2N® NetStar

Communication System

Manual NS Admin

Version 3.1.0

www.2n.cz

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The product family developed by 2N TELEKOMUNIKACE a.s. includes GSM gateways, private branch exchanges (PBX), and door and lift communicators. 2N TELEKOMUNIKACE a.s. has been ranked among the Czech top companies for years and represented a symbol of stability and prosperity on the telecommunications market for almost two decades. At present, we export our products into over 120 countries worldwide and have exclusive distributors on all continents.

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2N TELEKOMUNIKACE hereby declares that the 2N® NetStar product complies with all basic requirements and other relevant provisions of the 1999/5/EC directive. For the full wording of the Declaration of Conformity see the CD-ROM enclosed and at www.2n.cz.

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1. Introduction

Here is what you can find in this chapter:

- 1.1 About Help
- 1.2 About Application
- 1.3 Connecting to PBX
- 1.4 Configuration Menu
- 1.5 PBX Activation

1.1 About Help

Document format

Document version: 3.1.0

This document serves as a Help and Manual for the configuration of the 2N NetStar communication system by the NSAdmin program. 2N reserves the right to modifications.

Conventions

The following fonts are used in the text:

#Hypertext

- Hyperlink to another place in the document or outside the document.

#Document contents item

- All page headings are listed in the table of content. Clicking an item in the table of content will display the respective part of the page.

Part

Headings of the individual page topics listed in the table of contents. You can click this heading and jump to the top (table of contents) of the page where other topics can be selected.

Important words

- **The text used in NSAdmin and important words in the text we want to emphasise are highlighted.**

HW – Rack

- This form is used instead of a verbal description of the location of an item in the TreeView (tree in the left-hand part of the NsAdmin window). It means that in the TreeView you need to click the HW item and then the Rack item in it.

1.2 About Application

About Application

NSAdmin is a configuration tool that helps configure the **2N® NetStar** communication system, version 2. The application is designed for an x86 platform using the WINDOWS operating system connected with **2N® NetStar** through a network. It is controlled by a mouse and, secondarily, a keyboard. NSAdmin uses the TCP connection or modem and communicates with **2N® NetStar** basically via port 6992.

As a necessary condition for using this configuration tool under the Windows XP OS, a service pack 2 and Framework v.3. have been installed. The program does not work without these components.

Main menu of the configuration tool

Once the configuration tool has been started, a window is displayed helping to configure connections to PBXs, analyse traces and start up the Help. The main menu offers the following options:

- **Admin**
  - **Settings** – here open a dialogue with the global configuration tool settings.
  - **Language** – here choose one of the supported languages.
  - **Exit** – push this key to exit the configuration tool.
- **Trace** – operations with previously saved traces are only available here at the current stage.
  - **Load trace from file**
  - **Add trace from file**
  - **Trace analyser** – push to open a trace analysis window.
- **Help** – here start the Help in the chosen language.

Global settings of the configuration tool

This dialogue includes three tags with the following parameters:

- **XML script**
  - **Type of debugging** – use this section to enable and define the range of displaying the xml trace in the configuration tool. The PBX trace cannot be set here.
  - **Indent size** – use this parameter to define the indent size for the xml trace.
- **View** – this is an auxiliary function for trace analysis using database.
  - **Show current window name** – check this option to display the current window name in the right-hand bottom corner.
  - **Show object IDs** – check this option to display the current object Id in the right-hand bottom corner. This is an easy way how to know e.g. the

port Id and find the port in the trace.

- **General**
  - **Set colours of virtual ports** – here define the colour for each virtual port or disable this function.
  - **Set colours of tables** – here define the background colours for tables or disable this function.
  - **Set colours of logins** – here define the colour for each user login or disable this function.
  - **Set colours of extensions** – define the colour for each extension (SIP, external, etc.) or disable this function.

Click on **Default** in each of the **General** subtags to set the default colours.

- **Advanced** – Use the **Ask before delete if object is used in configuration** parameter to enable/disable warning display before removing the router or any of the routing objects from configuration.

Login

- **Copy devices structure from old version storage** – Disable/enable copying of the the PBX structure(s) from earlier configuration tool versions. Once copied, the selected PBXs can be removed by deleting only (not by unselecting).

To confirm a configuration setting, use the **OK** button and to exit the dialogue without saving, push the **Cancel** button.

1.3 Connecting to PBX

Icons of connection section

The figure below presents all icons of this section.

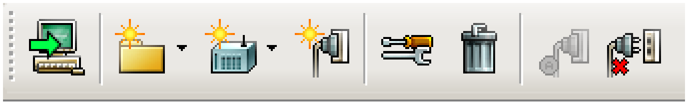


Figure: View of PBX Login Icons

Connect to PBX – use this icon to connect the configuration tool to the PBX via a selected connection.Icon meanings from the left:

- **Create group** – use this option to create a group of PBXs on the same level as the selected object or nested into the existing group.
- **Create PbX** – use this option to create a PBX on the same level as the selected object or nested into the existing group.
- **Create connection** – use this option to create a connection to the selected PBX.
- **Properties** – here set or change the properties of a selected object. A name is only assigned to groups. For details on PBX and connection settings see below.
- **Delete** – select this option to delete a group, PBX or connection.
- **Auto login** – use this option to enable an automatic PBX connection via the current connection after starting the configuration tool. One automatic connection may only be active at a time. By selecting another setting you cancel the previous one.
- **Cancel auto login** – use this option to cancel the automatic connection without specifying any object.

In addition, the following options are available in the context menu:

- **Import PBX structure** – select this option to import a predefined PBX structure as described below.
- **Export PBX structure** – select this option to export the current PBX structure for later PC connection use.
- **Import database** – select this option to import the database of a selected PBX in the off-line mode only. In the on-line mode, the database is replaced with the PBX data.
- **Export database** – select this option to export the database of a selected PBX in the off-line mode only.

Connection structure

In this menu you can create groups and subgroups (nested groups) and, subsequently, add PBXs to them. You can create PBXs without groups too, but this might be confusing if you use a higher number of PBXs. Then you can create connections for particular PBXs using the TCP/IP and modem protocols. The records are arranged alphabetically.

For easier administration of existing records, a record moving option using the mouse has been implemented on this screen, also designated as **drag & drop.**

Figure: View of Possible PBX Connection Structuret

To change the PBX connection settings use the **Properties icon**.

PBX properties

The dialogue shown in figure below helps you create a PBX or change the properties of an existing PBX. The parameter meanings are as follows:

- **Name** – define the name of the PBX to be connected.
- **Folder** – use this parameter to define the path to the folder for the configuration to be saved.
- **Autosave** – use this option to enable automatic database saving in the off-line mode.
- **Autosave database after** – here set the interval for automatic database backup. This function may only be used for the off-line mode.
- **Delete autosave item older than** – here set the maximum time for keeping old database backups in the storage folder. This function can only be used in the off-line mode.
- **Autosynchronisation** – If this selection is active, the tool tries to synchronise the data automatically with the PBX without inquiring. This function can only be used in the off-line mode. If this selection is active, 'Loading from PBX' is performed if the off-line database is empty. 'Display changes' is performed if the database has been synchronised, the identifier matches and 'Database from Autosave' has not been selected. Otherwise, the synchronisation type dialogue is invoked. If this selection is inactive, the dialogue is invoked in all cases.

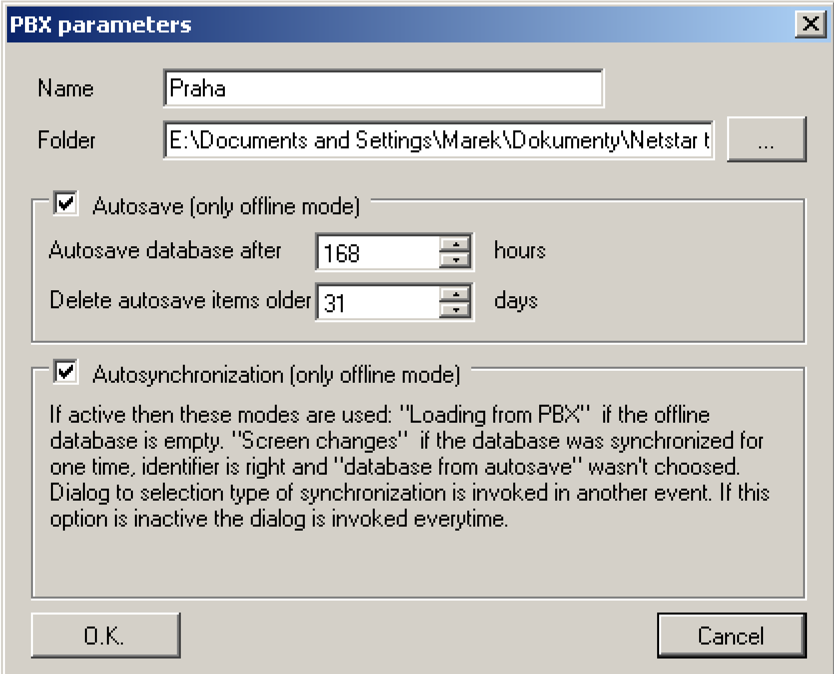


Figure: View of PBX Property Settings

Connection properties

The dialogue shown in figure below helps you create a PBX connection or change the properties of an existing PBX connection. The parameter meanings are as follows:

- **Connection name** – here enter the name of the selected connection.
- **Modes** – use this parameter to define whether the connection will support the on-line, off-line or both modes.
- **Download trace** – use this option to disable or restrict trace downloading from the PBX. Particularly useful for modem connections. *Any of the following modes can be selected:
  - **Only new** – Only the new trace is sent to the tool existent since the moment of PBX connection.
  - **Never** – No trace is sent to the tool (useful for trouble making modem connections).
  - **Always** – Approximately a 300kB – 1MB trace is read out of the PBX buffer upon connection and sent to the tool. Useful for sending the trace after the PBX start up.
- **Connection type** – use this option to define the PBX connection type. Typically, the TCP/IP or modem connection is selected. Use the TCP/IP, enter the PBX CPU IP address and the port to be used (6992 by default). With a modem choose the one that supports the X.75 protocol.
- **If unsuccessful try again** – define the time interval between the PBX connection attempts in case of failure (PBX switch-off or restart).
- **Connect as** – here define the login and password data for a secured PBX access.

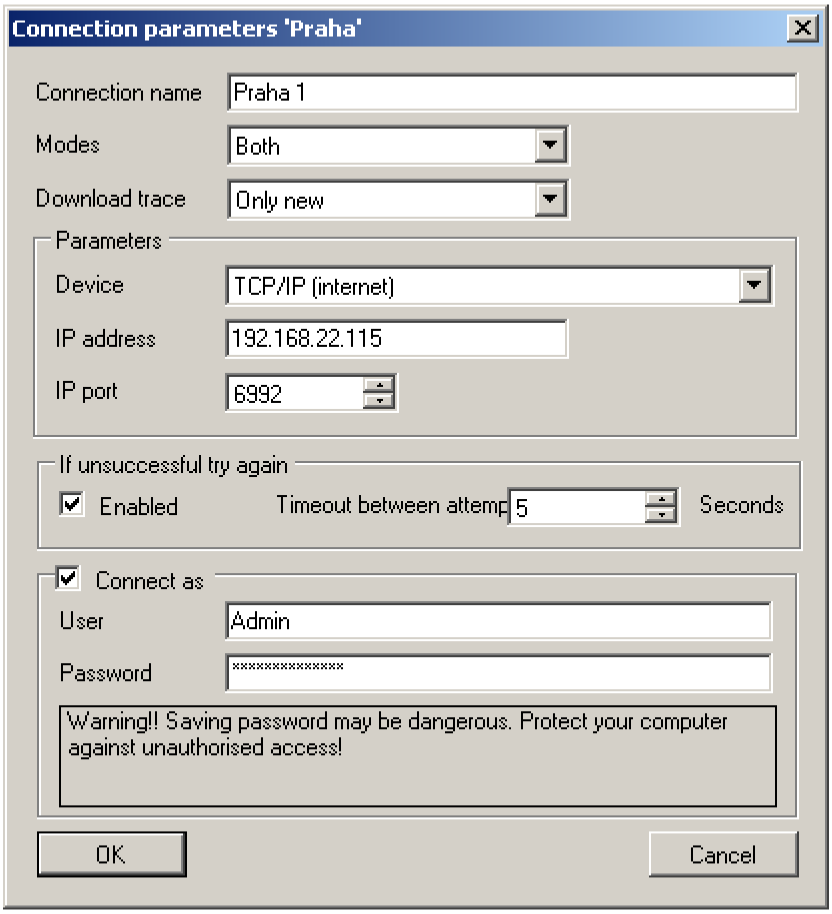


Figure: View of Connection Property Settings

Connecting to PBX

After an automatic or manual initiation of a PBX connection, the dialogue shown in figure below is displayed. It provides information on the PBX to be connected, the PBX firmware version (if detected), the last known connection error and, in the case of automatic connection attempts, also the remaining time to the next attempt. To connect immediately (before the timeout), push the **Try again** button. The **Cancel** but ton is used for leaving the dialogue.

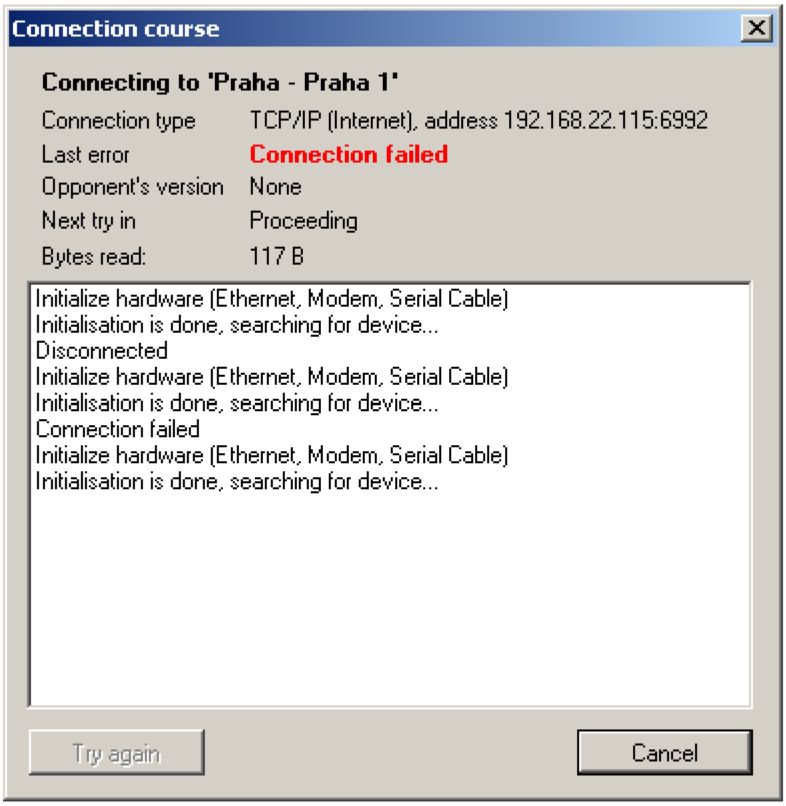


Figure: View of Connection Course Dialogue

If you are unable to connect to your PBX, please check whether:

1. the PBX has been switched on;
2. the PBX has been connected to the network;
3. both sides have the same IP address and port;
4. the used communication port has been opened;
5. the appropriate firmware and configuration tool versions are used;
6. the used communication port is not blocked by your antivirus software.

1.4 Configuration Menu

Main menu

After a successful connection to the PBX, the configuration part of the application is displayed. The main menu of this view is shown in figure below and contains the following options:

- **Administrator**
  - **Logout PBX** – use this option to logout the configuration tool from the PBX and return to the previous menu for another connection as described above.
  - **Connect/Disconnect** – use these options off-line to connect/disconnect the configuration tool to/from the PBX.
  - **Save changes** – here save all changes made since the last save.
  - **Undo changes** – here cancel all changes made since the last save in a menu.
  - **Settings** – use this option to invoke a global setting dialogue as described in Chapter 1.2 About Application.
  - **Language** – choose one of the supported languages.
  - **Exit** – use this option to exit the configuration tool.
- **Trace**
  - **Load trace from file** – use this option to load a trace from the file, thus clearing the previous one.
  - **Add trace from file** – use this option to load a trace from the file and add it to the existing one. You can interconnect traces for easy analysis.
  - **Save trace to file** – use this option to save the current on-line trace to a file. The configuration tool always saves an entire trace independently of whether the filter is being applied or not.
  - **Trace analyser** – here open the trace analysis window.
- **PBX**
  - **Upgrade** – select this option to display a firmware upgrade dialogue. Having been chosen, the firmware file is uploaded into the PBX and unpacked. After a restart, the new firmware is used.
  - **Import logs from PBX** – select this option to get an easy access to the PBX logs without using other applications. You can import **All** logs or selected logs only (**Selectively**).
  - **All** – after selecting a directory, the config.db and aoc.db files and the contents of the internal/log and /var/log directories are imported.
  - **Selectively** – select this item to display a dialogue for downloading selected logs from the PBX. The user can enter the files through a storage as defined in the **Global Data – Storage Manager** menu.
  - **Restart PBX** – use this option to initiate the PBX restart.
  - **Restart ústředny** – Volba umožňuje restart ústředny.
  - **Restore factory settings** – use this option to restore the factory default values. Choose one of the two options available in the dialogue window and press OK twice to confirm the selected action.
- **Wizards**
  - **Activation wizard** – for details refer to the next chapter, 1.5 PBX Activation.
  - **Import/export company structure** – here invoke the company structure

import/export dialogue. The csv and xml files are supported.

- **Database import** – click on this option to display the database importing window. Select **From file** or **From PBX**. If you choose **From PBX**, all the PBXs available for database import will be displayed. Use the **Rule** parameter to specify which of the original settings should not be overwritten by database import. The option is available in the off-line mode only.
- **Database export** – click on this option to export the PBX database into an xml file. The option is available in the off-line mode only.

- **Help** – select this option to start the help in the chosen language.

Figure: View of Configuration Tool Main Menu

Figure above also includes all configuration menu icons. Icon meanings from the left:

- **Logout PBX** – use this icon to logout the configuration tool from the PBX and return to the previous menu for another connection as described above.
- **Connect/Disconnect** – use these icons off-line to connect/disconnect the configuration tool to/from the PBX.
- **Save changes** – use this icon to save all changes made since the last save.
- **Undo changes** – use this icon to cancel all changes made since the last save in a menu.
- **Language** – flags are used to mark the configuration tool language versions.

Windows

On the left-hand side of the configuration tool you can find the TreeView where you can choose a menu item to be configured. The selected menu then opens on the right-hand side and is mostly divided into two subwindows; one for selecting and the other for configuring an object. The configuration is divided into tags for easier orientation. All the above mentioned windows are shown in the figure below.

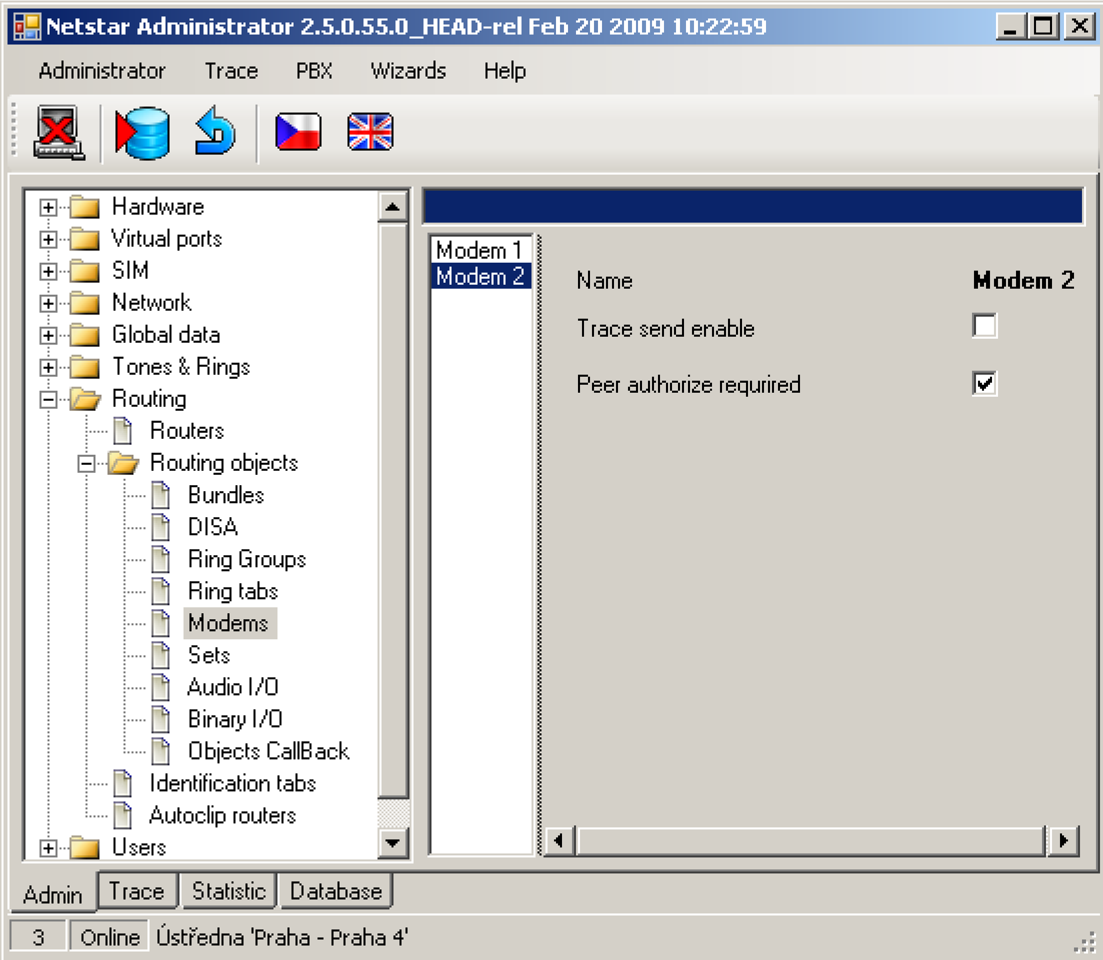


Figure: View of Configuration Tool Tags and Windows

The **Trace** and **Database** sections situated in the lower part of the screen above the status bar are important configuration tool components. **Trace** helps you monitor calls and analyse configuration errors if any. **Database** provides a direct view of the data stored (depending on the connection mode). **We strongly recommend that you should not change the data in this view if you do not know how!** This menu tag is governed by the read & write rights assigned to each login.

Status bar

The status bar is a lower task bar in the configuration tool. It provides two important pieces of information. The first one is the connection mode, which is either **on-line** or **off-line**. The other is the connected PBX name, which includes the **PBX** and **connection** data separated with a dash.

Figure: View of Configuration Tool Status Bar

1.5 PBX Activation

What you need

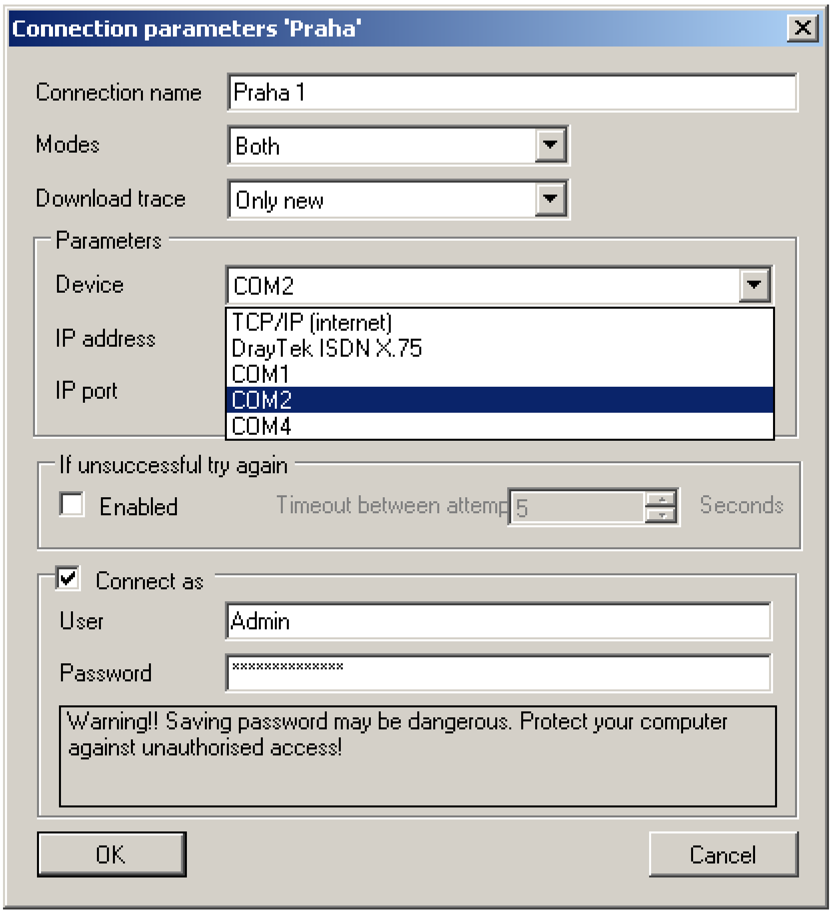
To activate and configure **2N® NetStar** you need the **2N® NetStar system**, a computer running the supported Windows version, a keyboard and a mouse. The PC and **2N® NetStar** PBX have to be interconnected in a LAN. Furthermore, it is necessary to display the redirected standard PBX input and output on your PC console. To do this, you need a six–wire, crossed cable with a six-pin RJ-12 connector on one end and a serial connector on the other. This cable is supplied as standard equipment. You can use a 'Tera Term Pro' console or any other functional console. You can also work in the HyperTerminal mode, but longer lines are interlaced.

Step 1: IP address setting

Prior to starting configuration you need to set up the PBX IP address to establish network communication between the system and the computer. To do this, you can use a serial console or the default IP address 192.168.100.100. The **2N® NetStar** – PC communication can be on-line or off-line. To configure the console use the table below.

Speed	115200
Bits	8
Parity	None
Stop bits	1
Flow control	None

If necessary, a serial port or modem may be used for PBX connection through the configuration tool, yet at a considerably lower bit rate.



Step 2: Hardware activation

After the first connection to the selected PBX according to the Connection to PBX chapter, a configuration wizard is displayed as shown in figure below. This wizard is displayed only if the PBX has a new empty database (has not been preconfigured according to your demands).

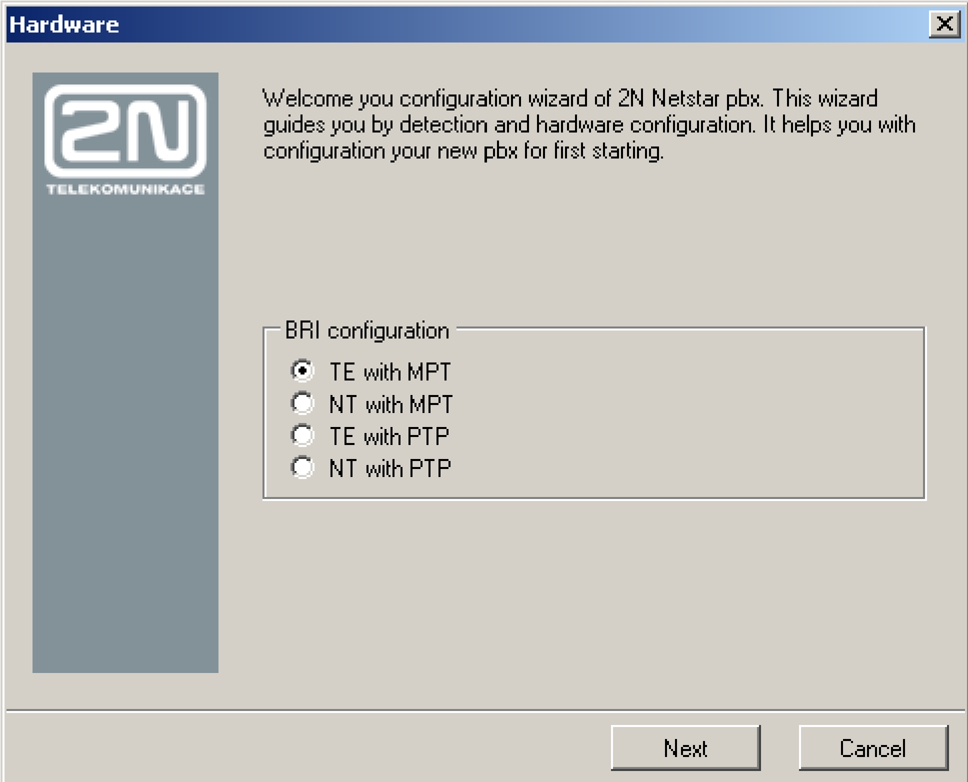


Figure: View of Hardware Configuration Wizard Dialogue

If the wizard is displayed, you can define the basic configuration settings of the virtual BRI ports. If you are not sure, you can push the **Next** button to proceed to the next configuration step because these settings may be changed any time later. Once you do that, the configuration tool (together with the PBX) starts to detect the hardware as shown in figure below.

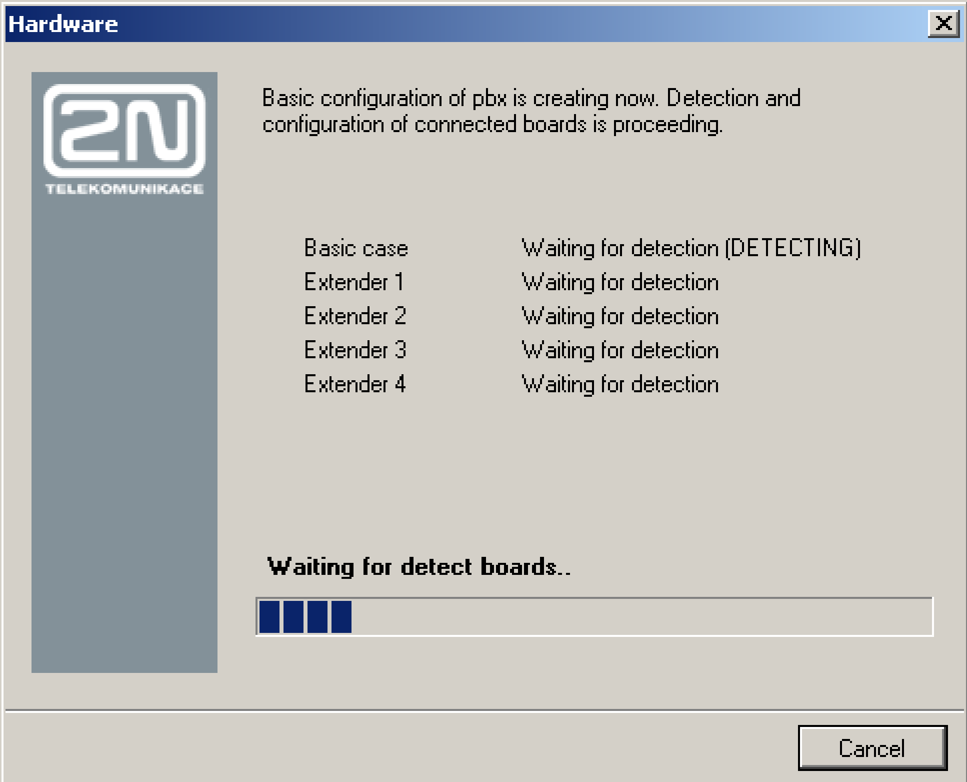


Figure: View of Wizard's Hardware Detection Operations

The boards are detected both in the basic unit and extenders. Once some hardware is detected, activation takes place, which means that virtual ports are assigned to all the boards detected (except for VoIP boards). The terminals connected are detected in the last stage of the wizard hardware configuration. This should make the PBX ready for further configuration, which is signalled by green board LEDs. The GSM board is the only board without LED indicators and so its ready status is signalled by the port LEDs. The current firmware version is loaded into the PBX after the first start–up or every firmware upgrade and may cause a short delay in the GSM board activation.

Step 3: Localisation setting

Localisation setting is another important step of the **2N® NetStar** configuration. In this step you can define the parameters shown in figure below and described in detail in the 6.3 Localisation chapter. In addition, you can add a language package of your own including texts and progress tones. Two language packages – Czech and English – are available in the PBX by default.

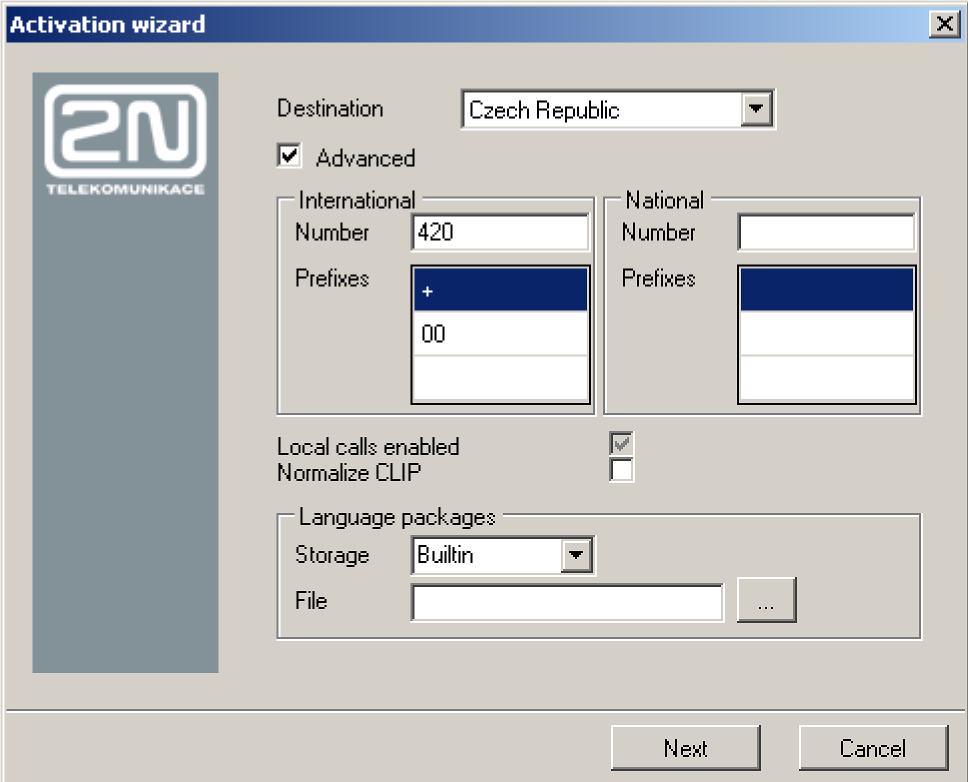


Figure: View of Wizard's Localisation Setting Operations

Step 4: Time setting

In this step, the wizard helps you set time, date and time zone. And also define the NTP server for automatic time synchronisation.

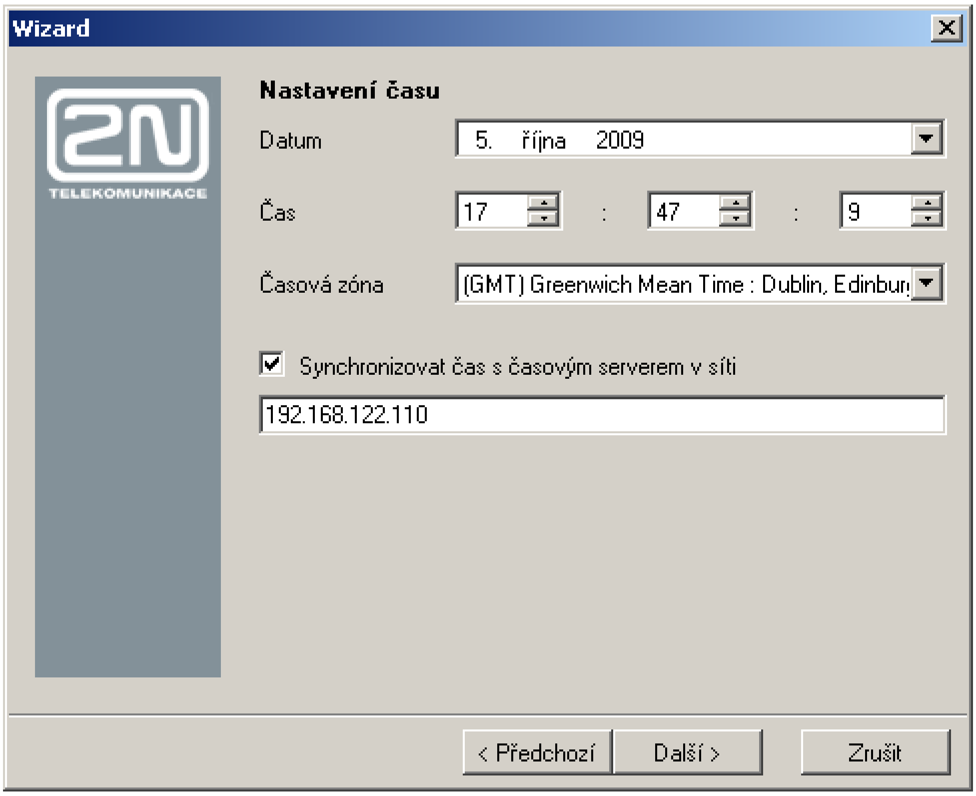


Figure: View of Configuration Wizard's Time Setting

Step 5: PBX function selection

In this step, choose one of the PBX modes. The setting is not final, it just defines the wizard's next configuration steps. The following options are available:

- **Private branch exchange**
- **Virtual branch exchange**
- **GSM gateway**
- **Hotel**

The offer of settings depends on the PBX mode selected. The lowest number of settings are available in the **GSM gateway**, where some steps are omitted and configuration starts as late as the SMTP. The settings are identical for the other modes except for step 1, which is always adapted to the particular version. The following steps are available in the **Private branch exchange**.

Step 6: Creation of groups, users and extensions

In this step, the configuration wizard enables an automatic creation of a group and its users and extensions. There are three types of extensions to be generated – analog, SIP and Cornet extensions. Analog extensions are used for the ASL virtual ports. SIP extensions are used for connecting SIP–supporting VoIP terminals and are assigned to the SIP proxy terminals. Cornet extensions are used for the StarPoint key (system) phones connected to the Cornet virtual ports. You can define the first extension number and count of extensions for each group (every other extension has a number increased by one). The extensions are then assigned to ports according to their types (if possible).

If you do not want to create extensions automatically, you can import the company structure from a pre–prepared file in the **xml** or **csv** format. In this way you can create a relatively complex company structure including user logins and multiple user extensions.

Three functions are added to this section, the first two of which are used for re–launching of the wizard.

- **Add new and remove deleted** – The existing structure of extensions is compared with the selected file. New extensions are added and those which are present in the PBX yet undefined in the file are removed.
- **Add new only** – The existing structure of extensions is compared with the selected file and new extensions are added only. Undefined extensions are retained in the configuration.
- **Assign ports randomly**–– select this item to enable random assignment of extensions to ports.

If you neither want to automatically create extensions nor to import the company structure, you can push the **Next** button and select **Don't create anything** to proceed to the next step.

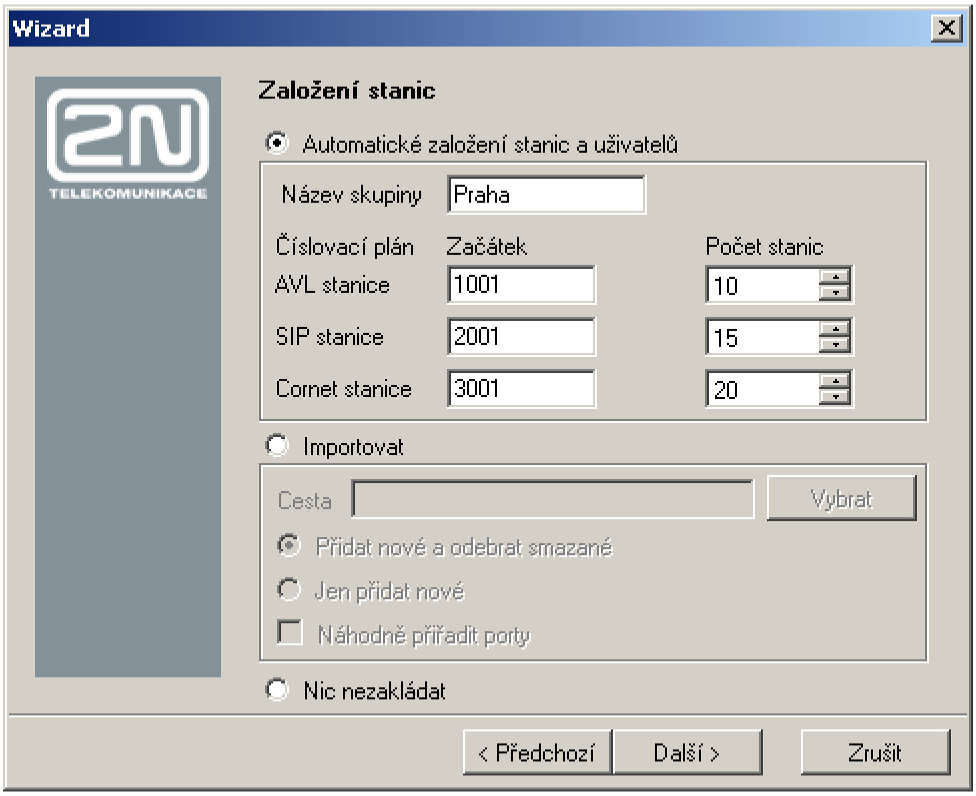


Figure: View of Wizard's Extension Creation or Intra–Plant Structure Import

Step 7: Settings for Assistant

This configuration step includes just two functions with the following meanings:

- **Launch web server** – this item launches the internal PBX web server, to which you can log in by entering the CPU IP address from your web browser.
- **Generate default logins** – this item generates logins for the users created in the preceding step. With them, you can log in to the web server as a user.

Step 8: SIP domain setting

This step helps you define a specific SIP domain. If this option is not selected, the CPU IP address is used as the domain.

Step 9: SMTP setting

Within this step you can define the SMTP server to be used by the PBX. Port 25 and the CPU Ethernet interface are set automatically for the SMTP. No security is used by default.

Step 10: Creation of routers

The wizard's last step is creating PBX routers. Routers are used for call/SMS routing from one PBX port to another. The wizard offers several default sets of routers, which are sufficient for your basic call routing. For special routing demands, reconfigure the routers and add new ones. All new routers are automatically filled with services, extensions and users and linked with each other.

Step 11: Data saving

Changes are not automatically stored into the PBX during the Wizard process. To save the changes, use the **Ctrl+S** shortcut or the **Save** button after completing the Wizard. To cancel all new settings, push the **Undo** button.

2. Hardware

Here is what you can find in this chapter:

- 2.1 Basic
- 2.2 Boards
- 2.3 Synchronisation
- 2.4 Board and Port List

2.1 Basic

Service mode

The section helps you put the PBX in the service mode and back if necessary. The service mode is used for quick changes such as card replacement. The PBX start is much faster after the service mode than after the PBX power off.

- **OFF** – a normal PBX running status. To reuse the PBX while in the service mode, select **OFF** and **save the changes**. Having returned from the service mode successfully, you can see in the section **Detected rack** in the column **Status** the state **RUN**.
- **ON** – the PBX service mode. To switch the PBX into the service mode while it is in a normal running status, select **ON** and **save the changes**. Having transferred into the service mode successfully, you can see in the section **Detected rack** in the column **Status** the state **STOP**.

Detected rack

The detected rack table gives you a clear overview of all parts of your PBX.

- **MAC address** – MAC address of the detected rack.
- **Serial number** – serial number of the detected CPU card.
- **State** – current rack state. This may differ from the CPU card state (e.g. the CPU is ON while the rack if OFF in the service mode).
    - **RUN** – normal rack operation. The power supply is connected to the cards.
    - **STOP** – the rack is stopped. The power supply is disconnected from the cards. Typical for the service mode.
    - **ERROR_LICENCE_EXPIRED** – the rack is running, but the trial licence or time-limited main licence has expired. A new licence has to be requested.
- **1:Basic** – indicates the basic unit state.
    - **PRESENT** – the basic unit is detected.
    - **MISSING** – the basic unit is not detected.
- **2-5:Extender** – displays the extender state.
    - **PRESENT** – the extender is detected.
    - **MISSING** – the extender is not detected. Check the power supply connection and the switch card - extender CPU interconnection.

Hardware profile

To set the hardware profiles and improve your system efficiency use the **HW – HW profiles** menu. The menu contains five different hardware profiles. For their benefits and disadvantages see the table below.

Each VoIP card allocates 32 channels in hardware profile 0 and 64 channels in all the other profiles. The PRI card always allocates 32 bus channels.

- **HW profile number** – since different hardware profiles use different bus frequencies, the new configuration saving system is switched into the service mode and back automatically to bring the bus frequency changes into effect. The bus clock can be 2, 4 or 8 MHz, which corresponds to 32, 64 nebo 128 extender channels respectively. The count of extender channels available for calls is always lower by 4 as one channel is normally occupied by signalling.

Profiles	0	1	2	3	4	5	6	7	8
Extenders	4	0	4	4	4	0	4	4	4
Extender channels	128	0	32	64	128	0	32	64	128
Trunk positions	128	256	224	192	128	224	192	160	96
Main case case – digital	128	64	64	64	64	64	64	64	64
Main case – analog	32	32	32	32	32	64	64	64	64
Detectors	32	64	64	64	64	64	64	64	64
Players	32	64	64	64	64	64	64	64	64

Table: Benefits and Disadvantages of Hardware Profiles

2.2 Boards

Arrangement of HW

Unfolding the **HW – Rack** menu you can see the rack fitting as shown in the figure below.

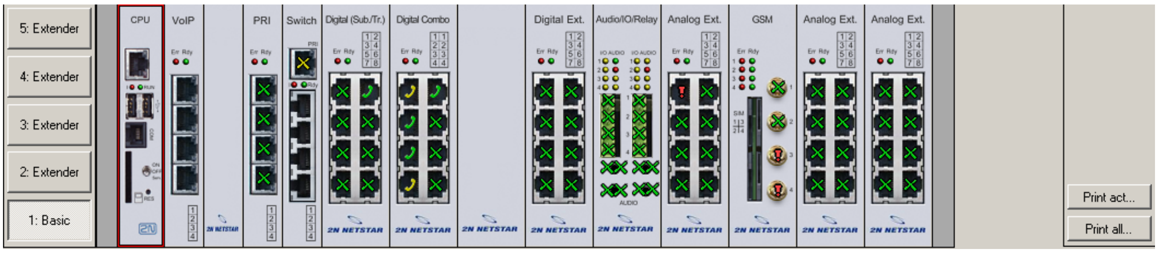


Figure: View of PBX Basic Unit Panel

Push the buttons on the left-hand side of the PBX to switch between the basic unit and extender views. Click on the right-hand mouse button in the basic unit or extender view to display the following options:

- **Add board** – click on an empty (no-board) PBX position to use this option. Add a board that has been detected by the PBX (using the Detected option) or a board from the list of supported boards for this position.
- **Remove board** – use this option to remove a selected board. If virtual ports or resources have been assigned to the board, you will be asked whether they should be removed or retained.
- **Migrate virtual port/resource** – use only if the context menu has been displayed on a port to initiate a virtual port substitution dialogue.
- **Synchronise with detected** – use this option to synchronise the current unit or extender with the detected PBX boards. Before a board is removed from the current configuration, you are asked to confirm the removal.
- **Expert menu** – use this option to access the advanced unit, board or port configuration functions. For details see later.

> Let us explain the terms "virtual port" and "physical port" and their difference for convenience. Basically, a virtual port is used for software setting of basic properties of a physical port. The advantage of this approach is the fact that the defined set of properties is attributed to a physical port only if the virtual port is assigned to it. Thus, you can esily move virtual ports between physical ports and change their functions as necessary.

- **Expert menu – Virtual port**
  - **Assign virtual port/resource** – use this option to assign an existing virtual port to a physical port. Select a virtual port from the list of existing ports.
  - **Create virtual port/resource** – use this option only for physical ports without any assigned virtual port. The new virtual port is automatically assigned to this physical port.
  - **Remove virtual port/resource** – use this option to remove a virtual port

from a selected physical port without deleting it. This virtual port can be used later including all settings (routing, assigned extensions, etc.). Rename the virtual port from XXX to UnassignedXXX.
    - **Delete virtual port/resource** – use this option to remove and delete a virtual port once forever. You will not be able to use this port any more.
    - **Regenerate name** – use this option to rename a selected virtual port according to its physical port.
- **Expert menu – Board and Case**
    - **Create virtual ports/resources** – use this option to create virtual ports (resources) for all of those physical board or unit ports at once that have not been assigned a virtual port.
    - **Remove virtual ports/resources** – use this option to remove all virtual board or unit ports at once without deleting them. These ports can be used later including their settings. Rename these virtual ports from XXX to UnassignedXXX.
    - **Delete virtual ports/resources** – use this option to delete all virtual board or unit ports once forever. You will not be able to use this port any longer.
    - **Regenerate unchanged names** – use this option to change all unchanged names of the virtual board or unit ports according to their physical ports.
    - **Regenerate all names** – use this option to change the names of all virtual board or unit ports according to their physical ports.

Board

The figure below shows all possible signalling statuses on the board ports.

Figure: View of Available Analog Board Signalling Statuses

- **Earphone**
- **Cross**
    - **Green** – signals a physical port with an assigned virtual port.
    - **Yellow** – signals a physical port with an assigned virtual port and active call (or call establishment).
    - **Green** – signals a physical port with an assigned virtual port and assigned extension.
    - **Yellow** – signals a physical port with an assigned virtual port, assigned extension and active call (or call establishment).

- **Exclamation mark**
  - **Yellow** – signals a physical port without any assigned virtual port or physical port without detected status.
  - **Red** – signals a hardware error, e.g. a low signal level for a GSM, GSM port without SIM card, ISDN virtual port with deactivated L1 or L2 (adjustable), etc.

PBX HW configuration print

The buttons to the right of the basic unit/extender figure help you print out the current PBX hardware configuration. Click on one of the buttons to display the print setting options. Click on Print to display the preview and on the button in the left-hand upper corner to print out the configuration. The buttons in the right-hand upper corner display the extender configuration previews.

- **Print current view** – print the view selected using the buttons to the left of the CPU board (basic unit or extender).
- **Print all** – print the basic unit and extender configurations.

Boards

A two-part **Boards tag** is available under the PBX view. The upper part shows basic information on the selected board. The parameters mean the following:

- **Position** – gives the board position number in the case as described below.
- **Type** – gives the type of the board to be configured.
- **Enabled** – disables the selected board. This option is useful, for example, while changing SIM cards without switching off the PBX.
- **State** – provides the current board status including information on a mismatch of the board to be configured with the detected one.
- **Detected** – shows the parameters of the board detected.
  - **Type** – type of the board detected on a selected position.
  - **Serial number** – serial number of the board detected on a selected position.
  - **MAC address** – MAC address of the board detected on a selected position.

A window showing the list of physical ports of the selected board is displayed under the above-mentioned part. The meaning of each list column is explained in the 2.4 Board and Port List chapter.

Tab Virtual port

The **Virtual port** tag helps you configure your virtual ports easily. You can set all parameters of the selected virtual ports and simultaneously see the panel layout. Use the **Virtual port** tag for an easy configuration of virtual ports. The tag includes all configuration settings for the selected virtual port while keeping the PBX view. Click on a card or its port to display the port assignment to a virtual port type in the left section of the screen. Use the drag&drop function to move a virtual port to another type. If you select the CPU card, all the virtual ports that use the card's LAN interface are displayed. Thus, you will see the SIP Proxy and SIP Gateway virtual ports as well as the SMTP and SMTPD virtual ports.

For details on the setting options associated with the parameters in the right-hand section of the tag refer to the User Manual chapters dedicated to particular virtual ports (especially 3. Virtual Ports).

Addressing

The position of each board is specified in the **R : C : B** format and the position of a port in the **R : C : B : P** format. The characters have the following meanings:

- **R** – rack number;
- **C** – rack unit number;
- **B** – unit board number;
- **P** – board port number.

Currently, **R** takes up the value of 0 and **C** ranges from 1 to 5, the basic unit being 1, the first extender 2 and so on up to the fourth extender with number 5. The board positions (**B**) in the basic unit are numbered 1 to 14 from the left. The extender positions are numbered similarly, from 1 to 12. The first basic unit and extender position is always reserved for the CPU board. The 1x/2x/4x ISDN PRI (with or without Zarlink) or Surf Ethernet boards can be mounted into positions 0:1:2 to 0:1:4 only. The position 0:1:5 is reserved for the Switch board, which contains the digital switching array.

2.3 Synchronisation

Upon connection to a public or private ISDN network, remember to configure one port for synchronisation at least. The PBX works in two modes at the same time: as a source of synchronisation (Master) and a device that receives synchronisation (Slave). There are two fields in the **HW – Synchronisation** menu. The left-hand one contains all digital virtual ports that can be selected for synchronisation, i.e. all PRI and BRI virtual ports in the TE mode. The other field contains a list of virtual ports that have been selected for synchronisation. All ports in the NT mode can be used as synchronisation sources.

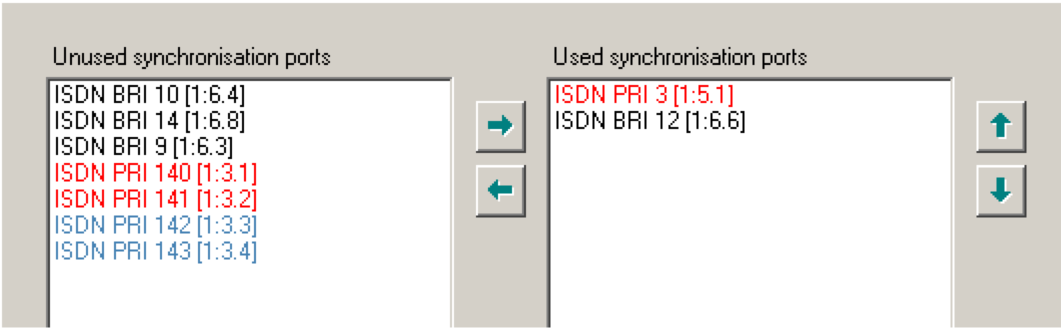


Figure: View of Synchronisation Port Assigning and Priorities

Use the **Up** and **Down** buttons to move the selected synchronisation up or down for a higher or lower synchronisation priority respectively – the port listed first has the highest priority (255). Every other carrier has a priority lower by 1 (254, 253, ..). A newly assigned carrier is always placed last (the lowest priority). In the case of synchronisation loss, the following port in the list (with a lower priority) is selected automatically. Upon synchronisation restoration, the PBX returns automatically to the port with the highest priority.

Push the **Right** and **Left** buttons to transfer the virtual ports from one field to another and thus ensure the PBX synchronisation.

2.4 Board and Port List

The **Hardware – Board list** menu contains a list of boards that are physically present in the PBX. The board list has four columns with the following meanings:

- **Address** – shows the physical board address within the PBX according to the Boards chapter.
- **Type** – shows the board type.
- **Serial number** – shows the factory-programmed board serial number.
- **MAC address** – shows the board MAC address.
- **Module IMEI** – shows IMEI of GSM module.
- **Virtual port** – shows the complete name of the carrier or resource assigned to a physical port.
- **Stack** – shows the general carrier / protocol stack (DSS1, ASL, CO, etc.).
- **Extension** – shows the list of extensions assigned to a physical port carrier.
- **User** – shows the users of extensions assigned to a physical port.
- **State** – shows the current port state.
- **Description** – provides associated information.

The context menu under the right button offers the following two options:

- **Export to CSV** – use this option to export the whole table into a *.CSV file. You can use this export for stocktaking purposes and/or for contacting the **2N® TELEKOMUNIKACE** Technical Support if necessary.
- **Move to port** – use this option to move quickly to the configuration of the virtual port on the selected row.

3. Virtual Ports

Here is what you can find in this chapter:

- 3.1 BRI and PRI
- 3.2 Cornet
- 3.3 ASL
- 3.4 CO
- 3.5 GSM
- 3.6 SIP
- 3.7 SMTP
- 3.8 Virtual Port Options

3.1 BRI and PRI

BRI

Refer to the Boards section for the meaning of the virtual port.

BRI virtual ports are assigned to physical ISDN ports for the Basic Rate Interface. For the hardware configuration of BRI virtual ports refer to the **Virtual ports – BRI/PRI** menu in the **Stack** tag. A list of all BRI virtual ports is displayed on the left and a window for the port parameter setting is available on the right. The configuration parameters are divided into logical parts.

Stack status

This field displays information on the stack and its current status including information on the L1 or L2 states, higher error rates or loss of synchronisation.

Digital interface parameters

- **Interface type** – cannot be selected, only shows the type of interface including bit rate information.
- **Interface mode** – use this parameter to switch between the NT (Network Termination) and TE (Terminal Equipment) modes. Some basic unit positions can only be used in the NT mode. Specifically, they are basic unit board positions 6, 9 and 12 if you follow addressing as described in Boards. A correct function requires a software–hardware matching and a proper jumper setting for each board port. Figure below may serve as a guide.

> Do not remove the board from the PBX without prior PBX switching off or into the service mode to avoid the PBX damage.

- **Bus mode** – use this parameter to switch between the MPT (point–to–multipoint) and PTP (point–to–point) modes. In the MPT mode you can connect up to eight terminals to one physical port. The PTP mode is mainly used for cross–connecting lines (trunks) between PBXs or for one terminal connection.
- **Enabled channels** – use the checkboxes to activate the B–channels. If no channel is checked off, you cannot use this port for communication (it behaves as if busy).
- **Deactivate L1 at relax** – use this parameter to deactivate the L1 layer on an inactive interface. The PBX automatically deactivates the layer after a timeout as defined in the **Deactivation timeout** item. Any incoming call automatically reactivates this layer.
- **Keep L1 active** – check this option to make the PBX keep L1 active on this interface without any incoming call. This option cannot be combined with the **Deactivate L1 at relax** option.
- **Inactive L1 as error** – use this option to activate a caution about the first layer being inactive. This is indicated by a red exclamation mark on the port in the **Hardware – Boards** menu and a red text in the upper stack status field. This option may not combined with the **Deactivate L1 at relax** option.
- **Set SLIP** – select **Nonsynchronous as error** to enable acceptable SLIP range parameters. Use this option in the TE mode only. If the SLIP rate gets over the

upper level, a red exclamation mark appears on the port in the **Hardware – Boards** menu and a red text is displayed in the upper stack status field. This error status gets changed after the SLIP rate falls below the **lower** level. The interval between these two values represents hysteresis.

- **Settings for BER** – select **BER as error** to enable acceptable interface error parameters. If the BER rate gets over the **upper** level, a red exclamation mark appears on the port in the **Hardware – Boards** menu and a red text is displayed in the upper stack status field. This error status gets changed after the BER value falls below the **lower** level. The interval between these two values represents hysteresis. The BER values are entered in an exponential format (e.g. 3e–5 means 3 errors in 100,000 bits). In practice, the BER may occur on a virtual port close after the cable connection. In that case, this is not an error if the BER disappears in a few minutes.

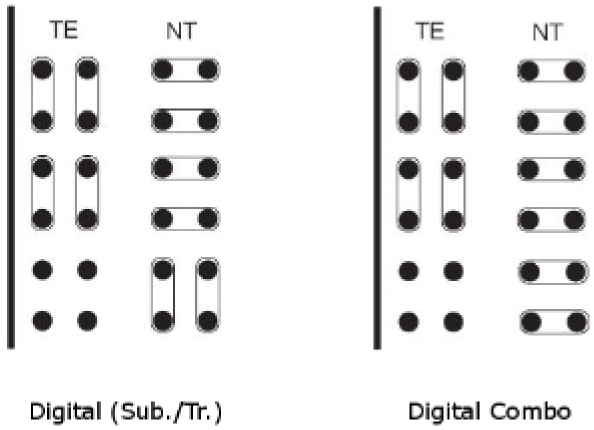


Figure: View of ISDN BRI Board Jumper Configuration (The thick line represents the board front)

Specific interface parameters

- **Multiframe** – is the first layer parameter of the So bus. For details refer to recommendation I.430.
- **Extended bus** – use this parameter to activate an extended bus. With just one terminal and proper terminating impedance you can extend the PBX–terminal distance up to 1,000 metres. This parameter can be set for an NT port only.
- **Priority 10** – is the first layer parameter of the So bus. This parameter can be set for a TE port only.

DSS1 protocol parameters

- **Reverse NT/TE mode** – this option refers to L3 signalling only. Check this option to make a TE port behave as an NT port (and vice versa).
- **Do not send time at NT** – use this option to disable sending of the connection date and time information within the CONNECT message from an NT port to a TE port. Available for NT ports only.
- **Ignore unset explicit channel** – use this option to enable call establishment without an explicitly set B–channel.
- **Always select B–channel** – use this option to disable sending the channel identification information within the SETUP message together with channel signalling. Available for TE ports only.
- **Disconnect L2 if there is no call** – use this option to disconnect the L2 layer on a inactive interface. The PBX automatically disconnects the layer after a timeout. An incoming call automatically reconnects the layer.
- **Keep L2 connected** – check this option to make the PBX to keep L2 connected on this interface without any incoming call. This option may not be combined with the **Disconnect L2 if no call** option.
- **Disconnected L2 as error** – use this option to activate a caution about the second layer being disconnected. This fact is indicated by a red exclamation mark on the port in the **Hardware – Boards** menu and a red text in the upper stack status field. This option may not be combined with the **Disconnect L2 if no call** option.
- **Terminals** – is active for virtual MPT NT ports. Enter all connected ISDN terminals including their **MSN numbers**. Assign a extension to these terminals using the **Extensions** tag. The terminal shall then identify itself as the selected extension within the PBX.

> **2N® NetStar** is able to process (i.e. resend to an interface other than ISDN) only **G.711 A-law** encoded calls on the ISDN interface. Calls encoded by **G.711 µ-law** can only be resent between the ISDN interfaces in the PBX.

Digital interface diagnostic

- **Line state** – the parameter cannot be set. It only shows the state of the first interface layer.
- **Number of SLIPs per minute** – this parameter gives the count of slips. A slip is caused by different clocks of the PBX and the active terminal. This value is updated every 6 seconds and represents a weighted average per minute.
- **Bit error rate per second** – the BER parameter gives a count of incorrectly transferred bits during transmission. The value is updated every 6 seconds and represents a weighted average per minute.

If the **SNMP** port supervision is used, enable some or all of the following parameters: **Inactive L1 as error**, **Disconnected L2 as error**, **Nonsynchronous L1 as error**, **BER as error**. If not, the port error will not be detected and the PBX will be unable to send a warning.

Tab Expert

- **Cause mapping** – is an additional function for modification of outgoing causes for a selected virtual port. It may be useful while adapting NetStar causes to the specific conditions of your network. You can choose a particular internal NetStar cause in the left–hand column and assign a cause to it to be sent to your network in the right–hand column.

PRI

Refer to the Boards section for the meaning of the virtual port.

PRI virtual ports are assigned to physical ISDN ports for the Primary Rate Interface. For the hardware configuration of the PRI virtual ports refer to the **Virtual ports – BRI/PRI** menu in the Stack tag. A list of all available PRI virtual ports is displayed on the left and a window for the port parameter settings is on the right. The configuration parameters are divided into logical parts.

Stack status

This field displays information on the stack and its current status including information on the L1 or L2 states, higher error rates or loss of synchronisation.

Digital interface parameters

- **Interface type** – the parameter cannot be configured. It only shows the type of interface including bit rate information.
- **Interface mode** – use this parameter to switch between the NT (Network Termination) and TE (Terminal Equipment) modes. A correct function requires a software–hardware matching and a proper jumper setting for each ISDN board port. Figure below may serve as a guide.

> Do not remove the board from the PBX without prior PBX switching off or into the service mode to avoid the PBX damage.

- **Enabled channels** – use the checkboxes to activate the B–channels. If no channel is checked off, you cannot use this port for communication or data transmission (it behaves as if busy). B–channels 0 and 16 cannot be used for call or data transmission under normal circumstances since they are blank. In PCMs of the 1st order are used for frame synchronisation and signalling transmission.
- **Deactivate L1 at relax** – use this parameter to deactivate the L1 layer on an inactive interface. The PBX automatically deactivates the layer after a timeout as defined in the Deactivation timeout item. Any incoming call automatically reactivates this layer.
- **Keep L1 active** – check this option to make the PBX keep L1 active on this interface without any incoming call. This option cannot be combined with the **Deactivate L1 at relax** option.
- **Inactive L1 as error** – use this option to activate a caution about the first layer being inactive. This is indicated by a red exclamation mark on the port in the **Hardware – Boards** menu and a red text in the upper stack status field. This option may not combined with the **Deactivate L1 at relax** option.
- **Settings for SLIP** – select **Nonsynchronous as error** to enable acceptable

SLIP range parameters. Use this option in the TE mode only. If the SLIP rate gets over the **upper** level, a red exclamation mark appears on the port in the **Hardware – Boards** menu and a red text is displayed in the upper stack status field. This error status gets changed after the SLIP rate falls below the **lower** level. The interval between these two values represents hysteresis.

- **Settings for BER** – select **BER as error** to enable acceptable BER range parameters. If the BER rate gets over the **upper** level, a red exclamation mark appears on the port in the **Hardware – Boards** menu and a red text is displayed in the upper stack status field. This error status gets changed after the BER rate value falls below the **lower** level. The interval between these two values represents hysteresis. The BER values are entered in an exponential format (e.g. 3e–5 means 3 errors in 100,000 bits). In practice, the BER may occur on a virtual port close after the cable connection. In that case, this is not an error if the BER disappears in a few minutes.

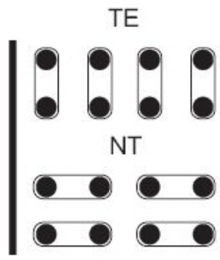


Figure: View of ISDN PRI Board Jumper Configuration (The thick line represents the board front)

Specific interface parameters

- **Prefer CRC** – use this option to enable preferring communication with the Cyclic Redundancy Check. In this mode, the PBX tries to establish connection with the CRC at first and, having failed, attempts to establish connection without the CRC.
- **Long haul** – use this parameter to activate an extended bus called the Long Haul. With just one terminal and holding impedance, you can extend the distance up to 1,000 metres. This parameter can be set only on an NT port.

DSS1 protocol parameters

- **Reverse mode NT/TE** – this option refers to L3 signalling only. Check this option to make a TE port behave as an NT port (and vice versa).
- **Do not send time at NT** – use this option to disable sending of the connection date and time information within the CONNECT message from an NT port to a TE port. Available for NT ports only.
- **Ignore unset explicit channel** – use this option to enable call establishment without an explicitly set B–channel.
- **Always select B–channel** – use this option to disable sending the Channel identification information within the SETUP message with channel signalling. Available for TE ports only.
- **Disconnect L2 if no call** – use to disconnect the L2 layer on an inactive interface. The PBX automatically disconnects the layer after a **timeout**. An incoming call automatically reconnects the layer.
- **Keep L2 connected** – check this option to make the PBX keep the L2 layer connected on this interface without any incoming call. This option may not be combined with the **Disconnect L2 if no call** option.
- **Disconnected L2 as error** – use this option to activate a caution about the second layer being disconnected. This fact is indicated by a red exclamation mark on the port in the **Hardware – Boards** menu and a red text in the upper stack status field. This option may not be combined with the **Disconnect L2 if no call** option.
- **Terminals** – not applied for PRI ports.

> **2N® NetStar** is able to process (i.e. resend to an interface other than ISDN) only **G.711 A-law** encoded calls on the ISDN interface. Calls encoded by **G.711 µ-law** can only be resent between the ISDN interfaces in the PBX.

Digital interface diagnostic

- **Line state** – the parameter cannot be set. It only shows the state of the first interface layer.
- **Number of SLIPs per minute** – this parameter gives the count of slips. A slip is caused by different clocks of the PBX and the active terminal. This value is updated every 6 seconds and represents a weighted average per minute.
- **Bit error rate per second** – the BER parameter gives a count of incorrectly transferred bits during transmission. The value is updated every 6 seconds and represents a weighted average per minute.

If the **SNMP** port supervision is used, enable some or all of the following parameters: **Inactive L1 as error**, **Disconnected L2 as error**, **Nonsynchronous L1 as error**, **BER as error**. If not, the port error will not be detected and the PBX will be unable to send a warning.

Expert tab

- **Cause mapping** – is an additional function for modification of outgoing causes for a selected virtual port. It may be useful while adapting NetStar causes to the specific conditions of your network. You can choose a particular internal NetStar cause in the left–hand column and assign a cause to it to be sent to your network in the right–hand column.

3.2 Cornet

Cornet is a digital virtual port for the StarPoint key phones with proprietary signalling (UPN interface). The Stack tag provides limited configuration capacities only. The parameters are divided into logical sections according to their respective functions. For the StarPoint configuration parameters refer to the Softphone subtag of the Properties tag. For more details on Softphone extensions refer to Chapter Setting Properties.

