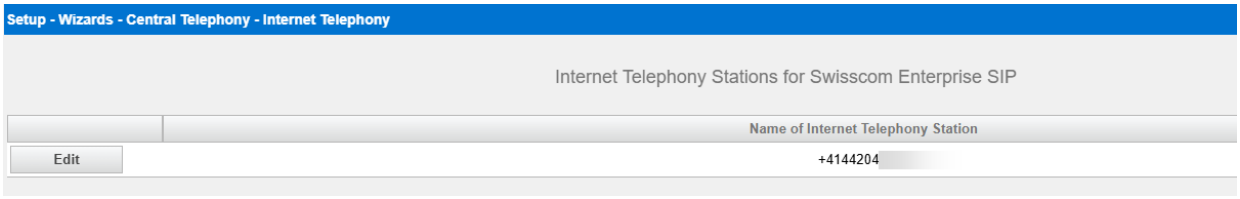
Now insert a new Internet Telephony station, click Add



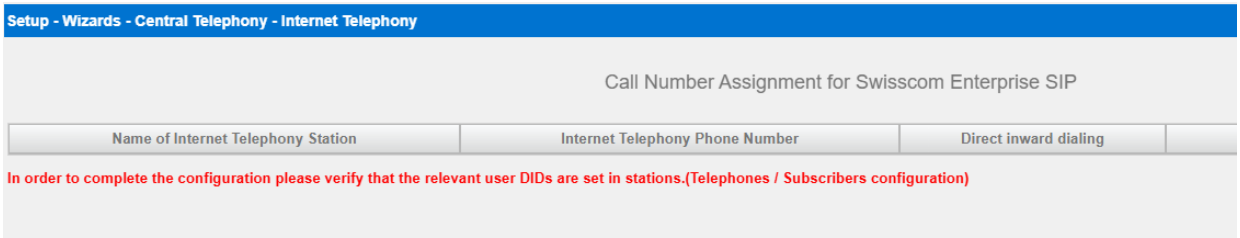
Now enter the information received from the provider

Internet telephony station:	**can be any name or number**
Authorization name:	empty
Password:	empty
Default Number:	**Master Number in E164 GNF format (required for PAI)**

