ConfigMAN 6.42

Programming manual for:

Slican IPS-08

Slican IPU-14

Slican IPM-032

Slican IPL-256

Slican CXS-0424

Slican CCT-1668

Slican MAC-6400

Slican MAC-ZERO

Release 1.40
for
6.42 firmware version

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FAIR PLAY 2006

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Manufacturer reserves right to make changes without prior notification.

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Table of content

1 <u>Preparing the PBX for programming</u>

1.1 Installing ConfigMAN and connecting the PBX to a computer

ConfigMAN is provided by the manufacturer as a software package to be installed on a PC. It contains:

- up to date version of ***ConfigMAN*** software,
- Windows USB drivers,
- current help file

During installation please execute commands displayed by setup application on PC screen. ConfigMAN application will be installed in default location.

By default the target directory for the program is:
C:\Program Files\Slican\ConfigMAN

The USB drivers are located in:
C:\Program Files\Slican\\Drivers\USB

Database directory for the MAN system (MultiMAN, ConfigMAN, BilingMAN):
C:\Program Files\Slican\pabx data

Above mentioned paths can vary slightly depending on operating system. Manufacturer assures that ConfigMAN package is backwards compatible with older versions of PBX firmware. On the other hand, each new version of the firmware is only compatible with the appropriate version of ConfigMAN.

Hardware requirements

- PC with x86 or amd64 microprocessor,
- 128 MB RAM,
- 50 MB free hard drive space,
- Windows 2000/XP/Vista/Windows 7/Windows 8/Windows 8.1/ Windows 10,

Attention: *Older versions of operating system ex. Windows 98, Windows Me <u>are not supported</u>.*

1.2 Local PBX managing

Configuring ConfigMAN for connecting via USB – driver installation

First connection to the PBX using USB requires the installation of USB port drivers. After connecting the computer to the USB port of the PBX, the operating system will display a driver installation wizard.

Fig. 1.1: New hardware installation wizard window

It is recommended to install the driver from the specified location. The next step requires the user to provide the access path to the driver files.
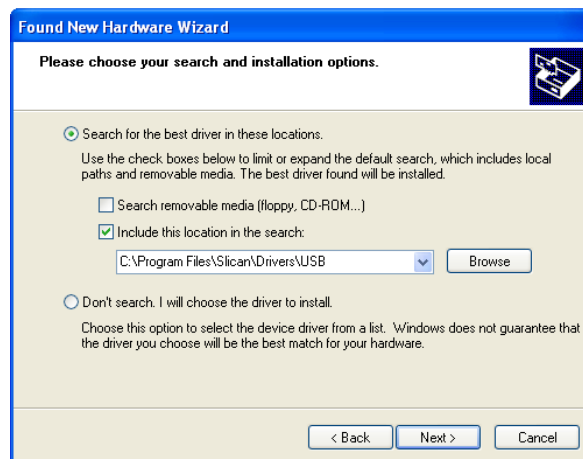
Fig. 1.2.: Hardware update wizard window

*By default, after installing **ConfigMAN**, the USB drivers are located in:*
C:\Program Files\Slican\Drivers\USB

After providing the directory, the system will automatically install the drivers. During installation, a window might pop-up, warning of a lack of compatibility testing for the operating system. You should nevertheless continue the installation process.
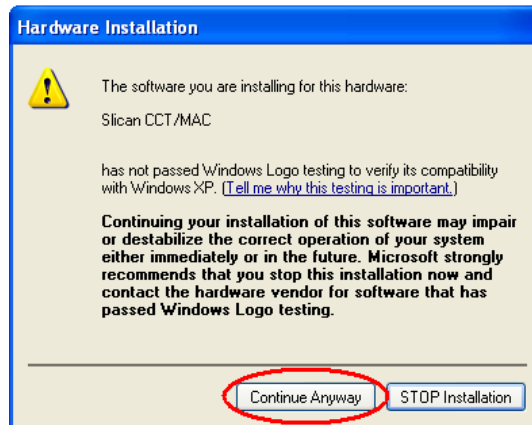
Fig. 1.3.: Window informing about lack of compatibility tests with Windows XP

For CXS-0424 after ConfigMAN installation and connecting cable to mini USB terminal, in device manager new network card named **USB Slican CXS-0424 Network Device #<no>** will be visible.

During installation, the system might again ask to be pointed to the location of some of the files. In that case, you should again select directory:

C:\Program Files\Slican\Drivers\USB.
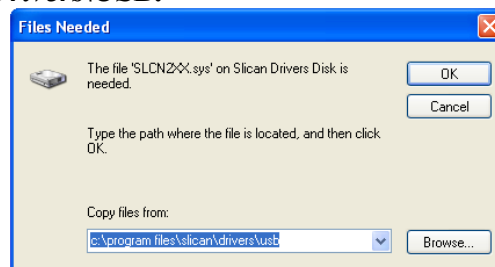
Fig. 1.4.: Window for selecting USB drivers folder

After installing the drivers, the system will confirm the success of the process.