Virtual port status

The upper menu field displays information on the stack type and its current status including information on the L1 and L2 states, increased bit error rate or nonsynchronous L1.

Digital interface parameters

- **Interface type** – the parameter cannot be configured. It only shows the type of interface.
- **Interface mode** – this parameter is set to NT and cannot be reconfigured. Thus, cross–connecting lines cannot be made using these ports.
- **Bus mode** – this parameter is always set to PTP and cannot be reconfigured. This port is used for one terminal only.
- **Enabled channels** – use this parameter to enable selected interface channels. If none is enabled, the interface cannot be used and behaves as if it was busy.
- **Keep L1 active** – check this option to make the PBX keep the interface active automatically.
- **Inactive L1 as error** – use this option to activate a caution about the first layer being inactive. This is indicated by a red exclamation mark on the port in the **Hardware – Boards** menu and a red text in the upper stack status field.
- **Set BER** – select BER as error to enable acceptable BER range parameters. If the BER rate gets over the **BER error level**, a red exclamation mark appears on the port in the **Hardware – Boards** menu and a red text is displayed in the port status field. This error status gets changed after the BER value falls below the **BER OK level**. The interval between these two values represents hysteresis. The BER values are entered in an exponential format (e.g. 3e–5 means 3 errors in 100,000 bits).

Master terminal

- **Type** – shows the type of the StarPoint terminal connected.
- **Firmware** – shows the current firmware version of the terminal connected.
- **Extenders** – shows information on the active extenders of the terminal connected.

Slave terminal

- **Type** – shows the type of the StarPoint terminal connected.
- **Firmware** – shows the current firmware version of the terminal connected.
- **Extenders** – shows information on the active extenders of the terminal connected.

Digital interface diagnostic

- **Line state** – the parameter cannot be configured. It only shows the state of the first interface layer.
- **Number of SLIPs per minute** – shows the count of SLIPs. A SLIP is caused by different clocks on the PBX and the terminal. This value is updated every 6 seconds and represents a weighted average per minute.
- **BER per second** – the Bit Error Rate shows the count of incorrectly transferred bits on the interface during transmission. The value is updated every 6 seconds and represents a weighted average per minute.

Supported terminals


3.3 ASL

The ASL virtual port is used for connecting common analogue telephones or fax machines. This virtual port enables DTMF and pulse dialling detection and as well as DTMF or FSK using CLIP transmission. The parameters are divided into logical sections.

Stack status

This field displays information on the stack and its current status. With an ASL virtual port you can see the following statuses:

- null
- config
- on_hook
- off_hook
- error_stop
- error_start_req

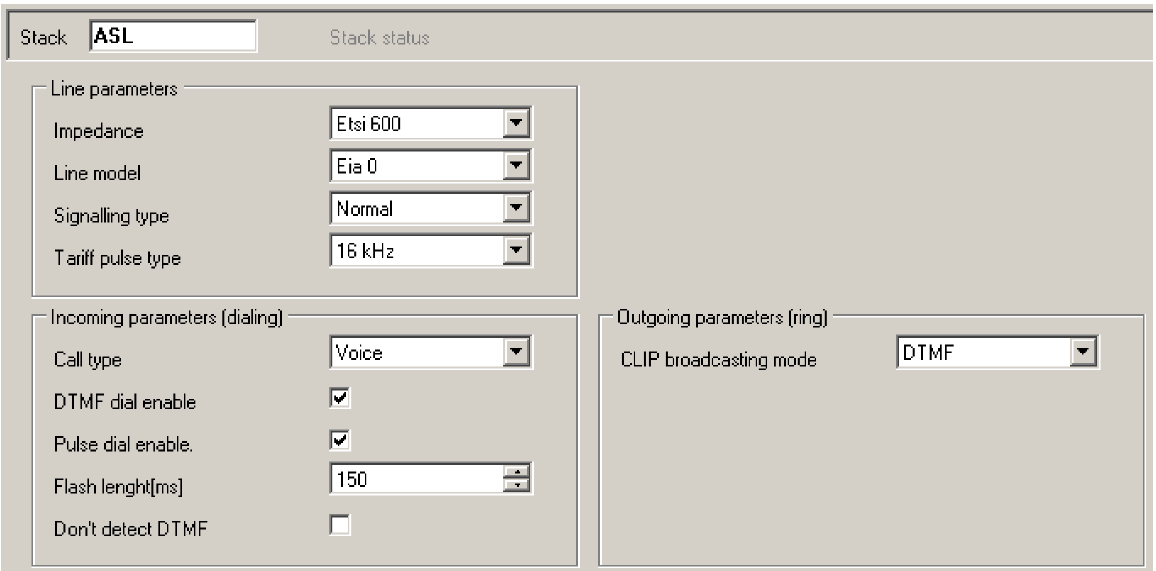


Figure: View of ASL Virtual Port Hardware Configuration

Line parameters

- **Impedance** – this parameter determines the impedance of the hybrid circuit according to preset models (User, ETSI 600, Germany and Real 600).
- **Line model** – this parameter provides further hybrid circuit parameters according to preset models EIA0 to EIA7 (e.g. EIA0 represents a 100m long line model).
- **Signalling type** – shows the type of active state signalling. Choose from **Reverse polarity**, **Tariff pulse** or **Simple**.
- **Tariff pulse type** – defines the tariff pulse sending source. Select **12 kHz**, **16 kHz** or **none**.

Incoming parameters (phone is dialling)

- **Call type** – determines the preferred type of communication on this port. Choose one of the **Voice**, **FAX**, **A3.1kHz Audio** and **56kb Modem** options.
- **DTMF dial enabled** – check this item to make the carrier detect DTMF dialling from an analogue phone.
- **Pulse dial enabled** – check this item to make the carrier detect pulse dialling from an analogue phone.
- **FLASH length [ms]** – the parameter sets the maximum time of the FLASH signal transmitted from a local phone to the PBX. The default value is 150 ms and the minimum value is 80 ms.
- **Disable port DTMF detection** – use this parameter to disable/enable DTMF detection using a detector on the board in order to save the internal PBX detectors.

Outgoing parameters (phone is ringing)

- **CLI broadcasting mode** – the parameter defines the preferred CLIP (Calling Line Identification Presentation) transmission type. The selections are **DTMF**, **FSK** and **none**.

3.4 CO

The CO virtual port is an analogue virtual port for connection of a CO (central exchange) analogue line. Since it has only a DTMF transmitter, it is unable to detect the DTMF. Therefore, route an incoming call directly to the final destination, or assign the DISA function to the virtual port to detect the DTMF symbols and route the call to the required destination. The parameters are divided into logical sections.

Basic

Stack status

This field displays information on the stack and its current status. With a CO virtual port you can see the following statuses:

- null
- config
- on_hook
- off_hook
- error_stop
- error_start_req

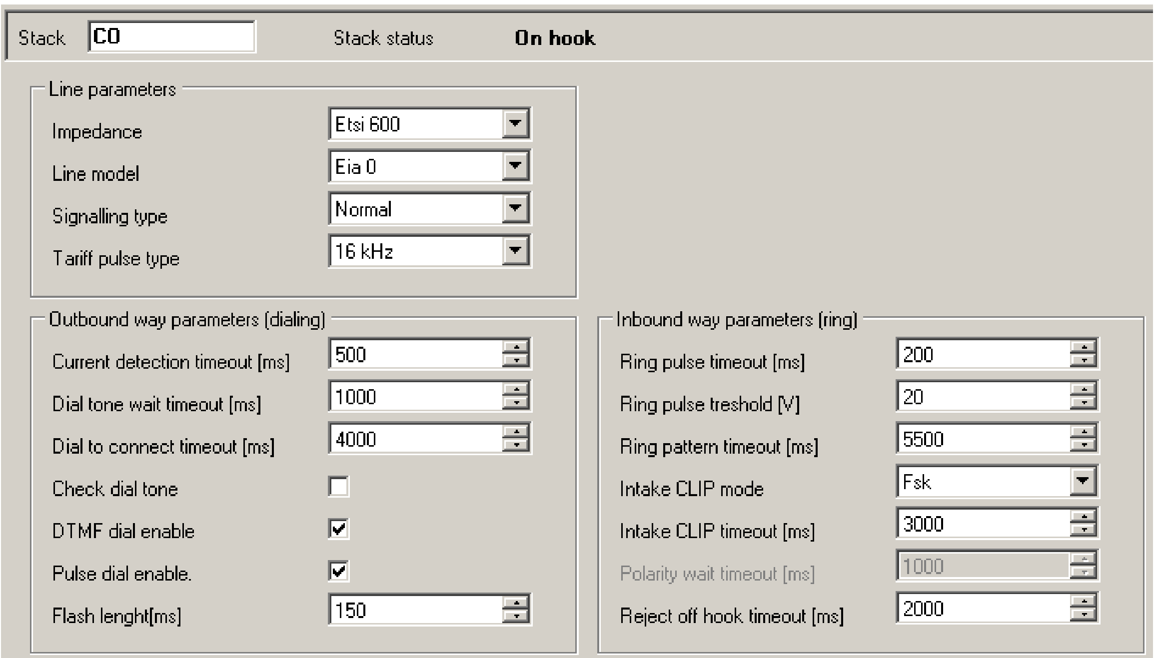


Figure: View of CO Virtual Port Hardware Configuration

Line parameters

- **Impedance** – determines the impedance of the hybrid coil according to preset models (User, ETSI 600, Germany and Real 600).
- **Line model** – determines further hybrid coil parameters according to preset models EIA0 to EIA7 (e.g. EIA0 represents a 100m long line model).
- **Signalling type** – shows the type of active state signalling. Choose one of the **Reverse polarity**, **Tariff pulse** or **Simple** options.
- Tariff pulse type – defines the tariff pulse sending source. Choose 12 kHz, 16 kHz or none.
- Dial tone –
- Congestion tone – this setting becomes available when you tick off the Check congestion tone parameter. Select the congestion tone mask for testing purposes.
- Check congestion tone – tick off the option to check the presence of the congestion tone. If the congestion tone is detected, the virtual port will pass into the On-hook mode.

Outbound way parameters (from PBX)

- **Current detection timeout [ms]** – here set the time for current detection on the picked-up carrier. If no current is detected within this timeout, a failure is reported.
- **Dial tone timeout [ms]** – here set the waiting time for dialling numbers to the carrier. Select the Check dial tone option to test the dial tone presence during this time.
- **Dial to connect timeout [ms]** – here set the maximum delay time for CO line dialling. Time monitoring is renewed after every digit entered. If no digit is detected within this timeout, the connection will be regarded as terminated and a new connection attempt will be made.
- **Check dial tone** – select this option to enable dial tone testing on the carrier for a period of time set in the **Dial tone wait timeout** parameter.
- **DTMF dial enabled** – tick off the option to enable DTMF dialling for the virtual port
- **Pulse dial enabled** – tick off the option to enable pulse dialling for the virtual port
- **FLASH length [ms]** – here set the maximum time of the FLASH transmitted from a local phone to the PBX. The default value is 150 ms and the minimum value is 80 ms.

Inbound way parameters (to PBX)

- **Ring pulse time [ms]** – this parameter sets the minimum time of the ring signal presence needed for ring detection. If the ring time is shorter than the preset value, ringing will be ignored.
- **Ring pulse treshold [V]** – this parameter sets the minimum ring voltage level needed for ring detection. If the ring voltage level is lower than the preset value, ringing will be ignored.
- **Ring pattern time [ms]** – this parameter sets the minimum period of time for alerting detection.
- **CLI reception mode** – the parameter defines the preferred CLIP (Calling Line Identification Presentation) reception type. Choose **DTMF**, **FSK** or **none**.
- **CLI reception timeout [ms]** – the parameter sets the CLI detection timeout as counted from the end of the first ring. This option is active only if the **DTMF** or **FSK** reception mode has been selected.
- **Polarity timeout [ms]** – the parameter sets the reverse polarity timeout. This option is active only if the **Reverse polarity** item has been selected for this virtual port.
- **Call reject timeout [ms]** – if the PBX needs to reject an incoming CO call, it has to pick up and hang up. Use this parameter to define the timeout for this action. If the action is too short, the other party will not recognise termination.

Expert

Chipset

- **Chipset Type** – this parameters determines the type of the chipset used. The SILABS_SI350 chipset is only supported at present.
- **Chipset Config** – activates one of the chipset configurations created.
- **New config** – helps create a new configuration for the selected chipset type.

Specific Chipset Config

- **Name** – sets the chipset type whose configuration is being set in the section.

> Do not change the above mentioned parameters unless absolutely necessary.

- **DCTerm** – sets the DC termination parameters (ringing voltage, minimum current, impedance). Hexadecimal values are used.
- **DAA Ctrl 5** – sets further parameters for analogue line matching (on/off-hook rate, low pass filter). Hexadecimal values are used.
- **ACIM** – helps set the proper impedance. Hexadecimal values are used.
- **Tx Gain** – sets the transmit gain.
- **Rx Gain** – sets the receive gain.

Refer to pages 78 to 81 of the User Manual for more details on the parameters.

The table below includes the ACIM settings including meanings.

ACIM [3:0]	Set	AC termination
0000	00	600 Ohm
0001	01	900 Ohm
0010	02	270 Ohm + (750 Ohm || 150 nF) and 275 Ohm + (780 Ohm || 150 nF)
0011	03	220 Ohm + (820 Ohm || 120 nF) and 220 Ohm + (820 Ohm || 115 nF)
0100	04	370 Ohm + (620 Ohm || 310 nF)
0101	05	320 Ohm + (1050 Ohm || 230 nF)
0110	06	370 Ohm + (820 Ohm || 110 nF)
0111	07	275 Ohm + (780 Ohm || 150 nF)
1000	08	120 Ohm + (820 Ohm || 110 nF)
1001	09	350 Ohm + (1000 Ohm || 210 nF)
1010	0A	0 Ohm + (900 Ohm || 30 nF)
1011	0B	600 Ohm + 2.16 μF
1100	0C	900 Ohm + 1 μF
1101	0D	900 Ohm + 2.16 μF
1110	0E	600 Ohm + 1 μF
1111	0F	Global complex impedance

3.5 GSM

The **Virtual ports – GSM** menu provides a list of all GSM virtual ports of the PBX. The parameters are divided into logical sections.

Basic

Stack status

This field displays information on the stack and its current status.

Network selection

- **Net type selection** – select the preferred network for module login. The following options are available:
  - **Any**
  - **Only GSM**
  - **Prefer GSM**
  - **Only UMTS**
  - **Prefer UMTS**
- **Roaming enabled** – use this option to enable roaming for a GSM virtual port.
- **Manual network selection** – if not checked, the SIM card tries to log into the preferred network automatically. If checked, enter the correct **Network code** to make the SIM card log into the selected network only. If the selected network is unreachable, the SIM card will not try to log into another network.
  - **Network code** – fill in a 5–digit international network code (e.g. T–mobile CZ=23001, O2 CZ=23002, Vodafone CZ=23003).
  - **Network name** – enter the name of the network as coded in the **Network code** parameter.
- **Cell selection** – select the network cell to which the module shall/may log in.
  - **Off** – the cell is selected automatically.
  - **Prefer selected** – the module tries to log in to the cell specified in the parameter below. If unsuccessful, the module tries other available cells.
  - **Only selected** – the module only tries to log in to the cell specified in the parameter below.
- **Cell ID** – set the network cell identifier for module login.

The following options are available under the right mouse button:

- **Known networks** – use this option to open a dialogue with the list of known networks and their international GSM codes. The networks are arranged according to countries.
- **Visible networks** – use this option to open a dialogue with the list of visible networks in the surroundings. By initiating the search you make your SIM card log out temporarily from the module.

Signal diagnostics

- **Signal measuring** – select this option to enable signal level measuring for a selected carrier.
- **Signal monitoring** – here enable signal level monitoring for a selected carrier. If the signal level drops below the value specified in the **Poor signal level** parameter, a red exclamation mark appears on the port in the **Hardware – Boards** menu and a red text is displayed in the upper stack status field. This poor signal status gets changed after the signal level exceeds the value specified in the **Good signal level** parameter. The interval between the values represents hysteresis.

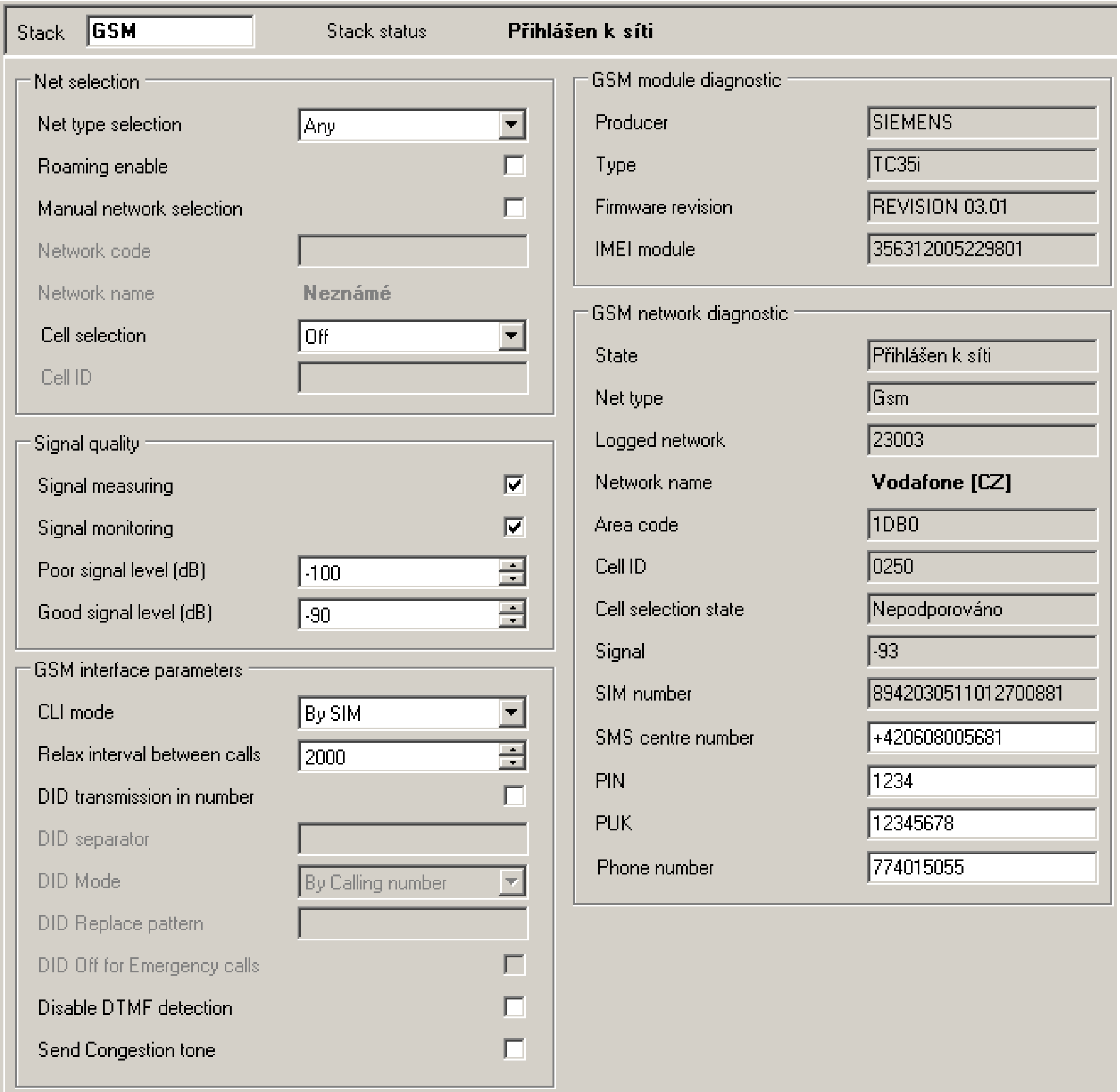


Figure: View of GSM Virtual Port Hardware Configuration

GSM interface parameters

- **CLI mode** – use this parameter to enable CLI restriction for the active SIM card. The following options are available:
    - **By SIM** – the SIM card default setting is respected.
    - **By calling number** – the calling user CLI setting is respected. If CLI is allowed, the SIM card uses CLI too. If not, the SIM card's identification is restricted.
    - **Presented** – SIM CLI is always restricted regardless of the SIM card or calling user settings.
    - **Restricted** – SIM CLI is always presented regardless of the SIM card or calling user settings.

> ⚠ This function must be supported by the network provider. Otherwise, calls with suppressed identification are rejected with a corresponding cause.

- **Relax interval between calls** – use this parameter to determine the idle period between two calls. This parameter only applies to calls going out from the PBX through the GSM carrier, not to incoming calls. During this time all outgoing calls are rejected with cause 34 – no circuit/channel available.
- **DID transmission in number** – use this parameter to enable a special direct dial–in transmission function within the called number. This function is supported by some networks only.
- **DID separator** – this character separates the called SIM card number and the DID (direct dial–in).
- **DID mode** – use this parameter to define how to work with the DID. Select one of the following options:
    - **By calling number** – if the calling user has CLI restriction, the calling number is not displayed in the DID (777982494#). If not, the calling number is displayed behind the DID separator (777982494#274).
    - **Always presented** – CLI is always presented in the DID regardless of the CLI setting (777982494#274).
    - **Restricted DID replace** – the mode is similar to the By calling number mode; the only difference being that, in the case of CLIR enable, the identifier specified in the DID replace pattern parameter is displayed behind the DID separator instead of the CLI (e.g. 777982494#888).
- **DID replace pattern** – specify the DID to be used as the caller's identification in the case of CLIR.
- **DID off for emergency calls** – disable the use of DID for specified emergency calls while the PBX is in one of the modes as described in detail in the Emergency Calls menu.
- **Diasable port DTMF detection** – use this parameter to disable/enable DTMF detection using a board detector in order to save internal PBX detectors.
- **Send congestion tone** – if this item is selected and the port requires tones, the PBX generates the congestion tone to the other party after the call end until the port receives the **Release** message or the **30s timeout** expires. If this item is not selected, the channel is closed practically on the telephone hang–up. If the port requires no tone, the channel is closed immediately too.

GSM modul diagnostic

- **Producer** – provides information on the board manufacturer.
- **Type** – provides information on the board type.
- **Firmware revision** – the software revision of the firmware uploaded into the board.
- **Module IMEI** – shows the detected IMEI code.

GSM network diagnostic

- **State** – shows the current port state for detection of network login problems if any. For example, **PIN REQUESTED** means that the SIM card requires the PIN code to log in. To log in successfully, you either enter the PIN or disable PIN requesting by the SIM card.
- **Net type** – displays the type of the network to which the module is currently logged in.
- **Logged network** – shows the international code of the network to which the SIM card is logged at the moment.
- **Network name** – shows the name of the network into which the SIM card is logged at the moment.
- **Area code** – shows the code of the area to which the SIM card is currently logged in.
- **Cell ID** – shows the ID of the cell to which the SIM card is currently logged in.
- **Cell selection state** – displays information on whether or not the given module supports manual cell selection.
- **Signal** – shows the current signal level (if activated). A low signal level may result in logout or call failure due to a high error rate.
- **SIM number** – shows the SIM card code detected.
- **SMS centre number** – fill in this parameter to enable SMS sending. In GSM networks, SMSs are not routed directly to the final destination, but through the provider's SMS centre. This is useful where an SMS cannot be delivered immediately (e.g. due to target phone unavailability). The SMS centre tries to deliver this message cyclically for a preset SMS validity time. This parameter is mostly automatically detected on the SIM card (preset by the provider). If not, fill it in manually.
- **PIN** – here enter the PIN code if it is required by the SIM card and has not been entered in the **SIM – SIM cards** cards menu for this SIM card.
- **PUK** – here enter the PUK code if it is required by the SIM card and has not been entered in the **SIM – SIM cards** cards menu for this SIM card.
- **Phone number** – this field is for information only. You can enter your SIM card telephone number for easier orientation. This parameter has no function.

Expert tag

AT commands

You can add AT commands here to set the module properties. These AT commands are executed upon every PBX restart or GSM/UMTS card restart. Use the arrows in the right–hand part of the section to specify the sequence of the commands. The **Timeout** column sets the time during which the answer to the command entered is awaited. The **Result** columns includes a brief statement on whether or not the command was successful. For specific answers see the **Answers for selected** section.

Answers for selected

Here find an answer to the AT command selected in the left part of the screen.

Net code locks

Use a special licence to restrict a module to a specific network. **This restriction is permanent.** Use a special licence again to unlock the status. The lock supports up to 8 networks.

Lock mode:

- **Unused** – the lock is not activated.
- **First enable** – the module is locked for the first network to which it logs in.
- **Enable** – the module may log in to the selected networks only.
- **Disable** – the module may log in to all networks except for those selected in the **Net code** section.

SIM num locks

Use a special licence to restrict a module to a specific SIM card. **This restriction is permanent.** Use a special licence again to unlock the status. The lock supports up to 4 SIM cards.

Lock mode:

- **Unused** – the lock is not activated.
- **First enable** – the module is locked for the first SIM card inserted.
- **Enable** – the selected SIM cards may log in to the module only.
- **Disable** – all SIM cards may log in to the module except for those selected in the **SIM num** section.

Audio parameters

This block of parameters helps you set audio profiles for various Siemens MC55/MC55i GSM module versions.

- **MC55 Audio Profile** – select the module version.
- **Tx gain** – set the transmitted audio signal intensity increase - **not implemented yet**.
- **Rx gain** – set the received audio signal intensity increase - **not implemented yet**.

USSD

USSD commands

In this section, you can enter the USSD commands (codes) for pre-paid SIM recharging or credit info display, for example. Click on **New** to enter the required command and on **Repeat** to execute the last-entered command. Press **Cancel** to terminate the current command processing. Refer to the Answer window for the command effect on the USSD code.

- **Network name** – the parameter shows the name of the network to which the SIM card is currently logged in.
- **Command** – shows the last-entered USSD command.
- **State** – shows information on the command processing.

3.6 SIP

SIP Gateway

The SIP Gateway virtual port is used for creating a trunk between two PBXs or connecting a PBX to the public network via a VoIP provider.

Stack status

This field displays information on the stack and its current status.

- **SOCK_TCP_ERROR** – the TCP socket has not been opened.
- **SOCK_UDP_ERROR** – the UDP socket has not been opened.
- **CREDS_IN_ERROR** – the authorisation server is unavailable.
- **CREDS_OUT_ERROR** – the authorisation client is unavailable.
- **REALM_CONFLICT** – the Realm collides with another port's Realm/Alias.
- **STUNNING** – the public IP address is being obtained from the STUN server.
- **STUN_TIMEOUT** – the STUN server is unavailable.
- **EXPIRED** – the public IP address validity has expired.
- **SIP_REGISTERING** – the gateway registration is in progress.
- **REG_TIMEOUT** – the REGISTRAR server is unavailable.
- **REG_NOT_AUTH** – the registration has not been authorised.
- **REG_REJECTED** – the registration has been rejected with an error.

Common parameters

- **Port** – here define the local gateway port to communicate with the other party.
- **Realm (Domain)** – define the domain over which the gateway communicates. The domain and ports specified here help route calls to the gateway. The Realm(Domain) + port items are checked in the Request–URI field for incoming INVITE messages. If the setting matches the gateway SIP, the packets are routed to the gateway. The INVITE messages whose Request–URI items are included in the Alias field are served too.
- **Via/Contact** – define the contents of the **Via** and **Contact** headers. The following options are available:
  - **IP address** – fill in the CPU IP address.
  - **FQDN** – fill in the PBX Hostname as entered for the PBX IP interface.
  - **NAT** – fill in the public IP address and NAT port for the opponent's sending of signalling messages. Packets are routed to the PBX according to the port routing and router IP address settings.
  - **STUN** – enter the STUN server address and port for finding the current address behind the NAT.
- **Authorisation required** – use this option to enable the other party's authorisation for incoming calls. User login data are used for this purpose. All logins are always used.
- **Send congestion tone** – enable transmission of the congestion tone from the PBX or network in case the opposite subscriber hangs up.

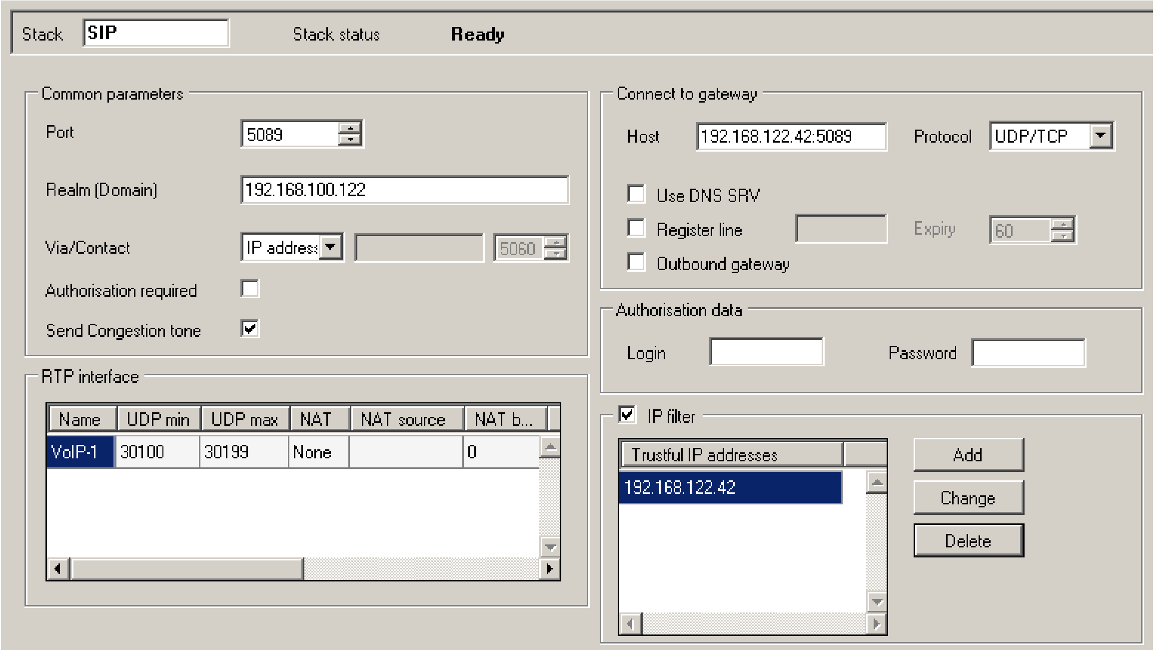


Figure: View of SIP Gateway Configuration Menu

Connect to gateway

- **Host** – here define the opponent's (provider's or other PBX's) IP address or DNS for trunk connection (call routing and registration request sending). If a port other than 5060 is to be used, it should be specified behind a colon (192.168.122.43:5071).
- **Protocol** – specify whether to use UDP and/or TCP, or just one of these protocols for transmission. If the NAPTR (Name Authority PoinTeR) option is selected, a query to the DNS is made first and the transmission protocol is selected depending on the reply. The **Use DNS SRV** parameter can only be used with this setting and a suitable DNS.
- **Use DNS SRV** – if this option is selected, a query to the defined DNS is made before the INVITE message is sent. The DNS reply defines two different call routing addresses. INVITE is routed to the first address and, if no reply comes after three INVITE sending attempts, the PBX sends INVITE to the other address included in the DNS reply.
- **Register line** – here enable line registration and specify the Caller ID. If a line is not registered, no call establishing requests are sent to it.
- **Expiry** – here define the registration expiry. The final value may be defined by the other party (e.g. shorter).
- **Outbound gateway** – if this option is selected, the **Contact** and **Record route** headers are ignored in replies and packets are routed directly to the address specified in the **Address** field.

Authorisation data

- **Username** – enter the username for login with authorisation.
- **Password** – enter the password for login with authorisation.

IP filter

The parameter helps you secure your PBX system against unauthorised call setup attempts via the given SIP gateway. Tick off this option to make your PBX process requests from trustworthy IP addresses only. Click on the buttons to the right of the IP address list or open the context menu in the IP address list using the right mouse button to add, remove or modify an IP address.

RTP interface

- **Name** – shows the name of the Ethernet interface (VoIP card) used.
- **UDP min** – here define the lower limit for the UDP ports used for RTP stream sending.
- **UDP max** – here define the upper limit for the UDP ports used for RTP stream sending.
- **NAT** – enable RTP stream routing through the NAT. If this selection is No, the opponent's RTP stream is sent to the VoIP interface. If a PBX is configured behind the NAT, one of the options in this menu has to be used for the VoIP interface to send a correct IP address to the WAN.
- **NAT source** – if you have entered the fixed IP address in the NAT column, now fill in the NAT IP address here for RTP streaming.
- **NAT base** – if you have entered the fixed IP address in the NAT column, now fill in the NAT port here for RTP streaming.

Codecs

- **Supported** – here find a list of supported codecs excluding the codecs that have been selected as Allowed.
- **Allowed** – here find a list of codecs to be used for communication on this virtual port. The context menu under the right–hand mouse button provides further **Codec setting** options.
- **DTMF according to RFC2833** – use this option to enable DTMF transmission according to RFC2833. With this option on, you can set the Payload type for DTMF transmission using the link below the name.
- **Fax T.38** – use this option to enable fax transmission according to the T.38 recommendation. If checked off, a link becomes available to the Advanced settings. The recommended setting is **TCF** – **Transfer**, **Error correction** – **Redundancy** and **No compression**.
- **FAX detection** – set whether 2N® NetStar shall detect FAX (send the re-INVITE message with T.38 in SDP) for Incoming and/or outgoing fax messages, Always or Never.

NetStar versions 2.6.0 and higher also support DTMF sending through SIP Info. This function requires no setting.

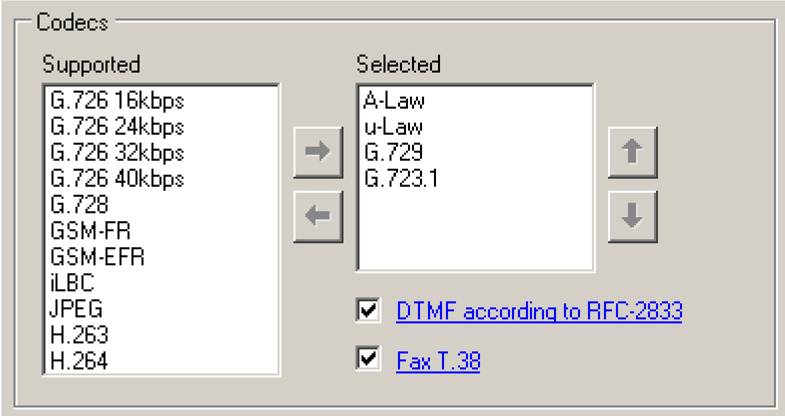


Figure: View of Codecs Setting Menu

SIP Proxy

The SIP Proxy virtual port is used for connecting SIP terminals to the PBX through terminal registration. All the parameters are divided into logical sections.

Basic

Stack status

This field displays information on the stack and its current status.

- **SOCK_TCP_ERROR** – the TCP socket has not been opened.
- **SOCK_UDP_ERROR** – the UDP socket has not been opened.
- **CREDS_IN_ERROR** – the authorisation server is unavailable.
- **CREDS_OUT_ERROR** – the authorisation client is unavailable.
- **REALM_CONFLICT** – the Realm collides with another port's Realm/Alias.
- **STUNNING** – the public IP address is being obtained from the STUN server.
- **STUN_TIMEOUT** – the STUN server is unavailable.
- **EXPIRED** – the public IP address validity has expired.
- **SIP_REGISTERING** – the gateway registration is in progress.
- **REG_TIMEOUT** – the REGISTRAR server is unavailable.
- **REG_NOT_AUTH** – the registration has not been authorised.
- **REG_REJECTED** – the registration has been rejected with an error.

Common parameters

- **Port** – here fill in PBX port for the SIP Proxy – terminal communication.
- **Realm (Domain)** – defines the domain over which the gateway communicates. The domain and ports specified here help route calls to the gateway. The Realm(Domain) + port items are checked in the Request–URI field for incoming INVITE messages. If the setting matches the SIP Gateway setting, the packets are routed to the gateway. The INVITE messages whose Request–URI items are included in the Alias field are served too.
- **Via/Contact** – here define the contents of the Via and Contact headers. The following options are available:
  - **IP address** – fill in the PBX IP address.
  - **FQDM** – fill in the PBX Hostname as entered for the PBX IP interface.
  - **NAT** – fill in the public IP address and NAT port for the opponent's sending of signalling messages. Packets are routed to the PBX according to the port routing and router IP address settings.
  - **STUN** – enter the STUN server address and port for finding the current address behind the NAT.
- **Authorisation required** – use this option to enable authorisation for all terminals. Logins and passwords of the users whose extensions are assigned to the given terminal in the **Extensions** tag are used for registration.
- **Send congestion tone** – enable transmission of the congestion tone from the PBX or network in case the opposite subscriber hangs up.

Proxy parameters

Registration validity – use this parameter to define the validity for terminal registrations. Every terminal has to send a new registration request upon expiry. The parameter range is 30 to 3,600s. The resultant registration term may be shorter than the value defined here (depending on the terminal setting).

RTP interface

- **Name** – shows the name of the Ethernet interface used.
- **UDP min** – here define the lower limit for the UDP ports used for RTP stream sending.
- **UDP max** – here define the higher limit for the UDP ports used for RTP stream sending.
- **NAT** – enable RTP stream routing through the NAT. If this selection is No, the opponent's RTP stream is sent to the VoIP interface. If a PBX is configured behind the NAT, one of the options in this menu has to be used for the VoIP interface to send a correct IP address to the WAN.
- **NAT source** – if you have entered the fixed IP address in the NAT column, now fill in the NAT IP address here for RTP streaming.
- **NAT base** – if you have entered the fixed IP address in the NAT column, now fill in the NAT port here for RTP streaming.

Codecs

- **Supported** – here find a list of supported codecs excluding the codecs that have been selected as Allowed.
- **Allowed** – here find a list of codecs to be used for communication on this virtual port. The context menu under the right–hand mouse button provides further **Codec setting** options.
- **DTMF according to RFC2833** – use this option to enable DTMF transmission according to RFC2833. With this option on, you can set the Payload type for DTMF transmission using the link below the name.
- **Fax T.38** – use this option to enable fax transmission according to the T.38 recommendation. If checked off, a link becomes available to the Advanced settings. The recommended setting is **TCF** – **Transfer**, **Error correction** – **Redundancy** and **No compression**.
- **FAX detection** – set whether 2N® NetStar shall detect FAX (send the re-INVITE message with T.38 in SDP) for Incoming and/or outgoing fax messages, Always or Never.

NetStar versions 2.6.0 and higher also support DTMF sending through SIP Info. This function requires no setting.

Terminals

This section is used for terminal management. If no terminal has been created, the VoIP phone cannot register to the SIP proxy. A registered phone is indicated by displaying the IP and MAC address for the connected terminal. Multiple phones may be registered to one terminal. In the case of an outgoing call, all of the registered phones are alerted until one of them answers the call. Incoming calls are identified according to the extensions that are assigned to the terminals in the **Extensions** tag.

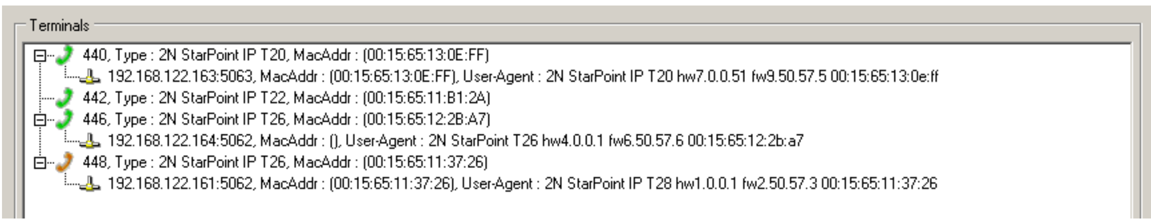


Figure: View of SIP Proxy Configuration

The following options are available under the right mouse button:

- **Add** – display a terminal adding dialogue. Select the **Name**, **Type** and **MAC address** for the terminal.
- **Edit** – edit the existing records.
- **Remove** – remove the selected record from the configuration.
- **Transfer MAC address to terminal** – transfer the MAC address of a registered terminal device into the terminal configuration.

The handset icon colour signals the terminal state: green means relax, red signals an active call, terminal alert or ringing outgoing call, and blue indicates an unknown terminal state or current state change.

Be sure to set the terminal type and MAC address correctly to make use of the **SIP Provisioning** function for the **2N® StarPoint** IP T2x phones.

Advanced settings

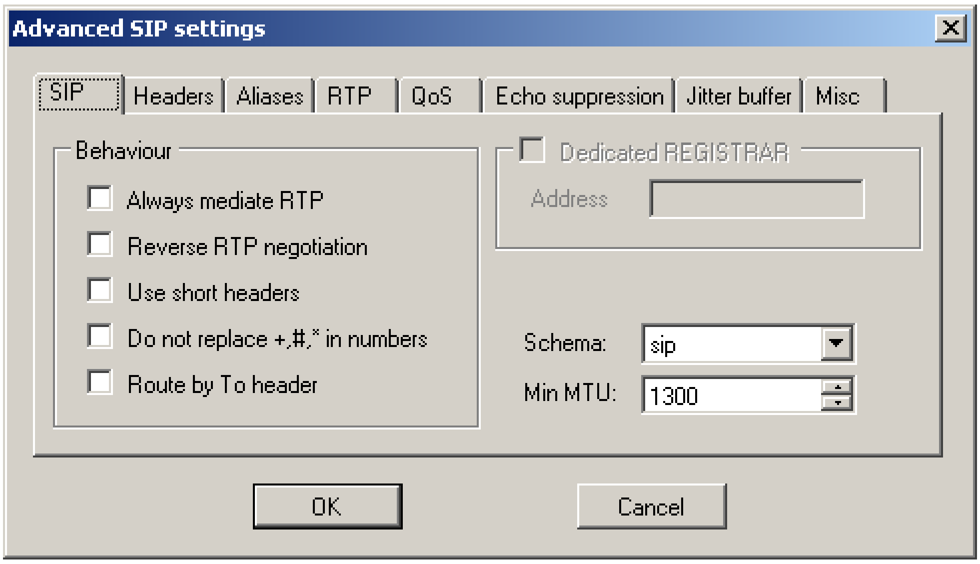


Figure: View of Advanced VoIP Parameter Setting Dialogue

SIP

- **Always mediate RTP** – if this option is checked off, the RTP stream is always routed through the PBX VoIP card. If not, the RTP stream is processed outside the PBX (in the case of VoIP – VoIP connection) and the PBX is reponsible for signalling only.
- **Reverse RTP negotiation** – use this option to define the way of codec negotiation. If this option is not checked, codecs are offered already in the **Invite** message.
- **Use short headers** – select this option to enable using short headers (e.g. From = f, To = t, Via = v) for data transfer minimisation.
- **Do not replace +,#,* in numbers** – these characters are replaced with corresponding strings %xx in numbers only if the option is not selected. If it is selected, they are sent.
- **Route by To header** – if this option is selected, the incoming calls of the given port are routed by the **To** header. In other cases (including the default setting!), the calls are routed according to the **Request URI** header.
- **Dedicated Registrar** – use this option to enable routing registrations from gateways to another destination.
- **Address** – the selected Registrar server address.
- **Port** – the selected Registrar server port.
- **Scheme** – set the **sip** or **tel** scheme in the "To" and "From" headers from the SIP. The **tel** selection is used for networks based on the numbering plan according to the E.164 recommendation.
- **Min. MTU** – set the packet size limit for obligatory TCP use in the UDP&TCP mode. The recommended maximum value is 1,448 bytes.

Headers

- **Complete domains** – use this section to specify the domain to be used for the **From** and **To** headers.
- **Send information** – **P–Asserted–Identity** – select this item to activate the P–Asserted–Identity header in the INVITE message for CLIR transmission. The opponent can thus obtain CLI information even if its CLIR (Calling Line Identification Restriction) is enabled. By default, CLIR is enabled on the SIP Gateway port (header active) and disabled on the SIP Proxy port (header inactive).

Aliases

This section enables to define further Realms(Domains) to be accepted. Those incoming calls (their INVITEs) will be accepted to the port whose Request–URI field shall match the SIP gateway or SIP proxy Domain and Alias settings.

RTP

- **DSP**– use this section for transferred data optimisation. When enabled, packets are not sent uselessly when the user does not speak. VAD stands for Voice Activity Detector.
  - **VAD off**
  - **VAD according to G.729 Annex B**
  - **VAD light**
- **Generate comfort–noise** – use this parameter to generate some noise into the call. Since analog line users are used to some background noise, similar noise is simulated here for comfort.
- **Mask lost packets** – use this option to activate lost packet masking optimisation.

QoS

The **TOS/DiffServ** section helps you set the outgoing packet parameters in order to define priority in processing by network elements.

- **SIP** – hexadecimal value of SIP packet priority.
- **RTP** – hexadecimal value of RTP packet priority.
- **Default values** – reset the default values for the two parameters. The default values are optimised for voice transmission.

Echo suppression

Use this tag to enable variable echo suppression modes.

- **Suppression off**
- **Profile G.168 8 ms**
- **Profile G.168 16 ms**
- **Profile G.168 32 ms**
- **Profile G.168 64 ms**
- **Profile G.168 128 ms**

- **Delay [ms]**
- **Adaptive suppression**
- **Non–linear processing**
- **Re–use coefficients**
- **Automatic control**

Jitter buffer

Use this tag to enable variable jitter suppression modes.

- **Delay [ms]**
- **Depth [ms]**
- **Automatic adaptation**
- **Short adaptation parameters**
- **Low limit [ms]**
- **High limit [ms]**
- **Threshold**

Miscellaneous

- **In–call mark receivin**
  - **Mode** – use this parameter to set the supported DTMF receiving method for calls.
- **Generating of INFO message**
  - **DTMF** – select one of the two available DTMF transmission modes using the SIP INFO method. The modes provide different formats of the DTMF character transmitting message.
- **Keep–alive**
  - **Interval** – defines the keep–alive packet sending interval. The default value is 10s.
- **STUN server** – allows the NAT clients (i.e. computers behind the firewall) to set up calls with a VoIP provider outside the local network.
  - **Host** – here enter the STUN server address (IP or domain name). The address is used in case the STUN IP method is selected in the RTP interface configuration. By default, **stunserver.org** is selected.
  - **Port** – use the parameter to define the port to be used for the STUN server. The default value is **port 3478**.

3.7 SMTP

SMTP

The **Simple Mail Transfer Protocol (SMTP)** is an Internet protocol used for e–mail transmissions across the Internet between clients and the server. The protocol provides deliveries via a direct sender–receiver connection.

E–mail messages can be sent via the SMTP client and received via the SMTP server in the NetStar PBX. See below for details. E–mail messages are routed by the PBX in the same way as SMS messages, i.e. using such objects as text routers and the **SMS routing** tag.

SMTP clients

The NetStar PBX provides more than one Ethernet interface. For communication with the SMTP server, however, the PBX always uses the CPU Ethernet interface. In the **Virtual ports – SMTP** menu you can create SMTP clients to log into the SMTP servers and send e–mail messages.

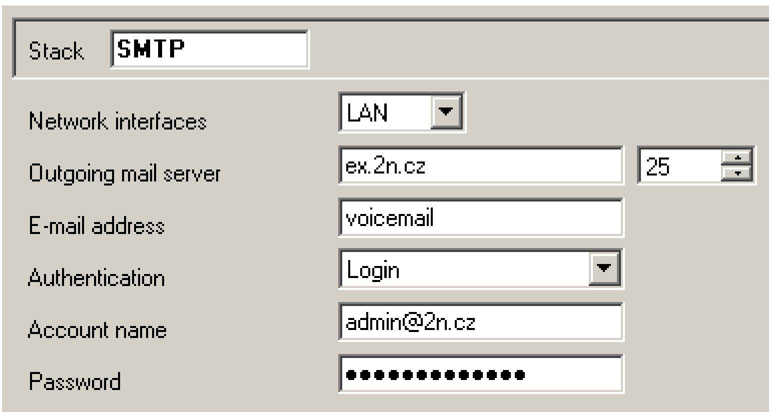


Figure: View of SMTP Client Configuration

- **Network interface** – here choose the network interface to be used for SMTP communication with the server. In this version you can use the CPU network interface only.
- **Outgoing mail server** – here enter the IP address of the SMTP server. If you use the DNS server, you can also use the domain name of your SMTP server.
- **Port** – here define the port to be used for communication with the SMTP server. Typically, port 25 is used.
- **E–mail address** – is used for identification of incoming messages within the SMTP server. Without a correct setup, the SMTP server will probably reject all connection establishment requests!
- **Authentication** – is used for choosing the type of authentication for access to your e–mail account on the selected SMTP server.
  - **None** –
  - **Plain** –
  - **Login** –
  - **Digest_MD5** –

- **Cram_MD5** –
- **Account name** – provides the name of the e–mail account registered by the selected SMTP server. Required by all methods.
- **Password** – account access password, required by all methods.

SMTP server (SMTPD)

The SMTP server processes incoming e–mails.

- **Port** – set the port on which incoming e–mails for this SMTP server are awaited. Two PBX SMTP servers may not have one and the same port.
- **Queue length** – set the count of e–mail messages to be queued and subsequently processed by the server (routed to final destinations in the PBX or resent to another interface). If you set 1, the server will not receive an e–mail processing request until it completes the preceding one.
- **Authorised group** – use this parameter to authorise incoming e–mail messages. The following options are available:
  - **Without authorisation** – incoming e–mail messages are accepted without authorisation.
  - **Any** – e–mail messages matching any PBX user login are accepted.
  - **Group of users** – e–mail messages matching user logins from a certain user group are accepted. If a superior group is selected, all subgroup users are included too.

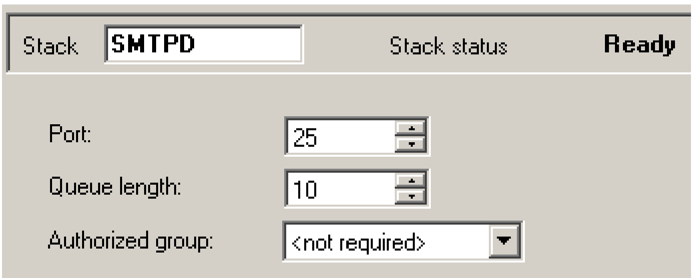


Figure: View of SMTP Server Configuration

3.8 Virtual Port Options

Introduction

The **Virtual ports** menu helps you configure all virtual port types and virtual ports. In the **Virtual ports – All** menu you can see all virtual ports regardless of their type. For easier orientation, the virtual ports are arranged according to port types and also colour–distinguished according to the stack type. To display a selected virtual port type use the **Virtual ports** submenus. By default, the following colours are assigned to virtual ports: **DSS1 BRI**, **DSS1 PRI**, **CORNET**, **ASL**, **CO**, **GSM**, **SIP Proxy**, **SIP Gateway**, **SMTP client**, **SMTP server**. These settings can be changed within the application setting as described in Chapter 1.2 About Application.

Creating virtual ports

By default, the database contains two basic virtual port types – **Default IN** and **Default OUT**. The virtual ports are created automatically in the **Hardware – Boards** menu. You can create more virtual ports and virtual port types manually using the following context menu options:

- **Add virtual port type** – use this option to create a new virtual port type. To assign a virtual port to a new virtual port type use the **Drag&Drop** function or the **Type** parameter in the **Basic** tag.
- **Add virtual port** – use this option to initiate a dialogue box for adding a new virtual port. Enter the virtual port name and choose the stack type from the available submenu list. The offerings depend on which menu you use. The **Virtual ports – All** menu includes a list of all stack types, but the **Virtual ports – Cornet** menu provides a list of Cornet stack types only. Manually created virtual ports are not assigned to physical ports automatically. They have to be assigned manually using the **Hardware – Boards**.
- **Delete** – use this option to delete a selected virtual port or virtual port type.
- **Rename** – use this option to rename a selected virtual port or virtual port type.
- **Copy** – use this option to make a copy of the selected virtual port or virtual port type retaining its settings (only the items that may not be identical are changed).
- **Assign name to unchanged according to physical port** – use this option to rename all the virtual ports that have not been renamed yet according to the physical ports they are assigned to.
- **Assign name to all according to physical port** – use this option to rename all virtual ports according to the physical ports they are assigned to.
- **Set parameters as Default IN** – this option is available for the virtual port type only and helps you set all the parameters for a new virtual port type as Default IN quickly.
- **Nastavit parametry jako na Default OUT** – this option is available for the virtual port type only and helps you set all the parameters for a new virtual port type as Default OUT quickly.

Moving records using the mouse, also called **drag & drop**, has been implemented in this menu for an easier transfer of existing virtual ports between the virtual port types.

The section below provides a description of the virtual port and virtual port type tags. All the tags and parameters defined below are common for all virtual port types. Some parameters or tags are omitted in some virtual ports because they have no sense there.

Basic

The **Basic** tag includes the following parameters:

- **Name according to the physic port** – use this option to rename a virtual port according to the physical port to which it is assigned. The name consists of the stack name and hardware address in the square brackets. In the event of a manual name change, the option keeps automatically unchecked.
- **Type** – use this parameter to assign a virtual port to a specific virtual port type, which represents another hierarchical level for some parameters.
- **Enable call without extension** – use this parameter to enable/disable answering of incoming calls without the CLI. This parameter is enabled by default. For example, it can be used where a terminal is connected to a certain physical port and no extension has been assigned to the virtual port.
- **Internal numbering plan** – here set the Calling Line Identification (CLI) subtype to **Internal**. **YES** is typically set for internal ports and **NO** for external ports. If **YES** is selected, no CLI normalisation is made and Mobility Extension terminals are not recognized.
- **Call on port is accounted** – an indicator (a = accounted) is inserted in the accounting sentence that is used for charging outgoing calls through this port. The accounting sentences are thus easily traceable by the accounting software.

CLI section

- **Identification tab** – use this parameter to assign an identification table to a virtual port. Choose any of the tables available in the **Routing – Identification tables** menu. The selected identification table is used for changing the Calling Line Identification (CLI) for outgoing calls through the corresponding virtual port.
- **Add prefix for external CLI** – use this option to assign a prefix to the virtual port as defined in the **Global data – Global parameters** menu. The prefix is then added to the Calling Line Identification for all external subtypes, but does not influence number assignment to a phone directory name. The prefix addition facilitates CallBacks for the virtual ports that do not support the number subtype (analog lines, SIP).

Keep number subtype

The parameter defines the final subtype of incoming and outgoing numbers for a virtual port, including the Calling Line Identification (CLI) and Called Party Number (CPN).

- **You can set the following:**
  - **Incoming CLI** – the parameter sets whether an incoming CLI will be retained or not.
  - **Incoming CPN** – the parameter sets whether an incoming CPN will be retained or not.
  - **Outgoing CLI** – the parameter sets whether an outgoing CLI will be retained or not.
  - **Outgoing CPN** – the parameter sets whether an incoming CPN will be retained or not.
- **Meanings of set values:**
  - **Default** – with this option settings from higher levels can be taken over.
  - **Replace unknown** – with this option the numbers are only normalised

that arrive in the PBX with the **Unknown** subtype. The other subtypes are retained. Normalising takes place as defined in the **Localisation** menu.

- **Replace always** – all incoming numbers are normalised.
- **Retain**– no number is normalised. The numbers are further processed with the subtype they arrive in the PBX with.

AutoClip routers

This section is used for assigning a selected AutoClip router to a virtual port. Assign the AutoClip routers for calls and messages separately but you can use one and the same AutoClip router. For details on AutoClip routers refer to Chapter 7.7 AutoClip Routers.

- **Calls** – here you can assign an AutoClip router for saving records on outgoing calls. To make the function work, assign the **AutoClip parameters** to the calling user in the **Routing – Users & groups** menu on the user or use group level. To assign the **AutoClip parameters** use the **Global data – Autoclip parameters** menu.
- **Messages** – here you can assign an AutoClip router for saving records on outgoing SMS. To make the function work, assign the **AutoClip parameters** to the calling user in the **Routing – Users & groups** menu on the user or use group level. To assign the **AutoClip parameters** use the **Global data – Autoclip parameters** menu. One and the same AutoClip router can be used both for SMS and calls.

Cause mapping

In this section, you can specify your own sets of causes to be used for signalling. To do this, use the **Global data – Causes – Cause mapping tables** menu. You can set a translation of a certain event into the given interface (SIP, GSM a ISDN) in the cause mapping table. Use these sets only if the predefined cause translations are inconvenient.

- **CP to stack** – define a specific cause translation for changes from the PBX to the virtual port. Hence, it is a change of a specific internal cause into any stack cause.
- **Stack to CP** – define a specific cause translation for changes from the virtual port to the PBX. Hence, it is a change of any stack cause into a specific internal cause.

You can choose a mapping table for each direction, **disable** the use of a table assigned to a virtual port type or use **Default** to enable the default table.

Name information sending

The parameters included in this section are intended primarily for the SIP Gateway virtual port. Depending on the selection of one of the two parameters below, information on the calling line can be inserted in the From field of the SIP INVITE message.

- **Find name in group phone book** – select a group in the telephone directory for

calling number and name matching.
- **Insert calling station name** – define whether the calling station name shall be added to the outgoing INVITE message.
- **Own channel count** – shows the count of voice channels that can be served by the virtual port.

Licences needed

In this section, you can check easily whether the **Mobility Extension** or **Call Recording** licence is required on the virtual port. If a licence is required yet absent or insufficient, it is in red letters here. If a licence is valid, the blue text **Valid licence** is displayed.

Properties

The Properties tag consists of a number of subtags, which are described in a separate chapter. This tag is exceptional because almost all of its parameters obey the hierarchical rules. For the hierarchy and parameter details refer to Chapter Setting Properties.

Progress info

The parameters in this tag help you enable/disable progress tones that are to be played back to the user. In some cases, the progress tones are generated by the PBX, in others they are transmitted from the network. The final effect depends not only on the user's virtual port, but also on a combination of the user's virtual port (mostly an internal PBX port) setup and the other party's virtual port (a public network port or another internal PBX port).

The port generates the network progress info to the opposite port:

- **Dial tone** – the port is a source of the network dial tone for the opposite port if the latter requires the dial tone.
- **Alert tone** – the port is a source of the network alert tone for the opposite port if the latter requires the alert tone.
- **Disconnect tone** – the port is a source of the network disconnect/congestion/busy tone for the opposite port if the latter requires the disconnect tone.

- **Setting options**
  - **Default** – provides fall–down to the next level (virtual port type).
  - **Yes** – enables use.
  - **No** – disables use.
  - **Conditionally** – if a tone is signalled by the network, the generated tone is played. If a tone is not signalled, the internal PBX tone is played. The **Reset condition** section is applied to this setting only. This setting is recommended especially for the SIP Gateway virtual port.

The port requests progress info from the PBX or the opposite port:

- **Dial tone** – the port requests the dial tone from the PBX or the opposite port that generates the dial tone.
- **Alert tone** – the port requests the alert tone from the PBX or the opposite port that generates the alert tone.
- **Disconnect tone** – the port requests the disconnect tone from the PBX or

the opposite port that generates the disconnect tone.

- **Setting options**
  - **Default** – provides fall–down to the next level (virtual port type).
  - **Yes** – enables use.
  - **No** – disables use.

Reset condition– enables playing of some PBX tones and some network tones for one call.

- **Parameters**
  - **Alert resets condition** – an incoming Alerting message resets the tone–generating condition and signalling of the played tone is awaited again.
  - **Connect resets condition** – an incoming Connect message resets the tone–generating condition and signalling of the played tone is awaited again.
  - **Disconnect resets condition** – an incoming Disconnect message resets the progress tone condition and signalling of the played tone is awaited again.
- **Setting options**
  - **Default** – provides fall–down to the next level (virtual port type).
  - **Yes** – enables use.
  - **No** – disables use.

The following examples are given for easier comprehension:

1. Suppose that user **A**'s phone is connected to an internal PBX port. Set the dial tone request for this port for user **A** to hear the dial tone after picking up the phone. If user **A** makes a call to user **B** connected to another internal PBX port and you want user **A** to hear the alert tone, set the alert tone request for user **A** 's port too.
2. Suppose that your PBX is connected to a public or private network that generates progress info. If you want the calling user to hear the alert tone, set the port used for calling into the public or private network to generate the alert tone and the user extension port to request the alert tone. Otherwise, the calling user would not hear the alert tone during outgoing calls. The disconnect tone is mostly generated by own PBX.
3. **Outgoing call to PSTN with conditioned generating**
4. Suppose that a call is going out to the PSTN, which generates the dialtone only. However, you want the user to hear the ringing and disconnect tones too. Therefore, set the **Conditioned option** in the **Generate tone** section and the **Alert resets condition** parameter at least in the **Reset condition** section for all the three tones for the port used for the outgoing call. Suppose that the three–tone requirement is set for the user's internal extension. Thus, when a CO line is seized, the user is played the PSTN dialtone first and then the internal PBX ringing tone after the dialling and ringing start (Alerting state signalling), because the PSTN generates no ringing tone. When the PSTN subscriber hangs up, the user is played the internal PBX disconnect tone.

Terminate call when PROGRESS_IND received – here enable call termination on selected ports in case the oppopnent signals progress tone playing during call setup. Moreover, you can define the call rejection cause. Two basic modes are defined for call setup termination:

- **Before alerting** – when the PROGRESS_IND message comes before the opponent's alert signalling.

- **During alerting** – when the PROGRESS_IND message comes after the opponent's alert signalling.

Overlap

Overlap is one of the Called Party Number (CPN) sending methods. If enabled, the CPN is not transmitted all in a SETUP message, but digit–by–digit in an INFO message.

The setup consists of the following parameters:

- **Overlap sending** – this parameter enables overlap sending in the port–to–PBX direction. It is primarily used for ISDN virtual ports.
- **Overlap receiving** – has not been implemented yet. The selection is inactive.
- **Overlap dialling** – has not been implemented yet. The selection is inactive.
- **First digit timeout [ms]** – this parameter sets the first digit dialling timeout starting at the moment of the microtelephone pick–up. When it expires, the user cannot go on dialling, obtains the disconnect tone and the call connection is terminated. The default timeout value is set to 14s
- **Next digit timeout [ms]** – this parameter sets the next digit dialling timeout restarting after each digit received. When it expires, the call establishment begins. The default timeout value is set to 6s. To initiate call establishing before the timeout, push the **#** button.

Extensions

The **Extensions** tag provides a list of extensions assigned to the virtual port. There are three forms of the tag depending on the virtual port type:

1. With the **BRI**, **PRI**, **SIP Gateway** and **SIP Proxy** virtual ports, the tag structure respects the presence of a terminal. To create a terminal, use the **HW** tag. Terminals are used for authorisation, MSN numbers and extension assignment. The **Extensions** tag consists of three parts. The first window from the left includes a list of terminals assigned to the virtual port. If you have not created any terminal, you can use the **Default** one. The central window provides a list of extensions assigned to the selected terminal. The field on the right–hand side of the menu helps you select a extension to be active among multiple extensions assigned to the terminal. You can make calls to all of the extensions, but all outgoing calls from this terminal are identified as the active extension and accounted to this active extension too (except for the **Private call from my extension** service).
2. With the **Cornet** virtual port, the situation is similar. The difference is that extensions are assigned as Master or Slave terminals because you can connect just one terminal to a Cornet port. If multiple extensions are assigned to this virtual port, specify the active extension (one for Master and one for Slave).
3. With the **ASL**, **CO** and **GSM** virtual ports, the situation is the simplest. The tag has two parts only. One is used for extension assignment and the other for active extension specification.

Free minutes/SMS

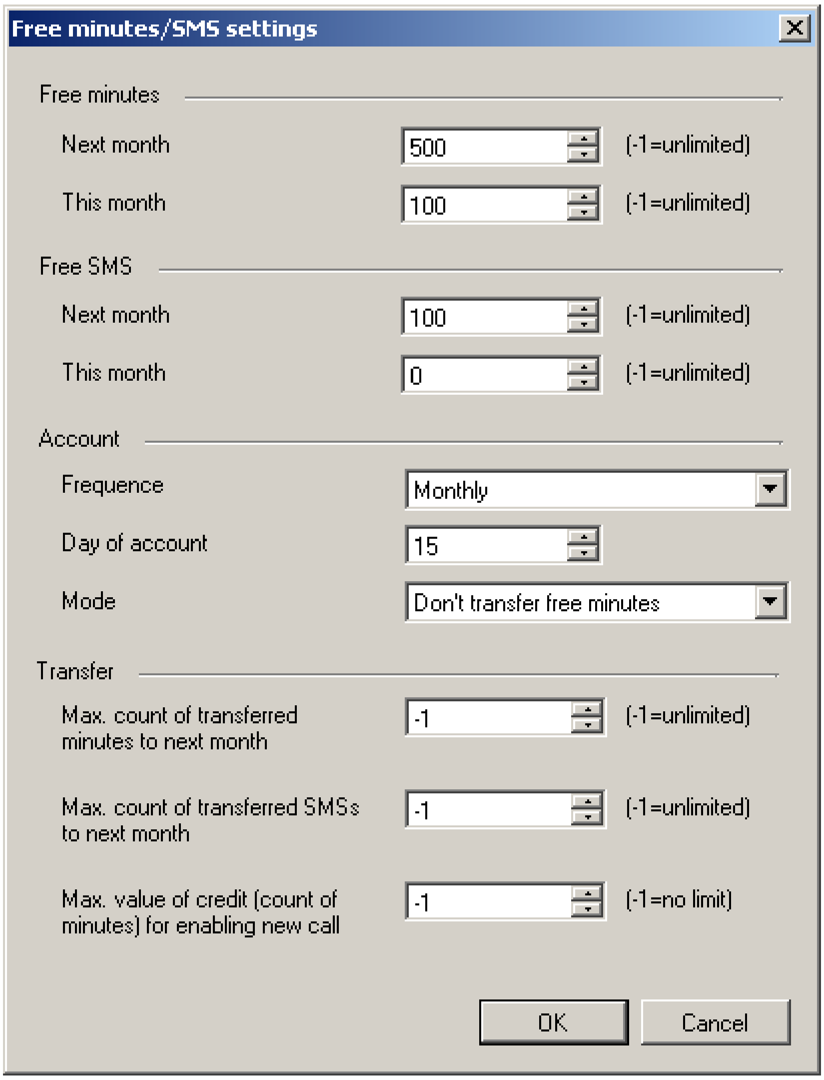
The tag helps you set free minutes and SMS for a selected virtual port.

Select tariff rate

Click on the **Set free minutes/SMS** button to display a dialogue and select one of the tariff rates as defined in the **Accounting and tariff rates** menu. In addition, you can assign here a setting to the selected virtual port tariff rate as defined earlier for any other virtual port. To change the tariff rate if necessary, use the **Used tariff rate** optio n. If you do so, you will lose all data saved on free minutes with the given tariff rate via this virtual port. To cancel the virtual port tariff rate, push the **Cancel free minutes/SMS** button.

Free minutes/SMS settings

Once a tariff rate is selected, the tariff rate credit rows are displayed in this section. Click on a row to display a setting dialogue for the count of free minutes, SMS messages and other credit parameters for the given virtual port. See the figure below for the dialogue.



The table includes columns with the following meanings:

- **Credit name** – the credit name as defined during tariff rate creation.
- **Free minutes for month** – the column includes the count of free minutes per

month for the given virtual port. This count is credited to the given virtual port at the beginning of the accounting period. If the free minute count changes within a month, the port credit is not increased until the beginning of the next accounting period unless provided otherwise in the setting dialogue.

- **Free minutes for this month** – the column shows the current count of free minutes to be used in this month. The value includes free minutes transferred from the previous accouting period if any.
- **Spent minutes** – displays the current count of minutes spent in the accounting period.
- **Free SMS for month** – the column includes the count of free SMS messages per month for the given virtual port. This count is credited to the given virtual port at the beginning of the accounting period. If the free SMS count changes within a month, the port credit is not increased until the beginning of the next accounting period unless provided otherwise in the setting dialogue.
- **Free SMS for this month** – the column shows the current count of SMS messages to be used in this month. The value includes free SMS transferred from the previous accouting period if any.
- **Spent SMS** – displays the current count of SMS sent in the accounting period.
- **Account** – use this option to set the billing frequency, i.e. the accounting period length. On this date, the free minute and SMS counts are increased according to the selected transfer mode. The mimimum values are set in the Free minutes for month a Free SMS for month columns.
- **Mode**– use this option to select the method of transfer of old free minutes into the next accounting period.
  - **Do not transfer** – no free minutes and/or SMS are transferred.
  - **First use new** – old free minutes and SMS are transferred but new ones are used first. Unused units older than one month are not transferred.
  - **First use transferred** – old free minutes and SMS are transferred and new ones are not used until these old units have been exhausted. Unused units older than one month are not transferred.

