OpenScape Business V1

Tutorial

Internet Telephony Configuration Guide

Version 1.2

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.

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.

Prerequisite for the configuration is that the Internet Explorer at a PC has a LAN connection to OpenScape Office, the **WBM is started**, and you are **logged on as an administrator**. Please use the menu items as described below.

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

For general information see www.unify.com

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

http://wiki.unify.com/wiki/Collaboration_with_VoIP_Providers

2. Internet Configuration

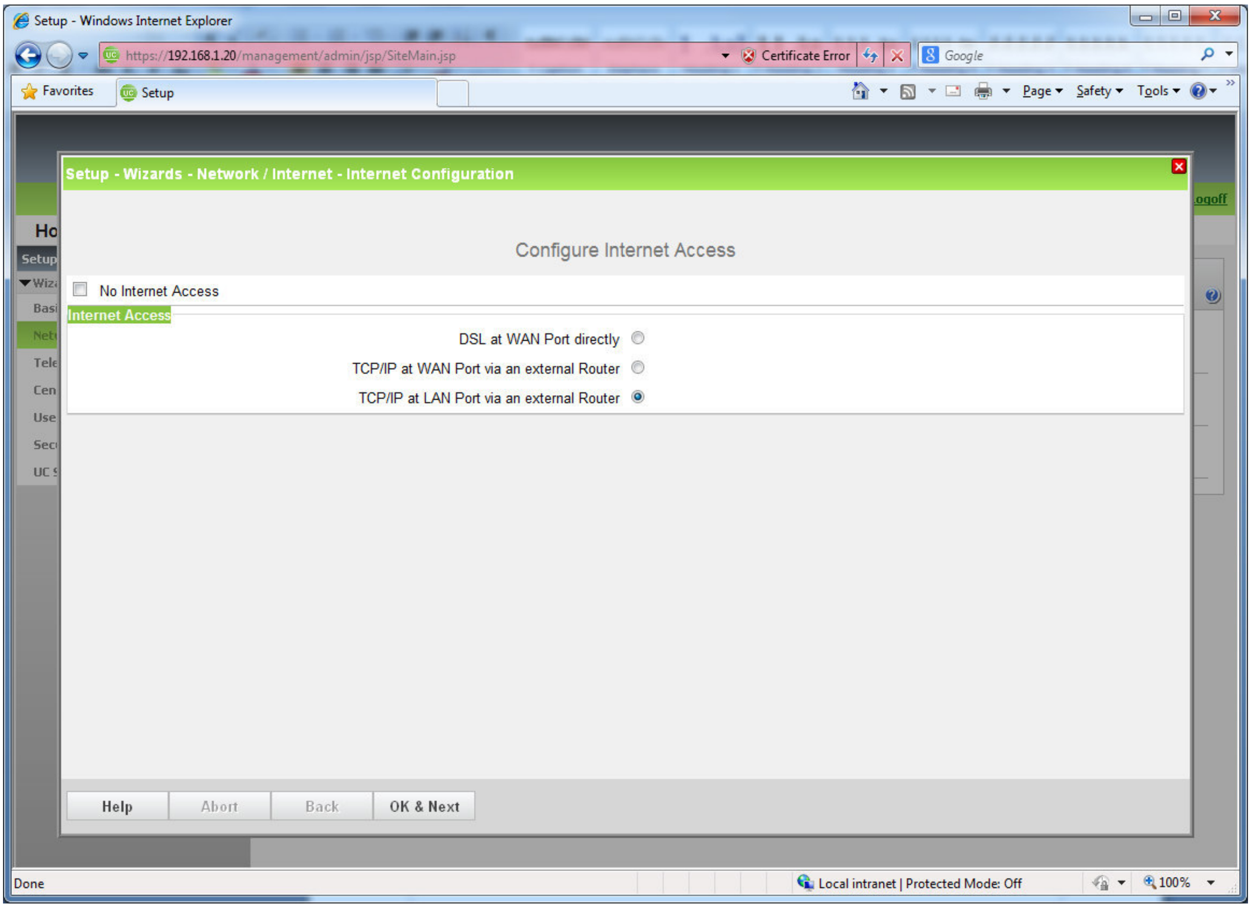
An internet connection from your ITSP or other from another Internet Service Provider (ISP) is required for Internet Telephony. The DSL bandwidth at the customer site determines the maximum number of concurrent calls (128 kbit/s for a G.711 call, up- and downstream).

2.1. OpenScape Business with external Router

If you have an OpenScape Business S or want to use the DSL router from your Provider or another external router, please enter its default Gateway IP address in the **Network Configuration** wizard. Connection to the external Router is done via LAN port. OpenScape Business does not support routers with symmetric NAT. An Application Layer Gateway (ALG) in the router should be disabled.

Please use the Internet Configuration wizard to set up your Network and Internet access.

1. Go to Setup > Wizards > Network / Internet. > Edit **Internet Configuration**



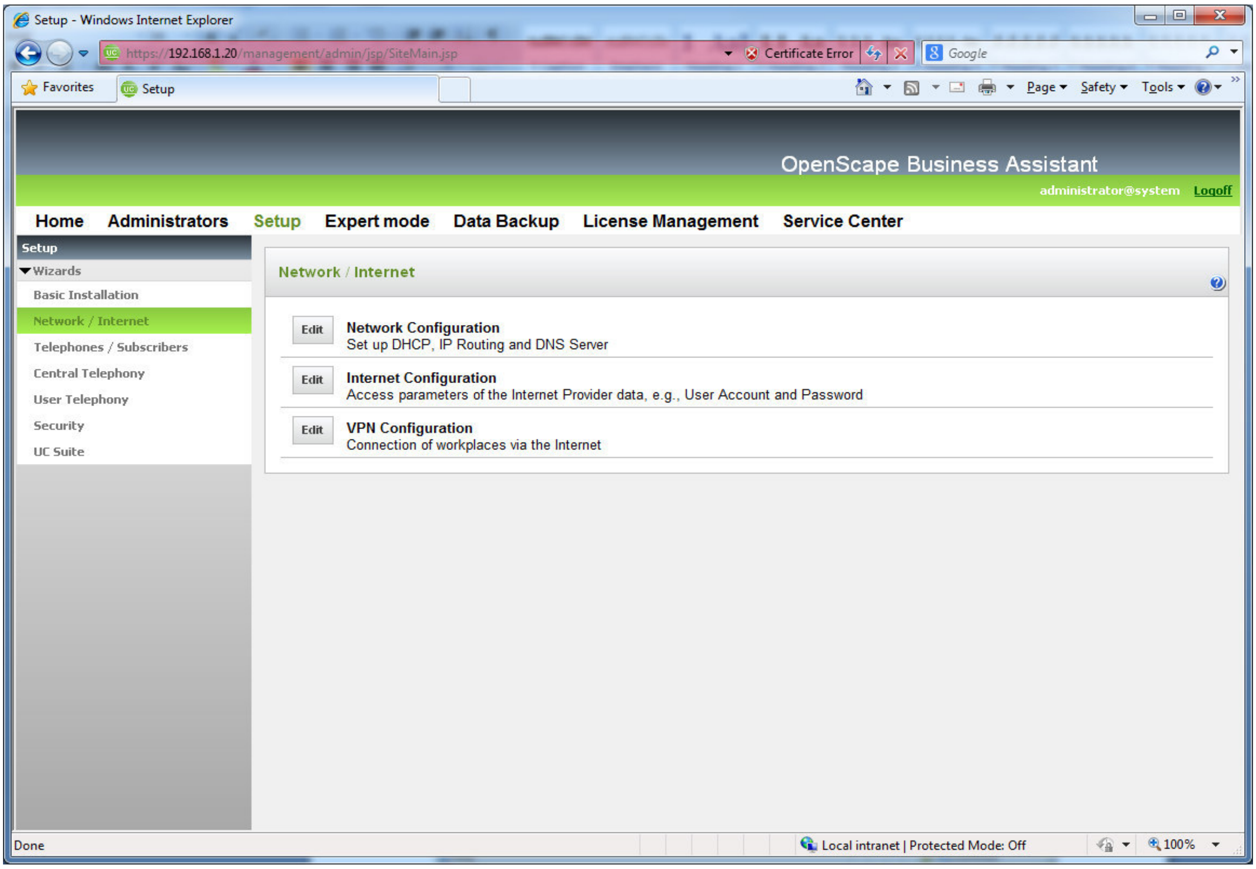
2. Choose “TCP/IP at LAN Port via an external router” and click OK & Next.
3. In the next window configure the DNS server and the IP of your default router
4. Click Finish

2.2. OpenScape Business used as Router

If you have an Internet connection with no router the OpenScape Business system is configured as a DSL router as described below. The connection to the DSL Router is done via WAN interface.

Please use the Internet Configuration wizard to set up your Internet access selecting a predefined provider or the most common type 'Internet Service Provider PPPoE'.

1. In the navigation bar, click **Setup**.
2. In the navigation tree, click **Wizards > Network / Internet**.

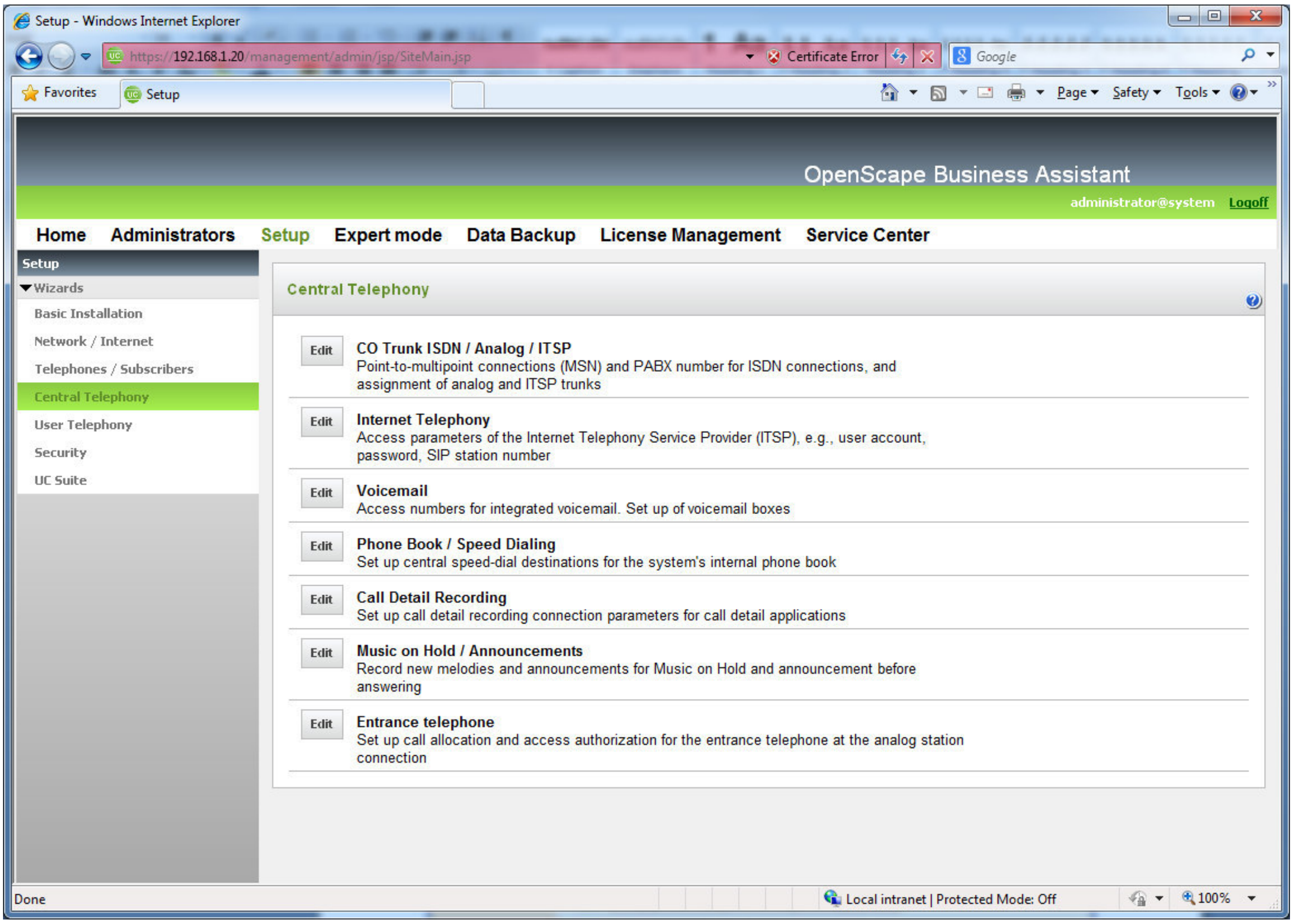


3. Click Edit to start the **Internet Configuration** wizard.
4. Activate the radio button **DSL at WAN Port directly** and click **OK & Next.**
5. From the **Internet Service Provider Selection** drop-down list, select the standard ISP Type **Provider PPPoE**.
6. The settings in the **IP Parameters** area depend on whether or not you obtain a dynamic or fixed IP address from your ISP.
   a) Dynamic IP address: Make sure that the **IP Parameters** check box is disabled.
   b) Fixed IP address: Enable the **IP Parameters** check box. Under **Remote IP Address of the PPP Connection**, **Local IP Address of the PPP Connection** and **Max. Data Packet Size (bytes)**, enter the values that you have received from your ISP. From the **IP Address Negotiation** drop-down list, select the item **Use configured IP address**.

7. For Internet telephony set **Full-Time Circuit** to **On** in the **Router Settings** area. Under **Forced Disconnect at (hour:min)**, enter the time (e.g., 04:59) at which the Internet connection is to be deactivated for a short time to avoid interruptions caused by internet provider's resets.
8. The settings in the **Authentication** area depend on whether or not the ISP requires authentication via PPP.
    a) Authentication required by ISP: Make sure that the check box **PPP Authentication** is enabled. Enter the Internet access name of the ISP as the PPP user name. The customary standard is the **CHAP Client** authentication mode.
    b) Authentication not required by ISP: Make sure that the check box PPP Authentication is disabled.
9. Select the **NAT** check box in the **Address Translation** area if you want to use NAT (selected by default). Hint: Please keep in mind that if NAT is unchecked, then the system is open against the Internet. Therefore due to security reasons it is strongly recommended to keep this flag enabled.
10. Select the **Address Mapping** check box in the **Address Translation** area if you want to use IP mapping (cleared by default).
11. Set the following values in the **QoS Parameters of Interface** area:
    a) Under **Bandwidth for Downloads** and **Bandwidth for Uploads**, enter the bandwidth in Kbps for downloads and uploads, respectively, as provided by your ISP.
    b) If you want to use Internet Telephony as well, open the drop-down list **Bandwidth Control for Voice Connections** and select the item **Upload only** or **Upload and Download**, as required. In the field **Bandwidth Used for Voice/Fax (%)**, enter how much bandwidth is to be reserved for voice and fax connections as a percentage value (default value: 80%).
12. Click **OK & Next**. You are taken to the **Configure DynDNS-Account** window.
13. If you want to use a VPN or remote access, you will need to have already applied for and set up a DynDNS account (at DynDNS.org, for example).
    a) Enter the data of your DynDNS account.
    b) Test the DynDNS account with **Connection test**.
    c) After the test succeeds, click **OK**.
    d) Click **OK & Next**.
14. If you want to use neither a VPN nor remote access, click **No DynDNS**.
15. Click **Finish** to exit the **Internet Configuration** wizard.
16. A system reboot is necessary.

