OpenScape Business V2 / V3

Tutorial

Internet Telephony Configuration Guide

Version 4.7

Table of Contents

History of Change

Date	Version	Changes
2013-07-15	1.0	Released for OpenScape Business V1
2014-05-07	1.1	Update for ITSP connectivity based on DID (V1R3) Configuration steps in Chapter 3 updated Appendix 4.5 and 4.6 created
2014-12-02	1.2	Update for MEX connectivity. Appendix 4.7 created. Update for Restart Button.
2015-04-17	2.0	Released for OpenScape Business V2 New chapter 4.8 for Multisite Configuration New chapter 4.9 for SPE activation for ITSP over TLS
2015-04-17	2.1	Rebranding and support of non T.38 fax for OpenScape Business UC
2016-09-07	2.2	Correction in paragraph 4.5 (Additional Notes).
2017-07-07	3.0	Update for V2R3
2018-02-28	4.0	Updated version for V2R4 Updated chapter 2 about Internet connection Add chapter "Know restriction Move MEX description to separate document For comments on this document please contact ulrich.abel@atos.net
2018-03-06	4.1	Update restrictions
2018-11-30	4.2	Update for V2R6
2019-01-11	4.3	Add hint about deactivation of RFC2198
2019-03-07	4.4	Add hint about early media handling
2019-10-25	4.5	Update for V2R7.1: change in Internet Telephony Station configuration
2020-01-31	4.6	Update STUN chapter
2020-03-03	4.7	Update limitations for local ringtone

1. Introduction

This document describes how to set up the OpenScape Business communication system for Internet Telephony via ITSP (Internet Telephony Service Provider) using Web-Based Management (WBM).

The guide covers mostly VoIP trunks with SIP protocol which provide a range of call numbers for business users (direct dialing inward, DDI).

General administration is covered by the respective WBM administrator documentation.

Current technical information on the products, applications and solutions available from Unify can be found under the following link: https://wiki.unify.com

For further ITSP issues, documentation and ITSP Certification Process see:

https://wiki.unify.com/wiki/Collaboration_with_VoIP_Providers

2. OpenScape Business Internet Configuration

An internet connection from your ITSP or from another Internet Service Provider (ISP) is required for Internet Telephony. The DSL bandwidth at the customer site determines the maximum number of concurrent calls (e.g. 128 kbit/s for a G.711 call).

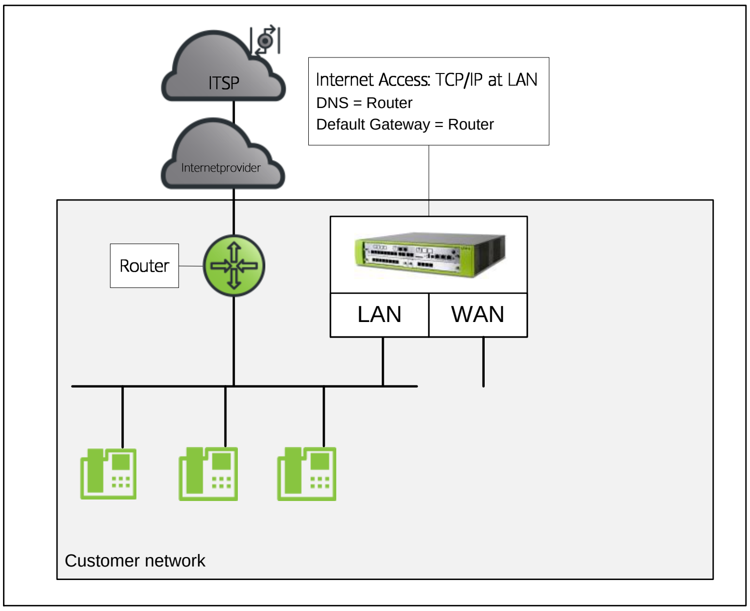
The most common connection scenarios are described in the following chapter.

A detailed description can be found in our Wiki under

https://wiki.unify.com/wiki/OpenScape_Business#Configuration_of_LAN.2FWAN_interface_for_VoIP

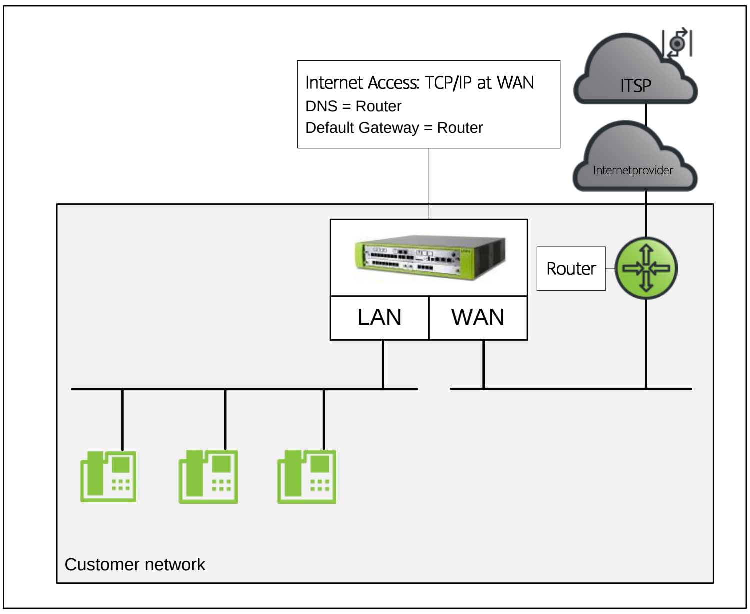
2.1. ISP and ITSP access with external Router at LAN

The most preferred configuration



Configuration 2.1: ISP and ITSP@LAN

2.2. ISP and ITSP access with external Router at WAN



Configuration 2.2: ISP and ITSP@WAN

Restrictions:

- Device@Home is not released in this configuration
- When the WAN interface is configured, all ITSP connections MUST use this interface

2.3. ISP with external Router at LAN and ITSP access at WAN

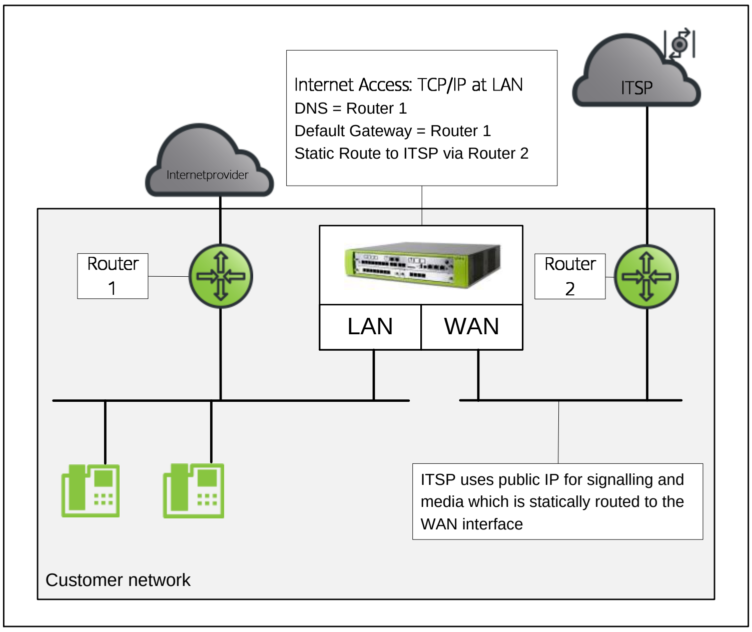
Several ITSPs deliver an access device to their customers and assign a dedicated network to the interface where the OpenScape Business is connected. In such scenarios the OpenScape Business WAN interface is used to connect to the access device. (if the assigned address does not fit in the customers network infrastructure) .

The access device may operate in different modes

a) the access device acts as router (e.g. Versatel Germany)
- static routes for all ITSP related traffic (signalling and media) must be defined
- SIP message content is generated in OpenScape business and sent via the router to the ITSP

b) the access device acts as SBC (e.g. Telefonica Germany)
- SBC is addressed in ITSP profile, no static routes necessary
- all ITSP related traffic (signalling and media) is routed via SBC
- It is the task of the SBC to pass the relevant information to the ITSP and change SIP message content if necessary

2.3.1. ISP@LAN and ITSP@WAN with access device acting as router

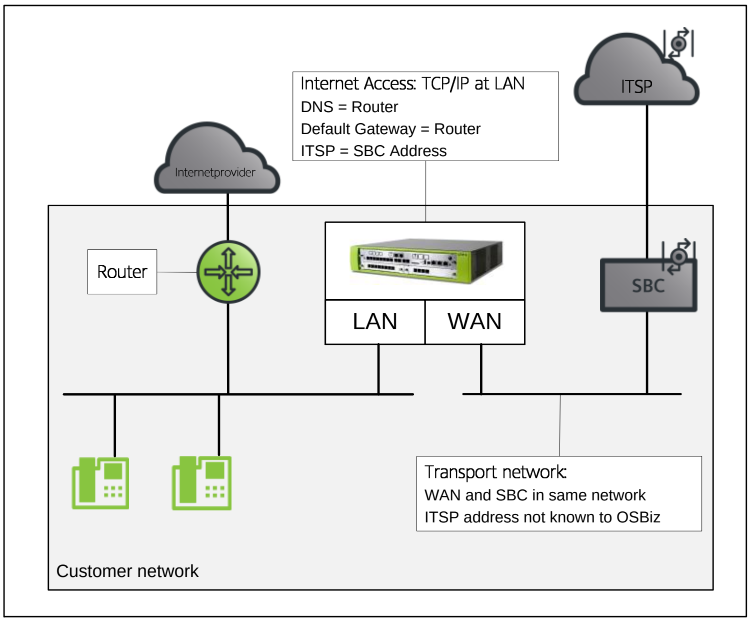


Configuration 2.3a: ISP@LAN and ITSP@WAN with access device acting as router

Restrictions:

- ALL ITSP's MUST use the same interface
- Device@Home is not released in this configuration

2.3.2. ISP@LAN and ITSP@WAN with access device acting as SBC



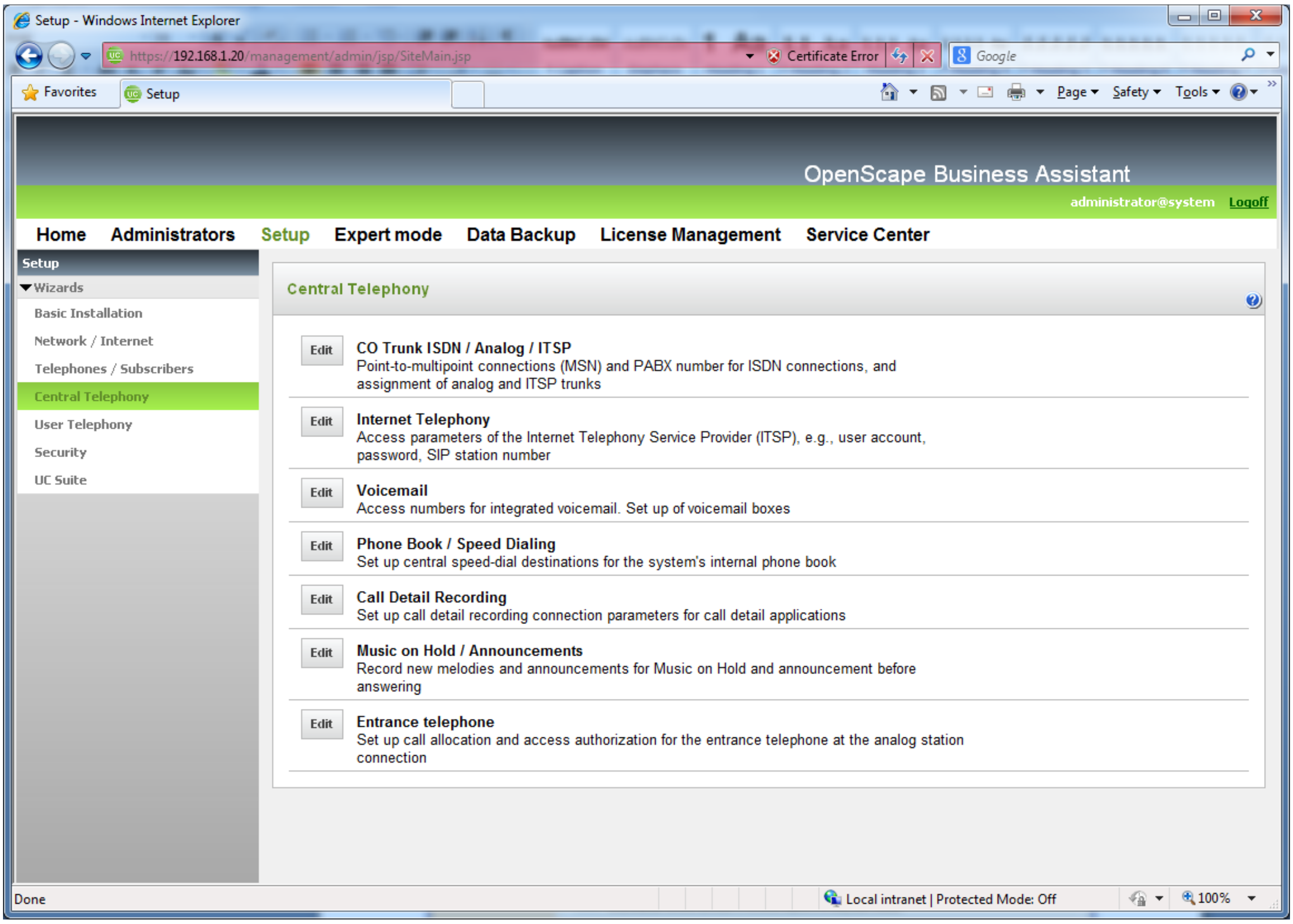
Configuration 2.3b: ISP@LAN and ITSP@WAN with access device acting as SBC

Restrictions:

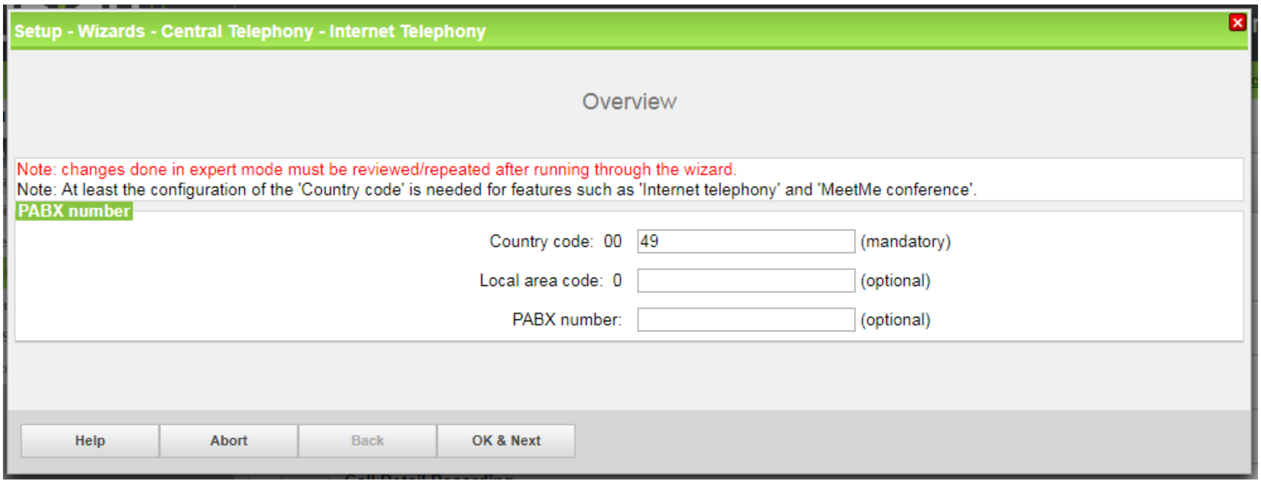
- Due to the restriction that ALL ITSP's MUST use the same interface, this configuration is limited to ONE ITSP only
- Device@Home is not released in this configuration