Files

The menu displays the current files with records of calls via the selected virtual port or virtual port type. The menu consists of a simple table with five columns with the following meanings:

- **Name** – name of the locked file.
- **File type** – type of the file.
- **Created** – the moment of file creation.
- **Valid for** – the file locking time, in other words a file storing time in a physical storage. When this time elapses, the file will be deleted.
- **Size** – size of the file.
- **CLIP Scheme** – calling number scheme.
- **CLIP Type** – calling number type.
- **CLIP Number/URI** – calling number or URI.
- **CPN Scheme** – called number scheme.
- **CPN Type** – called number type.
- **CPN Number/URI** – called number or URI.

Moreover, the context menu provides the following record handling options:

- **Save** – transfers the file from a storage to the PC.
- **Listen** – plays the selected file.

- **Remove** – removes the selected file from a storage.
- **Remove all** – deletes all files from a selected storage.

Stack

The **Stack** tag is described in Chapter 3. Virtual Ports depending on the stack type.

4. SIM

Here is what you can find in this chapter:

- 4.1 SIM Cards

4.1 SIM Cards

The **Virtual ports – GSM – SIM** menu includes a list of all PBX SIM cards. This menu is opened automatically whenever the SIM card is inserted in the PBX and the parameters filled in by the user (e.g. PIN) are used automatically for any future system detection of the SIM card. The menu includes two tags.

Basic

- **Card serial number** – this parameter shows the SIM card identification code detected. The NetStarAdmin uses this code automatically for SIM cards identification in the list on the left–hand side of the menu.
- **PIN** – Personal Identification Number – here insert the SIM card PIN if requested to be shared by the user and the authentication system. If the PIN is requested yet not entered or entered incorrectly, the SIM will not be logged into the network.
- **PUK** – Personal Unblocking Key – here insert the PUK to unblock the SIM card in case you have entered three incorrect values of the PIN code.
- **SMS centre number** – fill in this parameter to enable SMS sending. In GSM networks, SMSs are not routed directly to the final destination, but through the provider's SMS centre. This is useful where an SMS cannot be delivered immediately (e.g. due to target phone unavailability). The SMS centre tries to deliver this message cyclically for a preset SMS validity time. This parameter is mostly automatically detected on the SIM card (preset by the provider). If not, fill it in manually.
- **Phone number** – this field has an informative character only. You can enter your SIM card telephone number for easier orientation. This parameter has no function.
- **Name** – this field has an informative character only. It helps distinguish multiple SIM cards.
- **Port** – this field has an informative character only. It shows the physical port where the SIM card is being detected at the moment.

Free minutes/SMS

Use the tag to set the count of free minutes and SMS mesages via the selected SIM card. Refer to the Virtual Port Options menu in the Free minutes/SMS section for details on controls and tables.

5. Network

Here is what you can find in this chapter:

- 5.1 Network Interface
- 5.2 Service Settings
- 5.3 Supervision Services
- 5.4 DB connectors

5.1 Network Interface

The **Network – Network Interfaces** menu helps you manage all network interfaces available in the PBX. In addition to the CPU interface, there are Ethernet interfaces of VoIP boards. The bit rate of all the interfaces is 10/100 Mbit/s. These interfaces are used for communication with the PBX and SMTP clients, for signalling and RTP streams of VoIP calls. Having been opened, the **Network – Network interfaces** menu displays a list of Ethernet interfaces of the PBX on the left and selected interface settings on the right. With the CPU interface, the options are as follows:

Get IP address from DHCP server – use this option to enable obtaining IP settings from the DHCP server automatically. In this case the following sections are inactive.

- **Use following IP address**– use this option to enable the following static IP address and DNS server setting sections.
  - **IP address** – defines the static IP address for this interface.
  - **Subnet mask** – defines the subnet bit mask.
  - **Default gateway** – defines the IP address of the router or PC through which the PBX communicates outside the LAN.
- **DNS server addresses**
  - **Preferred server** – defines the IP address of the primary DNS server.
  - **Spare server** – defines the IP address of the secondary DNS server.
  - **DNS HostName** – defines the PBX Host Name.
  - **DNS Domain** – defines the PBX Domain Name.
- **Description** – this field has an informative character only.
- **Producer** – this field has an informative character only.

> You will lose connection with the PBX whenever you change the IP address. We recommend you to change the IP address via the console menu on the serial interface.

5.2 Service Settings

Here is what you can find in this section:

- Time Synchronisation (NTP)
- TFTP Root Storage
- TCP-IP Communication port
- System Services
- DHCP Server
- Directory Service (LDAP)

Time Synchronisation (NTP)

The menu helps you define the NTP server to be used for time synchronisation by the PBX. After checking the option in the upper menu part, enter the IP address or domain name of the existing NTP server into the field under the checkbox. After saving the data, the PBX will try to synchronise time with the preset NTP server. The result of this action is always shown in the **Synchronisation result** field together with information about the next planned synchronisation attempt.

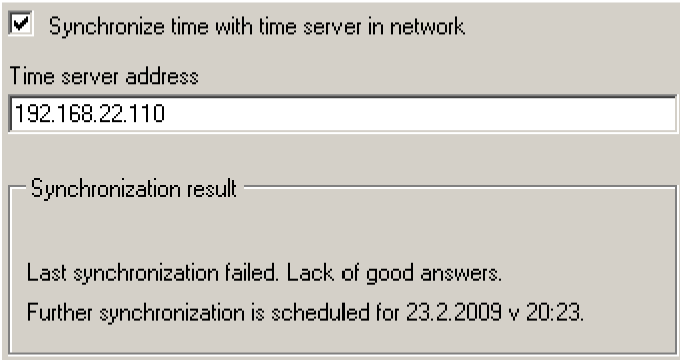


Figure: View of Time Synchronisation Menu

TFTP Root Storage

What Is TFTP?

The Trivial File Transfer Protocol is a very simple file transfer protocol, containing just basic FTP functions. The TFTP works over the connectionless **UDP** protocol. A single file can only be transferred through one connection. A single packet is only present in the network during communication. Having sent a packet, the program waits for confirmation and only then sends another one. Due to such simplification, the protocol provides just a **low transmission rate** to lines with a high latency. The TFTP uses **port 69**.

It has some restrictions and differences compared with the FTP:

1. Directories cannot be browsed through.
2. No user login and/or password entering are possible.
3. Can be used for data reading and/or remote writing only.
4. Supports the following three different transfer modes:
5. **netascii** – for an ASCII text with modifications from the Telnet protocol
6. **octet** – for raw binary 8–bit data
7. **mail** – for e–mail sending; this mode should not be used any longer
8. The maximum size of the file to be transferred is 32 MB.

TFTP in NetStar

In NetStar, the TFTP storage is used as a root directory for the TFTP server where the files provided by the TFTP server (which is part of every NetStar PBX) to TFTP clients are located. The typical TFTP client is an IP phone, which requests a configuration, telephone directory or new firmware from the TFTP server. The TFTP server searches this directory and provides the file, if found, to the client.

Using TFTP in NetStar

Basically, three situations may occur, for the first two of which the phone directory source has to be selected in the **SIP Phone Directories** menu:

1. NetStar receives a downloading request for the **gs_phonebook.xml** file, generates the file in the **GrandStream** telephone format from the selected source and sends it.
2. NetStar receives a downloading request for the **tftpPhoneBook.xml** file, generates the file in the **StarPoint IP Txx** telephone format from the selected source and sends it.
3. NetStar receives a downloading request for the **<MAC address>.cfg** configuration file and generates it for each **2N® StarPoint** IP T2x phone based on the MAC address.
4. NetStar receives a downloading request for another file, searches the TFTP storage for the file and sends the file if successfully found.

Configuration

The context menu provides the following options:

- **Refresh** – use this option to refresh the root storage for updated view.
- **Delete** – here remove a file from the root storage.

- **Rename** – here rename a file within the root storage.
- **Add file** – here add a PC file to the root storage.
- **Save file** – here save a root storage file into your PC.
- **The meanings of the table columns are as follows:**
- **Name** – displays the file name within the root storage.
- **Size** – displays the size of the file added.
- **Changed** – displays the date and time of the last file update.
- **Attributes** – gives additional information on the file.

Example for StarPoint IP T28

Log in to the telephone web interface (default login data: admin, admin) and move to the **Phone directory** tag. Here select the **Remote phone directory** middle link in the upper part. Enter the following string into one of the fields: **tftp://PBX_IP_address/tftpPhoneBook.xml**. Save the data. Go to the **Users – Phone directories – SIP phone directories** menu in **2N® NetStar** and select the phone directory source (group/user). Now push the directory access button on your phone to download the directory from the PBX.

TCP-IP Communication port

The **TCP/IP Communication port** menu is used for management of ports through which you can access your PBX. Basically, you can only **Add** or **Remove** a port in this menu, enabling/disabling the authorisation requirement. It is only port 6992 that requires authorisation after initialisation.

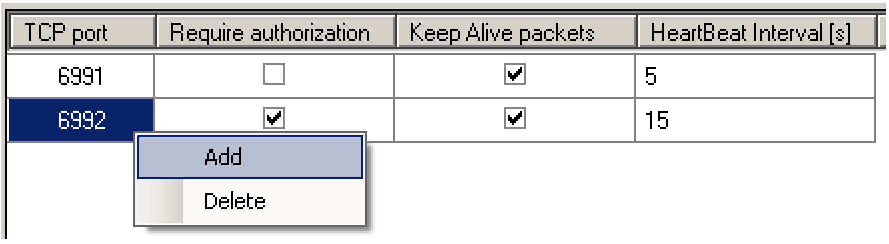


Figure: View of TCP/IP Communication port Menu Configuration

If the PBX is accessed through a password-protected port, the **Database** tag for direct configuration is not displayed for the user or administrator by default. To display it, assign the Read and Write rights to the user or administrator using the **Users – Users Rights** menu, or use a PBX port without authentication. In such case, however, you expose your PBX to the risk of unauthorised access. The database access is unnecessary for common configuration needs and should be granted to experienced technicians only.

As shown in figure above, you can define more parameters for each port (except for authorisation requirement).

- **HeartBeat packets** – enable/disable sending of keep-alive packets on a port to keep communication.
- **HeartBeat interval** – define the time interval between the keep-alive packets.

System Services

Communication ports

Ports 22 (SSH) and 23 (TELNET) are closed for security reasons in the default configuration in the firmware version 2.3.0 and later. To open them use a console or this configuration tool. Enter the "root" login for Telnet or SSH. The password is not defined by default. You are strongly recommended to set the password! The password setting command syntax is as follows:

- passwd root <Enter>
- Changing password for root
- New password: <Enter>
- Retype password: <Enter>

The password will be changed regardless of the original password. If you forget your password, you can change it any time in the same way. All you have to know is your console login name.

Use the "Admin" login for the console and password "2n" (by default).

> If you, despite recommendations, use the TELNET and SSH protocols for login to the NetStar PBX, any software warranty provided by the manufacturer shall be null and void. The system access is logged and intended for servicing purposes only.

Menu System

To open or close a port using the configuration tool, go to the **Network – Service setting – System Services** menu and use the **Enabled** or **Disabled** options. The meanings of the options are as follows:

- **Internal server of Assistant** – use this option to open or close access to the Assistant web server (user web application).
- **Telnet server** – use this option to open or close access to the system via the Telnet protocol.
- **SSH server** – use this option to open or close access to the system via the SSH protocol.
- **Trace level** – you can enable the Linux level for trace message displaying. This option is especially useful if you need to supervise the PBX remotely with a poor connection.
    - **All** – all available trace messages. Basic settings.
    - **None** – discontinue all trace messages.
    - **Call mng** – messages related to call routing by the PBX.
    - **Stack** – just messages related to call processing on the PBX interfaces (GSM, ISDN, SIP, analog).
    - **HEAP** – this option is reserved for listing of PBX memory capacity messages in special development firmware versions only.

DHCP Server

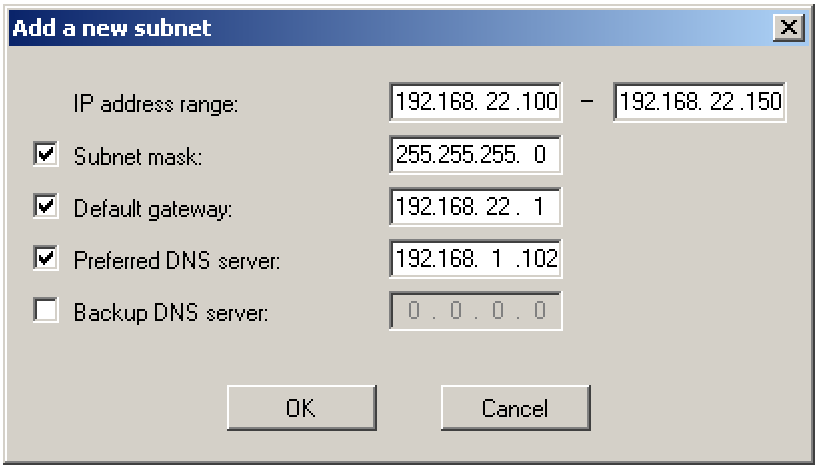
The DHCP server is used exclusively for assigning IP addresses and other parameters to SIP terminals with the specified MAC address in NetStar.

The menu consists of two basic sections. A field for setting ranges of subnet IP addresses to be assigned is in the left part of the screen. More parameter settings for the selecetd subnet are available on the right.

Subnet

The context menu provides the following options:

- **Add new subnet** – use this option to display a dialogue to define the required subnet parameters, see below.
- **Change range** – use this option to edit the existing subnet range. It has the same function as a double click on the selected subnet.
- **Delete subnet** – use this option to delete the selected subnet from configuration.



You can specify the following in this dialogue:

- **IP address range**
- **Subnet mask**
- **Default gateway**
- **Preferred DNS server**
- **Backup DNS server**

The first obligatory step is to enter the range of IP addresses. The remaining parameters need not be filled in immediately if you unselect them. You can edit the parameters later in the **Subnet properties** section.

Subnet options

The context menu provides the following options:

- **Add subnet mask** – the option is only active for subnets with no mask defined so far.
- **Add default gateway** – the option is only active for subnets with no default gateway defined so far.

- **Add DNS server(s)** – the option is only active for subnets with no DNS defined.
- **Edit value** – use this option to edit the existing values. It has the same function as a double click on the selected parameter.
- **Remove option** – use this option to remove a parameter from the selected subnet configuration.

Directory Service (LDAP)

The LDAP will be launched in one of the following versions of the 2N® NetStar PBX.

5.3 Supervision Services

Here is what you can find in this section:

- Remote Control (SNMP)
- Event Reporter

Remote Control (SNMP)

What is SNMP?

The **Simple Network Management Protocol (SNMP)** forms a part of the Internet protocol suite as defined by the Internet Engineering Task Force (IETF). The SNMP is used in network management systems for data acquisition and network monitoring for administration purposes. It consists of a set of network management standards, including the Application Layer protocol, a database schema and a set of data objects. The SNMP is available in three versions. Compared with the first version, version 2 is enhanced with authentication and version 3 with encryption. A majority of today's devices support the SNMP version two.

There are two sides in the SNMP communication – the monitoring one and the monitored one. These parts can run on separate physical devices or within one piece of equipment. The monitored side is often called **Agent** and the monitoring side **Manager**. The monitoring side flexibly collects information on the system state. The **Manager** sends requests to the **Agent**, mostly requesting some system state information. The **Agent** provides responses to the **Manager**. The **Agent–Manager** communication is often marked as an **SNMP operation**.

OID alias **Object Identifier** is an identifier used for explicit identification of each value in the SNMP communication. The OID is composed of a dot–separated sequence of numbers where each dot represents one level of the OID tree structure. The numerical identifications in subtrees are not unique and that is why the OID is always sent as a whole string. Each company and each of its SNMP supporting devices has an international OID of its own.

MIB alias **Management Information Base** is used for translation of OID strings into a more comprehensible text. The **MIB database** can be extended to include more **MIB files**.

Users

The SNMP v3 is a user oriented communication protocol. The user created in this part of configuration corresponds to the **USM** (User Security Model) in the SNMP v3 and to **Community** in the other versions. In addition to standard options **Add**, **Delete** and **Rename** user, the **Default** option is available, which helps introduce the default SNMP setting, including creation of the **public** user, right line **Unrestricted** and filters **Internet** and **NetStar Traps**.

- **Authentication** – here define the password and way of encryption for authentication.
  - **Protocol** – use the **MD5** or **SHA** methods to secure your password.
  - **Password** – here enter the user password.
- **Privacy** – here define the password and way of encryption for data transmission.
  - **Protocol** – use the **DES** or **AES** methods to secure your transmission.
  - **Password** – here enter the encryption password.
- **Access** – here assign rights to a selected user using the list of available rights.

Rights

- **Name** – define the name of the right to be created. This name is displayed in the **Users** selection.
- **Context** – use a text string to identify the SNMP module within the client address. This parameter need not be filled in.
- **Full match context** – use this option to enable requirement of full match including context. It is mostly unnecessary.
- **Security model** – in this parameter choose either a specific security model (SNMP v1, SNMP v2c, USM = SNMP v3) or the **Whatever** option. Any selection has to be supported by the other party too since no communication feedback is available.
- **Minimum security level** – the parameter offers three different models:
  - **Authentication and privacy**
  - **Without authentication and privacy**
  - **Authentication only**
- **Read filter** – use this parameter to set the Read filter by choosing an item from the list of available filters in the **Filters** tag. The filter restricts access to the PBX information for selected users.
- **Write filter** – use this parameter to set the Write filter by choosing an item from the list of available filters in the **Filters** tag. The filter restricts writing within the PBX for selected users.
- **Notify filter** – use this parameter to set the Notify filter by choosing an item from the list of available filters in the **Filters** tag. The filter restricts notifications from the PBX for selected users.

Title	Context	Full match context	Security model	Minimal security level	Read filter	Write filter	Notify filter
Internet		☐	Whatever	Without authentication and privacy	Internet	Internet	Internet
Restricted		☐	User's (USM)	Authentication and privacy	Restricted	Restricted	Restricted

Figure: View of SNMP User Right Setting Menu

Filters

- **Derive** – use this option to create a filter with the same setting as the currently selected one has.
- **OID root** – use the parameter to set the OID tree root to be used as a base for filter setting. You can view the OID structure in a tree or an alphabetical list.
- **Exception** – use the parameter to change the meaning of a filter rule. If this option is not checked, the defined OID subtree is used. If it is checked, the use of a subtree from this row is denied. With this parameter you can specify that the whole section 2.1 will be used except for subsection 2.1.3.
- **OID subtree** – here choose restrictions for a subtree. If the row is empty, the whole of the above specified OID root is used. Every filter should have one rule at least, even an empty row. It is because the filter compares rules with the subtrees instead of the OID root. If some subtrees overlap, the most common (the shortest OID) rule is applied.

MIB files

- **Add** – use this option to add a selected MIB file to the MIB database.
- **Delete** – use this option to delete a selected MIB file.
- **Recompile** – use this option to recompile a selected MIB file.
- **File** – this column shows the path to the MIB file source. This path is relevant for the **Recompile** option.
- **Status** – this column shows the current status of a MIB file. The options are **Compiled**, **Not compiled** and **Not found**. The MIB file statuses are also indicated by the icons on the line beginnings as shown in Figure 2.
- **Additional information** – displays additional information.

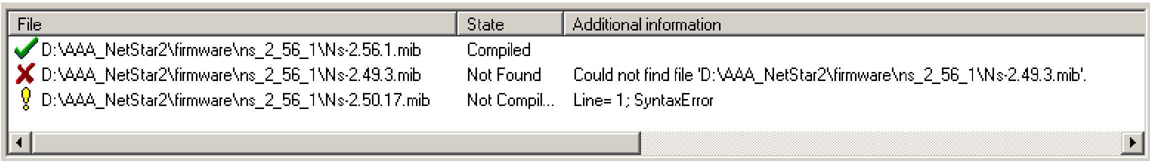


Figure: View of MIB File Management Section

Default Notify filter options according to RFC3415

- **Internet access**:
  - subtree 1.3.6.1
- **Restricted access**:
  - **System** – subtree 1.3.6.1.2.1.1 according to **RFC3918**
  - **SNMP** – subtree 1.3.6.1.2.1.11 according to **RFC3918**
  - **snmpEngine** – subtree 1.3.6.1.6.3.10.2.1 according to **RFC3411**
  - **snmpMPDStats** – subtree 1.3.6.1.6.3.11.2.1 according to **RFC3412**
  - **usmStats** – subtree 1.3.6.1.6.3.15.1.1 according to **RFC3414**

When you choose the Internet filter, the following traps are transmitted upon the PBX restart:

1. ColdStart is sent notifying of the PBX restart or switch–off.
2. Information is sent on all L1 Active ports, i.e. Cornet, BRI and PRI.
3. Information is sent on all CO and ASL ports.
4. Information is sent on all successfully logged–in GSM ports with SIM cards.
5. Finally, information on error statuses and their elimination is sent: DSS1 deactivation or higher BER or SLIP (according to setting), Cornet deactivation (according to setting), ASL – error, CO – error, GSM – logout or bad signal (according to setting).

Answer

In this tag specify the ports and the client from which the PBX is able to receive requests.

- **Peer port** – here the PBX expects the SNMP requests and acknowledgements. The default SNMP port is 161.
- **From this client address only** – use this option to lock receiving requests from a selected IP address or domain name.

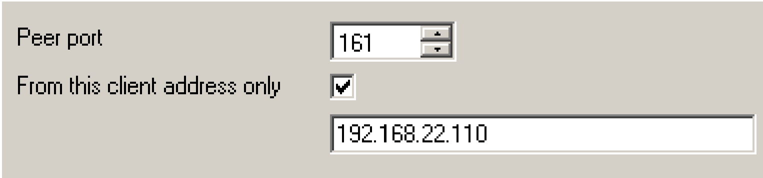


Figure: View of Listening Port Setting Section

Notification

- **Client address** – here define the client's IP address or domain name to which notifications are filtered as mentioned below are sent.
- **Client port** – here define the client port to which notifications are sent.
- **Used local port** – here specify the PBX port to be used for sending notifications if necessary. And for receiving info request confirmations. If this option is disabled, the port is selected randomly.
- **Notification type** – here select the type of notification to be used. For SNMP v1, Traps may be selected only, for higher versions info requests are also available.
    - **Trap** – is an SNMP message sent to the client about an event that should be notified. The message does not require acknowledgement.
    - **Inform request** – is an SNMP message sent to the client about an event that should be notified. Unlike traps, inform requests can be resent if undelivered within the acknowledgement timeout.
        - **Repeat** – defines the count of notification sending attempts.
        - **In interval** – defines the time interval during which confirmation is awaited from the client.
- **Version**– here specify the notification coding type according to the SNMP version used.
    - **SNMP v1**
    - **SNMP v2c**
    - **SNMP v3**

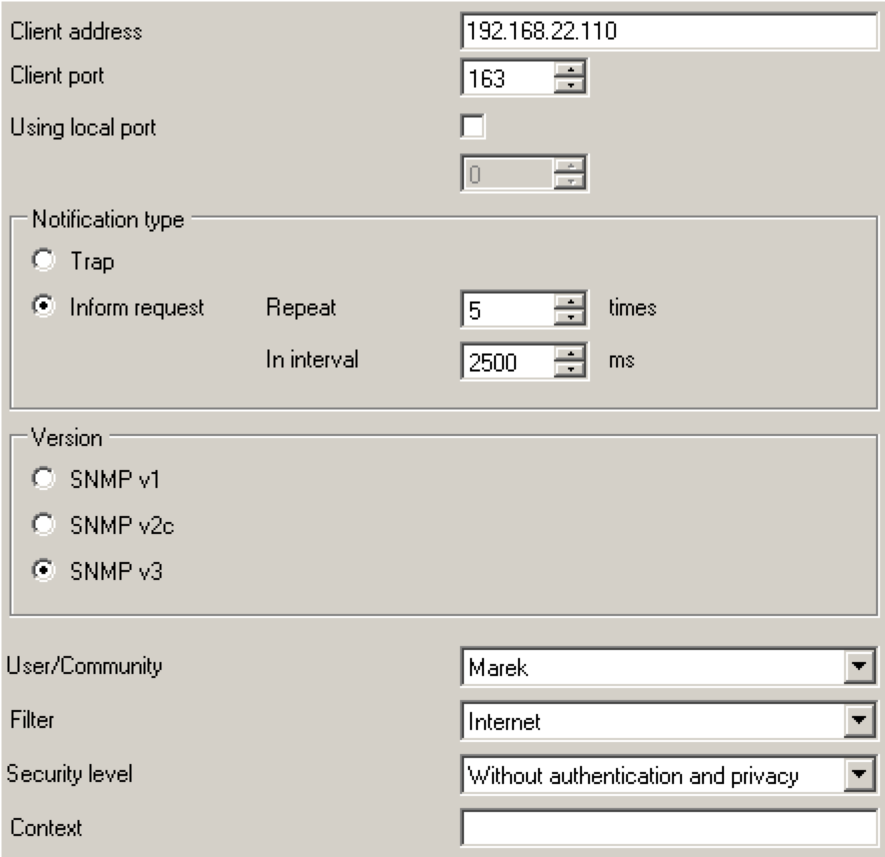


Figure: View of Notification Configuration Menu

- **User/Community** – here define the SNMP user that corresponds to the **USM** for SNMP v3 and **Community** for the other versions.
- **Filter** – here define the Notify filter. The longer the root and subtree OID, the stricter the filter.
- **Security level** – this parameter can be used for SNMP v3 only and defines the notification security level. Choose one of the following options:
  - **Authentication and privacy**
  - **Without authentication and privacy**
  - **Authentication only**
- **Context** – use a text string to identify a SNMP module within the client address. This parameter need not be filled in.

Event Reporter

Find the Event reporter in the **Network – Supervision Services – Event Reporter** menu. Here you can set the basic rules for sending info SMS on system parts. **This function is subject to licence!**

- **Event**– use this parameter to set the event type to be SMS–reported. Choose one of the following options:
  - **PBX restart** – PBX restart notification.
  - **PBX keepalive** – sending of keepalive messages for PBX checking purposes. Set the keepalive sending period in the Planned events in the **Global data** menu.
  - **Port ready** – notification of a virtual port function reactivation.
  - **Port busy** – notification of a virtual port occupation. The BRI and PRI ports are considered busy whenever all channels have been occupied. This function cannot be used for the SIP or SMTP port.
  - **Port error** – virtual port error notification.
  - **Storage full** – notification of a functionless logical storage due to overfilling.
  - **No call port credit** – notification of low credit for a defined virtual port.
  - **No call SIM credit** – notification of low credit for a defined SIM card.
  - **No call terminal credit – not implemented yet.**
  - **No call user credit** – notification of low credit for a defined user.
  - **No SMS port credit** – notification of SMS limit exhaustion for a defined port.
  - **No SMS SIM credit** – notification of SMS limit exhaustion for a defined SIM card.
  - **No SMS terminal credit – not implemented yet.**
  - **No SMS user credit** – notification of low credit for a defined user.
  - **State of status control object Error** – notification of passing of the selected status control object into the Error state.
  - **State of status control object OK** – notification of passing of the selected status control object into the OK state.

Do not report more often than [s] – here set the report sending hysteresis to avoid excessive report sending due to repeated port changes. The states to which a hysteresis timeout applies are reported at once (within one report) after the timeout.

Report on active notification

The block sets how an active event shall be reported.

- **Message** – specification of the text of the SMS to be sent whenever the selected event happens (Port error, Storage full, etc.), i.e. becomes active. If the **Message** field is left blank, no SMS will be sent. In addition to standard texts, the following dynamic strings may be entered.
  - **%n** - name - here enter the name of the event reporter that recorded the event.
  - **%d** - date - here enter the PBX date and time valid at the instant of event recording.
  - **%k** - key - here enter the name of the port to which the event relates.
  - **%v** - value - here enter the event value. At present, there is no event to meet this parameter

- **Relay action** – the option is not accessible until the virtual port is selected in the **Used relay** block. Only then you can select the required relay action.
  - **Switch on** – this option helps close the relay of the below–specified port whenever some of the conditions defined in the **Event parameters** block is met.
  - **Switch off** – helps open the relay of the below–specified port whenever some of the conditions defined in the **Event parameters** block is met.
  - **Positive pulse** – helps activate the relay of the below specified virtual port for a period defined in the **Relay pulse width** parameter after one of the conditions specified in the **Event parameters** is met.
  - **Negative pulse** – helps deactivate the relay of the below specified virtual port for a period defined in the **Relay pulse width** parameter after one of the conditions specified in the **Event parameters** is met.
- **Relay pulse width** – if **Positive pulse** or **Negative pulse** is set above, define the pulse width in milliseconds.

Report on inactive notification

The block sets how an inactive event shall be reported.

- **Message** – specification of the text of the SMS to be sent whenever the selected event ceases to exist, becomes inactive. If the **Message** field is left blank, no SMS will be sent. In addition to standard texts, selected dynamic strings may be entered (see above).
- **Relay action** – the option is not accessible until the virtual port is selected in the **Used relay** block. Only then you can select the required relay action.
- **Relay pulse width** – if **Positive pulse** or **Negative pulse** is set above, define the pulse width in milliseconds.

Parameter evaluation

- **Active notify evaluation** – specify the SMS sending conditions. The following options are available:
  - **Independent by parametr** – reports on active/inactive events are sent for each parameter separately, independently of the states of the other parameters.
  - **At least one parameter active** – reports are sent for each parameter with an active event separately, independently of the states of the other parameters. If the **%k** string is used in the SMS text, the SMS always includes a list of parameters with an active event. The inactive event report is not sent until the event ceases to exist for all the parameters.
  - **All parameters active** – this selection represents a logical AND of all selected parameters. The report is sent only if the event condition is met for all the selected parameters. The inactive event reports are sent for each parameter separately, independently of the states of the other parameters. If the **%k** string is used in the SMS text, the SMS always includes a list of parameters with an inactive event.
- **Deglitch active change shorter than [s]** – this parameter limits evaluation of active events in the case of rapid changes. If a selected event takes a shorter time than as defined, it will not be evaluated as active.
- **Deglitch inactive change shorter than [s]** – this parameter limits evaluation of inactive events in the case of rapid changes. If a selected event ceases to exist in a shorter time than as defined, it will not be evaluated as inactive.

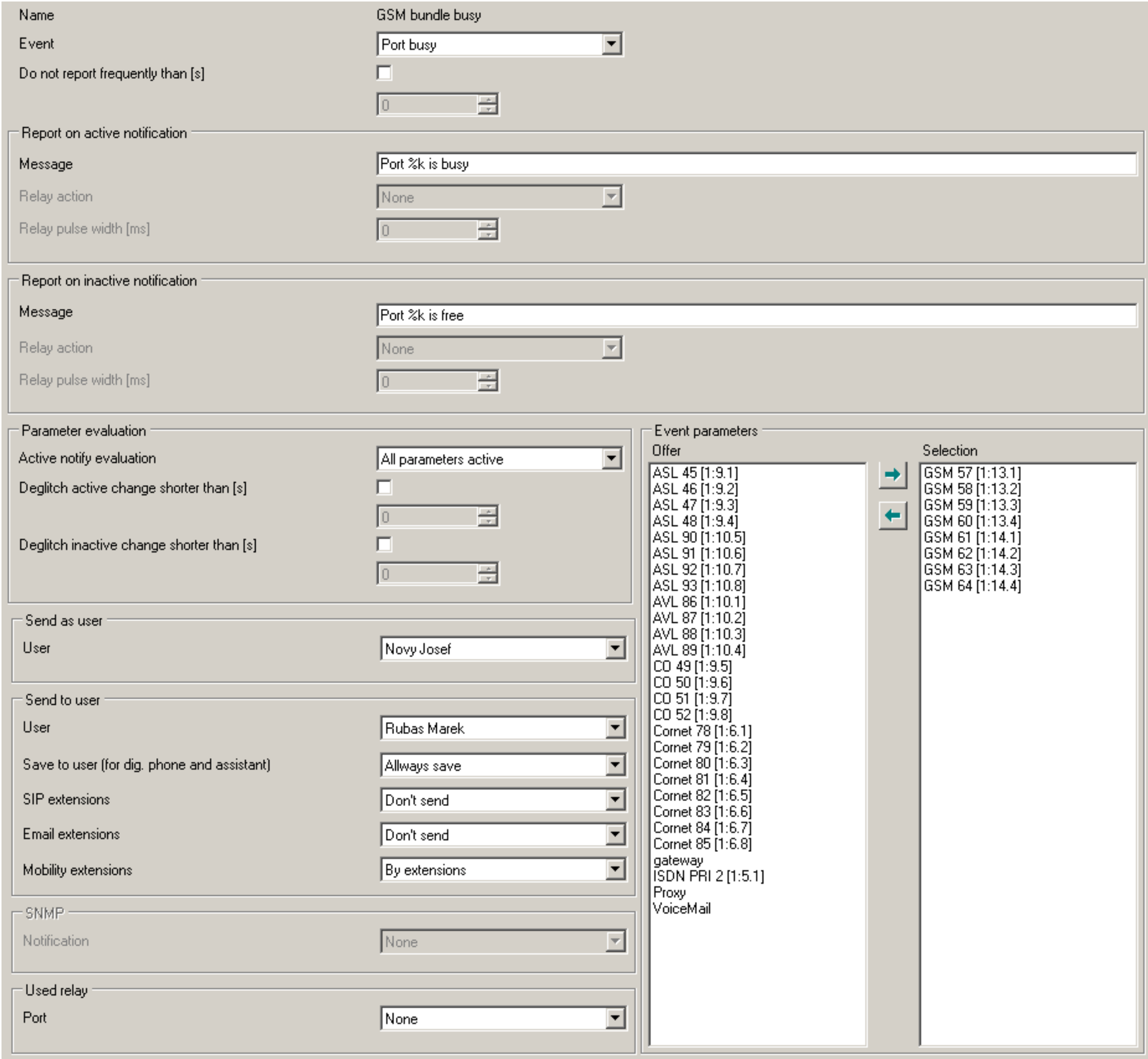


Figure: Pohled na nastavení Reportéra událostí

Send as User

- **User** – here define the user to be presented as the message author.

Send to User

- **User** – use this parameter to define the user to which the message shall be sent.
- **Save to User** – use this parameter to enable/disable saving messages at the user regardless of the user settings, or respecting the user settings.
- **SIP extensions** – use this parameter to enable/disable resending messages to user SIP stations regardless of the user settings, or respecting the user settings (According to stations).
- **Email extensions** – use this parameter to enable/disable resending messages to user email stations regardless of the user settings, or respecting the user settings (According to stations).
- **Mobility Extensions** – use this parameter to enable/disable resending messages to user external stations regardless of the user settings, or respecting the user settings (According to stations).

SNMP

- **Notification** – here specify the SNMP user for notifications. The SNMP block is not available yet.

Used relay

- **Port** – here specify the port whose relay is to be closed whenever some of the conditions defined in the **Event parameters** block is met.

Event parameters

The block is accessible if any of the above mentioned options (PBX restart, Port ready, Port error, etc.) is selected in the **Event parameters**. A survey of available objects related to the event is on the left and a list of objects currently monitored by the given Event reporter is on the right. Use the arrows to move the objects from one side to the other.

Example

Let us observe a bundle of GSM virtual ports (see Fig. 1). If all of the GSM virtual ports are occupied, the "All ports are busy" SMS is sent. When the GSM virtual ports get released gradually (the observed event is inactive), the "Port %k is free" SMS is sent, where %k means the list of free virtual ports.

5.4 DB connectors

The **Network – DB connectors** menu is used for setting communication with the External Routing Machine (ERM server). The ERM server partially replaces or complements the 2N® NetStar internal routing mechanisms. Having received a call/SMS routing request, the PBX sends a query to the ERM server. If a matching record is found in the ERM server database table, the ERM server sends back a parameter specifying further call/SMS routing in the PBX.

The DB connectors created are assigned to the external routers in the **Routing – External routers** menu for call/SMS routing as specified in the ERM server answer.

You can add, rename or remove the DB connectors using the context menu to the left. The right-hand part of the menu is divided into two sections – the upper section helps you set the DB connector properties, in the bottom section you can set parameters for DB connector - ERM server communication.

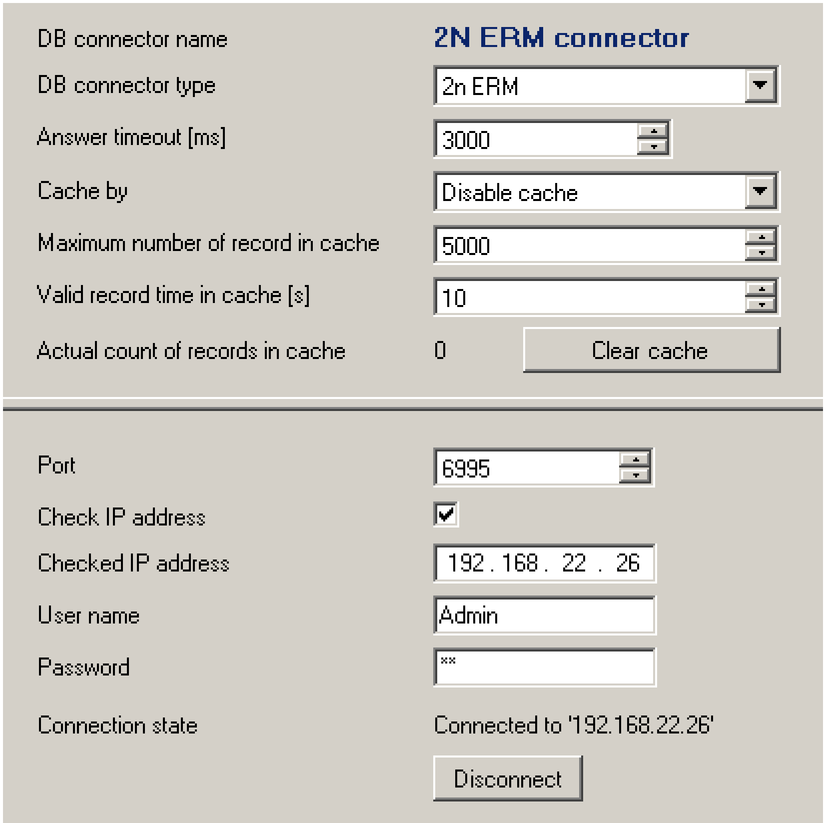


Figure: Pohled na nastavení DB konektoru

- **DB connector type** – type of the DB connector.
- **Answer timeout** – set the timeout for the PBX to wait for the ERM server answer. If no reply comes within the timeout, the call is routed to the default destination in the ERM server.
- **Cache by** – define how to store replies from the ERM server. If a match is sound in the cache, no query is sent to the ERM server. Records can be stored according to the calling or called subscriber. Record storing is disabled by default.
- **Maximum number of record in cache** – set the maximum count of records to

be stored.

- **Valid record time in cache** – set the validity time for a record in the cache.
- **Actual count of records in cache** – this field displays the count of records currently stored in the cache. Click on **Clear cache** to delete all the cache records.
- **Port** – set the port number for PBX - ERM server communication.
- **Check IP address** – having ticked off this option, complete the **Checked IP address** parameter to enable communication with the ERM server with this IP address only.
- **User name** – use this option to verify the user name in ERM server communication.
- **Password** – enter the user password as set in the ERM server.
- **Connection state** – monitor the ERM server connection state. Click on **Disconnect** to disconnect from the ERM server temporarily.

6. Global Data

Here is what you can find in this chapter:

- 6.1 Global Parameters
- 6.2 Emergency Calls
- 6.3 Localisation
- 6.4 Licences
- 6.5 Language Packages
- 6.6 Services
- 6.7 Conference Rooms
- 6.8 Progress Tones
- 6.9 Ring Tones
- 6.10 AutoClip Parameters
- 6.11 Storage Manager
- 6.12 Scheduled tasks
- 6.13 Status Control parameters
- 6.14 DTMF
- 6.15 Causes
- 6.16 Time Parameters
- 6.17 Assistant

6.1 Global Parameters

Disable all new calls

Tick off the parameter to switch the PBX into a mode in which no new calls can be made but active calls are not forcibly terminated. Trying to set up a call, the user fails being played a defined message. This function is useful for servicing purposes.

Switch on regime ME

Use this option to switch your PBX into the Mobility Extension mode, which is specifically used whenever the PBX is connected as a gateway between another PBX and various types of private or public networks. When this mode is active, all the Flash patterns and DTMF characters are sent directly to the opposite port of the PBX, which does not respond to them. **This function has nothing to do with the Mobility Extension used for authorised external extensions!**

Unselected as missed

This option sets the way of displaying missed calls. It refers to cases when an incoming call is routed to a group of extensions or to one user with multiple extensions and is answered by one of these extensions. If this option is not checked, missed calls are not displayed. If it is checked, missed calls are displayed at all extensions except for the one that has answered it.

Generate phone directories from users

Use this option to define the way of automatic generation of phone directories using the list of users or extensions in the **Users – Phone directories – Group generated** menu. If this option is checked off, the phone directory is filled with user names and respective internal numbers. If not, the phone directory is filled with extension names and respective numbers.

Repeat destinations

With firmware v. 2.7.0 and higher, you can repeatedly route a call to one and the same object. This is used, for example, for new routing in a bundle if the **Repeat cyclically** parameter is enabled.

- **Timeout** – define the time interval after which the same call can be routed to the selected object. If routed before the timeout end, the call is ignored by the PBX.
- **Count** – define the count of routing repetitions via one object for the given call. When this count is completed, call routing is terminated in the next routing attempt to this object.

> **An unduly low timeout value may result in a considerable PBX overload due to call deadlock.**

Example:

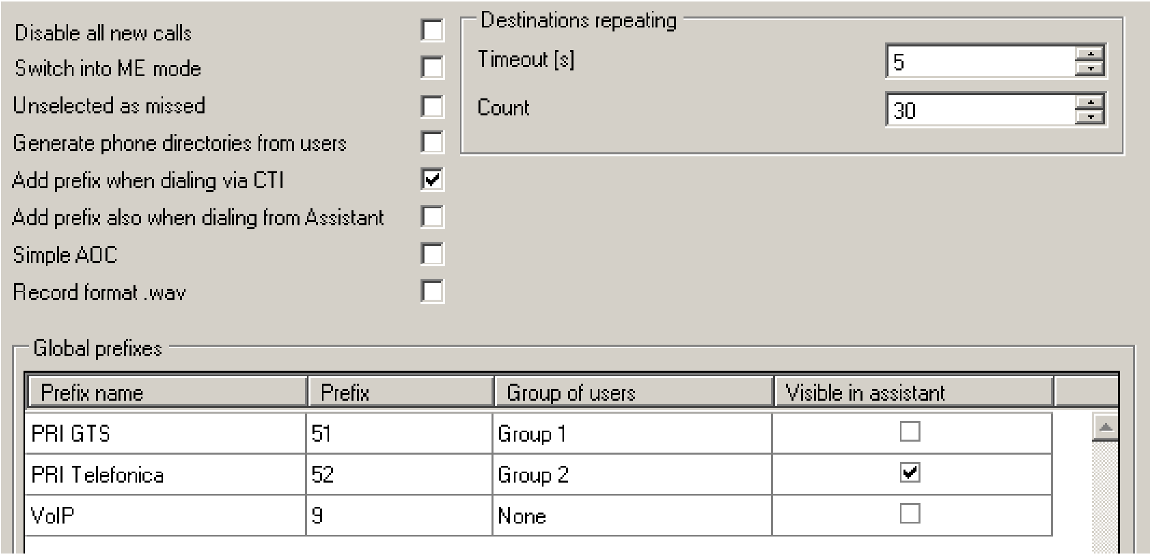


Figure: View of Global Parameter Configuration Menu

Global prefixes

Global prefixes are primarily used for **Analogue** and **VoIP** virtual ports for easier dialling (CallBacks) even to public networks from the list of missed calls. The prefix is not added where the CLI has the **Internal** subtype. Assign the respective prefixes to the virtual ports using the **Added prefix for external CLIP** included in the **Basic** tag. Unlike the frequently used identification table, this option is applied close before departure to the selected port, i.e. after pairing with the telephone directory. It would be necessary to keep different phone directories for digital, analogue and SIP virtual ports if this option were not used.

Add prefix when dialling via CTI – use this option to enable dialling prefixes also for CTI calls, i.e. calls from the Tray icon, Assistant and–or Communicator. In order to set up a call, the PBX checks the length of the called number against the **Numbering plan length** parameter in the Localisation menu. If the called number is longer, the prefix matching the calling user group is dialled.

Add prefix also when dialling from Assistant – use this option to enable dialling prefixes also from the Assistant. The key fact for whether to enable or disable this item is whether or not the prefixes are included in the phone directories. If so, do not check this option off to avoid double prefix dialling. You can select this option only if the preceding one is enabled.

The meanings of the tag columns are as follows:

- **Prefix name** – here define the prefix name to be used in other menus for identification.
- **Prefix** – here define the prefix to precede the calling party identification.
- **Group of users** – here define the group of users to use the given prefix. If no group is selected, the prefix is valid for all groups of the PBX.

Example

Suppose a call is coming from a public network extension with the number 777123456. The call is routed through the PBX to the user Karel Furst, who belongs to user group 'Skupina 1'. His VoIP phone is registered to the SIP proxy, which has been assigned prefix PRI GTS (Figure 1) in the **Added prefix for external CLIP** parameter. If the number 777123456 is found in the phone book, the calling user name is sent to the terminal including the calling user number and the added prefix 51, i.e. 51777123456. To call this number back, the user Karel Furst can dial it directly from the list of missed calls (or received calls). The call is correctly and least–cost routed to the appropriate external port.

Billing

Simple AOC - this function helps you modify the billing rercords of the PBX. It allows you to simply distribute costs of such functions as call forwarding, Mobility Extension (bidirectionally) and CallBack. You are recommended to tick off the parameter before exporting data from the PBX at the latest. This simplification applies to CDRs created by firmware versions 3.0.0 and higher. **The internal subscriber numbers are crucial for billing in these versions.**

Recording

Record format .wav – tick off to save all recorded calls in the .wav format. Calls are recorded in the .alaw format by default.

6.2 Emergency Calls

This menu helps you route emergency calls properly when the PBX is in one of the pre–defined emergency modes. Of course, this setting does not solve the PBX error states. In error states, analog CO lines and an analog telephone connected to the corresponding port of the same card can be used, for example. If the card is not powered, these ports are disconnected and you can make PSTN calls via the card directly.

- **List of emergency numbers** – here specify all necessary emergency numbers. Enter the numbers into one row using the comma separator. The count of numbers is unlimited.
- **Set exceptional situations** – define how the PBX should process calls other than the emergency ones. A call rejection cause and a voice message are defined for each contingency.
  - **Licence expired** – the PBX licence has expired.
  - **Emergency mode** – PBX emergency mode.
  - **Disable calls** – the **Disable new calls** option from the Global parameters menu has been activated. No new calls may be set up but active calls are not terminated (the PBX is waiting for the users to hang up).
- **Destination for emergency calls** – define a virtual port or bundle of ports to be used for emergency call routing.

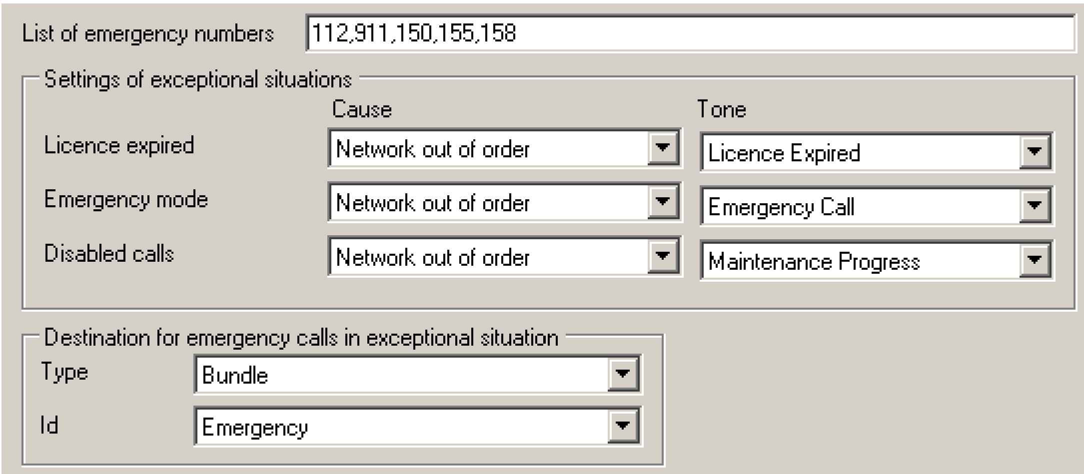


Figure: View of Possible Emergency Call Menu Configuration

6.3 Localisation

Destination selection

In this field enter the numbers and prefixes according to the international numbering plan. This subsequently facilitates normalisation of incoming and outgoing numbers and call routing:

- **Destination** – here choose a **Localisation** (country) from the list and the appropriate country code and access codes will be assigned automatically. The settings can be changed manually if needed.
- **Number** – this number represents the country code within the international numbering plan. For example, the Czech Republic has number 420 and Slovakia 421.
- **Prefixes** – this prefix represents access codes into the international telephone network. By default it is 00 and + for the GSM network.

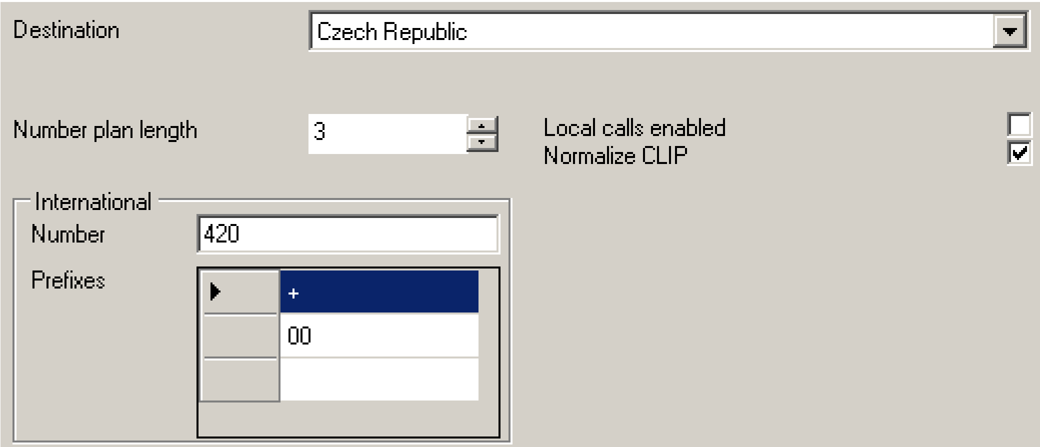


Figure: Basic PBX Localisation Parameter Setting Menu

Local settings

Like the **International** option, the **Local calls possible** tag helps you define the national parameters:

- **Number** – represents the national access code (area code). For example, the town of Bratislava, Slovakia, has the area code 2.
- **Prefixes** – represents the access codes into the national telephone network. The default value is 0.

Normalise CLIP

- **Normalise CLIP** – check this option to cut automatically the Calling Party Number (CPN) to the shortest known format according to the CLIP routing **Localisation** setting. If this option is not checked, you have to route incoming calls to the requested destinations via the CPN routers. As a matter of fact, this setting means that numbers +421XXX, 00421XXX, 0XXX and XXX are identical in terms of routing.
- **Number plan length** – use this parameter to define the PBX numbering plan length. The setting affects number normalising.

The settings included in this chapter are particularly utilised by the Initial Wizard for the first system login in the on-line mode with a new database.

6.4 Licences

Licence files

This section provides a list of installed licence files including basic descriptions. Here you can install, uninstall or download the licence files to your computer. The field consists of several columns with the following meanings:

- **File** – shows the absolute path to a licence file within the system data space.
- **State** – shows the current state of a licence file within the system (e.g. **Loaded**, **Not loaded**, **Bad CPU**, etc.).
- **E1 ports** – shows the count of licensed ports for ISDN PRI.
- **E1 channels** – shows the count of licensed channels for ISDN PRI.
- **SIP terminals** – shows the count of licensed terminals for VoIP telephones that are necessary for logging your VoIP extension to the SIP proxy.
- **ME** – shows the count of licensed external extensions (ME – Mobility Extensions). To enable the **Transfer** parameter in the **Properties – ME** tag on any of the hierarchical levels, you need more licences. Setting this parameter to **YES** on the user level needs as many licences as many extensions the user has. Setting this parameter to **YES** on the carrier level needs as many licences are there are extensions logged to the carriers of the selected type.

Licence files

File	ID	Status
/data/netstar/licence/0.key	NS2LIC-G8cc5b35d9663f34298	OK
/data/netstar/licence/1.key	NSLIC-TEST-11	OK
/data/netstar/licence/2.key	NS2LIC-Ge8d6e86bc05dd6c0cf	OK

Figure: Example of Three–Licence NetStar

Licences

This part displays a well–arranged table showing details on a selected licence file. The field consists of several columns with the following meanings:

- **Feature** – shows the type of a licensed service, interface or object within the system.
- **Type** – defines a licence within its type.
- **Licensed** – shows the count of licensed channels, terminals or service accesses.
- **Requested** – the currently requested count of channels, terminals or service accesses. The red-marked rows indicate lack of licences.

If no licence is available, the PBX works in the trial mode. After the trial licence expiry (800 hours), the system is blocked and will not work until the relevant licence is installed.

Licence features

Feature	Type	Licensed	Requested
Conference subscriber		5	0
Mobility Extension user		Unlimited	1
CallBack user		0	2
VoIP channel	VoIP	0	0
Virtual port	Common BRI	Unlimited	0
Virtual port	S0	Unlimited	8
Virtual port	UPN	Unlimited	8
Virtual port	E1	13	5
Virtual port	ASL	Unlimited	8
Virtual port	CO	Unlimited	8
Virtual port	GSM	Unlimited	14
Virtual port	AUX	Unlimited	0
Virtual port	Binary I/O	Unlimited	0

Figure: View of Licence Features Table

The most important licences

The survey below includes the most important licences including their function descriptions.

- **SIP terminal** – shows the count of licensed terminals for VoIP phones. You cannot log in a VoIP extension to the SIP proxy without a terminal.
- **Mobility Extension user** – shows the count of Mobility Extension licences (external extensions). They are necessary for enabling the **Transfer** parameter on the **Properties – ME** tag on some of the hierarchical levels. Setting this parameter to **Yes** on the user level needs as many licences as there are user extensions (excluding external extensions). Setting this parameter to **Yes** on the carrier type level needs ME licences for all extensions logged in to the carriers of this type.
- **CallBack user** – the CallBack licence shows the count of extensions that are allowed to use and that are currently using the function.
- **Conference subscriber** – the licence shows the highest count of conference participants during the PBX operation instead of the current count of participants for the Requested column. The function helps identify the need for licence expansion.
- **Conference rooms** – shows the count of licensed and currently existing conference rooms. The licence is also allocated to the rooms that are currently inactive due to a time condition.
- **VoiceMail user** – gives the count of users to whom the VoiceMail function may be assigned and the count of users who are currently using the VoiceMail function.
- **Modem** – the modem licence shows no count. The function is either licensed or unlicensed.
- **Event reporter** – the Event reporter licence shows no count. The function is either licensed or unlicensed.

- **Call recording** – the count of recording users or channels is licensed. One licence is allocated to one virtual port channel or one station of an authorised user. This means that 30 licences are needed to enable recording over the whole ISDN PRI port. If, for example, your licence is limited to 10, calls via 10 channels of this port will only be recorded.

6.5 Language Packages

The list of available language packages finds itself in the **Global data – Language packages** menu. In addition to default packages, new languages packages can be installed here. A language package consists of progress tones, service messages and StarPoint key telephone menus.

You can create a language package easily using any of the existing packages. Open the **Language.ini** file (a common text file). Change the **Language ID** into the number corresponding to the required localisation. Now translate the file rows into the selected language. To add a message, record one and name it exactly as specified in the English/Czech or any other package. Finally, zip all the files into a folder (hungary.zip, e.g.).

To install a new language package, click on the right-hand mouse button and select the **Install language package** option in the context menu. Then choose the path to the packed file. To uninstall a language package, select the package and use the **Uninstall language package** option with the right-hand mouse button. The language package table consists of four columns.

It is unnecessary to define a storage for the language package in this menu as the storage set in the Global data – Storage Manager menu is used automatically.

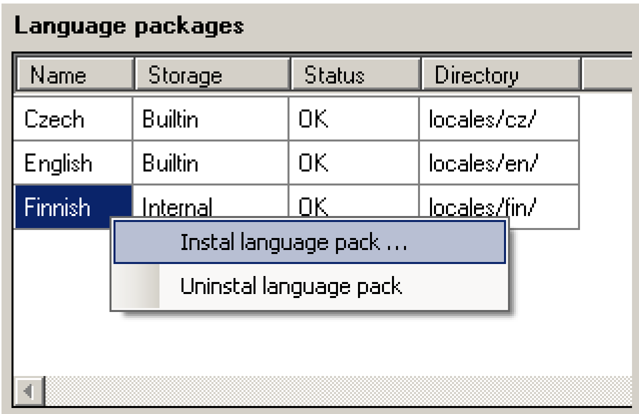


Figure: View of Language Package Adding Menu

Column meanings:

- **Name** – shows the name of the language package. The default packages are named after their respective languages.
- **Storage** – defines the path to the package storage within the system data space. **Built-in** means the **/opt/netstar** directory and **Internal** means the **/data/netstar** directory. Together with the **Directory** column, this parameter gives the absolute path to the storage.
- **Status** – shows the package installation status.
- **Directory** – shows the path to the package storage within the system data space. Together with the **Storage** column, it gives the absolute path to the storage.

6.6 Services

Service Division

The services supported by the NetStar PBX can be divided into three groups – **User**, **Extension** and **Others**.

- **User** – this group consists of user call forwarding settings, including forwarding to VoiceMail, VoiceMail progress tone/message recording, playing and deleting, PIN changing, user bundle login and so on.
- **Extension** – this group consists of extension forwarding settings, extension ringing settings for user calls, extension bundle and carrier logins, private calls, CallBack to a extension, call takeover from a extension and so on.
- **Others** – this group consists of all the remaining services – progress tones recording, PBX date and time setting, CallBack to a number, global setup, call assumption from a group, conference calls, connection to calls, profile activation and so on.

Service Setting

The **Global data – Services** menu displays a list of available services on the let. To create the default list use the **Default** option in the context menu. Set the selected service on the right–hand side of the menu. Programmable service attributes are added to each service such as progress tones, setting status messages, PIN activation, alert time and default routing destination. If a service requires the PIN, assign the PIN to the calling user to avoid service unavailability!

To activate a service, dial the service prefix into the routing table according to the called number. For the default service router refer to the **Services** in the **Routing – Routers** menu as shown below:

Name SERVICES

Type Called number

Prefix	Digit...	Remo...	Add to ...	Remove fro...	Add to ...	Scheme	Type	Destinatio...	Destination	Tone
*21	0	3		0		Preserve	Preserve	Service	Forward user-Uncond.	None
*22	0	3		0		Preserve	Preserve	Service	Forward user-Busy	None
*23	0	3		0		Preserve	Preserve	Service	Forward user-No answer	None
#20	0	3		0		Preserve	Preserve	Service	Cancel all user fwdg.	None
#21	0	3		0		Preserve	Preserve	Service	Cancel user fwdg. -Uncond.	None
#22	0	3		0		Preserve	Preserve	Service	Cancel user fwdg. -Busy	None
#23	0	3		0		Preserve	Preserve	Service	Cancel user fwdg. -No answer	None
**	0	2		0		Preserve	Preserve	Service	Take over from extension	None
##	0	2		0		Preserve	Preserve	Service	Take over from user	None
#*	0	2		0		Preserve	Preserve	Service	Take over from group	None
*#	0	2		0		Preserve	Preserve	Service	Take over from my group	None
*50	0	3		0		Preserve	Preserve	Service	Alarm	None
#50	0	3		0		Preserve	Preserve	Service	Cancel alarm	None

Figure: View of Service Call Routing Menu

Or, activate a service by sending an SMS. This method can only be used for groups of services that need not be active during extension setting (e.g. call forwarding, PIN/user changing, profile activation, etc.).

Description of Selected Services

For some services, more parameters should be defined in addition to progress tones or messages. Below are some of them:

Private call

Here set the destination type in the **Destination** field to **Nothing** (the calling user settings are used) or **Router** (select a router).

Call parking

Here define the **Maximum parking time**. The default value is set to 180s. The parked user hears the Music on Hold. After the time limit, the parking place is cleared and the call returns to the extension that parked it before. The parked user hears the alert tone.

Set presence

You can use an SMS message (not a call) to set the presence text. The SMS is routed to the text router where the text section is removed that is used for SMS routing to the **Set presence** service. The default code is ***61**. The rest of the text is used as a new presence text of the SMS sending user. If the user has an active profile at the time of sending, the presence text is assigned to the profile.

Add to conference

Use this service to add held subscribers to a conference. Two extensions at least have to be on hold for a correct function (speech slots are used). Now use the **Add to conference** service. The default code is**0#*. The two held extensions and the caller now join the conference.

Call to conference room

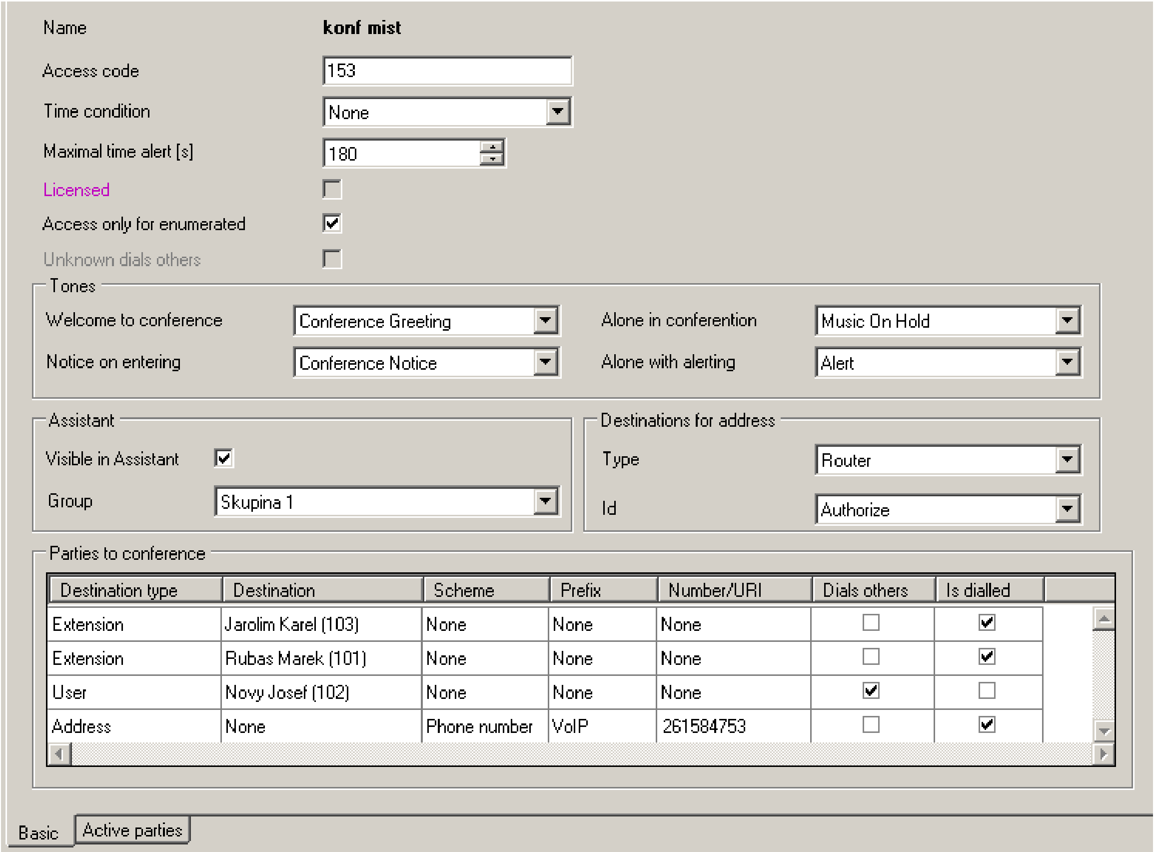
Use this service to call a conference for a defined conference room. Having dialled the service code (***2#** by default), you are invited to enter the conference room access code. Enter the code and then a 'hash'. If the calling user has the right to call together the conference room subscribers, the other users are called subsequently. For more information refer to the Conference Rooms section.

6.7 Conference Rooms

For conference room settings refer to the **Global data – Conference rooms** menu. Use this menu to configure the conference rooms and define the authorised users. This function is subject to licence, so make sure that you have the required count of licences for operating all of your conference rooms.

Basic

- **Access code** – is used for distinguishing your conference rooms within the service. Therefore, assign a unique access code to each conference room.
- **Time condition** – define a time interval to limit the conference room use. If no time condition is defined (– – –), the conference room is accessible continuously.
- **Maximal time alert [s]** – use this parameter to define the maximum ringing time for each of the extensions called together within a conference room. After this time interval, ringing to unanswered extensions is terminated. The default value is 180s.
- **Licensed** – if this checkbox is ticked off, the conference room is licensed and may be used. If not, check the count of licences in the Global data – Licences menu and purchase new licences or delete unused conference rooms as necessary.
- **Access only for enumerated** – use this parameter to lock the conference room and give access to selected users only. Unauthorised users are denied access to the conference room.
- **Unknown dials others** – use this parameter to assign the conference–calling right to a user that is not included in the conference room.



Tones

- **Welcome to conference** – this tone is played to the user after the user's joining the conference called by the selected conference room (handset pick–up).
- **Notice on entering** – this tone is played to the conference participants after joining of the user that was not dialled during the conference calling or got out of the conference and is now trying to rejoin the conference room.
- **Alone in conference** – this tone is played to the user that remains alone in the conference room (no other extension is even ringing).
- **Alone with alerted** – this tone is played to the user that is the first or only to answer during ringing to the conference room users. As soon as another user answers the phone, the tone is disconnected.

Assistant

- **Visible in Assistant** – here enable displaying of the particular conference room in the Assistant.
- **Group** – with the Assistant display enabled, specify here the user group for which the conference room should be visible. To make it visible for all user groups, select (– – –).

Destination for addresses

Use this parameter to define the routing destination if the address specified in the Conference subscribers block is used for conference call set–up. If the Default destination type is selected and the calling party is an address, you cannot call the specified addresses but can call the extensions and users that are dialled directly. If the calling party is a extension or user, you can call the addresses too (routing From port of the calling party is used).

- **Type** – define the destination type for the address.
- **Id** – define the specific destination of the selected type.

Parties to conference

In this configuration section, you can specify users, extensions or just telephone numbers including properties for a conference room. For this purpose, seven columns with the following meanings are available:

- **Destination type** – here select a user, extension or address. The following columns are available or not depending on your selection.
- **Destination** – here define a extension or user.
- **Schema** – here enter the Number or URI scheme for the address.
- **Prefix** – select the required prefix from the list of prefixes defined in the Global data – Global parameters menu for the address. Use this prefix for dialling the given conference subscriber.
- **Number/URI** – enter a specific Number of URI for the address.
- **Dials others** – here set whether the selected user has the right to call a conference for the selected conference room.
- **Is dialled** – here set whether the selected user shall be dialled or not during conference set–up.

Active users

This tag includes an on–line list of all active conference participants. The extension name (if available in the telephone directory), scheme (Number/URI), number type (national, internal, ...) and the number are defined here for each of them.

6.8 Progress Tones

Introduction

The "Progress" is a general name for all tones and announcements injected into the speech channel by the PBX. When a new database has been created, the PBX provides a set of default progress tones depending on the language packages installed. The basic set can be extended to include own (user recorded) files and tones, or external audio inputs (e.g. mp3 player) can be connected. The menu is logically divided into tags.

Figure: View of Progress Tones Menu Tags

Progress list

The progress tones represent the highest level of tone and message processing. They are ready–made tones and announcements that are played back to the user by the PBX. Each progress tone has to contain one source at least.

Progress list

The Progress list tag displays all progress tones available in your PBX including those created by the user. The following functions can be used through the context menu:

- **Add** – use this option to add a new progress tone.
- **Rename** – use this option to rename a selected progress.
- **Delete** – use this option to delete a selected progress.
- **Delete all** – use this option to delete all progresses.
- **Add default progresses** – use this option to update the default progress set preserving the changes made in other default and user progress tones.
- **Restore default progresses** – use this option to restore the default set of progress tones without changing or deleting the user created progress tones.

Information about progress

- **Name** – the parameter shows the name of a selected progress tone and cannot be configured in this section.
- **Number** – is the progress tone number to be used for user progress tone recording, playing and deleting services.
- **Allow progress sharing** – use this parameter to enable sharing of a selected progress tone by multiple users with the aim to save the PBX internal sources (players) during increased traffic. Since the progress tone is played back from the current position in this mode, this function is mostly used for the progress tones that need not necessarily be played back from the beginning (e.g. Music on Hold).
- **Language** – here select the progress tone language version.
- **Play** – use this option to play back a selected progress tone.
- **Stop** – use this option to stop playing a selected progress tone.