3. Internet Telephony Configuration

The **Internet Telephony** wizard can be used to activate a predefined Internet Telephony Service Provider (ITSP) for the Internet Telephony user connection. You can configure Internet telephony stations for up to four 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. Clear the **No call via Internet** check box. A list of the configured ITSPs is displayed. The list contains the predefined ITSPs and custom created ITSP Profiles. 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. You can activate a maximum of four ITSPs. Click **OK** when finished

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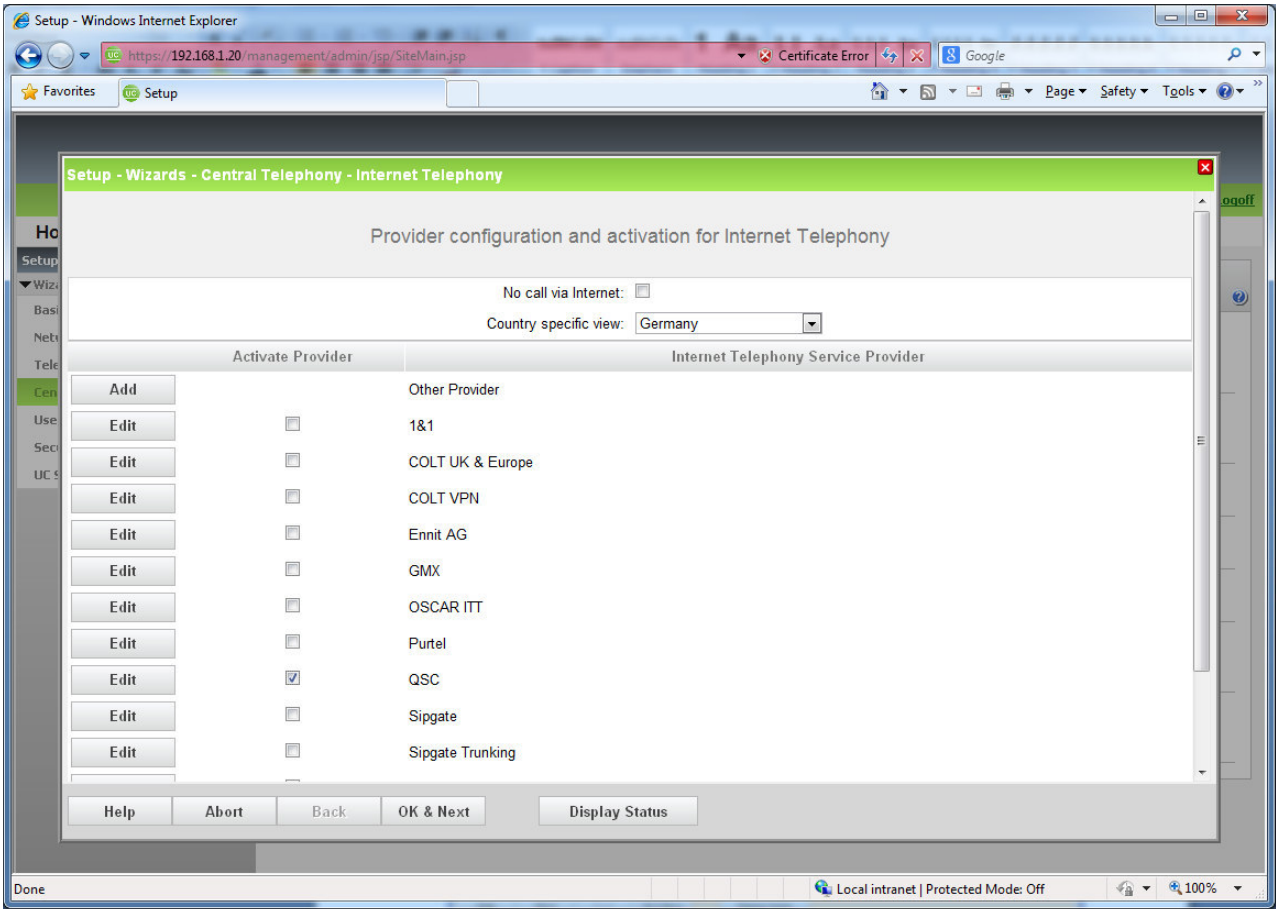


5. Click Edit at your ITSP Profile to manage your accounts and ITSP Stations.

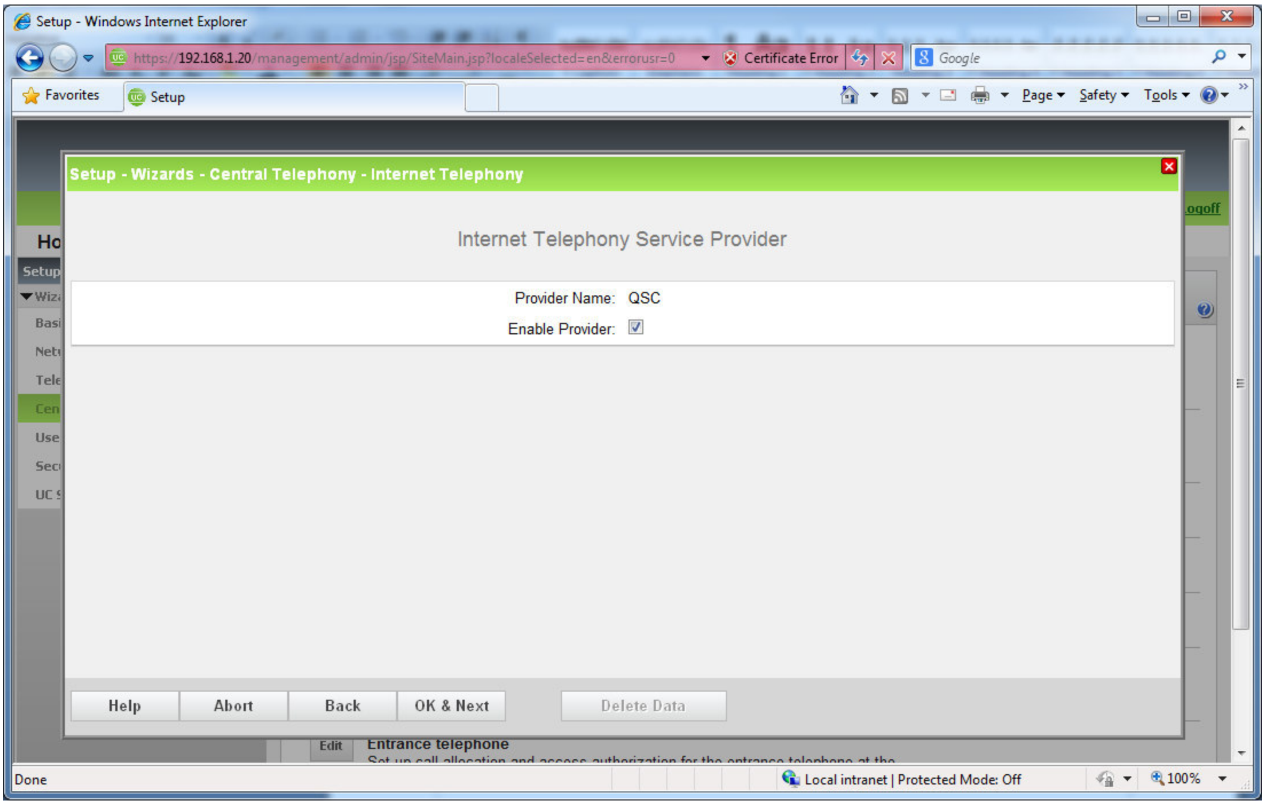
Remark:

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 15.

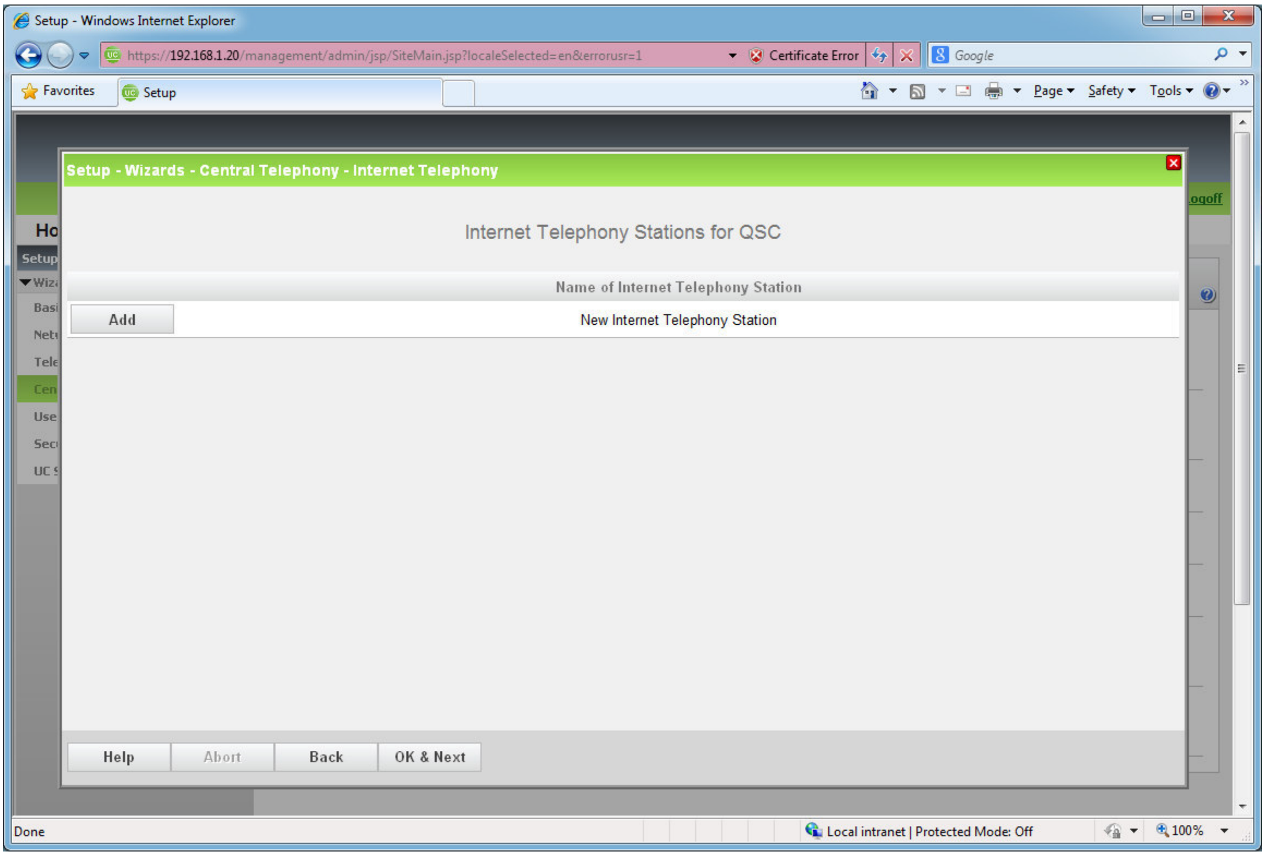
Warning:

Deactivation and reactivation of your ITSP may delete/change some data in the Trunks Routing section (especially when ITSP is configured in Public Number DID mode). In that case a reconfiguration of trunks and routes is necessary.

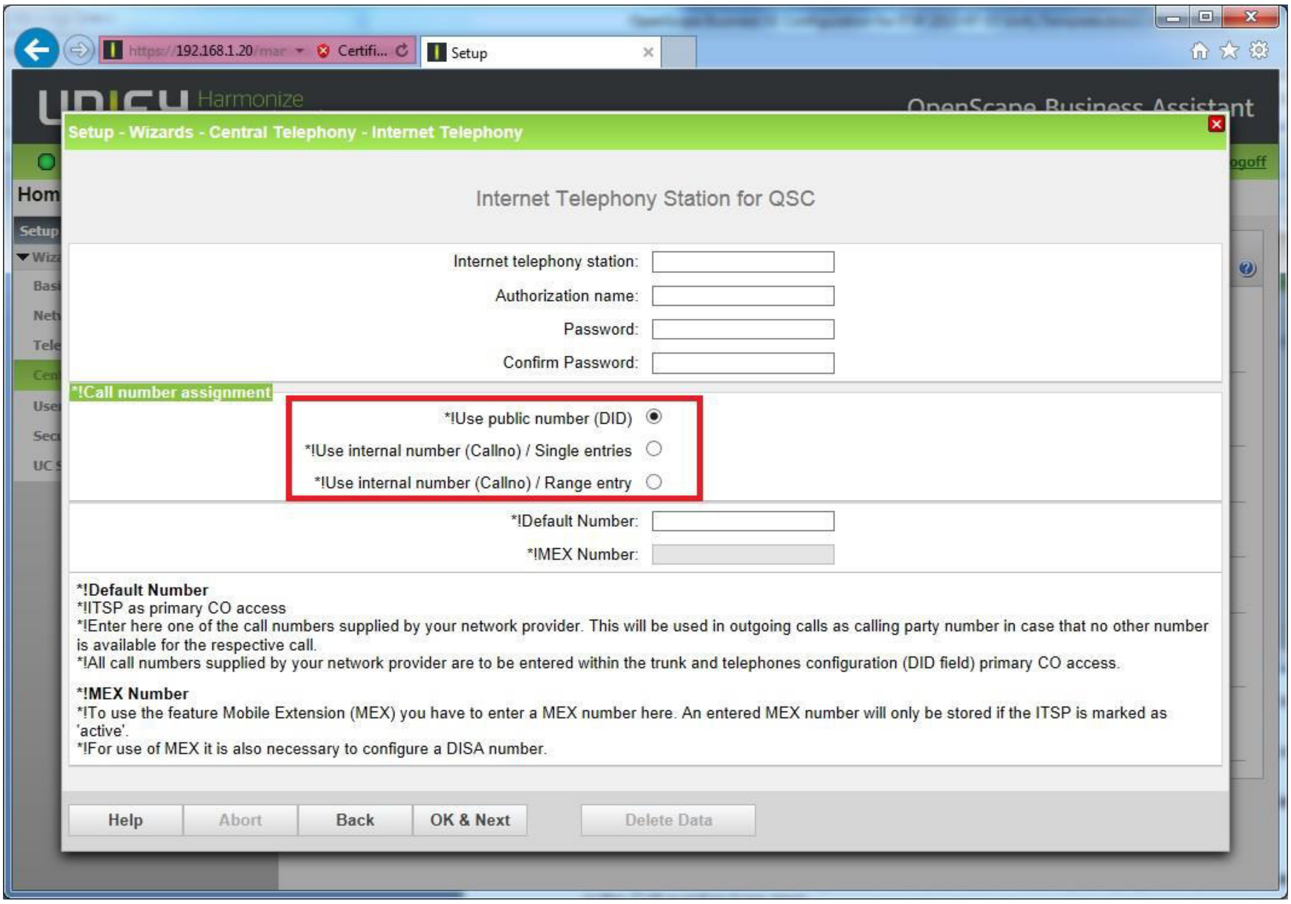
Alternatively instead of deactivating and activating an ITSP profile you may “Restart” the profile. The Restart button is available in the Display Status screen. In this case no need for reconfiguration in trunks/routes is necessary. For more information please refer to step 23 below.