Fig. 1.5.: Window for finishing USB driver installing process

After installation, the device manager will contain a new USB device called **Slican CCT/MAC.**
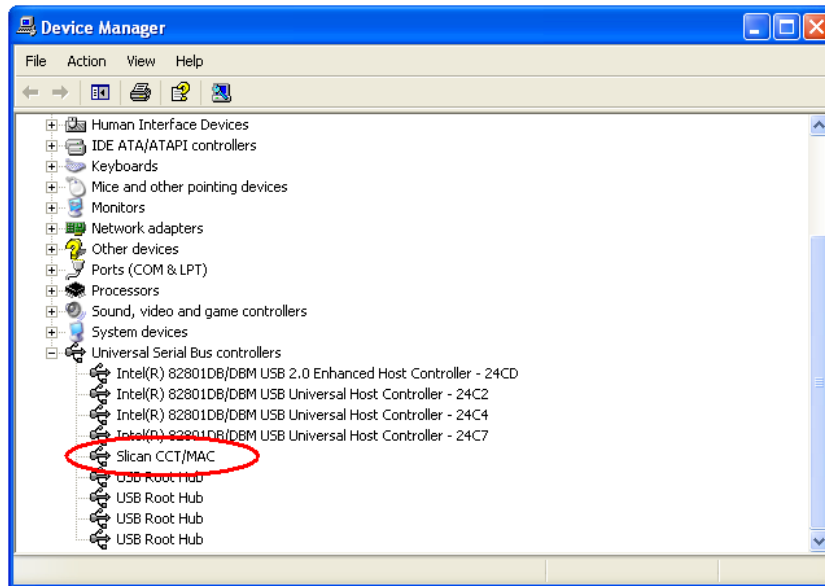
Fig. 1.6.: Window device manager

After completing the installation process select USB as the transport medium in the **PABX >> Connection configuration** dialog box.
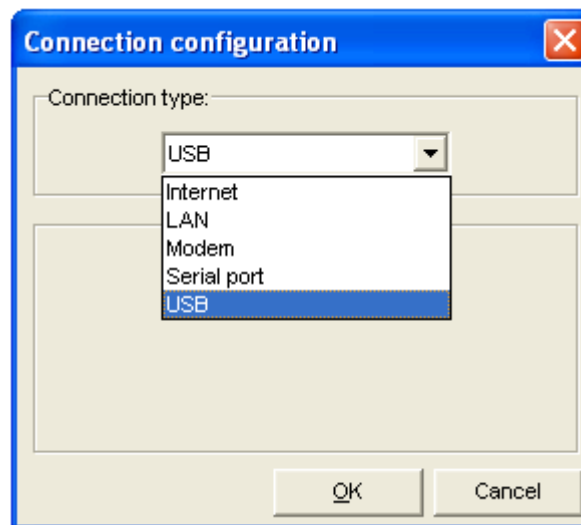
Fig. 1.7: Connection configuration window

Configuring ConfigMAN for connecting via LAN

To connect to the exchange using LAN you need to select **LAN** in the **PABX >> Connection configuration** dialogue box, and provide the IP address of the PBX.
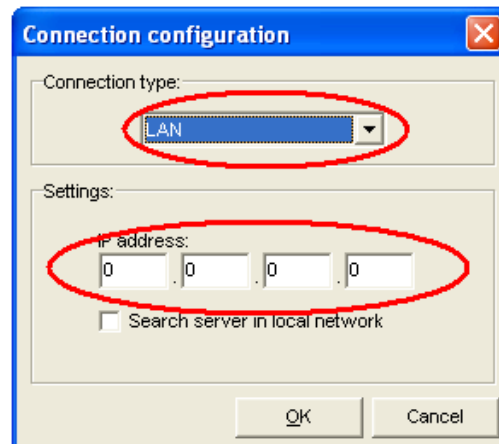
Fig. 1.8.: Connection configuration via LAN

Configuring the IP address of PBX

By default, the PBX is set with the IP 192.168.0.248. If the computer is given an address from the 192.168.0.xxx range and a mask 255.255.255.0 (ex. 192.168.0.1) it can be used to connect to the PBX via LAN and change its IP address.