3. Internet Telephony Configuration

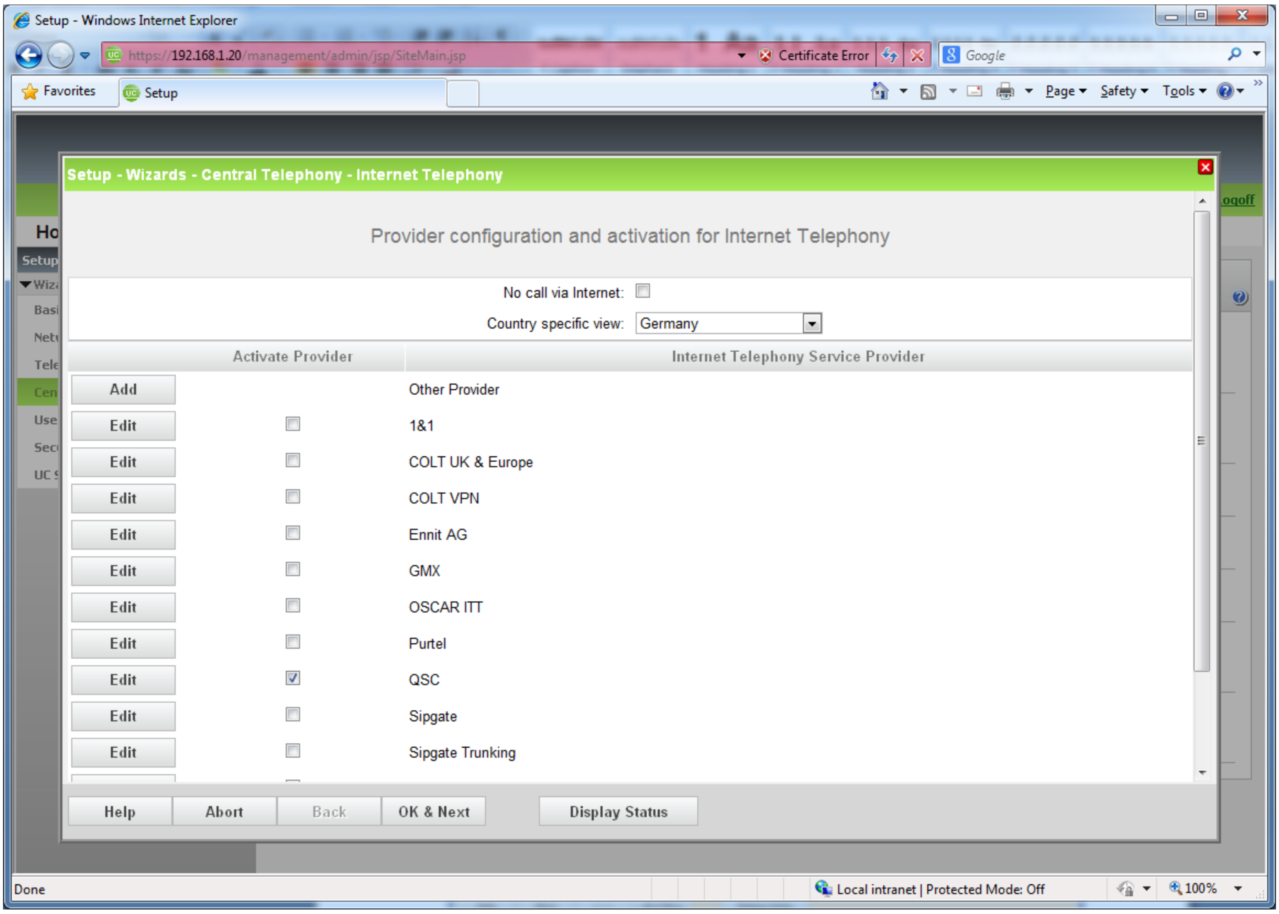
The **Internet Telephony** wizard must be used to activate an Internet Telephony Service Provider (ITSP). You can configure Internet telephony stations for up to eight (8) ITSPs.



1. In the navigation bar, click **Setup**.
2. In the navigation tree, click **Wizards > Central Telephony**.
3. Click **Edit** to start the **Internet Telephony** wizard.
4. Insert the location data (if not previously configured). Please note that country code is mandatory. The number are entered without prefixes and leading "+"



5. Click OK & Next
6. Clear the **No call via Internet** check box. A list of the configured ITSPs is displayed.
   By default the country specific ITSPs are shown. By selecting ‘all countries’ you can see all providers.
   If required, click **Display Status** to check which ITSPs have already been activated and which Internet telephony subscribers have already been configured under each ITSP.

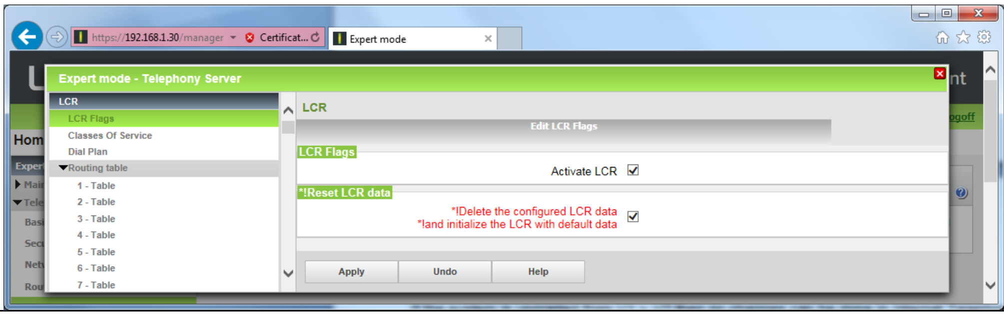


Note:

If the system is upgraded from an older version or older minor releases then ITSP activation/deactivation may not be available in Internet Telephony wizard until a reset to LCR is done.

Already activated ITSPs will continue to work even without the LCR Reset. In order to make any activation/deactivation change in the wizard, the LCR Reset is needed.

Please go to Expert Mode > LCR > LCR Flags and click the “Reset LCR Data” flag as seen below:.



7. Click **Edit** at your ITSP Profile to manage your accounts and ITSP Stations.

   If you have already configured the accounts and ITSP stations and just want to activate your existing profile then click OK & Next, skip the next steps and continue with number 21.

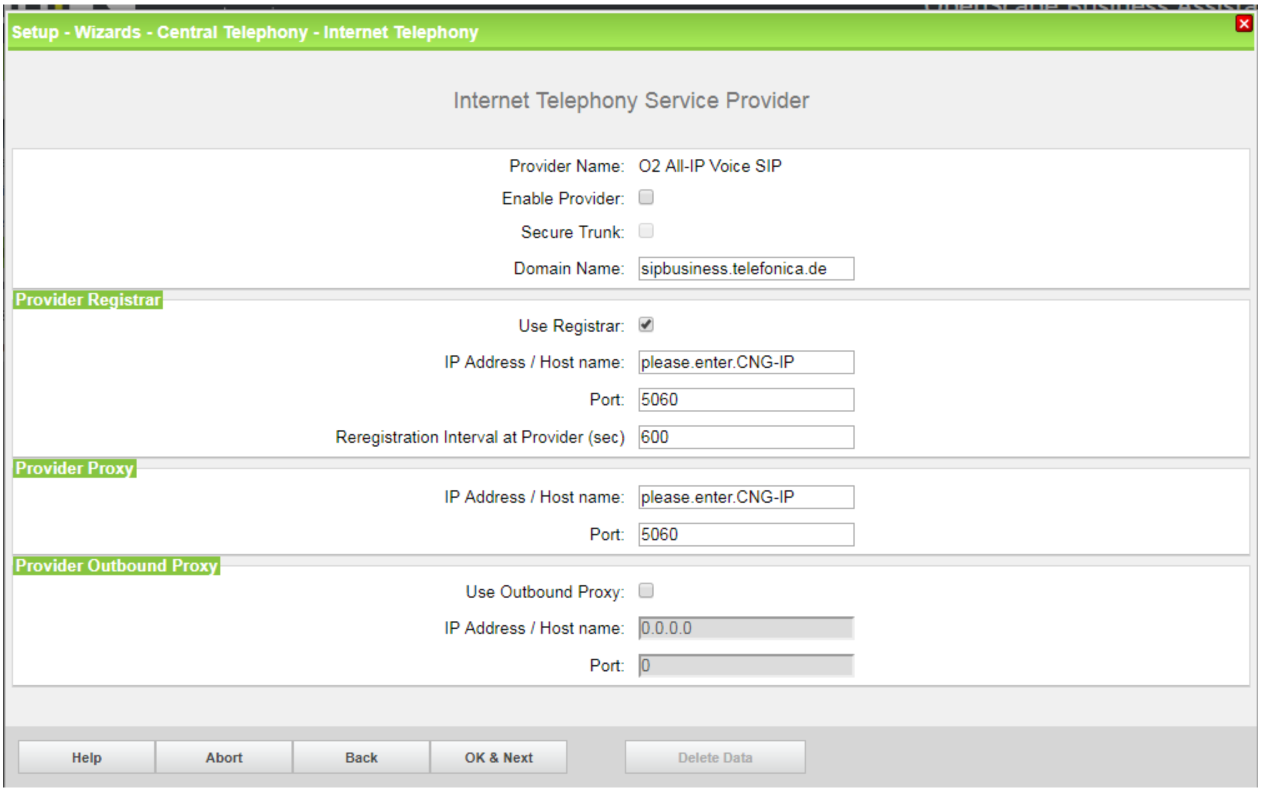
Warning:

If you perform manual changes in Expert mode -> Trunks Routing section deactivation and reactivation of your ITSP may delete/change these manual data. Manual changes have to be performed again after each activation.

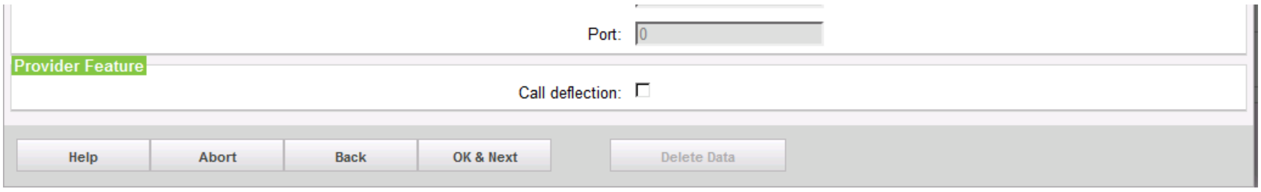
To avoid these configuration please use the “Restart” function for the ITSP. The Restart button is available in the Display Status screen. For more information please refer to step 24 below.

The next page gives an overview of the predefined ITSP address configuration and allows for the activation of a secure trunk, if available for your ITSP.

If your ITSP uses customer specific address values these addresses must be entered here.



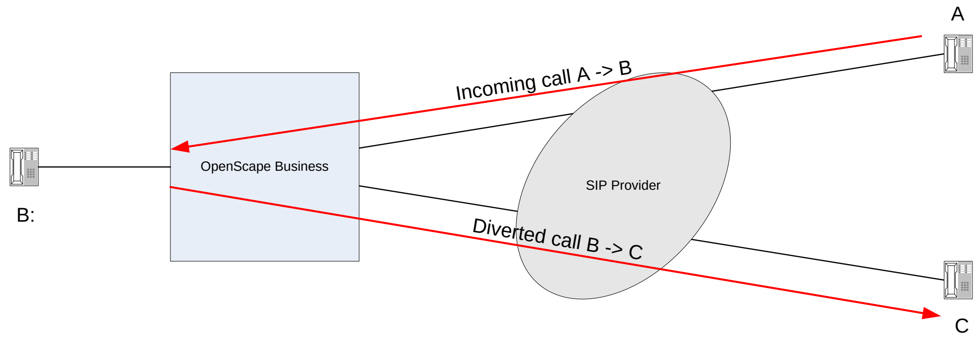
Provider Features / Call deflection: If the Call forwarding by rerouting is supported by your ITSP an additional configuration option is shown on this page:



In default call forwarding is performed by initiating a new call (Forward switching)

+ call management can be used after call forwarding is initiated

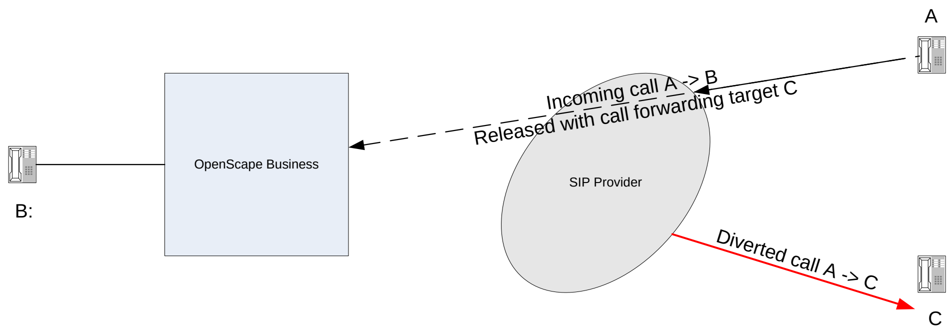
- Two ITSP channels are used when call is active



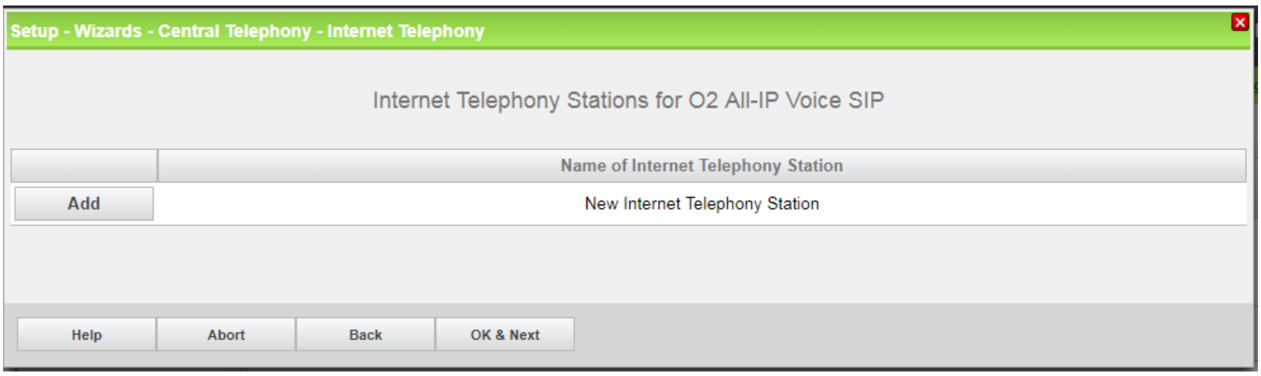
When Call deflection is activated the call is released with a 302 SIP response and call forwarding by rerouting is performed by the ITSP.