6. Click OK & Next



7. Click **Add** in this screen



In this screen you must first choose the type of Call Number Assignment.

Use public number (DID):
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. This mode can be used also for central access in networked systems. One ITSP access will be configured in one node (gateway). All other network nodes will have ITSP access via the gateway node. Local CO lines may be used but only for outgoing calls (e.g. emergency calls).

Use internal number (Callno) / Single entries or Range entry:
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 (red box) is available only for the first time you configure an Internet Telephony Station for an ITSP.

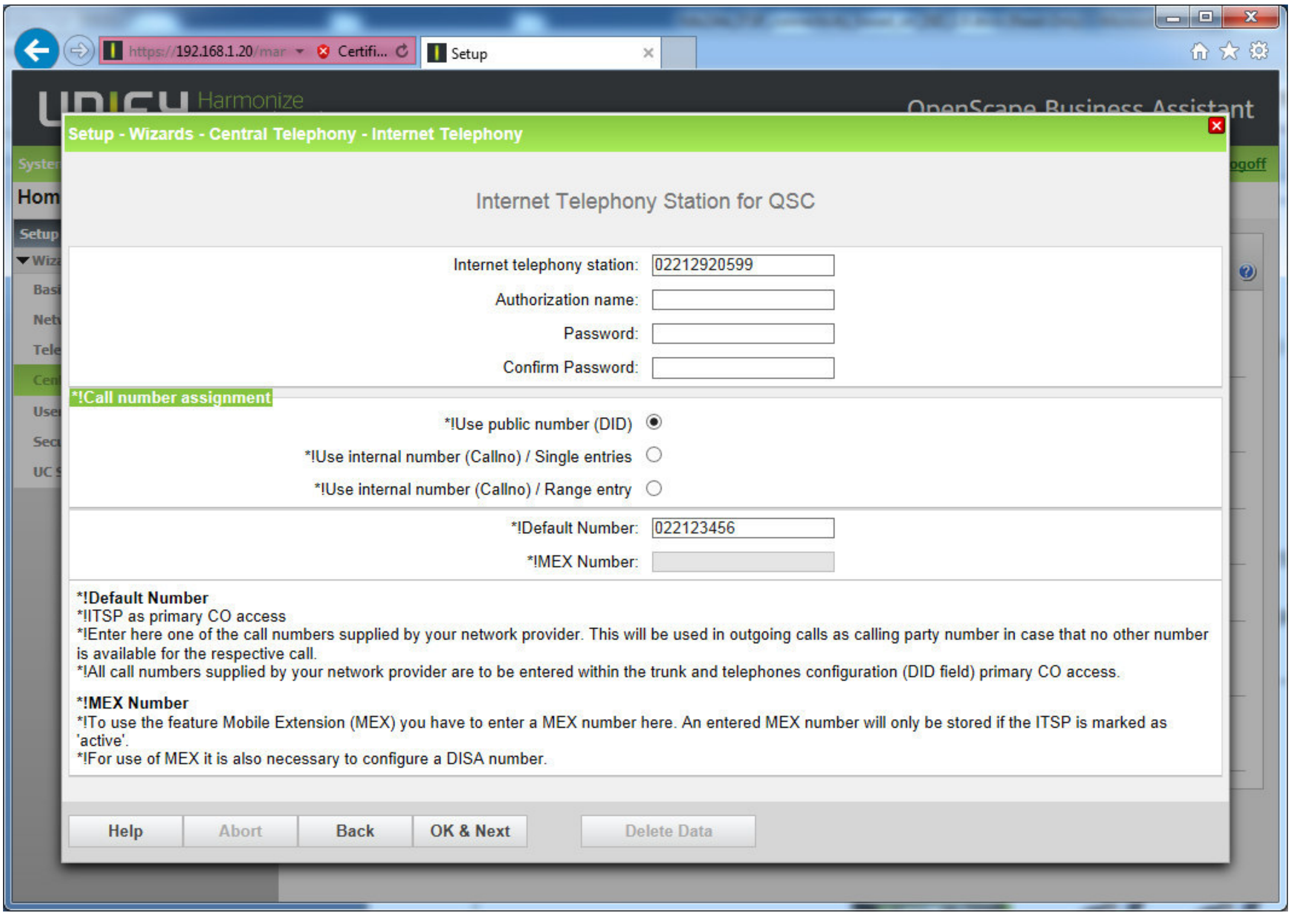
- The DID mode ("Use public number") should not be used when the system has also ISDN CO access. System must have only ITSP CO access.

Please refer to Appendix 4.5 for more details and examples of the DID mode

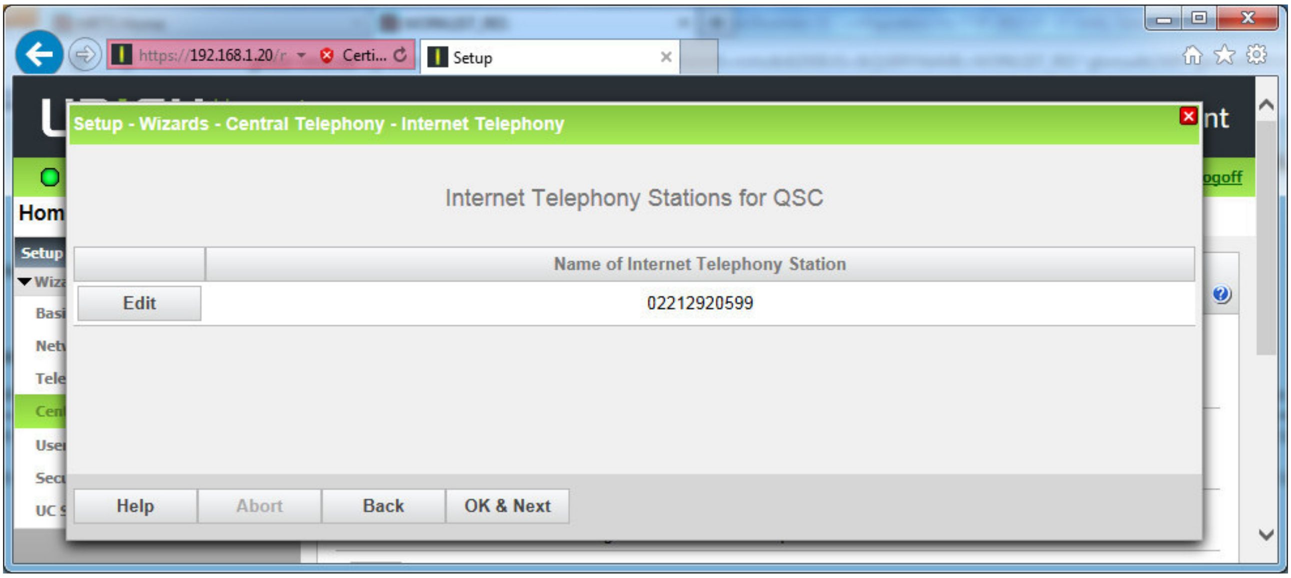
Below are the next steps (8-12) for configuring DID mode.

If you want to use the Internal Call number (Callno) mode please skip to step 19.

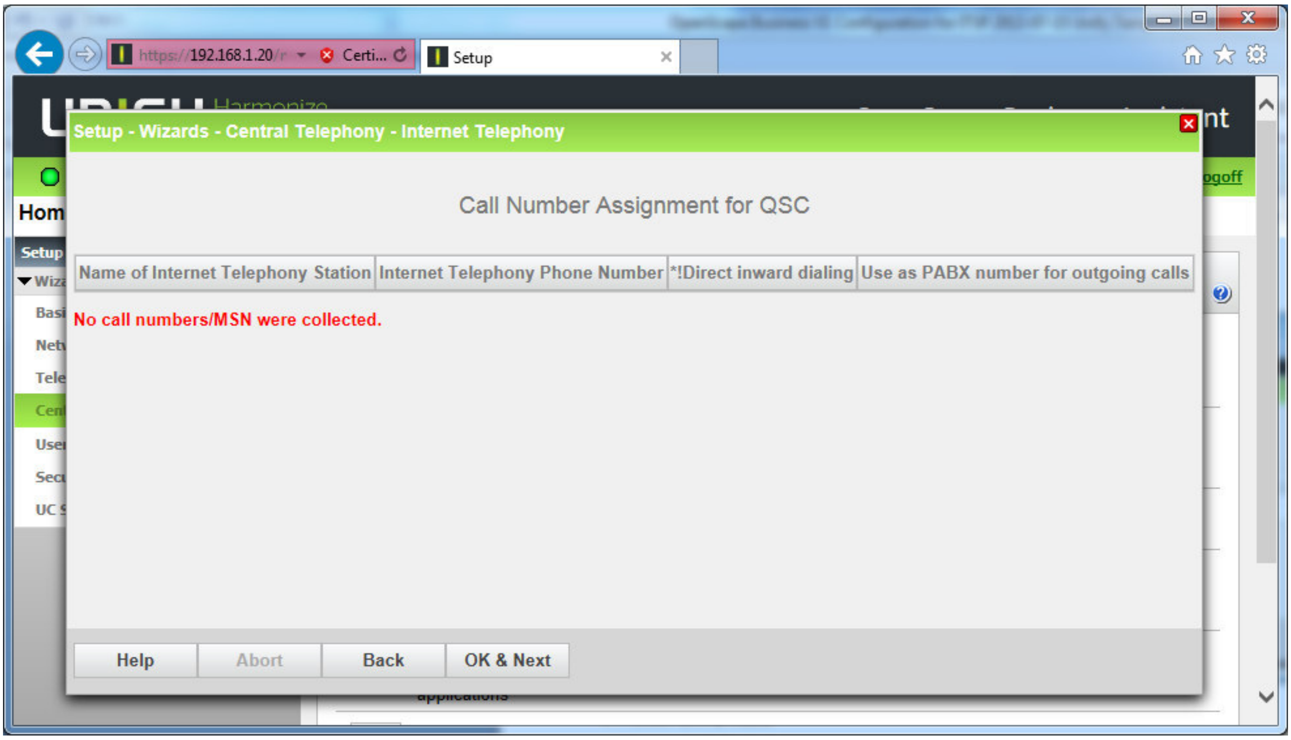
“Use public number (DID)” Mode



8. Enter the **relevant account (username) or number in Internet Telephony Station**. (if none was provided by your ITSP then enter here the pilot number of the DDI range)
9. Enter the **Authorization Name** and **Password** which was given to you for the VoIP account by your provider, if necessary
10. For subscribers without their own DID, one number can be used as **Default Number** for outgoing calls. 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. Please also note that if there is a DID Configured in the Intercept/Attendant station then this number will be used for outgoing calls from stations without their own DID. **Default Number** is used in outgoing calls when there is no DID configured in the Intercept/Attendant.



11. Click **OK & Next**

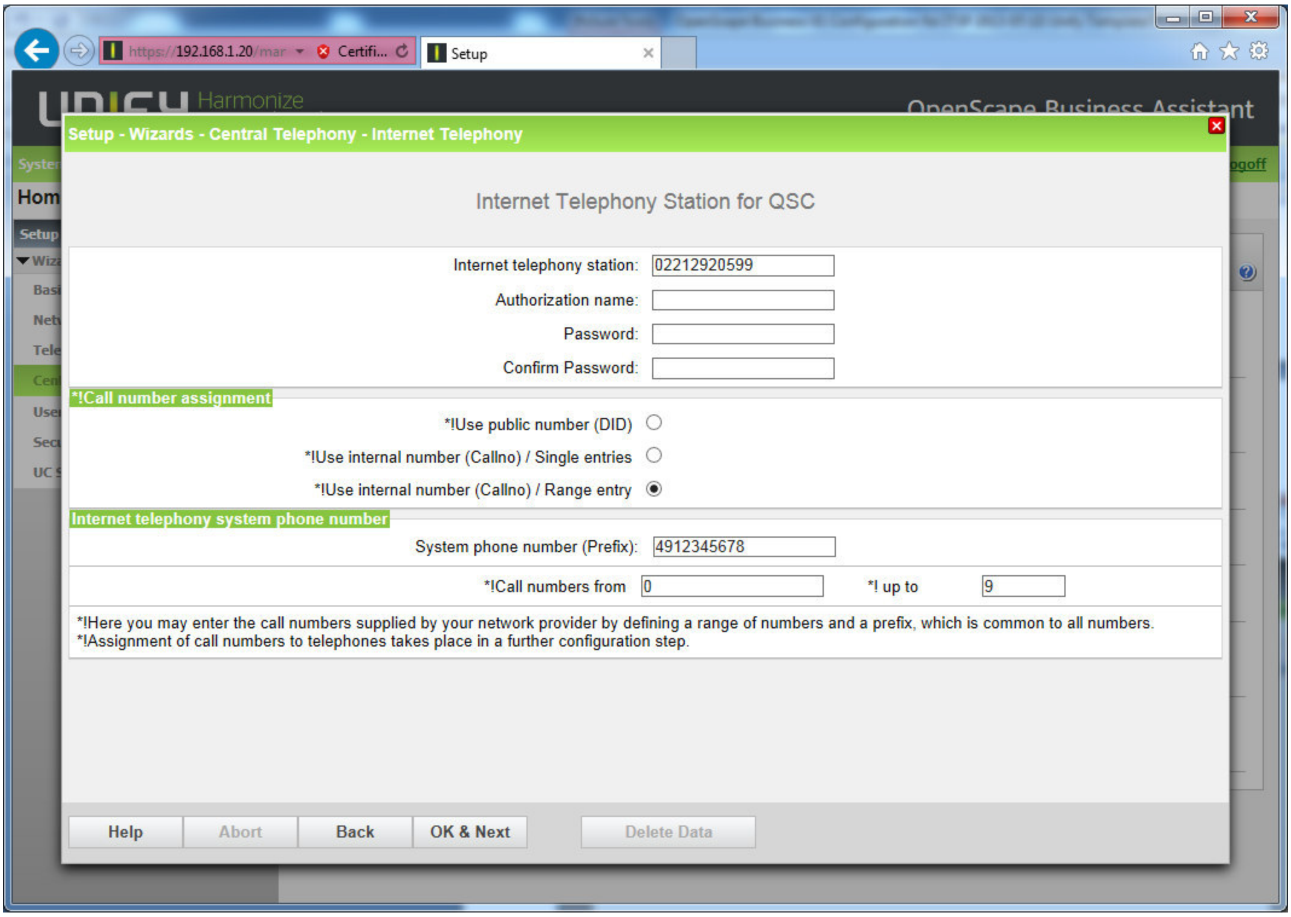


12. In this screen we get this message because there are no MSNs created since we are in the DID mode. Click **OK & Next**

Continue in step 19.

“Use internal number (Callno)” mode

If you have used the DID Mode, then please skip these steps (13-18) and go to step 19.