Progress configuration

- **Action**– here choose one of the listed commands to define the meaning of the row.
    - **Repeat** – use this command to set the count of repetitions from the last Repeat command, or from the beginning of the progress tone till this moment. Set the count of repetitions in the Repetitions column. If the parameter is set to 2, the sequence is played once and then repeated twice.
    - **Repeat from beginning** – use this command to set the count of repetitions from the beginning of the progress tone till this moment. Set the count of repetitions in the Repetitions column. If the parameter is set to 2, the sequence is played once and then repeated twice.
    - **Pause** – use this command to create delays between the progress tones. Do not define the delay in the **Duration [ms]** column until you have set the **Play** or **Play progress** row.
    - **Play sound** – use this command to play a selected element for the period of time as set in the **Duration [ms]** column. If you set 0, the element will be played till the end.
    - **Play sequence** – use this command to play other progress tones. For the progress to be played refer to the **Progress** columns. The other columns are unused.
    - **Off** – use this option for the progress tones played within a notification of a new incoming call. Fill in the **Duration** column to set the tone off–time. If repetition is enabled, then the selected tone is replayed after re–connection. The **Queue Alert** is an example of this type of tone. To add this tone to your list, click on **Add default progresses**.
- **Priority** – here define the priority of sources for the row set by the **Play** command. If a source is unavailable at the moment, another source with a lower priority is used.
- **Progress** – here define the progress tone to be played within the **Play progress** command.
- **Own file** – here define the source file as listed in the **Own files** tag to be played within the **Play** command.
- **Tone** – here define the source tone as listed in the **Tones** tag to be played within the **Play** command.
- **Input (AUX in)** – here define the source audio input as listed in the **Audio inputs** tag to be played within the **Play** command.
- **Repetitions** – here define the count of repetitions for the **Repeat** command. If you select 0, the progress tone is played all around. If you select 1, the sequence is played once.
- **Duration** – here define the duration for the **Play** or **Pause** commands.

To change the order of the progress rows use the arrows in the right–hand part of the menu. To add new rows to a certain position use the **Insert ahead selected** and **Insert behind selected** options. The **Add** selection adds a record after the last one.

Language pack files

Language pack files list

This section shows all available language package files that can be used as sources for progress tones. Save these files to your local PC disk if necessary using the context menu.

Related progress list

This section gives a list of all the progress tones that use the above selected file. You can use all the context menu functions as available in the **Progress list** tag.

Other sections

The **Information about progress** and **Progress configuration** sections are common for all tags and their parameters. For the configuration options refer to the Progress list section.

Own files

Own files

This section shows all the files uploaded by the user to the PBX that can be used as progress tone sources. **The voice message must have the following format: wav 8kHz, 8bit, mono, aLaw**. You can also use the service as described in the User Manual for recording. The context menu of this section provides the following functions:

- **Add** – use this function to add a record. The record is then used as a progress tone source. It has no file after creation. It has to be uploaded via the **Own files source** section.
- **Rename** – use this function to rename a selected record.
- **Delete** – use this function to delete a selected record.
- **Delete record, keep file** – use this function to delete a selected record while keeping its uploaded file in the PBX.
- **Backup file to local disk** – use this function to download a file of a selected record to your local disk. First select the file to be saved in the NetStar data space and then enter the name and storage on your local disk.

Own files sources

Within this section you can upload a file of your own assigning it to the created record in the **Own files list** section. The context menu of this section provides the following functions:

- **Add** – use this option to upload a file with announcement for a selected record.
- **Add record for existing file** – use this option to create a record for an existing file.
- **Delete** – use this option to delete a file of a selected record.
- **Delete record, keep file** – use this function to delete a selected record while

keeping its uploaded file in the PBX.

Related progress list

This section displays a list of all the progress tones that use the above selected own file. You can use all the context menu functions as available in the **Progress list** tag.

Other sections

The **Information about progress** and **Progress configuration** sections are common for all tags and their parameters. For the configuration options refer to the Progress list section.

Tones

Tones

This section displays all the tones of the PBX that can be used as progress tone sources. The context menu of this section provides the following functions:

- **Add** – use this function to add a new tone.
- **Rename** – use this function to rename a selected tone.
- **Delete** – use this function to delete a selected tone.
- **Delete all** – use this function to delete all tones.
- **Derive** – use this option to create a copy of a selected tone.
- **Add default tones** – use this option to complete the list of default tones while preserving any changes in the existing tones.
- **Restore default tones** – use this option to complete the list of default tones restoring the default values of all the changed tones.

Tone configuration

In this section you can configure a tone using a three–column table as follows:

- **Language** – this column defines the language version for each tone row. Thus, you can create different forms of a tone for different languages.
- **Action** – this column helps you set one of the following actions for each row:
  - **425Hz** – plays a tone with the frequency of 425Hz. Set the tone duration in the **Duration [ms]/Repetitions** column.
  - **Repeat** – use this option to repeat rows from the beginning to this row. Set the count of repetitions in the **Duration [ms]/Repetitions** column. If you select 0, the section is played back all around. If you select 1, the sequence is played just once.
  - **Silent** – use this option to define the delays between the **425Hz** function rows. When used on the first row, this function has no meaning.
  - **Duration [ms]/Repeatitions** – use this column to enter the duration for the **425Hz**, **Silent** and **Through** functions or the count of repetitions for the **Repeat** function.
- **Through** – has the same function as **Silent**, yet you can hear the line sounds if any.

Related progress list

This section provides a list of all the progress tones that use the above selected tone. You can use all the context menu functions as available in the **Progress list** tag.

Other sections

The **Information about progress** and **Progress configuration** sections are common for all tags and their parameters. For the configuration options refer to the Progress list section.

Audio inputs

Audio inputs

This section displays all the audio inputs of the PBX that can be used as progress tone sources. The context menu of this section provides the following functions:

- **Add** – use this function to add a new audio input.
- **Rename** – use this function to rename a selected audio input.
- **Delete** – use this function to delete a selected audio input.

Audio input sources

In this section you can assign a virtual port of the Audio/IO/Relay board to a selected Audio input. For each port define the language to be used for the input.

Related progress list

This section provides a list of all the progress tones that use the above selected audio input. You can use all the context menu functions as available in the **Progress list** tag.

Other sections

The **Information about progress** and **Progress configuration** sections are common for all tags and their parameters. For the configuration options refer to the Progress list section.

6.9 Ring Tones

To set the ringing tones use the **Global Data – Ring Patterns** menu. Each ringing tone consists of a ring pattern and Cornet tune. Some terminals are unable to change the ring tune and use the ring pattern only. See a list of available ring patterns on the left. You can add, remove or rename the ring patterns using the context menu. During database creation, default progress tone patterns are created that can be edited or removed as necessary. To restore the default settings without removing the tones created by you, use the **Update** option. To restore the default settings and remove all the ring tones, use the **Default** option in the context menu. To create the ring tones, use the following parameters:

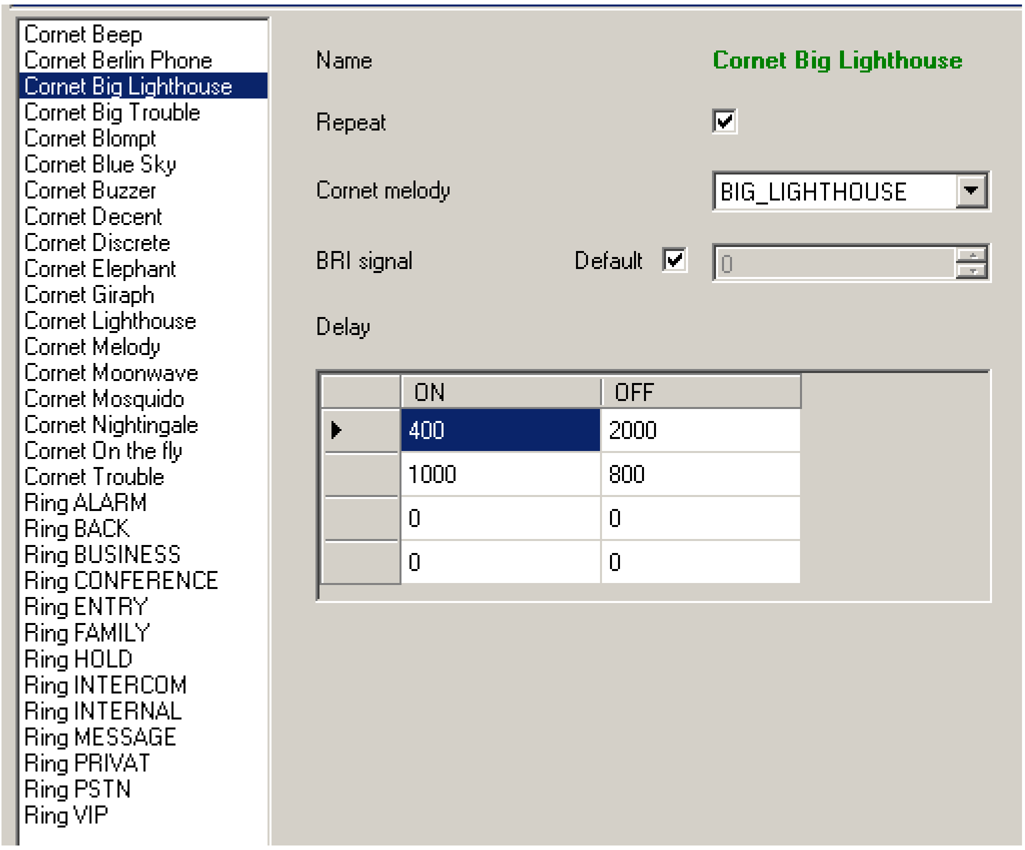


Figure: View of Ring Tone Configuration Menu

- **Repeat** – check this option to make a tone pattern being played repeatedly. If this option is not checked, the tone pattern is used only once.
- **Cornet tune** – use this parameter to assign a ring tone tune to a selected ring pattern in the StarPoint key telephones.
- **BRI signal** – use this parameter to set the ring tone signalling for the ISDN terminals that support multi–tune ringing.
- **Delay** – here set the ring pattern using two columns and four rows. Set the **ON** and **OFF** columns for each row to be used. **ON** represents the ring current time and **OFF** represents the row resumption delay. Both the parameters are set in milliseconds.

6.10 AutoClip Parameters

AutoClip Routing

AutoClip routing is used for routing of incoming calls and SMS messages in NetStar mainly through the carriers that do not transfer the PBX CLI. For example, an outgoing call via the GSM carrier identifies itself as a SIM card number assigned to a port, not as a calling user. For these cases, the information on outgoing calls and messages is saved into AutoClip routing tables, which help find the originally calling user and route the incoming call or SMS to this user. For more details on AutoClip routing refer to Chapter 6.10 AutoClip Parameters.

AutoClip Parameters

You can save records on outgoing calls/messages including user defined parameters into the AutoClip table with the aid of AutoClip parameters. To define the AutoClip parameters, use the **Global Data – AutoClip Parameters** menu. The menu is divided into two parts. A list of available AutoClip parameter sets is on the left. Here add, remove or rename the sets using the context menu. On the right, you can configure the parameters of a selected AutoClip parameter set.

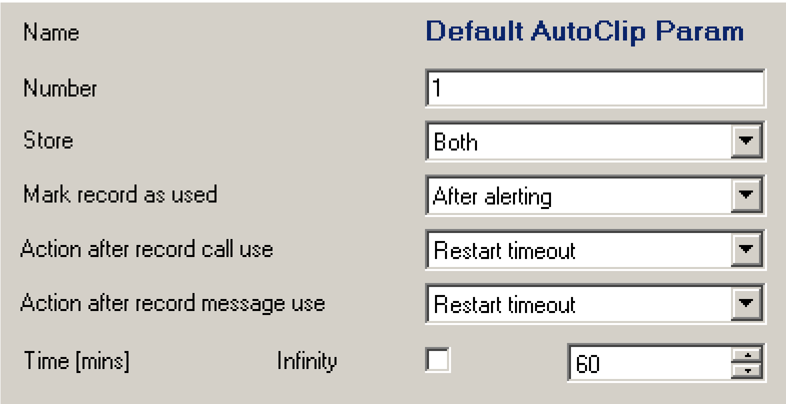


Figure: View of AutoClip Parameter Setting Menu Used for Saving

- **Name** – is the name of the AutoClip parameters set.
- **Number** – is the number of the AutoClip parameters set. It has no function in the current firmware, but is ready for later use.
- **Store:**
  - **Missed** – use this option to store only records on unanswered calls (including rejected) in the AutoClip router.
  - **Answered** – use this option to store only records on answered calls (signalling connections) in the AutoClip router.
  - **Both** – use this option to store records on all calls made (answered, missed, rejected) in the AutoClip router.
- **Mark record as used:**
  - **After alerting** – a record is marked as used after the originally calling user extension starts ringing (Alerting message).
  - **After active** – a record is marked as used after the originally calling user

receives an incoming call (Active message).

- **Action after record call/message use:**
  - **None** – no action is done after use and the record may be reused for the next matching call(s)/message(s) (until its validity has expired).
  - **Restart timeout** – the record validity is restarted after use and the record may be reused for the next matching call(s)/message(s) (until its validity has expired).
  - **Delete record** – the record is deleted after use.
  - **Time [mins]** – use this parameter to set the validity period for each record of the AutoClip router. When it is checked, the given record has an unlimited validity.

6.11 Storage Manager

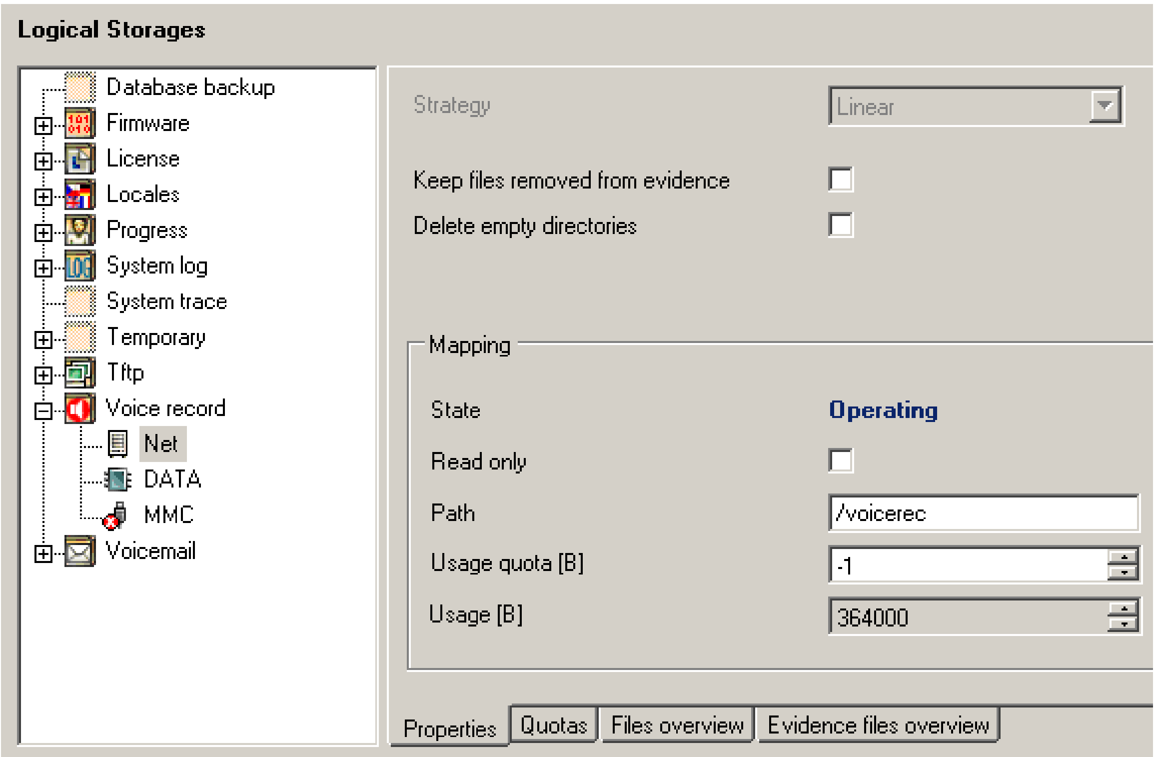
Find the storage manager in the **Global data – Storage manager** menu. This menu helps you define all storages necessary for the PBX operation and services. In addition to classical internal storages (such as DATA, NAND), you can map network disks and MMC cards, which make the usable space almost unlimited and provide access to such services as call recording, for example.

Logical storages

Logical storages represent the basic storing units for all PBX services and functions. You can add logical storages to the PBX from a pre–defined set but cannot create logic storages of your own. Logical storages themselves have no reserved data space. Hence, you have to map one physical storage at least to each logical storage, such as the internal memory, MMC card or network disk (CIFS – Common Internet File S.).

Right–hand button context menu actions:

- **Expand all** – unfold the logical storage tree into a view of the physical storages.
- **Collapse all** – fold the tree structure into a view of the logical storages.
- **Add logic storage** – select and add a pre–defined logical storage.
- **Add physical storage** – add a physical storage to the currently selected logical storage. You can only add physical storages as created in the **Physical storages** section.
- **Remove** – remove the selected physical storage or logical storage including all of its physical storages.
- **Remove all** – remove all logical and thus all physical storages from this section.
- **Default** – reset the current structure of logical/physical storages to the default values. All the physical storages added are ignored and remain in the structure.



Properties

- **Strategy** – select how to choose physical storages for the given logical storage. A linear strategy is only available at present. The selection is active when you click on any of the logical storages. **You can change priority of the physical storages by Drag&Drop function.**
  - **Linear** – data are stored in a sequence starting from the first physical storage. When the first storage is full, the next physical storage in the sequence is used.
- **Keep files removed from evidence** – if you tick off this option, the removed files will be deleted from the register but remain in the physical storage.
- **Delete empty directories** – a file is deleted automatically including the directory when its validity terms expires.
- **Mapping**– the section is active when you click on any of the physical storages and includes the following parameters.
  - **Status** – this parameter displays the current status of the selected physical storage within a specific logical storage, including whether the storage is available or full.
  - **Read only** – this parameter disables file editing for the given physical storage. The files can be read only.
  - **Path** – this parameter defines the path to a directory within the logical storage mapped. This accelerates physical storage mapping as you just search the selected directory instead of the entire structure. Make sure that the selected directory really exists in the specified location.
  - **Usable space** – the parameter specifies some space in the currently selected physical storage to be occupied by the given logical storage. It is because one physical storage may be used by multiple logical storages (by defining the used directory, e.g.), each of which may be assigned a different data space size. Set **–1** to leave the space unlimited on this level.
  - **Used** – this non–editable parameter informs of the current occupation of the space reserved for the selected physical storage.

Quotas

- **Table columns:**
- **Subject type** – this column displays the subject type to which the row relates. Choose User, Group, Virtual port, or Virtual port type.
- **Subject** – defines a subject of the above selected type.
- **Usage quota** – defines the usable space for the given subject within the logical storage data space (for all its physical storages).
- **Item size** – defines the maximum file size for the given subject.
- **Number of items** – defines the maximum count of files to be stored for the given subject within the logical storage data space.
- **Item life [s]** – sets the time for which the file stored for the given subject shall be kept in the logical storage data space.
- **Delete oldest achieving quota** – enables deletion of the oldest user files within the logical storage data space (when the file retaining time expires).

List of files

If you are on the logical storage level, you can see all files contained in the corresponding physical storages. If you select a physical storage, you can only see the files saved in the particular physical storage.

Right–hand button context menu actions:

- **Re–read view** – you can refresh the current file list within the logical storage data space.
- **Remove** – use this option to remove the selected file.
- **Rename** – use this option to rename the selected file.
- **Create directory** – use this option to create a directory within the data space of the currently used physical storage.
- **Add file** – add a file from a PC to the currently used physical storage.
- **Save file** – load a file from the logical storage data space into a PC.

Table columns:

- **Name** – file name.
- **Size** – file size.
- **Changed** – last file revision date and time.
- **Attributes** – additional information on the file to be used by the system.

List of locked files

Right–hand button context menu actions:

- **Save** – load a file from a storage to a PC.
- **Listen** – play the selected file.
- **Remove** – remove the selected file from the storage.
- **Remove all** – delete all files from the selected storage.

Table columns:

- **Name** – name of a locked file.
- **Created** – file creation date/time.
- **Validity** – file locking time, or file lifetime in the physical storage. The file will be deleted when this time expires.
- **Size** – file size.
- **Subject type** – subject type to which the file belongs.
- **Subject** – file owner.
- **Media** – type of the memory medium connected (MMC, USB, ...). **Not implemented yet.**

Physical storages

Right–hand button context menu actions:

- **Add** – add a row for your own physical storage mapping.
- **Rename** – rename the selected physical storage.
- **Remove** – remove the selected physical storage.
- **Remove all** – remove all physical storages defined.
- **Default** – reset the default physical storages for the PBX.

Table columns:

- **Name** – name of the physical storage.
- **Type** – basic type of the physical storage. Choose **Built–in**, **Network**, or

Removable.

- **Access point**– define the path to the storage.
    - **Removable or built–in** – a set of pre–defined paths to specific parts of the internal data space or the MMC card slot.
    - **Network** – define the path to the shared space of the network disk as for classical sharing (e.g. //192.168.122.110/netstar_data).
- **Usage quota** – define the total space to be used by a physical storage for all of its PBX functions. When the limit is exceeded, the physical storage will be put out of operation.
- **Network type** – choose either **Microsoft Windows** or **Nfs**. Used for network connections only.
- **Login** – set the login for connection to the shared space on the network disk. Used for network connections only.
- **Password** – set the passwod for connection to the shared space on the network disk. Used for network connections only.
- **Connection attempt in [s]** – define the intervals in which the PBX attempts to get connected to the given storage.

Physical Storages

Name	Type	Access point	Usage quota ...	Network type	Login	Password	Connect atte...
ROOTFS	Onboard	Nand0 - rootfs partition	-1				-1
DATA	Onboard	Nand0 - data partition	31457280				-1
TMP	Onboard	Tmpfs - temporary partition	8388608				-1
LOG	Onboard	Tmpfs - log partition	6291456				-1
MMC	Removable	Mmc - slot 1	-1				-1
Net	Network	\\192.168.122.110\sdileni	-1	Microsoft windows	rubas	***************	-1
Net 2	Network	\\192.168.122.110\voicemail	-1	Microsoft windows	rubas	***************	-1
Net 3	Network	\\192.168.22.33\nahravky	-1	Microsoft windows	TP	***************	-1

Detected storages

Table columns:

- **Physical storage** – name of the physical storage.
- **Type** – type of the physical storage (Built–in, Network, Removable).
- **Access point** – path to the storage as described in the Physical storages above.
- **State** – current state of the physical storage.
- **Root path** – root directory of the physical storage.
- **Media** – type of the memory medium connected (MMC, USB, ...). **Not implemented yet.**
- **Usage** – amount of data saved in the physical storage.
- **Free size** – current free space in the physical storage.
- **Total size** – total amount of data space in the physical storage.

Detected Storages

Physical storage	Type	Access point	State	Root path	Media	Usage [B]	Free size [B]	Total size [B]
ROOTFS	Onboard	Nand0 - oddíl rootfs	Ok	/opt/netstar/		24924312	11223040	41943040
DATA	Onboard	Nand0 - oddíl data	Ok	/data/netstar/		14192572	34992128	41943040
TMP	Onboard	Tmpfs - oddíl temporary	Ok	/tmp/		0	8388608	8388608
LOG	Onboard	Tmpfs - oddíl logování	Ok	/var/log/		2662797	3584000	6291456
Net	Network	\\192.168.122.110\sdileni	Ok	/media/net_00/		12107776	80073867264	131069833216
Net 2	Network	\\192.168.122.110\voicemail	Ok	/media/net_01/		0	80073871360	131069833216
Net 3	Network	\\192.168.22.33\nahravky	Ok	/media/net_02/		0	16722554880	320070479872

6.12 Scheduled tasks

This menu helps you schedule your database back-up, PBX restart, UMTS board restart or sending keepalive messages informing of the PBX operation. Click on **Add** in the context menu to add an event. Select the event type, name and repetition mode in the next window.

Database back–up

This option helps you schedule your database back-up intervals easily, especially in case of incidental data loss or configuration changes. The database is stored in the physical storage defined in the Storage manager in pre-set intervals. The storage can be a MMC card or a shared directory on a network disk. The database is stored with a timestamp designating the storing date/time and the current firmware version for later use.

PBX restart

Deferred PBX restart is applied, e.g., in upgrades as the PBX cannot be restarted as soon as the new firmware is downloaded due to being currently used.

UMTS boards restart

Use this option to schedule the restart of the UMTS boards.

PBX Keepalive

Keepalive messages are used for PBX operation checks. To send the keepalive reports successfully, set the Event Reporter in the **Network – Supervision services** menu. Set the event type to **PBX keepalive** for the object created. **The Event reporter is a licensed function.**

The following options are available in the Scheduler menu:

- **Type** – select the type for the event scheduled.
- **Name** – enter the name for the event scheduled.
- **Schedule task** – select the repetition frequency for the event scheduled.
    - **Not scheduled** – the event is not scheduled in this mode.
    - **Daily** – specify in how many days the event shall be repeated.
    - **Weekly** – specify in how many weeks and on which days the event shall be repeated.
    - **Monthly** – specify on which days or days of which weeks of the selected month(s)the event shall be repeated.
    - **Once only** – select the execution date for the event scheduled.
- **In time** – define at what time the event shall be executed on the selected day. We recommend you to select a time off the PBX peak load (i.e. off working hours) for all events except for the keepalive messages.

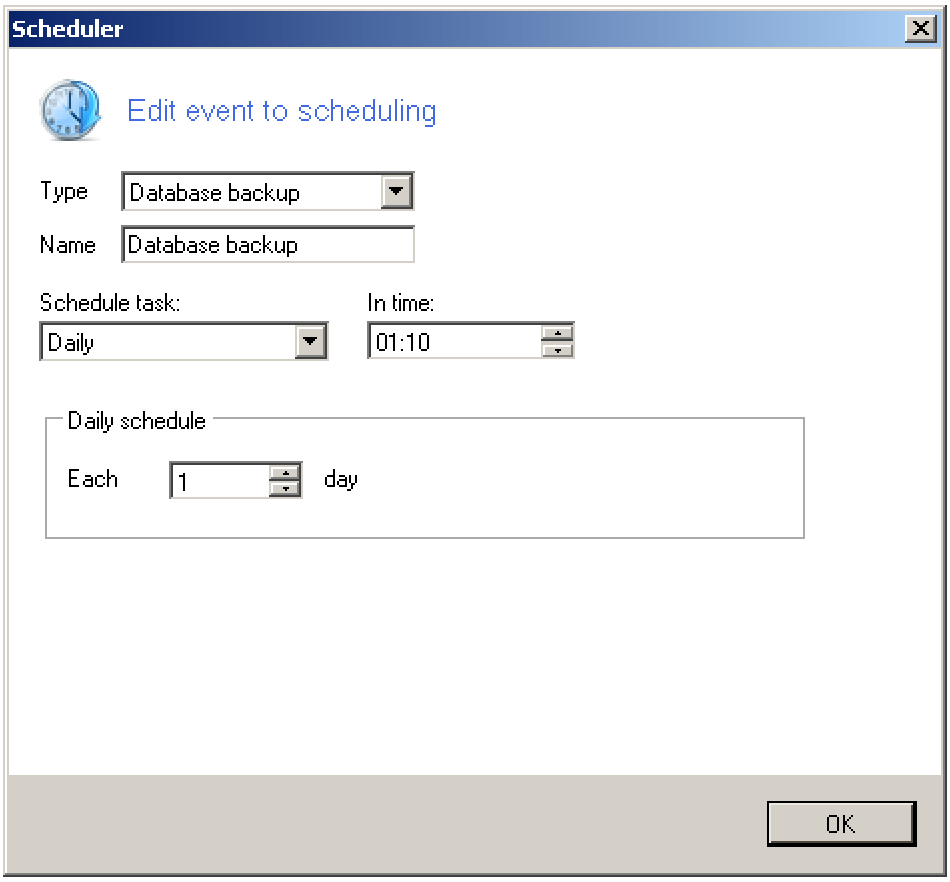


Figure: Daily Database Back–Up Configuration Example

Do action immediately – make the selected event be executed the moment the button is pressed.

6.13 Status Control parameters

The **Global data – Status Control parameters** menu is used for defining states of the Status Control objects created in the **Routing – Routing objects – Status Control objects** menu. Figure 1 shows an overview of the parameters to be defined. You can add and remove the rows using the right-hand mouse button context menu

State definitions

State	State name	Message text	Called number	Tone after set	State color
Unready		unready	804	Service Set	Maroon
Ready		ready	801	Service Set	Green
Error		err	802	Service Set	Lime
OK		ok	803	Service Set	Yellow

Figure: Pohled na menu Status Control parametry

The meanings of the columns are as follows:

- **State** – define the state of the Status Control object. The following options are available: **Ready**, **Unready**, **OK** and **Error**.
- **State name** – this optional parameter makes identification of multiple states easier.
- **Message text** – based on a match of the text entered here and the text of the message sent to the Status Control object, the proper state is selected for the Status Control object.
- **Called number** – based on a match of the number entered here and the CPN, the proper state is selected for the Status Control object.
- **Tone after set** – this tone is played to the calling subscriber whenever a Status Control object state change occurs. If the tone is not set, the call will not be answered in the Status Control object (i.e. the call will not pass into CONNECT), but the state change will be executed.
- **State color** – assign one of the available colours to the object. Whenever the Status Control object state changes, its colour will change accordingly in the Operator menu of the 2N® NetStar Assistant. Should one state be assigned multiple colours in the table, the colour of the first state is used for all identical states.

> A user must have the **Operator administration** right in order to display the Operator menu in the **2N® NetStar** Assistant. Refer to the **Users – User rights** menu for the user right settings.

Click on the column header to arrange the rows in the ascending or descending order according to their names or numerical values in the cell.

6.14 DTMF

Refer to the **Global parameters – DTMF** menu for DTMF profile settings. Select the profile for DTMF detection using this menu.

6.15 Causes

Here is what you can find in this section:

- Cause Objects
- User Causes
- Cause Mapping Tables

Cause Objects

In this menu, you can create sets of causes to be used for modifying bundle parameters. The menu is divided into two parts. You can add, remove and rename objects on the left and edit the selected objects on the right. The following options are available on the right-hand side:

- **Name** – name of the selected cause object.
- **Respond to** – use this parameter to specify the object's behaviour with respect to the causes entered.
  - **Unspecified** – the object shall respond to all causes unspecified in the Cause item.
  - **Specified** – the object shall only respond to the causes specified in the Cause item.
- **Cause** – define causes by selecting any of the pre-defined ones. In addition to common causes, you can use specific PBX causes or user causes defined in the **Global data – Causes – User causes**menu. See below for some specific causes. The following options are available under the right-hand mouse button:
  - **Add** – add a row.
  - **Remove** – remove the selected row.
  - **Remove all** – remove all rows all at once.
- **Specific causes**
  - **Invalid licence** – the cause warns that the licence is invalid.
  - **Low credit** – the cause warns that the virtual port credit has been exhausted.
  - **Recording not ready** – the cause notifies a call recording error due to inaccessibility or unavailability of the storages mapped, for example.

User Causes

In this menu, you can add user causes to be used within other objects if necessary (Cause objects or Cause mapping tables, e.g.). The following options are available under the right-hand mouse button:

- **Add** – add a row.
- **Remove** – remove the selected row.
- **Remove all** – remove all rows all at once.

The table consists of two columns with the following meanings:

- **Assigned Id** – the columns shows the Id that is automatically assigned to this user cause and used by the PBX.
- **Desription of cause** – the column defines the user description of the cause, which replaces the Cause ID in other menus.

Cause Mapping Tables

In this menu, you can specify changes in selected causes. By assigning causes to different types of virtual ports you can present identical causes in a different way on the PBX interfaces. To assign a mapping table to a virtual port use the Basic tag for the particular virtual port. You can also specify in which direction the mapping table should be used. One and the same table can be used for different interfaces and both directions at the same time. Hence, an internal caue can be translated into a cause towards the ISDN, SIP or GSM interface and also in the opposite direction.

The menu has two sections. You can add, remove and rename mapping tables on the left. The right–hand section includes two parameters and the mapping table.

- **Name** – name of the selected mapping table.
- **Mask stack cause** – use this option to disable displaying of the original cause in the PBX trace for the whole mapping table. In that case, the trace displays **Type:None**.

Name **Cause map 1**

Mask stack cause ☐

Cause	Msk	Q.850 Val	Valid	Q.850 Loc	Test	Set	GSM Type	GSM Value	Valid	SIP value	Valid
Call rejected	☐	21	☑	1	☐	☐	8	31	☑	486 Busy Here	☑
User not responding	☐	19	☑	3	☐	☐	0	0	☐		☐
REC Error	☐	0	☐	0	☐	☐	0	0	☐	405 Method Not Allowed	☑

Mapping table

The context menu helps you add, remove and remove all rows of the table. The table consists of twelve columns and an unlimited number of rows. The sequence of rows is irrelevant unless there are two rows with an identical cause and different settings. In that case, the earlier–added row is applied (the one higher in the configuration).

- **Cause** – choose one of the pre–defined PBX causes. Here user causes are applied.
- **Mask** – you can disable displaying of the original cause in the PBX trace for a selected mapping table row.
- **Q.850 value** – you can enter the particular cause value according to Q.850 to be assigned to the cause in the given row.
- **Valid** – enable translation for an ISDN stack.
- **Q.850 location** – define the Location value to be used in DSS1 for specification of the network or user from which the cause is coming. For the acceptable values see the table below.

Dekadicky	Význam
0	User
1	Private network serving the local user
2	Public network serving the local user
3	Transit network
4	Public network serving the remote user
5	Private network serving the remote user
7	International network
10	Network beyond interworking point

- **Test** – this option relates to column Q.850 loc and is used in the inbound direction (Stack to CP). If it is not checked off, column Q.850 loc need not match and the row is recognised according to Q.850 val. If it is checked off, both the values have to match.
- **Set** – this option relates to column Q.850 loc and is used in the outbound direction (CP to Stack). If it is not checked off, column Q.850 val is and column Q.850 loc is not included in the outgoing message. Otherwise, the message includes both the values.
- **GSM type** – define the GSM message type.
- **GSM value** – enter a GSM cause value to be assigned to the cause in the given row.
- **Valid** – enable translation for a GSM stack.
- **SIP value** – choose one of the available causes for the given row.
- **Valid** – enable translation for a SIP stack.

6.16 Time Parameters

Here is what you can find in this section:

- Date and Time
- Time Conditions
- Holidays

Date and Time

In this menu you can find the current date and time of your PBX including the time zone. Figure below shows a basic view of the **Date and time** menu. The date format is **year/month/day** and time is displayed in the **24–hour** format.

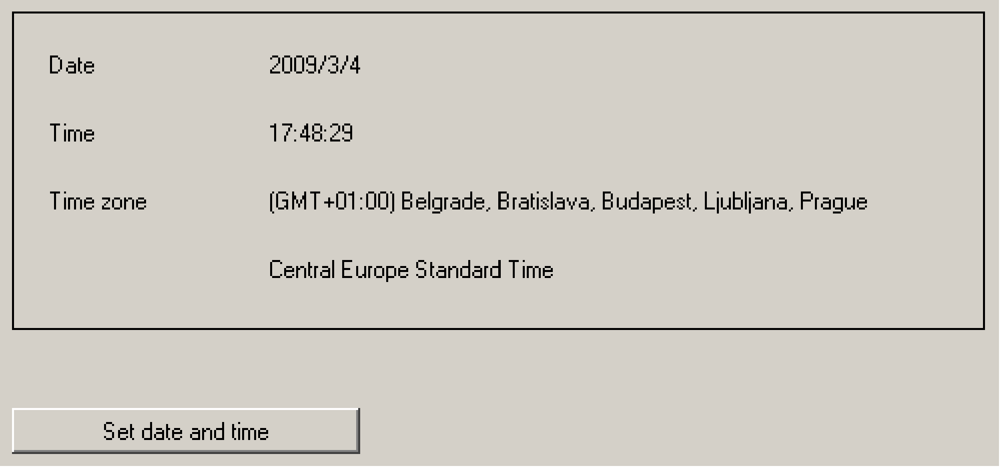


Figure: View of Date and Time Setting Menu

Push the Set date and time button to display a dialogue box as shown in figure below. Select a calendar item or use the arrows in this window to change the date. Type the day/year values to set the date.

To set time, type the values or use the arrows. Standard 0–23 hour and 0–59 minute/second limitations are applied.

Choose a time zone from the list of time zones.

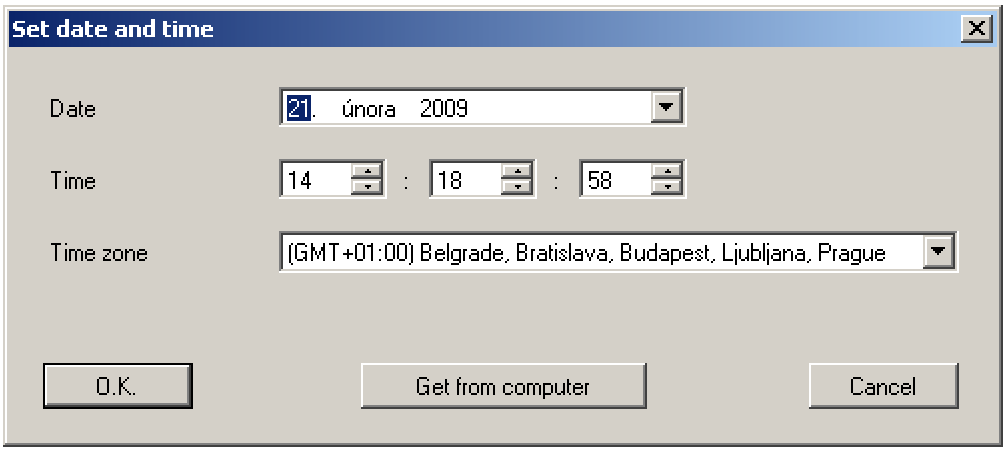


Figure: View of PBX Date and Time Setting Dialogue

Time Conditions

To define the time conditions use the **Global Data – Time Parameters – Time Conditions** menu. The menu is divided into two parts. A list of available time conditions is on the left and can be created, removed or renamed here via the context menu. On the right you can compile the time conditions. A time condition can consist of several simpler rules that are added up. You can specify, add, remove or edit the selected time condition rules in the context menu.

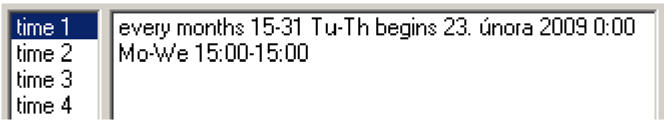


Figure: Basic View of Time Condition Menu

First select the time condition rule to be added or modified using the **Add** or **Edit** optio ns. A dialogue box as shown in figure below is displayed for you to select the rules.

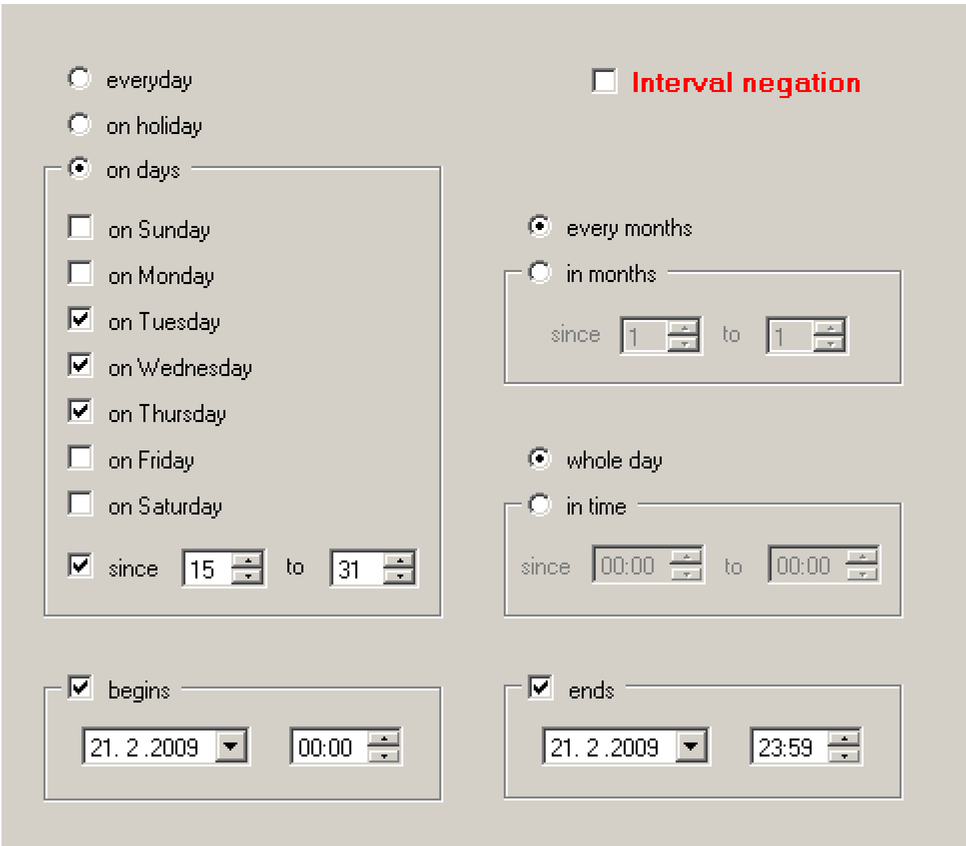


Figure: Part of Time Condition Editing Menu

With regard to a complexity of time conditions, the following time condition setting rules have been defined:

1. Parameters that optionally define the absolute time limits (i.e. beginning and end) have been introduced. The time interval has to obey the limitation if applicable regardless of any other settings (including **Interval negation**). To

define the time limit, use the **From** and **To** checkboxes and **Date** and **Time** fields in the upper part of the time limit setting window.

2. The other fields except for **Interval negation** define a repeat rule for each part of the definition. An interval is valid for a selected time point if:
    a. Holiday is not checked or the selected time point represents any of the defined holidays;
    b. No weekday is checked or the selected time point represents a checked weekday; and simultaneously
    c. no day is checked or the selected time point represents a day within the specified day range;
    d. no month is checked or the selected time point represents a month within the specified month range;
    e. no time is checked or the selected time point represents some time within the specified time range;

- If the **Interval negation** is checked, rule 2) is inverted. The limits described in item 1) are not affected.
- Below, the parameter criteria are specified:
- If defined, the **From** and **To** options have to contain valid day and time data. If you set the **From** and **To** parameters at the same time, then the **From** value may not be greater than the **To** value;
- If the holiday option is selected, no other options may be checked except for **Interval negation**;
- If the day option is selected, the **From** and **To** parameters must range between 1 – 31 (cum) and the **From** value may not be greater than the **To** value;
- If the month option is selected, the **From** and **To** parameters have to range between 1 – 12 (cum) and the **From** value may not be greater than the **To** value;
- If the time option is selected, the **From Hour** and **To Hour** parameters have to range between 0 – 23 (cum) and the **From Minute** and **To Minute** parameters have to be in the range of 0 – 59 (cum). The composed time parameter **From** (Hour + Minute) may not be greater than the composed time parameter **To** (Hour + Minute).

The time conditions can be used for call routing or user profile switching.

Holidays

To define holidays and important days use the **Global Data – Time Parameters – Holidays** menu. The menu is divided into two parts. A list of available holidays is on the left and the setting options are on the right. To add a holiday, choose the **Add** opti on in the context menu. Then choose a day in the calendar to the right. You can define holidays for the current year or select a holiday that repeats periodically using the **Valid every year** item below the calendar. The holidays are not arranged alphabetically but according to their dates. You can also load the holiday list from a predefined file via the **Update from file** option. You can remove and rename holidays as necessary.

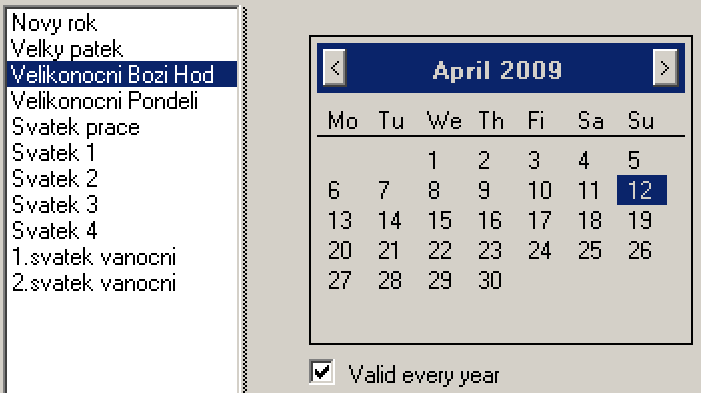


Figure: View of Holiday Adding Menu Used by Time Conditions

6.17 Assistant

Here is what you can find in this section:

- Administration Settings
- User Relations

Administration Settings

What is Assistant?

The Assistant is a web application for user account supervision. The web server for this application can be run from a PBX or an external computer. The web server version has to be the same as that of the PBX firmware. In the **Assistant** menu you can find three submenus for an easy Assistant managing and active session monitoring.

Administration settings

The **Assistant – Administration Settings** menu provides the following basic application settings:

- **Confirm deleting** – use this option to enable confirmation of record removing from the call history. If this option is checked, the user is asked for confirmation before removing a record.
- **Default language** – use this option to select the application language from a list. Currently, the list includes three languages - Czech, English and Finnish.
- **Image directory** – use this option to select one of the predefined image sets.
- **CSS style file name** – use this option to set the CSS style to be used for the application.
- **Maximum user session time [min]** – use this parameter to set the logout timeout for an inactive user.

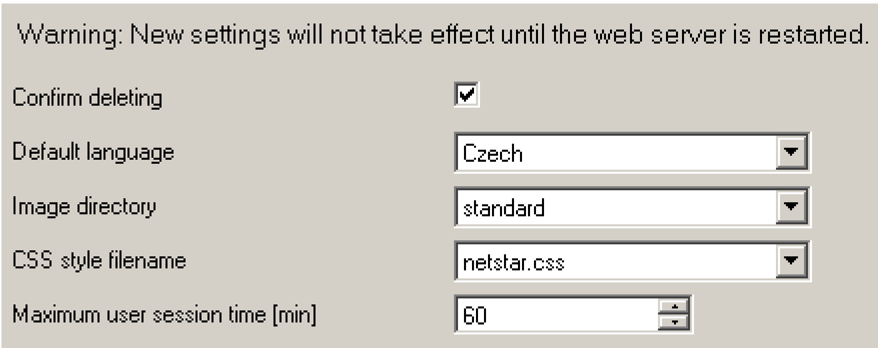


Figure: View of Assistant Web Server Setting Menu

User Relations

In the **Assistant – User Relations** submenu you can find the list of all active sessions. There are three columns in the list with the following meanings:

- **Username** – shows identification of each user session within the database.
- **Session ID** – shows the user that corresponds to a specific session.
- **Last access time** – shows the last user activity time in a specific session.

7. Routing

Here is what you can find in this chapter:

- 7.1 Routers
- 7.2 External Routers
- 7.3 Complex Routers
- 7.4 Switch Routers
- 7.5 Routing Objects
- 7.6 Identification Tables
- 7.7 AutoClip Routers

7.1 Routers

Router

The router is a set of rules used for incoming call routing through the PBX. Routers are defined in the **Routing – Routers** menu, which consists of two windows. The left window displays a list of available routers. The right–hand window helps you configure a selected router. On the left–hand side of the menu you can use the context menu with the following options:

- **Add**– use this option to initiate a router adding dialogue. Then enter the name and type of the new router. After creation, the router types are colour distinguished for convenience. Choose any of the following router types:
  - **Called number** – use this option to add a router that routes calls according to the CPN.
  - **Calling number** – use this option to add a router that routes calls according to the CLI.
  - **Called number type** – use this option to add a router that routes calls according to the called number subtype (CPN subtype), i.e. **Internal**, **Local**, **National**, **International** or **Unknown**.
  - **Calling number type** – use this option to add a router that routes calls according to the calling number subtype (CLI subtype), i.e. **Internal**, **Local**, **National**, **International** or **Unknown**.
  - **Call type** – use this option to add a router that routes incoming calls according to the call type, i.e. **voice**, **fax** or **data** calls.
  - **Port** – use this option to add a router that routes calls according to the incoming carrier.
  - **Text** – use this option to add a router that routes incoming SMS messages according to the text.
- **Delete** – use this option to delete a selected router. If the router is not empty (has some rows), you will be asked for confirmation. If you delete a router, all the associated data are deleted too.
- **Delete all** – use this option to delete all of the created routers.
- **Rename** – use this option to rename a selected router. If you fill in an already used name, you have to change it or abort renaming.
- **Add router group** – use this option to add a new group of routers.
- **Add router subgroup** – use this option to add a subgroup to the currently selected group of routers.
- **Move to root level** – use this option to move the selected object to the highest level beyond all groups and subgroups created.
- **Move group content** – use this option to move the content of a group or subgroup into another router group or subgroup.
- **Default** – use this option to delete all the current routers and create new default routers according to the router list. These new routers are automatically filled with services, users and extensions.
- **Default from file** – this option has a similar function as Default, but in this case you can choose a file of your own for creating new routers.
- **Update** – use this option to update the currently used routers including settings.
- **Update router** – use this option to fill a router with services, users or extensions. If the given records already exist in the router, they are ignored, if not, they are added to the router end.

- **Update from file** – this option has a similar function as **Update**, but in this case you can choose a source file of your own. The existing routers are not deleted but completed with missing records.
- **Export to file** – use this option to back up all routers including records in the xml file format.
- **Export router to file** – use this option to back up the currently selected router in the xml file format.
- **Copy router** – use this option to make a copy of the currently selected router. All you have to do is enter a new name. The router copy contains the same data as the original router (including the default destination).
- **Show objects routed to router** – use this option to open a side window to see the list of all objects to be routed to the selected router. This function helps you check the PBX routing settings.
- **Expand all** – use this option to open the whole structure of groups and subgroups with routers easily.
- **Collapse all** – use this option to close the whole structure of groups and subgroups with routers easily.

Moving records using the mouse, also called **drag & drop**, has been implemented in this menu for easier moving of existing routers, or router groups and subgroups.

Call Routing

Call routing is executed similarly in all router types. First, the row that matches the incoming information (CPN/CLI, CPN/CLI subtype, call type, incoming carrier or SMS text) is found and then the rule specified therein is applied. In the case of prefix congruence the following columns are applied and in the case of absolute congruence of all criteria the higher row is preferred. To change the row priority use the arrows on the right–hand side of the screen. To insert rows with a certain priority use the **Insert ahead selected** and **Insert behind selected** options. The **Add** option is used for adding a record behind the currently last one (i.e. the lowest priority record). **Add copy of row** and **Add copy of row to another router** are also useful functions, allowing you to add a selected row to a router of the same type or a router with the same column header. Some router types also enable to change the number or SMS text used for routing. The sections below describe all available router types and their configurable parameters.

By Called Number

This router is based on called number (CPN) routing. The router consists of twelve columns with the following meanings:

- **Prefix** – sets a part or the whole of the called number. When this prefix matches the incoming CPN, this row can be used for routing. In this column you can use all digits, characters ***, #, +** and letters **A**, **B**, **C**, **D**, which can also be DTMF dialled. The question mark (**?**) can substitute any digit (or character), but not the whole number (or prefix). Therefore, to substitute all of the three–figure prefixes **xyz**, you have to use three question marks, i.e. **???**. Generally, the ***** character is used for services like the **#** character, which is also used for dialling end signalling.
- **Digits after**– this column provides details on the called number length for a row (the prefix can be followed by a different count of digits). This number sets how many digits are to be awaited after the prefix before the call is routed to another

destination according to the preset rule.

  - **"0"** – no more digits are awaited.
  - **">0"** – the process waits for a given count of digits (characters).
  - **"–"** – the dash indicates an unknown length of the called number. Dialling should be terminated by adding a **#** or by the timeout expiry.

In the case of an 'unknown length' of the called number, the call is routed immediately upon prefix recognition and the following digits are transmitted to the destination according to the rule (generally to another router or to the public network). Otherwise, the call is not routed until the whole number has been dialled (according to the preset prefix and count of expected digits, so the number need not be complete at all). Therefore, remember to sort prefixes from the longest to the shortest ones while using the 'collision routing'.

The called number can also be changed in this router type. Having passed through the router, the call can be routed to another router of the same type where, however, it is routed according to the number modified by the preceding router. Use the following columns for CPN changing:

- **Remove from beginning** – here define the count of digits to be removed from the called number beginning.
- **Add to beginning** – here fill in the string to be added to the called number begin. Use this column only if the called number length is other than '–' (dash) in the **Digits after** column. Doing this use the following symbols:
- **Number** – means digits, letters A, B, C, D and characters *, #, +.
- **,** – the comma means waiting for one second.
- **p(X)** – **X** represents the count of seconds of waiting. This instruction is equivalent to entering an X number of commas.
- **t** – determines whether the preset number will be dialled after connection to a voice channel (**t** used), or whether dialling will be delayed before connection (**t** unused).
- **Remove from end** – here define the count of digits to be removed from the called number end. Use this column only if the called number length is other than '–' (dash) in the **Digits after** column.
- **Add to end** – here fill in the string to be added to the called number end. Use this column only if the called number length is other than '–' (dash) in the **Digits after**column. Doing this use the following symbols:
  - **Number** – means digits, letters A, B, C, D and characters *, #, +.
  - **,** – the comma means waiting for one second.
  - **p(X)** – **X** represents the count of seconds of waiting. This instruction is equivalent to entering an X number of commas.
  - **t** – determines whether the preset number will be dialled after connection to a voice channel (**t** used), or whether dialling will be delayed before connection (**t** unused).
- **Scheme** – here you can change the called number scheme to **Number** or **URI**. The default value of this column is **Preserve**.
- **Subtype** – here select the called number subtype as **Internal**, **Local**, **National**, **International** or **Unknown**. The default value of this column is **Preserve**.

Examples

1. The instruction **t1p(5)3,,*6** means that after the other party answers the call, you dial digit 1, wait for five seconds, dial digit 3, wait for two seconds and, finally, dial * and digit 6.
2. The instruction **1,2,,3p(3)456** means that digit 1 is dialled followed by a one–second delay, then digit 2 is dialled followed by a two–second delay, digit 3 is dialled followed by a three–second delay and, finally, digits 4, 5 and 6 are dialled.

- **Destination type**– sets the type of destination to which an incoming call should be routed. Choose an item from the list of available PBX routing objects. There are three options in the column that need more explanation:
  - **Default** – use this option to route the incoming call directly to the next routing level (if any). It is generally used for sorting objects into sets. With the Default option, the incoming call is routed back to the superior set and the next set item is used.
  - **Disabled** – use this option to terminate the incoming call routing immediately. The calling user will hear the congestion tone.
  - **Origin** – use this option to return a modified number from the given router back to the incoming port (through which it came to the PBX).
- **Destination** – this column selects a destination within the above–selected destination type.
- **Tone** – in this column define the tone to be played to the calling user after prefix dialling in the case of overlap sending. The tone is played after dialling end only in case the called number length has not been defined and a router is the next destination.
- **Time condition** – use this column to set a time condition for each router row. The routing rule is valid only during the time condition validity period. Time conditions help you create sophisticated routing schemes according to time. You can route a call to different destinations for the same incoming conditions (except for time).
- **Default destination**– if no match is found in the Prefix column, the call is routed as defined in this option (located below the routing rule table):
  - **Type** – in this part set the type of destination to which an incoming call is to be routed. Choose an item from the list of all available PBX routing objects.
  - **Id** – in this part select the destination of the above–selected destination type.

By Calling Number

This router is based on routing according to the calling number (CLI). The router consists of twelve columns with the same meanings as the case is in the By called number router. The only difference lies in that the CPN prefix is used and no instructions for delayed dialling are included. **All completed changes affect the resultant CLI!**

By Called Number Subtype

This router is based on routing according to the called number subtype (CPN subtype). The called party number subtype is the only parameter that comes into the router and cannot be changed there. The router consists of five columns with the following meanings:

- **Subtype**– is a part of the identification to be used for call routing. You can set five subtypes:
  - **Internal** – represents an internal phone number specified by the PBX administrator.
  - **Local** – represents a private network phone number in the local format.
  - **National** – represents a public network number in the national format with prefixes.
  - **International** – represents a public network phone number in the international format with prefixes.
  - **Unknown** – an unknown number format relating to none of the above mentioned subtypes.
- **Destination type**– this column sets the type of destination to which an incoming is routed by this rule. Choose an item from the list of all available PBX routing objects. In this column you can find three options that need more explanation:
  - **Default** – use this option to route the incoming call to the next routing level (if any). It is generally used for sorting objects into sets. With the Default option, the incoming call is routed back to the superior set and the next set item is used.
  - **Disabled** – use this option to terminate the incoming call routing immediately. The calling user will hear the congestion tone.
  - **Origin** – use this option to return a modified number from the given router back to the incoming port (through which it came to the PBX).
- **Destination** – here select a destination of the above–selected destination type.
- **Tone** – in this column define the tone to be played to the calling user after prefix dialling in case that a router is the next destination.
- **Time condition** – use this column to set a time condition for each router row. The routing rule is valid only during the time condition validity period. Time conditions help you create sophisticated routing schemes according to time. You can route a call to different destinations for the same incoming conditions (except for time).
- **Default destination** – if no match is found in the **Prefix**column, the call is routed as defined in this option (located below the routing rule table):
  - **Type** – in this part set the type of destination to which an incoming call is to be routed. Choose an item from the list of all available PBX routing objects.
  - **Id** – in this part select the destination of the above–selected destination type.

By Calling Number Subtype

This router is based on routing according to the calling number subtype (CLI subtype). The router consists of five columns with the same meanings as the case is with the By called number subtype. The only difference lies in that the CLI subtype is the only parameter coming into the router and cannot be changed there. The calling party number subtype changes made in this router are only used for routing and not for call identification.

By Call Type

This router is based on routing according to the call type (voice, data, video, etc.). All the columns have the same meanings as the case is with the By called number subtype except for the first one. The first column defines the call type. When a preset call type is recognised, the call is routed to the preset destination.

By Port

This router is based on routing according to the incoming port (the call comes into the PBX through this port). All the columns have the same meanings as the case is with the By called number subtype except for the first one. The first column defines the port. When a preset port is recognised as the incoming port, the call is routed to the preset destination.

Message routing

The last router type is an SMS router, which routes SMS messages according to their texts. This router can also be created in the **Routing – Routers** menu and cannot be used for call routing. It consists of five columns with the following meanings:

- **Prefix** – use this column to enter a text string to be recognised at the SMS beginning. After recognition, the SMS message is routed through the PBX according to the preset rule.
- **Replace**– use this column to edit SMS messages. You can either replace the existing text with another one or insert instructions with the following meanings:
  - **%c** – inserts the sender number (CLI).
  - **%l** – inserts the receiver number (CPN).
  - **%se** – erases the whole text of any length. If you leave the **Replace** column empty, it is translated as 'Don't change the incoming text'.
  - **%sr(B,E)** – inserts the original string omitting the first **B** number of characters and the last E number of characters.
  - **%ss("STRING",X,N)** – finds the **X–th** appearance of the **STRING** in the incoming SMS message. From this point on, you can leave **N** letters of the incoming text deleting all the others. By setting **N** to zero, you insert the whole text (from the mentioned point to the message end).
  - **%sm(B,L)** – from the **B–th** character of the SMS message on, you can insert **L** characters of the original text. By setting **L** to zero, you insert the rest of the text.
- **Destination type**– this column sets the type of destination to which an incoming SMS message is routed by this rule. Choose one of the destinations used for SMS routing only. In this column you can find two options that need more explanation:
  - **Default** – use this option to route the incoming SMS to the next routing level (if any). It is generally used for sorting objects into sets. With the **Default** option, the incoming SMS is routed back to the superior set and the next set item is used.
  - **Disabled** – use this option to terminate the incoming SMS routing immediately. **The SMS message will not be delivered!**
- **Destination Id** – this column sets a destination of the above–selected destination type.
- **Time condition** – use this column to set a time condition for each router row.

The routing rule is valid only during the time condition validity period. Time conditions help you create sophisticated routing schemes according to time. You can route a call to different destinations for the same incoming conditions (except for time).

- **Default destination**– if no match is found with any of the preset strings, the SMS message is routed as defined in this option (located below the routing rule table):
  - **Default type** – in this part set the type of destination to which an incoming SMS message is to be routed. Choose only one of the destinations that can be used for SMS routing.
  - **Default Id** – this part sets a destination of the above–selected destination type.

7.2 External Routers

External routers use the External Routing Machine (ERM server) for call and SMS routing. The ERM server partially replaces or complements the **2N® NetStar** internal routing mechanisms. Having received a call/SMS routing request, the PBX sends a query to the ERM server. If a matching record is found in the ERM server database table, the ERM server sends back a parameter specifying further call/SMS routing in the external router. DB connectors, which are used for setting communication with the ERM server, are an inseparable part of the external routers. Refer to the **Network – DB connectors** menu for DB connector settings.

The **Routing – External routers** menu consists of two windows. The left window includes a list of available external routers. The right window helps you configure the selected external router. The context menu opened in the left part of the menu offers the following options:

- **Add** – display the router adding window and enter the router name.
- **Delete** – remove the selected router including all database links associated with this object.
- **Delete all** – remove all the routers listed. You will be asked for confirmation before deleting.
- **Rename** – rename the existing router. If you enter an existing name, you will be warned and no change will be executed.
- **Add router group** – add a new group of routers.
- **Add router subgroup** – add a subgroup to the currently selected group of routers.
- **Move to root level** – move the selected object to the highest level beyond all existing groups and subgroups.
- **Move group content** – move the whole content of a group or subgroup to another group or subgroup of routers.
- **Copy router** – all you have to do to copy the currently selected router is enter a new name here. The copied router contains the same records as the original one (including the default destination).
- **Show objects routed to router** – display a side window for the selected router inclduing a listing of all objects routed to this router. This function is useful for PBX routing checks.
- **Expand all –** unfold the whole router group/subgroup structure easily.
- **Collapse all** – fold the whole router group/subgroup structure easily.

Call Routing

The process of call/SMS routing via an external router is simialr to that via an internal router. The only difference is that the external router uses a single input parameter, i.e. the **Parameter**. If the parameter value obtained from the ERM server is identical with the value included in an external router row, routing to the specified destination is executed.

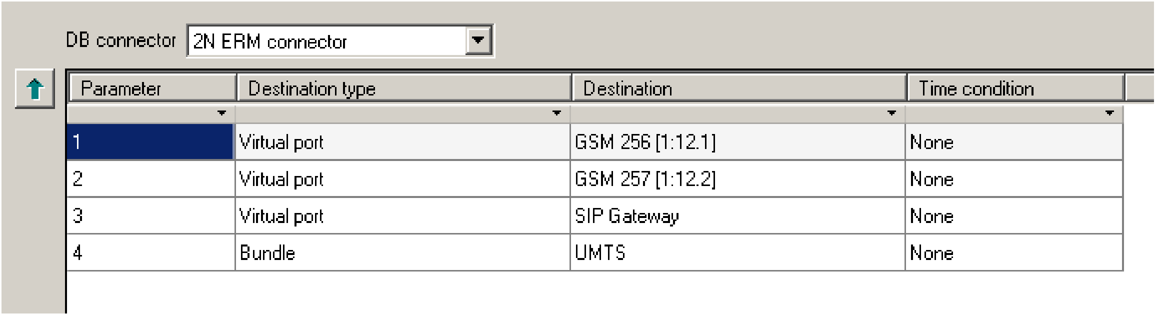


Figure: Pohled na nastavení externího routeru

- **DB connector** – this option helps you select the DB connector for communication with the ERM server. The external router does not work without DB connector assignment.
- **Parameter** – this column gives a string of characters to be compared with the string sent back by the ERM server. Alphanumerical characters can be used.
- **Destination type** – this column specifies the type of the destination to which the call is to be routed. All the PBX routing objects are available (if created). Moreover, there are three options where the destination is not obvious at first sight:
  - **Default** – with this option, the next routing level is used if existent. This function is primarily used for arranging objects into sets. The **Default** option returns the call to the superior set from the router and the next set item is used.
  - **Disabled** – use this option to discontinue the call routing process. The calling subscriber will get the busy tone.
  - **Origin** – use this option to return the modified number from the given router to the port from which the call came in.
- **Destination** - select a destination of the type selected in the preceding column.
- **Time condition** - assign a time condition to each router row. Such row will only be valid during the time condition validity term. Time conditions help you create more sophisticated call routing rules in dependence on time. Thus, you can route calls with the same input conditions to different destinations at different times.

- **Default destination** – if a match is not found in the **Parameter** column for the value returned by the ERM server, the call is routed to the default destination (item below the routing table):
  - **Type** - this parameter specifies the type of the destination to which the call shall be routed. All the PBX routing objects are available (if created).
  - **Id** - this parameter helps you select a destination of the selected type.

7.3 Complex Routers

This menu is used for complex routing of incoming calls through the PBX.

7.4 Switch Routers

> **The Switch Router function is available in 2N® NetStar firmware versions 3.1.1. and higher.**

Switch Router helps you modify call/SMS routing via the **2N® NetStar** PBX using a service called **Set switch router**. Make a call or send an SMS to the service to select a switch router and one of its predefined parameters, thus specifying the call/SMS destination.

The **Routers – Switch routers** menu consists of two windows. The left window displays a list of available switch routers. The right window helps you configure the selected switch router. The context menu opened in the left part of the menu offers the following options:

- **Add** – display the router adding window and enter the router name.
- **Delete** – remove the selected router including all database links associated with this object.
- **Delete all** – remove all the routers listed. You will be asked for confirmation before deleting.
- **Rename** – rename the existing router. If you enter an already existing name, you will be warned and no change will be executed.
- **Add router group** – add a new group of routers.
- **Add router subgroup** – add a subgroup to the currently selected group of routers.
- **Move to root level** – move the selected object to the highest level beyond all existing groups and subgroups.
- **Move group content** – move the whole content of a group or subgroup to another group or subgroup of routers.
- **Copy router** – all you have to do to copy the currently selected router is enter a new name here. The copied router contains the same records as the original one (including the default destination).
- **Show objects routed to router** – display a side window for the selected router incldUing a listing of all objects routed to this router. This function is useful for PBX routing checks.
- **Expand all** – unfold the whole router group/subgroup structure easily.
- **Collapse all** – fold the whole router group/subgroup structure easily.

Call Routing

The process of call and SMS routing via a switch router si similar to that via a standard router. The only difference is that the switch router uses a single input parameter, i.e. the **Parameter**. Routing obeys the curently active parameter or row. Use the **Set switch router**service to select a switch router and the parameter. Use the **Get switch router** service to know the currently active switch router row. The service informs the caller of the active row setting by playing the appropriate **Info tone**. You can also use the **2N® NetStar Assistant** for switch router configuration.

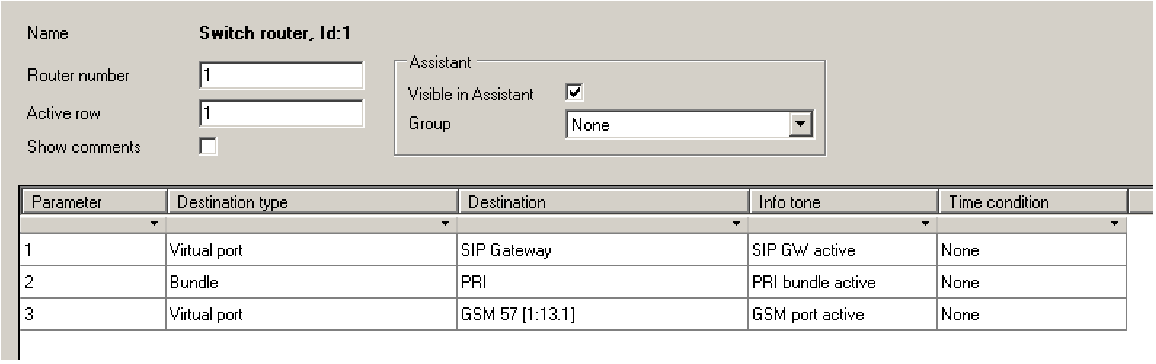


Figure: Pohled na nastavení přepínacího routeru

- **Router number** – router identification, entered into the service during router selection.
- **Active row** – currently active parameter of the switch router.
- **Show comments** – tick off to display the **Comment** column in the table. Enter a comment related to the row without affecting the call or SMS routing process. The comment is displayed automatically next to each switch router row in the **2N ® NetStar Assistant**.
- **Assistant** – use this block of parameters to set the switch router with respect to the Assistant user application.
    - **Visible in Assistant** – enable displaying of the switch router in the application. If you do not tick off this option, the given switch router will not be available within the application.
    - **Group** – set a group or subgroup of users authorised to work with the switch router within the application. If the given (sub)group has more subgroups, the switch router is only available to the users assigned to the (sub)group, not to the users of inferior subgroups.
- **Parameter** – this column gives a string of characters for identification of a router row for service based router setting. The column may include **numerical** characters only.
- **Destination type** – this column specifies the type of the destination to which the call is to be routed. All the PBX routing objects are available (if created). Moreover, there are three options where the destination is not obvious at first sight:
    - **Default** – with this option, the next routing level is used if existent. This function is primarily used for arranging objects into sets. The **Default** option returns the call to the superior set from the router and the next set item is used.
    - **Disabled** – use this option to discontinue the call routing process. The calling subscriber will get the busy tone.
    - **Origin** – use this option to return the modified number from the given router to the port from which the call came in.
- **Destination** - select a destination of the type selected in the preceding column.
- **Info tone** – select a tone to be played back to the calling subsscriber as row identification if the **Get switch router** service is used. **Make sure that the Info tone parameter is set to avoid Get switch router errors.**
- **Time condition** - assign a time condition to each router row. Such row will only be valid during the time condition validity term. Time conditions help you create

more sophisticated call routing rules in dependence on time. Thus, you can route calls with the same input conditions to different destinations at different times.