+ No ITSP channel used when call is active

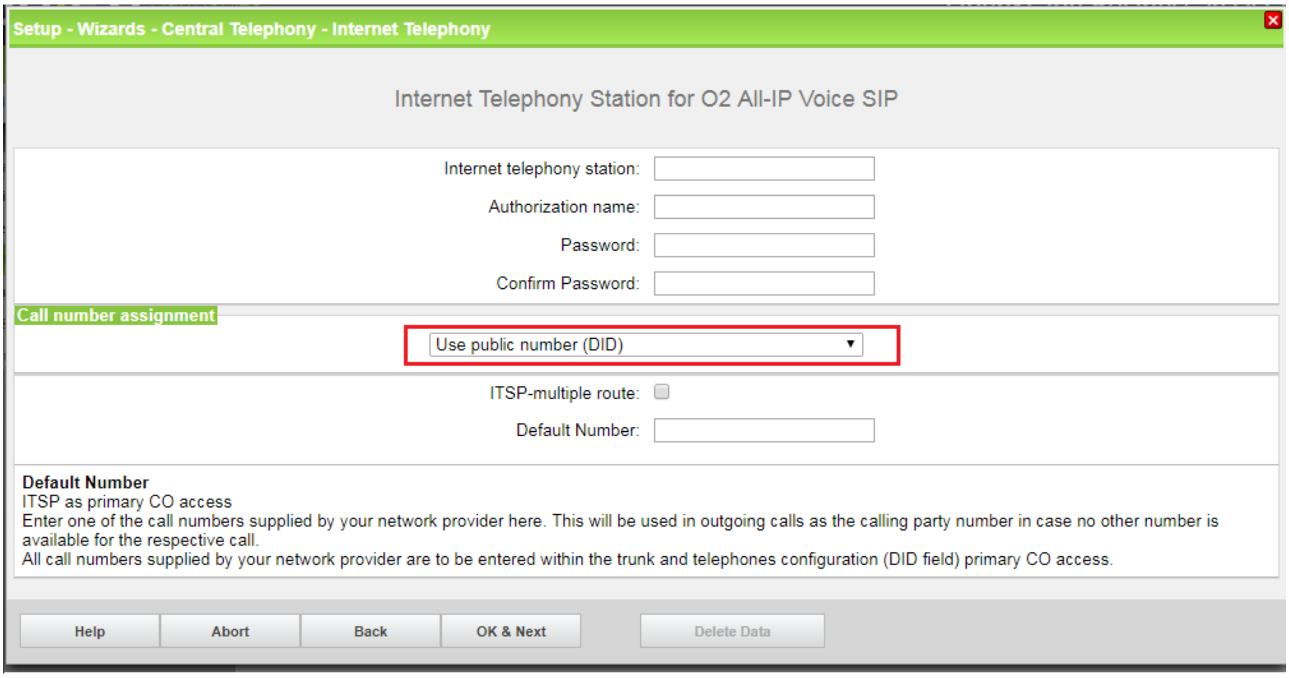
- No control of the call when call deflection is initiated



8. If you are done with this page click OK & Next



9. Click **Add** in this screen



On this screen you must choose the type of Call Number Assignment first:

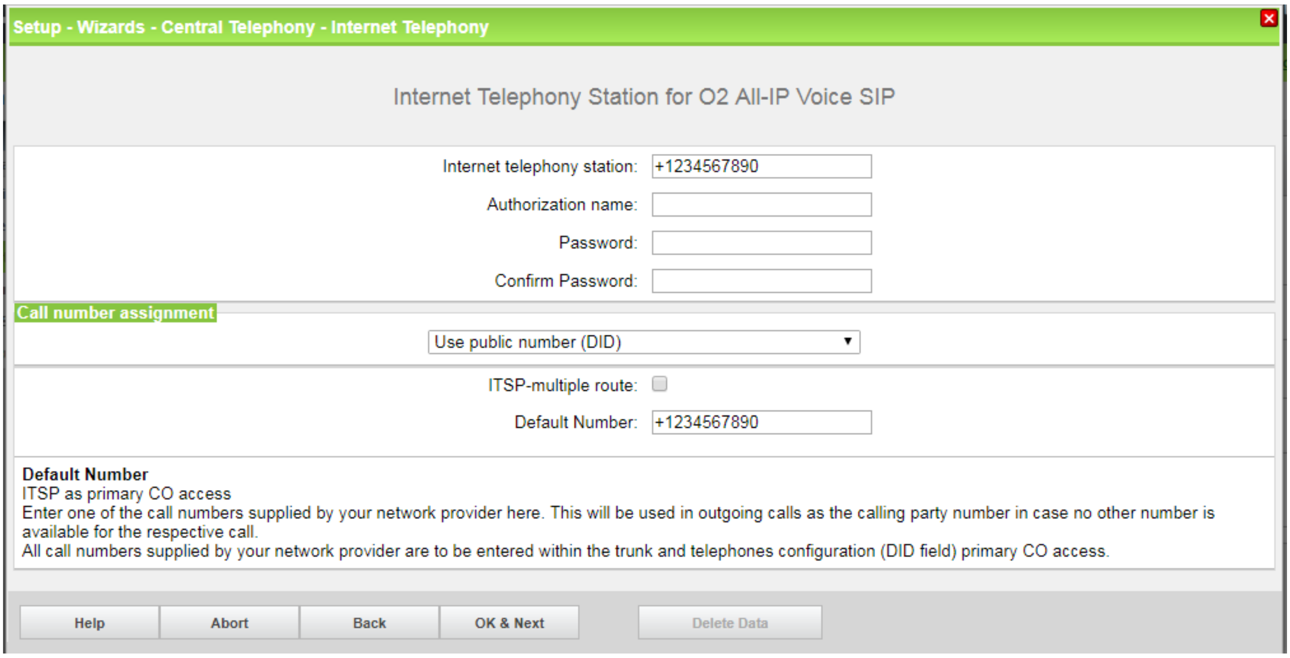
- **<u>Use public number (DID):</u>**
  This is the preferred and most common mode for an ITSP trunk access.
  In this mode all ITSP numbers are based on the station's DID, Location data and Route settings. No mapping is done, just like on ISDN CO interfaces.
- **<u>Use internal number (Callno) / Single entries or Range entry:</u>**
  In this mode all ITSP numbers must be created separately and assigned/mapped to internal call numbers based on the station's Call Number (Callno). This mode cannot be used for central access in networked systems. Each node must have its own ITSP access.

	Remarks: Please note that the selection between the types is available only for the first time you configure an Internet Telephony Station for an ITSP.

Below are the next steps (10-14) for configuring DID mode.

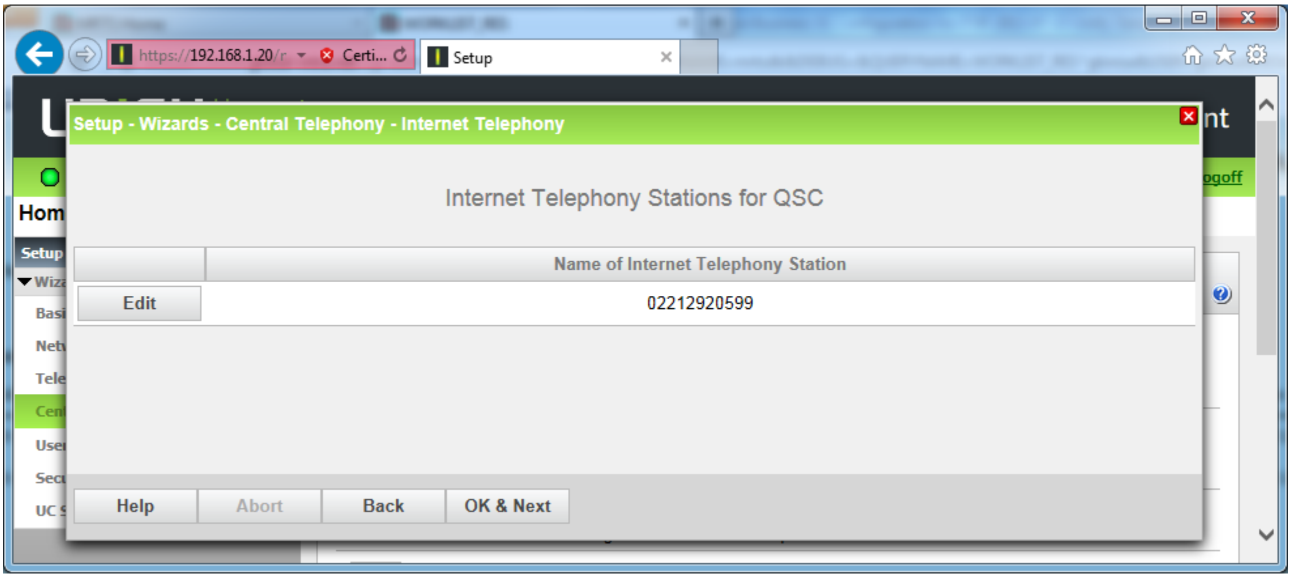
If you want to use the Internal Call number (Callno) mode please skip to step 15.

“Use public number (DID)” Mode

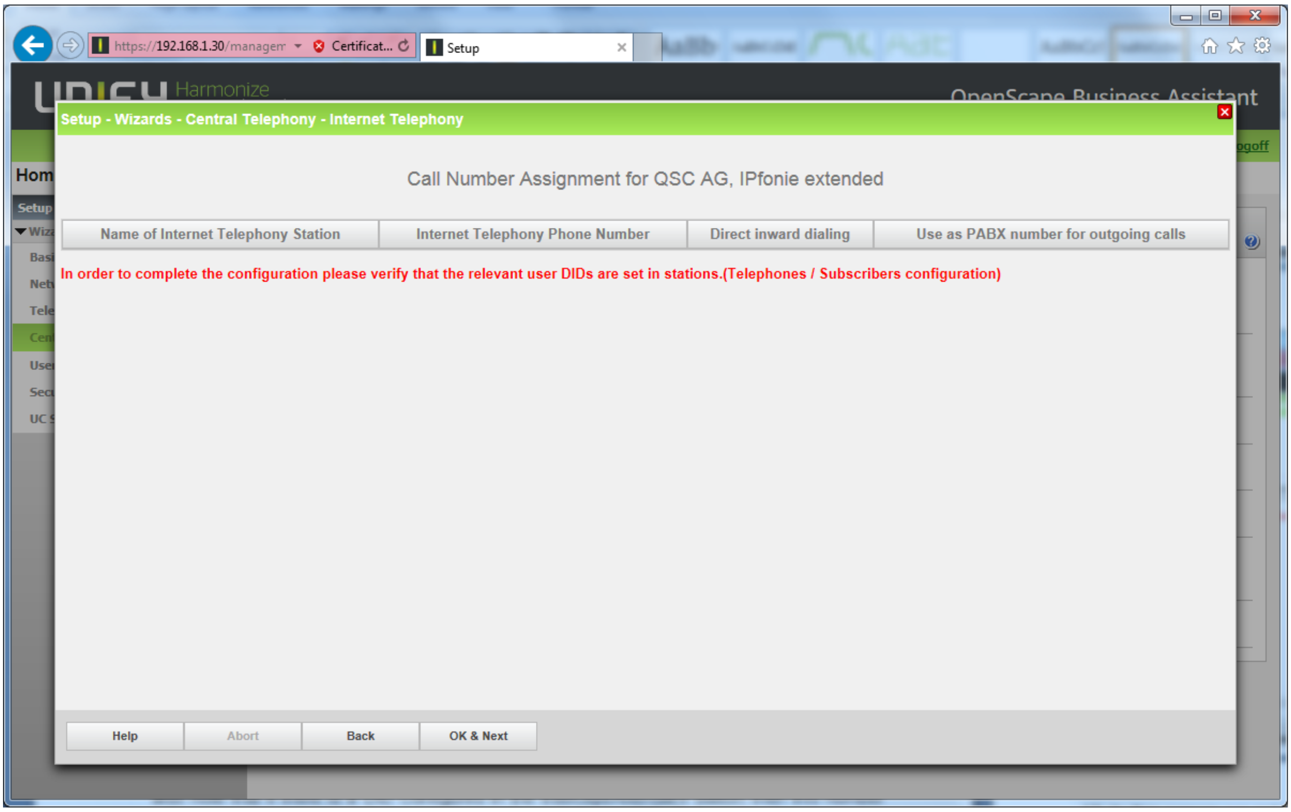


10. The Telephony Station configuration has two options:
    - If your ITSP is using registration:
      enter the name or number provided by your ITSP in **Internet Telephony Station**.
      Enter the **Authorization Name** and **Password** which was provided by your ITSP
    - If the ITSP does not use registration
      enter a name of your choice (e.g. the pilot number of the DDI range or a name) in **Internet Telephony Station**
      Most ITSPs does not use authentication if no registration is used. In this case nothing is entered in Authorization name and Password
      If your ITSP has provided credentials for Authentication (Username/Password) enter them here
11. The ITSP-multiple route flag is needed for configurations where more than one registration / route is needed for configuring the ITSP trunk. More information can be found in a separate guide available in our wiki:
    https://wiki.unify.com/wiki/Collaboration_with_VoIP_Providers#General_Configuration_guides
12. Enter the Default Number in the format requiered by your ITSP
    Please note that this number will be used in the SIP headers exactly as entered here. No change of number format based on location data and route settings will apply.
    In outgoing calls the calling number is presented according to the following rule:

    - use the configured DID or CLIP number
    - if no DID / CLIP is configured use the DID/CLIP of the Intercept/Attendant station
    - if no DID / CLIP for Intercept present, use the **Default Number**



13. Click **OK & Next**

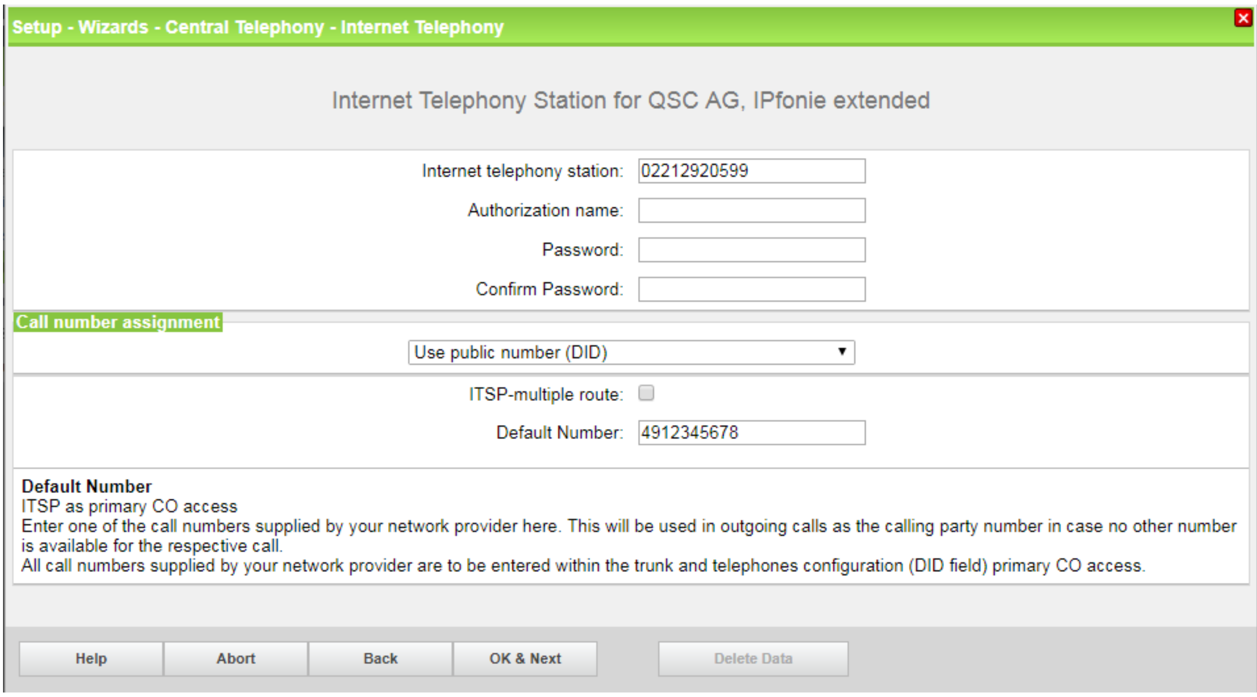


14. Nothing to be entered on this page, in public number DID mode the DID numbers are entered in the station configuration. Click **OK & Next**

Continue in step 21.