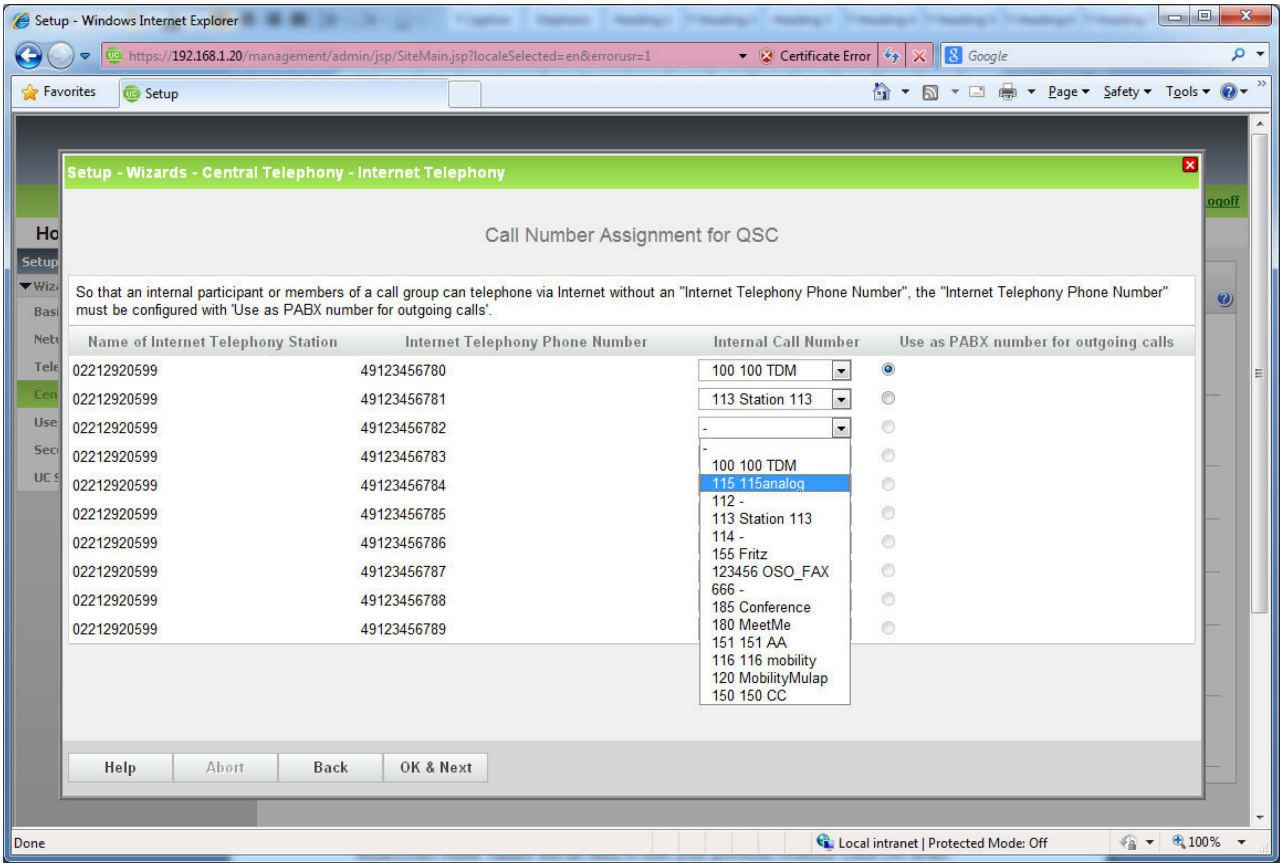
13. Enter the **relevant account (username) or number in Internet Telephony Station**. (if none was provided by your ITSP then enter here the pilot number of the DDI range)
14. Enter the **Authorization Name** and **Password** which was given to you for the VoIP account by your provider, if necessary
15. a) For Accounts with single call numbers select the option **Internet Telephony Phone Number**

    Enter the phone number and click **Add** for every phone number you received from your provider.

    b) For DDI trunks / SIP trunking select the option **Internet telephony system phone number** in the Call number type area.

    Enter **System phone number** e.g. 4912345678

    Enter the DID number range for the Internet telephony station in the 'from' and 'to' fields after **Direct inward dialing band**. The range entered by default is 100 - 147.
16. Click **OK & Next**. An overview of your ITSP providers is shown. Click OK & Next



17. Assign one **internal call number** each to all Internet telephony phone numbers. For subscribers without Internet telephony phone number one number can be selected as **PABX number for outgoing calls**.
18. Click OK & Next. An overview of your ITSP providers is shown. Click OK & Next

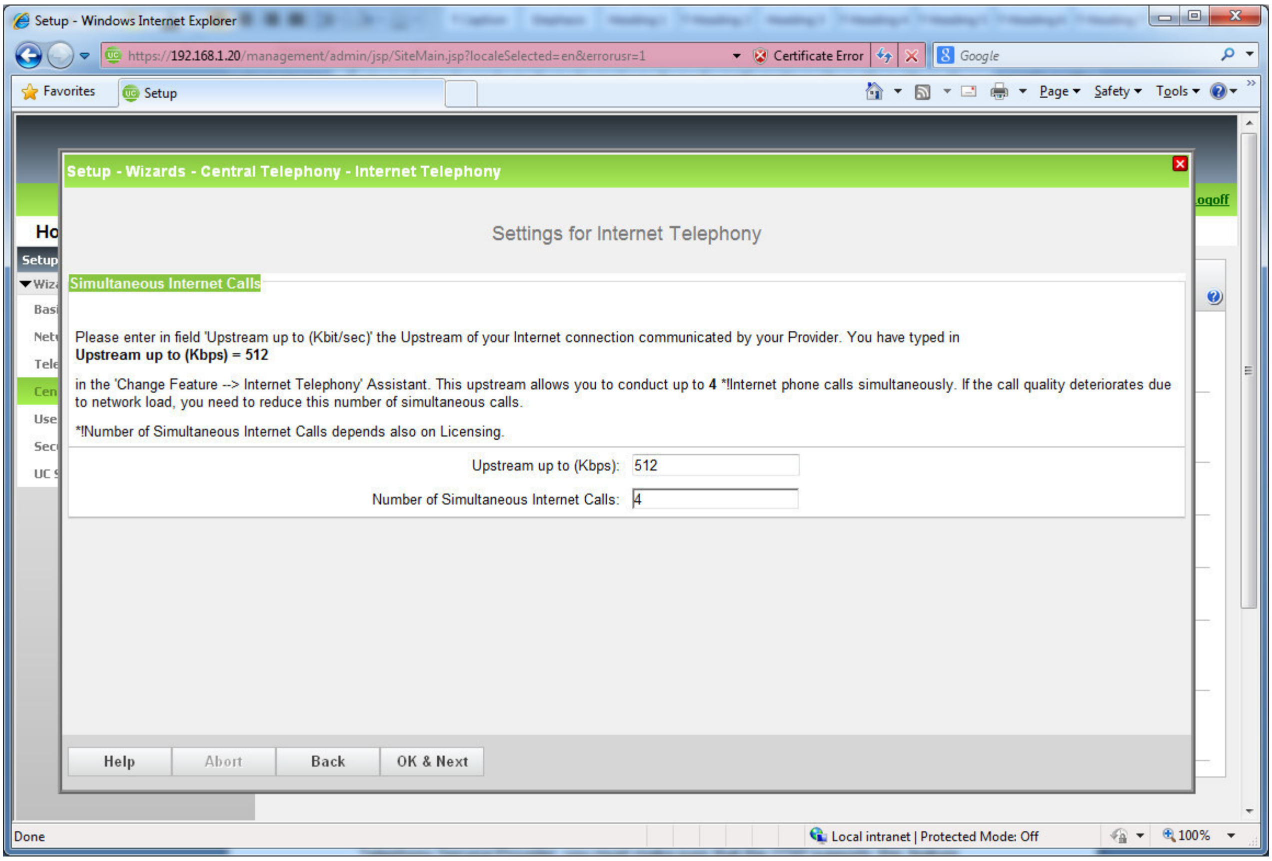
19. At this step you will have to configure the Upload Bandwidth of your internet connection.

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 upload preconfigured value derrives from the value used in the Internet Wizard.

E.g. for 512 kbps upload you can have up to 4 calls. If this is not the first time you run this wizard then the upload value will be filled in with your previous choice.

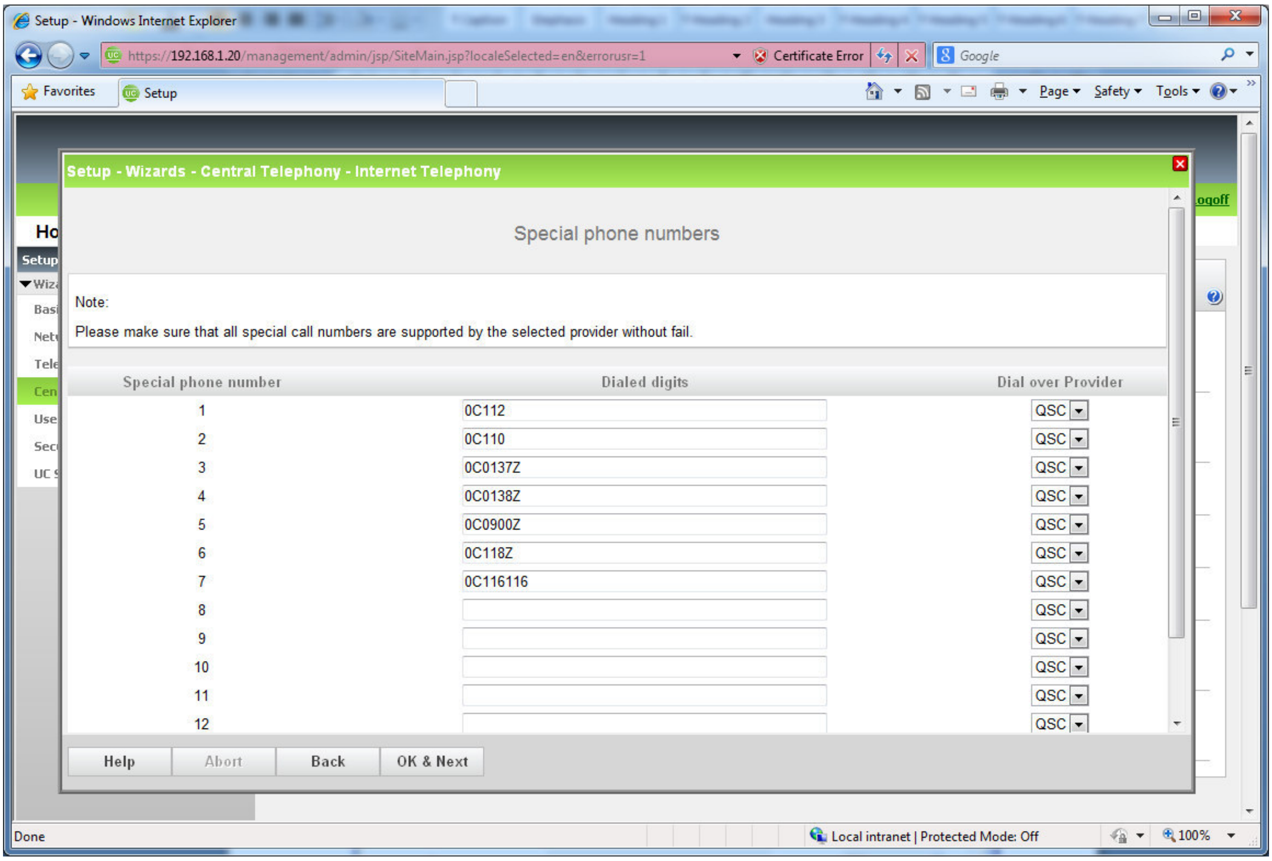
Click OK when finished.



20. Next you can define the handling of special numbers in the **Dialed digits** column.The following station number entries are valid:
    - 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)

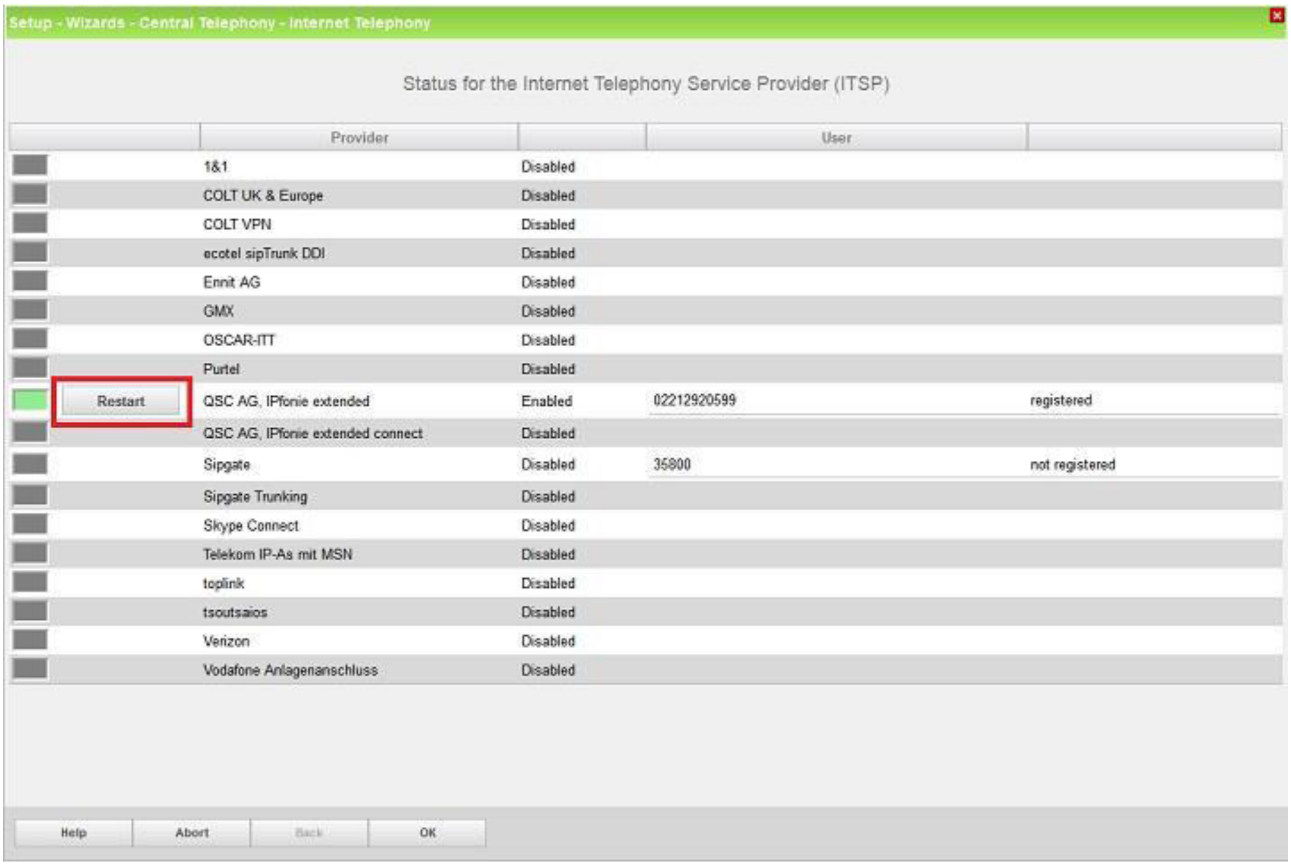
21. Use the **Dial over Provider** column to specify whether the special number should be dialled via ISDN or an ITSP. Only the active ITSP is displayed. Ensure that emergency numbers can always be dialled. If you want to dial emergency numbers via an Internet Telephony Service Provider, you must make sure that the ITSP supports this feature.