- **Default destination** – if a match is not found in the **Parameter** column for the value returned by the ERM server, the call is routed to the default destination (item below the routing table):
  - **Default type** - this parameter specifies the type of the destination to which the call shall be routed. All the PBX routing objects are available (if created).
  - **Default Id** - this parameter helps you select a destination of the selected type.

7.5 Routing Objects

Here is what you can find in this section:

- Bundles
- DISA
- Ring Groups
- Ring Tables
- Modems
- Sets
- Audio Inputs and Outputs
- Binary Inputs and Outputs
- CallBack
- Status Control Objects

Bundles

Bundle

The bundle is a routing object that enables to route an incoming call to one (or all) of the objects specified in the bundle. Choosing an object within a bundle depends on the selected strategy. The fact that an routing object is busy need not necessarily lead to routing termination. The call can be routed to another routing object either upon a busy router recognition or after a timeout as preset. For bundle parameters and their usage see below.

Bundle Setting

Bundles can be configured in the **Routing – Routing objects – Bundles** menu. A list of available bundles is displayed on the left. Add, delete or rename bundles using the context menu. You can also create predefined bundles with the Default option. The parameters of the selected bundle are shown on the right. The figure below shows a possible bundle configuration.

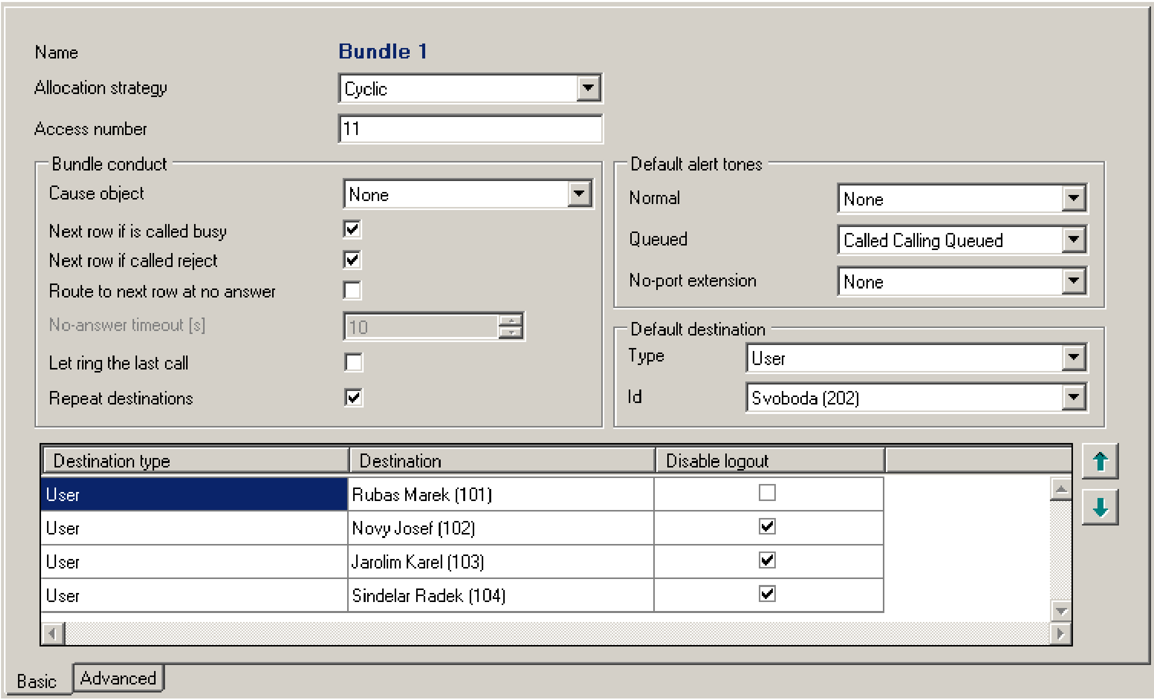


Figure: View of Bundle Configuration Menu – Basic

The above mentioned menu consists of the following parameters:

- **Allocation strategy** – here select the way of object choosing within a selected bundle. Choose **Linear**, **Cyclic**, **All** or **By credit**.
  - **Linear strategy** – an incoming call is always routed to the first bundle row. If this object is busy or unavailable, the call is routed to the next row or terminated (as preset).
  - **Cyclic strategy** – an incoming call is routed to the bundle row that comes immediately after the one used for the previous routing to this bundle. If

this object is busy or unavailable, the call is routed to the next row or terminated (as preset).

  - **All** – an incoming call is routed to all objects at the same time. Basically, the strategy substitutes the ring group function. The main difference, however, is that stations and users can login to a bundle using a service.
  - **By credit** – this strategy is intended for credit–monitored bundles with virtual ports. An incoming call is routed to the virtual port of the bundle with the currently highest count of free minutes. If there are more rows with the same count of free minutes, the sequence of rows in the bundle is respected. If a row object is busy or unavailable, the call is routed to the next row, or terminated.
- **Queue on bundle** – enable incoming call queuing for a bundle. Depending on the selected strategy, the bundle queue is active immediately upon the first, or after the second passage of the call through the bundle. With the **All** strategy, the queue is forced for all destinations. With the **Linear** or **Cyclic** strategy, destinations are called according to the strategy and the queue is disabled. Upon the first attempt to call all the bundle destinations, another cycle is completed with an active queue. This means that the call is routed with queue to all the bundle destinations that have not returned cause 21 – Call Reject or 18 – No user responding and/or a cause defined in the **Cause object for queue**.
- **Access number** – here enter the bundle number to be used for identification in the **Bundle login** and **Bundle logout** services.

Bundle conduct

- **Cause object** – use this option to select one of the cause objects as pre–defined in the **Global data – Causes – Cause objects** menu. These objects represent a set of causes to be responded to by the bundle. When one of the cause objects has been selected, the **Respond to busy** and **Respond to reject** options are disabled automatically.
- **Cause object for queue** – select this option to route an incoming call with queue to all the bundle destinations that have not returned the cause defined in the selected cause object.
- **Respond to busy** – use this option to route an incoming call to the next row in case the object is busy. This increases the successful routing rate. Here cause 17 – User Busy – is relevant only.
- **Respond to reject** – use this option to route an incoming call to the next row if rejected by the called user. Here cause 21 – Call reject is relevant only.
- **Route to next row at no answer** – use this command to proceed to the next bundle row in case the call is not answered within the timeout defined in the **No answer timeout [s]** parameter or under causes 18 – No user responding – and 19 – No answer from user.
- **Let ring the last call** – if this option is checked and **Route to next row no answer** is used, the incoming call is not routed to the Default destination after routing to the last unused object, but it rings at the last destination. The last unused object need not be the object from the last row of the bundle.
- **Repeat destinations** – if this option is cheked off and proceeding to the next row on busy is applied, the call is not routed to the default destination after the last bundle record, but routing starts from the first bundle record again. **First** means the **First used** within this incoming call. A timeout for repeating destinations and acceptable count of repetitions is defined in the Global parameters for convenience and deadlock protection. If a call is routed to a destination before the timeout expires, the destination pretends to be unavailable.

Default alert tones

In this section select variable alert tones for specific situations.

- **Normal** – sets the alert tone to be used in all situations except for the two cases mentioned below.
- **Queued** – sets the alert tone to be used for routing to a extension with an active queue. The incoming call has to be queued, otherwise the normal alert tone is used.
- **No–port extension** – sets the alert tone to be used for routing to the user with a no–carrier extension (external extension). Such user has to be assigned an external extension and another internal (active) extension at least. Otherwise, you will hear another alert tone.

Default destination

Here select a destination to be used whenever the call is rejected on the last bundle destination, the next row proceeding timeout expires, or the call cannot be delivered for any other reasons (all destinations are busy or logged out). Default destination routing depends on the **Bundle conduct** settings.

In addition to the above mentioned parameters, an object adding table is available.

The table consists of two columns with the following meanings:

- **Destination type** – in this column select the type of the routing object to be used for incoming call and SMS routing. Define the extension, user, carrier, set, ring group, another bundle, ringing table and VoiceMail, or disable the selected line. Remember that a call is answered immediately when routed to the VoiceMail. **Also keep in mind that if an SNS is routed to the bundle, the ringing table and ring group object rows are not applied!**
- **Destination Id** – use this column to set an object of the selected type.
- **Disable logout** – select this option to disable user or station logout from a bundle using the Logout from bundle service. Any logout attempt is notified as an error.

Advanced settings

- **Send CLIP** – this is a table facilitating incoming call identification. Having passed the table, the call Id is modified as required. The **Send as** parameter helps you set two identification display modes. Select **Display** to display the **Number/URI** as the CLIP on the telephone, but the original CLIP will be stored in the CPN history. Select **Force** to modify both the CLIP displayed during ringing and the CLIP stored in the CPN history. Use **Scheme** to select either number or URI and **Type** to set the number subtype (Unknown, Internal, Local, National or International).
- **Force Facility** – refers to the called number. It is used in DSS1 messages for communication with Ericsson exchanges for billing purposes. Again, set the **Scheme** (Number or URI), **Subtype** (Unknown, Internal, Local, National, International) and **Number/URI** (specific number or address).
- **Force Redirecting** – refers to the called number. It is used in DSS1 messages for communication with Nokia exchanges for billing purposes. Again, set the **Scheme** (Number or URI), **Subtype** (Unknown, Internal, Local, National, International) and **Number/URI** (specific number or address).
- **Assistant** – in this section set up a bundle with respect to the Assistant user application.
  - **Visible in Assistant** – use this option to display a bundle within the application. If it is not checked, the bundle is not available for use.
  - **Group** – use this option to select a group or subgroup of users who are allowed to work with a selected bundle within the application. If the selected group (or subgroup) contains subgroups, the bundle is available only to the users who are assigned directly to the group (or subgroup) to which the bundle is assigned.
- **Accounting group** – use this option to enable adding the selected group number to the accounting sentence for a selected object for later cost distribution purposes.

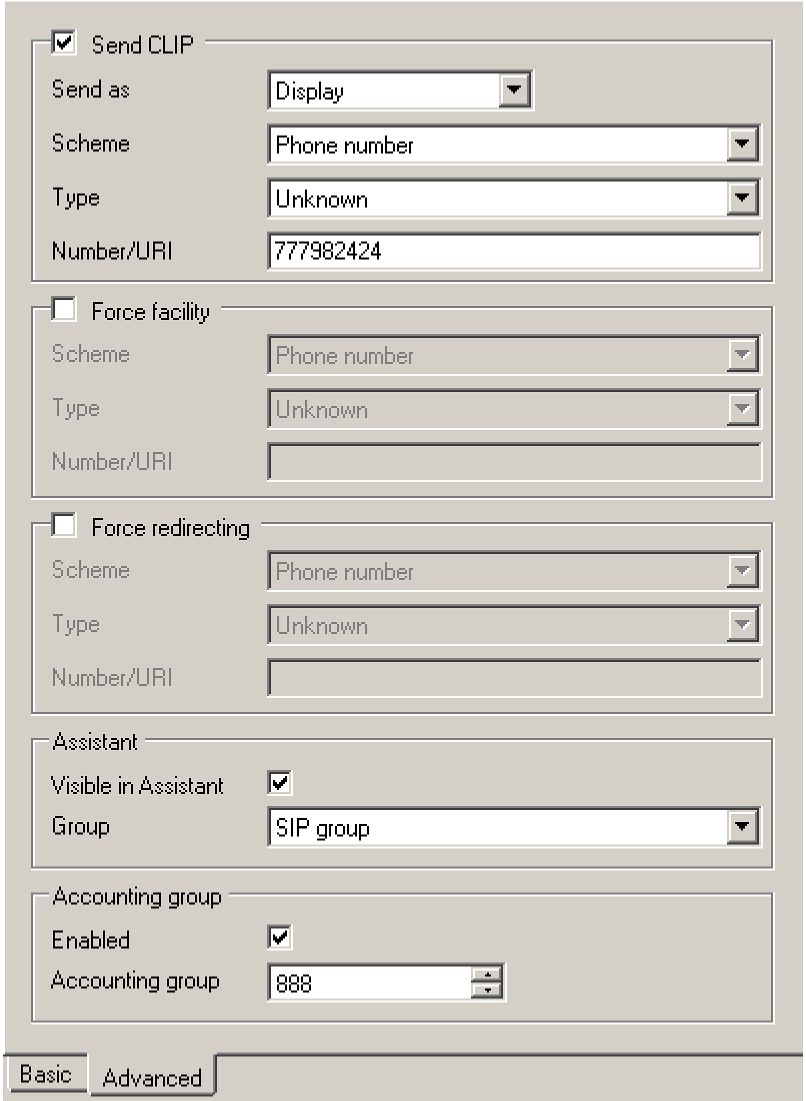


Figure: View of Bundle Configuration Menu – Advanced

Service Login to Bundle

The **Station/User Login to bundle** services have been enhanced with the option to specify the bundle position to which the station/user will be assigned. If a '0' is selected for the bundle position or no selection has been made, the station/user is placed last in the bundle (as before). Selecting a '1' means the first position, '2' means the second, '3' the third, and so on. Refer to the example below for illustration.

Example

Suppose you want to log in a station to the third position of bundle 151. Dial the service access number *64 from the station and enter the four–digit user PIN (1111, e.g.) when requested so. Now you will be asked to dial the bundle number. Dial 151 and press * for confirmation. Then dial the required bundle position for your station, i.e. 3, and press **#** for dialling end. The service has been completed successfully and your station is logged in as the third in the bundle.

Refer to the User Manual for details on the Login to bundle service.

DISA

DISA

The DISA (Direct Inward System Access) routing object is used for automatic call acceptance by the PBX with a subsequent DTMF transfer option and playing of the selected tone. In conjunction with suitable routers, you can create the IVR structure. This routing object is particularly suitable for GSM and CO virtual ports where you have to answer incoming calls 'Manually' to give the calling user an opportunity to influence routing (these virtual ports do not support the dial-in option).

DISA Setting

To configure the DISA routing object use the **Routing – Routing objects – DISA/IVR objects** menu. A list of available DISA objects is displayed on the left. Add, delete or rename the DISA objects using the context menu. Moreover, the following three options are available:

- **Default** – use this option to add three basic DISA modes – DISA_DEN, DISA_NOC, DISA_ME.
- **Update** – use this option to update the existing default DISA services.

Once a DISA object is selected, the configuration options get displayed to the right. **Strategy** is the first parameter to be configured, determining the behaviour of the entire DISA routing object. Choose **Immediate** or **Alerting**. Different DISA object parameters can be configured according to the strategy selected.

Immediate strategy

This strategy represents a common DISA concept. When a call comes to the port, it is answered, the DTMF detector is connected and the selected progress tone is played to the calling user. The numbers to be dialled are searched in the preset router. If no digit is detected before the timeout expiry, the call will be routed to the selected default destination. The DTMF detector is active only in the period between the call answer and the timeout end.

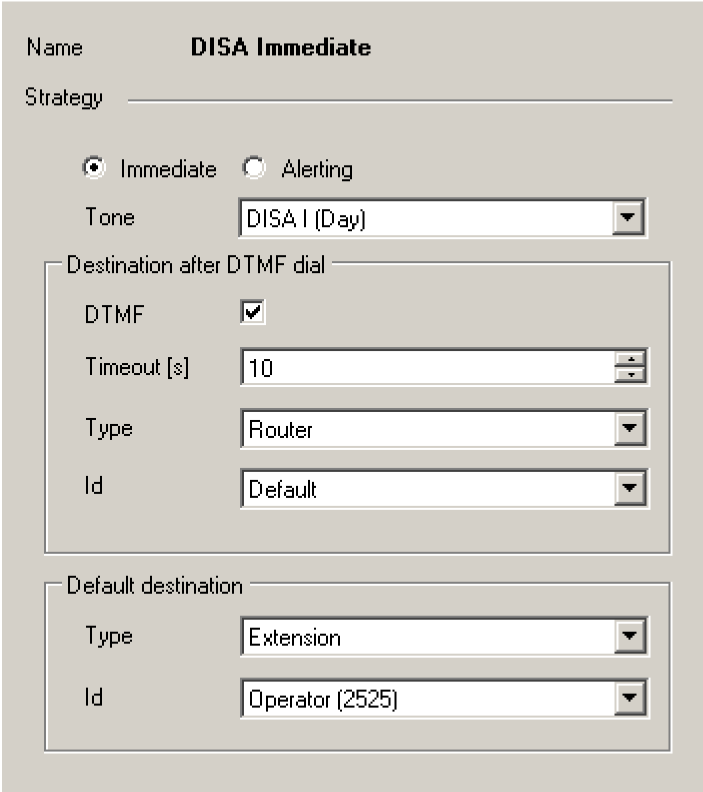


Figure: Příklad konfigurace objektu DISA se strategií Ihned

The menu consists of the following parameters:

- **Tone** – use this option to choose a suitable progress tone from the list. Add progress tones and messages of your own in the menu in Chapter 6.7, Progress tones, if desired.
- **Destination after DTMF dial** – use this option to set a router to be used for call routing by the PBX upon DTMF dialling.
  - **DTMF** – use this option to set whether the DTMF detector should be allocated for the DISA. The count of DTMF detectors is determined by the available hardware profile. If you do not select this option, the following three parameters are not available for configuration.
  - **Timeout [s]** – set how long the DISA object should wait for DTMF. If you select **0**, the whole voice message will be played back and the call will be routed to the default destination. Therefore, never choose the **endless voice message** in the **Tone** parameter in this case.
  - **Type** – set the router type for call routing.
  - **Id** – choose a router of the selected type.
- **Default Destination** – set an object for call routing if no DTMF dialling is detected within the timeout.
  - **Type** – set the destination type for call routing.
  - **Id** – choose a destination of the selected type.

Alerting strategy

This strategy represents a new DISA concept. When a call comes to the port, it is immediately routed to the preset Alerting destination and this destination is being alerted till the end of timeout. The timeout is set by the **Timeout** parameter. The call is not answered during the timeout and the calling user hears the alert tone from the network. After the timeout, the call is answered, but only towards the calling user, who is played the predefined progress tone. The Alerting destination is still being alerted. If the DTMF option is checked, the DTMF detector is connected after the timeout and remains active until the end of routing (answer, reject, etc.). The DTMF digits are transferred into the router selected in the **Destination after DTMF dial** option.

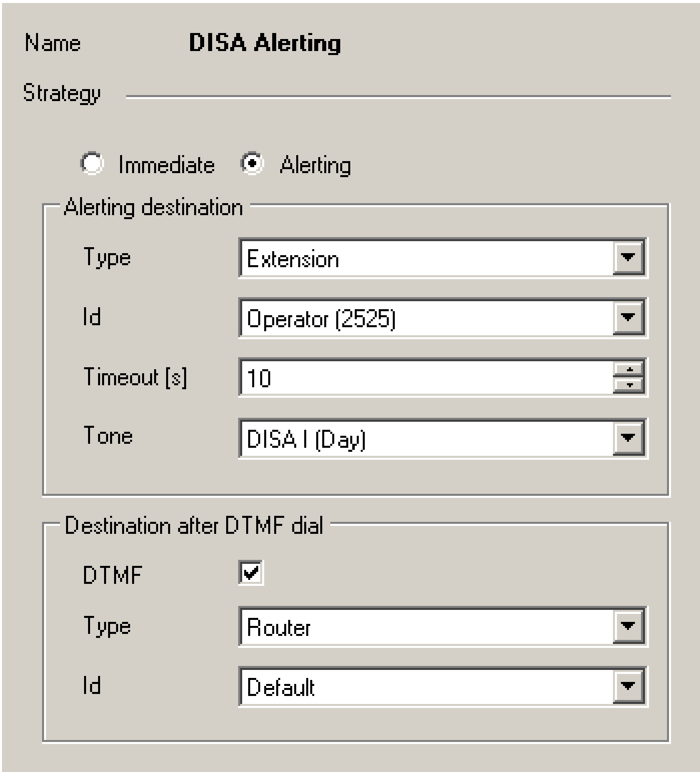


Figure: Příklad konfigurace objektu DISA se strategií Vyzváněcí

The menu consists of the following parameters:

- **Alerting destinace** – set the alerting destination parameters.
  - **Type** – set the destination type for call routing.
  - **Id** – choose a destination of the selected type.
  - **Timeout [s]** – set the timeout for playing the predefined voice message. The '0' selection is not suitable for the **Alerting** strategy since the call would be answered immediately, which is undesirable in this strategy.
  - **Tone** – select a tone from the current list of PBX progress tones.
- **Destination after DTMF dial** – set a router for call routing by the PBX after DTMF dialling.
  - **DTMF** – define whether or not the DTMF detector shall be allocated for this DISA object.
  - **Type** – set the router type for call routing.

- **Id** – choose a router of the selected type.

Ring Groups

Ring Group

The ring group is a routing object that is used for routing an incoming call or SMS message to more destinations at the same time. When the call is answered, the other destinations stop ringing and display the Missed call message. For more information refer to the Global Parameters menu, the **Unselected as missed** item.

The ring groups are also used as user groups for taking over calls. The users who miss their calls due to absence may use the **Take over from group** and **Take over from my group** selections. For these purposes, the ring group has to contain extensions only!

Ring Group Setting

To set the ring groups use the **Routing – Routing objects – Ring groups** menu. A list of available ring groups is displayed on the left. Add, delete or rename the ring groups using the context menu. Moreover, the following options are available:

- **Default** – use this option to add default ring groups. Rings groups are added to any group or subgroup that contains a user.
- **Update** – use this option to update the currently selected default ring group.
- **Update all** – use this option to update the contents of all default ring groups at once.

If you choose one of the ring groups, you can set its parameters on the right–hand side of the menu.

The menu consists of the following parameters:

- **Number** – is used as a ring group identification for taking over calls. If this number is not filled in, the ring group cannot be used for the **Take over from group** and **Take over from my group** services. Use the **Default alert tones** to select specific alert tones for specific situations.
- **Default alert tones**– in this section you can set up different alert tones for specific situations.
  - **Normal** – set the alert tone that is used in all situations except for the two cases mentioned below.
  - **Queued** – set the alert tone to be used for routing to a extension with an active queue. The incoming call has to be queued, otherwise the normal alert tone is used.
  - **No–port extension** – set the alert tone for routing to a no–port extension user (external extension). An external extension and one internal extension (active) at least have to be assigned to the user. Otherwise you will hear the alert tone.

In addition to the above–mentioned parameters, an object adding table is available. The table consists of two columns with the following meanings:

- **Destination type** – in this column select the type of the routing object to be used for incoming call routing. Define the extension, user, virtual port, set, ring group, bundle, ringing table and such objects as DISA, VoiceMail and service, or disable the selected line. Remember that a call is answered immediately when routed to the DISA (Immediate), VoiceMail and service and thus it makes no sense to add other objects to the ring group!

- **Destination** – use this column to select an object of the selected type.

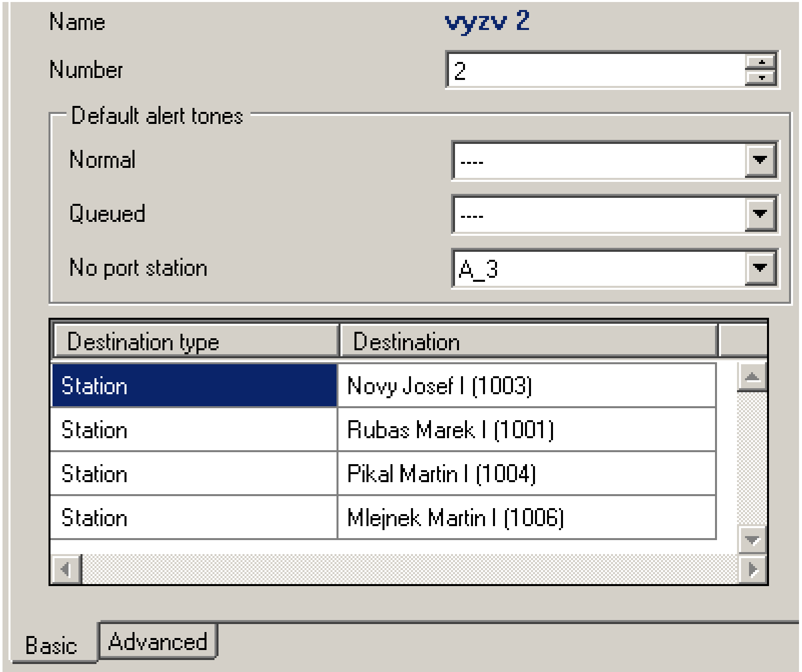


Figure: View of Ring Group Configuration Menu – Basic

Advanced settings

- **Send CLIP** – this is a table facilitating incoming call identification. Having passed the table, the call Id is modified as required. The **Send as** parameter helps you set two identification display modes. Select **Display** to display the **Number/URI** as the CLIP on the telephone, but the original CLIP will be stored in the CPN history. Select **Force** to modify both the CLIP displayed during ringing and the CLIP stored in the CPN history. Use **Scheme** to select either number or URI and **Type** to set the number subtype (Unknown, Internal, Local, National or International).
- **Force facility** – refers to the called number. It is used in DSS1 messages for communication with Ericsson exchanges for billing purposes. Again, set the **Scheme** (Number or URI), **Subtype** (Unknown, Internal, Local, National, International) and **Number/URI** (specific number or address).
- **Force forwarding** – refers to the called number. It is used in DSS1 messages for communication with Nokia exchanges for billing purposes. Again, set the **Scheme** (Number or URI), **Subtype** (Unknown, Internal, Local, National, International) and **Number/URI** (specific number or address).
- **Assistant** – use this section to set the selected ring group with respect to the Assistant user application.
  - **Visible in Assistant** – use this option to display a ring group within the application. If it is not checked, the ring group is not available for use.
  - **Group** – use this option to set a group of users to be able to work with the selected ring group within the application. Assign ring groups to so–called root groups only (never to subgroups). The assigned ring group is available to all the root group and subgroup users.
- **Accounting group** – use this option to enable adding the required group number into the accouting sentence for a selected object for later cost distribution purposes.

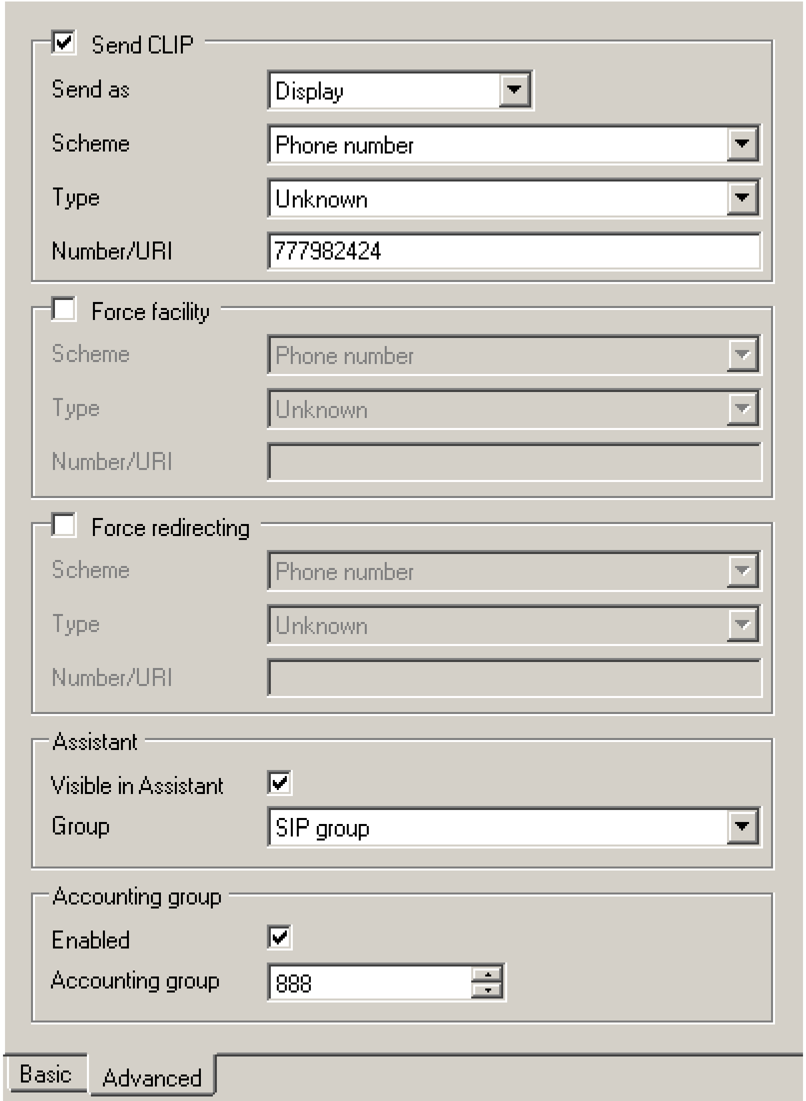


Figure: View of Ring Group Configuration Menu – Advanced

Ring Tables

Ring Table

The ring table is a routing object used for sequential routing of incoming calls to multiple objects, thus combining the advantages of a bundle and a ring group. The incoming call routing obeys predefined rules, which are always searched from the beginning. If an incoming call is answered by the destination to which it has been routed, the ring table routing process is terminated.

Ring Table Setting

To set a ring table use the **Routing – Routing objects – Ring tables** menu. A list of available ring tables is displayed on the left. Add, delete or rename the ring tables using the context menu. The configuration menu of a selected ring table is displayed on the right, providing the following parameters:

- **Default alert tones**– use this section to define specific alert tones for specific situations.
    - **Normal** – here set the alert tone to be used in all situations except for the two cases mentioned below.
    - **Queued**– here set the alert tone to be used for extension routing if one of the following commands is used.
        - **Route with queue** – the selected alert tone is used regardless of the **Queue** setting at the final destination.
        - **Route** – the selected alert tone is used only if the **Queue** parameter is enabled at the final destination.
    - **No–port extension** – sets the alert tone to be used for routing to a no–port extension user (external extension). Such user has to be assigned an external extension and one internal (active) extension at least. Otherwise, you will hear the alert tone.

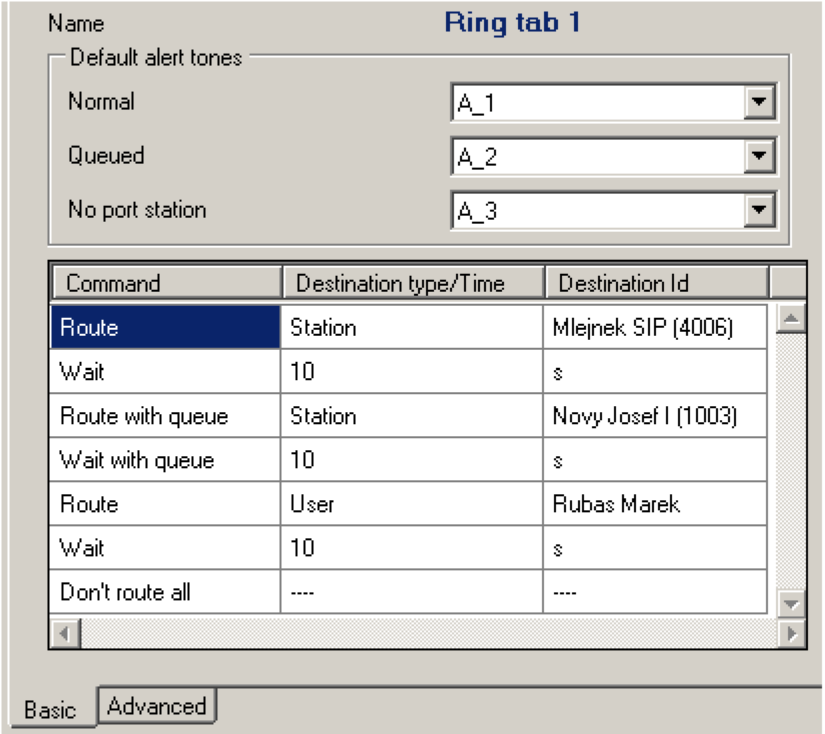


Figure: View of Ring Table Configuration Menu – Basic

The most important part of the ring table setup is the table located in the bottom part of the menu. Use this table to define the call routing rules. For this purpose, you can combine a few commands, which can be divided into three logical groups according to function.

- **Routing** – these commands determine the object to which an incoming call will be routed.
  - **Route** – this command routes an incoming call to the object defined in the remaining table columns. First select an object type and then an object of the selected type. Choose the extension, user, carrier, set, ring group, bundle, another ring table, AutoClip router and also such objects as DISA, VoiceMail and service. Remember that a call is answered immediately when routed to the DISA (Immediate), VoiceMail and service and it makes no sense to add other objects to the ring group!
  - **Route with queue** – this command routes an incoming call to the object defined in the remaining table columns. If the object is busy, the incoming call is queued regardless of the **Queue** setting for the object.
- **End of routing** – these commands terminate call routing to the object to which the call was routed using the **Route** or **Route with queue** commands.
  - **Do not route** – use this command to terminate routing to an object. Be sure to terminate call routing only to the object to which the call has been routed to by the ring table. For example, you cannot terminate routing to a user extension if you have routed the call to a user.
  - **Do not route all** – use this command to terminate all active routing settings in the ring table.
- **Waiting** – these commands are used for setting the time intervals between the routing commands and routing termination commands. To specify the time period, use the second column of the table.

- **Wait** – use this command to set the timeout for proceeding to the next row of the table. The timeout is not applied if the previous command has routed the incoming call to a busy destination and the call has been rejected or queued. In this case, the routing proceeds immediately to the next row. If 0 is used, the PBX waits for an indefinite period of time and the next row is only used in the event of busy destination or call rejection.
- **Wait always** – use this command to set the timeout for proceeding to the next row of the table. The incoming call is not routed to the next row before the timeout expiry. If 0 is used, the PBX does not wait and immediately proceeds to the next row (such row has no sense).
- **Wait with queue** – use this command to set the timeout for proceeding to the next row of the table. The timeout is not used if the previous command has routed the incoming call to a busy destination and the call has been rejected (not queued). In this case, the routing proceeds immediately to the next row. If, however, the call has been queued, the routing waits for a preset timeout or busy object answer (whatever comes first) before proceeding to the next row. If '0' is used, the PBX waits for an indefinite period of time and the next row is only used in the event of busy destination or call rejection.

The last command cannot be included in any of the above–mentioned groups.

- **None** – has the same function as an empty row (= no function).

Advanced settings

- **Send CLIP** – this is a table facilitating incoming call identification. Having passed the table, the call Id is modified as required. The **Send as** parameter helps you set two identification display modes. Select **Display** to display the **Number/URI** as the CLIP on the telephone, but the original CLIP will be stored in the CPN history. Select **Force** to modify both the CLIP displayed during ringing and the CLIP stored in the CPN history. Use **Scheme** to select either number or **URI** and **Type** to set the number subtype (Unknown, Internal, Local, National or International).
- **Force facility** – refers to the called number. It is used in DSS1 messages for communication with Ericsson exchanges for billing purposes. Again, set the **Scheme** (Number or URI), **Subtype** (Unknown, Internal, Local, National, International) and **Number/URI** (specific number or address).
- **Force forwarding** – refers to the called number. It is used in DSS1 messages for communication with Nokia exchanges for billing purposes. Again, set the **Scheme** (Number or URI), **Subtype** (Unknown, Internal, Local, National, International) and **Number/URI** (specific number or address).
- **Accounting group** – use this option to enable adding the required group number into the accounting sentence for a selected object for later cost distribution purposes.

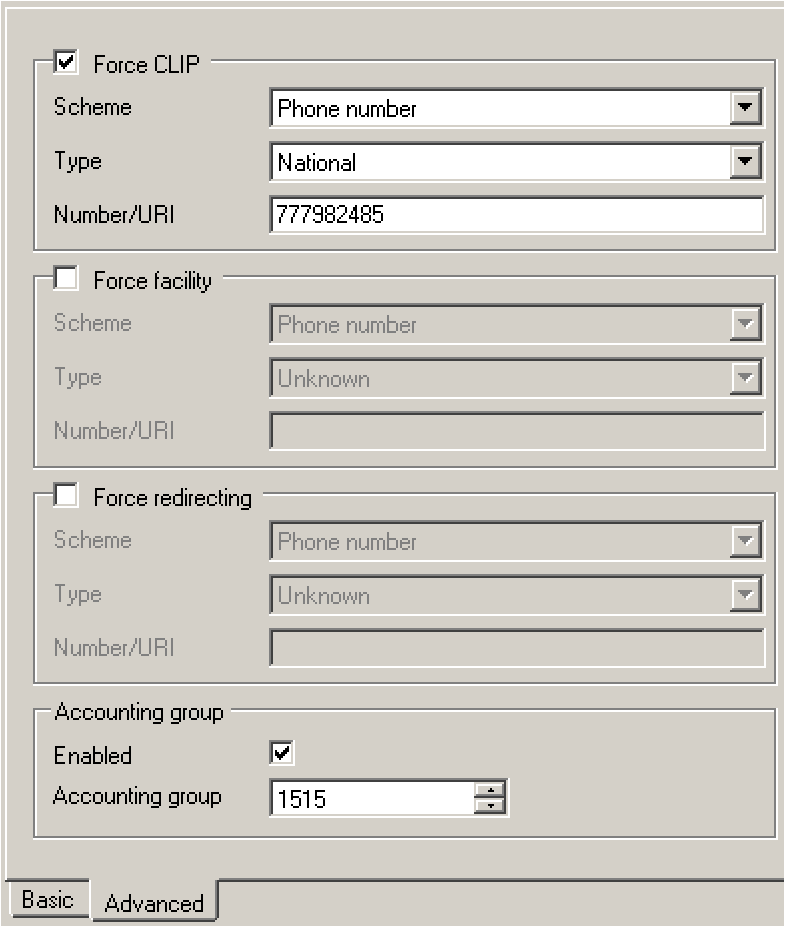


Figure: View of Ring Table Configuration Menu – Advanced

Modems

Modem Connection

Modem connection is used for remote PBX access where no TCP/IP connection is available. A modem also provides remote access to the database and enables to receive current system traces via the TraceView application. Modem access, however, is considerably limited by a low data rate and thus is not recommended for the Localisation where the TCP/IP access can be used. The current NetStar PBX firmware version supports the **ISDN modem with protocol X.75**. The figure below shows an example of modem configuration for remote access to the PBX.

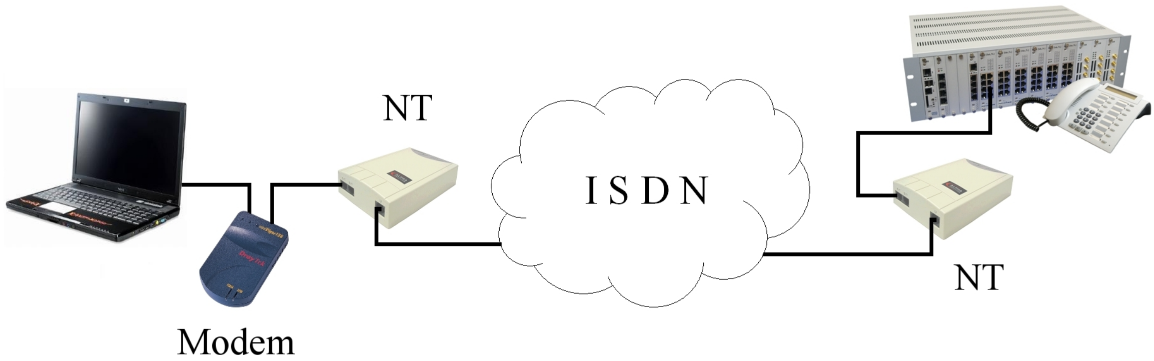


Figure: View of PBX Remote Access Modem Configuration

Connection Setting

To enable modem connection, select the required modem from the list of devices (if unavailable in the list, the modem is probably not installed in the PC) in the Properties and select the number to be dialled for PBX connection. This number must then be routed to the routing object created in the **Routing - Routing objects – Modems** me nu.

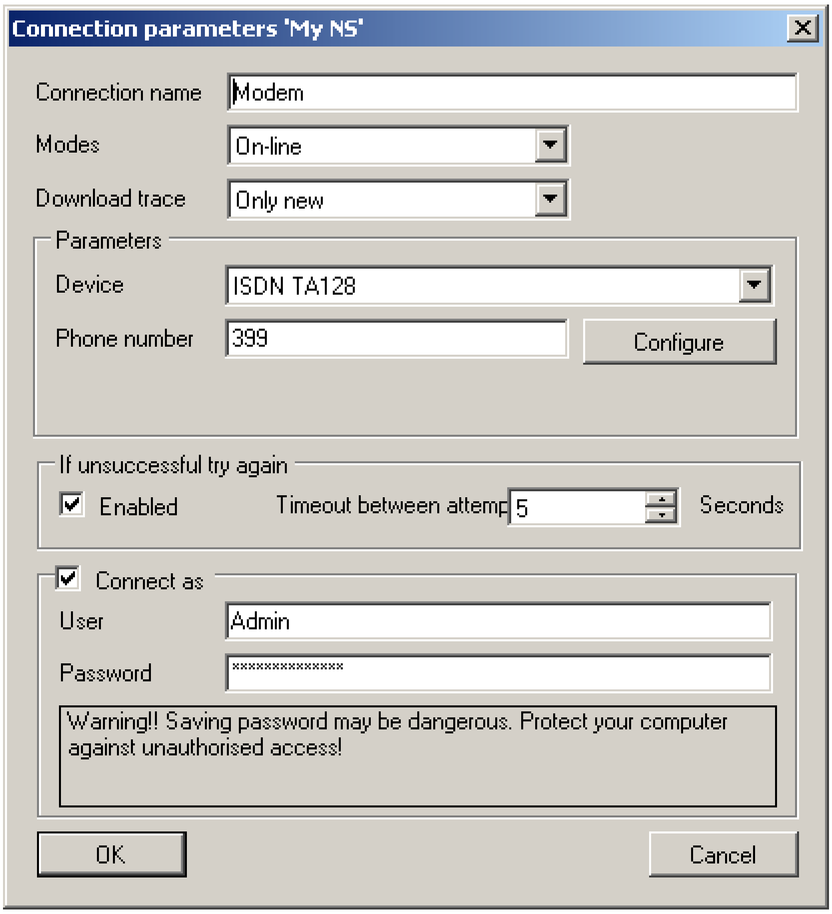


Figure: Nastavení parametrů připojení pro vzdálený dohled prostřednictvím ISDN modemu

Modem Setting

- **Trace send enabled** - use this option to enable trace sending for the TraceView application via a modem. If this option is not checked, the application is connected but no system information is sent to the remote user. In this mode you can view the database only.
- **Peer authorisation required** - use this option to enable a login dialogue request for modem connection. If this option is not checked, the connection is established without requiring the login and password settings. This option is used for connections via the NSAdmin configuration tool and the TraceView application.

Sets

Set

The set is a routing object that is used for an easy object sequencing. For example, sequencing of routes with the aid of default destinations is not flexible enough, being obligatory for all incoming calls. Connecting into various parts of the string may be very tying. Sets enable you to create different sequences for different situations as necessary. In addition to routers, you can add AutoClip routers, ring groups, bundles, ring tables and other sets to the sets. Furthermore, you can add extensions, users, virtual ports, modems, DISA functions and services. Remember to include the DISA and services at the end of the structure to avoid any premature chain termination.

Having joined a set, a call is always automatically routed to the first object. To route the call back to the original set, set a row or default destination in the **Default** option. This option is used as a signal for return to the set and proceeding to the next object (row) of the set. That is why routing to an extension mostly terminates the process. You cannot set the **Default** destination in the event of unsuccessful routing and the call has no opportunity to return to the set.

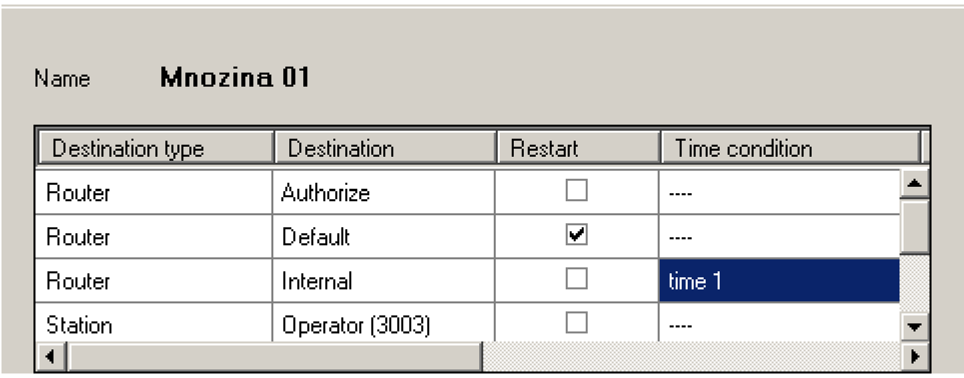


Figure: View Set Configuration Menu

Set Setting

To set up a set use the **Routing – Routing objects – Sets** menu. A list of available sets is displayed on the left. Add, delete and rename the sets using the context menu. The configuration of the selected set is displayed on the right–hand side of the menu. Unlike the other routing objects, the sets have no configuration parameters in the menu. The menu contains only an adding table for the objects to which incoming calls are to be routed. The table consists of four columns with the following meanings:

- **Destination type** – use this column to select the type of object to be used for incoming call routing. Choose the routers, AutoClip routers, ring groups, bundles, ring tables, other sets, extensions, users, carriers, modems, DISAs and services. Remember that the process will be terminated when you select an object that has no opportunity to return to the set. You are recommended to add such objects to the end of the structure. Use the **Default** option to return to the higher level set (if you are using a set in a set).
- **Destination** – use this column to select an object of the selected type.
- **Restart** – this option relates to the called party number (CPN). If this number

has been changed since it arrived in the PBX and there is a **True** setting somewhere in the set, then the original, unchanged number is being searched for in the routers from this object on. Again, if the CPN is changed again in or behind the object and the **False** parameter is set for the subsequent objects somewhere in the set, the call is routed according to this changed number until an object with the **True** selection is found.

- **Time condition** – use the time conditions to change a set in time. You can define a different time condition for each row. The rows are then valid in the time of the preset time condition validity only.

Audio Inputs and Outputs

What is Audio I/O?

The Audio I/O ports are routing objects that cooperate with the audio ports of the Audio/IO/Relay board. Sounds enter the PBX or are played back through these inputs. The inputs can be used as a source of external progress tones and the outputs as a broadcast, for example.

Audio Ports

The Audio/IO/Relay board can be equipped with two or four stereo jack ports with the diameter of 3.5 mm, which represents four or eight ports (each stereo port is used as two mono ports). No port can be used as an input and output at the same time. Define the attenuation value for each port ranging from –70 to 70 dB.

The audio ports cannot be used as standard PBX virtual ports and should be assigned to a specific routing object of the Audio I/O type. Each audio port can be assigned to multiple Audio I/O routing objects.

Audio I/O Setting

The menu for the Audio I/O routing objects is divided into two parts. You can add, delete or rename the routing objects on the left and define a selected object on the right using the following parameters.

- **Name** – displays the name of a selected routing object only. It cannot be directly configured here.
- **Audio I/O** – assigns a selected Audio/IO/Relay board source to an object.
- **Cancelled by incoming call** – is a radio function option (has not been implemented yet). If a call comes during radio playing, the radio function is terminated and the terminal starts ringing. If this option is not checked, the incoming call is rejected with the user busy cause (or queued).
- **Turn on tone to caller** – enables playing of the below–defined tone to the calling user. The calling user should always hear the tone. It means that you cannot play a tone to the assigned source only but you can disable tone playing for both directions.
- **Turn on tone to Audio I/O** – enables playing of the below–defined tone to a selected Audio/IO/Relay board source.
- **Tone** – defines the tone to be played to the calling party or/and to the selected Audio/IO/Relay source.
- **Turn off tone after a time limit** – enables the tone time limit. If the time parameter is switched off, the whole tone is always played back. After the tone has been played, the calling user is connected to the assigned source for broadcasting.
- **Tone time [ms]** – use this parameter to define the tone playing time. It may be shorter or longer. After this period, the calling user is connected to the assigned source for broadcasting.

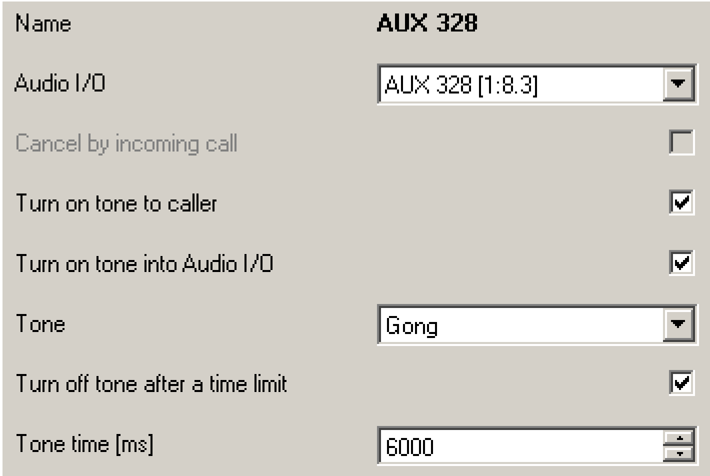


Figure: View of Audio I/O Configuration Menu

Example 1 – Broadcast

To use the audio port for broadcasting set the selected port onto **Output** in the Boards menu and then assign it to the selected Audio I/O routing object. The broadcast function is activated by an incoming call to this routing object. To play an announcement (e.g. We are beginning ... 5, 4, 3, 2, 1, on air...), select the message in the **Tone** parameter. When a call comes to the routing object, the selected message is played in the selected direction (calling or both) first and then the calling user is connected to the assigned source for broadcasting.

Example 2 – External Music on Hold source

To use an Audio/IO/Relay board source as an external Music on Hold (or other progress tone) source, configure it as **Input**. You do not even need the Audio I/O routing object for this purpose. Add a new input to the **Audio inputs** tag in left–hand upper part of the **Global data – Progress tones** menu and then assign Localisation and the Audio/IO/Relay board port to it in the right–hand upper part of the menu. Subsequently, use the **Progress list** tag to select this Audio input as a Music on Hold input. Now assign the progress tone to a selected user group in the **Properties – Basic** tag in the **Hold tone** parameter. The selected group of users will thus get music played to the port from an mp3 player or another source instead of the Music on Hold.

Binary Inputs and Outputs

What Is Binary I/O?

The Binary I/O ports are routing objects cooperating with the binary ports on the Audio/IO/Relay board. Each port consists of a relay and a detector. Thus, the ports can be used both for relay switching and relay state detection. The port has only a weak current source and is not intended for switching door locks and similar equipment. If completed with an appropriate external source, however, the port can be used for this purpose too.

Binary Ports

The Audio/IO/Relay board can have four or eight binary ports. Each of them can be used in the Output (switch), Input (detector) or bi-directional mode (switch and detector). The function of each port also depends on the jumper hardware setting. For available modes refer to the Hardware Manual, Chapter Audio Inputs and Outputs. The setting contains a non-programmable **Pulse filtering [ms]** parameter, which is set to a fixed value of 100ms. This means that changes on the input are detected every 100ms even if they came more frequently. The parameter helps protect the PBX against overload.

The binary ports cannot be used as standard PBX virtual ports and should be assigned to a specific routing object of the Binary I/O type. Each binary port can be assigned to multiple Binary I/O routing objects.

Binary I/O Routing Object

The Binary I/O routing object can be set as a switch or a detector. The following parameters are common for both the modes.

- **Name** - displays the name of the selected object only. It cannot be directly configured here.
- **Binary I/O** - assigns an Audio/IO/Relay board source to a selected object.
- **Direction** - defines whether the selected routing object controls the binary port switch or detector.

Switch Setup

- **Switch status** - this parameter displays the current status of the switch/relay (Active, Inactive, Unknown). With the **Unknown** option, the assigned binary port is probably configured as an input or the port or board is unavailable.
- **Do not pick up** - this parameter helps execute the actions specified below without the call being answered (the call remains in the alerting mode). Thanks to this, for example, the relay can be activated and the call can be routed by the PBX at the same time.
- **Tone** - sets the announcement to be played to the calling user whenever a call comes to this routing object.
- **Action at pick up** - this section defines the action to be executed upon pick up of a call or upon SMS coming to this routing object.
    - **None** - the relay does not respond.

- **Connect** - the relay is activated unless activated before.
    - **Disconnect** - the relay is deactivated unless deactivated before.
    - **Connect pulse** - the relay is activated for the time defined in the **Pulse width [ms]** parameter and then re-deactivated. If activated earlier, it is only deactivated at the end of the pulse.
    - **Disconnect pulse** - the relay is deactivated for the time defined in the **Pulse width [ms]** parameter and then re-activated. If deactivated earlier, it is only activated at the end of the pulse.
- **Action at hang up** - this section defines the action to be executed upon hang up of a call coming to this routing object.
    - **None** - the relay does not respond to the hang up.
    - **Connect** - the relay is activated unless activated before.
    - **Disconnect** - the relay is deactivated unless deactivated before.
    - **Connect pulse** - the relay is activated for the time defined in the **Pulse width [ms]** parameter and then re-deactivated. If activated earlier, it is only deactivated at the end of the pulse.
    - **Disconnect pulse** - the relay is deactivated for the time defined in the **Pulse width [ms]** parameter and then re-activated. If deactivated earlier, it is only activated at the end of the pulse.
- **Action after timeout/tone** - this section defines the action to be executed after the **Timeout** expiry or after **Playing whole tone**.
    - **None** - the routing object does not respond to the timeout expiry or the end of the played tone.
    - **Hang up** - after the timeout expiry or playing the whole tone, the call is hung up in the routing object with cause no. 16 - normal call clearing.
    - **Call destination** - after the time expiry or playing the whole tone, the call is routed as configured in the **Destination** option.
    - **Play whole tone** - use this option to enable playing of the whole tone independently of the preset time limit.
    - **Destination** - use this section to define the next routing destination after the time limit or playing the whole tone.
- **Connect by time conditions** - use this option to enable the relay activation according to the selected time conditions. The relay is activated whenever one time condition at least is valid. If this option is not checked, the preset time conditions are not used.

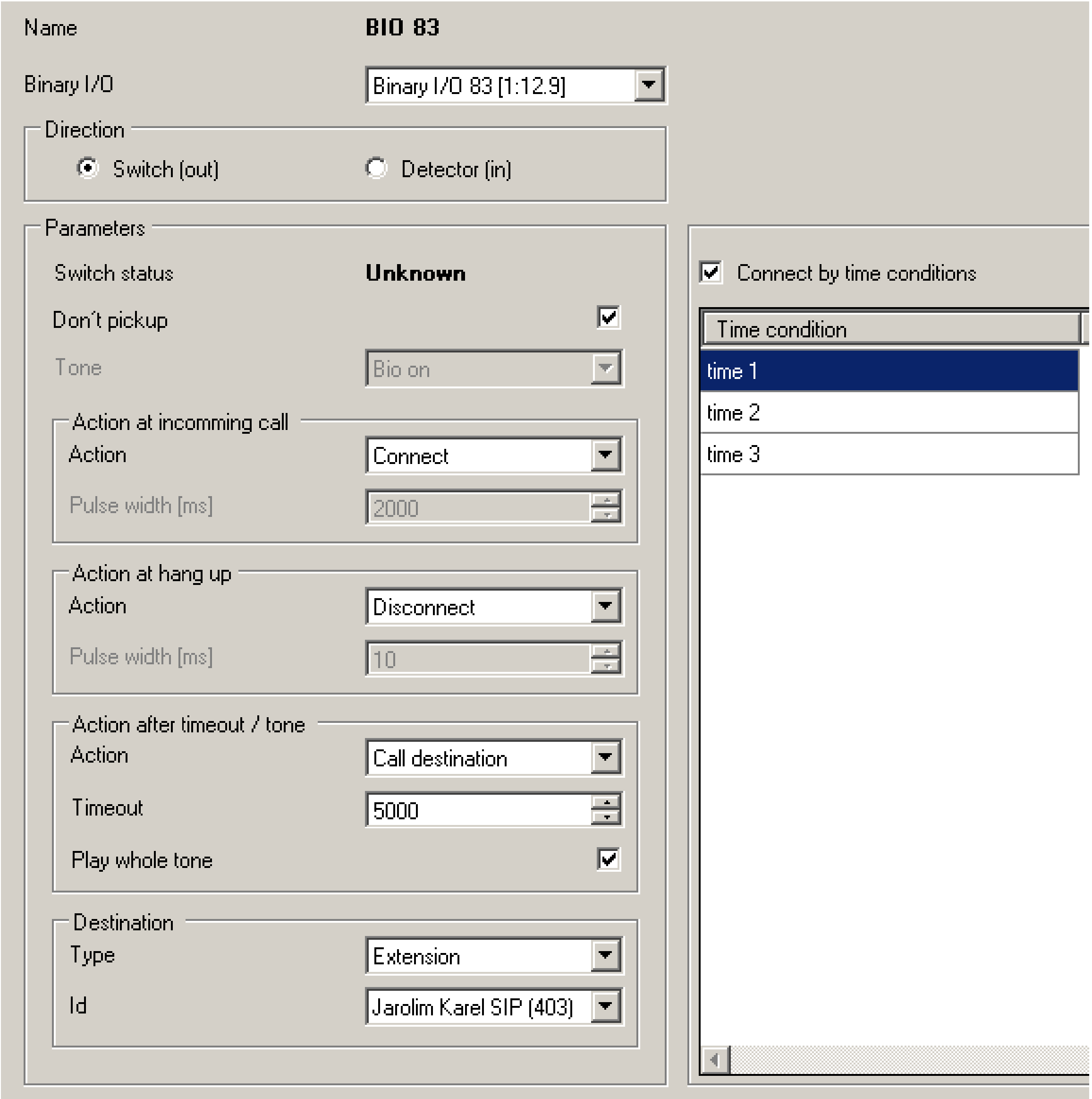


Figure: View of Binary I/O - Switch Configuration Menu

Example

Switch activation/deactivation by incoming SMS.

To activate the switch, route the incoming SMS using the text router to the particular binary object of the switch type where the **Action at pick up** parameter is set to **Connect**. The other actions are ignored. To deactivate the switch, route the SMS with a different text through the text router to a different binary object than that used for activation. This binary object, however, is assigned to one and the same binary source. But the **Action at pick up** parameter is set to **Disconnect** this time.

Detector Setup

- **Detector status** - this parameter displays the current status of the detector (Active, Inactive, Unknown). With the **Unknown** option, the assigned binary port is probably configured as an output or the port or board is unavailable
- **Tone connected** - sets the tone to be played to the calling user when the detector gets in the active state upon pick up. The playing mode depends on the **Timeout** and **Play whole tone** parameters.
- **Tone disconnected** - sets the tone to be played to the calling user whenever the detector gets in the inactive state upon pick up. The playing mode depends on the **Timeout** and **Play whole tone** parameters.
- **Tone events enabled** - sets the tone to be played to the calling user when the **Send events** is checked. The playing mode depends on the **Timeout** and **Play whole tone** parameters.
- **Timeout** - use this parameter to set the call duration. After the time limit, the call is hung up (unless the following option is checked).
- **Play whole tone** - use this option to enable playing the whole tone independently of the preset time limit.
- **Send events** - use this option to enable SMS sending for predefined events.
- **Send as user** - use this parameter to define a user as an SMS sender. Be sure to select the SMS routing parameters for the SMS sender.
- **Destination for events** - use this section to define the destination for sending the detector status messages. Select a user, extension or number. SMS messages to users and extensions are routed directly but those routed to an address use the From port of user routing as set in the Send as user parameter.

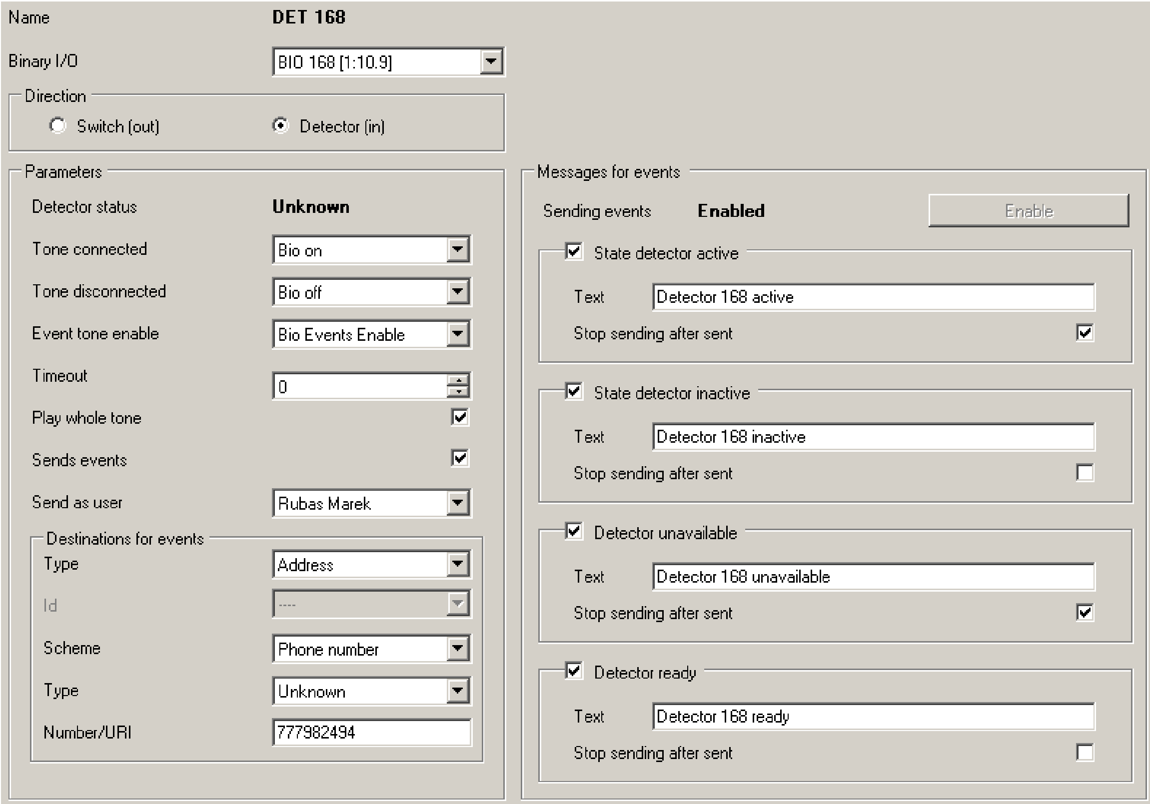


Figure: View of Binary I/O - Detector Configuration Menu

- **Messages for events**
  - **Sending events** - displays the current state of event sending. If such sending is enabled, the messages can be sent and the **Enable** button is inactive. If the sending is stopped, the messages are not sent and the **Enable** button is ready for use.
  - **Active detector state** - use this option to enable a message about the active state of the detector. Within this section you can define the message text to be sent. Optionally, you can stop sending after this message by selecting the **Stop sending when message was sent** option.
  - **Inactive detector state** - use this option to enable a message about the inactive state of the detector. Within this section you can also define the message text to be sent. Optionally, you can stop sending after this message by selecting the **Stop sending when message was sent** option.
  - **Detector unavailable** - use this option to enable a message about an unavailable detector. Within this section you can also define the message text to be sent. Optionally, you can stop sending after this message by selecting the **Stop sending when message was sent** option.
  - **Detector ready** - use this option to enable a message about a ready detector. Within this section you can also define the message text to be sent. Optionally, you can stop sending after this message by selecting the **Stop sending when message was sent** option.

CallBack

What Is CallBack?

CallBack is a function used for external PBX extensions. With the CallBack you can easily reduce costs of external extensions. The extension with the CallBack enabled only alerts the PBX or sends an SMS in the appropriate form and the PBX calls back to this extension. After answering the call, the external extension can dial through the PBX as in the case of a direct call. So you do not need an expensive fixed payment tariff for all external extensions but only for your PBX SIM cards. **This function is subject to licence.**

CallBack Setting

Find the CallBack configuration menu in the **Routing – Routing Objects** menu. The menu is divided into two parts. The left side is used for management and the right side for configuring the selected object. The context menu on the left consists of the following options:

- **Add** – use this option to add a new object for CallBack.
- **Delete** – use this option to delete a selected object.
- **Rename** – use this option to rename a selected object.
- **Default** – use this option to delete all current objects of this menu and create two default CallBack objects – one for calls and one for SMS messages.
- **Update** – use this option to add the default CallBack objects preserving the current ones. If default objects have already been created, their parameters are set to default values.

On the right–hand side of the menu you can find the following parameters:

Ring CallBack

- **Name** – only displays the name of the selected object.
- **CallBack delay [s]** – defines the delay between the CallBack recognition and execution.
- **Ring destination** – this destination is used in the case of successful CallBack immediately after call answering by the external extension.
- **Ring detection time [s]** – defines the ringing time for incoming calls from a extension with the CallBack. If this timeout expires (the calling user does not hang up), the call is routed to Destination after timeout and behaves as a normal call and the CallBack function is not used.
- **Destination after timeout** – this destination is used after the Ring detection timeout for another call routing. In the case of SMS CallBack, this destination is not used.

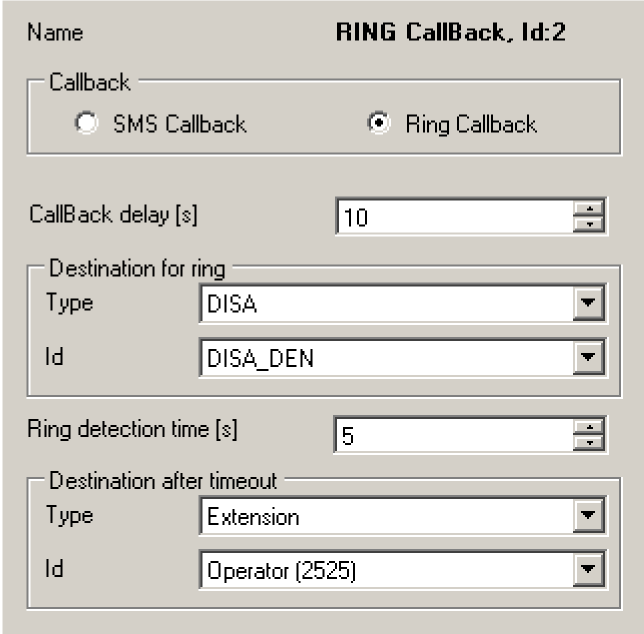


Figure: View of CallBack Configuration Menu for calls

SMS CallBack

- **Name** – only displays the name of the selected object.
- **CallBack delay [s]** – shows the delay between the CallBack requesting SMS reception and CallBack execution.
- **Delay in SMS content** – the delay data can be omitted in the SMS if set so here.
- **Alerting destination** – set the destination for routing the numbers included in the SMS.

SMS format

An incoming SMS for the CallBack function has to be routed into the text router for routing into the selected CallBack object. The SMS format depends on the **Delay in SMS content** setting. If this parameter is set to **Yes**, the SMS shall be as follows:

- **Called number, Delay, Calling number**

If the **Delay in SMS content** parameter is set to **No**, the SMS shall be as follows:

- **Called number, Calling number**

- **Called number** – this parameter is mandatory. Calls are routed according to the **Destination for ring** settings.
- **Delay** – this parameter is optional and has the same function as **CallBack delay** , yet a higher priority.
- **Calling number** – this parameter is optional and identifies the calling party if necessary. If absent, the SMS sender is used as the calling party.