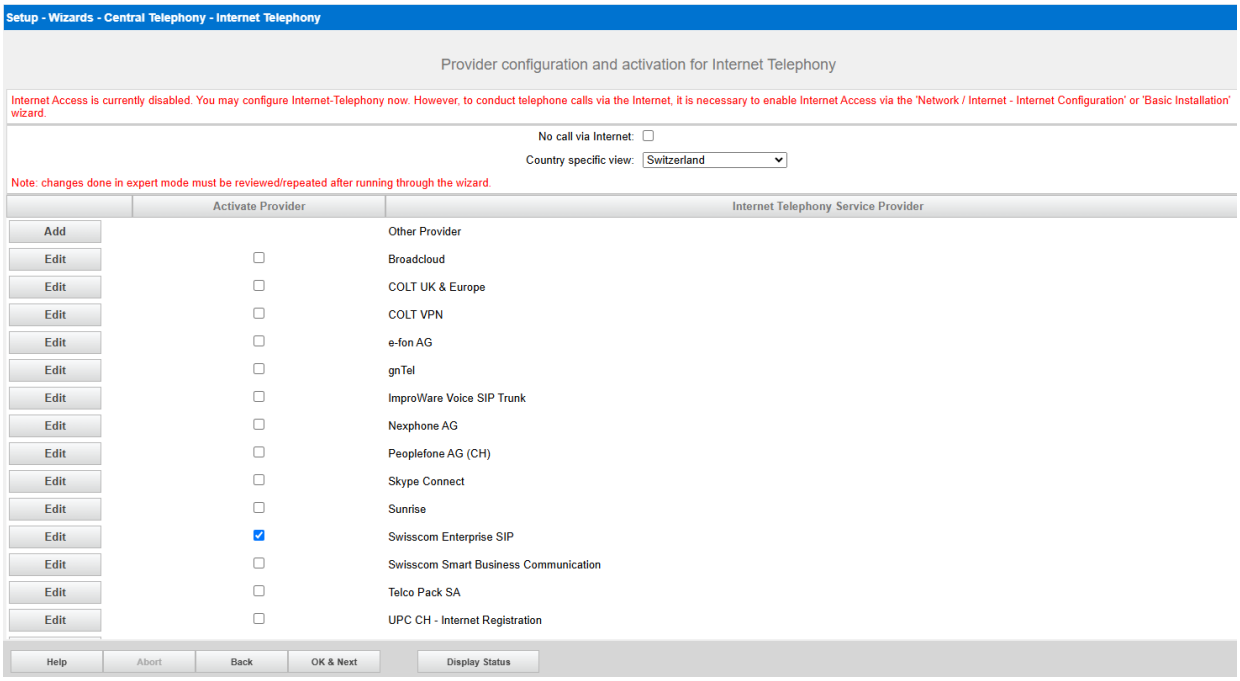




Ok & Next



Ok & Next

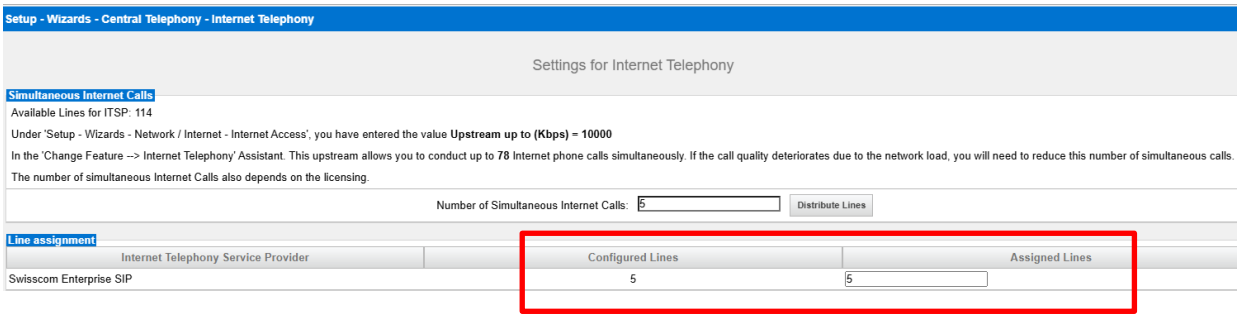

Setup - Wizards - Central Telephony - Internet Telephony
Provider configuration and activation for Internet Telephony
Internet Access is currently disabled. You may configure Internet-Telephony now. However, to conduct telephone calls via the Internet, it is necessary to enable Internet Access via the 'Network / Internet - Internet Configuration' or 'Basic Installation' wizard.
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
Broadcloud
COLT UK & Europe
COLT VPN
e-fon AG
gnTel
ImproWare Voice SIP Trunk
Nexphone AG
Peoplefone AG (CH)
Skype Connect
Sunrise
Swisscom Enterprise SIP
Swisscom Smart Business Communication
Telco Pack SA
UPC CH - Internet Registration
Help
Abort
Back
OK & Next
Display Status

Ok & Next

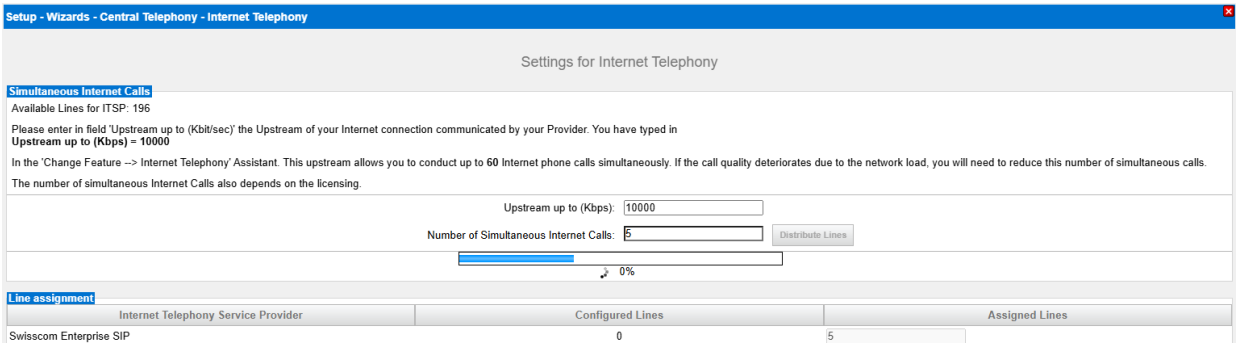
8 Internet telephony settings (simultaneous internet phone calls)