22. Click **OK & Next**.



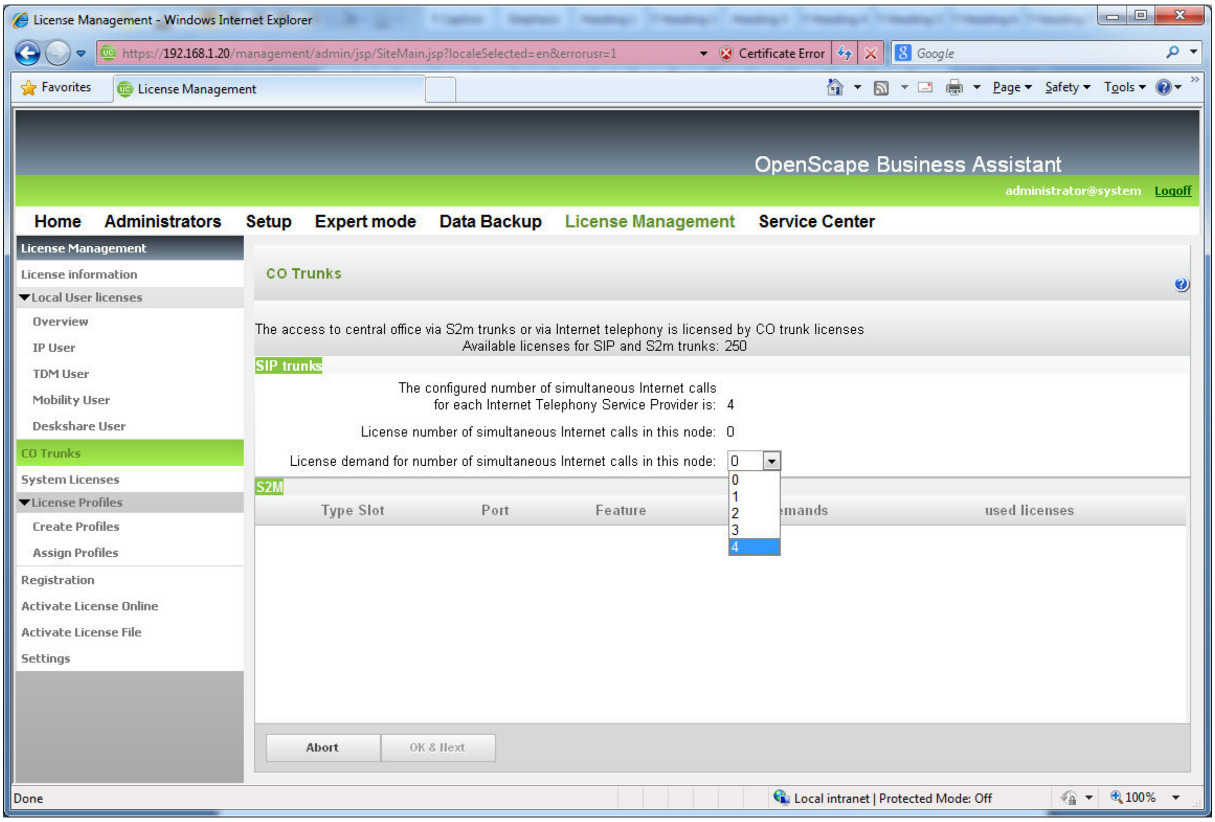
23. Here you can see the status of your ITSP. If the profile is succesfully activated then you should see the status in green color. If the status color is orange, then this means that the activation was not succesful. In this case please verify that you configured the correct credentials for your account. If the problem still exists then please check the STUN mode configuration. (For more details please see chapter 4.3)

    In this screen you can also restart your ITSP by pressing the relevant button as seen below. In case that the ITSP is using registration, this will result to a de-registration and a re-registration.



24. Click **Next** and then **Finish** to exit the **Internet Telephony** wizard.

25. 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. The licenses required for each SIP Trunk are “OpenScape Business V1 S2M/SIP Trunks”



26. It is recommended to perform a data backup by clicking on Data Backup in the navigation bar and then on Backup- Immediate in the navigation tree.

Now your ITSP is ready to use. Outgoing calls via the first configured provider can be made with default prefix 855. Further providers (up to 4) can be used via default prefixes 856, 857 and 858 respectively.

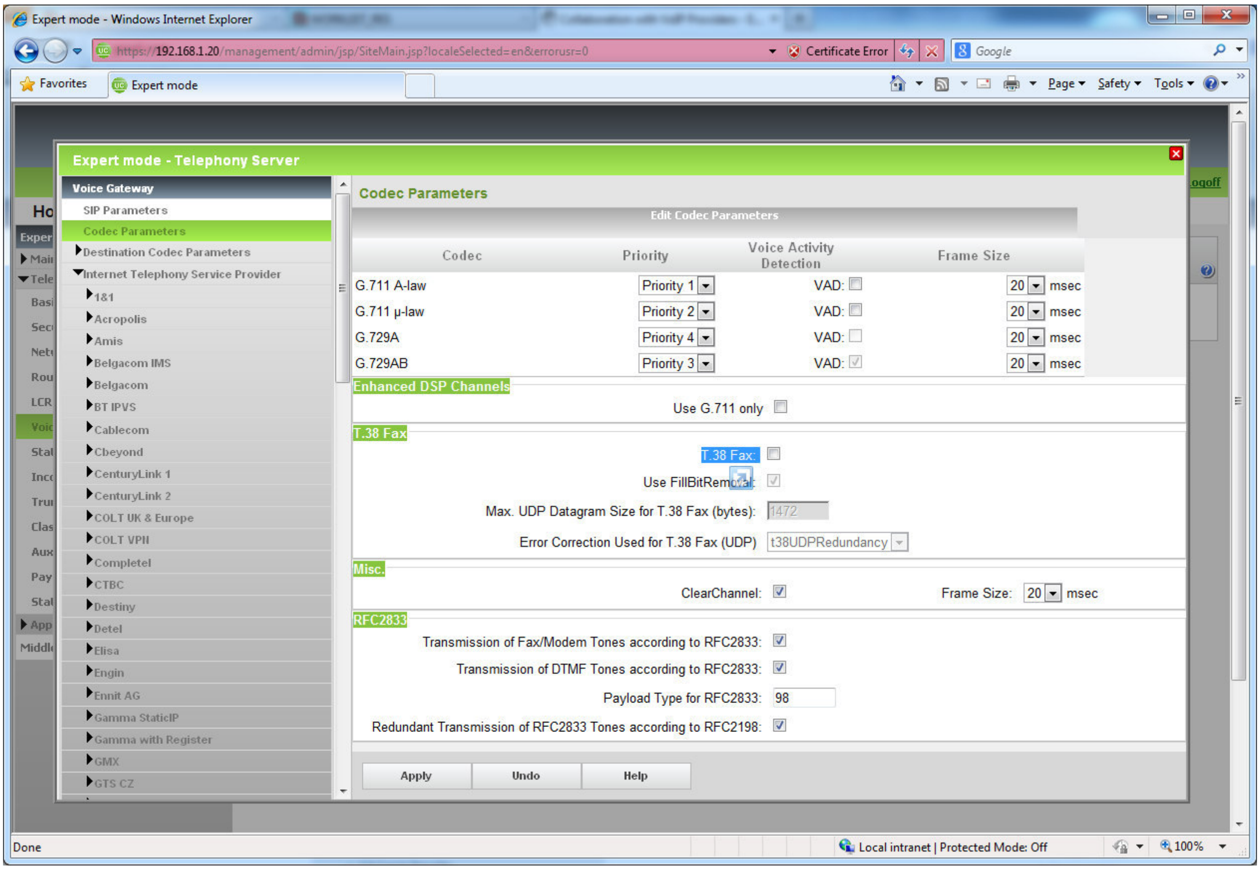
4. Appendix

4.1. Fax Setup

Fax is possible in two ways, either by protocol T.38 or by using clear 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, then T38 needs to 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.



→ It is strongly recommended to disable T.38 only if your provider does NOT support it.

→ If the ITSP does not support T.38, then fax via the OpenScape Business UC application is NOT possible.

4.2. 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"

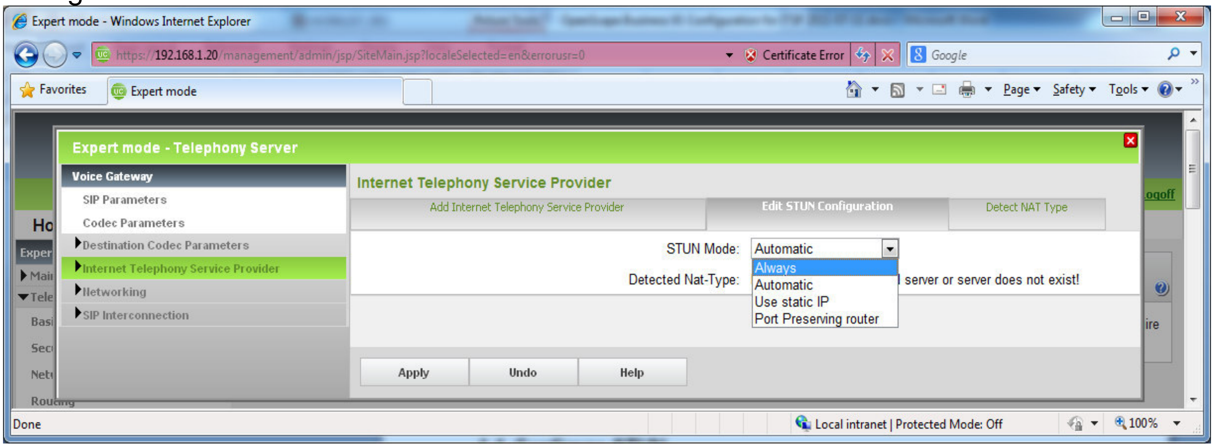
4.3. Provider Hints

Profile Settings:
This guide's goal is to present basic configuration and usage of already certified ITSPs. The preconfigured profiles already have the correct basic and advanced settings and the only thing needed is to enter the account and MSN/DID info. Please note that some ITSP profiles do not have fixed IPs/Domains for their Servers and these have to be entered manually. For more info please check the specific ITSP release notes in our official wiki page.
Nevertheless if for some reason you need to edit the basic or advanced Profile settings then you may do it in "Expert Mode > Voice Gateway > Internet Telephony Service Provider". Please note that such changes must be done by experienced personnel, otherwise they may cause malfunction of your ITSP. For more information please refer to the "VoIP Provider Data Collection.doc" at our official ITSP wiki page here.

Account Settings:
You can find configuration hints how to enter account data for a specific provider at
http://wiki.unify.com/index.php/How_to_enter_SIP_Provider_Account_Data
Please feel free to add information from your experience to this web page.

4.4. Configure STUN

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



Notes on setting STUN mode:

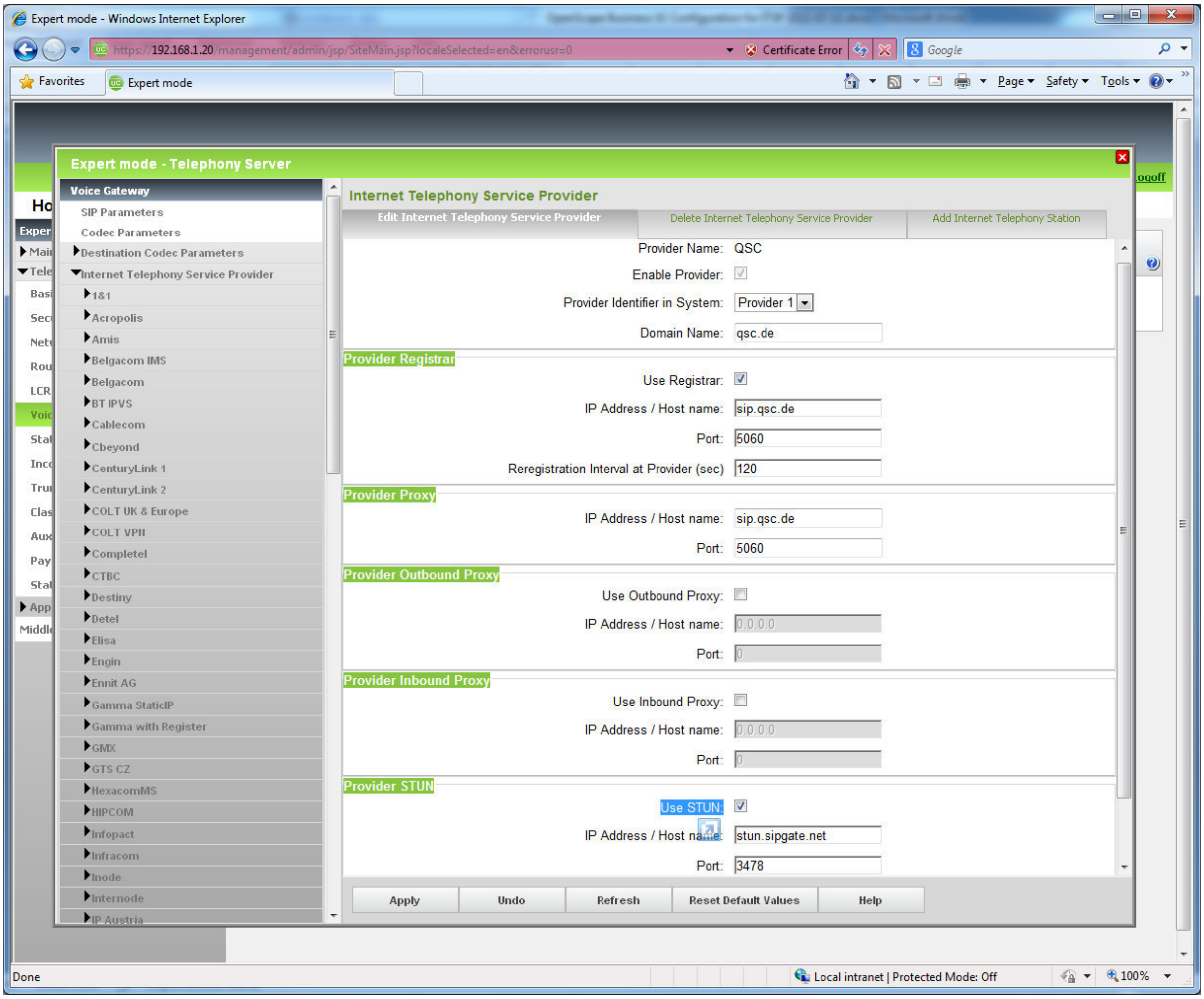
The necessary STUN mode depends on ITSP infrastructure and the used router. STUN is not required for ITSPs that resolve NAT traversal using infrastructure components in the provider network such as Session Border Controller (SBC). See also:http://wiki.unify.com/index.php/Network_Configuration_for_VoIP_Providers

- **“Automatic”(Default)** If no ITSP is active, STUN is fully disabled. With an active ITSP, it depends on the “Use STUN” Flag per ITSP Profile. If “Use STUN” flag is disabled the STUN is switched OFF. If “Use STUN” flag is enabled then STUN determines the used firewall type (NAT type) at system startup and detects IP address changes during runtime by using the configured STUN server in the ITSP profile. Depending on the detected NAT type, STUN changes certain parameters in SIP messages (NAT traversal).
  Please note: symmetric NAT is not supported.
- **“Always”** STUN is always active, even if no ITSP is active, for example.
  Depending on the detected NAT type, some parameters in SIP messages (NAT traversal) are adapted.
- **“Use static IP”** If you are using a static IP on your ADSL modem/router then use this mode and enter here the static IP and port.
- **“Port Preserving Router”** (Use this option if none of the above is working, there are some specific Modem/Routers that have a special port for NAT and need this option to work properly

Switching off STUN completely:

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

In other words, if the “Use STUN” flag is enabled, then the Global STUN Mode configuration (illustrated above) takes effect. If the flag is disabled then STUN is totally disabled.