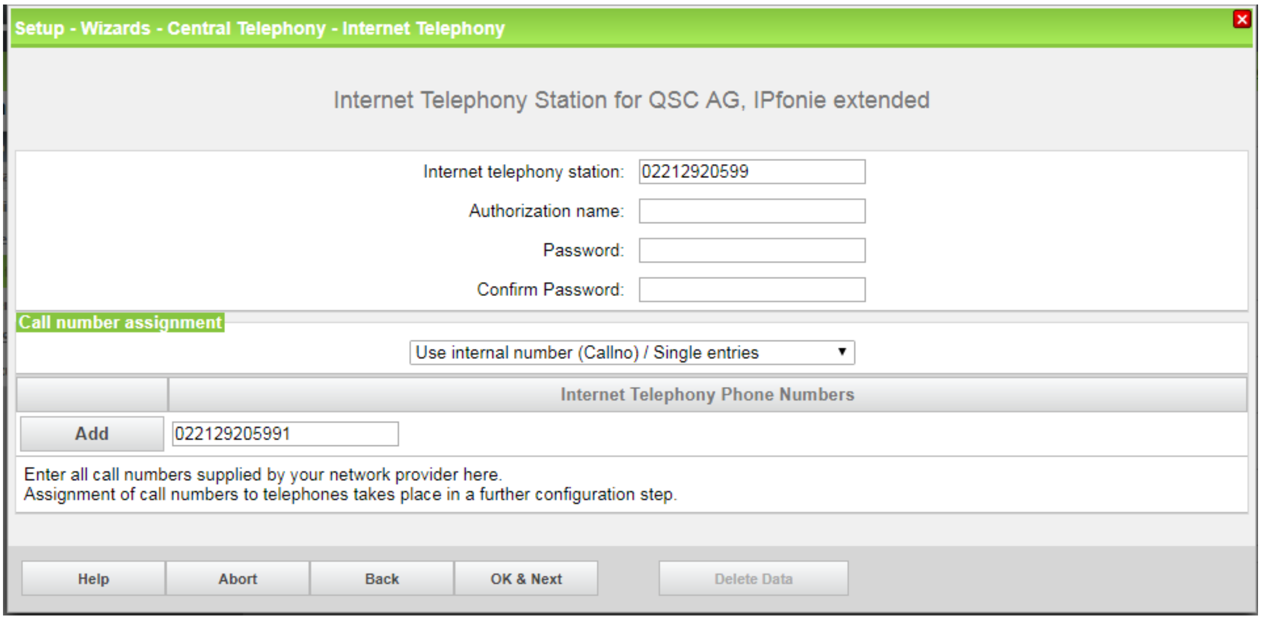
“Use internal number (Callno)” mode

This mode is intended to be used for accounts where each single call number needs a SIP registration. But with some ITSP’s this mode is even used for registration of several call numbers. For Accounts with a range of consecutive call numbers you may select this option but DID mode is prefered due to its simplicity. Also the configuration and allocation of numbers is easier. Usually DID is offered from the ITSPs when there is a range of number

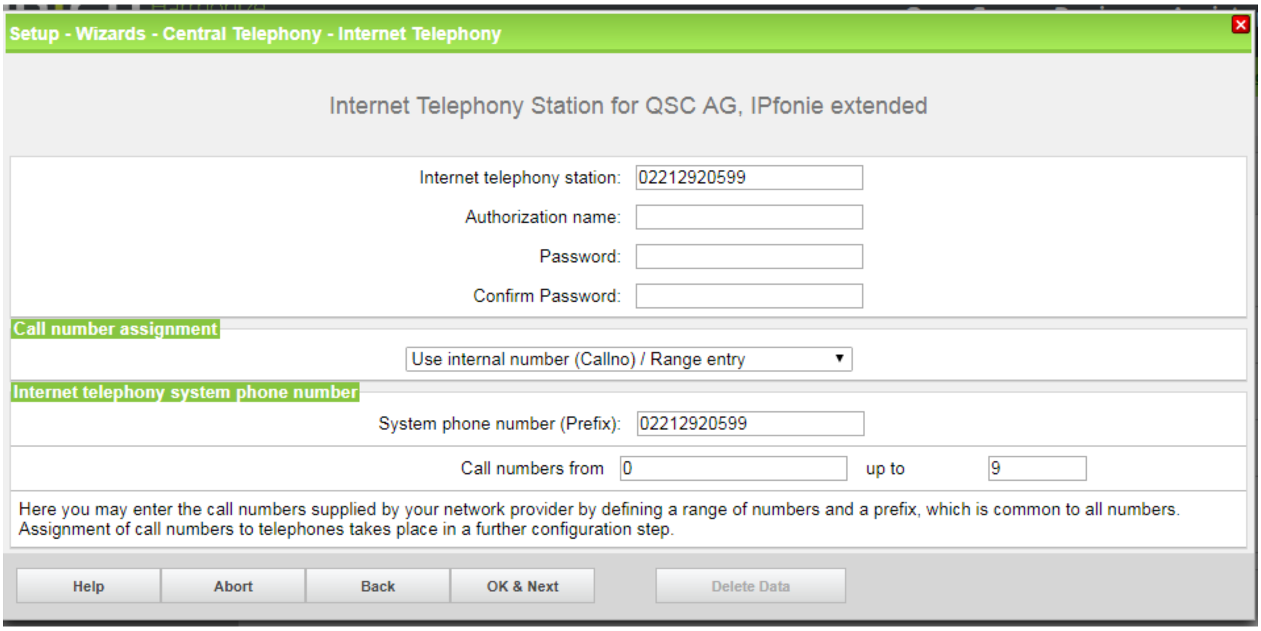


15. Enter the **relevant account (username) or number in Internet Telephony Station**.
16. Enter the **Authorization Name** and **Password** which was given to you for the VoIP account by your provider, if necessary
17. “Use internal number (Callno)” mode has two options:

    a) **single number**: For Accounts with single call numbers select this option and enter the phone number and click **Add** for every phone number you received from your provider.



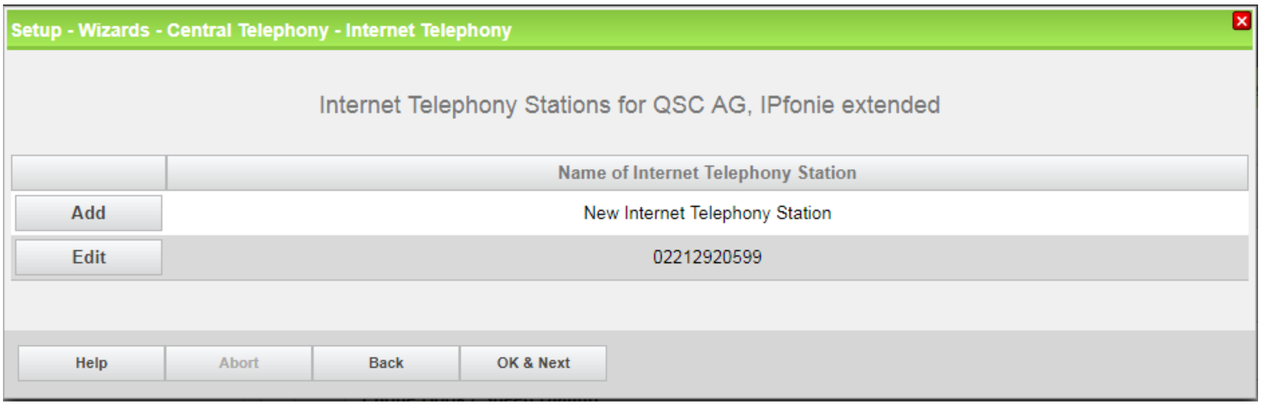
b) For Accounts with a range of consecutive call numbers you may select this option



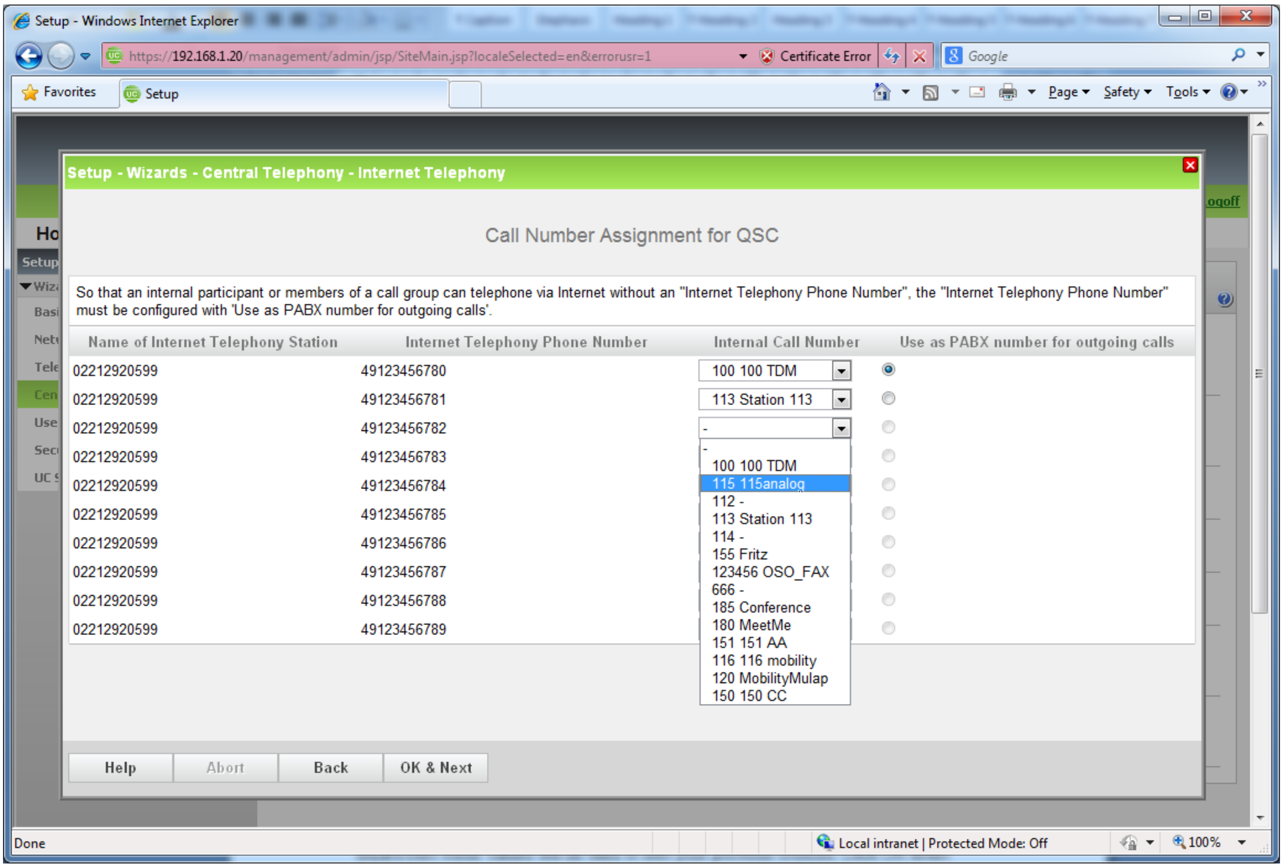
Enter **System phone number** which is the common prefix for all numbers in the range e.g. 02212920599

If the range of numbers is from 022129205990 to 022129205999 then the **Call numbers from** we add 0 and **up to** we add 9.

18. Click **OK & Next**.
An overview of all Internet Telephony Stations is shown.



Click **OK & Next**.



19. Assign one **internal call number** each to all Internet telephony phone numbers.
    For subscribers without Internet telephony phone number one number MUST be selected as **PABX number for outgoing calls**.
20. Click OK & Next.