Enter the upstream

This corresponds to the licensed SIP trunks channels (B-channels).



Under Number of Simultaneous Internet calls and under Assigned Lines, enter the desired number of B-channels



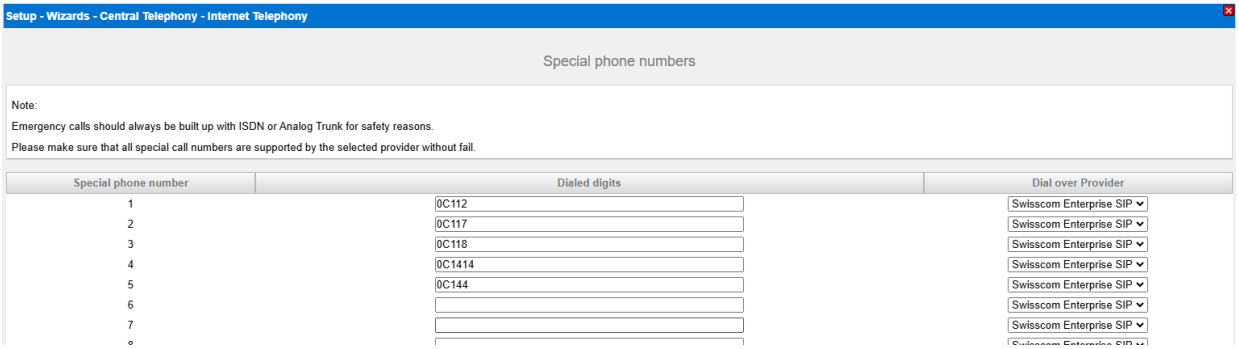
Push OK & Next

The "Number of Simultaneous Internet calls (B-channels) will be assigned to the direction 12 (1. ITSP direction).

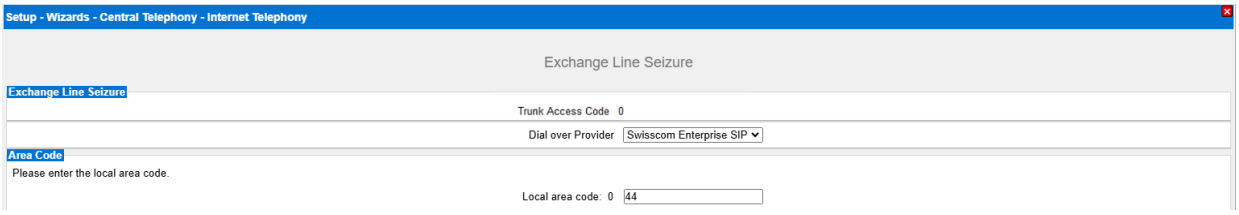
8.1 Special phone numbers

Enter the special phone numbers. Select the provider you want to choose from.