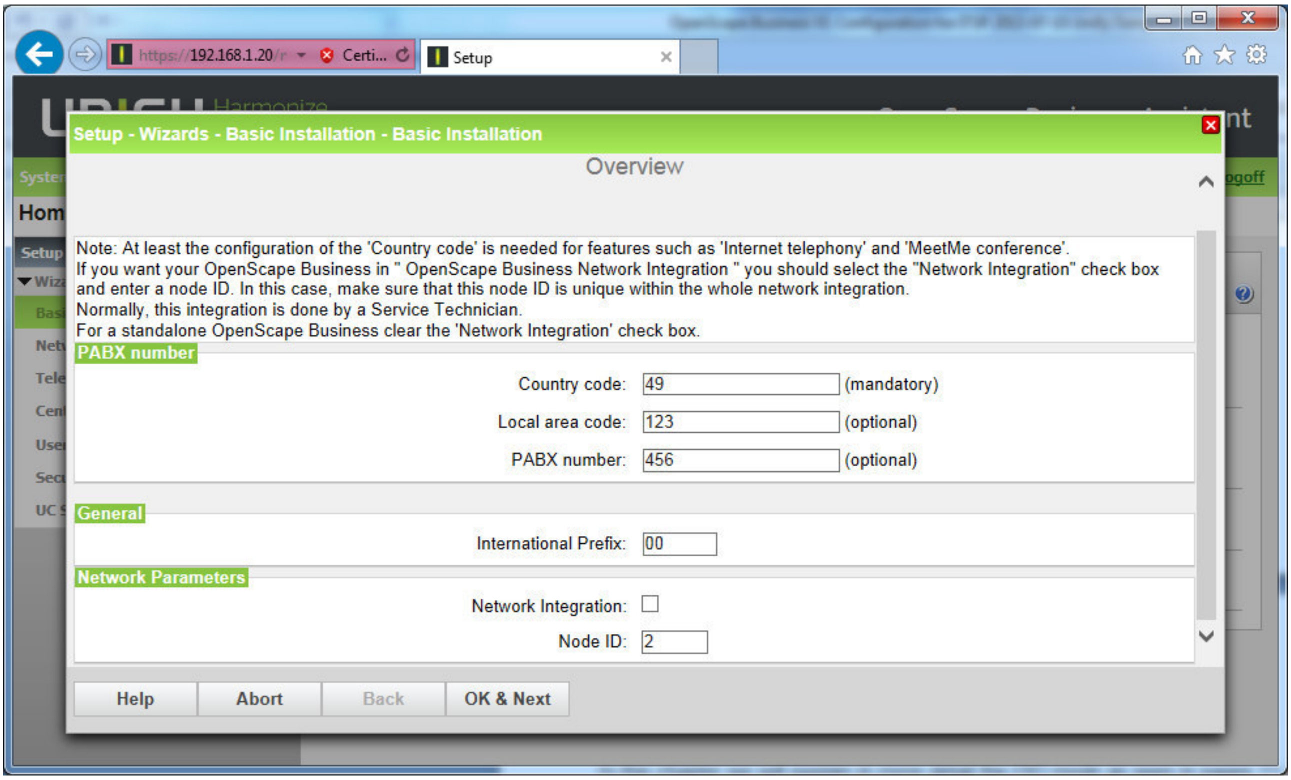
4.5. "Use public number (DID)" mode configuration examples

In this chapter we will explain in more detail the DID mode as seen in pages 11-13. As mentioned earlier, in this mode all ITSP numbers are based on the station's DID, Location data and Route settings. Just like on ISDN CO interfaces. We will use an example to make it clearer.

Let's say the ITSP has provided the following number: 0049123456780
Let's assume that:

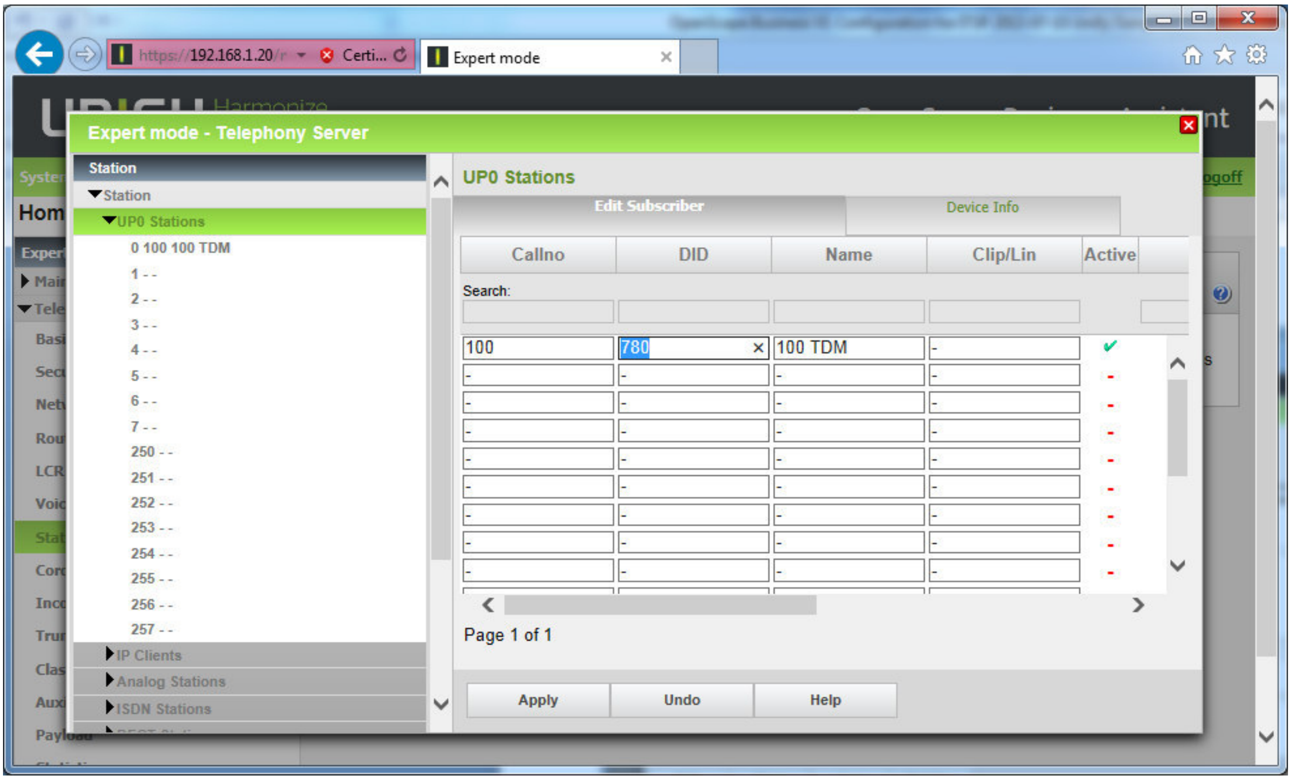
00 is international prefix
49 is the Country code
123 is the Local Area Code
456 is the PABX number
780 is the DID.

ITSP Route should always be the Gateway Location and we should have the following settings:



The same values should be used in the PABX number-incoming, in the ITSP Route in expert mode.

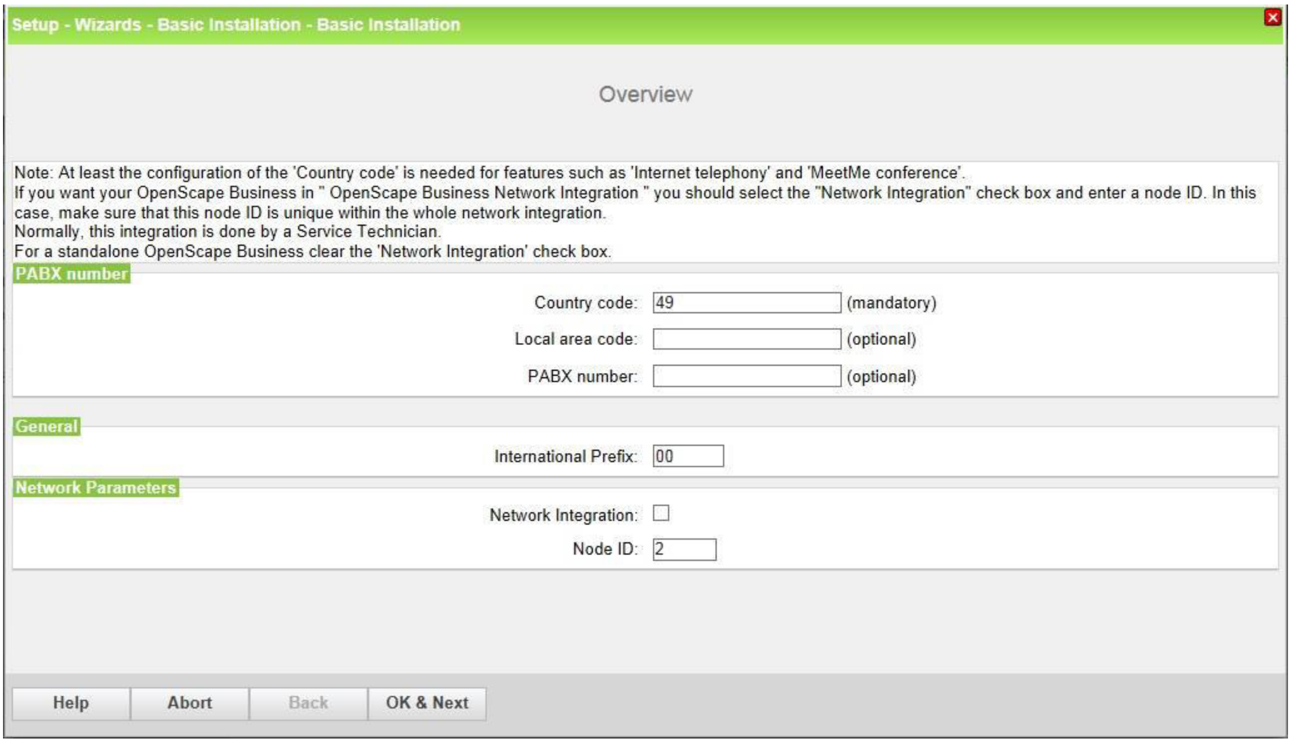
The next step is to set the DID (780) in the relevant station's field:

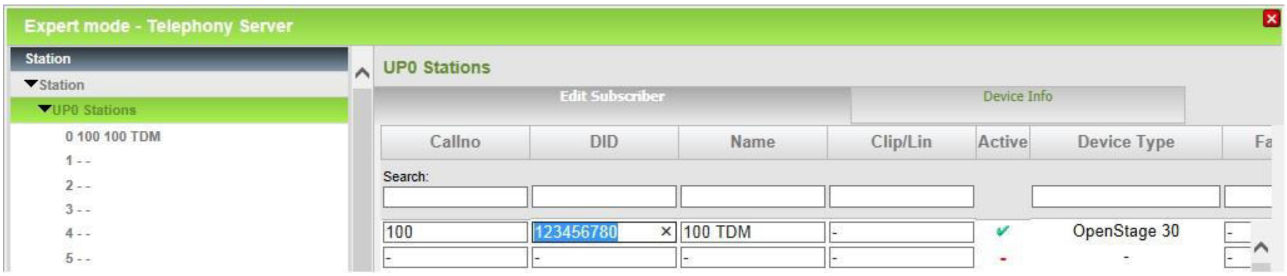


Now our number is assigned to internal station 100. The final format of the number that will be used (in the SIP headers) depends also on the routing parameters. We may use 0049123456780 or 49123456780 or +49123456780 or 0123456780 depending on the routing parameters. In our case if we want to use 004949123456780 then the "No and type outgoing" should be set to "International" and also "Call No. with International / National prefix" flag should be activated. We will explain below all formats and needed configuration.

If you have a DDI range from the ITSP, e.g. 0049123456780-9 then you should repeat the same steps for every number (e.g. set DID 781 to internal 101 etc.)

Alternatively we could make also various assumptions and combinations and get exactly the same result e.g.:





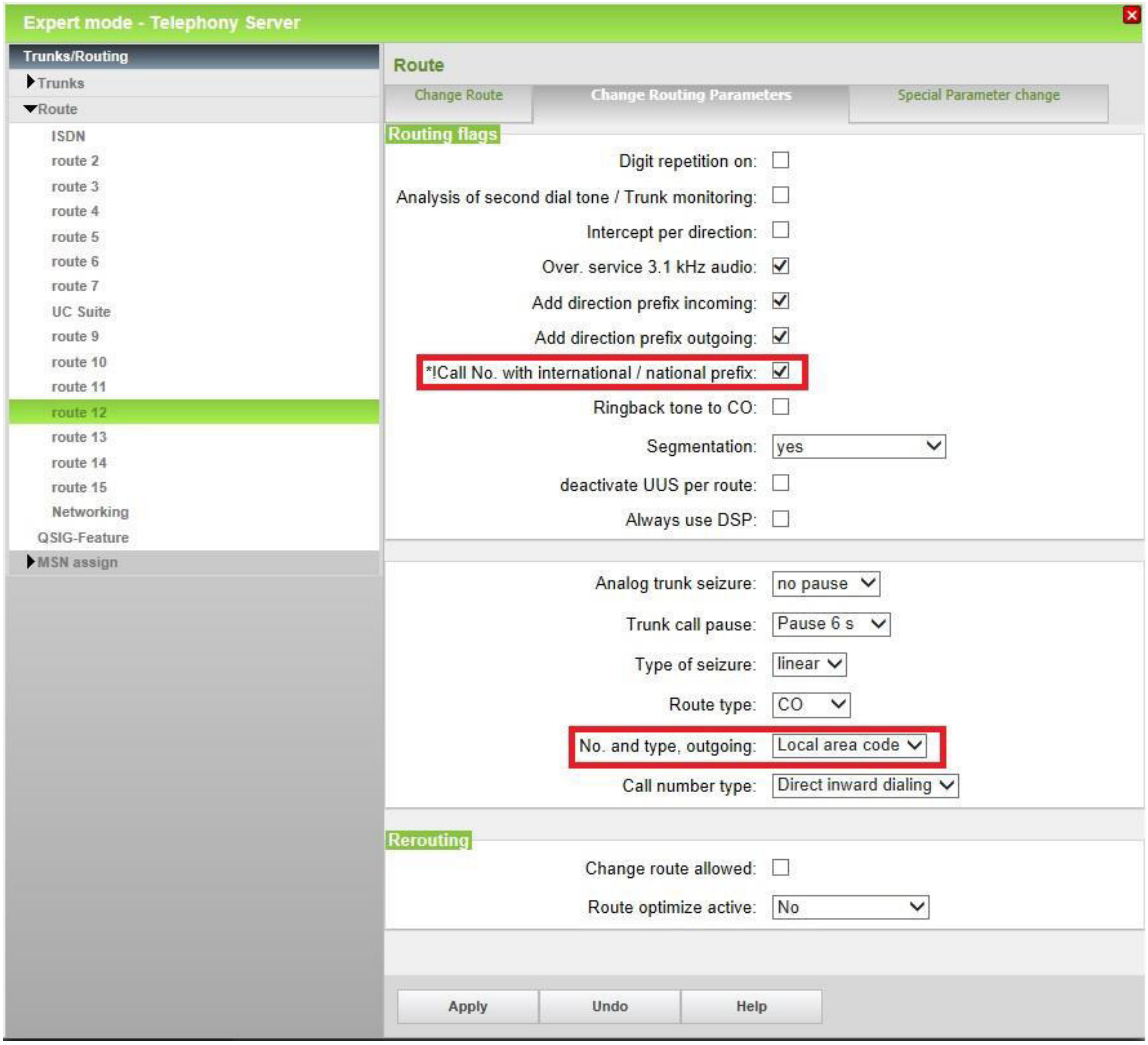
00 is international prefix
49 is the Country code
123456780 is the DID
In the ITSP Route (Gateway Location) we have the following settings:
Country Code 49, Local Area Code and PABX number left empty.
Then by adding 123456780 in the DID field of station 100, we could get the same result as before.