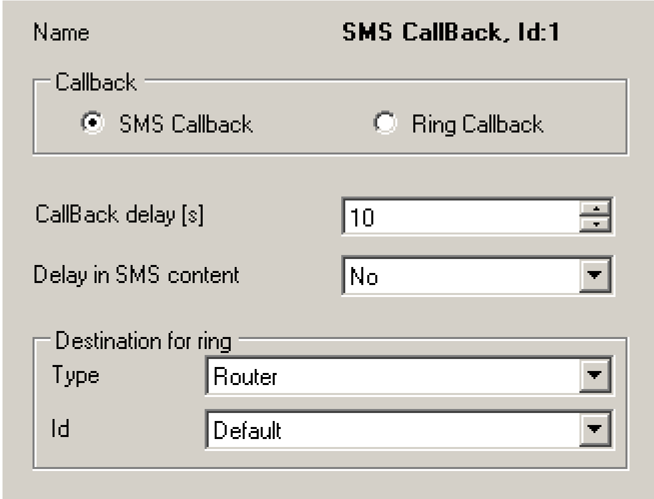


Figure: View of CallBack Configuration Menu for SMS

Example 1 – Initiated by call

The external extension with an enabled and licensed CallBack function dials a PBX SIM card number. The call is routed to the CallBack object. When hearing the alert tone, the calling user can wait for the end of the **Ring detection timeout**. In that case, the CallBack function is not activated and the call is automatically routed according to the **Destination after timeout**. When the calling user hangs up before the timeout expiry, the **CallBack** and **CallBack delay** are activated. After the **CallBack delay** expi ry, a CallBack to the external extension is established. The external extension user answers the call and can go on dialling through the PBX according to the **Destination for ring**.

Example 2 – Initiated by SMS

The external extension with an enabled and licensed CallBack function sends an SMS message to a PBX SIM card. The SMS is routed to the CallBack object. The SMS text may be **800123456,30**, for example. The PBX waits for a timeout (30s) and after that establishes a CallBack to the external extension. The external extension user answers the call and the PBX builds up a call to the called party 800123456 using the **Destination for ring** parameter.

Status Control Objects

A Status Control object is a routing object used for keeping the defined state (information) on the basis of received information. Received information here means the called number or text message. The state of the Status Control object is determined by the called number or received SMS. Use the **Routing – Routing objects – Status Control objects** menu to create the Status Control objects. The Status Control parameters are an inseparable part of the Status Control objects and help define their states. Refer to the **Global data – Status Control parameters** men u for the Status Control parameter settings.

You can also be informed of a change of the selected Status Control object state by the Event Reporter, which can notify transition to the Error or OK state if programmed so.

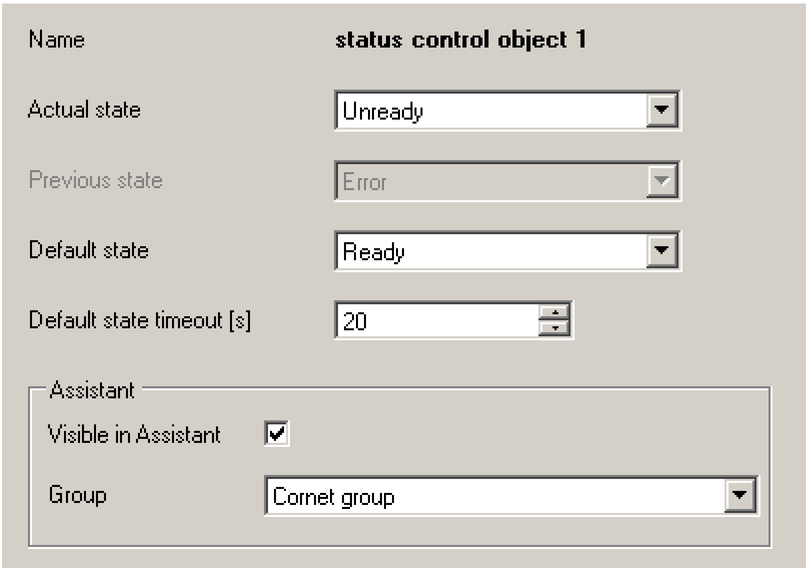


Figure: Pohled na menu pro nastavení routovacího objektu Status Control

- **Name** – this parameter gives the name of the selected routing object and cannot be configured.
- **Actual state** – informs about the current Status Control object state.
- **Previous state** – gives information on the previous Status Control object state. This parameter is used for information only and cannot be edited.
- **Default state** – this parameter defines the state to which the Status Control object passes after the timeout. This option is only available if the **Default state timeout** is non-zero.
- **Default state timeout** – here set the period of time after which the default state is set automatically. Select '0' to disable the automatic state change function.
- **Assistant** – use this block of parameters to set the bundle with respect to the Assistant user application.
    - **Visible in Assistant** – use this option to display a bundle within the application. If it is not checked, the bundle is not available for use.
    - **Group** – use this option to select a group or subgroup of users who are allowed to work with a selected bundle within the application. If the selected group (or subgroup) contains subgroups, the bundle is available

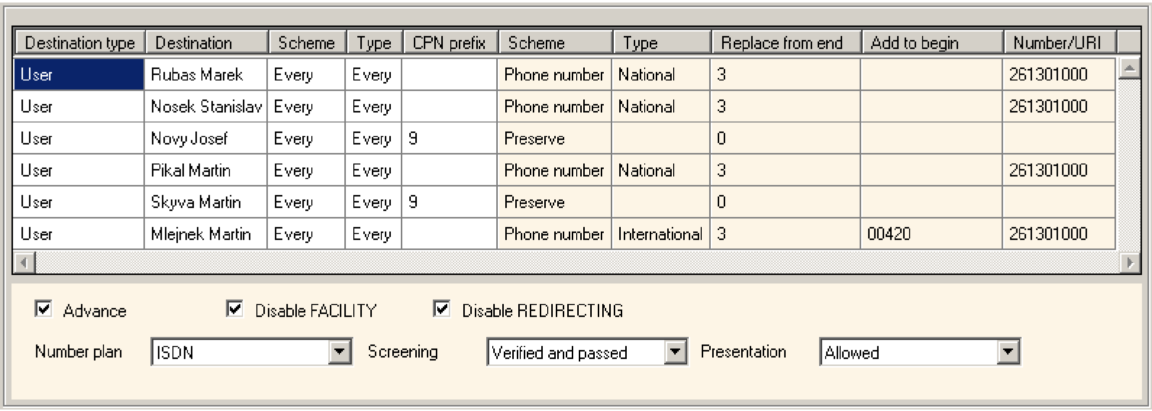
only to the users who are assigned directly to the group (or subgroup) to which the bundle is assigned.

7.6 Identification Tables

What Is Identification Table?

The identification tables are used for changing the calling extension numbers. To create and modify them use the **Routing – Identification tables** menu. To view an identification table, assign it to a virtual port or a virtual port type. The setup menu consists of two windows. A list of available identification tables is on the left. To configure a selected identification table, use the right–hand window. The context menu on the left side of the menu consists of the following options:

- **Add** – use this option to add an identification table.
- **Delete** – use this option to delete a selected identification table. If you delete an identification table, all the associate settings are removed from the database (e.g. assignment to a carrier).
- **Rename** – use this option to rename an existing identification table.
- **Default** – use this option to delete all current identification tables and create default identification tables. These tables have already been filled with corresponding objects (extensions, users, etc.).
- **Update** – use this option to update a selected identification table according to the selected type. Select the type via a dialogue box. Earlier identification table records are not deleted.
- **Update all** – use this option to add default identification tables and preserve all already existing ones.



Destination type	Destination	Scheme	Type	CPN prefix	Scheme	Type	Replace from end	Add to begin	Number/URI
User	Rubas Marek	Every	Every		Phone number	National	3		261301000
User	Nosek Stanislav	Every	Every		Phone number	National	3		261301000
User	Novy Josef	Every	Every	9	Preserve		0		
User	Pikal Martin	Every	Every		Phone number	National	3		261301000
User	Skyva Martin	Every	Every	9	Preserve		0		
User	Mlejnek Martin	Every	Every		Phone number	International	3	00420	261301000

Figure: View of Identification Table Configuration Menu

Example

Suppose extension 1234 assigned to user Rubas Marek is calling through the virtual port with the assigned identification table from Figure 1. To change the extension identification proceed as follows:

- First create number 261 301 000 of the National subtype.
- Then, change the last three digits according to the original extension number – 234 as shown in row 1 to get number 261 301 234.
- Finally, set the ISDN numbering plan, Screening and Presentation, which are transmitted via DSS1 signalling. Also, disable the Facility and Redirecting parameters independently of the previous routing.

Identification Table Setting

In the right–hand part of the menu, set the parameters of the identification table as selected on the left. The configuration window can be logically divided into four parts: **Calling party determination**, **New identification determination**, **Advanced settings** and **Default destination**. The table rows are arranged according to priorities. To change a row priority use the arrows on the right–hand side of the screen. To add rows with a certain priority use the **Insert ahead selected** and **Insert behind selected** options. The **Add** option is used for adding a record behind the currently last one (i.e. the lowest priority record).

Calling party determination

Calling party determination is performed at the beginning of each identification table row. Here define the object to which the below–selected identification rule will be assigned. For this purpose, use the following parameters:

- **Destination type** – in this column select a type of the calling party for the rule. Choose **Every**, **Extension**, **Extension type**, **User**, **Group**, **Virtual port** or **Virtual port type**.
- **Destination** – use this column to define a calling party of the selected type (e.g. a extension).
- **Scheme** – use this column to specify if the calling party identification should be presented as a **Number**, **URI** or non–specified (**Every**).
- **Subtype** – use this column to define the calling party number subtype before identification changing. Choose one of the **Unknown**, **Internal**, **Local**, **National** , **International** and **Every** options. Use **Every** if you are not sure which number subtype is used. Use **Subtype** only if the **Scheme** parameter is set to **Number**.
- **CPN prefix** – use this column to ensure that one and the same extension can identify itself differently depending on the called number.

New identification determination

New identification determination is executed in the second part of each row. For convenience, this part can has a yellow background. The identification rule sets a completely new calling party identification using five columns with the following meanings:

- **Scheme** – use this column to define whether the calling party shall identify itself by a number or URI or use the previous identification after passing through this row.
- **Subtype** – If the **Scheme** column is set to **Number**, choose **Unknown**, **Internal**, **Local**, **National** or **International** as the new CPN subtype.
- **Number** – set a number to be used for creating the new CPN identification within this row.
- **Replace from end** – use this parameter to define the count of new CLI digits (as set in **Number**) to be replaced by the original CLI digits. If the CLI is not transmitted, the resultant CLI is as set in the **Number** parameter.
- **Add to beginning** – use this parameter to add selected digits to the beginning of the new CLI.

Time condition

You can set a time condition in the last identification table column to define the validity time for each row. If the time condition is valid, the particular identification table row can be applied. If not, the row is ignored. This helps identify users and/or virtual ports differently for different parts of the day, week or month. You can assign the time conditions created in the Time Conditions menu.

Advanced settings

You can define advanced parameters for each identification table row – **Numbering plan**, **Screening** and **Presentation** – to be transmitted via DSS1 signalling. In addition, you can disable the advanced **Facility** and **Forwarding** functions, which are used in some networks.

- **Numbering plan** – use this column to set the used numbering plan for each table row.
- **Screening** – use this column to set the screening information for each table row.
- **Presentation** – use this column to set the CLI presentation restrictions for each table row.
- **Disable facility** – use this option to disable **Facility** for a selected identification table row.
- **Disable forwarding** – use this option to disable **Forwarding** for a selected identification table row.

Default identification

The lower menu is called **Default** and helps you set identification parameters for all the calling parties that have not been found in the table. The functions of these parameters are the same as those of the yellow–highlighted parameters as described earlier.

Consider forwarded user – use this option to enable the use of the identification table for a changed CLI in the event of call forwarding. Refer to the example below for more details.

Example

Suppose a call is coming to user B from user A. User B's calls are forwarded outside the PBX via the port to which the identification table with the **Consider forwarded user** p arameter enabled is assigned. The identification table has records for both user A and user B, and, in this case, the record assigned to user B is used. If the above mentioned parameter were disabled, the record assigned to user A would be used.

7.7 AutoClip Routers

AutoClip Router

The AutoClip routers are used for automatic routing of incoming calls and SMS messages in case a match is found in the assigned AutoClip router. Records are added to the AutoClip routers while outgoing calls or SMS messages are passing through the carriers to which the AutoClip routers are assigned. All you need to add a record on an SMS is to send it. A record on an outgoing call can be added only if the call has been rejected or unanswered by the called party. For easier comprehension and use, examples are provided at the end of this chapter.

AutoClip Router Use

To set the AutoClip routers use the **Routing – AutoClip Routers** menu. The menu is divided into two parts. A list of available AutoClip routers is displayed on the left. Add, delete or rename the AutoClip routers using the context menu. Moreover, there is an **Update** option, which enables you to add the default Autoclip router. Having selected a router, you can see its record listing (the last 100 records) and also set some of its parameters in the right–hand side of the menu.

You can assign the AutoClip routers virtual ports or virtual port types in the **Basic** tag. Assign the AutoClip router to calls and SMS messages separately. However, one and the same AutoClip router may be assigned in either case. Each AutoClip router record (row) has a flag, identifying an outgoing call or an SMS message. Each record is stored with a set of parameters. Some of the parameters depend on the call (CPN, CLI) and some on the AutoClip parameters assigned to the calling user or to the incoming virtual port. To set the above–mentioned AutoClip parameters use the **Global data – Autoclip parameters** menu. Remember to assign the parameters to calls and SMS separately (using the same parameter sets again). To assign the parameters to outgoing calls use the **Properties** option of the **Routing** tag on the user or group level. To assign the parameters to outgoing SMS messages use the Messages tag on the user or group level.

> **If these AutoClip parameter sets are not assigned to a user (or group), the user's records cannot be added to the AutoClip router!**

AutoClip Router Setting

You can set four parameters for a selected AutoClip router:

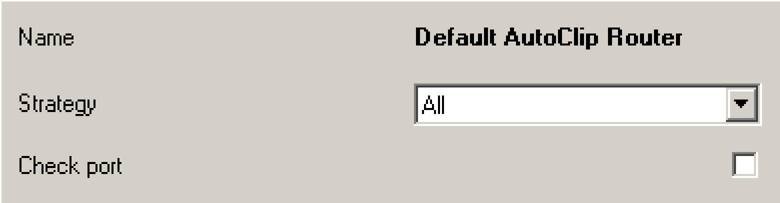


Figure: View of Identification Table Basic Settings

- **Strategy** – use this option to define the way of handling records from multiple users calling one number. This strategy refers both to record storing and subsequent record retrieving. Choose one of the following three strategies:
  - **All** – choose this option to save all records to the database. If an incoming call matches more AutoClip router records, all the matching users are alerted at the same time.
  - **Sequentially** – choose this option to alert all the matching users sequentially (starting from the latest record) if an incoming call matches more AutoClip router records. The next record is used for the next matching incoming call only if the previous record was marked as used and not deleted, the timeout validity was restarted or no action was made after the record use.
  - **Last one** – choose this option to add the latest record to the matching user (deleting the earlier ones) if calls are made or SMS sent by multiple users and routed through the carrier to which the Last one AutoClip router is assigned. If an incoming call matches more AutoClip router records (e.g. after a router strategy change), the user matching the newest record is only alerted.
- **Check port** – use this option to define whether or not to check the used ports. If this option is enabled, it is checked whether the incoming call came to the PBX through the same port as had been used for the outgoing call making the record. If not, the AutoClip router record will not be used.
- **Default destination** – in this field set the default destination to be used for incoming call routing in case no AutoClip router record match is found.
- **Destination for address** – use this section to define the destination to be used for incoming call routing in case there is no user but only the CLI information in the **Final destination** column.

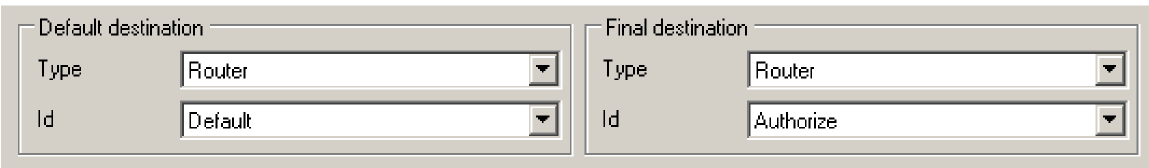


Figure: View of Identification Table Call Routing Settings

AutoClip Record Table

The AutoClip router table consists of eleven columns with the following meanings:

- **Validity** – this column displays the validity time for each record. Set the time limits in the AutoClip parameter set.

- **Last change with** – this column defines whether the record was created/changed by a call or message.
- **Scheme** – this column shows the CPN scheme for each record. Select **Number** or **URI**.
- **Number/URI** – this column shows the called party number (CPN). This number is necessary for finding a match with the calling subscriber. Therefore, make sure that the CPN is saved in the appropriate format. Always consider specific network properties and incoming normalising if applicable.
- **Time [mins]** – this parameter shows the validity time for each record.
- **Action after call use** – this parameter defines whether the record shall be kept valid or deleted after being used by a call.
- **Action after message use** – this parameter defines whether the record shall be kept valid or deleted after being used by a message.
- **Record is used** – this parameter defines whether the record shall be designated as used when it has passed through alerting, i.e. when the alerting message has been signalled, or when it has passed through active, i.e. when the call has been answered.
- **Virtual port** – this column shows the port used for routing of the outgoing call that created this AutoClip router record. It is used if the **Check port** option is selected.
- **Final destination** – this column shows the calling party that created this AutoClip router record. The name is displayed for a PBX user and the CLI is displayed for an external user. Any incoming call or SMS is then routed to such user.

Validity	Last change ...	Scheme	Number/URI	Time [mins]	Action after call use	Action after message use	Record is used	Virtual port	Final destination
13.10.2011 10:57:42	Call	Phone number	774123456	60	Delete record	Restart timeout	After active	GSM 94 [1:12.1]	User 'Rubas Marek'
13.10.2011 11:00:40	Message	Phone number	723001002	60	Delete record	Restart timeout	After active	GSM 94 [1:12.1]	User 'Novy Josef'
13.10.2011 11:11:08	Call	Phone number	225588000	60	Delete record	Restart timeout	After active	ISDN PRI 2 [1:5.1]	National Number '261301276'

Figure: View of Identification Table Record

Example 1

Suppose user **A** is calling to the public network via a GSM port to which an AutoClip router is assigned. The called user **B** does not answer the call. A new call record is added to the above–mentioned AutoClip router containing the CPN, record validity time, calling user, information on the carrier used for such call establishment and other parameters. Having found a missed call, user **B** cannot identify the calling user because the CLI is represented by your PBX SIM card number. User **B** tries to call back to that number and the call is coming to your PBX carrier that was used for the outgoing call earlier. The CLI of this incoming call matches the CLI stored in the AutoClip router. If this record is still valid, the incoming call is routed directly to user **A**.

Example 2

This example relates to calls that are not established by a PBX user but pass through the PBX from one port to another. In this case, the AutoClip parameters have to be assigned to the incoming port. The record added to the AutoClip router includes the CLI in the **Final destination** column instead of the user name. If an incoming call matches a table record, it is routed to the stored number as defined in the **Destination for address** option.

8. Users

Here is what you can find in this chapter:

- 8.1 Users and Groups
- 8.2 User Rights
- 8.3 Extension Types
- 8.4 Extensions
- 8.5 Phone Directories

8.1 Users and Groups

User Creation

To set users use the **Users – Users and Groups** menu. In this menu you can also manage groups and extensions. A list of available groups, subgroups, users and extensions is displayed on the left.

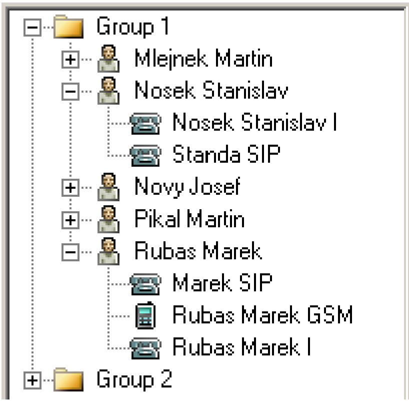


Figure: PBX User Structure from Groups to Extensions

In the context menu you can find the following options:

- **Add user** – use this option to add a user to a selected basic group/subgroup.
- **Add extension** – use this option to add a new extension to a selected user.
- **Copy extension** – use this option to create a extension for the user with the same settings as the currently selected extension has. All the user has to do is enter a new name for the extension.
- **Create by wizard** – use this option to initiate the automatic extension–creating wizard. With it you can import the extension list or create a extension according to preset numbers and ranges.
- **Add group** – use this option to add a basic user group. You cannot add subgroups and users unless one basic group has been created at least (the options are unavailable).
- **Add subgroup** – use this option to add a subgroup to a selected basic group/subgroup. The subgroups can be nested on several levels.
- **Move to root level** – use this option to move the selected object to the highest level beyond all groups and subgroups created.
- **Delete** – use this option to delete existing basic groups, subgroups or users. It is unavailable if the basic group or subgroup contains any nested object (subgroup or user). To delete such items, delete or transfer all the objects nested therein.
- **Rename** – use this option to rename a selected existing basic group, subgroup or user.
- **Move to** – use this option to move users to another basic group/subgroup.
- **Find (F3)** – use this option to initiate a searching dialogue within this menu using the preset rules. Items are searched for on all levels from groups to extensions.
- **Find next (F5)** – use this option to enable repeated searching of the string that has been entered using the **Find (F3)** function.
- **Expand all** – use this option to open the whole structure of groups and

subgroups with users and stations easily.

- **Collapse all** – use this option to close the whole structure of groups and subgroups with users and stations easily.

Moving records using the mouse, also called **drag & drop**, has been implemented in this menu for easier moving of existing extensions, users, groups and subgroups.

While creating the basic groups or subgroups you are requested to set the group or subgroup name only. For user creation, however, a dialogue is displayed for you to define more parameters and even assign extensions to such user as shown in figure below.

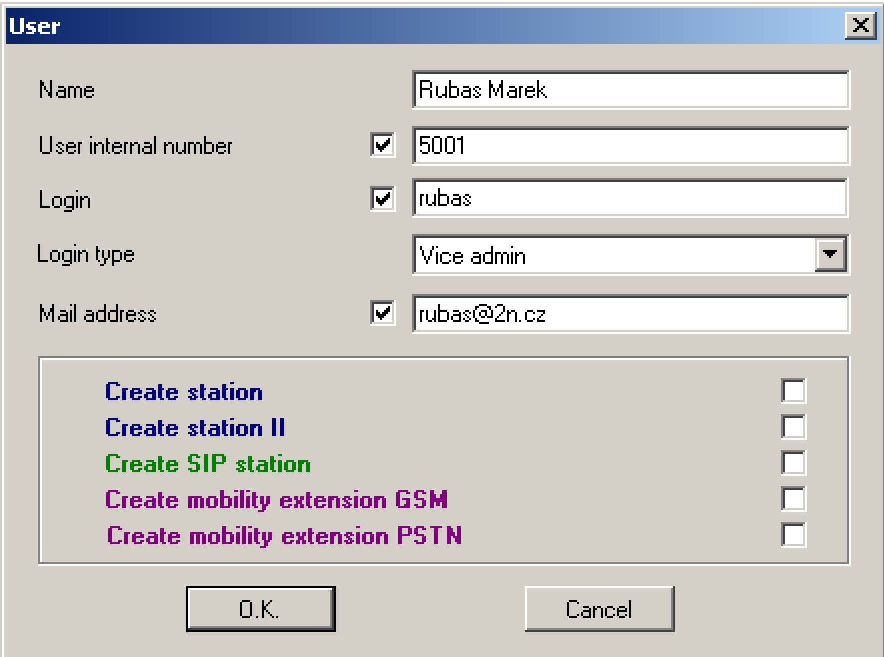


Figure: View of User Creating Dialogue

The following part of this chapter describes the **Users and Groups** menu tags:

Basic

In case a group of users is selected, two programmable parameters are displayed to the right with the following meanings:

- **Save messages** – here enable message saving into the PBX memory. If message saving is disabled for a group, messages are not displayed on the group's Cornet ports.
- **Maximal number of messages** – use this parameter to define the maximum count of messages to be saved in the PBX for a user. Whenever this limit is achieved, the messages are deleted as necessary (starting from the oldest ones).
- If a user is selected, the tag includes the following additional parameters:
- **PIN** – here fill in the Personal Identification Number (PIN). This number should contain four digits and is used for access to protected PBX services (e.g. Private call). The default value is 1111.
- **Internal number** – this number is primarily used for user identification within the PBX and represents the necessary condition for SMS messages routing.

- **E–mail address** – here fill in the user e–mail address to be used for user VoiceMail forwarding. If this field is empty, the user will not be able to use service call forwarding to VoiceMail because the voice messages created will have no target destination.
- **Alias** – this parameter is used by the PC operator and Application server external applications. Alias in the PBX corresponds to the user name in the Active Directory. Alias and e–mail are used in the exchange server for checking user identifications and user profiles may subsequently be switched according to the calendar events. For detailed information refer to the Application server manual.
- **Status** – here select one of the pre–defined statuses to be used for user status identification by co–operating applications. The pre–defined statuses correspond to the standard statuses applied in instant messaging applications.
- **Presence string** – here fill in the text to be displayed to the user calling to one of your extensions.
- **Active profile** – use this field to define the current active profile of the user. You can also select an item from the list of available profiles.
- **Automatic profile switching** – use this option to enable automatic profile switching according to time conditions as defined in the Time Conditions tag.

If you select a extension, you will see more options. For all of them refer to the 8.4 Extensions menu.

Properties

The **Properties** tag consists of a lot of subtags, which are described in a separate chapter for convenience. This tag is exceptional because almost all of its parameters follow the hierarchical structure. For the structure and all the parameters refer to Chapter 9. Setting Properties.

Profiles

User profiles facilitate user setting handling by changing multiple parameters in one step. Each user can use up to eight profiles (or nine if we include the no–profile setup), which feature an unlimited count of different parameters. To create a profile, use this configuration tool, a key phone or the Assistant user web application. In the context menu of this tag you can find the following options:

- **Add** – use this option to add new user profiles. This option is disabled once the eighth profile has been created. A profile number is assigned to each profile automatically. This number is always greater by one than the current largest profile number assigned to this user. Moreover, the dialogue helps create profiles according to the existing user profiles. The profiles are copied including all respective settings.
- **Delete** – use this option to remove a selected profile from the database.
- **Rename** – use this option to rename a selected profile.

The user profile configuration is divided into the three tags:

Basic

- **Name** – shows only the name of the selected user profile. It has an informative character only and cannot be changed here. To change it, use the **Rename** option in the context menu as described above.
- **Number** – represents a profile identifier used primarily for **Profile activation**. If you do not fill in this field, you will not be able to use this service.
- **Bundle** – use this option to assign a selected profile to one of the available PBX bundles. Upon activation of a profile to which a bundle has been assigned, the user is automatically added to this bundle. Upon deactivation, the user is automatically removed from this bundle.
- **Presence string** – here type the text to be displayed to the user calling to one of your extensions. This setting has a higher priority than the same setting on the no–profile user level. It means that if this profile is active, this text will be displayed independently of other settings.
- **Status** – here select one of the pre–defined statuses to be used for user status identification by co–operating applications if the given profile is active. The pre–defined statuses correspond to the standard statuses applied in instant messaging applications.

VoiceMail

This tag is similar to the VoiceMail tag on the user level. However, this tag does not support all parameters. It is only used for more precise settings of the user profile. The parameters of this tag have a higher priority. You can set the following:
Content

- **Progress** – here set the progress tone to be played to the calling user in the case of call forwarding to VoiceMail.
- **CFNA (Forwarding at no answer)** – here set the forwarding to VoiceMail in case the incoming call is not answered before the timeout end. To specify the timeout, use the **Forwarding** subtag in the **Properties** tag for the respective user. The default value is 30 seconds.
- **CFU (Forwarding unconditional)** – here set the unconditional forwarding to VoiceMail. It means that all incoming calls will be forwarded directly to the VoiceMail if this profile is active (unless there is a hierarchical exception).
- **CFEC (Forwarding on error cause or busy)** – here set the forwarding to VoiceMail in the case of busy user or another error cause detection (e.g. call rejection).

Properties

The **Properties** tag consists of a lot of subtags, which are described in a separate chapter for convenience. This tag is exceptional because almost all of its parameters follow the hierarchical structure. For the structure and all the parameters refer to Chapter Setting Properties.

Profiles & Time Conditions

In the Profiles & Time conditions tag assign time conditions to the user profiles created using the **Users – Users and Groups – Profiles**. The context menu of this tag has two options only:

- **Add** – use this option to add a new row to the table. Doing this choose one of the given time conditions for this row. You can assign one time condition just once to one user. After all the available time conditions have been used, the **Add** option becomes unavailable until you create another time condition.
- **Remove** – use this option to remove table rows.

One profile may only be assigned to one time condition within the time condition validity period. However, different time conditions can be assigned to one user profile. To make the user profiles switch according to the preset time condition rules, check the **Automatic profile switching** option in the **Users – Users and Groups** menu of the **Basic** tag.

Phone Directories

The **Phone directories** tag is located in the **Users – Users and Groups** menu.

- If you select one of the user groups on the left–hand side of the tag, you will see a list of phone directories assigned to this user group. In the context menu you can use the following options:
  - **Add** – use this option to add a phone book to a selected user group. Choose one of the items of the list of all available phone directories.
  - **Delete** – use this option to remove a selected phone book from a user group.

You can make use of the benefits of the hierarchical structure while assigning phone directories as described in Chapter Setting Properties. The phone directories assigned on the group level are also available to the users of these groups and subgroups.

- If you select one of the users on the left–hand side of this tag, you will see the phone book assigned to the user on the right. The count of the phone book records is limited by the **Maximum user tel. nums.** in the **Basic** subtag of the **Properties**tag. The default value is 1000 records. The context menu contains the following options:
  - **Add** – use this option to add a row to the user phone book. This option becomes unavailable when you reach the maximum count of the phone book records.
  - **Delete** – use this option to remove a selected row from a user phone book.
  - **Delete all** – use this option to remove all rows from a user phone book at once.
  - **Export** – use this option to export the current user phone book in the **xml** or **csv** format.
  - **Import** – use this option to import the user phone book in the **xml** or **csv** format.
  - The user phone book consists of twelve columns with the following meanings:
  - **Name** – shows the name of the user who appertains to this record. This name is shown on the calling/called user's display.
  - **Nickname** – shows the nickname of each record. It is primarily used for easier searching of the phone book.
  - **Scheme** – this column sets the user identification scheme. Choose either **Number** or **URI**.
  - **Prefix** – this column sets the access prefix as defined in the **Global data – Global parameters** menu. This prefix shall be dialled automatically before the user number included in the phone directory.

- **Number** – here fill in the user **Number** or **URI** according to the **Scheme** column.
- **Ring pattern** – choose a specific ring tone for each user phone book record. If the PBX accepts an incoming call with a CPN matching this record, your extension will use this ring tone.

The remaining six columns are used for forwarding incoming calls to a specific destination. The call forwarding settings in the phone directories have the highest priority of all within the PBX.

VoiceMail

The **VoiceMail** tag is used for configuring the user VoiceMail and is available in the **Users – Users and Groups** menu. The tag occurs in two forms depending on the level to be configured.

VoiceMail for Groups

If you select one of the user groups in this tag, you can change only some parameters. The settings on this level have a lower priority than those on the subgroup or user level.

- **Progress** – use this option to choose a VoiceMail progress tone from a list.
- **Message** – use this parameter to set the text of the SMS informing of a new VoiceMail message. In addition to a static text, you can use dynamic strings with the following meanings:
  - **%u** – called user name;
  - **%n** – calling user name;
  - **%c** – calling user number;
  - **%d** – VoiceMail creation date and time.
- **Save to User** – use this parameter to enable/disable saving messages at the user regardless of the user settings, or respecting the user settings. The selection is intended for displaying messages on Cornets and in **2N® NetStar Assistant**.
- **SIP extensions** – use this parameter to enable/disable resending messages to user SIP stations regardless of the user settings, or respecting the user settings (According to stations).
- **Email extensions** – use this parameter to enable/disable resending messages to user email stations regardless of the user settings, or respecting the user settings (According to stations).
- **Mobility Extensions** – use this parameter to enable/disable resending messages to user external stations regardless of the user settings, or respecting the user settings (According to stations).

VoiceMail for Users

If you select one of the users, you can set more parameters than on the group level. The meanings of the parameters are as follows:

- **Enabled** – use this option to enable VoiceMail function for a selected user. It is subject to licence and after saving the setup you have to check whether the **Licensed** option is ticked off. If not, then you do not have enough licences for this function.

Forwarding

- **CFNA** – Call Forwarding at No Answer – use this parameter to set forwarding to VoiceMail in case the incoming call is not answered within the preset period the time. To specify the time limit, use the Forwarding subtag in the Properties tag for the respective user. The default value is 30 seconds.
- **CFU** – Call Forwarding Unconditional – here set the unconditional forwarding to VoiceMail. It means that all incoming calls will be forwarded directly to the VoiceMail if this profile is active (unless there is a hierarchical exception).
- **CFEC** – Call Forwarding on Error Cause – here set the forwarding to VoiceMail in the case of busy user or another error cause detection (e.g. call rejection).

Welcome note

- **Welcome note** – use this option to choose a VoiceMail progress tone from a list.
- **Set welcome note** – use this option to enable/disable recording of a VoiceMail progress tone via the VoiceMail Record welcome note (*35) service.

Messages

- **Maximum record length [s]** – use this parameter to set the maximum voice message recording time. After this time limit, the incoming call will be cleared automatically.
- **Do not store** – use this option to enable/disable saving of VoiceMail to the PBX. Voice messages are only resent to the corresponding e–mail (according to the setting).
- **Maximum record term [s]** – use this parameter to set the maximum voice message storing time in the PBX. After this time limit, the voice message will be removed the moment another message is saved.
- **Delete oldest at no space** – use this option to remove the oldest voice messages in order to get more space for new voice messages.
- **Maximum record count** – use this parameter to set the maximum count of voice messages to be stored in the PBX.

Notification

- **Message** – use this parameter to set the text of the SMS informing of a new VoiceMail message. In addition to a static text, you can use dynamic strings with the following meanings:
  - **%u** – called user name;
  - **%n** – calling user name;
  - **%c** – calling user number;
  - **%d** – VoiceMail creation date and time.
- **Save to User** – use this parameter to enable/disable saving messages at the user regardless of the user settings, or respecting the user settings. The selection is intended for displaying messages on Cornets and in **2N® NetStar Assistant**.
- **SIP extensions** – use this parameter to enable/disable resending messages to user SIP stations regardless of the user settings, or respecting the user settings (According to stations).
- **Email extensions** – use this parameter to enable/disable resending messages to user email stations regardless of the user settings, or respecting the user settings (According to stations).
- **Mobility Extensions** – use this parameter to enable/disable resending messages to user external stations regardless of the user settings, or respecting the user settings (According to stations).
- **Send e–mail without voice message** – use this option to enable e–mail sending even if the calling user hangs up before the recording starts. Remember to enable **Send to user mail** to make this function work.

Files

This tag is located on the group and user levels and used for viewing files with calls recorded via these objects. Refer to the 3.8 Virtual Port Options chapter, the Files section, for details on the overview table columns and/or context menu options.

Assistant

This tag can only be used on the user level. The default Assistant settings are available in the **Global data – Assistant – Administration setting** menu. To change a user setting, create an individual user setting using the following parameters:

- **Application main page** – this option to define the Assistant home page for a selected user.
- **Default language** – use this option to define the Assistant default language for a selected user.
- **Image directory** – use this option to define the set of images to be used by the Assistant for a selected user.
- **CSS style filename** – use this option to define the Assistant's appearance for a selected user.

Free minutes/SMS

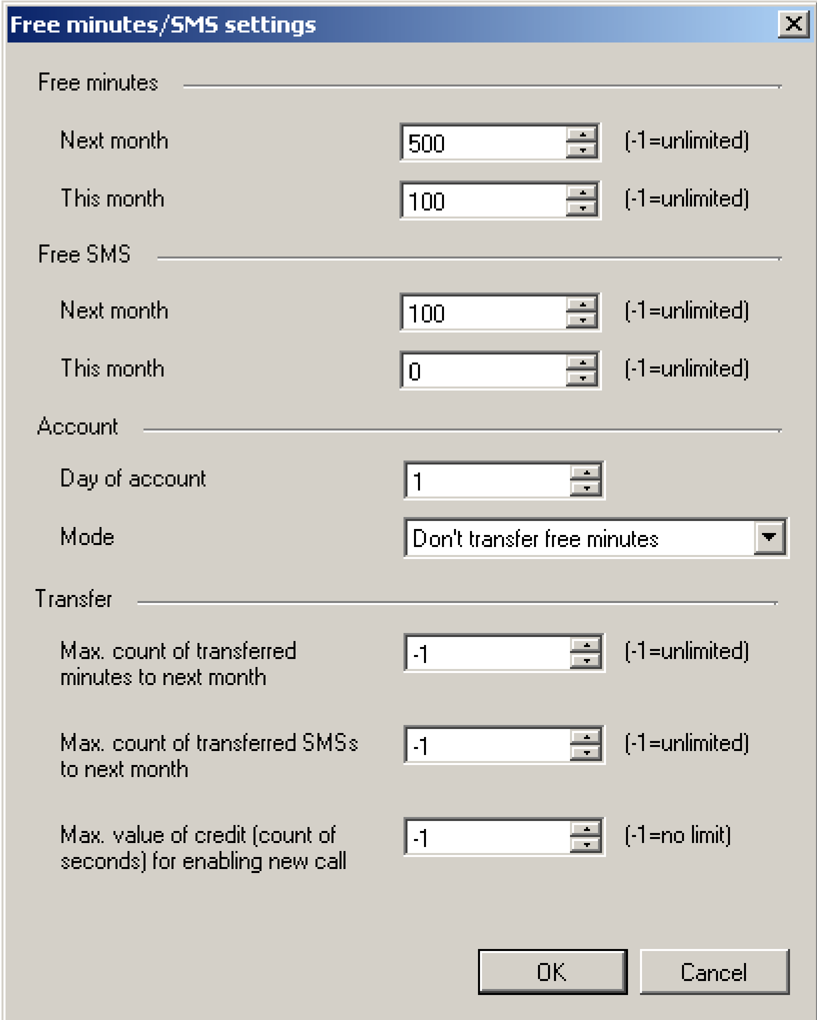
The tag helps you set free minutes and SMS for a selected user. The set count of minutes and SMS shall only be deducted on the ports via which the call goes out of the PBX and that are selected for call billing on port (**Basic** tag of the given port). All the **Default OUT**ports are such ports by default.

Select tariff rate

Click on the **Set free minutes/SMS** button to display a dialogue and select one of the tariff rates as defined in the **Accounting and tariff rates** menu. In addition, you can assign here a setting to the selected user tariff rate as defined earlier for any other user or virtual port. To change the tariff rate if necessary, use the **Used tariff rate** opt ion. If you do so, you will lose all data saved on free minutes with the given tariff rate via this user. To cancel the user tariff rate, push the **Cancel free minutes/SMS** butto n.

Free minutes/SMS settings

Once a tariff rate is selected, the tariff rate credit rows are displayed in this section. Click on a row to display a setting dialogue for the count of free minutes, SMS messages and other credit parameters for the given user. See the figure below for the dialogue.



The table includes columns with the following meanings:

- **Credit name** – the credit name as defined during tariff rate creation.
- **Free minutes for month** – the column includes the count of free minutes per month for the given user. This count is credited to the given user at the beginning of the accounting period. If the free minute count changes within a month, the port credit is not increased until the beginning of the next accounting period unless provided otherwise in the setting dialogue.
- **Free minutes for this month** – the column shows the current count of free minutes to be used in this month. The value includes free minutes transferred from the previous accouting period if any.
- **Spent minutes** – displays the current count of minutes spent in the accounting period.
- **Free SMS for month** – the column includes the count of free SMS messages per month for the given user. This count is credited to the given user at the beginning of the accounting period. If the free SMS count changes within a month, the port credit is not increased until the beginning of the next accounting period unless provided otherwise in the setting dialogue.
- **Free SMS for this month** – the column shows the current count of SMS messages to be used in this month. The value includes free SMS transferred from the previous accouting period if any.
- **Spent SMS** – displays the current count of SMS sent in the accounting period.

- **Day of account** – here set the day in the month on which a new accounting period shall start. On this date, the free minute and SMS counts are increased according to the selected transfer mode. The mimimum values are set in the Free minutes for month a Free SMS for month columns. Setting **0** means **Never** (Manually) and setting **32** means **Every day**.
- **Mode**– use this option to select the method of transfer of old free minutes into the next accounting period.
  - **Do not transfer** – no free minutes and/or SMS are transferred.
  - **First use new** – old free minutes and SMS are transferred but new ones are used first. Unused units older than one month are not transferred.
  - **First use transferred** – old free minutes and SMS are transferred and new ones are not used until these old units have been exhausted. Unused units older than one month are not transferred.

8.2 User Rights

Logins

A list of all users and logins is displayed on the left-hand side of the **Users – Users rights** menu. The list is divided into sections according to user groups and subgroups. The user name is on the left and the respective login name, if any, on the right. You can use the following context menu options here:

- **Create login** – use this option to create a login for a selected user. This option is active only if the user has not been assigned any login. You can choose one of the types specified below.
- **Change login** – use this option change the login type. The option cannot be used for Admin login. Each login is also assigned a type that defines the respective right assignment level. Choose one of the following options:
  - **Vice Admin** – is a login with all rights except for **Delegate logins** and **Manage own group**.
  - **Super** – is a login with the same rights as the Admin login.
  - **Vice Super** – is a login with the same rights as the Vice Admin login.
  - **Manager** – is a login designed for the manager of all user logins in a group. Hence, the **Delegate logins** and **Manage own group** rights are assigned to the manager.
  - **Vice Manager** – is a login to be used by the **Manage own group** user.
  - **User** – use this option to create a general user login.
- **Change password** – use this option to change the password of a selected login.
- **Remove login** – use this option to delete a selected login. The Admin login cannot be deleted.
- **Generate missing logins** – use this option to automatically create logins for all the users who have not been assigned one. Such logins are of the user type, have no password and their names are respective user names (without spaces and diacritic marks, with small letters).

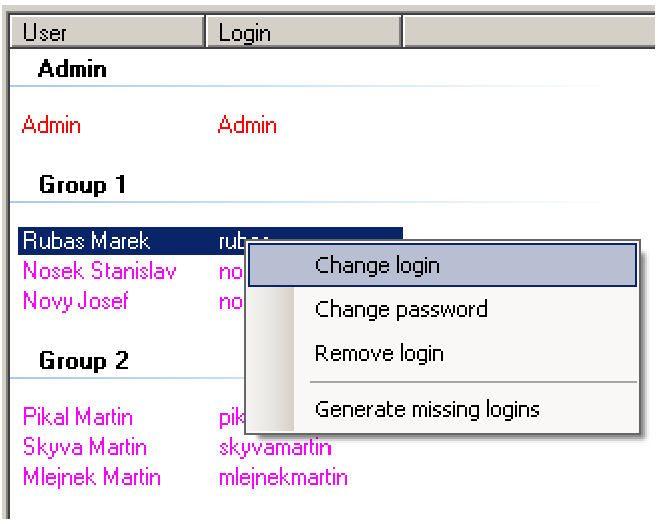


Figure: View of Logins According to Groups

Basic

After selecting a user, a list of all the users of the respective group including logins and rights is displayed on the right-hand side of the **Basic** tag. This view is useful for setting similar rights in the user group. The table of rights is divided into sections with the following meanings.

- **Basic**
  - **Disable** – use this option to disable a login for a period of time without deleting it.
  - **Must change password** – use this option to set automatic advice of a password change upon access to the Assistant application.
- **Tab directly**
  - **Read** – use this right to enable reading of the database via the configuration tool.
  - **Write** – use this right to enable writing into the database via the configuration tool.
- **Database**
  - **Write** – use this command to save completed changes into the database.
- **Trace**
  - **See** – use this command to display the Trace tag in the tool.
  - **Enable** – use this command to enable trace downloading from the PBX.
- **Statistics**
  - **See** – use this command to view statistic data through the Statistics tag in the tool.

Assistant

After selecting a user, a list of all the users in the respective group including logins and rights is displayed on the right-hand side of the **Assistant** tag. This view is useful for setting similar rights in the user group. The table of rights is divided into sections with the following meanings:

- **User management** – enables to view and change the settings of other users.
- **Telephone directory management** – enables to view and change the directories of other users.
- **Call history management** – enables to view the call history of other users.
- **Telephone management** – enables to view and change the telephone settings of other users.
- **Extension management** – enables to view and change the extension settings of other users.
- **Global configuration management** – enables to view and change the global configuration settings.
- **Operator management** – enables to view and manage the operator settings.
- **Alarm management** – enables to view and change the alarm settings.
- **SMS management** – enables to view and manage the SMS messages of other users.
- **Conference room management** – enables to view and manage the conference rooms.
- **Hotel view** – enables to view and manage alarm clocks and emergency alarms in a hotel structure.
- **Recorded calls** – enables to view and manage recorded calls of the user.

8.3 Extension Types

Extension Type Creation

This tag gets displayed whenever you click on the **Users – Extension Types** menu. The extension types are used for easier setting of groups of extension. A list of available extension types is displayed on the left and you can set a selected extension type on the right. On the left, you can use the context menu with the following options:

- **Add** – use this option to add a extension type.
- **Delete** – use this option to delete a selected extension type.
- **Rename** – use this option to rename a selected extension type.
- **Copy extension type** – use this option to create a extension type with the same settings as the currently selected extension type has.

Extension Type Properties

The Properties tag consists of a number of subtags, which are described in a separate chapter for convenience. This tag is exceptional because almost all of its parameters follow the hierarchical structure. For the structure and all the parameters refer to Chapter Setting Properties.

8.4 Extensions

Extension Creation

This tag gets displayed when you click on the **Users – Extensions** menu. A list of available extensions is on the left and settings for a selected extension on the right. On the left, you can also use the context menu with the following options:

- **Add** – use this option to add a extension. After clicking on this option you will see a dialogue box as shown in Figure 1. First define the extension name. If you choose an already existing name, the extension will not be created and you will be warned. Assign each extension to a specific user in this dialogue box too. Furthermore, fill in the **extension number** and, if you are creating an external extension, fill in the **Routing number** too (for call routing within other networks) and, if applicable, check the option **Resend SMS** to resend all incoming SMS messages to the external extension.

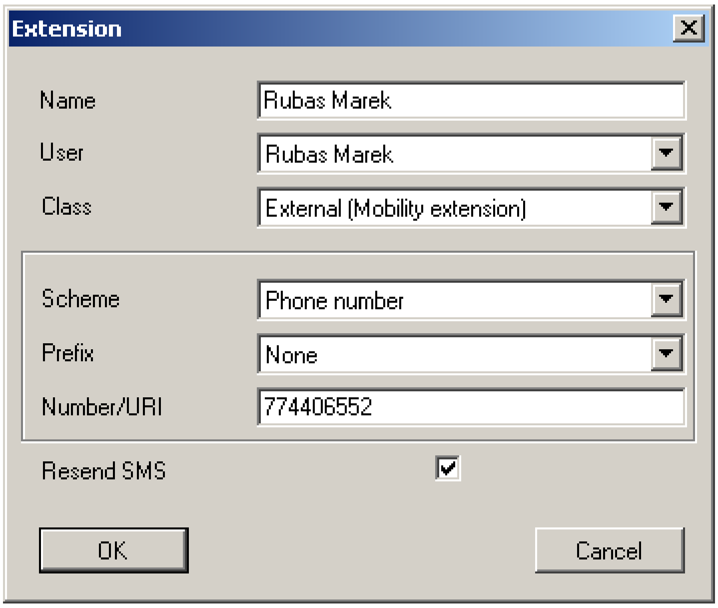


Figure: Extension Creation Dialogue

- **Delete** – use this option to delete a extension.
- **Rename** – use this option to rename a extension. If you fill in an already existing name, you will be warned.
- **Copy extension** – use this option to create a extension for the selected user with the same settings as the currently selected extension has. All the user has to do is enter a new extension name.
- **Find** – use this option to search extensive corporate databases for a extension by its name or number. The name or number to be searched need not be complete (may be a part of the string only).
- **Find next** – use this option to enable repeated searching of the string that has been entered in the **Find** function. This option is unavailable until you fill in the

string to be searched by the **Find** function.

Basic Settings

If you select a extension on the right-hand side of the screen, three tags will get displayed on the right: **Basic**, **Properties** and **Profiles**. The **Basic** tag contains the following parameters:

- **Object** – specifies the object type.
- **Name** – shows the name of the selected station.
- **Station type**– defines the station type. The following options are available:
  - **Normal** – a normal internal station.
  - **SIP** – a SIP station. It should be assigned to a terminal on the SIP Proxy.
  - **E-mail** – an e-mail station. Not intended for calling.
  - **External** – a Mobility Extension station.
- **Scheme** – use this option to define the station identification scheme. Choose either a telephone number, or URI.
- **Prefix** – here choose one of the prefixes defined in the Global parameters menu. This prefix partly substitutes the number subtype and facilitates CallBacks.
- **Number/URI** – define the station identification. Enter a number, e-mail address, or URI. The function of the parameter depends on the Station type setting. When an external station is identified, the originally dialled number is changed and the call is routed with this number via the defined destination.
- **User** – shows the name of the current user. Use this option to assign a extension to another user too.
- **Type** – use this option to assign a extension to a specific extension type. It can facilitate setting of the common parameters for a group of extensions (e.g. outgoing routing via a GSM bundle for all external extensions).
- **Ring group** – use this option to select the ring group in which you may take over calls from the members of the group without being a member of the group (default dial *#).
- **Active** – use this option to activate/deactivate a selected extension. A deactivated extension becomes unreachable for other extensions (incoming calls are rejected) but is able to establish outgoing calls.
- **Do not ring at call to user** – use this option to route a call to this extension in case it is routed to the user. If it is checked, only the calls routed directly to this extension alert the extension.
- **Resend SMS** – use this option to enable/disable SMS resending. If this option is checked, all the SMS messages delivered to the user are resent to this external extension.
- **Enable CallBack object** – use this option to enable the CallBack function for a selected extension. The function is subject to licence and so make sure that the **Licensed** option has been selected after data saving. If not, check your licence in the **Global data – Licences** menu.

Others

- **Virtual port** – this parameter shows the port to which the extension is currently assigned. The parameter has an informative character only and cannot be changed in this menu.
- **Protocol** – this parameter defines the communication protocol to be used by the virtual port to which the extension is currently assigned. The parameter has an informative character only and cannot be changed in this menu.
- **Terminal** – this option provides a correct identification of the calling user. It is used only for the extensions that are assigned to the ISDN, SIP or Cornet ports. In other cases, you can connect one terminal only to each physical port and so the terminal identification matches the extension number.
  - You can connect two terminals to the Cornet port – **Master** and **Slave** but the PBX can only connect one digital telephone to the physical port and so you are recommended to keep the **Master** setting.
  - You can connect a bus with up to eight terminals to the ISDN BRI port. Each terminal has its own identification (Multi Subscriber Number, MSN). Assign the MSN numbers to the terminals created in the **Stack** tag for the BRI port. Use the **Terminal** option to assign a selected extension to one of the available terminals.
  - The SIP terminals identify themselves with their SIP URI. Define the terminals on the SIP proxy level. Use the Terminal option to assign a selected extension to one of the available terminals.
- **Active** – this option means that the station on the given port is the main station. Its outgoing calls are identified as this station. Incoming calls are routed to secondary stations too.
- **Goto virtual port** – click on the button to pass to the settings for the virtual port to which the station is logged in.

Required licences

This section displays the licence requirements and statuses for the CallBack, Mobility Extension and Call recording services. The fact that a licence is required yet absent or insufficient in the PBX is signalled by a red text. The fact that a licence is required and present and valid in the PBX is signalled by a blue text.

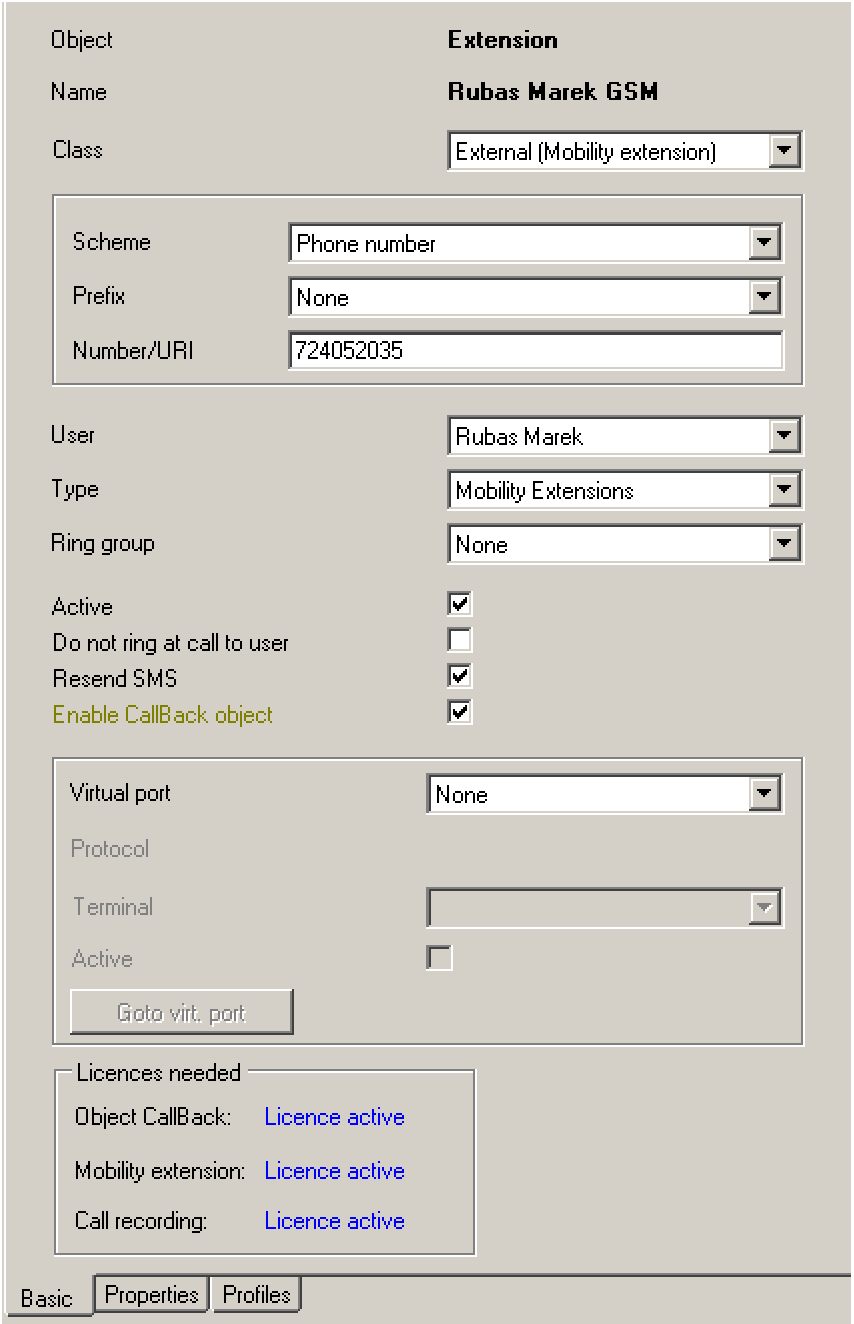


Figure: View at Extension Options

Extension Properties

The Properties tag consists of a lot of subtags, which are described in a separate chapter for convenience. This tag is exceptional because almost all of its parameters follow the hierarchical structure. For the structure and all the parameters refer to Chapter Setting Properties.

Profiles

In this tag, define the properties of a extension within a selected user profile. The extension profile is the highest priority setting. You cannot create new profiles but can edit the existing ones. A list of the profiles created on the user level is displayed on the left. When you select one of these profiles, you will see two new tags – **Basic** and **Properties**. Find the following parameters in the **Basic** tag of the extension profile:

- **Active** – use this option to activate an extension within a selected user profile. If it is not checked off, all calls coming to this extension are rejected. The extension can establish outgoing calls.
- **Do not ring at call to user** – use this option to enable call routing to an extension within call routing to an extension user when the user profile is active. If it is checked off, only the calls routed directly to this extension alert the extension.

The Properties tag consists of a lot of subtags, which are described in a separate chapter for convenience. This tag is exceptional because almost all of its parameters follow the hierarchical structure. For the structure and all the parameters refer to Chapter Setting Properties. **Remember that the extension profile level setting has the highest priority!**

8.5 Phone Directories

Here is what you can find in this section:

- User Phone Directories
- Group Phone Directories
- Group Phone Directories (Generated)
- Common Phone Directories
- SIP Phone Directories

User Phone Directories

Having been created, each user is automatically assigned a private phone directory. A list of user phone directories is displayed on the left-hand side of the **Phone directories – User phone directories** menu. The phone directory has a limited capacity of records. The default value is 10 records per user. This limit can be changed using the Maximum user tel. nums parameter in the **Basic** tag in the user settings. To edit the records use the **Users – Users & Groups** menu in the **Phone directory** tag.

In the context menu on the right-hand side of the menu you can use the following options:

- **Add** – use this option to add a row to a selected phone directory.
- **Delete** – use this option to remove a selected row from a selected phone directory.
- **Delete all** – use this option to remove all rows from a selected phone directory.
- **Find (F3)** – use this option to search a selected phone directory for a record. Enter complete initial words respecting the lower and upper cases.
- **Find next (F5)** – use this option to enable repeated searching of the string that has been entered in the **Find (F3)** function.
- **Export** – use this option to export the current phone directory into an **xml** or **csv** file.
- **Import** – use this option to import a phone directory saved in the **xml** or **csv** format.
- The phone directory table in this menu consists of records divided into six columns with the following meanings:
- **Nickname** – in this column set the name to be used for easier phone directory searching.
- **Name** – in this column set the name of the extension for which the record has been created. This name will be displayed on your phone.
- **Scheme** – in this column define whether the entered string represents the **Number** or **URI**.
- **Prefix** – here set the access prefix as defined in the **Global data - Global parameters** menu. This prefix shall be dialled automatically before the user number included in the phone directory.
- **Number/URI** – in this column define the phone number (or URI) to be entered in the format corresponding to the selected subtype.
- **Ring pattern** – in this column assign different ring patterns to each record of the phone directory. The ring tone will be used if the calling party number of the incoming call matches the phone directory record. If this ring pattern is not set, the default pattern for the final destination will be used.

The last six columns of this tag are used for call forwarding settings for each contact. One pair of columns is intended for each forwarding type. In the first column of each pair set the destination type for forwarding and in the other set a destination of the selected type. **This forwarding mode has a higher priority than the mode selected in the Forwarding and Forwarding-exceptions tags!**

Group Phone Directories

For each group of users, a group phone directory is created automatically and filled with the user telephone numbers. You cannot add or remove records manually in this directory. You can just edit appropriate parameters in the **Scheme**, **Subtype**, **Ring pattern** and call forwarding columns. For the group phone directory refer to the **Phone directories – Group phone directories** menu.

In the context menu on the right–hand side of the menu you can use the following options:

- **Add** – use this option to add a row to a selected phone directory.
- **Delete** – use this option to remove a selected row from a selected phone directory.
- **Delete all** – use this option to remove all rows of a selected phone directory.
- **Find (F3)** – use this option to search a selected phone directory for a record. Enter complete initial words with respect to lower and upper cases.
- **Find next (F5)** – use this option to enable repeated searching of the string that has been entered in the **Find (F3)** function.
- **Export** – use this option to export the current phone directory into an **xml** or **csv** file.
- The phone directory table in this menu consists of records divided into six columns with the following meanings:
- **Nickname** – in this column set the name to be used for easier phone directory searching.
- **Name** – in this column set the name of the extension for which the record has been created. The name will be displayed on your phone.
- **Scheme** – in this column define whether the entered string represents the **Number** or **URI**.
- **Prefix** – here set the access prefix as defined in the **Global data - Global parameters** menu. This prefix shall be dialled automatically before the user number included in the phone directory.
- **Number/URI** – in this column define the phone number (or URI) to be entered in a format corresponding to the selected subtype.
- **Ring pattern** – in this column assign different ring patterns to each record of the phone directory. The ring tone will be used if the calling party number of the incoming call matches the phone directory record. If this ring pattern is not set, the default pattern for the final destination will be used.

The last six columns of this tag are used for call forwarding settings for each contact. One pair of columns is intended for each forwarding type. In the first column of each pair set the destination type for forwarding and in the other set a destination of the selected type. **This forwarding mode has a higher priority than the mode selected in the Forwarding and Forwarding–exceptions tags!**

Group Phone Directories (Generated)

For each group of users, a dedicated phone directory is generated and filled with the users or extensions as defined in the **Generate phone directories from users** param eter in the **Global Data – Global parameters** menu. Every change in the name, number, scheme or subtype is automatically made in the generated phone directory too. For group phone directories refer to the **Users – Phone directories – Group phone directories (Generated)** menu.

In the context menu on the right-hand side of the menu you can use the following options:

- **Find (F3)** – use this option to search a selected phone directory for a record. Enter complete initial words respecting the lower and upper cases.
- **Find next (F5)** – use this option to enable repeated searching of the string that has been entered in the **Find (F3)** function.
- **Export** – use this option to export the current phone directory into the **xml** or **csv** file.
- The phone directory table in this menu consists of records divided into twelve columns with the following meanings:
- **Nickname** – in this column set the name to be used for easier phone directory searching.
- **Name** – in this column set the name of the extension for which the record has been created. This name will be displayed on your phone.
- **Scheme** – in this column define whether the entered string represents the **Number** or **URI**.
- **Prefix** – here set the access prefix as defined in the **Global data - Global parameters** menu. This prefix shall be dialled automatically before the user number included in the phone directory.
- **Number/URI** – in this column define the phone number (or URI) to be entered in a format corresponding to the selected subtype.
- **Ring pattern** – in this column assign different ring patterns to each record of the phone directory. The ring tone will be used if the calling party number of the incoming call matches the phone directory record. If this ring pattern is not set, the default pattern for the final destination will be used.

Common Phone Directories

To create common phone directories use the **Phone directories – Common phone directories** menu. You can create an 'unlimited' number of phone directories and assign them to selected groups of users.

The context menu on the right–hand side of this menu offers the following options:

- **Add** – use this option to add a row to a selected phone directory.
- **Delete** – use this option to remove a selected row from a selected phone directory.
- **Delete all** – use this option to remove all rows of a selected phone directory.
- **Find (F3)** – use this option to search a selected phone directory for a record. Enter complete initial words respecting the lower and upper cases.
- **Find next (F5)** – use this option to enable repeated searching of the string that has been entered in the **Find (F3)** function.
- **Export** – use this option to export current phone directory into the **xml** or **csv** file.
- **Import** – use this option you import phone directory from the **xml** or **csv** file.
- The phone directory table in this menu consists of records divided into six columns with the following meanings:
- **Nickname** – in this column set the name to be used for easier phone directory searching.
- **Name** – in this column set the name of the extension for which the record has been created. This name will be displayed on your phone.
- **Scheme** – in this column define whether the entered string represents the **Number** or **URI**.
- **Prefix** – here set the access prefix as defined in the **Global data - Global parameters** menu. This prefix shall be dialled automatically before the user number included in the phone directory.
- **Number/URI** – in this column define the phone number (or URI) to be entered in a format corresponding to the selected subtype.
- **Ring pattern** – in this column assign different ring patterns to each record of the phone directory. The ring tone will be used if the calling party number of the incoming call matches the phone directory record. If this ring pattern is not set, the default pattern for the final destination will be used.

The last six columns of this tag are used for call forwarding setting for each contact. One pair of columns is intended for each forwarding type. In the first column of each pair set the destination type for forwarding and in the other set a destination of the selected type. **This forwarding mode has a higher priority than the mode selected in the Forwarding and Forwarding–exceptions tags!**

SIP Phone Directories

Find the SIP phone directories in the **Users – Phone directories – SIP phone directories** menu. You can define one general phone directory source for the whole PBX and distribute it to the SIP extensions.

Phone directory source

Here define the phone directory source. On the basis of the source, a phone directory is generated for the SIP extensions and stored in the TFTP storage for the SIP terminals. The following options are available:

- **Disabled** – no directory is generated.
- **Group** – the directory is generated from the directories assigned to the selected group.
- **User** – the directory is generated from the directories of the selected user.
- **Extension** – the directory is generated from the directories of the selected extension.

1. Having received the **gs_phonebook.xml** downloading request, NetStar generates the file in the **GrandStream** telephone format from the selected source and sends it.
2. Having received the **tftpPhoneBook.xml** downloading request, NetStar generates the file in the **2N® StarPoint IP T2x** telephone format from the selected source and sends it.
3. Having received a downloading request for another file, NetStar searches the TFTP storage and sends the file if available.

9. Setting Properties

Here is what you can find in this chapter:

- 9.1 Setting Properties

9.1 Setting Properties

Fall-Down Hierarchy

All the **Properties** tag parameters are used according to a fall-down hierarchy of the PBX. It means that setting a parameter on one level you cannot be sure that it will be used. Each level of this fall-down hierarchy has a preset priority. The following figure defines all the fall-down hierarchy levels. The higher the level, the higher the priority.

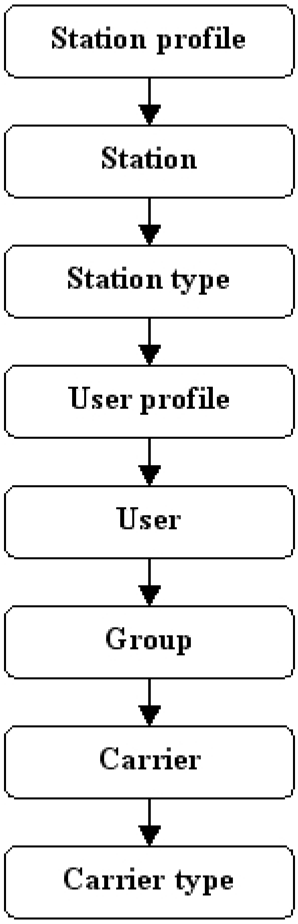


Figure: View of PBX fall-down Hierarchy. Higher levels have higher priorities

It implies from the figure above that the parameters set on the extension profile level have the highest priority and the parameters set on the virtual port type level have the lowest priority. If a parameter is set to the **Default** value on a level, a different setting on a lower level is searched for this parameter. If a parameter is not set on any level (Default is set on all levels), the PBX uses the value preset by the source code.

Properties Tag

The **Properties** tag is situated in the menus of all routing objects as mentioned above (Figure 1). By default, the properties are not set on all levels as they are unnecessary for normal PBX operation. To set a parameter for an object, simply push the **Create properties** button. To cancel a parameter, push the **Reset default properties** button . The **Properties** tag consists of fourteen subtags, which are logically divided according to functions. Some are used only on certain levels because they have no sense on others. The text below explains all the parameters available in the subtags.

Basic

The parameters of this subtag are mostly divided into sections according to their functions:

- **No answer timeout [s]** – use this parameter to set the maximum time of alerting the called phone. After the timeout, the call establishing process is terminated with the 'User not responding' cause and the calling user hears the congestion tone. The default value is 180s. The maximum value is 1 hour.

Holdoff parameters

- **Call hold** – use this option to enable holding of a call. The default value of this parameter is **YES** (Hold enabled).
- **Hold tone** – use this option to set the tone to be played to the user during call holding by the other party. The default tone is the **Music on Hold** progress tone.
- **Maximum hold level** – use this parameter to define the maximum count of held calls per extension. Push the call holding button again to get connected to the first held subscriber, push it once again to get connected to the second held subscriber, and so on. If a held subscriber hangs up, the released position can be used for another hold.

Transfer parameters

- **ICT enabled**– (Implicit Call Transfer) – use this parameter to enable call transfer via a connected terminal (phone). You can use one of the following options:
  - **Blind transfer (with hang–up)** – the call is not answered by the other party before transfer. E.g. user **A** calls to user **B**. User **B** answers the call and users **A** and **B** are talking with each other. User **B** holds the call. User **A** hears the Music on Hold and user **B** establishes a new connection to user **C**. User **C** does not answer the call and user **B** terminates the call. User **A** is transferred to user **C** and hears the alert tone instead of the Music on Hold. When user **C** answers the call, users **A** and **C** can talk to each other.

1. **Transfer without hang–up** – the call is answered by the other party before transfer. E.g. user **A** calls to user **B**. User **B** answers the call and users **A** and **B** are talking to each other. User **B** holds the call. User **A** hears the Music on Hold and user **B** establishes a new connection to user **C**. User **C** answers the call and user **B** terminates the call. User **A** is connected to user **C** and they can talk to each other.

- **Transfer return timeout [ms]** – use this parameter to define the timeout for return from the blind transfer. After this timeout (if not answered), the call returns to the extension that transferred it. The default value of this parameter is

20s.

Queue parameters

The queue parameters are only available on the group and user levels. The **Station polling timeout** is the only parameter on the station level.