- Rega
- Police
- Fire Brigade
- Emergency Number
- etc.
-



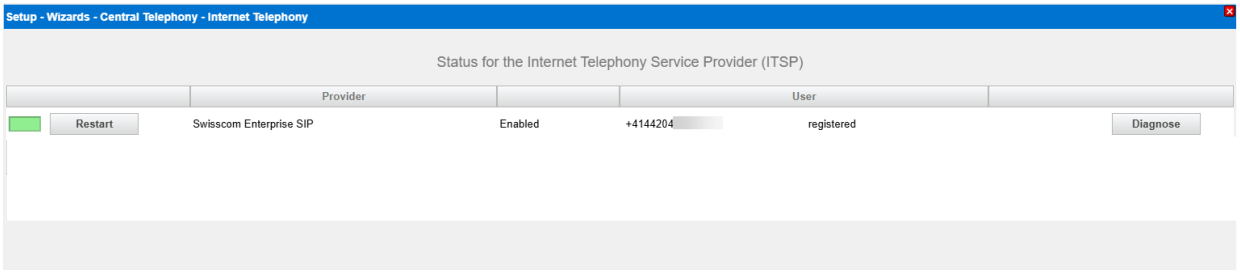
Push OK & Next



The Configuration with Internet Telephony (Wizard) are completed The following Setting must be done in Expert mode.

8.2 ITSP Status for the Internet-Telephony Service Provider (ITSP)

Open Service Center > Diagnostic Status > ITSP Status



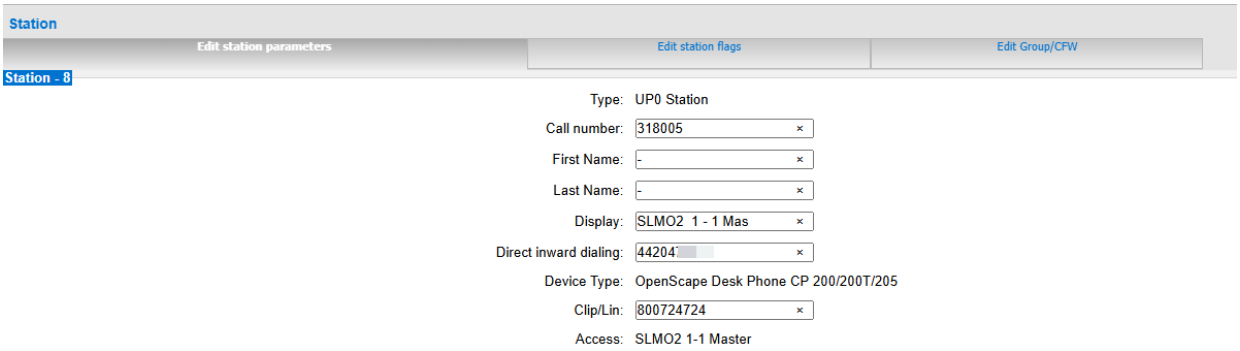
If the Provider is not active (registered) you can start first analyses by click on the Button "Diagnose". (Summary of the config and Status messages is shown.)

9 Station creating

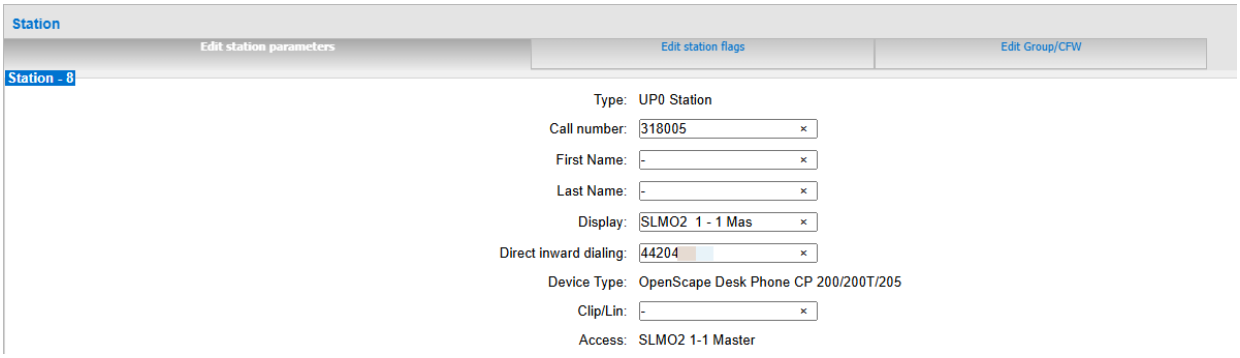
Call number: internal Number of the station
DID: Public Number over which the user can be reached from Public Network
It's recommended to add DID with 9 digits.