The bottom line is that we may have a specific number from the ITSP, but we can “create” the same number based on various combinations depending on the system configuration every time.

Routing Parameters:

As mentioned above the call number format that will be used depends also on the routing parameters (exactly the same way as on ISDN trunks). In the example above, the ITSP number 0049123456780 was created, Country Code is configured but if the "No and type outgoing" is set to Local Area Code, then the country code will be omitted. In our example we would have 123456780 (or 0123456780 if 0 is set as national prefix).

When activating a preconfigured ITSP profile, these Routing parameters are set automatically depending on specific ITSP advanced profile settings that were determined during certification process. Therefore nothing special should be done if using a preconfigured profile. In any case if the number format needs to be changed e.g. if you are using a custom ITSP profile during ITSP certification, then these settings have to be configured manually.
The routing parameters that affect the final format of the number are marked in below image:



Remark:

When we are in DID mode the "Call number type:" is always set to "Direct inward dialing".

In the following table we have all the call number formats and relevant values for these settings based on the first example above.

Outgoing Number Format	No. and type outgoing	Call No. with international / national prefix
International dialable 0049123456780	Country code	Enabled
International canonical +49123456780*	Country code	Disabled
Implicitly international 49123456780**	Country code	Disabled
National dialable 0123456780	Local area code	Enabled
National without prefix 123456780	Local area code	Disabled

* This configuration will affect the originator field (FROM and PAI/PPI headers) in SIP headers. If you want to use canonical format (+49xxx) also in the destination field (To: header) then additional configuration is needed in LCR. Based on our example above we must make the following changes in some already preconfigured Dial Plans and Dial rules:

LCR Dial Plan 16 = 0C0-Z
LCR Dial Plan 34 = 0C00-Z
LCR Dial rule 2 is set to e.g. D49E3A type international
LCR Dial rule 3 is set to e.g. D49123E2A type international
LCR Dial rule 8 is set to E3A type international

**Only for the special case of Implicitly International format 49123456780, an additional parameter in the ITSP profile must be configured: Outgoing call – Type of Number (calling): = International

Additional Notes for the DID Mode:

- DID mode is available since OpenScape Business V1R3.
- One ITSP access (only) is supported with this kind of connectivity (DID Mode). This interface defines the gateway location for the system.
- Additional ITSP accesses may be configured in the same manner (DID based, no mapping) only, if collisions regarding the DID numbering scheme can be avoided.
- Additional ITSP accesses based on internal call number (Callno mapping mode) may be used in the same node.
- For central ITSP access in a network scenario, only one node should be configured in DID mode. Local CO lines may be used but only for outgoing calls (e.g. emergency calls). LCR configuration should be configured accordingly in all nodes in the same way as in central CO/ISDN access in networking scenarios.
- Starting with V1R3 this configuration is mandatory when CLIP shall be used. The CLIP configuration based on mapping mode as described in Wiki can only be used in older SW versions. Reconfiguration is required, when upgrading such a system to V1R3 (see 4.6.)

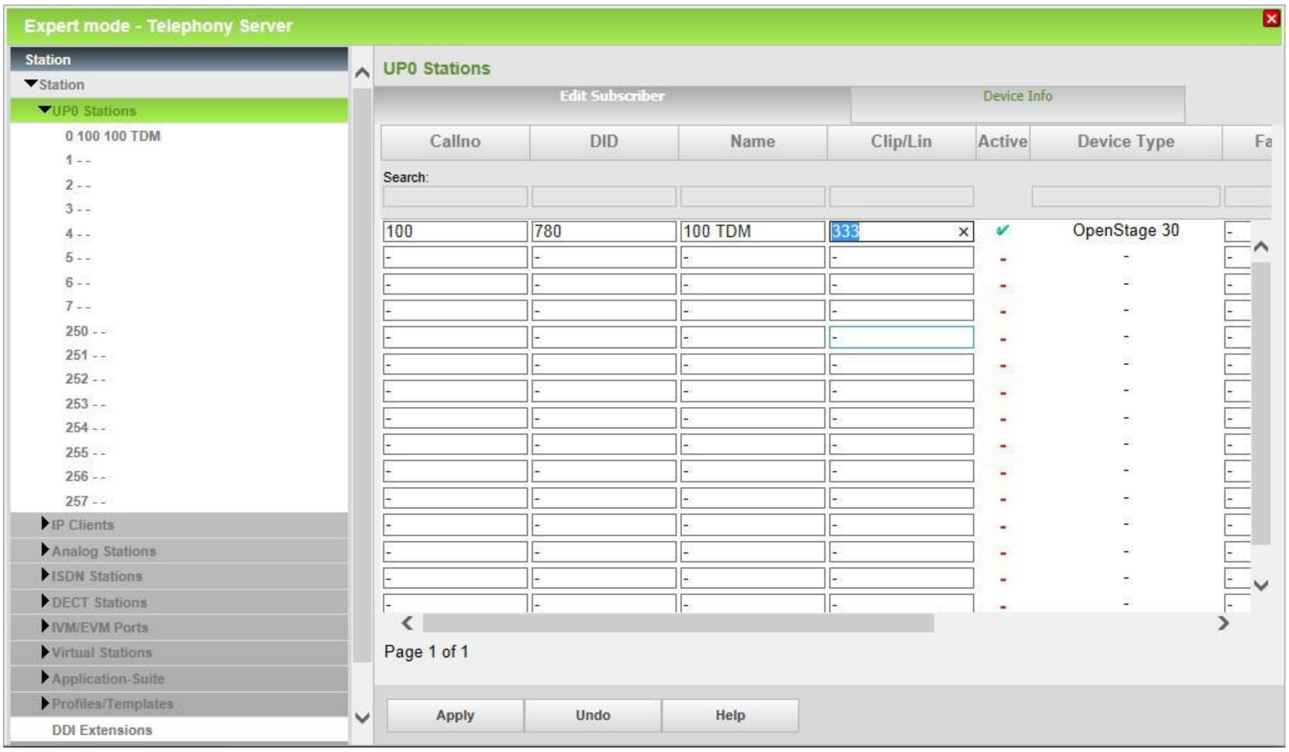
For further details on the SIP headers Formats please refer to the VoIP Provider Data Collection doc (available in wiki) or contact the SIP Certification team.

4.6. Clip / Lin

It is possible to use a different DID or a dummy entry in outgoing ITSP calls. This is possible by editing the Clip / Lin field in the station configuration. This functionality is available since OpenScape Business V1R3 and an essential precondition is that the ITSP must be configured in DID Mode (refer to appendix 4.5 for more details).

As described in the first example in 4.5, let's assume that we have the ITSP Number 0049123456780. The DID 780 is assigned to internal station 100. This will result to an outgoing call that will have 0049123456780 in FROM and PAI/PPI (if used) headers in outgoing INVITE.

If we want to have a different DID (e.g. 333) in these headers, then this possible by entering this value in the Clip / Lin field as seen in below screenshot:



This will result in an outgoing INVITE containing 0049123456**333** in FROM and PAI/PPI headers.

- Please note that this has to be supported by the ITSP, otherwise the ITSP may not authenticate the call and reject it.

CLIP no Screening

Additionally we may use a different flavour of this functionality also known as "CLIP no Screening". By activating the relevant ITSP profile flag (*"CLIP no Screening support"*), we can achieve to have the Clip/Lin in FROM field and the "Default Number" in PAI header. For more details about this functionality please refer to the "VoIP Provider Data Collection.doc" (available in wiki).

4.7. Mobile Extension (MEX) Connectivity via ITSP

The main concept of the feature is that the MEX device can be used as an internal station over the GSM network. MEX device can make calls to other internal OSBiz stations and also dial out via the ITSP trunk. MEX terminated calls will be handled as standard mobility calls.

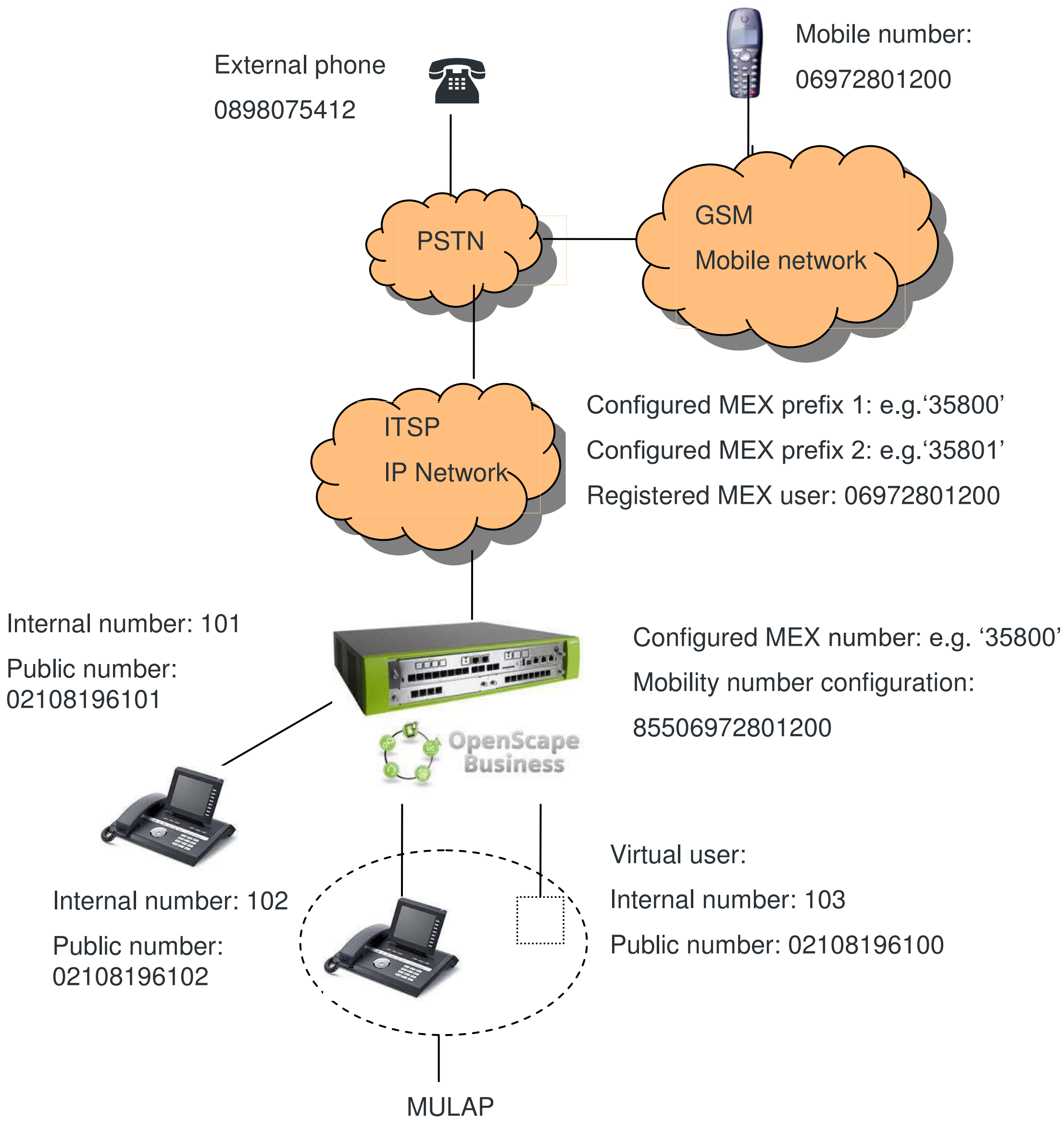
OSBiz supports this feature for ITSP trunks only. SIP MEX implementation is based on the existing Mobility feature. Therefore the conditions and relations to other features are identical to mobility, if not otherwise mentioned.

Preconditions:

- SIP MEX feature must be supported by the relevant ITSP/GSM Carrier.
- ITSP must be configured in public number / DID mode (see chapters 3 and 4.5)
- LCR must be activated
- ITSP must use dialable number format. In other words including international/national prefixes (e.g. 0049xxx / 089xxx). E.164, implicit international/national formats are not supported (e.g. +49xxx, 49xxx, 89xxx are not supported).
- Since the feature is based on Mobility, a mobility license is required for every SIP MEX user.
- A DISA number must be configured
- ITSP access and configured mobility extension must be located at the same node.

In the following pages a detailed configuration example is available.

Scenario Example:



OpenScape Business Configuration:

Based on above example image:

ITSP is using national dialable format

0 is the national prefix

855 is the ITSP LCR prefix (by default)

35800 is the MEX prefix

The following use cases will be supported:

Example scenario 1: MEX originated call:

a) MEX user (mobile phone) calls internal station 102 by dialling 102. ITSP will route the call to OSBiz with the following info:

Incoming INVITE:
FROM: 06972801200
TO: 35800102

OSBiz will recognize that this is a MEX call from the number that is configured as mobility user. The destination number 102 will be extracted from TO and then handling will be mapped to feature ‘Call Through’ so that a call to the internal station 102 is initiated.

b) MEX user (mobile phone) calls external phone by dialling 8550898075412. ITSP will route the call to OSBiz with the following info:

Incoming INVITE:
FROM: 06972801200
TO: 358008550898075412

OSBiz will recognize that this is a MEX call from the number that is configured as mobility user. The destination number 8550898075412 will be extracted from the TO and then handling will be mapped to feature ‘Call Through’ so that a call to 0898075412 is initiated over the ITSP trunk. In the outgoing INVITE the originator will be the public number of the MULAP (or the virtual station’s that is binded with Mobility-if no MULAP is set).

This example described operating mode “Dialling external numbers with seizure code”. In other words the mobile user will dial also the ITSP seizure code 855 (or the ITSP will send it in TO field). Therefore the system flag “Add Seizure code for MEX” must be deactivated. If the incoming INVITE will not include 855 in the TO header (user will not dial 855 or the ITSP will not send it) then OSBiz can add it automatically. In that case system flag “Add Seizure code for MEX” must be activated.