    An overview of your ITSP providers is shown. Click OK & Next

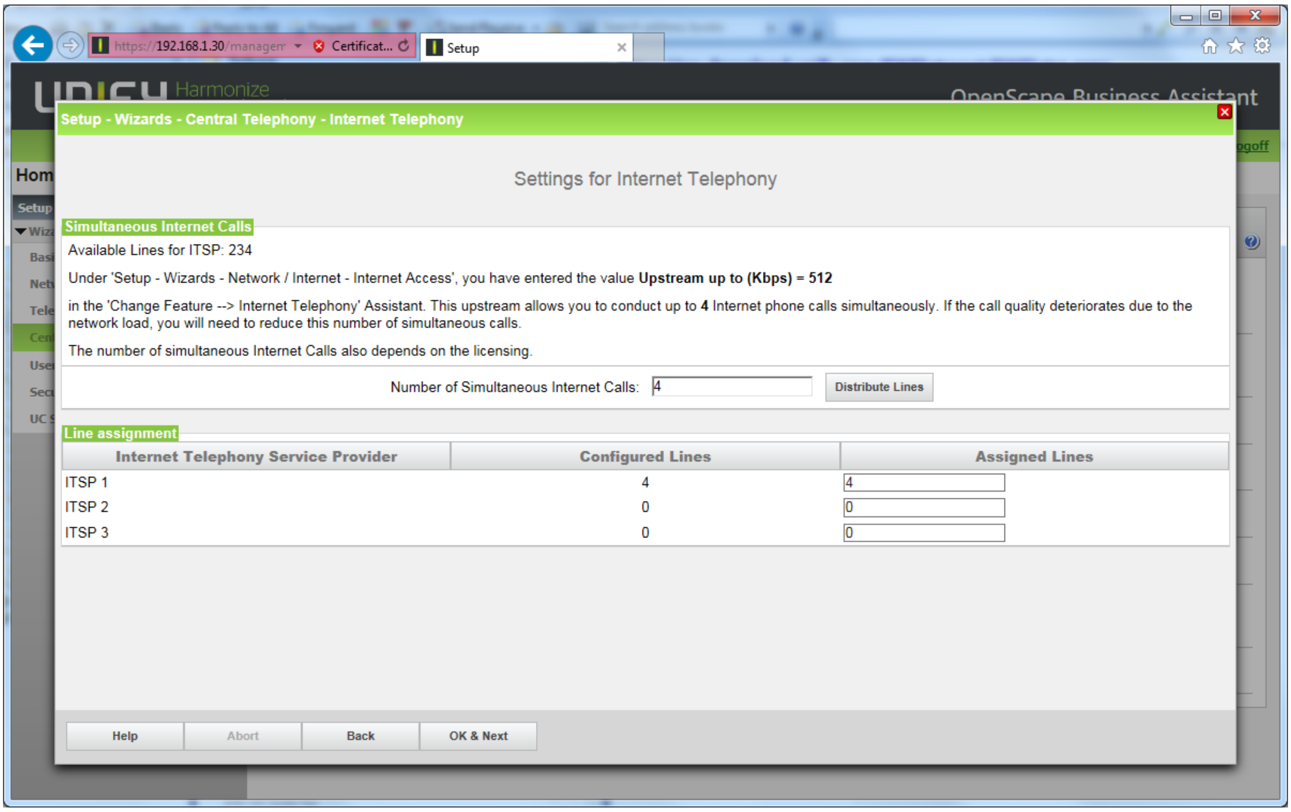
21. At the next step you have to configure the maximum number of simultaneous calls based on the Upload Bandwidth of your internet connection and distribute the lines per ITSP.

    The maximum number of simultaneous calls depends on the Upload. If voice quality falls as a result of network load, you must reduce the number here.

    The preconfigured upload value derives from the value used in the Internet Wizard (e.g. for 512 kbps upload you can have up to 4 calls).

    If all configured ITSP's should share the same number of simultanous calls, press "Distribute Lines" in order to distribute the defined number to each ITSP. If you have more than one ITSP and the amount of calls should be different, you can edit the values in the column "Assigned Lines".

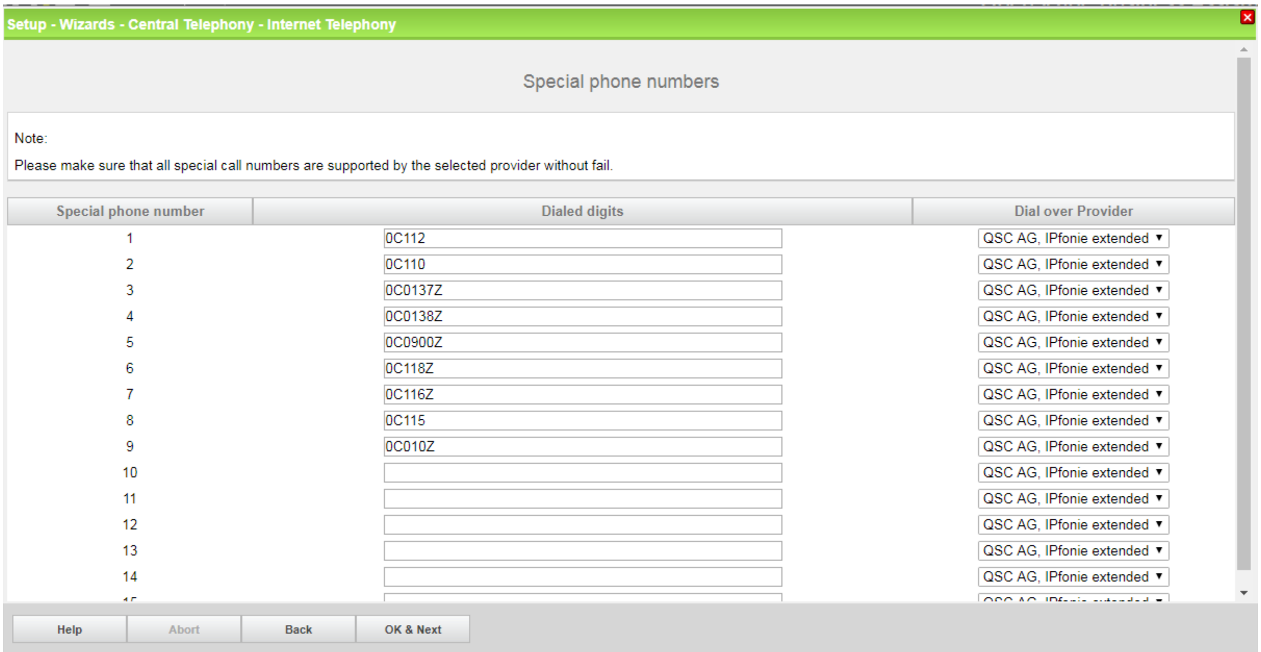
    The number of assigned lines should be equal to the value of "parallel calls" booked at the resprective ITSP.



Click OK & Next when finished.

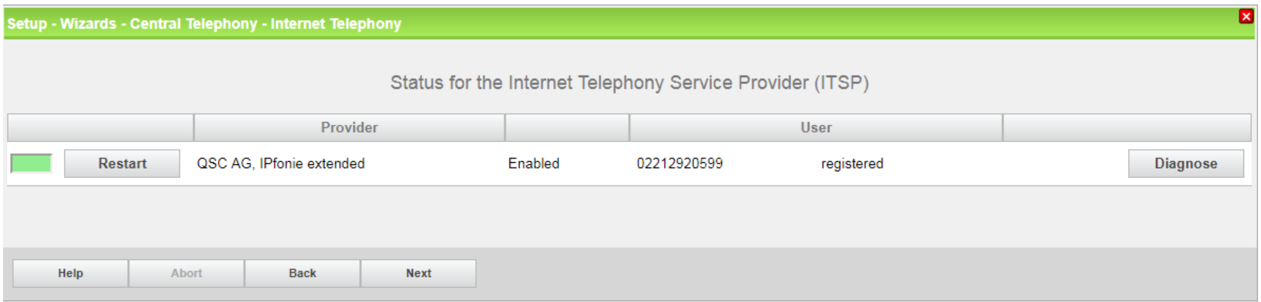
22. Next you can define the handling of special numbers. The table consists of a country specific default of numbers which should be dialed without further manipulation.
    If you want to add additional numbers which needs to be dialed as entered you may add them in the **Dialed digits** column. The entries will be stored in the LCR dial plan with the following syntax ::
    - 0 to 9: allowed digits
    - -: Field separator
    - X: Any digit from 0 to 9
    - N: Any digit from 2 to 9
    - Z: One or more digits to follow up to the end of dialling
    - C: Simulated dial tone (can be entered up to three times)

    Use the **Dial over Provider** column to specify which trunk should be used for the special number. Please make sure that emergency numbers are allowed by the specified provider.

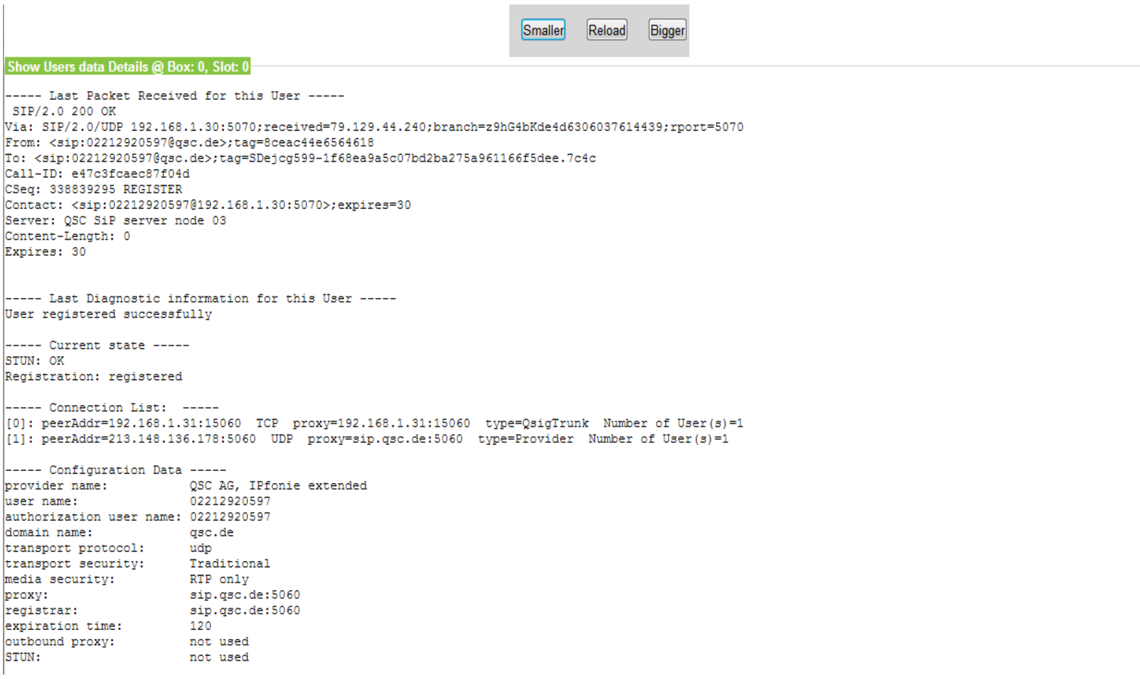


23. Click **OK & Next**.
24. The following picture shows the status of the ITSP connection.

    In this screen you can also restart your ITSP. In case that the ITSP is using registration, this will result to a de-registration and a re-registration.



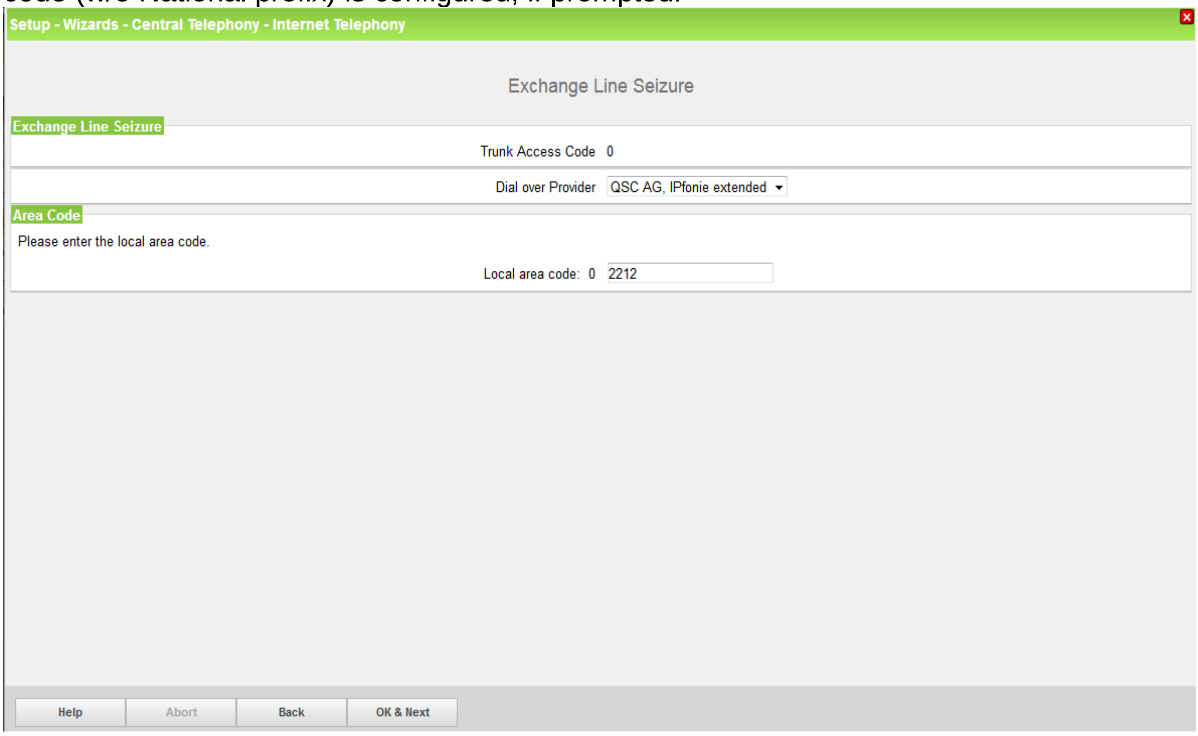
Pressing the Diagnose button will open a new browser window to display status and configuration information. This information should help to analyse problems:



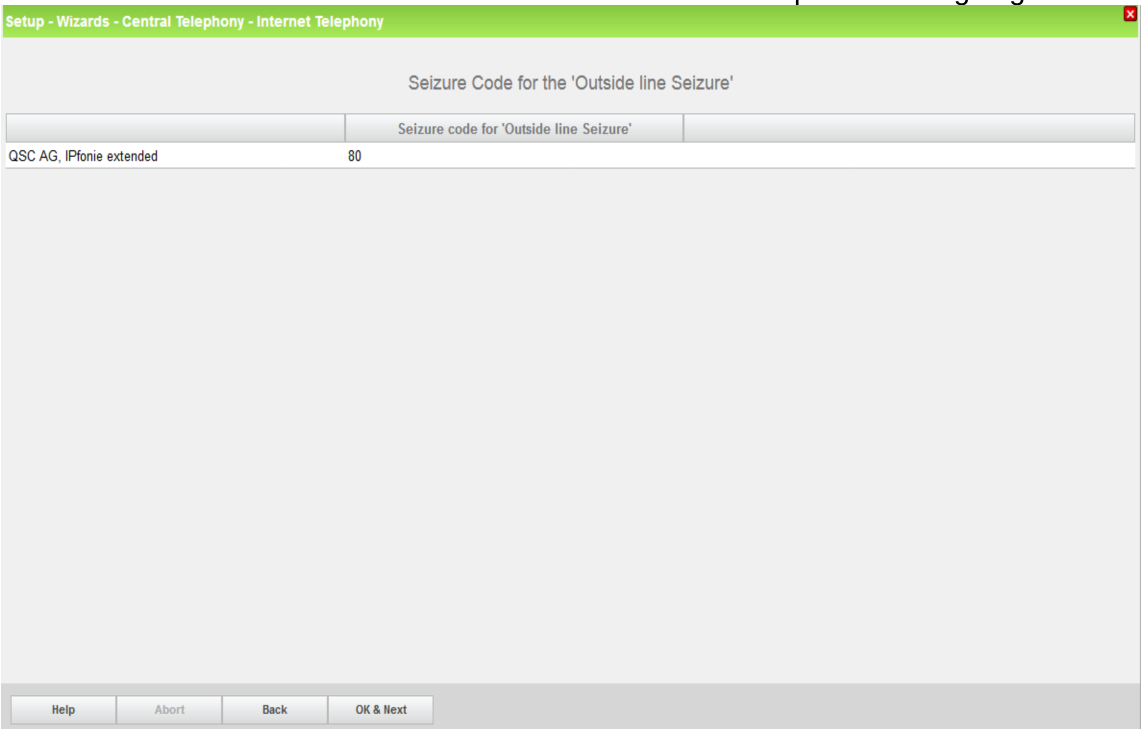
Close the window and return to the Status page

25. Click **Next**

In this page we choose the “Provider” which is selected by seizure code 0. In addition the area code (w/o National prefix) is configured, if prompted.