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



User without Clip No Screening => Clip Setting



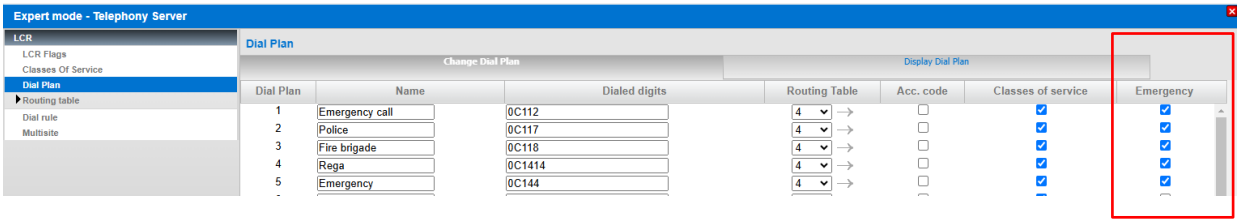
10 LCR Least Cost Routing

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

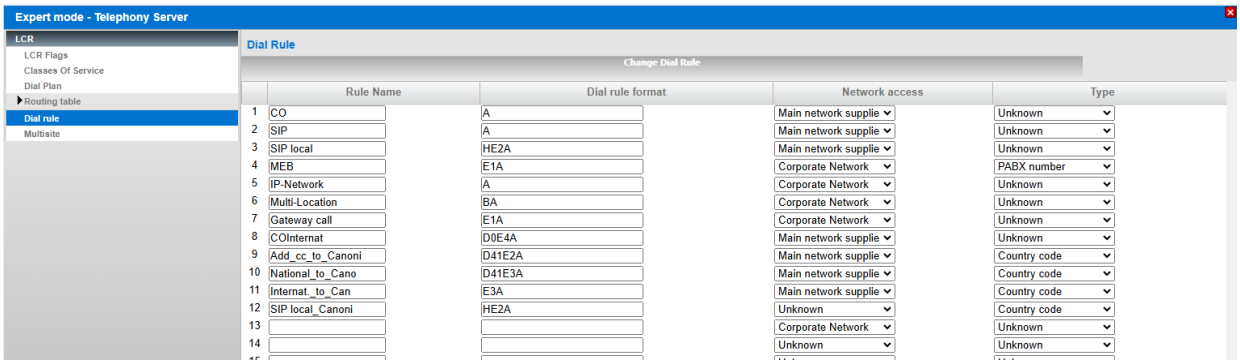
You can check the Dial plan afterwards:

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).

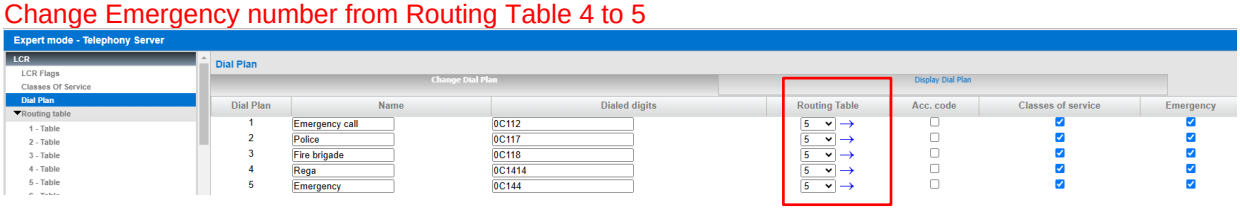


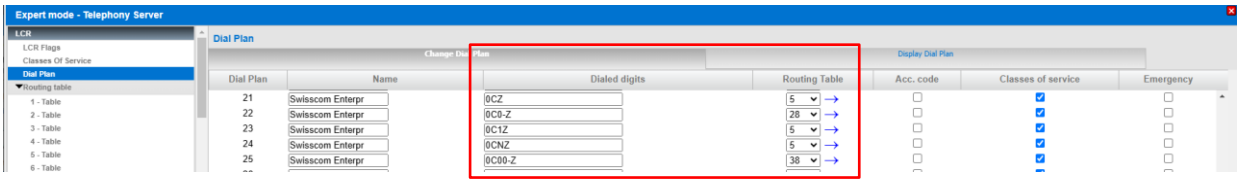
Expert mode > Telephony Server > LCR > Dial rule