The algorithm to differentiate between internal and external destinations when flag ‘Add seizure code for MEX’ is activated is the number of digits. If destination is >7 digits then number is assumed as external, thus seizure code is added. If destination is <=7 digits then it is considered as internal destination.

As the only indication for the algorithm to differentiate between internal and external destinations in the operating mode ‘add seizure code for MEX’ (= dialing external numbers without seizure code) is the number of digits, the functionality depends on the numbering scheme of the node /network and this mechanism can activated / deactivated by WBM configuration. It is only suitable in networks with internal numbers with a maximum length of 7 digits and must be deactivated, when this condition is not met.

Example scenario 2: MEX terminated call (to fixed network number):

External phone calls MEX user by dialling the public number of the virtual user of the OSBiz. The incoming INVITE from ITSP will contain:

FROM: 0892108075412
TO: 02108196100

The further handling is the processing of a call to a mobility user as it is already implemented so that a call to mobile 06972801200 is initiated (via ITSP trunk). In the outgoing INVITE the originator will be the public number of the virtual station (02108196100). * Please note that if the ITSP is configured to display the original caller in transit scenarios, then the FROM field will have 0892108075412

Example scenario 3: MEX terminated call (to mobile number):

This is not a typical scenario. Some ITSP's require that the MEX mobile number is accessible with second MEX prefix. This second MEX prefix can be configured only in Expert Mode and is not available in wizards.

External phone 0892108075412 calls MEX mobile phone 06972801200. ITSP will route the call to OSBiz. The TO field will/may be equipped with an additional prefix2 (e.g. 35801). The incoming INVITE will contain:

FROM: 0892108075412
TO: 3580106972801200

The destination will be mapped to the MEX mobile number using the resp. ITSP tables (see Voice Gateway configuration). In other words even if the ITSP is set to operate in public number / DID mode, only for this case a manual MSN entry must be created in ITSP profile in Voice Gateway with following info: MSN: 3580106972801200 mapped to internal virtual station 103. The further handling is the processing of a call to a mobility user as it is already implemented; a call to MEX mobile 06972801200 is initiated.

Remark: Some providers may use a leading prefix 3. This is not supported in OSBiz.

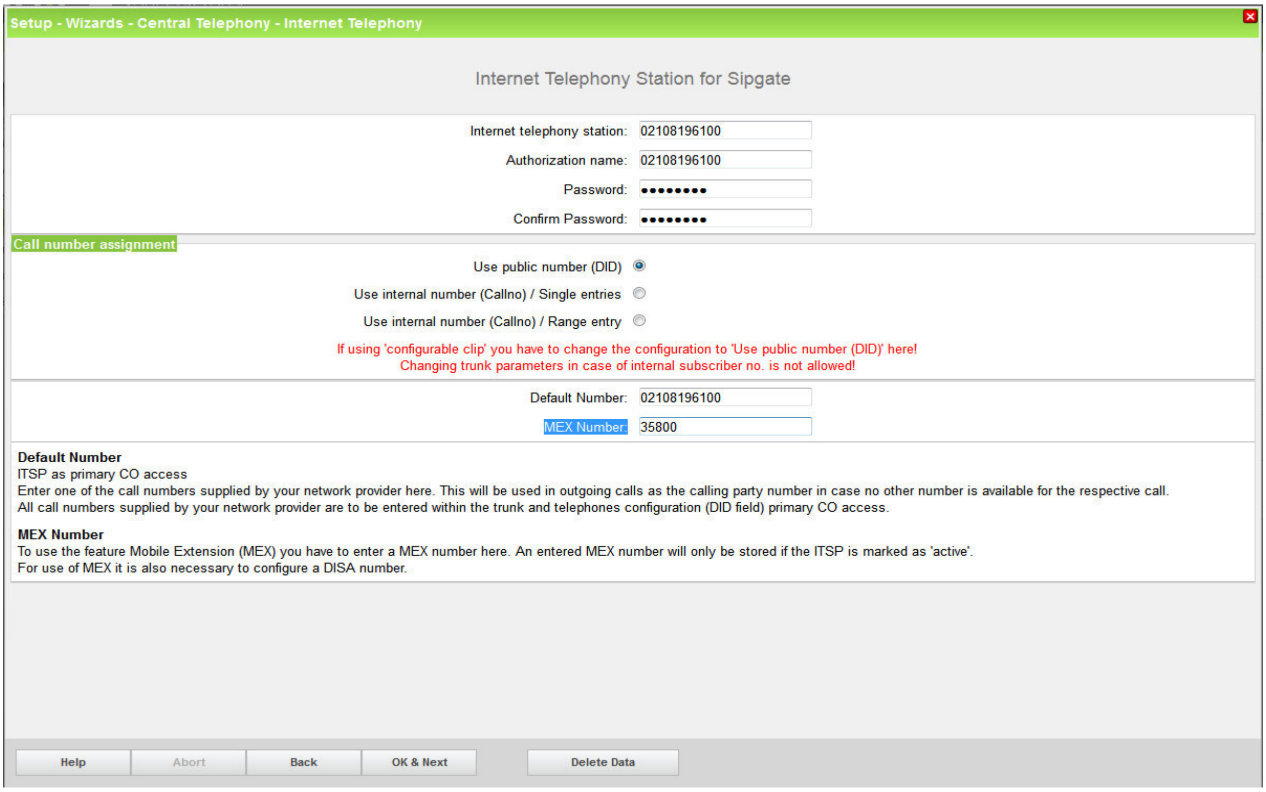
Additional info:

- In OSBiz Embedded features in call state (e.g. consultation) can be activated with DTMF dialling (standard use of feature 'Mobility'). Here '*' and '#' can be used or the substitution codes for '*' and '#'. The configuration as for standard use of feature 'Mobility' is necessary i.e. station orientated bit 'DTMF based feature activation' has to be set in WBM.
- For the virtual user a MULAP or a single configuration can be used (as standard use of feature 'Mobility')
- For emergency numbers (e.g. 110 and 112) respective LCR rules have to be defined (e.g. '11' -> …). Here collisions with the call numbers of the systems have to be avoided (no call numbers should begin with 11).

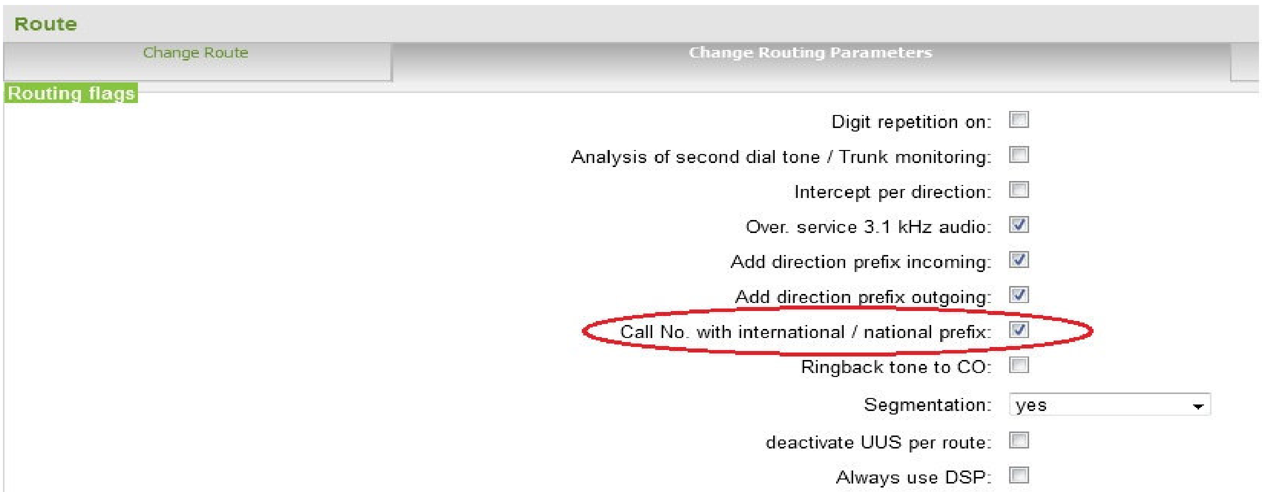
How to Configure SIP MEX

Since MEX feature is based on Mobility feature, then a Mobility user must be configured over the ITSP trunk. The same applies also for DISA number (please refer to official documentation for Mobility for further information and configuration hints).

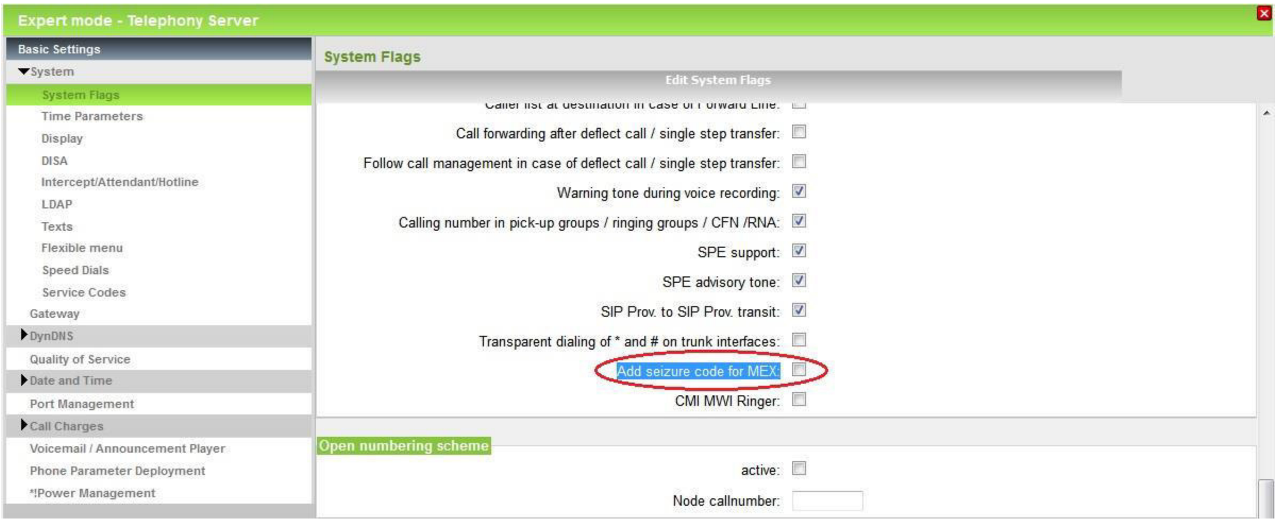
MEX prefix is configured in ITSP wizards in Internet Telephony Station (see image below). As already mentioned, ITSP must be configured in “Public Number (DID)” mode. Please note that the ITSP must be have been activated in order to add the MEX prefix or activate the enable ITSP checkbox when creating the profile for the first time. If the ITSP is not activated, then the MEX number field will not be editable. MEX prefix can be up to 8 digits.



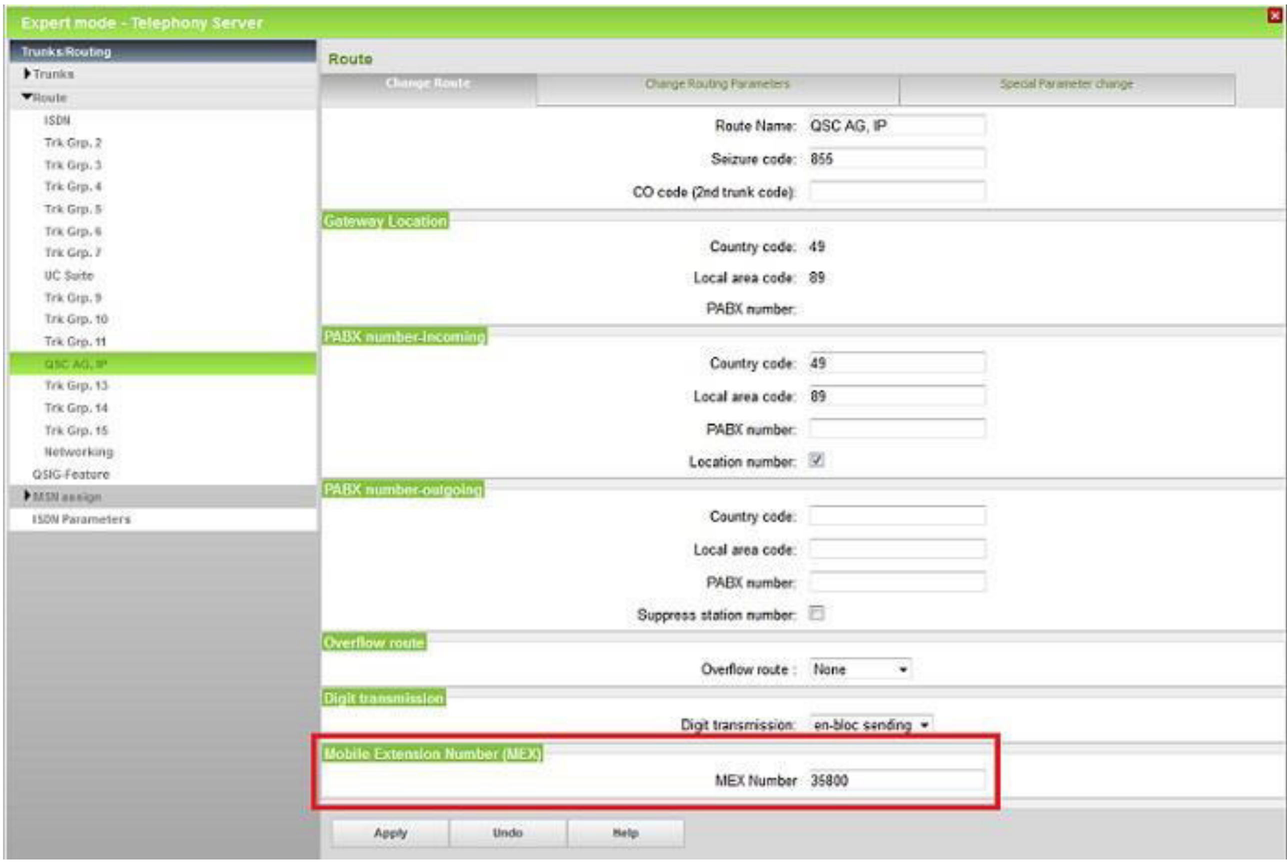
Please keep in mind that the ITSP must be configured to use dialable international/national format. Therefore flag “Call No. with international / national prefix“ must be activated in ITSP Route > Routing Parameters (for more info please refer to the “use Public Number DID” Mode chapter).



Depending on the dialling behaviour the flag “Add seizure code for MEX” must be configured accordingly (see example 1b). Default value is “false”, so the user has to dial external destinations with leading seizure code (e.g. “0” or “9”). If it shall be possible to dial external destinations without seizure code then this flag must be activated. This flag is located in System Flags:



MEX number that was set in the wizard is stored under the ITSP Route. You may find/edit it in Expert Mode > Trunks Routing:



Now the system is ready to work with MEX prefix as described in configuration examples.

About Unify

Unify is one of the world's leading communications software and services firms, providing integrated communications solutions for approximately 75 percent of the Fortune Global 500. Our solutions unify multiple networks, devices and applications into one easy-to-use platform that allows teams to engage in rich and meaningful conversations. The result is a transformation of how the enterprise communicates and collaborates that amplifies collective effort, energizes the business, and enhances business performance. Unify has a strong heritage of product reliability, innovation, open standards and security.

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