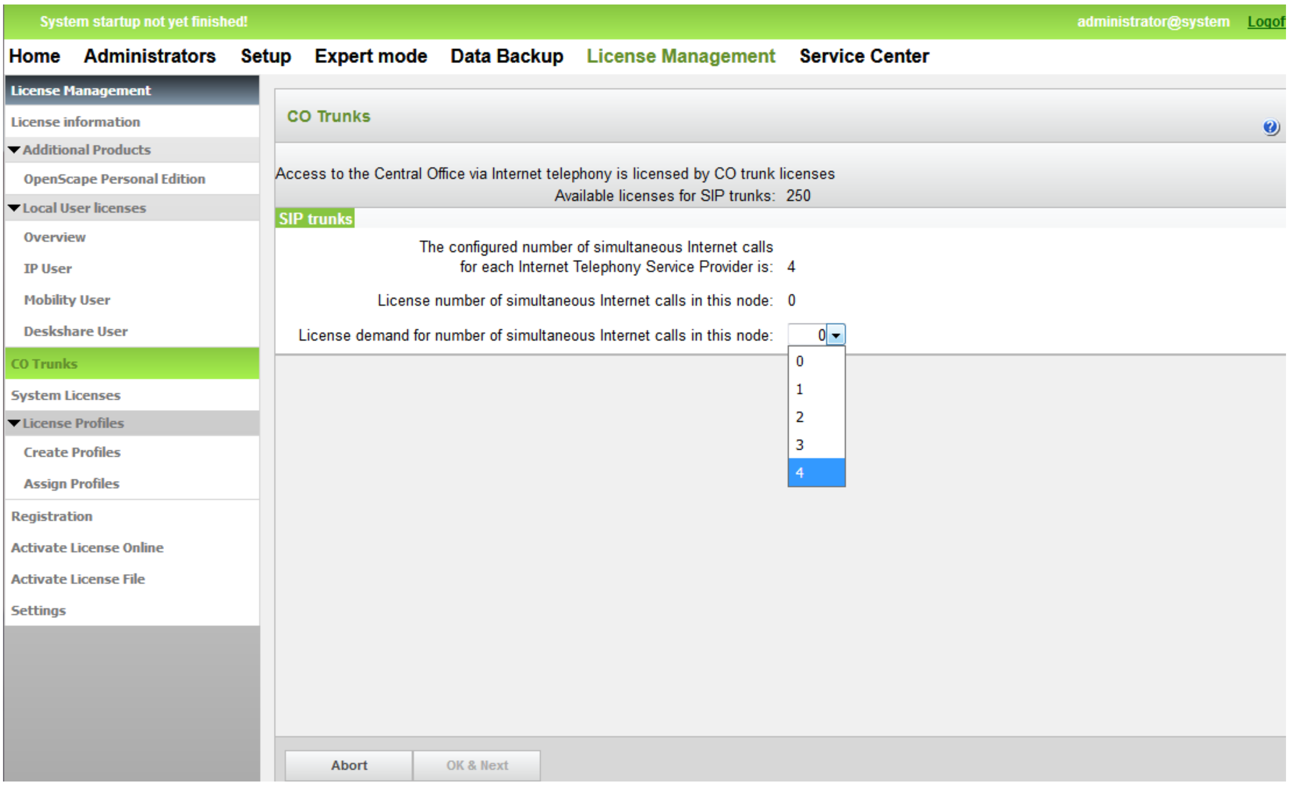


26. Click **OK & Next**. for an overview about the seizure codes to place an outgoing call



27. Click **OK & Next** and then **Finish** to exit the **Internet Telephony** wizard

28. The last step is to configure the licenses for the SIP Trunks.
    Go to tab **License Management > CO Trunks** and set the ITSP/SIP Trunks you want to activate.



4. Appendix

4.1. Known restrictions

Due to different reasons, OpenScape Business does not support some features, which may be offered by an ITSP. This section contains a list of feature limitations at the ITSP access.

Media Types

OpenScape Business offers support for the media types **audio** (Voice and voice band data) and **image** (fax) on the ITSP interface. All other media types (e.g. Video) are NOT supported on the ITSP interface. This limitation does not apply to the internal SIP interfaces.

Media Transport

The integrated SBC function terminates all media streams of the ITSP. Only one UDP stream is supported per connection (RTP/T.38).

DTMF

The system supports DTMF according to the RFC2833/4733 standard. Sending and receiving of DTMF-relay in the body of an INFO or NOTIFY message is NOT supported. If an ITSP does not support RFC2833/4733, control of the UC application is NOT possible.

Early media rendering

OpenScape Business will render early media based on the information in the received provisional responses. If SDP is present the `P-Early-Media` header field will be used (if present) to decide if early media needs to be rendered.
OpenScape Business has no means to supervise the RTP stream.

If early media is rendered, OpenScape Business stays connected to that stream until the call is answered. There is no fallback to local ringback tone if early media stops (regardless of received `P-Early-Media` header fields).

Early media in combination with forking

OpenScape Business will render the media announced in the last provisional response containing valid media information when a forked call results in several early media streams. During early media a change in the codec is not supported.

Subscribe/Notify for Message Waiting Indication

As OpenScape Business systems have internal voicemail systems they do not support subscriptions for message waiting to public voicemail systems provided by the ITSP.

Transfer

REFER requests are rejected on the ITSP interface due to security considerations.
Handling of these messages would mean to create new calls to a number that was provided by an external, possibly untrusted, party, which may result in high costs (toll fraud).

The OpenScape Business system does not send REFER to the ITSP leg.

There is no signaling for call number update towards the ITSP during transfer. In call transfer scenarios, where an OSBiz user transfers a call with an external subscriber or transfers a call to a

PSTN subscriber, the PSTN subscribers don't display the connected party, but continue to display the OSBiz user.

In blind transfer scenarios from an OSBiz user, during the transfer the transferred party hears MOH instead of ring back tone.

Call forwarding

302 (Redirect/Diversion) responses are rejected on the ITSP interface due to security considerations.
There is no signaling for call number update towards the ITSP on the incoming leg during call forwarding. In call forwarding scenarios, where an OSBiz user forwards a call with a PSTN subscriber, the forwarded PSTN subscriber will display the original (dialed) OSBiz user and not the connected (forwarded to) party.

In case of diversion, OSBiz does not forward the Diversion information header from initial INVITE. Moreover, diversion counter is not incremented.

Feature activation using Keypad

Some ITSP allow feature activation by sending of the characters * and/or # followed by a feature code (stimulus feature activation). Using * and # on the ITSP interface has to be explicitly allowed by the system configuration (system flag).

In a SIP-URI "#" is an invalid character which has to be escaped. Therefore the ASCII representation in Hex is sent on the line: # is 0x23 -> %23 in SIP-URI

! **Note: When invoking features by "stimulus" procedures, no indication might be given to the user (e.g. no special dial tone, no display information). Thus using stimulus features is not recommended and requires a careful handling by the user.**

STUN (see 4.4 for details)

STUN can be enabled/disabled for each individual provider separately. Nevertheless all providers which have STUN enabled use the same STUN server and STUN mode.

Trunk selection for myPortal, myAttendant, myAgent

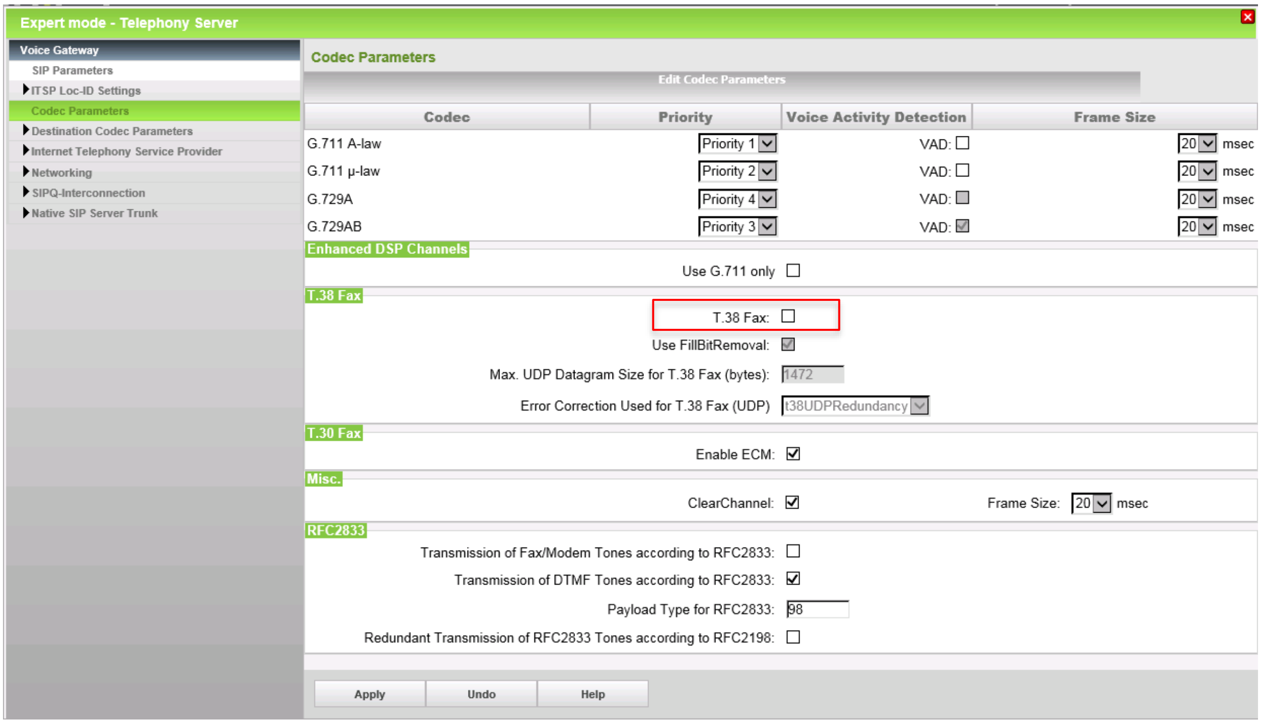
A trunk prefix for selecting a SIP trunk as an alternate trunk can be used from the OpenScape Business client's dial box but not from client's directories and settings. Due to this limitation SIP trunks cannot be used for presence based call forwarding if a TDM trunk is present in parallel.

4.2. Fax

Fax is possible in two ways, either by protocol T.38 or by using a transparent channel with codec G.711. Fax over T.38 is more reliable and secure than fax over G.711.

- For fax T38, nothing special needs to be configured. It is enabled by default.
- If the ITSP does not support T38, T38 should be disabled in order to send the fax via G.711:

Expert Mode > Telephony Server > Voice Gateway > Codec Parameters > Disable the flag “Fax T.38”. All the other settings should remain at the default values.



For more detailed information about each ITSP settings, please check the document available under

https://wiki.unify.com/index.php/Collaboration_with_VoIP_Providers#Released_SIP_Providers_in_Detail

4.3. Codecs and RFC2833 Setup

In the above screen you can also configure the codecs and its priorities for Gateway calls (calls via TDM stations). If G729 is used by the provider, then both G.729A and G.729AB MUST be activated.

Also RFC2833 is configured here. The RFC2833 dynamic payload type is negotiated between the OpenScape Business system and the ITSP. If the provider does not support negotiation and request for a specific value, this must be entered under "Payload Type for RFC2833".

For more detailed information about each ITSP settings, please check its Configuration Guide document (if available) located at:

https://wiki.unify.com/wiki/Collaboration_with_VoIP_Providers#Tested_VoIP_Providers_by_Countries

	Redundant Transmission of RFC2833 Tones according to RFC2198: Since V2R5 this parameter is deactivated by default in new systems. In systems which have been configured before the parameter may be activated. Even if the support of RFC2198 is negotiated as part of the SDP offer/answer procedure some ITSP reject a call setup if RFC2198 is offered. For that reason it is recommended to deactivate RFC2198!

4.4. Configure STUN

In case where the OpenScape Business system is connected behind a Router and the Interface used for ITSP calls has a private IP address the address information in SIP and SDP contains this private IP. Some ITSPs require to receive the public IP of the router in SIP (e.g. in Via:/Contact:) and SDP (c= line/m=line) to route packets correctly. For this use case the system has a STUN component to provide the correct address information in the SIP and SDP. The STUN component has different operation modes which needs to be configured. As the requirements differ between the various ITSPs the necessary configuration is identified during the ITSP certification.

To allow a better understanding of the STUN component, the configuration options are described below.

The STUN configuration is divided in two parts:

1. the global STUN configuration
2. the ITSP specific STUN configuration

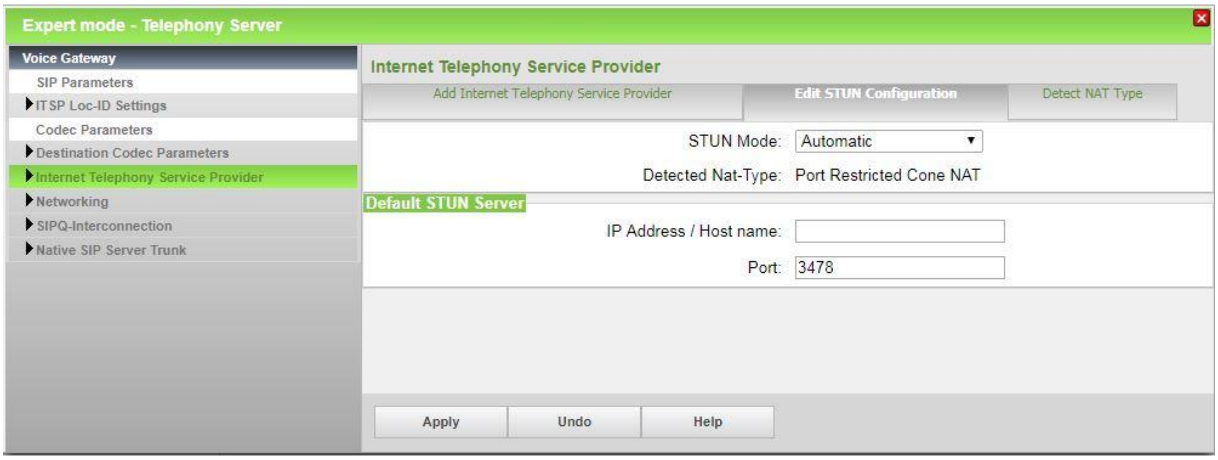
More information about STUN can be found in the wiki under:

https://wiki.unify.com/index.php/Network_Configuration_for_VoIP_Providers

	Importanrt: If STUN is needed for an ITSP both parts MUST be configured. The global STUN mode AND the activation in the profile

4.4.1. Global STUN configuration:

"Expert Mode > Voice Gateway > Internet Telephony Service Provider > Edit STUN Configuration"



On this page you can change the **global** STUN mode used for **ALL** VoIP traffic going to the internet (e.g. ITSP, Device@Home, Circuit).