Expert mode > Telephony Server > LCR > Dial Plan

Change Emergency number from Routing Table 4 to 5

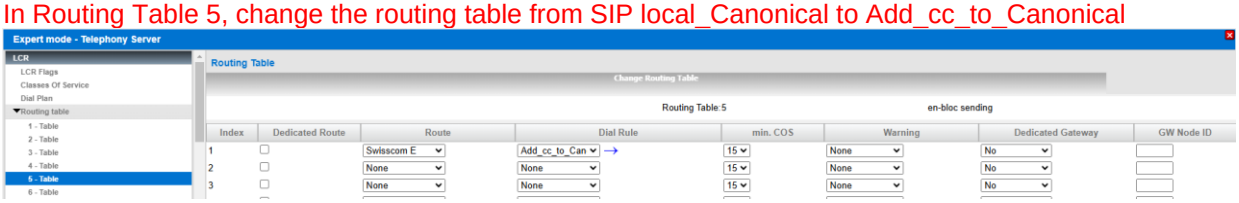




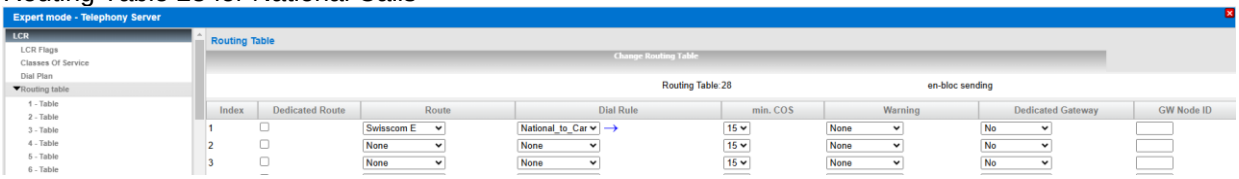
Expert mode > Telephony Server > LCR > Routing Table

Routing Table 5 for Local Calls

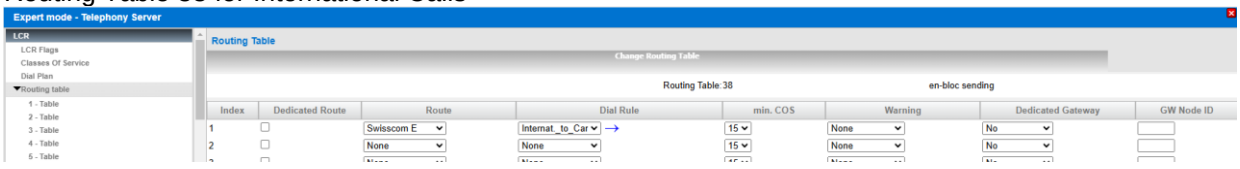
In Routing Table 5, change the routing table from SIP local_Canonical to Add_cc_to_Canonical



Routing Table 28 for National Calls



Routing Table 38 for International Calls



11 SIP Trunk requirements and recommendations

11.1 DTMF

11.1.1 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.

11.1.2 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.

11.2 Best Practices

11.2.1 Fax over IP Recommendations & Settings

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

Swisscom Recommendation:
https://www.swisscom.ch/content/dam/swisscom/en/about/company/network/all-ip/documents/fax-over-ip-recommendations-for-swisscom-enterprise-customers.pdf.res/fax-over-ip-recommendations-for-swisscom-enterprise-customers.pdf

Swisscom White Paper Fax over IP:
https://www.swisscom.ch/content/dam/swisscom/de/about/unternehmen/netz/all-ip/sonderthemen/documents/whitepaper-fax-over-ip-de.pdf.res/whitepaper-fax-over-ip-en.pdf