- **Queue** – use this option to enable call queuing. It means that if an incoming call is routed to a busy extension with a queue, the call is not terminated, the calling user hears the alert tone and can wait for connection. After the current call is terminated, the phone of the called user is alerted again with your call from the queue. If the queue is disabled, the incoming call on a busy extension is terminated with the 'User busy' cause. The default value of this parameter is **NO** (queue disabled).
- **User busy when station busy** – use this option to select whether or not an incoming call shall make the user busy. The option is enabled by default, which means that the user cannot use two own stations at the same time or call from one own station to another.
- **Queue depth** – use this parameter to set the maximum count of calls to be queued. All excessive calls will be rejected as if the queue had been disabled (User busy). If you set this parameter to zero, the count of queued calls will be unlimited. The default value of this parameter is zero (unlimited).
- **Queue timeout [ms]** – use this parameter to define the delay between the end of the previous call and the beginning of the next queued call alerting. The default value 0 s (alerting starts without delay).
- **Maximum time in queue [s]** – use the parameter to define the period of time for which the calling subscriber may stand in the queue. When this time elapses, the incoming call is terminated.
- **Repeated attempt timeout for stations [s]** – use this parameter to set the time interval for the PBX to re–try to route a call to the selected station. The selection is designed primarilly for external stations, which may be occupied by a call that is not made via the PBX. The busy status is identified the moment the PBX tries to make a call and the network sends the busy tone (cause 17 – User busy is only respected). Call routing is terminated temporarily, but, with an active queue, it is necessary to know when the given station is free and ready to answer. In these cases, the other user stations are not called repeatedly. The parameter is also useful for internal ports with multiple logged–in stations. While one station is speaking on one port, the other cannot be used and is attempted repeatedly until it gets free or the calling subscriber hangs up. In this case, the other user stations are alerted too since the busy cause from the virtual port does not mean a busy user.
- **Next call** – use this parameter to compile a caution on an incoming queued call. The default value of this parameter is **NO** (caution disabled).
- **Next call tone** – set the progress tone to be played to the user during an active call if there is a call in the queue for the user. The progress tones with the **Off** option can only be used for this function. The **Queue Alert** tone is a good example. If you do not have this progress tone in your list, click on **Add default progresses**.

Identification parameters

All identification parameters are available on all hierarchical levels of the PBX except for the **Outgoing hold CLIP** parameter, which is active on the group and user levels only.

- **Incoming hold CLIP** – use this parameter to forward the called party number to the called user in the case of call transfer made by the extension where this parameter is being enabled. It means that, if **YES** is selected, the transferred call will be identified by the CLI of the transferred user (**A**) instead of that of the user who transferred it (**B**). The default value of this parameter is **NO**.
- **Outgoing hold CLIP** – use this parameter to display the original calling party number in the case of call transfer. It means that, if **YES** is selected, you will see the calling party number of the transferred user (**A**) instead of that of the user who transferred the call (**B**). The default value of this parameter is **NO**.
- **Use Replace in identification table** – use this parameter to replace a part of the preset identification (as specified in the identification table) with your own calling extension identification. The default value of this parameter is **YES** (replacement enabled).
- **CLIR**– use this parameter to restrict the calling line identification. Use a service or a pre–programmed StarPoint phone button to change the station settings. The parameter is set to **NO** by default.
- **Language** – use this option to select the language to be used by the StarPoint terminals. Choose one of the languages listed. The default value corresponds to the preset **Localisation** of the PBX.
- **Max phone directory item count** – use this parameter to set the maximum count of records in a user private phone directory. This parameter is available on the user level only! The default value of this parameter is 10 records.

Routing

- **To port** – use this option to route an outgoing call through the selected port. Outgoing means the outward direction from the PBX.
- **From port** – the section includes two types of call routing.
  - **Normal** – define routing of the calls that arrive in the PBX via the port.
  - **For services and calls on hold** – set the routing destination for calls on hold or services without a destination of its own. This helps you easily create a complete routing system for PBX external stations.
  - **AutoClip parameters for calls** – use this option to choose an AutoClip parameter set for storing of AutoClip router records. This option can be used on the user profile, user and group levels only.
- **No port** – use this option to set routing for extensions not assigned to a virtual port. It is primarily used for the external, email and virtual port extensions that are used for special routing cases. This setting is available on the virtual port and virtual port type levels only.

For more information on call and SMS routing refer to Chapter Routers.

Message routing

The **Message routing** tag is available on all hierarchical levels. Its structure is similar to that of the Call routing tag but includes several additional parameters.

- **To port** – use this option to set routing rules for the messages that go out of the PBX through the port.
- **From port** – use this option to set routing rules for the messages that come into the PBX through the port.
- **AutoClip parameters for messages** – use this parameter to assign a set of AutoClip parameters as defined in the **Global data – AutoClip parameters** menu. All records on outgoing SMS messages are stored in the AutoClip router

including these parameters.

- **No port** – use this option to set routing rules for the extensions that are not assigned to any port. Such extensions include, in particular, PBX external or virtual port extensions used for special routing cases.
- For call routing by the PBX refer to the Routers chapter.
- **Parameters of unsuccessful sending**
- **Repeat at fail** – use this option to enable repeating of a failed SMS sending attempt. An attempt may fail due to a GSM network rejection or bad signal quality. This option does not refer to SMS delivery failures due to switch–off or temporary unavailability of a remote GSM extension.
- **Number of repeated attempts** – use this parameter to set the maximum count of sending attempts at an SMS sending failure due to network rejection or temporary signal unavailability. The default value is 4000.
- **Time for message repeat [s]** – use this parameter to set the interval between successive SMS sending attempts. The default value is 180s.

ME

The ME subtag is used for setting parameters for external extensions. They mostly include parameters for call holding using an external extension. Mobile phones are not equipped with a standard PBX call holding key and that is why this function has to be replaced with a sequence of DTMF symbols. This tag contains the following parameters:

- **Transfer** – use this parameter to enable call holding from an external extension. This function is subject to licence and hierarchy. If you define this parameter on the user level, you need as many licences as many extensions the user has (the external extension is not included because it needs licence for use).
- **Pattern time interval [ms]** – use this parameter to set the time interval between individual characters of the FLASH and DISCONNECT patterns. This time interval is important for a correct recognition of the pattern. If the delay between the pattern characters is longer than the preset time value, the pattern will not be recognised.
- **FLASH pattern** – use this parameter to set the sequence of the FLASH pattern characters used for call holding. Re–enter the FLASH pattern to reconnect a held call or switch between two calls (one active and one on hold).
- **DISCONNECT pattern** – use this parameter to set the sequence of the DISCONNECT pattern characters. When you have one active call and one call on hold, use this pattern to terminate the active call and reconnect the call on hold.
- **Do not end outgoing call** – use this parameter to select that an outgoing call from the PBX to an external station shall not be terminated after the calling subscriber hangs up. Instead, the user gets the busy tone from the PBX and can, using the DISCONNECT PATTERN, return to the dialtone and dial the required station through the PBX. The selection is disabled by default.
- **Do not end incoming call** – use this parameter to select that an incoming call from an external station to the PBX shall not be terminated after the calling subscriber hangs up or the DISA fails. Instead, the user gets the busy tone from the PBX and can, using the DISCONNECT PATTERN, return to the dialtone and dial the required station through the PBX. The selection is disabled by default.

Forwarding

The whole tag is available on the group and user levels only.

This subtag is used for call forwarding setups. The unconditional call forwarding (CFU) has the highest priority. The other two forwarding types have the same priority and each is used in a different situation. The call forwarding settings in this tag can be changed for a selected group of users in the **Forwarding exceptions** tag, which has a higher priority. Furthermore, it holds true that if extension **A** forwards its calls to extension **B**, then extension **B** can call to extension **A** without being forwarded. This function is called **Boss–secretary**. The following parameters are available in the **Forwarding** subtag:

- **CFNA** – Call Forwarding at No Answer – use this parameter to set call forwarding in case the called user fails to answer within a timeout.
- **CFNA timeout [ms]** – use this parameter to set the timeout for CFNA forwarding. After the timeout expiry, the call is forwarded to the preset destination.
- **CFU** – Call Forwarding Unconditional – use this parameter to set call forwarding of all incoming calls (highest priority). Each incoming call is forwarded to the preset destination regardless of other settings of this subtag.
- **CFEC** – Call Forwarding on Error Cause or Busy – use this parameter to set call forwarding in the case of routing on a busy extension or routing ending up with an error cause.
- **Cause object** – select one of the cause objects as pre–defined in the **Global data – Causes – Cause objects** menu. Basically, a cause object is a set of error causes, which are subsequently respected in the CFEC forwarding mode. You can disable the cause objects or use the default settings of any of the lower levels (Default).

Forwarding – exceptions

The whole tag is available on the group and user levels only.

The **Forwarding – exceptions** subtag is used for specifying exceptions from the forwarding rules set in the **Forwarding** subtag. The exceptions are also applied when no forwarding rule has been set in the **Forwarding** subtag. This tag is also called a Black/White list. Use the phone directory to fill the list with addresses.
A field is available here for each type of forwarding (CFU, CFNA and CFEC), where you can add a limited number of exceptions. Each row represents one exception and is divided into two parts. In the first part, the calling party is defined for which this exception will be valid, and in the other part, a new call routing rule is determined. Setting the calling party, select a extension, user, ring group, virtual port, virtual port type and extension type. Setting the rule, choose any of the destinations available in the **Forwarding** subtag, or select one of the following three options:

- **Disabled** – use this option to disable call forwarding as defined in the **Forwarding** subtag for a selected calling user (users) and allow this user (users) to call the selected destination.
- **Enabled** – use this option to route a forwarded call to the destination preset in the **CFW (Forwarding enabled)** field. If this field is not filled in, the call is routed as with the **Disabled** option.
- **Rejected** – use this option to terminate a forwarded call with the 'Call reject' cause. The calling user hears the congestion tone.

Tones

Use this tag to define the basic tones of the PBX to be played to the calling user. The menu is divided into three parts. The first part, **Dial**, helps you set various dial tones, the second part, **Alert**, helps you set various alert tones and the third part, **Congestion**, helps you set various congestion tones. To add a row defining which tone would be used for which situation use the context menu. A list of situations (states) related to specific types of tones is displayed in the **Type** column. A list of available progress tones is displayed in the **Tone** column. It holds true for the dial tones that a higher row has a higher priority. It is because there may be more valid conditions than one in the dial tones. To change the priority of the rows, use the two arrow buttons on the right. To add a row with a certain priority easily use the **Insert ahead selected** an d **Insert behind selected** options. The **Add** option is used for adding a record behind the currently last one (i.e. the lowest priority record).

Ring patterns

In this tag assign the ring tones according to the calling party. Add more parameter rows to the table using the context menu. The meanings of the columns are as follows:

- **Destination type** – in this column choose one of the listed objects. When this object is recognised, the extension rings according to the **Pattern** column. Choose one of the CLIP, extension, extension type, user, group, virtual port and virtual port type options.
- **Destination** – in this column set an object of the type selected in the **Type** column. If **CLIP** is selected, fill in **CLI** (Calling Party Number).
- **Scheme** – this column is active only if the **CLIP** option has been selected in the **Type** column. It defines whether the incoming CLI (Calling Party Number) scheme is **Number** or **URI**. If this column fails to match the incoming scheme, this row will not be used.
- **Pattern** – in this column select a the ring pattern from the list of available PBX patterns.

Again, it holds true that a row situated higher in the configuration has a higher priority. To change the priority of the rows, use the two arrow buttons on the right. To add a row with a certain priority easily use the **Insert ahead selected** and **Insert behind selected** options. The **Add** option is used for adding a record behind the currently last one (i.e. the lowest priority record).

Softphone

The **Softphone** subtag is used for setting parameters of the StarPoint key phones. These settings are not created automatically (except for the Default IN virtual ports), but on all levels of the fall-down hierarchy using the **Create Softphone extension** but ton. To delete a setting, use the **Remove Softphone extension** button. The **Softphone** tag has two subtags: **Keypad** and **Parameters**. The **Parameters** subtag is used exclusively for setting the StarPoint key phone parameters. The **Keypad** tag helps you set the type of the terminal connected. The key phones are detected automatically on the extension level only. The other subtag settings relate to the StarPoint key phones again.

Terminal – choose an item from a list of available terminals:

- **ANALOGUE** – indicates any analogue terminal.

- **ECONOMY** – indicates a **2N® StarPoint** key phone **– Economy** type.
- **ADVANCED** – indicates a **2N® StarPoint** key phone – **Advanced** type.
- **ENTRY** – indicates a **2N® StarPoint** key phone **– Entry** type.
- **BASIC** – indicates a **2N® StarPoint** key phone – **Basic** type.
- **STANDARD** – indicates a **2N® StarPoint** key phone – **Standard** type.
- **ISDN** – indicates any ISDN terminal.
- **GSM** – indicates any GSM terminal.
- **VoIP** – indicates any VoIP terminal.
- **Optiset Advance** – indicates a **2N® Optiset** key phone – **Advanced** type.
- **Optiset Standard** – indicates a **2N® Optiset** key phone – **Standard** type.
- **Optiset Entry** – indicates a **2N® Optiset** key phone – **Entry** type.
- **OpenStage 10** – indicates a key phone **2N® OpenStage** type 10.
- **OpenStage 15** – indicates a key phone **2N® OpenStage** type 15.
- **OpenStage 20** – indicates a key phone **2N® OpenStage** type 20.
- **OpenStage 30** – indicates a key phone **2N® OpenStage** type 30.
- **OpenStage 40** – indicates a key phone **2N® OpenStage** type 40.
- **2N StarPoint IP T20** – indicates a IP phone **2N® StarPoint IP** type T20.
- **2N StarPoint IP T22** – indicates a IP phone **2N® StarPoint IP** type T22.
- **2N StarPoint IP T26** – indicates a IP phone **2N® StarPoint IP** type T26.
- **2N StarPoint IP T28** – indicates a IP phone **2N® StarPoint IP** type T28.

Extenders - Having chosen one of the **2N® StarPoint** key phone, you can connect extenders with further programmable keys. You can connect up to four extenders with sixteen keys (extender type S16) or up to two extenders with ninety keys (extender type S90).

An eighteen-button extender (S18 type) can only be connected to the **2N® OpenStage 15, 2N® OpenStage 30** and **2N® OpenStage 40** key phones.

Up to two 38-button extenders (IP key modules) can be connected to the **2N® StarPoint IP** phones of the types T26 and T28.

With the **Entry**, **Economy**, **Basic**, **Standard** and **Advanced** terminals, you can set the following parameters:

- **Restart IP terminal** – push the button to restart the given IP terminal. The function is only available for the **2N® StarPoint** IP T2x terminals. **Make sure that the correct terminal type is entered in the * Virtual ports – SIP – Stack – Terminals * menu to avoid errors.**

Key setting

To program the phone keys display a dialogue box by clicking on the selected key. Select the key function and legend in the window and choose any of the following functions:

- **Auto answer** – use this function to set the timeout after which the incoming call is answered automatically.
- **CLIR** – use the function to restrict the CLI. If active, the CLIR function is indicated by a button LED light.
- **DEFAULT** – use this function to clear all the key functions on the given fall-down level.
- **DO NOT DISTURB** – push the button to enable the Do not disturb state, during which the station is inaccessible and the calling subscriber gets the busy tone.

The outgoing calls are not limited in the Do not disturb mode.

- **ESC** – push the Escape key to reject incoming calls, return to a superior level or clear a character in an item.
- **FLASH** – push the Flash key to hold calls. If a call is on hold, you can dial another user or service number. Re–push the key to switch between two calls (one active and the other on hold).
- **STATE**– use the State button to set speed dialling for the selected number and monitor the state of the selected virtual port, user or extension at the same. The user state displays all user extensions. The state is indicated by a LED at the button:
- **INTERCOM** – push the **Intercom** key to interconnect two StarPoint key phones. All you have to do is enter the CPN in the dialogue box. After you push the key, the connection is established automatically. The called user needs a HandsFree key phone for this function.
  - **"Quit"** – the LED is off if all the user extensions are at relax (the selected extension or virtual port is at relax).
  - **"Hook off"** – the LED is on if one of the user extensions at least is off–hook (the selected extension or virtual port is off–hook).
  - **"Is alerted"** – the LED is flashing if one of the user extensions at least is ringing (the selected extension or virtual port is ringing).
- **PHONE DIRECTORY** – use this option to display the phone directory. This key has the same function as the right arrow.
- **MUTE** – push the **Mute** key to temporarily deactivate your key phone microphone. Push the key again to re–activate it.
- **REDIAL** – push the Redial key to dial the last–dialled number.
- **HANDSFREE** – push the **HandsFree** button to switch a call from the HandPhone to the HandsFree mode and back for the StarPoint key phones only. The HandsFree mode uses a microphone and loudspeaker placed on the terminal body.
- **MISSED CALLS** – use this function to enter the **Missed calls** menu.
- **DIALED CALLS** – use this function to enter the **Dialled calls** menu.
- **ANSWERED CALLS** – use this function to enter the **Received calls** menu.
- **NO FUNCTION** – this option has no function and ignores any fall down from lower–priority levels.
- **NEW MESSAGES** – use this function to enter the **Received messages menu**.
- **PROFILES** – use this function to enter the **Profiles** menu for profile activation/deactivation.
- **ACTIVATE PROFILE** – use this function to directly activate or deactivate a selected profile. **The option is not available until the user profile has been created.**
- **CALL SLOT** – use the button as a slot for another call on the selected extension. The count of the call slots is limited by the **Maximum hold level** parameter in the **Properties - Basic** tag. Push this button during a call to hold the call, the held subscriber hears the dialling tone and can route the call to another extension. If the **Queue** parameter is enabled in the **Properties - Basic** tag, the call slots are occupied by incoming calls. Thus, you can switch the callers, putting the inactive call on hold during which the caller hears the Hold tone.
- **CALL RECORDING** – push the button to start recording the current call. Repush the button to stop recording. Refer to the Recording. subtag for call recording details.

Parameter setting

The Parameters subtag offers the following parameters:

- **Key volume** – use this parameter to set the loudness of the key pushed in the handset or HandsFree. The parameter may range from 0 to 15.
- **Ring volume** – use this parameter to set the loudness of the ring tone. The parameter may range from 0 to 8.
- **HandsFree volume** – use this parameter to set the loudness of the HandsFree. The parameter may range from 0 to 15.
- **Headset volume** – use this parameter to set the loudness of the headset. The parameter may range from 0 to 15.
- **Display contrast** – use this parameter to set the display contrast. The parameter may range from 0 to 7.
- **Time format** – use this parameter to set the time format. Choose either a twenty–four–hour or twelve–hour format.
- **Call list type** – use this option to set the displaying of call records in one of the following formats: **Name and time, Number and time, Name list or Number list**.
- **Call list type** – use this option to set the call list displaying in one of the following formats: **Name and time, Number and time, Name list or Number list**.
- **Phone list type** – use this option to set the phone directory displaying in one of the following formats: **Name list or Name and number**.
- **Message list type** – use this option to set the displaying of received messages in one of the following formats: **Name and time, Number and time, Name list or Number list**.
- **Default tune** – use this option to set the ring tune for the StarPoint key phone. The key phone uses a predefined ring pattern, but this pattern is played with the tune defined in this parameter. The default tune is Cornet Elephant.
- **Hang up timeout** – use this parameter to set the hang–up timeout after the opposite party terminates the call. Having received disconnect, the Cornet port hangs up after the timeout regardless of whether the call is HandsFree or uses a handset. **When used with applications and HandsFree, this timeout should be considerably reduced to approximately 5s.**
- **Message ring tone** – use this option to define the ring tone for the caution on a received message.
- **Intercom ring tone** – use this option to define the ring tone for the caution on an incoming intercom call.
- **Information type at relax** – use this option to set the format of information to be displayed on the second line of the key phone at rest. Choose any of the following options:
  - **Name** – shows the extension name.
  - **Number** – shows the extension number.
  - **Profile** – shows the active profile name.
  - **Name and profile** – shows the extension and active profile names.
  - **Number and profile** – shows the extension number and active profile name.
  - **Name and number** – shows the extension name and number.
- Information type at incoming call – use this option to set the format of information to be displayed on the first line of the key phone at the time of incoming call ringing. Choose one of the following options:
  Waiting for next key – use this option to set the cursor rate for proceeding from

one position to another while typing a text on a StarPoint key phone. Choose one of the seven levels, starting from 'extremely fast' to 'extremely slow'.

- **CLIP** – shows the calling party number (CLI) only.
- **CLIP and CPN** – shows the calling party number (CLI) and originally called party number (original CPN).
- **CLIP and CPN list** – shows the calling party number (CLI) and originally called party number (original CPN). In both cases, the numbers are compared with the phone directories. If a match is found, the name is added to the number.

AoC

The whole tag is available on the group and user levels only.

The **AoC** subtag helps you define the warranted count of call records to be displayed on the key phone and in the NS Assistant. Set the count of **Missed**, **Received** and **Dialled** calls separately. The default count of records is twenty. This limit applies to PBXs with a high number of users (carried calls) only. If a PBX has a capacity to store more records, all of them are displayed at any time.

SMS at no answer

Use the SMS at no answer subtag to set parameters for the **SMS at no answer** functio n. An **SMS at no answer** is a message that is sent automatically to the called user if the following conditions are satisfied:

1. The called party rejects or does not answer the incoming call.
2. The **SMS at no answer** is activated on one of the levels of the fall-down hierarchy for the calling or called user.

The SMS at no answer is sent through the destination defined here. Usually is this destination defined as bundle of virtual GSM ports. Type the SMS text to be sent to the called users in the SMS subtag. You can also add information on the calling number (string %c) and the calling extension name (%n) to the SMS body.

SMS setting:

- **Send SMS at no answer** – send this SMS as an SMS at no answer to the called extension. This setting has a higher priority than that for an external extension.
- **Send SMS at no answer of external extension** – send this SMS as an **SMS at no answer** to the called external extension. This setting has a lower priority than the general one mentioned above.

Moreover, you can define the sending timeout. If you set this timeout to 20s, the outgoing call has to alert the called extension for more than 20s so that the SMS can be sent after the call end. **The timeout is the fourth condition for SMS at no answer sending**.

Examples of SMS at no answer sending:

1. User **A** calls to an external GSM phone via a GSM port of the PBX. The SMS at no answer has been enabled and the timeout is 20s. The GSM phone begins to ring, but the called user does not answer the call. After the timeout (20s), user **A** hangs up (or the call is rejected by the other party). Subsequently, the SMS set in the upper configuration row is sent to the selected destination. If **Origin** is set as the destination, the SMS will be sent via the same GSM port as was used for the call.
2. User **A** calls to an external GSM extension of user **B**. The SMS at no answer is

disabled for User **A** while enabled with no timeout for user **B**. After no answer or call rejection by the external GSM extension, an SMS at no answer is sent containing the text as defined in the bottom row of the configuration of user **B**. When the SMS at no answer is enabled for user **A** too, an SMS at no answer is sent containing the text from the upper row of the configuration of user **A**.

Services

The Services subtag helps you create individual service settings, thus replacing the global ones. You can modify such parameters as progress tones, timers and routers, or activate the PIN request in the service settings. To disable individual settings push the **Remove individual setting** button.

Recording

The whole tag is available on the user, group, virtual port and virtual port type levels only.

You can set the call recording parameters in this tag. Find the tag in the Properties on the user, group, virtual port and virtual port type levels.

- **Recording**– specify the recording mode for the level.
  Pattern length [ms] – set the time for which the recording ON/OFF pattern characters are to be awaited. If a character comes after the timeout, it is identified as invalid (the pattern is not recognised).
  - **Default** – settings from other hierarchical levels are used.
  - **Disabled** – disable call recording regardless of lower level settings.
  - **Upon request** – enable call recording activation/deactivation during the call. The parameters Pattern length, ON pattern and OFF pattern are used.
  - **Turn on at alerting** – call recording is activated by ringing detection and deactivated by the call end.
  - **Turn on at connect** – call recording is activated by the call connection and deactivated by the call end.
- **ON pattern** – define the recording ON pattern during the call. The parameter is applied by PBX extensions only if the **Upon request** recording mode is active. The default pattern is 1*.
- **OFF pattern** – define the recording OFF pattern during the call. The parameter is applied by PBX extensions only if the **Upon request** recording mode is active. The default pattern is 3#.
- **Hang up at unsuccessful recording** – use the parameter to enable call termination in case call recording fails (due to an unavailable storage, invalid licence, etc.).
- The following parameters relate to the saved files with call records. If they are inactive, create new properties of the object used. Thus, you add a row to the Call recording logical storage.
- **Item lifespan [s]** – set the storing time for a file with a call record. After the time expires, items are deleted sequentially. The item lifespan is automatically set to 10 years if this option is not selected.
- **Maximum count of items** – set the count of records in a storage for an object. After the limit is reached, no more calls are recorded unless deleting of oldest files is enabled.
- **Usable space for all items** – define some space in a storage to be reserved for records of the given object. When the space is full, no more calls are recorded unless deleting of oldest files is enabled.

- **Delete oldest after reaching limit** – enable deleting of oldest record files if necessary.

> ⚠ 2N TELEKOMUNIKACE, a.s. shall not be held liable for any recording errors due to unavailable network disks and/or exceeding of the maximum storage capacity.

Customer

The Customer subtag provides parameters for functions that have been implemented for a specific customer and so their meanings will be explained marginally only. This subtag is divided into three sections. Define the supported method of the CPN sending for call billing purposes in the first section. Set the called party identification to be transmitted via a DSS1 message in the remaining parts.

- **Forwarding number** – is used in DSS1 messages for communication with **Nokia** exchanges for billing purposes. Set the **Scheme** (Number or URI), **Subtype** (Unknown, Internal, Local, National, International) and **Number/URI** (specific number or address).
- **Facility** – is used in DSS1 messages for communication with **Ericsson** exchanges for billing purposes. Set the **Scheme** (Number or URI), **Subtype** (Unknown, Internal, Local, National, International) and **Number/URI** (specific number or address).

The number for call billing is sent using this information element. Make sure that the parameter is set for the virtual port too. Choose **Yes, No**, or **Default**.

In addition to DSS1, the parameter is also used in SIP signalling, where it sets the **Diversion** header.

10. Billing and Tariffs

Here is what you can find in this chapter:

- 10.1 Billing and Tariffs

10.1 Billing and Tariffs

This menu describes tariffs offered by network providers. The tariffs are then used for deducting free minutes and SMS messages for virtual ports. In future, the menu should facilitate accounting and least cost routing.

Provider

Add a provider in the left menu column. The item is just a group including all call billing rules.

Context menu options:

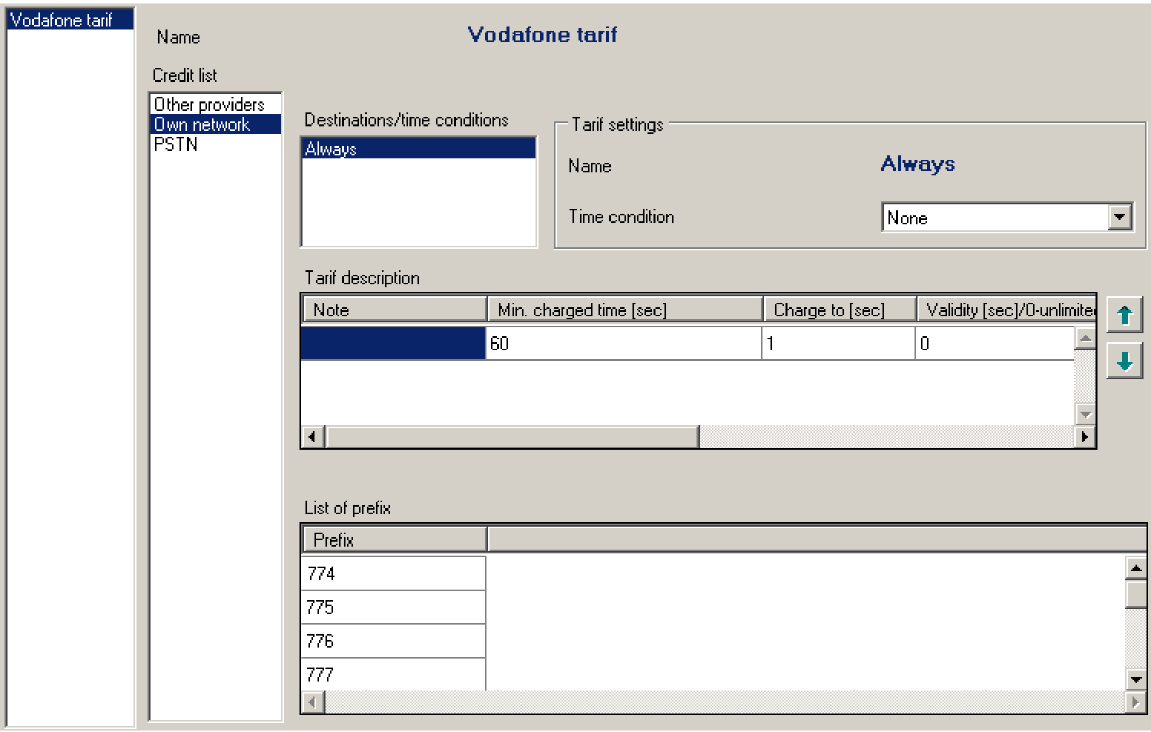
- **Add** – add a provider.
- **Rename** – rename the selected provider.
- **Delete** – remove the selected provider.
- **Default** – reset the pre–defined provider.

Credit list

You can enter any number of credits for each provider and describe each credit with a different set of properties.

Context menu options:

- **Add** – add a credit.
- **Rename** – rename the selected credit.
- **Delete** – remove the selected credit.



Destinations/Time conditions

You can add a destination and time condition to each credit in this section. Destination means the target network to be dialled.

Context menu options:

- **Add** – add a destination.
- **Rename** – rename the selected destinaci.
- **Delete** – remove the selected destinaci.

Tariff setting – Here you can set or change the time condition for the selected destination.

Tariff description

Context menu options:

- **Add** – add a row to the destination describing table.
- **Delete** – remove the selected description table row.

Column description:

- **Note** – for information only.
- **Minimum charged time [s]** – set the minimum call cost. If a call is answered, these seconds are charged to the calling subscriber regardless of the duration of the call. Typically, this value is set to 60s.
- **Charge to [s]** – set the call billing interval after the Minimum charged time elapses.
- **Valid to [s]** – set the end of the interval to which a row applies. For example, if you set 360s, the row will be used for the first 6 minutes only and then the next row will be applied. If you set **0**, the row is valid without limitation.
- **Tariffication impulse time [s]** – use the parameter to set the sending frequency for the tariffication pulses for DDS1-supporting devices. By default, the tariffication pulses are not sent - the parameter value is zero.

Arrows next to the window are used for modifying the row sequence.

Prefix list

Here you can manage the prefixes that relate to the selected destination and credit. If one of these prefixes is dialled via a specific virtual port and the time condition is met, the rules for this credit are used for billing.

Context menu options:

- **Add** – add a row to the prefix table. Matching prefixes are highlighted.
- **Delete** – remove the selected prefix table row.

Column description:

- **Prefix** – this item shows the prefixes that are assigned to the selected credit destination. If * is included, the rule applies to all prefixes.

11. Configuration Examples

Here is what you can find in this chapter:

- 11.1 Other Useful Information
- 11.2 Mobility Extension Configuration
- 11.3 NetStar Installation Guide

11.1 Other Useful Information

COM port and communication program setting

Basic equipment of OS Miscrosoft Windows, Hyperterminal application, is used for connection. The whole setting of this application is shown in the figure below.

Console setting

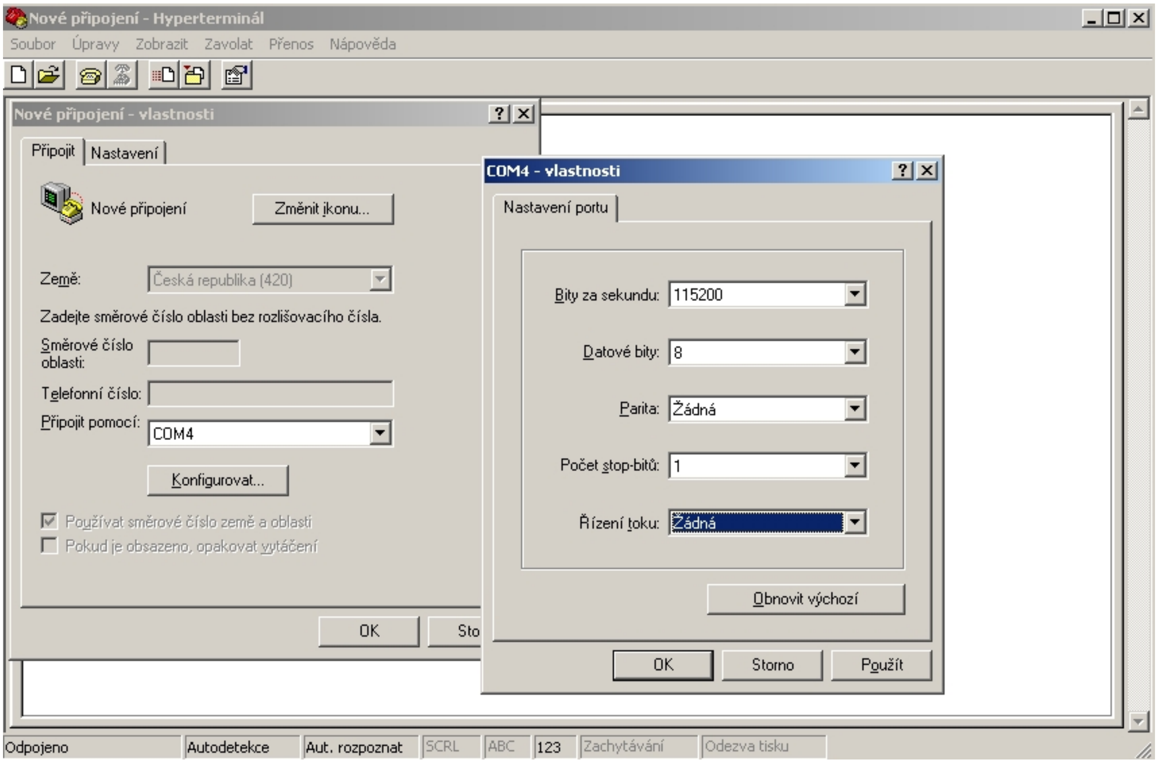


Figure: View of Hyperterminal Application Settings

Console structure

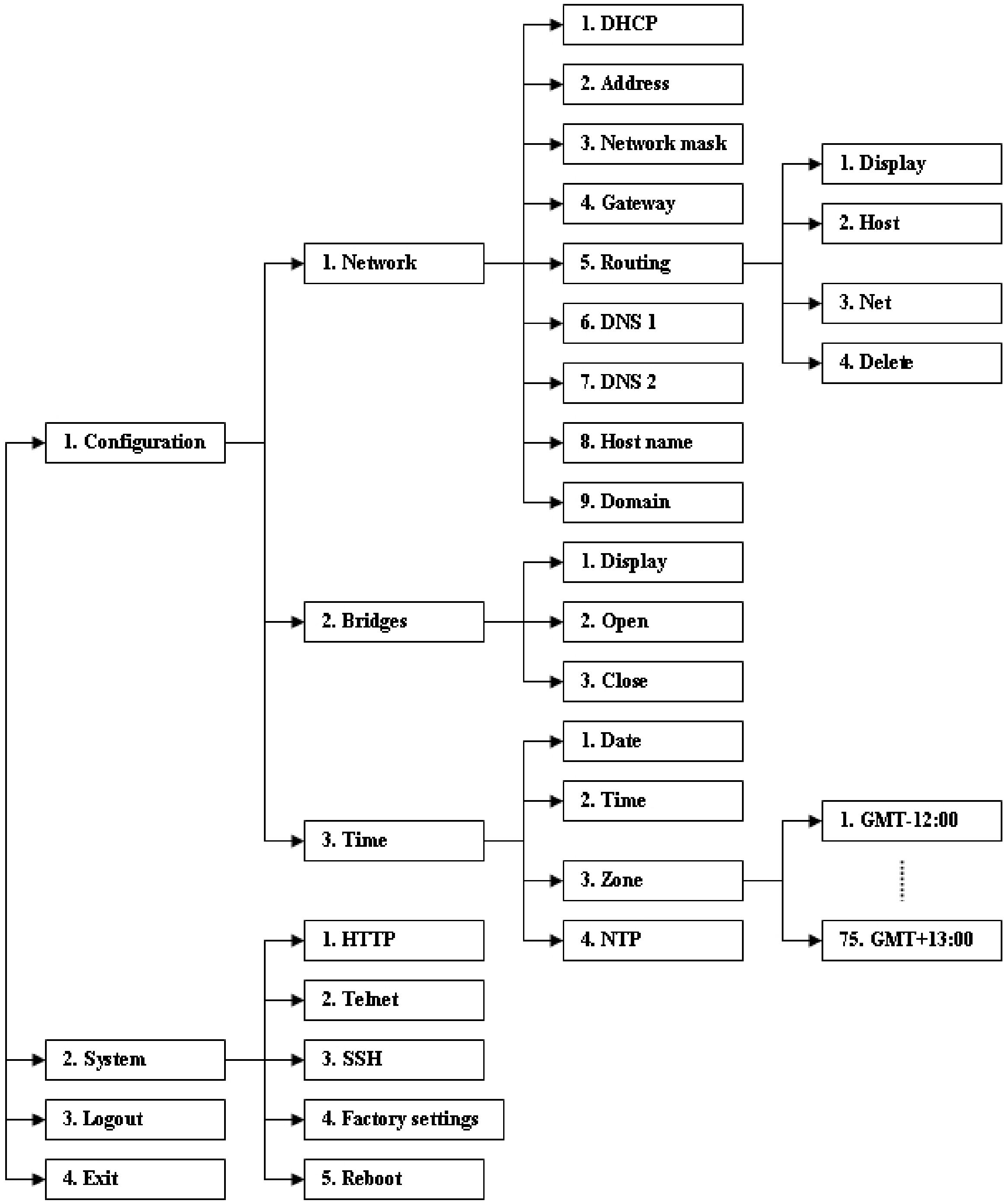


Figure: View of Console Structure for Easier Orientation

11.2 Mobility Extension Configuration

Mobility Extension

Mobility Extension is an extension feature of the **2N® Netstar** PBX, which enables external extensions to use also features that are not normally available as well as practically all PBX services. The Mobility Extension feature is connected with the existence of external extensions. Before you start creating external extensions and configuring their routing make sure that you have a valid licence and if so, for how many extensions the licence is valid. This can be verified in the **Global data – Licences** menu. Here, in the lower–right part of the screen you will find the licence tables. In a row called "Mobility Extension user" you will see the number of licences owned (third column) as well as the number of licences required by the PBX (last column). The last column shows not only the external extensions created but also the parameter called "Transfer" in the "Properties" tag on the extension and user levels. If you set this parameter to "Yes", only one licence is required for the extension, and the number of licences required for the user is the same as the number of extensions it has (external extensions are counted only once).

External Extension Creation

In principle, an external extension can be created in three ways. The first one is creating an external extension when adding a user. This situation is shown in **Figure 1**. This way, an external GSM or PSTN extension can be created for the user. A conventional GSM extension (cellular telephone) is configured in the GSM part and a normal fixed line of a public network subscriber is configured in the PSTN part. In the case of GSM external extension, the text "GSM" is automatically added after the name, and in the case of PSTN external extensions it is the text "PSTN". If you wish to forward SMS messages delivered to the user to the cellular telephone of an external extension, check the option "Resend SMS". Configuration of SMS routing is described in the text below.

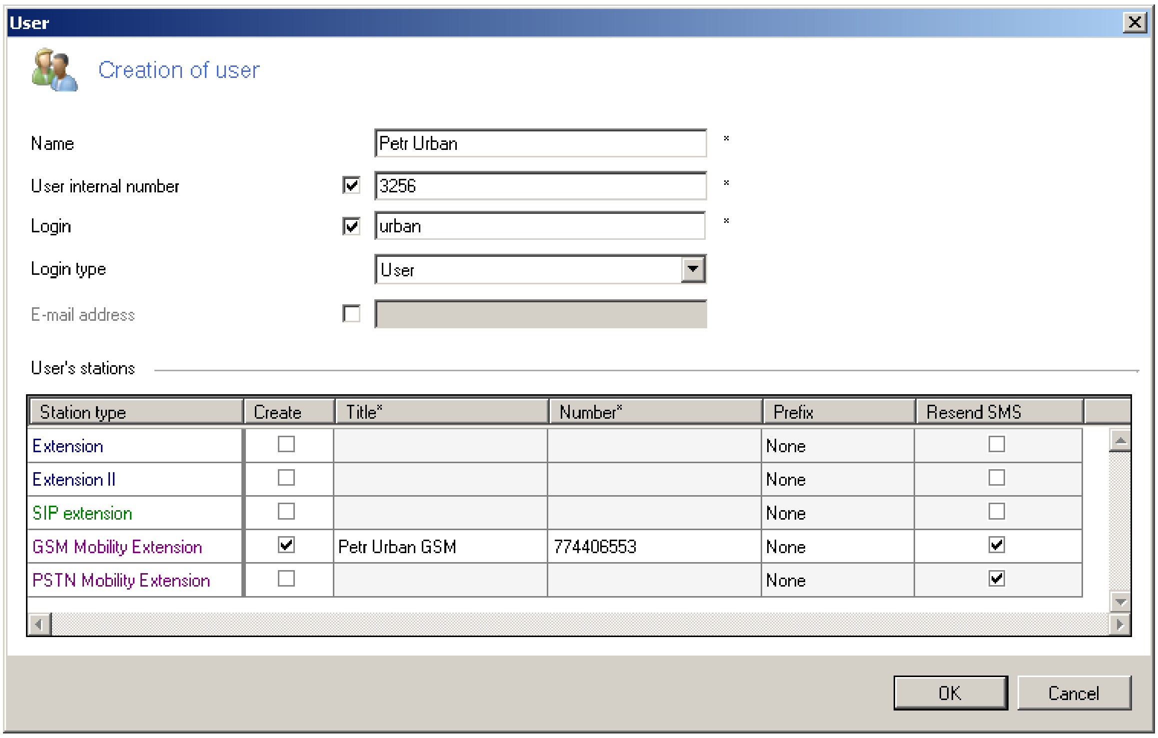


Figure 1: Creating External Extension by Adding a User

The second way to create an external extension is to create an external extension and then assign it to a specific user (an external extension may not exist without its user). This can be done in the **Users – Extensions – External** menu, where, via the context menu, you select Add station and, having completed the name and number, assigned the user and, if necessary, ticked off the SMS resending option, press OK to confirm station creation. Refer to Figure 2 for details.

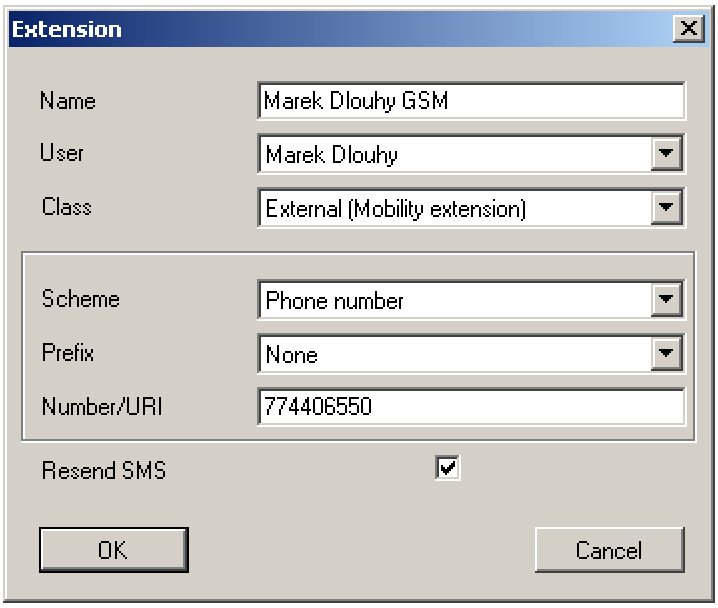


Figure 2: Dodatečné vytvoření externí stanice uživateli

The third and last way is to create an external station over the user. This can be done in the **Users – Users & Groups** menu, where you, having selected a user, open the context menu and select **Add station**. Complete the station name and number and, if necessary, enable SMS resending in the dialogue window. Again, refer to Figure 2 for details.

Having added an external station and saved the configuration, check this external station in the **Required licences** section of the **Basic** tag on the station level for a valid lincence. Refer to the **Global data - Licences** menu for the total count of licences for the Mobility Extension users.

Routing Incoming Calls with Mobility Extension

What is necessary:

1. external extension
2. routing "From port"
3. DISA direct inward dialing
4. permitted transmission for putting a call on hold

Routing of incoming calls of external extensions is associated with the recognition of these calls immediately after the call has arrived at one of the ports of the **2N® Netstar** PBX. Recognition is based on compliance of CLIP (caller identification) with its subtype. Further routing is then governed by the settings at the recognized external extension ("Routing" tag in the extension properties). This tag is shown in **Figure 3**. T he From port item relates to incoming call routing. The item consists of two sections: Normal and Services and on-hold. The earlier sets the destination for primary incoming call routing and the latter is used for call hold or access to a service without a destination of its own. For easier understanding the annex (**Annex 1**) shows a flow chart of processing of an incoming call of an external extension.

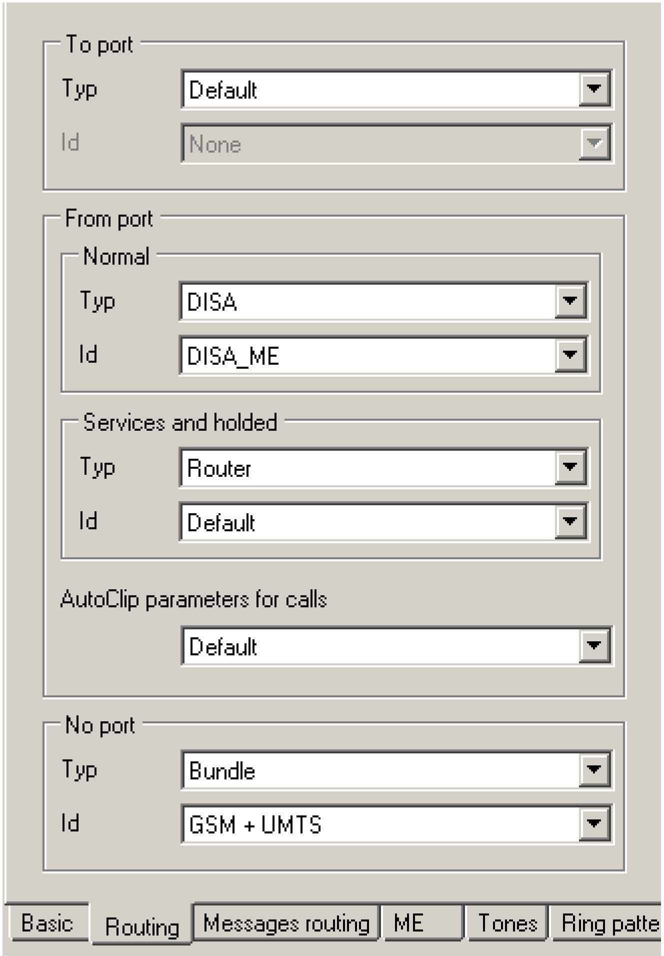


Figure 3: View of external extension "Routing" Tag

As shown in Figure 3, incoming calls from an external station are routed to DISA. Specific settings of the DISA direct inward dialling are shown in **Figure 4**. With this configuration, the incoming call is routed to the Default router and then a 10-second dialling timeout follows, which is then detected by the DTMF detector (its inclusion requires checking the DTMF option in the lower part). If the 10-second time limit expires without detecting the dialling, the call is then routed to the Operator extension.

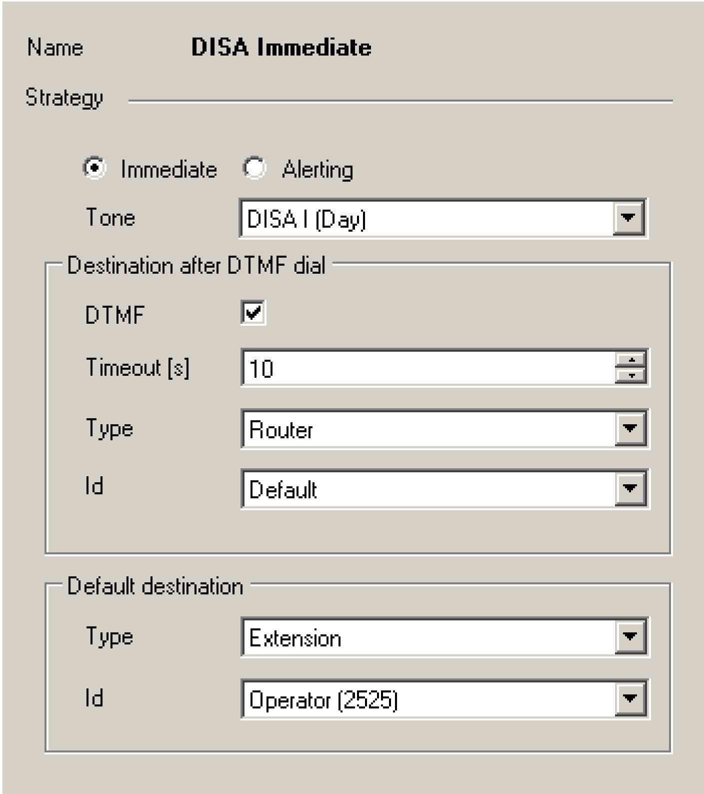


Figure 4: View of Configuration of DISA for Mobility Extension

The Default router usually gives the user a much broader scope of operation than the above–mentioned Internal router because it is one level higher in the hierachical structure. It allows the user to call internal extensions, use the services and also call public networks. Restriction of calling international numbers can be achieved for example by including an authorisation router.

To distinguish external station rights, simply create several DISA services and routers and assign a different router to each group with identical rights in the DISA to define the possibilities of the calling subscriber.

After the call has been made, it is possible to put it on hold any time. In order to access this function it is necessary to permit it at one of the hierachical levels. Permission is made in the "ME" tag in the Properties (**Figure 5**). Flash pattern serves for putting a call on hold and for switching between active calls. Disconnect pattern serves for terminating one of two active calls and for returning to the other one. The Pattern time interval specifies the possible delay between the entering of the characters of one pattern (time between pressing 7 and *). If the delay is longer than the set length, the entered pattern will be evaluated as invalid.

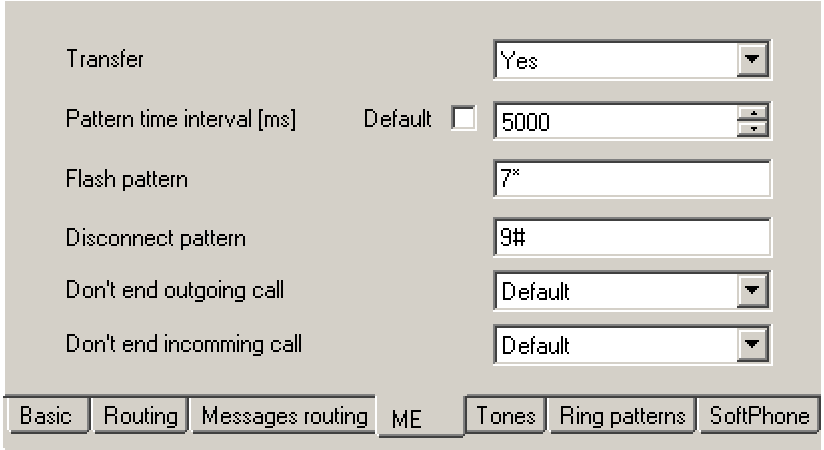


Figure 5: Common Configuration of "ME" Tag with Enabled Transmission of Dialling of External Extension

What is necessary:Routing Outgoing Calls with Mobility Extension

1. an external extension;
2. "No port" routing;
3. "From port" routing;
4. a bundle of ports;
5. permitted transmission for holding a call.

Routing of outgoing calls to an external extension mainly depends on the configuration of this extension and on the way it is called. In principle, an external extension can be called in two ways. The first is to call a number, which is then routed to the respective external extension. The second way is to call a user of this extension. In such case, it is necessary to uncheck the option "Do not ring when calling a user" in the "Basic" tag at the extension, as shown in **Figure 6**.

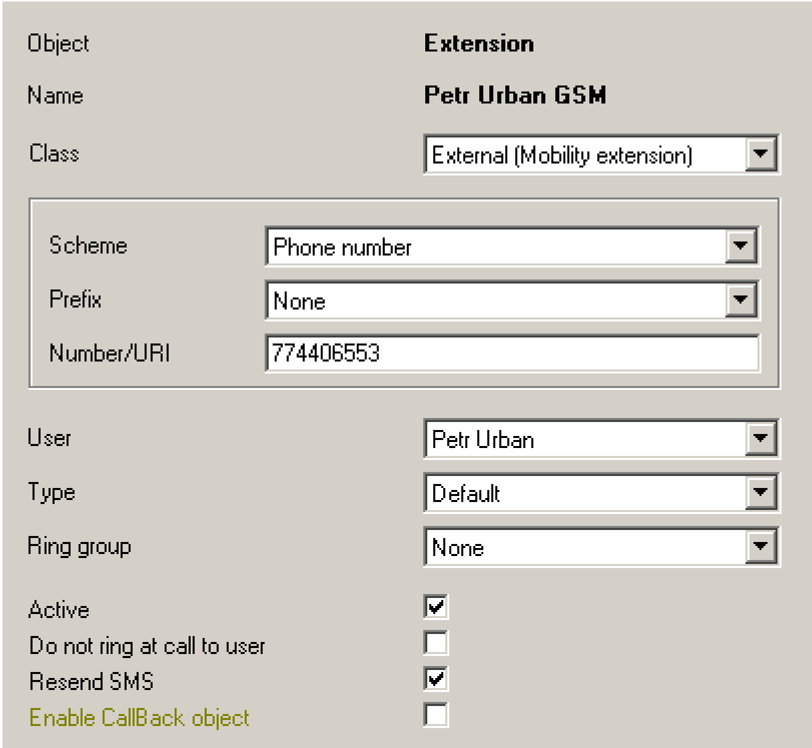


Figure 6: Part of Configuration of External Extension with Number Used for Routing

The port over which the call will be made determines the setup of routing "Without port" in the "Routing" tag in the extension properties (or, as the case may be, the type of the extension when using a mass configuration by the fall–down structure). **Figure 3** shows a suitable solution of routing of an outgoing call of the external extension. Inclusion of a bundle of GSM ports in the routing "Without port" reduces the probability of rejection of a call at the respective external extension when one GSM port is busy at the moment.

One of the possible settings of the bundle is shown in **Figure 7**. The selected cyclic allocation strategy means that at the first attempt to call one of the external extensions, which has this bundle set in the routing "Without port", it will be routed to the first row of the bundle, i.e., to port GSM 1. At the second call to port GSM 2, subsequently to port GSM 3, and at the next making of a call again to port GSM 1. In the event that the specific port is busy at the moment of routing, it will automatically go to the next row. This is ensured by checking the "Next row if is called busy" option. In case all the ports are busy, the call can be routed ot the default destination used as a back-up route.

The By credit option is another suitable strategy for this case. This strategy is intended for credit-monitored bundles with virtual ports. An incoming call is routed to the virtual port of the bundle with the currently highest count of free minutes. If there are more rows with the same count of free minutes, the sequence of rows in the bundle is respected. If a row object is busy or unavailable, the call is routed to the next row, or terminated.

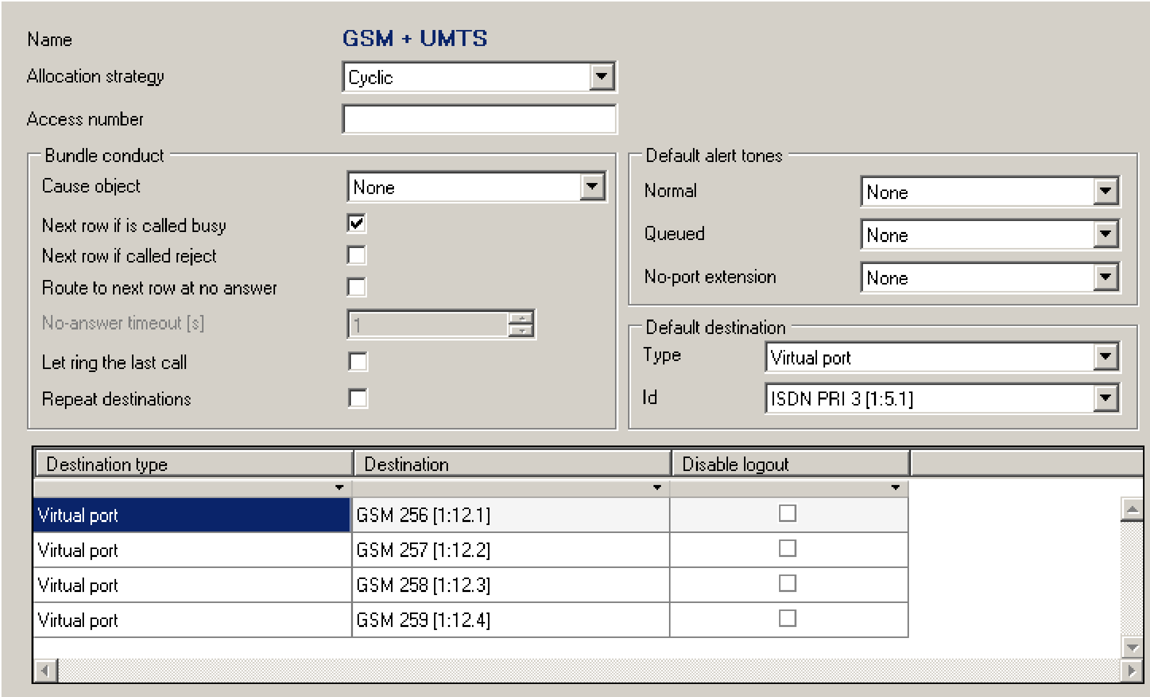


Figure 7: Typical Configuration of Bundle of GSM Ports for Outgoing Routing of External Extension

What is necessary:SMS at No Answer

1. an external extension;
2. "SMS at no answer" setting in Properties.

These SMSs are used for information on missed calls. Make sure that the SMS at no answer tag in the Properties is set correctly on one of the hierarchical levels to send the SMS successfully. Refer to **Figure 8** for a typical configuration. As you can see, the configuration is divided into two parts. The first part represents configuration for SMS sent by the PBX to the counterparty when the call initiated by an internal or external extension of the PBX with these settings is not answered or is rejected. The other part of configuration is used for notification of a missed or rejected call of the external extension from the PBX, which was initiated over the GSM network.

You can enable or disable the SMS and set the minimum alerting timeout after which the SMS is sent if the counterparty fails to answer the call. The two SMS configuration parts can be active at the same time as shown in Figure 8. If two SMS messages should be sent at one moment, the SMS from the first part will be preferred (for an unanswered incoming call). The SMS will be sent via the selected destination, which is a GSM port or a GSM port bundle typically. If you set **Origin** as the destination, the SMS will be sent directly via the GSM port over which the unsuccessful call setup was attempted.

The possibility to use the %n and %c strings for caller identification is very useful. The %c string inserts the caller's number (CLIP) and the %n string inserts the name as specified in the respective telephone directory.

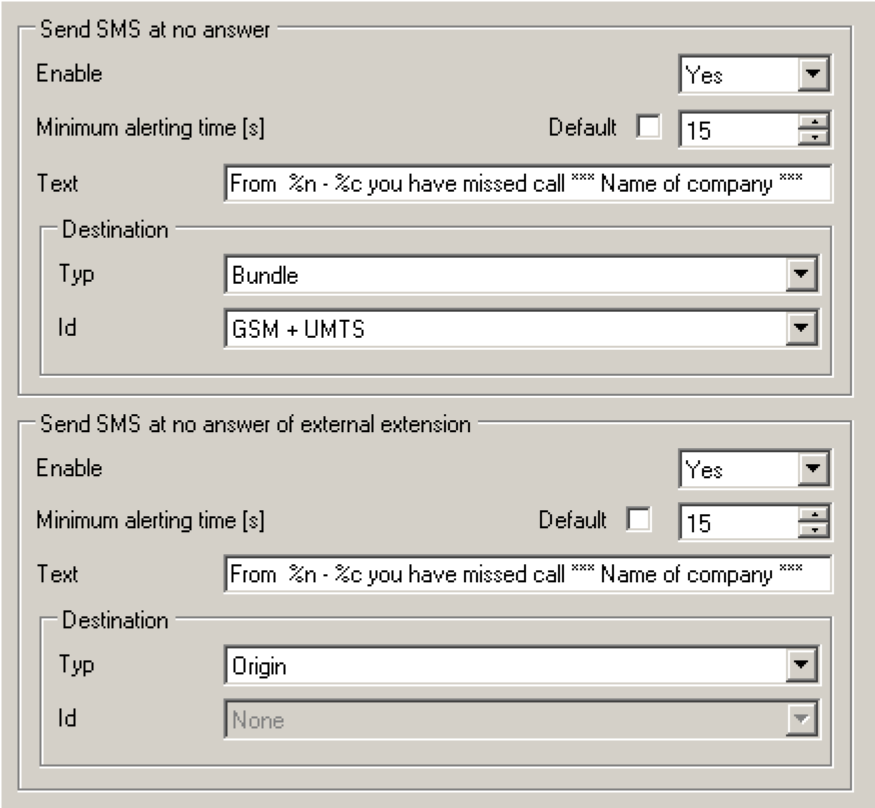


Figure 8: Typical Settings for Sending SMS at No Answer

What is necessary:Outgoing SMS to External Extension

1. an external extension;
2. "No Port" Messages routing

Outgoing SMS are routed according to the Message routing tag in the Properties on one of the hierarchical levels (mostly Group, User or Station) in 2N® NetStar. This tag, together with the typical settings, is shown in **Figure 9**. The part of configuration marked as "No port" is used for routing or redirecting of SMS at an external extension. Here a specific port is set over which the SMS will be sent to the routing number of the external extension, but it is also possible to set a bundle of ports.

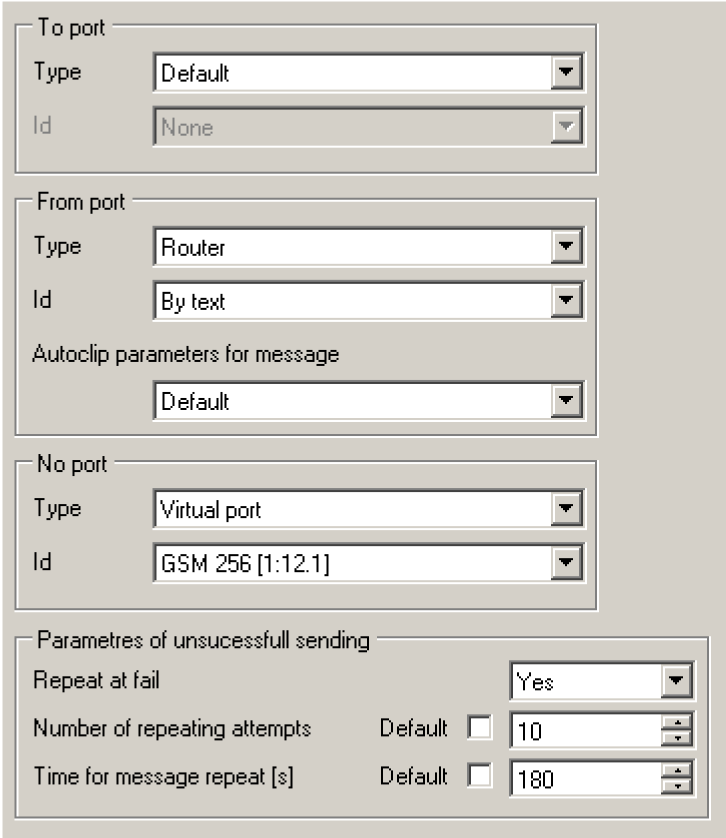
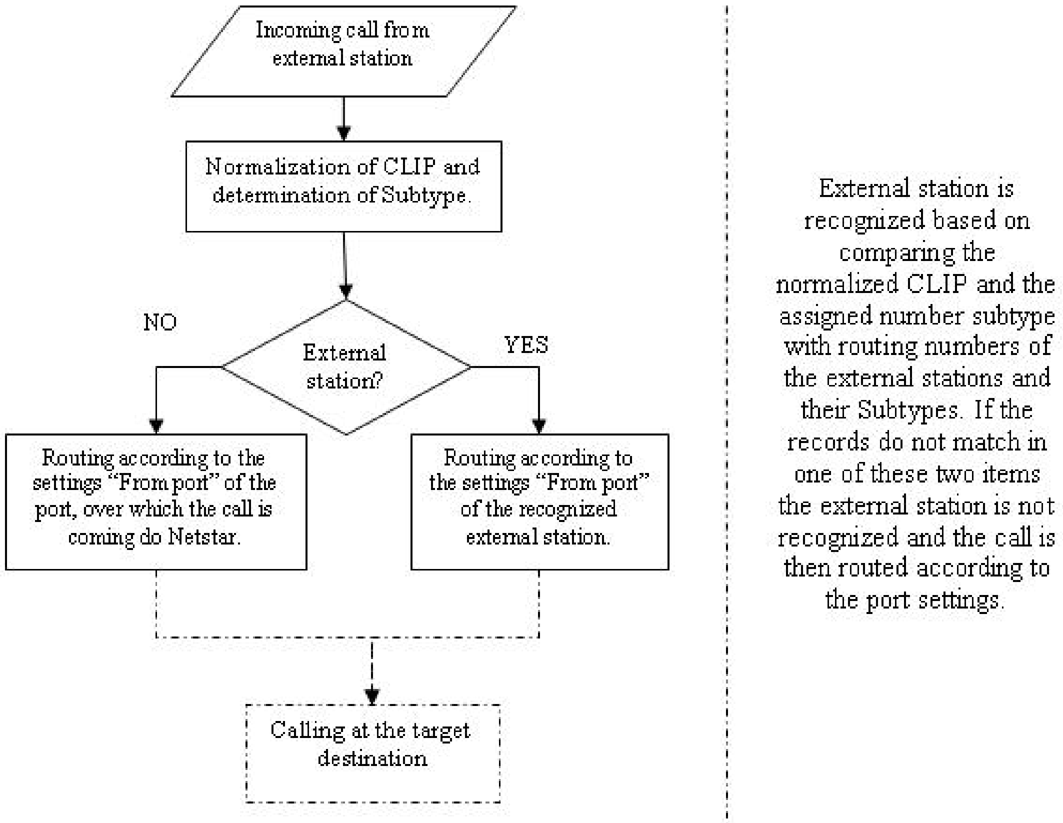


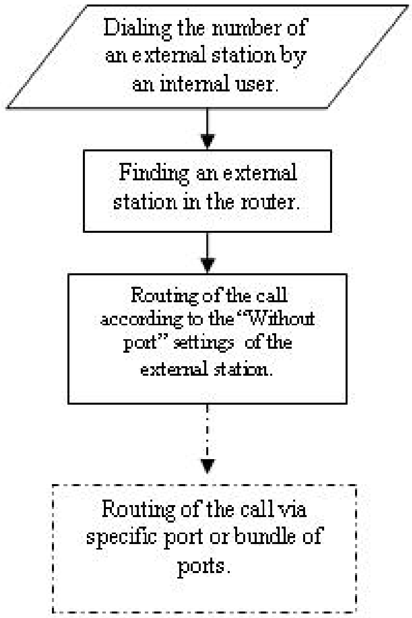
Figure 9: View of Typical Settings in "Messages routing" Tag for User with External Extensio

The condition of a correct function of SMS sending to an external extension is the checked "Resend SMS" option in the configuration. The flow chart for SMS sending to an external extension is included in **Annex 3**. The procedure of forwarding of SMS received by the user at the external extension is shown in **Annex 4**.