In addition this page is used to define a "**Default STUN Server**" which is used for Device@Home if no ITSP is used. Please note that this setting is **NOT** used if an active ITSP has a STUN server configured.

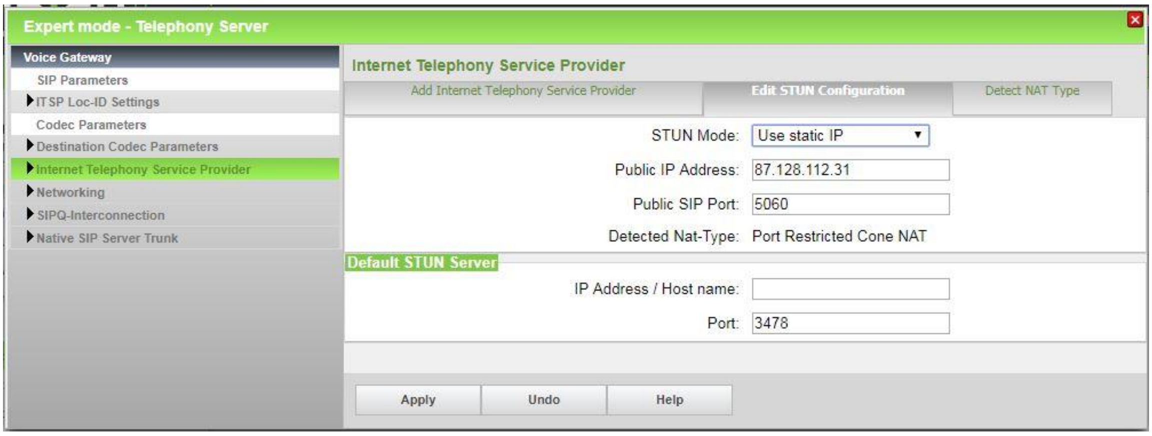
STUN mode:

- **"Automatic"(Default)** The system determines if STUN is needed based on the configuration and the detected NAT type.
  Please note: symmetric NAT is not supported.

If STUN usage is possible, the STUN protocol is used to determine the public IP address and port to be used in SIP signaling and media (SDP).

- **“Always”** STUN is always active, even if no ITSP is active.
  The STUN protocol is used to determine the public IP address and port to be used in SIP signaling and media (SDP).

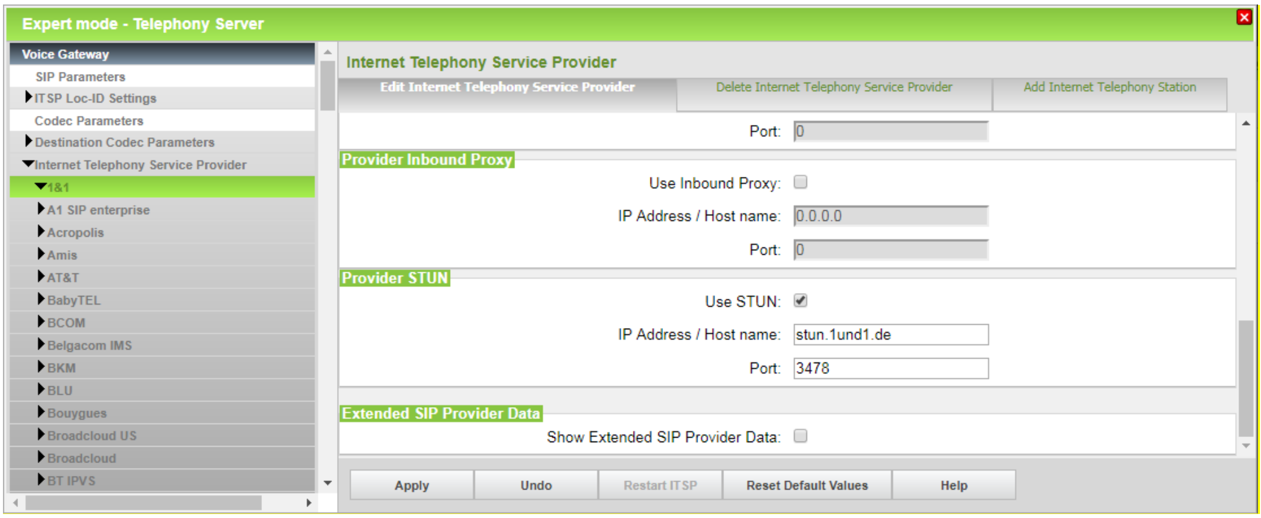
- **“Use static IP”** In this mode the IP address and port to be used in SIP and SDP is configured. For SIP signaling the public port is configured here, for media (SDP) the ports configured in port management are used.



- **“Port Preserving Router”** In this mode the STUN protocol is used to determine the public IP address to be used in SIP signaling and media (SDP). The port is used unchanged in SIP/SDP

4.4.2. ITSP specific STUN configuration:

The usage of STUN can be activated / deactivated individually for each provider. This is possible with the profile parameter: “Use STUN” as seen below.



Importanrt:

If the global STUN mode is set to **Use static IP** the STUN server address must be filled in as well (even if no server is used). You can use a place holder like **use.static.ip** for the server name

4.5. Multisite Configuration

The main concept of the feature is to host several locations and ITSPs (up to 8) in a centralized OSBiz system and use the appropriate ITSP depending on the station location. Calls within the same area will be possible without dialing the area code (functionality is supported by some countries only, e.g Germany). All ITSP calls will have access code “0” for all users regardless of the station’s location. For all these, two new concepts are introduced:

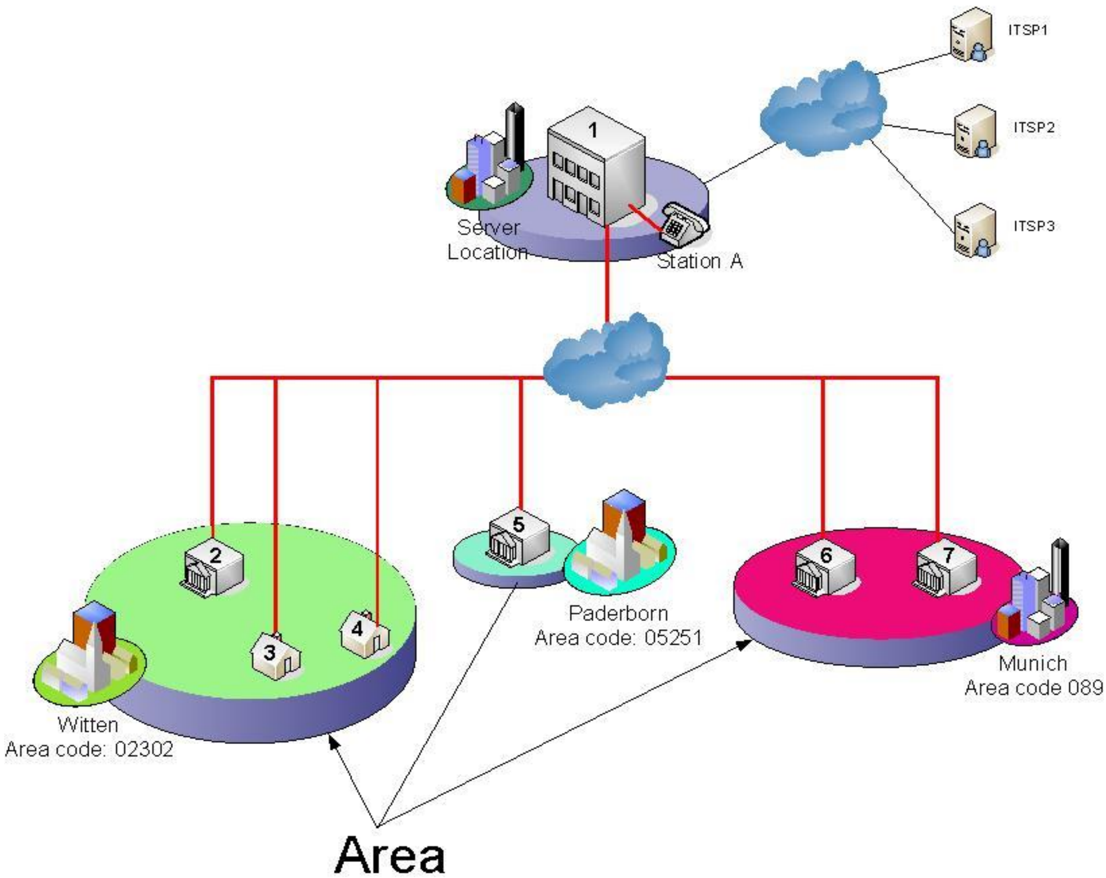
Area assignment:

Each station belongs to a preconfigured location/area. Dial Rule “SIP local” based on location.

Dedicated Route:

Each location will be configured to use a specific ITSP. Calls from specific station will overrule static “Route” entry in LCR Routing Table.

Scenario Overview:



*For this example we will use 3 locations and 3 respective ITSPs (maximum can be 8).

The concept is to use a specific ITSP for each area. In the above scenario we assume:

Area Witten > ITSP 1

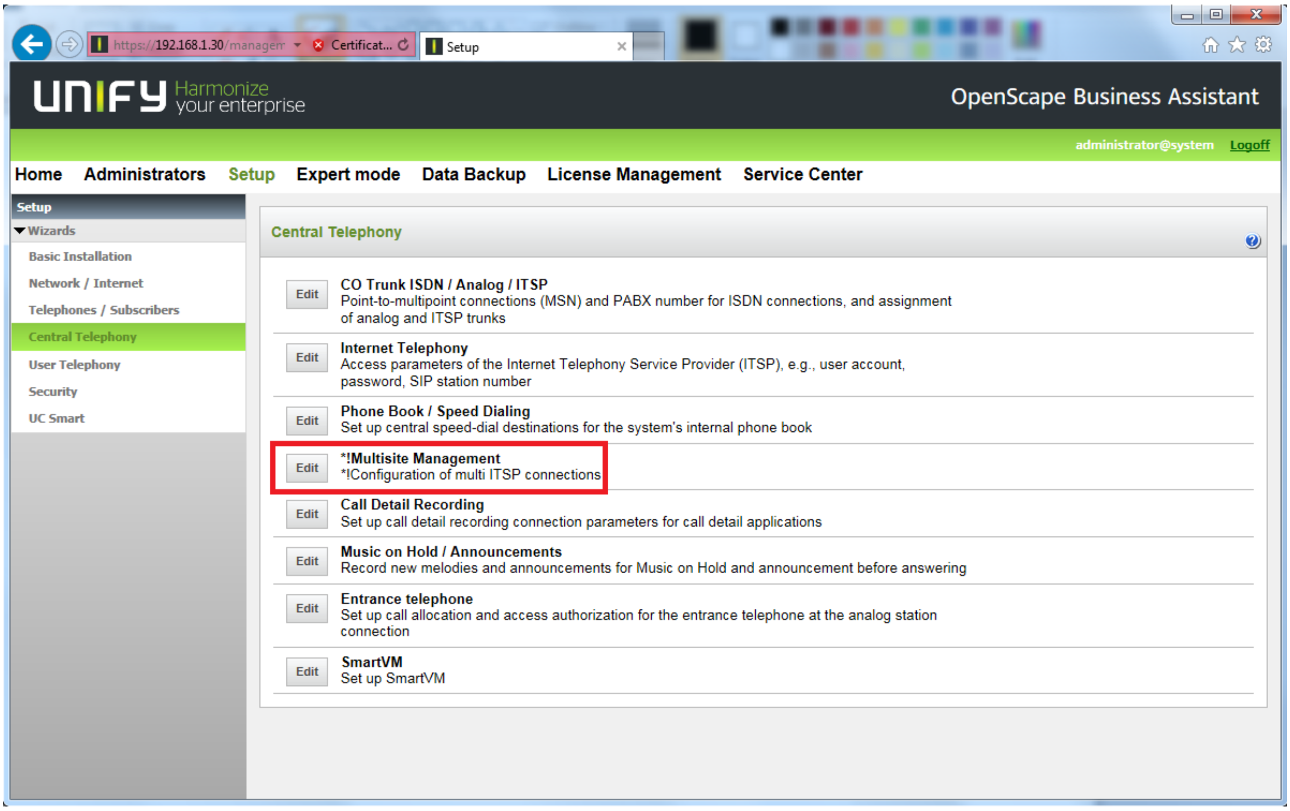
Area Paderborn > ITSP 2

Area Munich > ITSP 3

In order to configure a Multisite scenario, a new wizard “Multisite Management” is introduced in Setup > Central Telephony as seen below.

Please note that the wizard is visible only when at least one ITSP is already activated and system is not part of an OSBiz network.

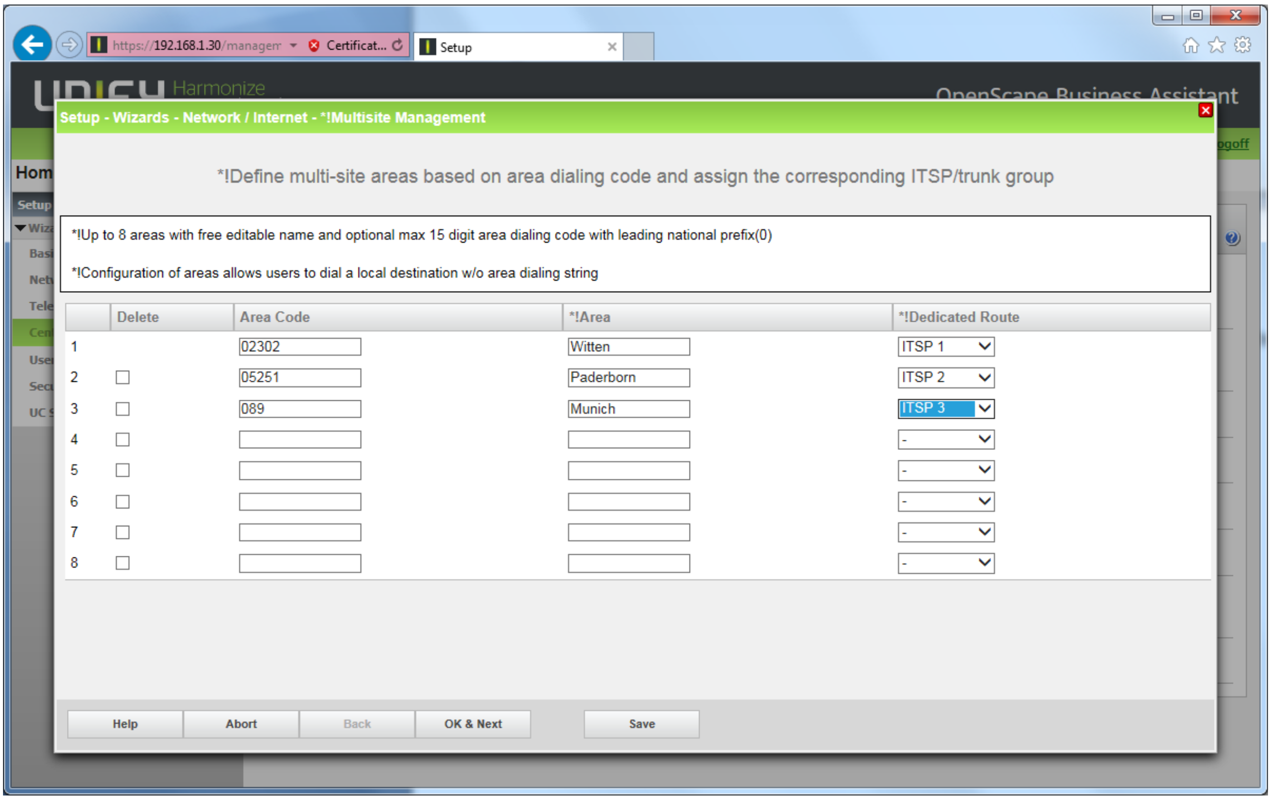
For this example we assume that all 3 ITSPs are already activated.