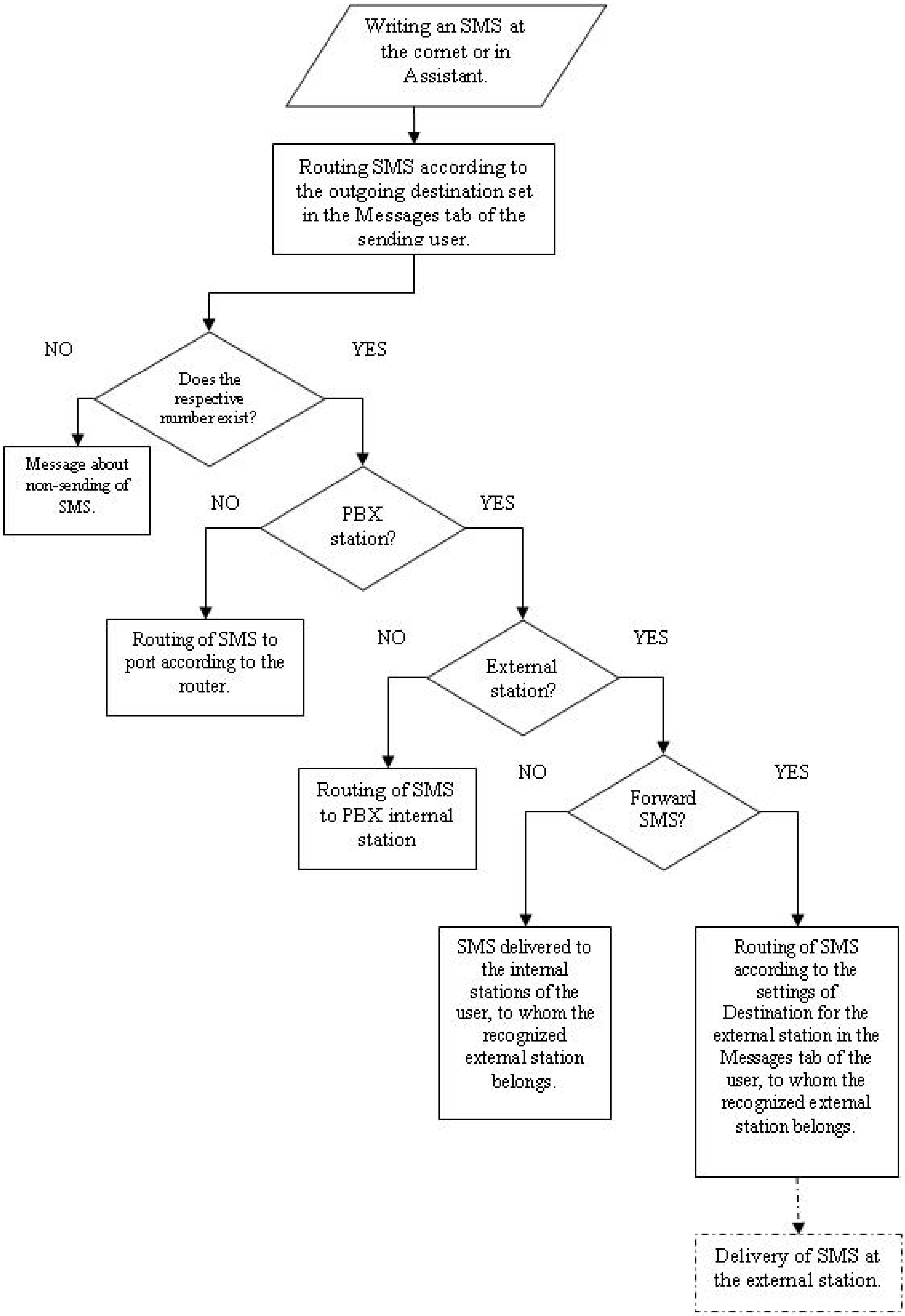
Appendix



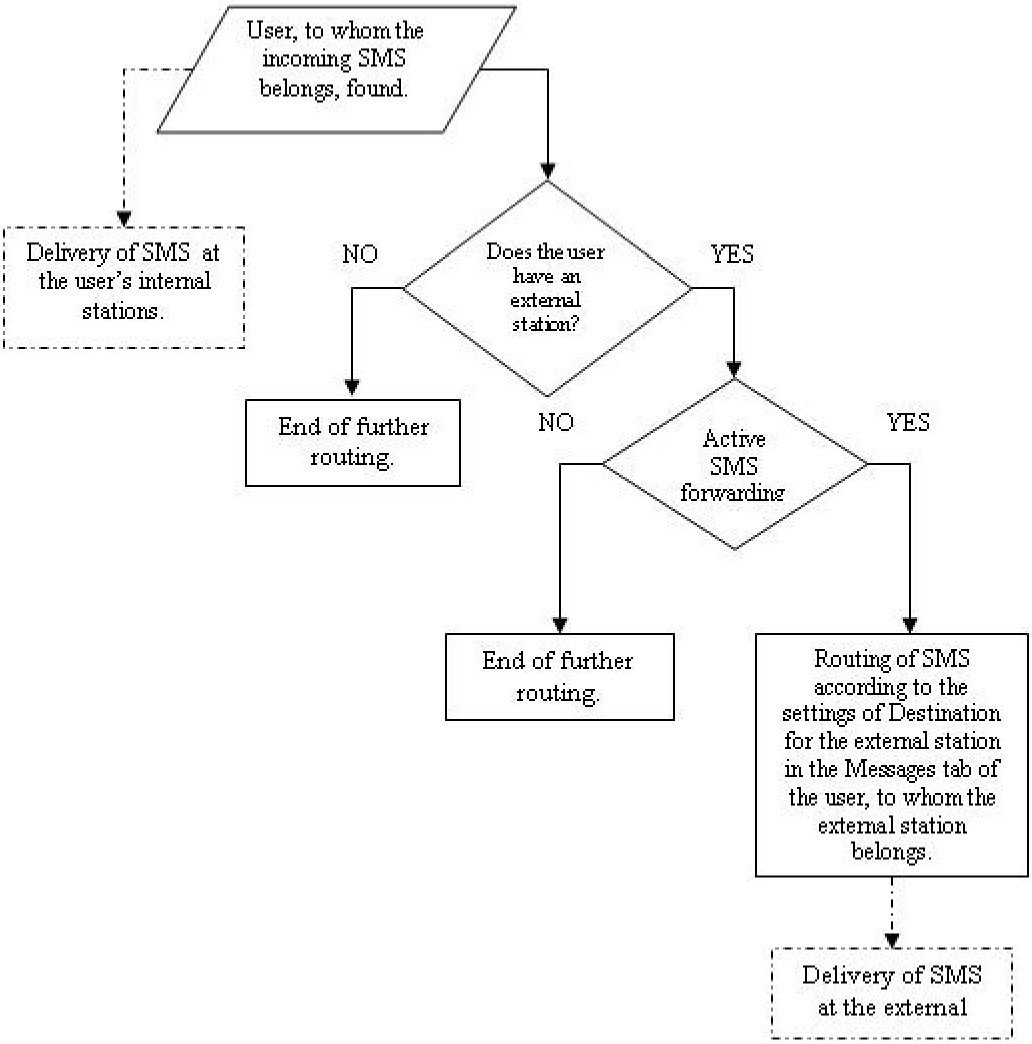
Annex 1: Flow chart showing the processes for an incoming call from an external extension



Annex 2: Flow chart showing the processes for an outgoing call to an external extension



Annex 3: Flow chart showing the processes for sending SMS to an external extension



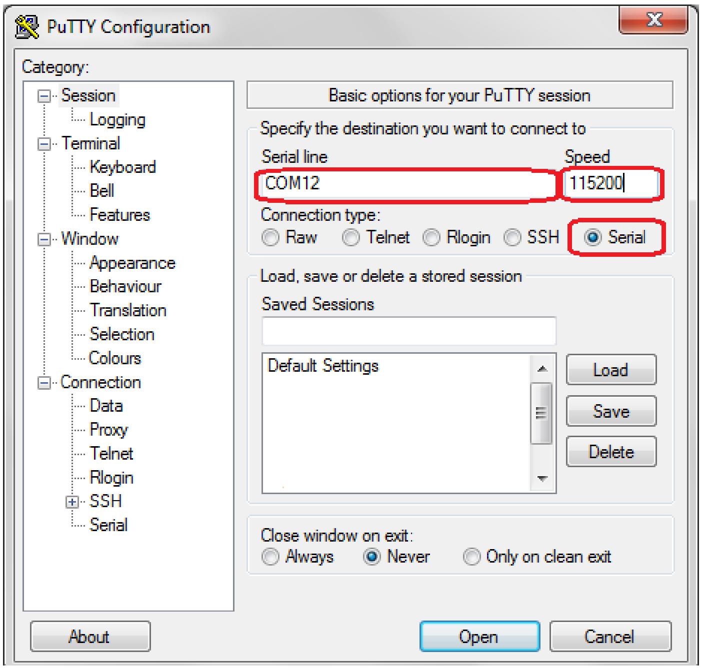
Annex4: Flow chart showing the processes for forwarding SMS to external extension

11.3 NetStar Installation Guide

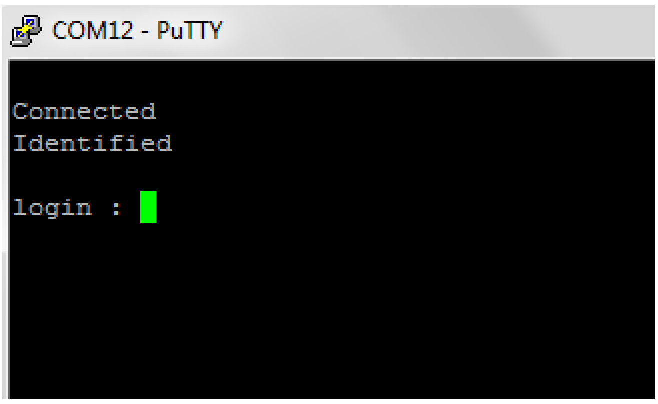
Setting IP Address and Time

IP address

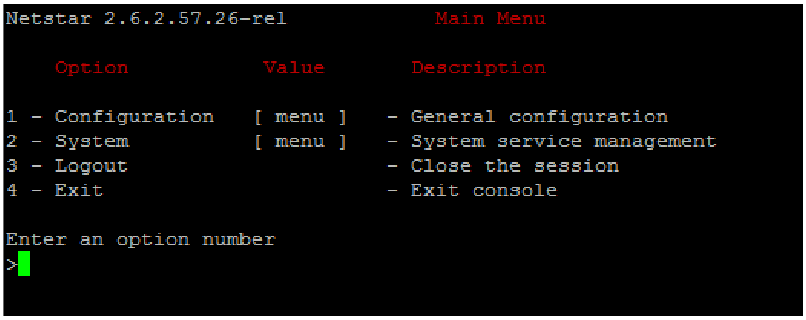
- Connect to NetStar with HyperTerminal tool
  - Rate: 115200
  - Flow control: None
- You can also use Putty
  - Rate: 115200
  - Serial line: set the COM interface number you are using for connection between your PC and NetStar



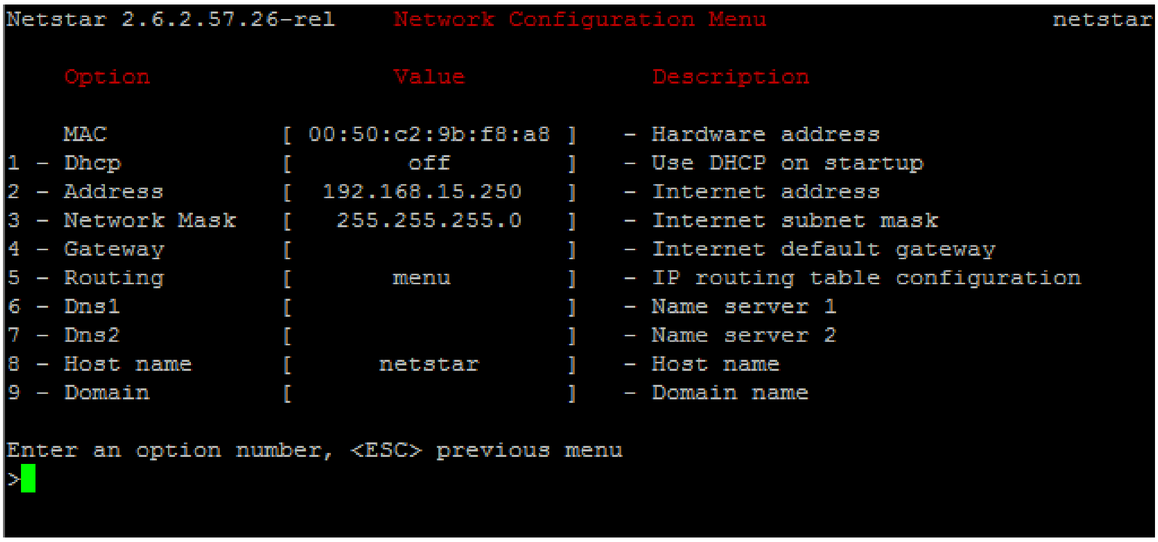
Once you are connected, press <ENTER> to get the login screen. The default login information is **Admin** and **2n**. If there is a # character and no login request on the screen, type **NsCon** and push Enter for confirmation. The login request should get displayed.



Having inserted correct login information, you will get the initial configuration screen.



By pressing the proper digits you will get to the configuration menus. Press 1 and 1 for IP configuration.

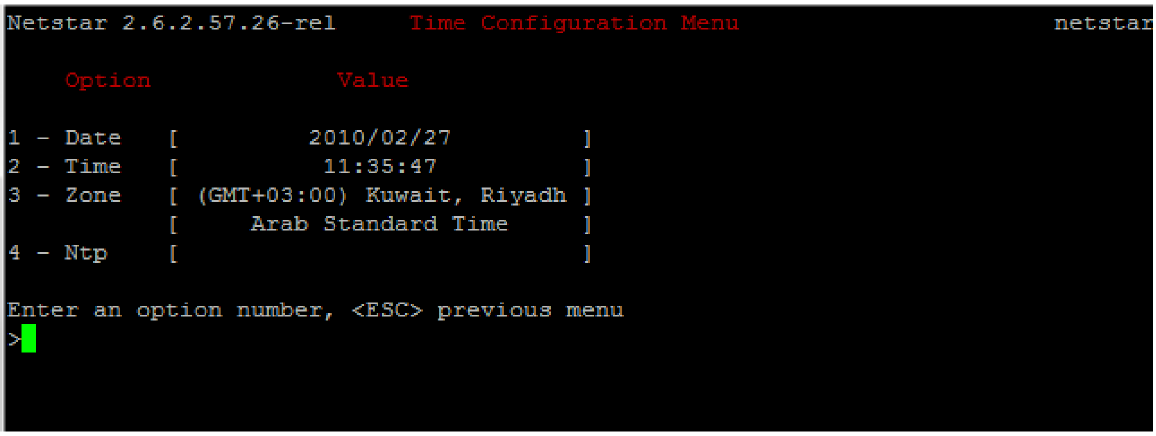


The IP configuration screen gets displayed. Press options 2 to 4 for IP setting. To escape from the menu or cancel the current operation, use the <ESC> key. Having completed all settings, push <ESC> twice to get to the default menu.

Time

When you are in the initial screen for time configuration, press 1 and 3. You will get into the **Time setting** menu. For correct configuration, push 3 to set the time zone and then enter the number according to your location. Then you can modify time and date using options 1 and 2.

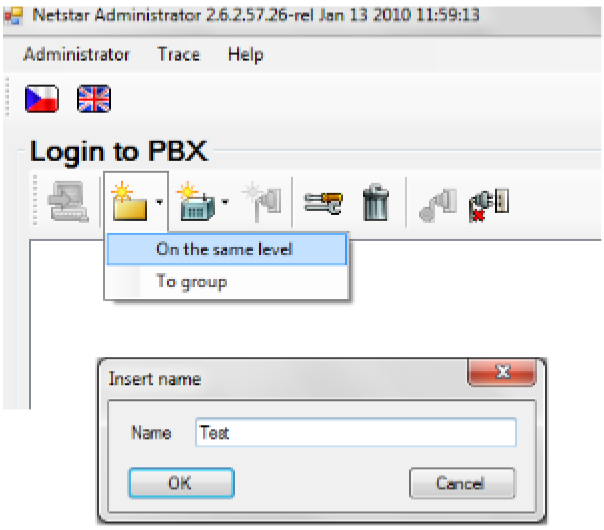
You can also use an NTP server is available on the LAN. After pressing option 4 set the IP address or domain name of the NTP server (one DNS at least has to be set in the IP settings).



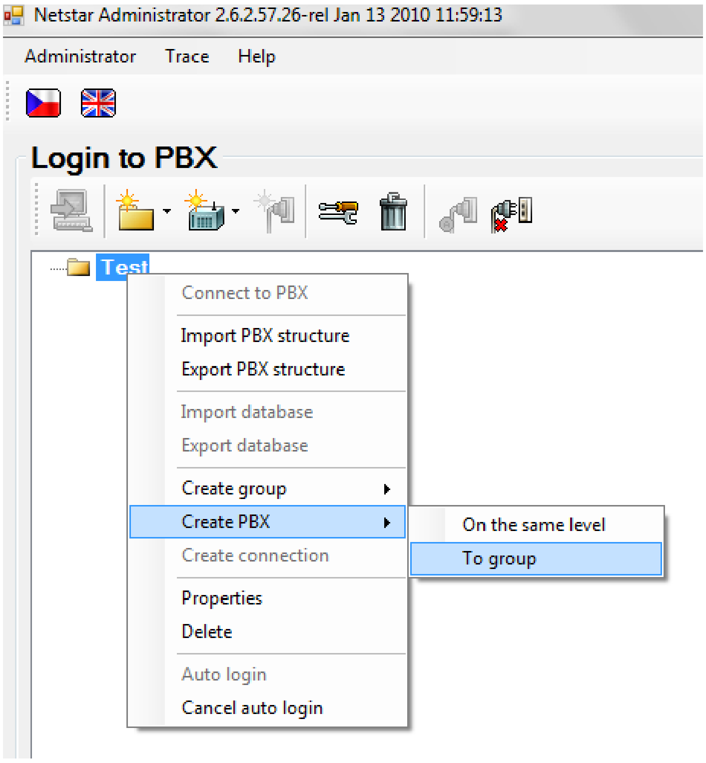
Connection of Configuration Tool to NetStar

Start the NetStar configuration tool. In case no connection to NetStar has been created, create a new one.

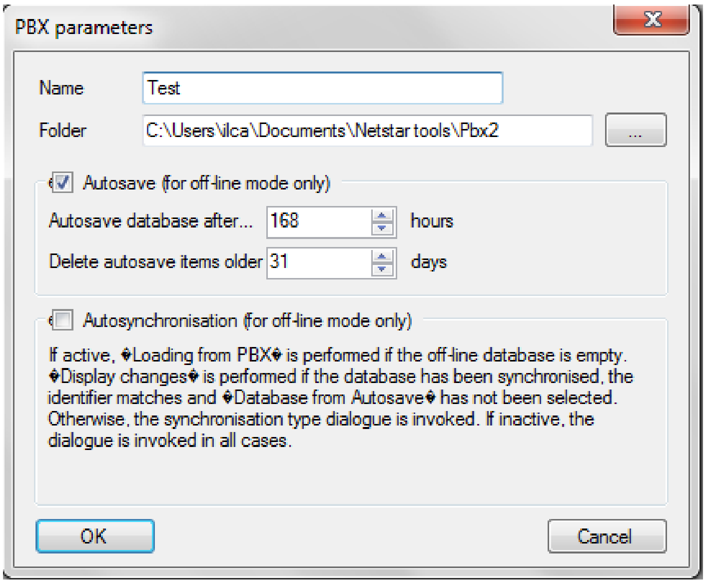
Create a new group for the customer by choosing the Folder icon and the **On the same level** option and name the new group. In our case it is called **Test**.



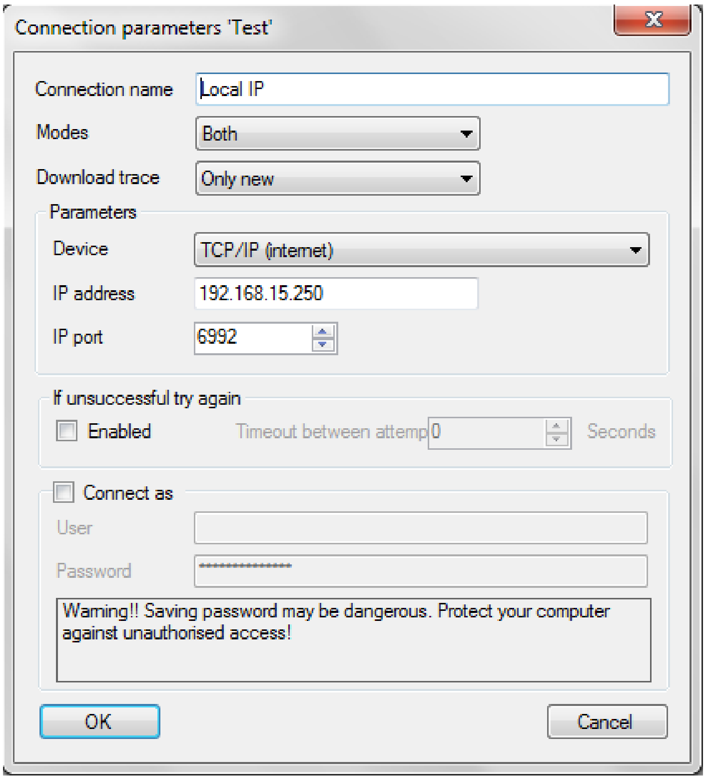
Once a new group is created, right click on it and go to the **Create PBX – To group**



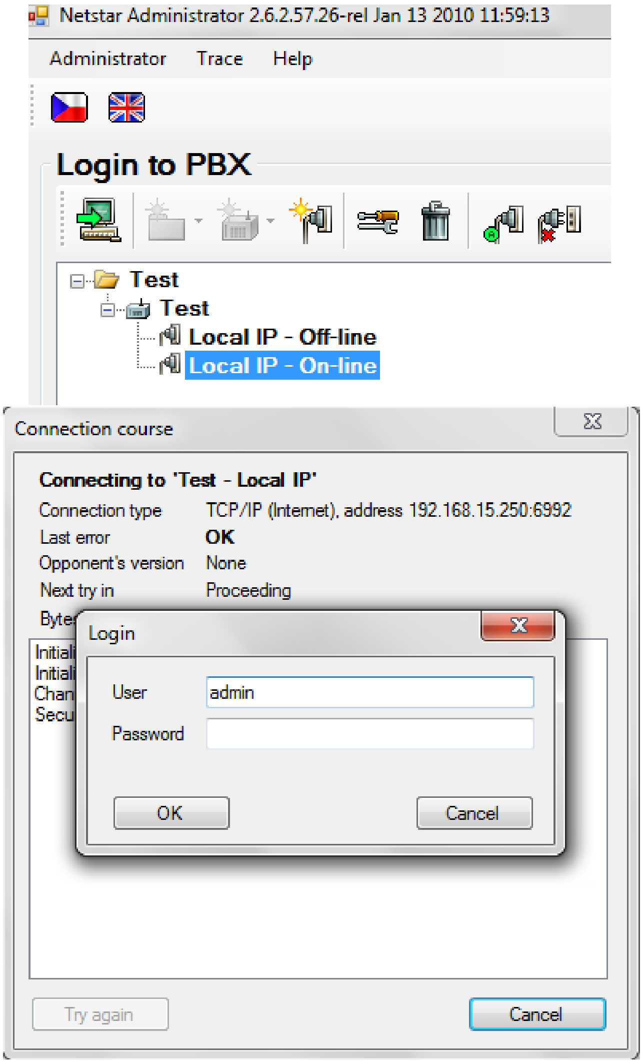
and create a new PBX called **Test**.



Push OK to open the IP setting screen. Set any name you want. In our case we use **Local IP** to mark that the local IP will be used. Fill in the IP address into the IP address field that you set for NetStar in the first step.

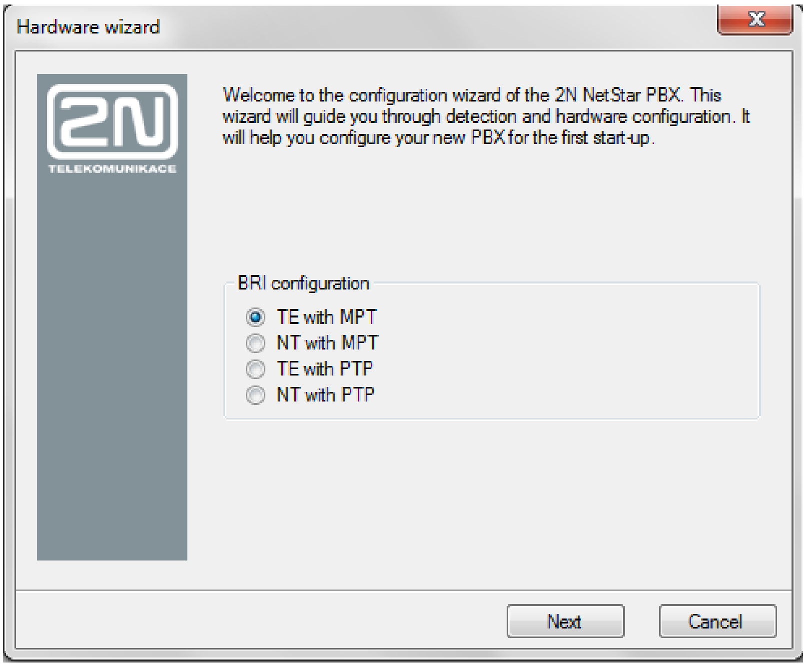


After all the steps are finished, you will create the connection to NetStar. To get connected, just double click on the option with the On-line text at the end of the line. Before connection you will be asked to enter your user name and password.



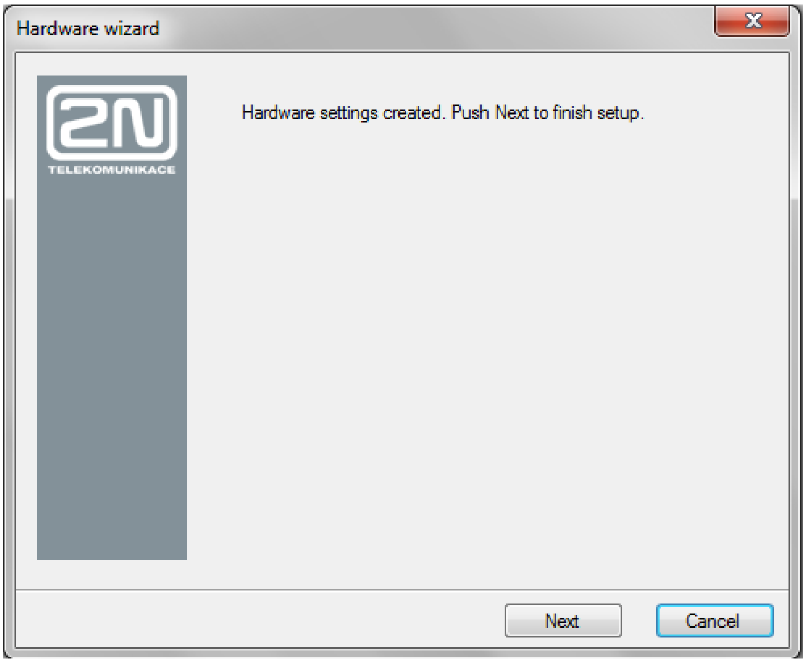
Configuration Wizard

The aim of the configuration wizard is to provide you with an easy basic installation. The ISDN BRI parameters are specified during configuration (click on Next not to use ISDN BRI).

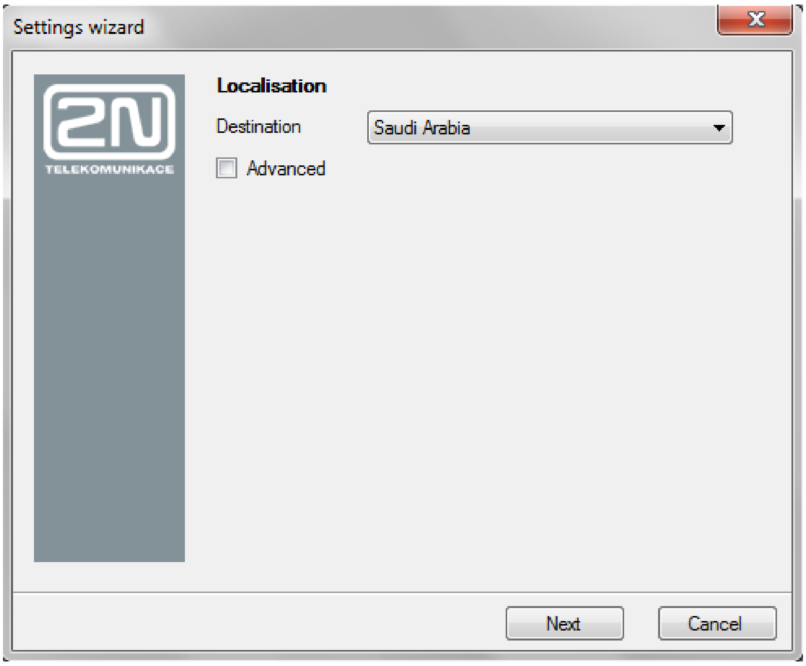


Then the hardware is activated. When the activation is finished, you will get the screen below. Please note that hardware activation can take more than 5 minutes depending on the hardware configuration used.

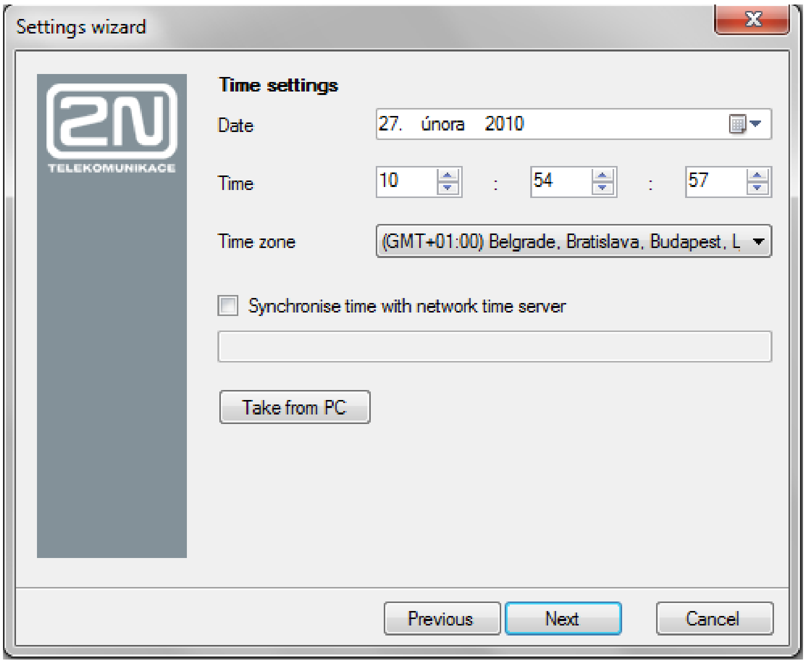
When the hardware detection is finished, click on **Next** to continue.



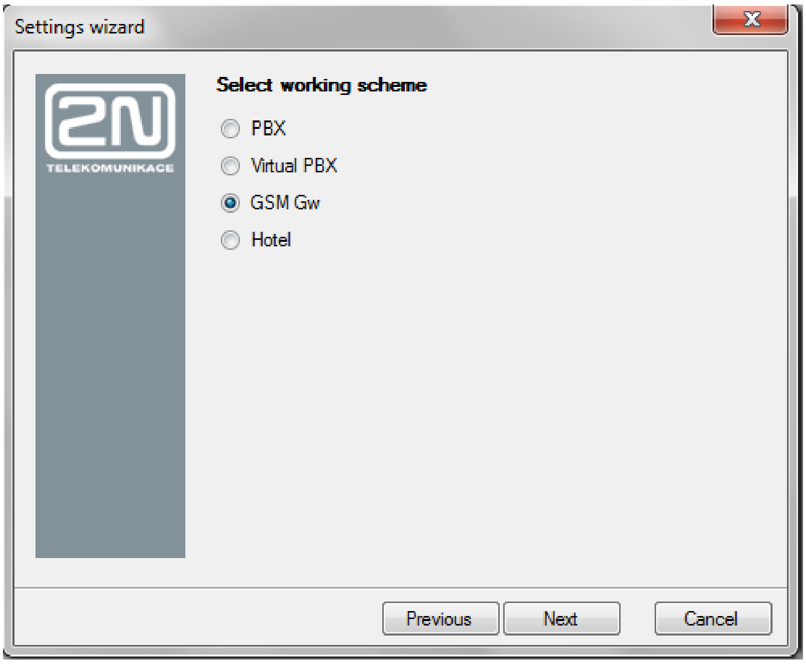
After the hardware is activated, the wizard will guide you through the basic gateway configuration settings like localisation, where you have to choose the country where NetStar will be installed,



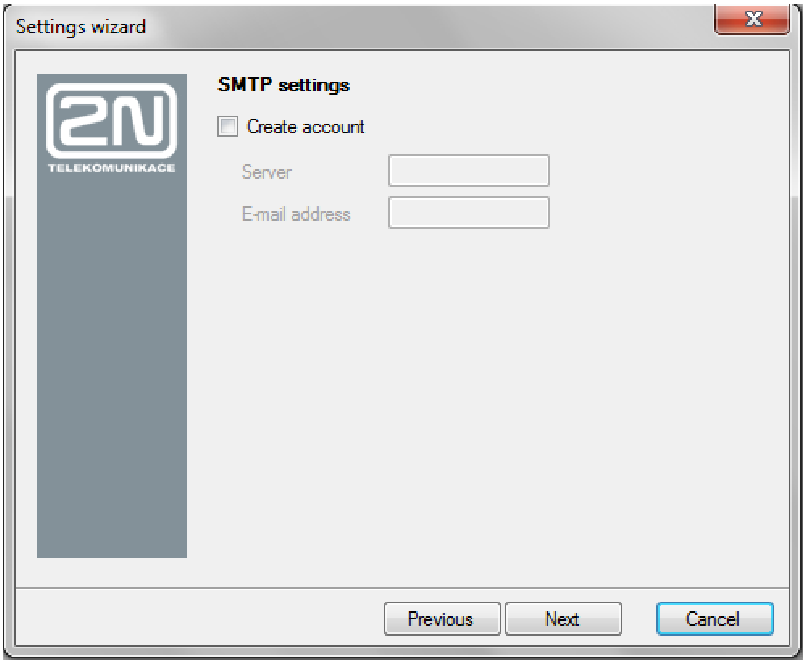
time zone settings



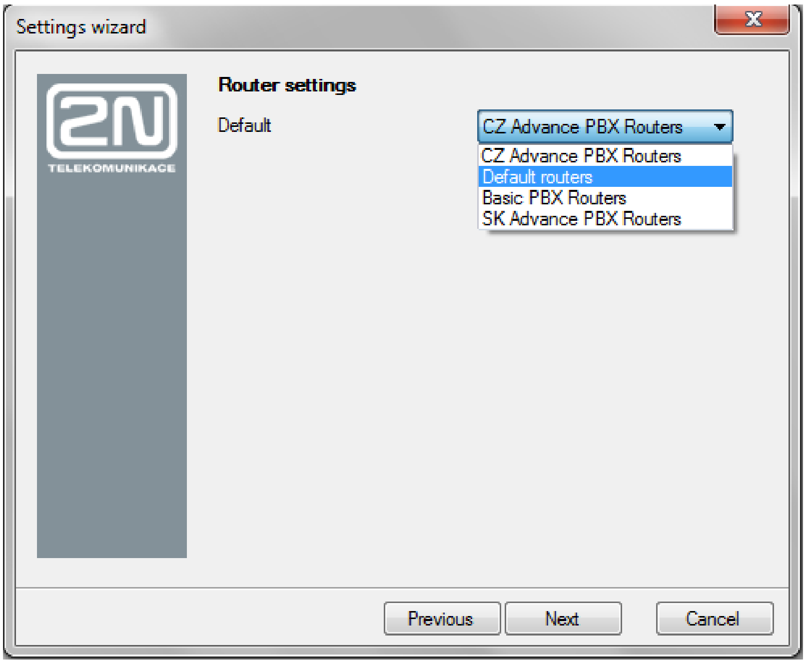
and purpose of the NetStar. Here choose the GSM GW option.



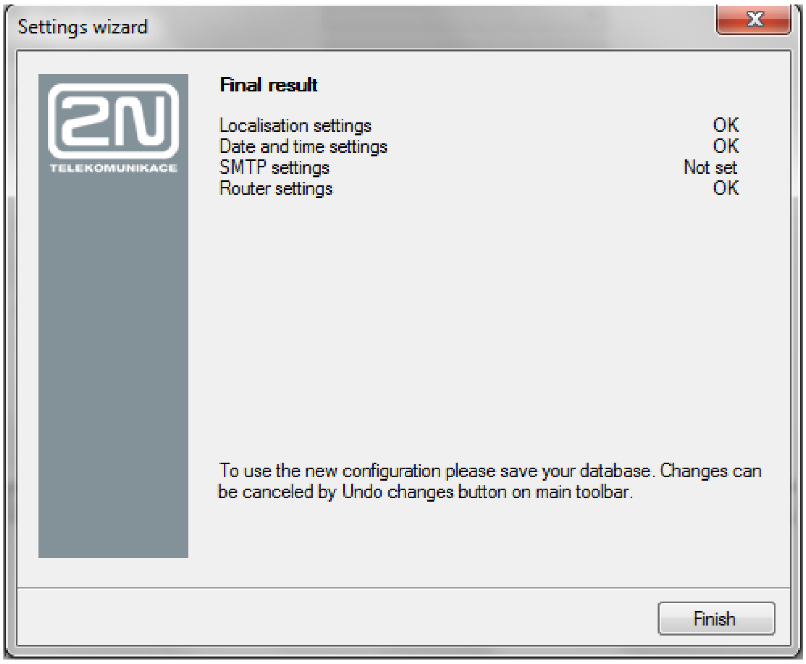
When asked for SMTP settings, choose **Next**.



The last screen will ask you for Router settings. Here choose the preferred LCR structure. In our case it will be Default routers. Then choose **Next**.



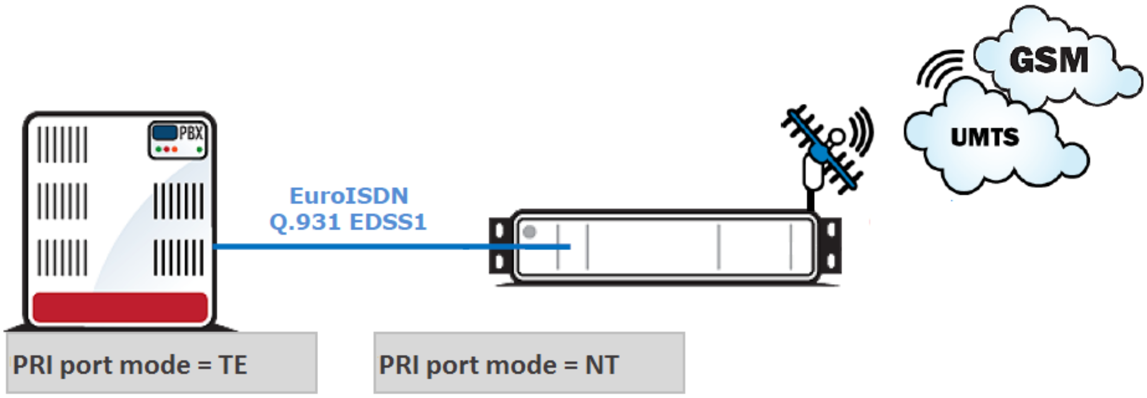
When you get to the final overview, press **Finish** to get to the configuration interface.



To apply the configuration created by the wizard scrip, save the changes to NetStar using the saving icon.

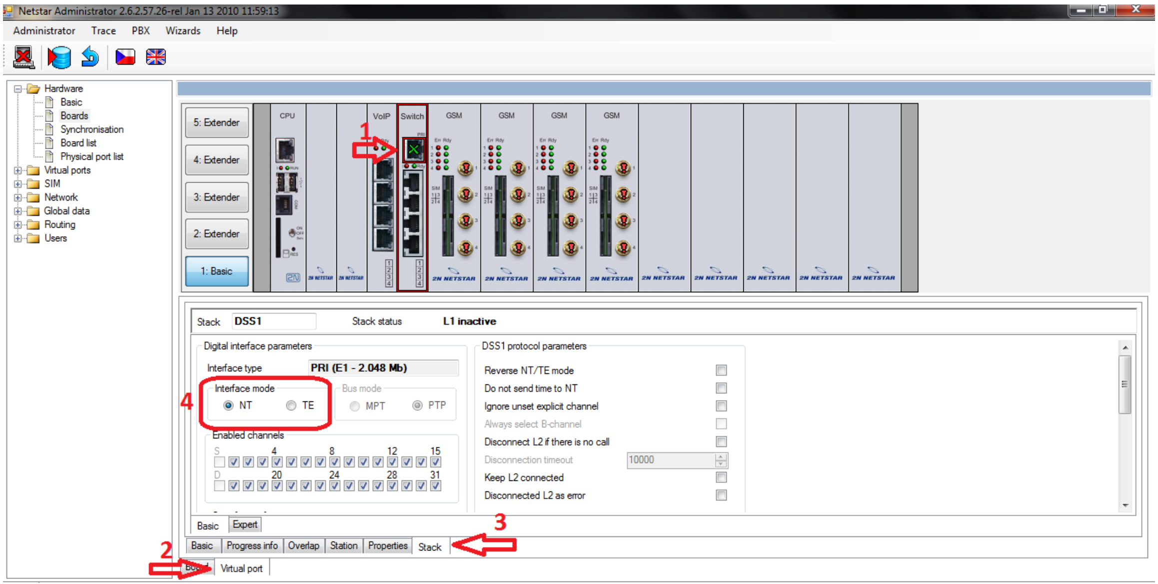
Interface Configuration

PRI ISDN



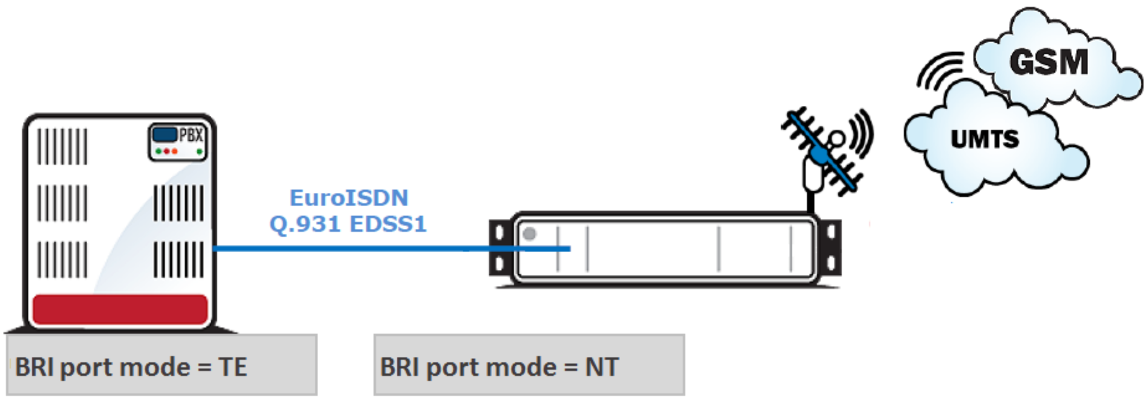
The most important aspect of PRI interface configuring is the configuration of the PRI line between the NetStar and PBX systems. The first information we need is the PRI interface configuratin on the PBX. In case you are not sure about the PBX PRI port configuration, contact a person responsible for the PBX maintenance without delay. In our case, the PBX was connected to the PSTN and so the PRI port is configured as TE. For correct interconnection, NetStar has to be configured as NT. To do so you have to:

- **Set PRI card jumpers** – to do so switch NetStar into the service mode. When the light goes off on the PRI card, remove the card from the rack and check the jumper configuration. For a correct jumper placing use the sticker on the PRI port.
- **Set correct communication protocol** in the NetStar configuration tool. To do so go to the **Boards** menu in **Hardware** and choose the port to be configured. Set the **Virtual port** and **Stack** tags for this port. When you are in the Stack menu, set the interface mode to **NT**.



To apply your new configuration, save the changes to NetStar using the saving icon (or Ctrl+S). To configure the interface into the TE mode, take the same steps and set the jumpers and interface mode to TE.

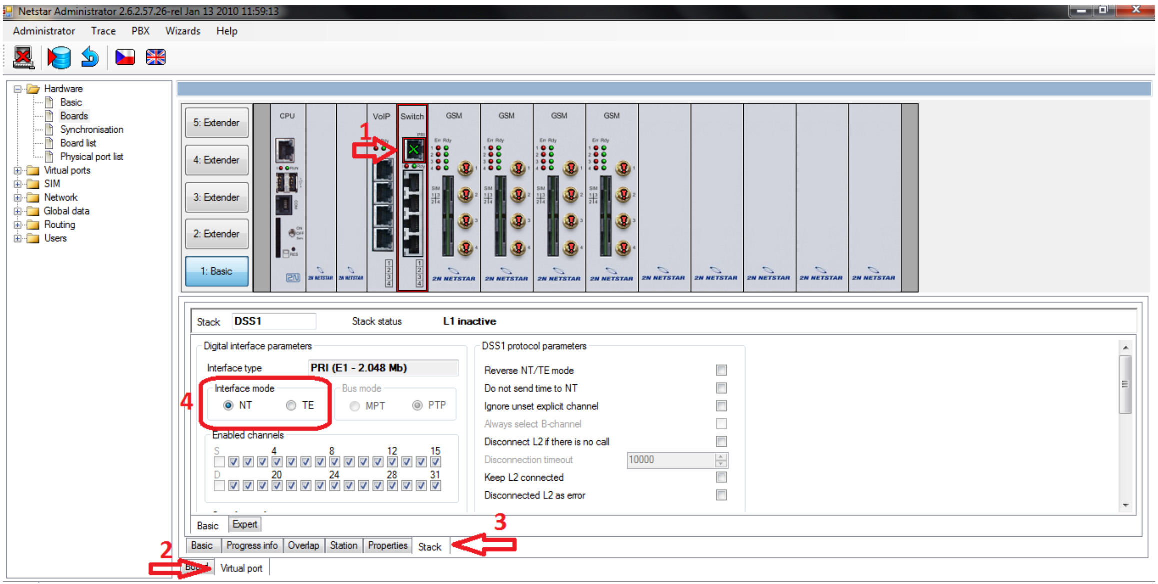
BRI ISDN



The most impotant aspect of BRI interface configuring is the configuration of the BRI line between the NetStar and PBX systems. The first information we need is the BRI interface configuration on the PBX. In case you are not sure about the PBX BRI port configuration, contact a person responsible for the PBX maintenance without delay. In our case, the PBX was connected to the PSTN and so the BRI port is configured as TE and Point-to-Point. For correct interconnection, NetStar has to be configured as NT and also PTP. To do so you have to:

- **Set BRI card jumpers** – to do so switch NetStar into service mode. When the light goes off on the BRI card, remove the card from the rack and check the jumper configuration. For a correct jumper placing use the sticker on the BRI port.
- **Set correct communication protocol** in the NetStar configuration tool. To do so go to the **Boards** menu in **Hardware** and choose the port to be configured.

Set the **Virtual port** and **Stack** tags for this port. When you are in the Stack menu, set the interface mode to NT and mode to PTP.

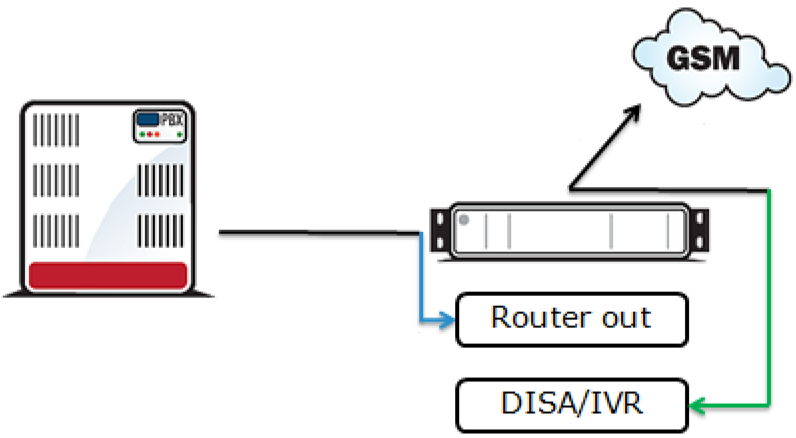


To apply your new configuration, save the changes to NetStar using the saving icon (or Ctrl+S). To configure the interface into the TE mode, take the same steps and set the jumpers and interface mode to TE.

LCR Creation

The final configuration step is to create the LCR rules and configure the interfaces to work properly according to these rules.

Our task is to enable all outgoing calls to be passed to GSM and all incoming calls to play the DISA welcome note or be passed to the PBX IVR.



Outbound calls

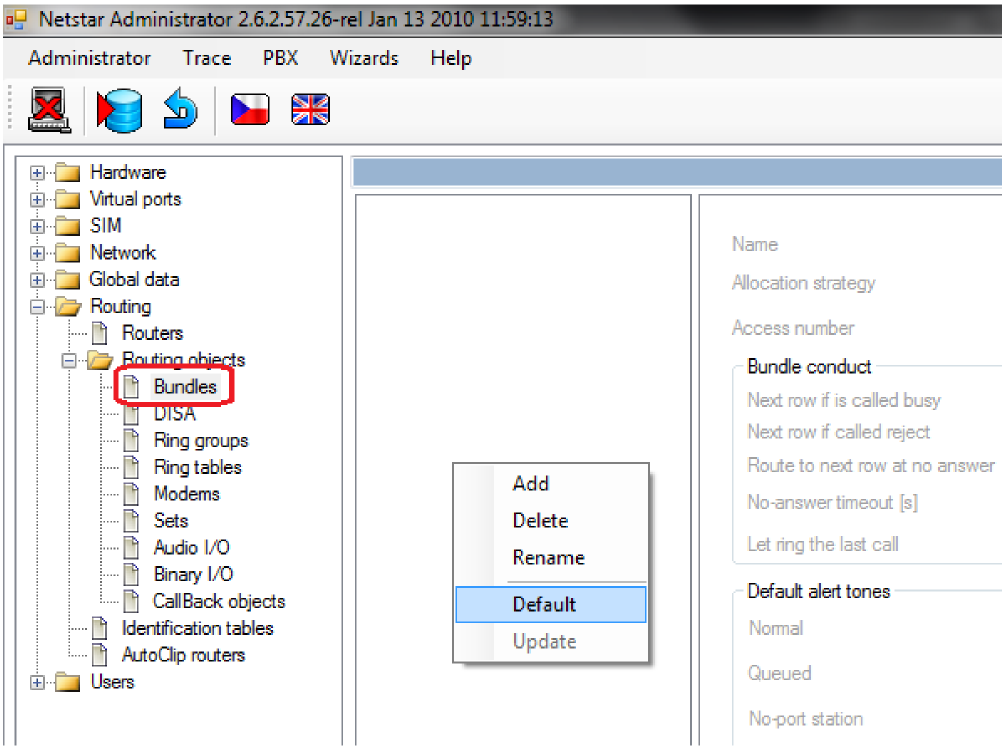
We need to take three steps for outbound calls.

1. Create a GSM bundle responsible for a correct and well-balanced use of all GSM

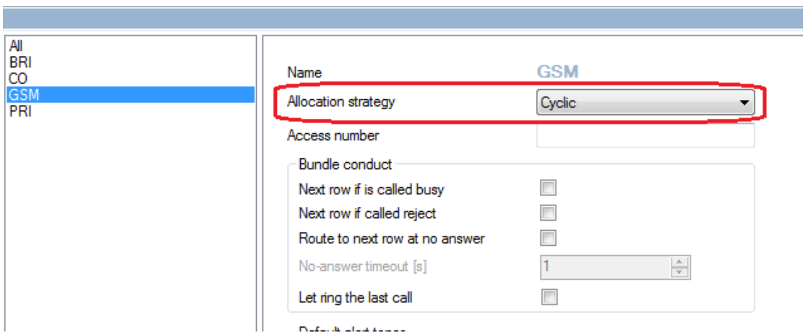
modules.
2. Create a router responsible for routing calls to the GSM bundle.
3. Assign this router to a virtual port connected to the PBX.

Create GSM bundle

- Go to **Routing – Routing objects – Bundles**, click on the right mouse button and choose **Default** to create the default set of bundles. One of them is called GSM and filled with all GSM ports.

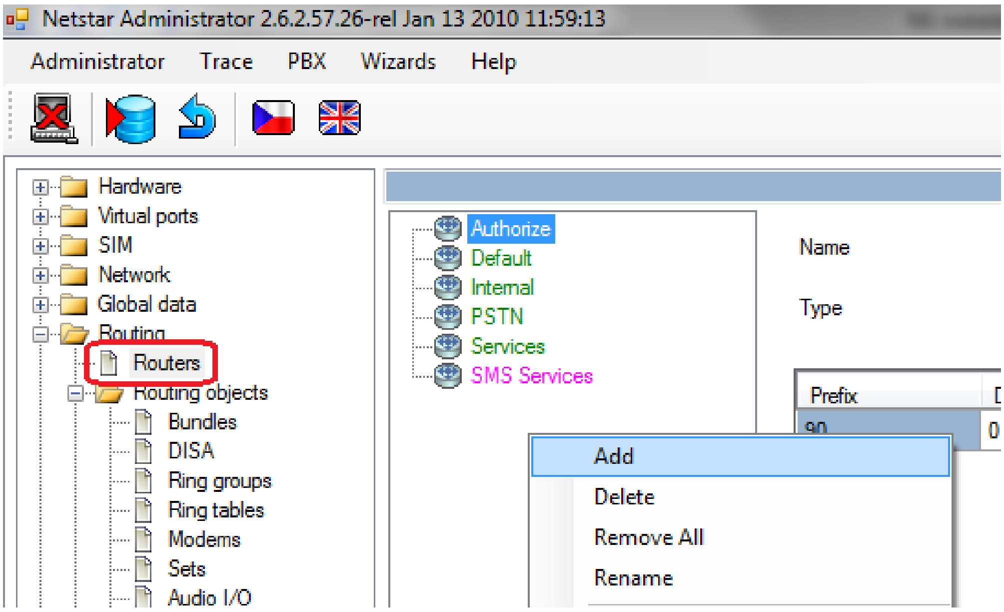


- Make sure that the count of the GSM ports in the GSM bundle matches the count of ports available in NetStar.
- Configure the bundle – set the allocation strategy to **Cyclic**.

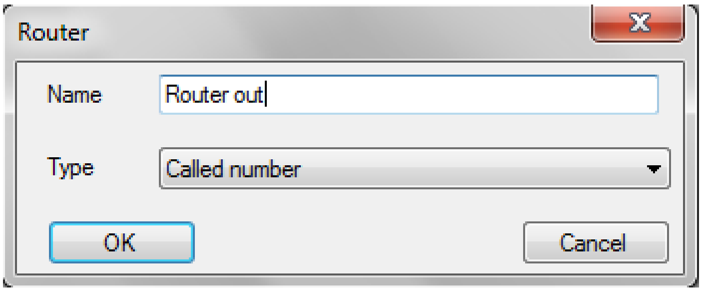


Create router

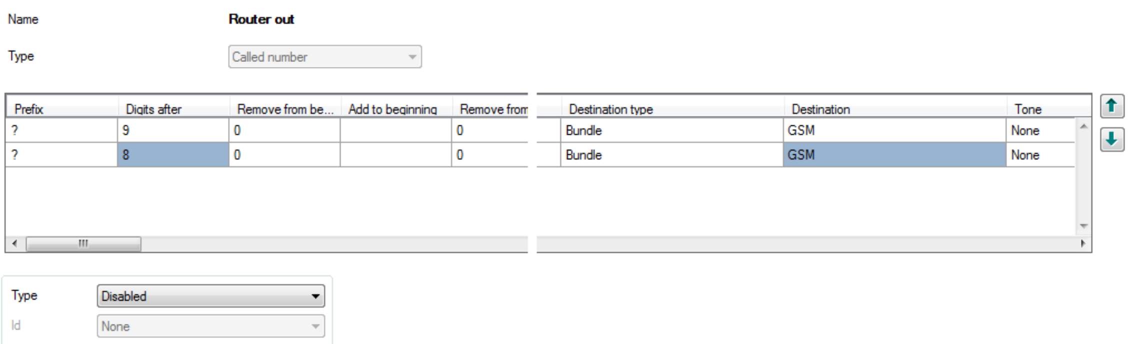
- Go to **Routing – Routers**, click on the right mouse button and choose **Add**.



- Fill in the router name and keep the **Called number** type selection.

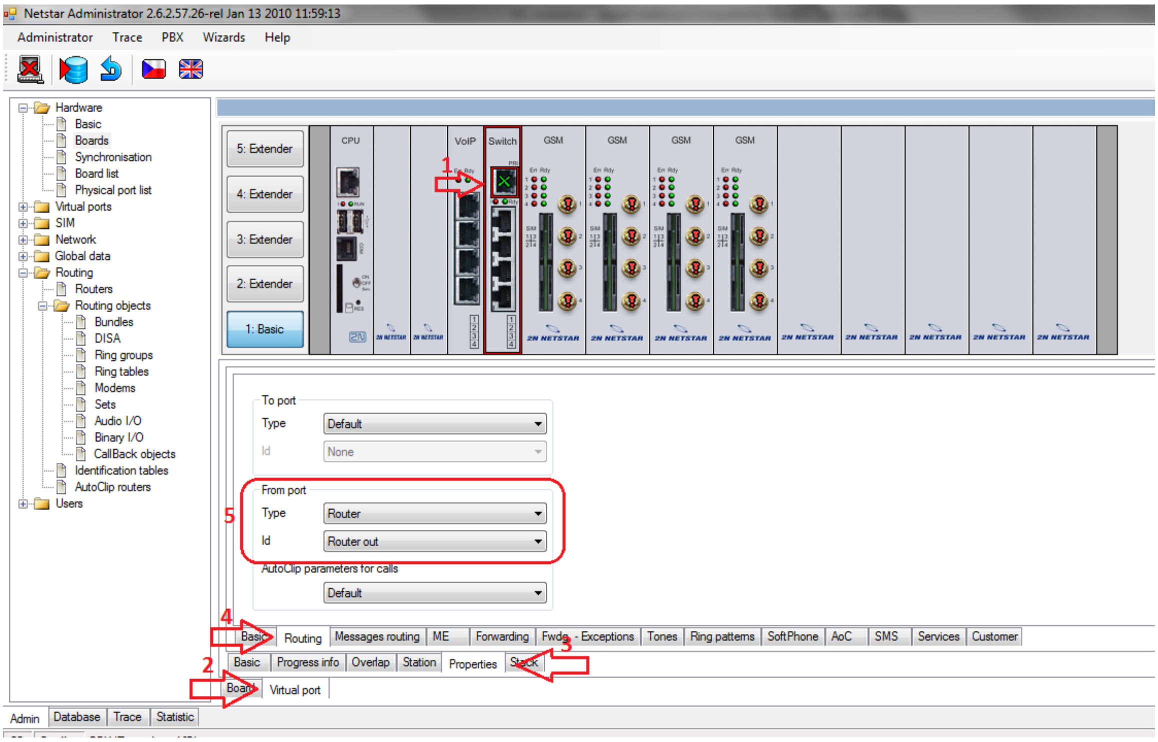


- Add 2 rows as shown in the figure below (click on the right mouse button and choose **Add**).



Assign router to PRI/BRI port

- Go to the **Hardware – Boards** and choose a port connected to the PBX.
- On the bottom side of the configuration tool choose the **Virtual port** tag, then **Properties** and finally **Routing**.
- In the **Routing** tag set **From port**, Type to Router and Id to your router.



Save your new configuration to NetStar using the icon.

Now NetStar is properly configured to pass calls from the PBX to GSM or PSTN through a bundle of GSM ports.

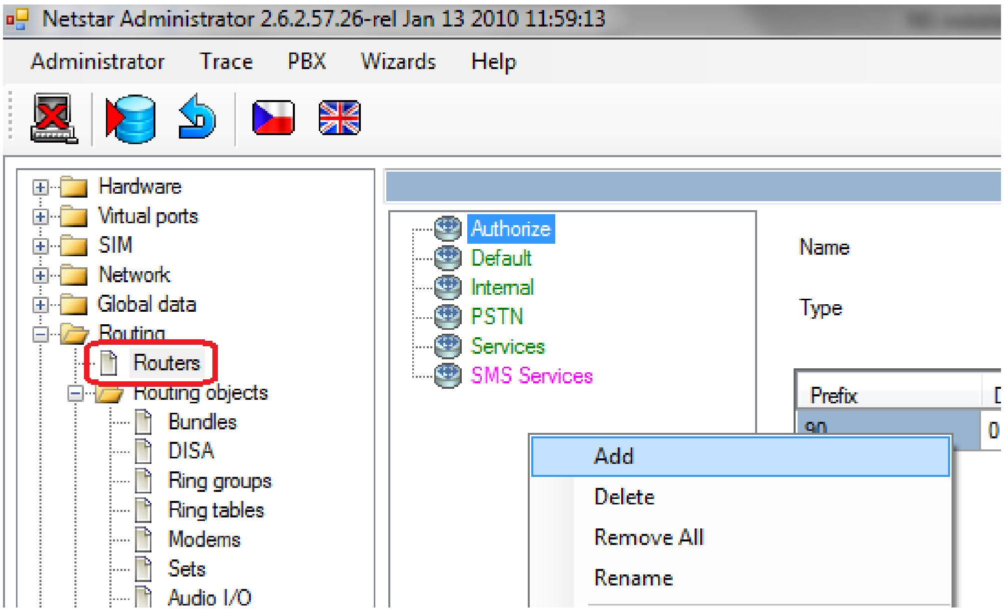
Inbound calls

We need to take two steps for inbound calls.

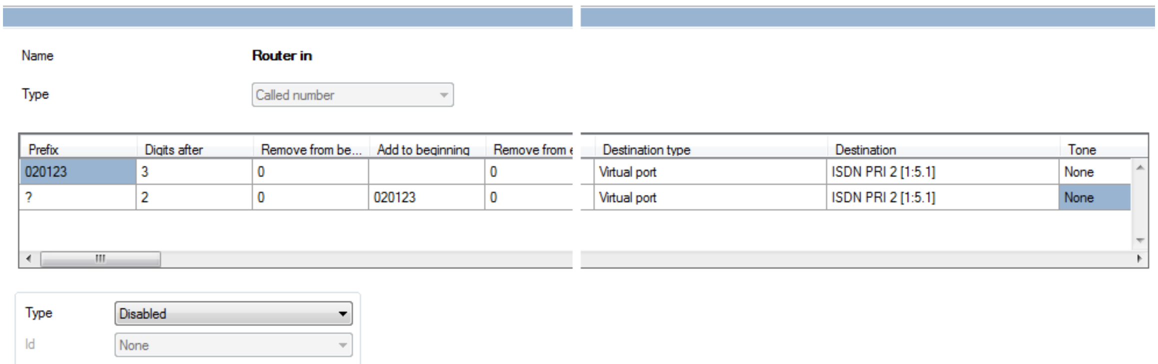
1. Create an incoming router responsible for routing calls into the connected PBX and assign the router to the virtual port through which NetStar is connected to the PSTN (GSM ports).
2. Set the DISA function for processing incoming calls.

Create router

- Go to **Routing – Routers**, click on the right mouse button and choose **Add**.

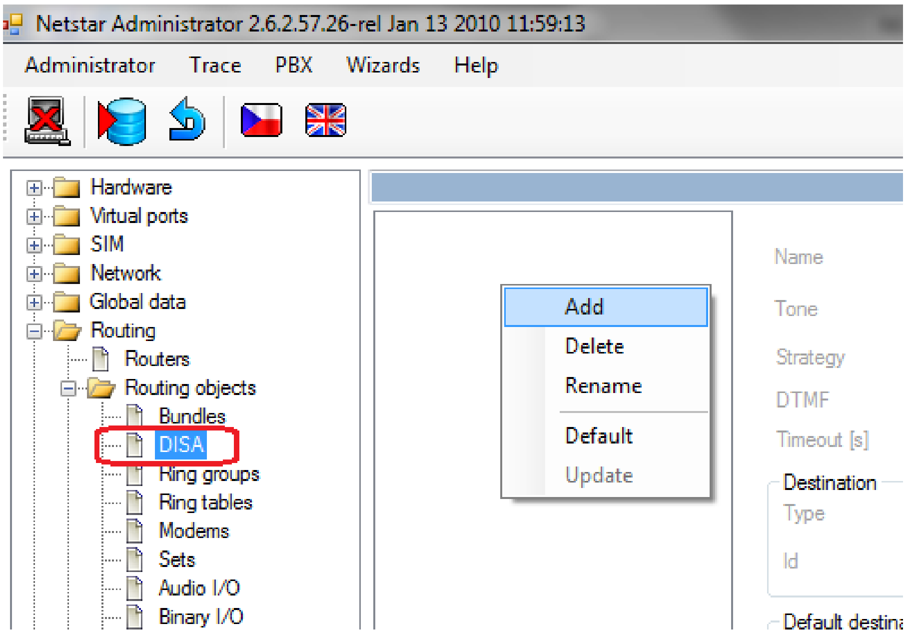


- Fill in the router name and keep the **Called number** type selection.
- Suppose that the PBX PRI port cannot be re-programmed. In this case we have to send a call request in the same format as the PSTN. In our example the company number is 020123xxx. When DISA passes the digits to our router, we have to take into account that the user can dial the number as a full PSTN number (first line) or as a short extension number (second line). In the latter case, we have to add 020123 to make the PBX receive the number in same way as from the PSTN. In both cases, the call will pass to the PRI interface that is connected to the PBX.

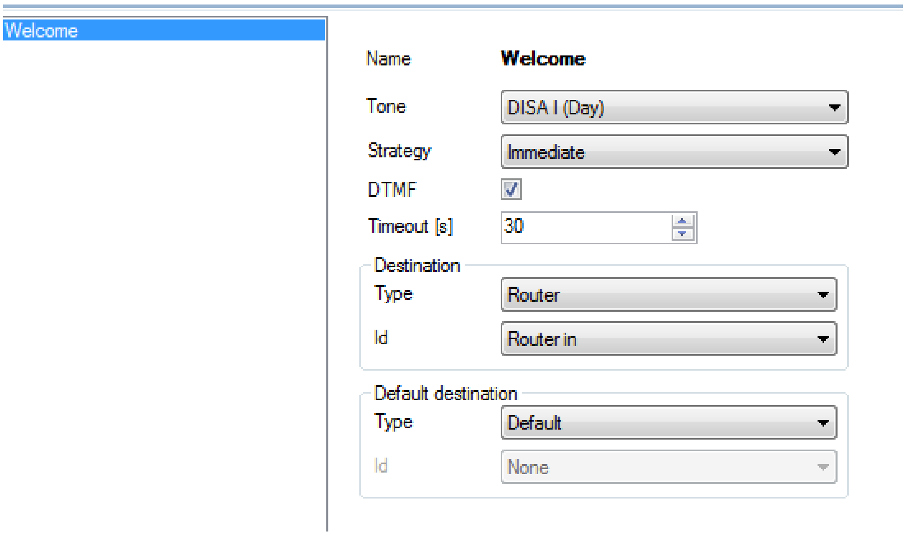


Configure DISA for incoming calls

- Go to **Routing – Routing objects – DISA**, click on the right mouse button and choose **Add**.

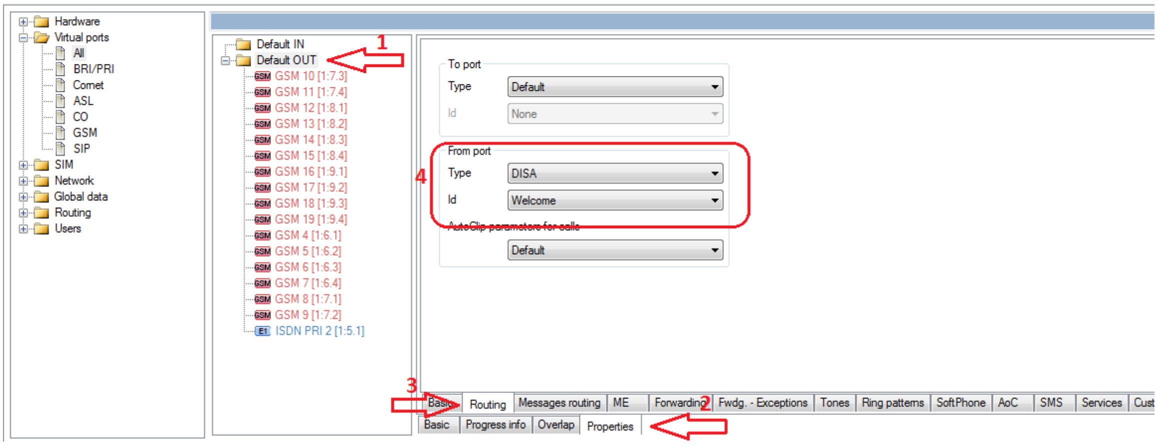


- Name your new DISA and set it as shown in the figure below.



Assign DISA for all GSM channels

- Go to **Virtual ports – Default OUT – Properties – Routing**.
- In the **Routing** tag set **From port**, Type to DISA and Id to your new DISA.



Save your new configuration to NetStar using the icon.

Now NetStar is properly configured to answer incoming calls from GSM, play DISA to them and pass dialled numbers to the PBX.

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