This is the first screen of the Multisite Management wizard. Here the areas must be defined. The Area Code for the first area is provided by default (derives from Location data).

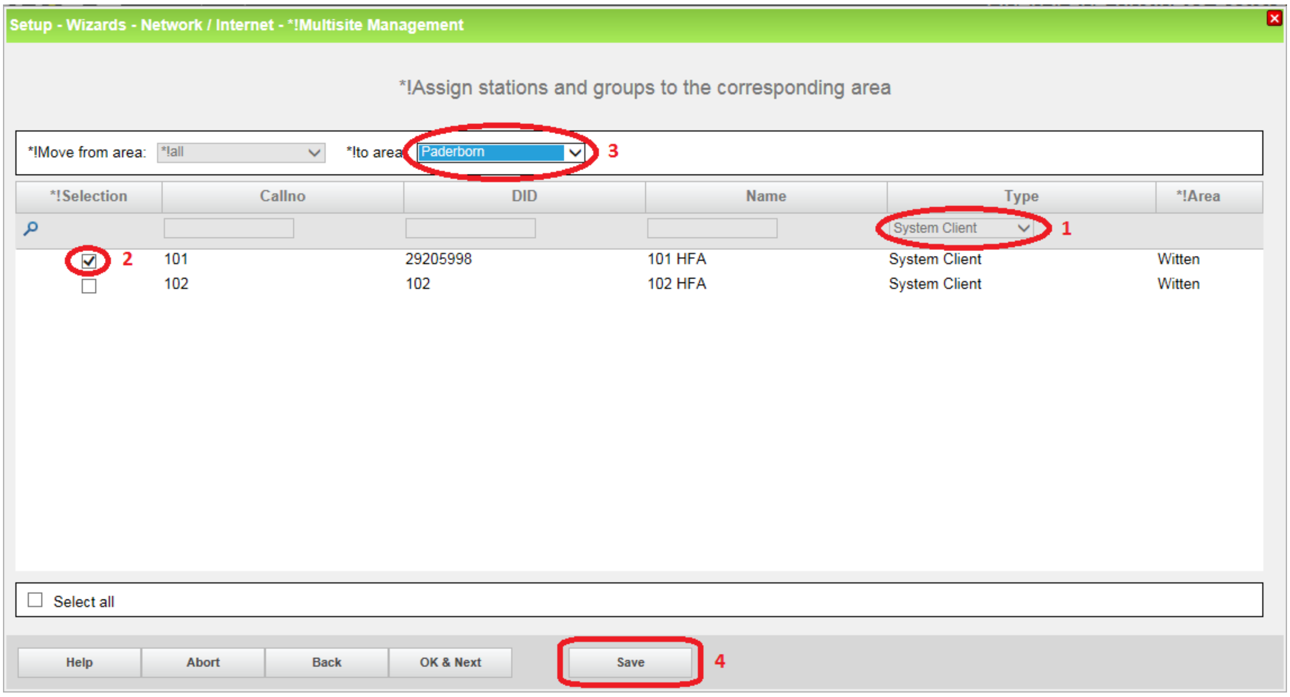
Please keep in mind that the Area Code field configured in this screen is not the same as the Local Area Code in Route settings. The Area Code set here will be used later in the LCR (H parameter). Please also note that all ITSP Routes must have only the Country Code configured, no Local Area Code should be configured for the Multisite functionality.

Here also a dedicated route must be configured for each area (one ITSP per location).



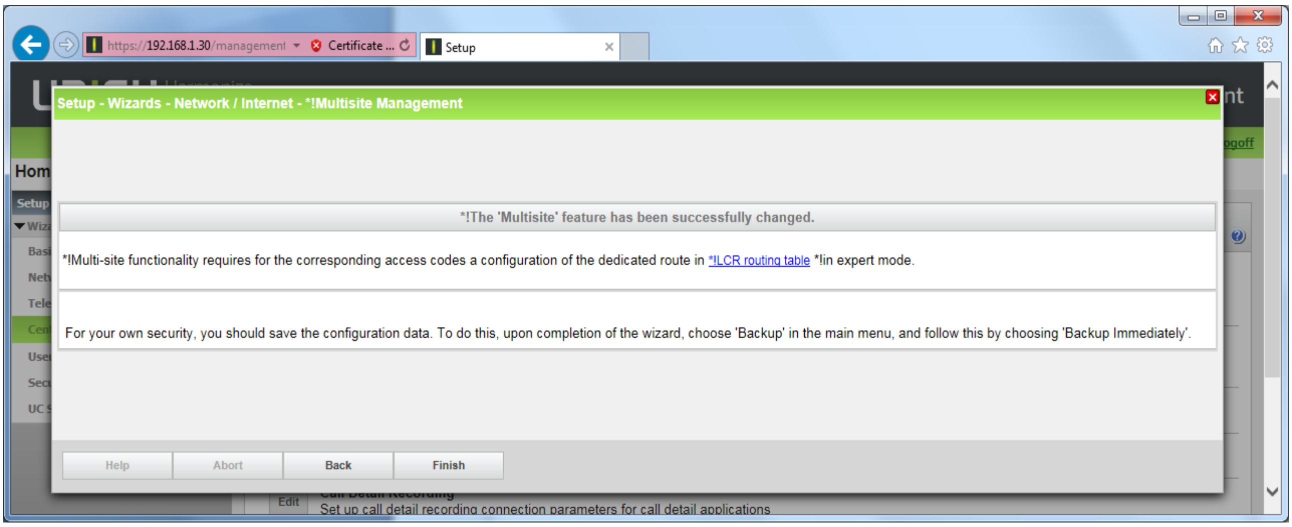
Click OK & Next

The next step is to assign stations to a specific area. By default all stations belong to the first area. Search is possible by various criteria/ filters. Specific stations can be selected or whole groups and moved to another Area. Here we have filtered to system clients (1), then chose station 101 (2) and then chose move to area Paderborn (3). In order to apply this change, please press Save (4). After pressing the Save button, the change is applied and you may continue with other changes in the same screen. After finishing all changes please press OK & Next.



Respectively we assign also station 102 to area Munich. So the final assignment is:

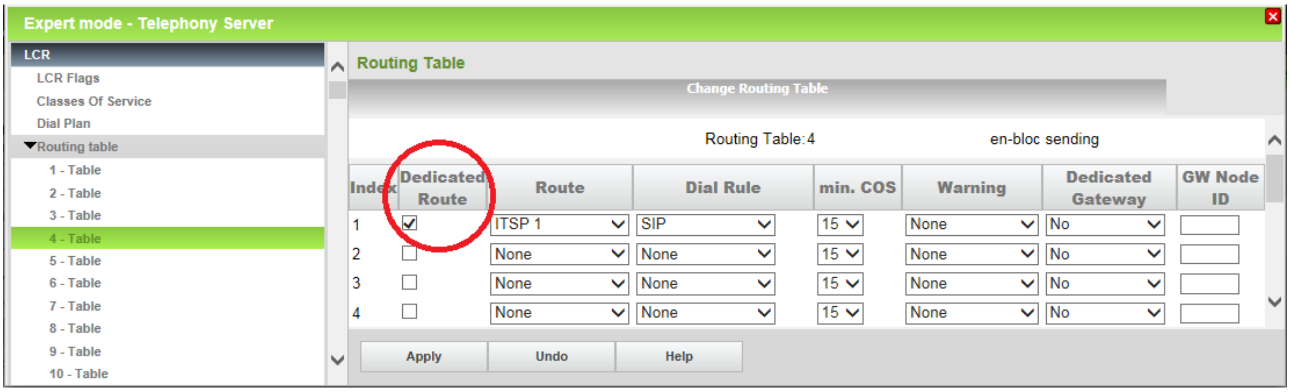
Station 100 belongs to Area Witten
Station 101 belongs to Area Paderborn
Station 102 belongs to Area Munich

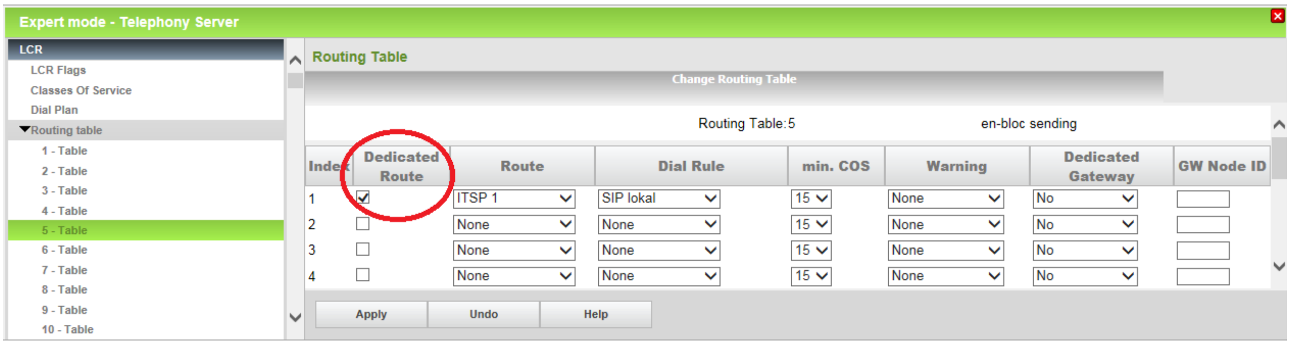


The next steps must be done in Expert Mode > LCR:

In LCR the Dial Rule “SIP Local” is set to HE2A (Provided by default in V2 or after LCR reset) where H is the new parameter that reflects the respective area configured earlier. All ITSPs will be accessible via seizure code 0 (dial rules 0CZ, 0C1Z and 0CNZ). Therefore all the seizure codes in all ITSP routes should be changed to 0.

The next step is to go to LCR and activate dedicated route for the Default ITSP for the respective routing tables (by default: table 4 with dial rule “SIP” and table 5 with dial rule “SIP Local”):

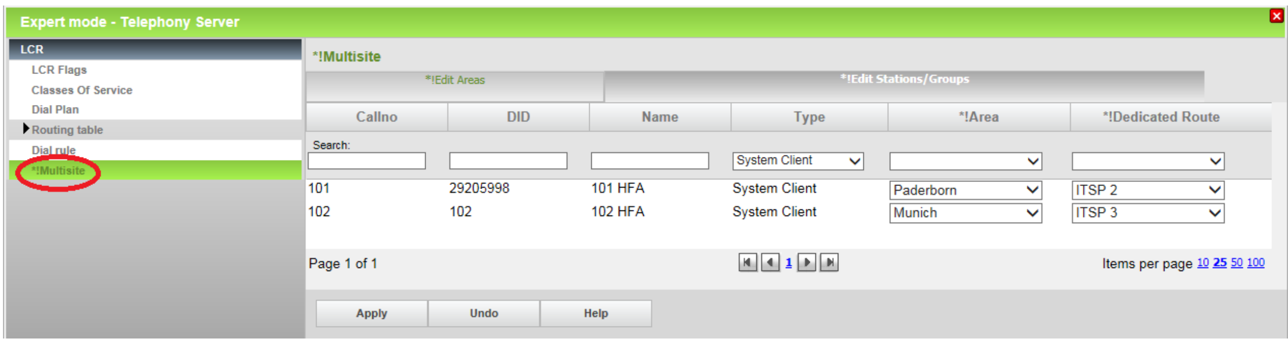




In our example we have assumed that default ITSP is ITSP 1. The default ITSP is used in cases where no (user specific) dedicated route can be determined. E.g. calls initiated for conference.

Remark: In these cases an area cannot be determined as well, therefore the H-Rule will add the area code of Area 1 (relevant for destination numbers without area code)

Areas and Stations assignment configuration done earlier in Multisite Wizard is accessible also in Expert Mode > LCR:



The configuration is now finished. Here are some use case examples for better understanding of the Multisite functionality:

Station 102 belongs to Area “Munich” with area code “089”

Example Case 1:
Station 102 wants to dial number 123456 that belongs to the same area Munich.
Station 102 will dial 0123456 (no local area is dialled).
This will result in an outgoing INVITE to the ITSP 3 where the TO field will be: 089123456.

Example Case 2:
Station 102 wants to dial an international destination, e.g. 0030210123456
This call will be routed via ITSP 3 and TO field will be 0030210123456

Example Case 3:
Station 102 wants to dial a destination by dialling the ITSP 3 prefix.
In this case station 102 can dial the relevant ITSP 3 prefix (e.g. 857) and call will be routed as normal ITSP call without the location algorithm.

Example Case 4:
Station 102 wants to dial a destination via another ITSP (e.g. ITSP 1 or ITSP 2).
In this case station 102 can dial the relevant ITSP prefix (e.g. 855 or 856) and call will be routed as normal ITSP call without the location algorithm.

General Conditions and Limitations

- Multisite is available on both Server and Embedded System.
- All sites must have the same country code, same CO access code.
- ITSPs must be configured in public number DID mode.
- Multisite is recommended to be used in a single node system.
  (Multisite node may be part of a network, but configuration can be done only via expert mode)
- Multisite wizard is available when at least one ITSP is active.
- Application controlled call scenarios:
  Destination call numbers configured on application side e.g. UC suite must be in dialable format including area code (national) or canonical format (recommended).
- IP Mobility not supported / allowed. Reason: Potential problems concerning handling of emergency calls.

The example described was a multi site and multi ITSP.
We can also have all type of combinations:

A) Multiple locations with multiple ITSPs > the example that was analyzed
B) 1 location with multiple ITSPs > Location algorithm can be used as well as described for dialing within the same area. Alternatively the default seizure codes for each ITSP can be used as well (without the location algorithm).
C) Multiple locations with 1 ITSP > Similar configuration with case A can apply.

About Unify

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

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

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