To change the IP address of the PBX, you need to provide new settings in the **IP Network/LAN interfaces** sheet, or set the PBX to use DHCP.
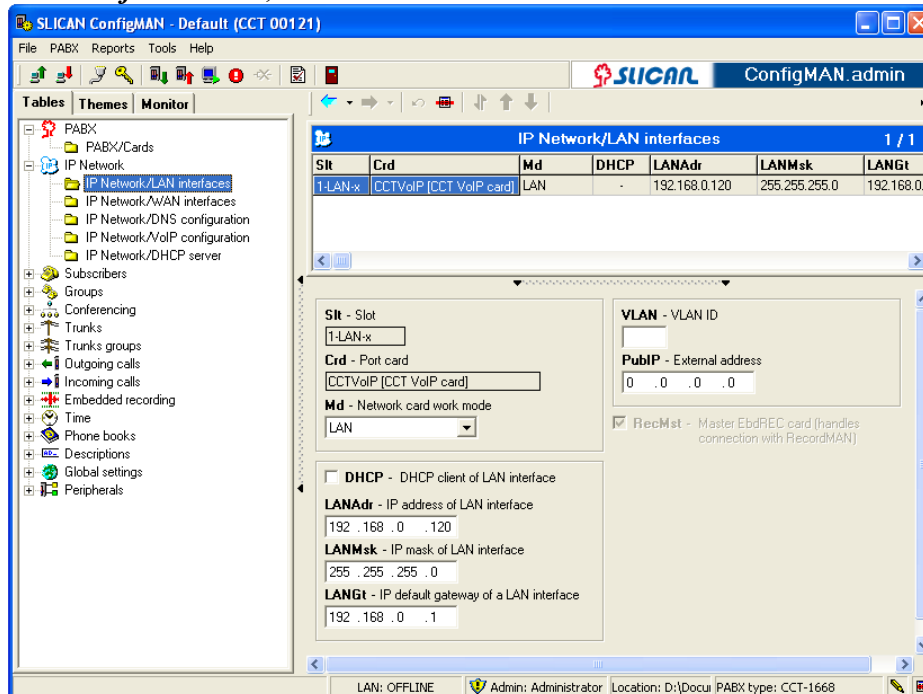
Fig. 1.9.: LAN configuration

1.3 Remote PBX managing

Configuration of global settings concerning with remote PBX managing

Remote PBX managing can be realized using modem connection or over the internet. For both configuration is described in following chapters. Remote access to PBX can be:

- **Switched on** – access over the internet and using modem connection is switched on for all the time,
- **Switched off** – access over the internet and using modem connection is switched off for all the time,
- **Temporary** – access to PBX over internet and using modem can be switched on temporarily by user with appropriate privileges from internal phone only, using service *987KKKK, where KKKK – four digit access code granted by admin in sheet *Global settings / PABX access codes.* Temporarily access is limited to 30 minutes between service selection and establish connection. Every next connection extend access time by 30 minutes.
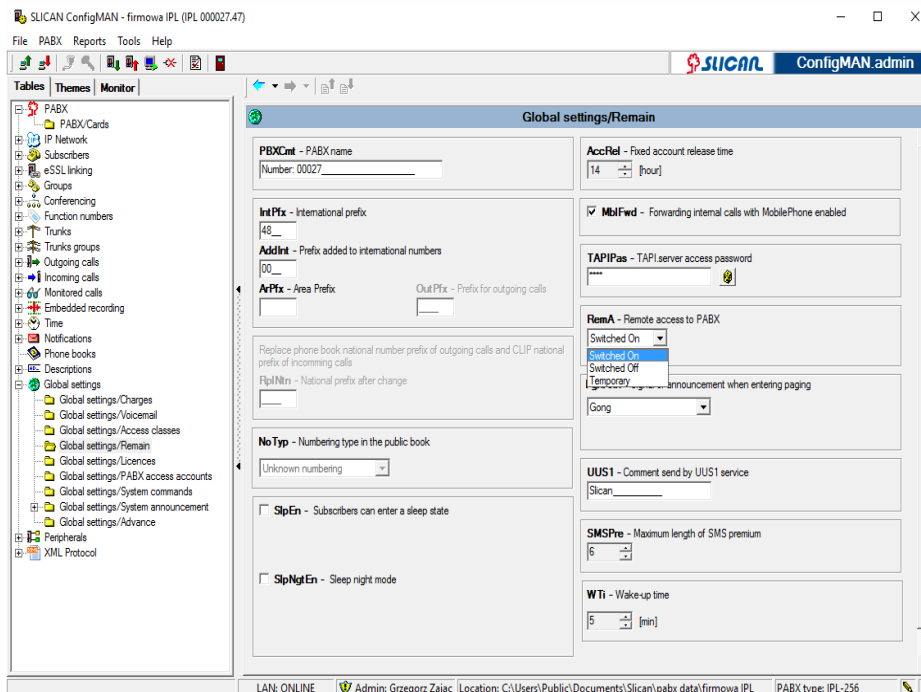
Settings of remote access to PBX can be done in field *RemA* of sheet *Global settings / Remain.*

Configuring ConfigMAN for connection via Internet

Companies can manage their PBXes using the Internet thanks to the built-in Gatekeeper mechanism, which authorize access to the equipment. All connections to the PBX over the Internet are **encrypted**. To PBX communication **port 5525** is used.

To obtain connection with the PBX using the Internet, you must first provide your **Service identifier** (the same as the one used to log in on the www.slican.com webpage in the *ServNET area*) and **Password** (the same as the one used to log in on the www.slican.com webpage in the *ServNET area*), as well as the PBX's serial number.

Fig. 1.11.: Configuration for connection over Internet

As the connection to the PBX goes via several links (i.e. user – keeper, keeper – router, router – LAN and LAN – Slican PBX), the cause of any errors needs to be precisely defined. The dialog box on the image below shows an example of a connection error.

Fig. 1.12.: Window of connecting with PABX over Internet – connection error identification

Information about Keeper connection can be found in ***Monitor***, in ***Port status*** sheet.
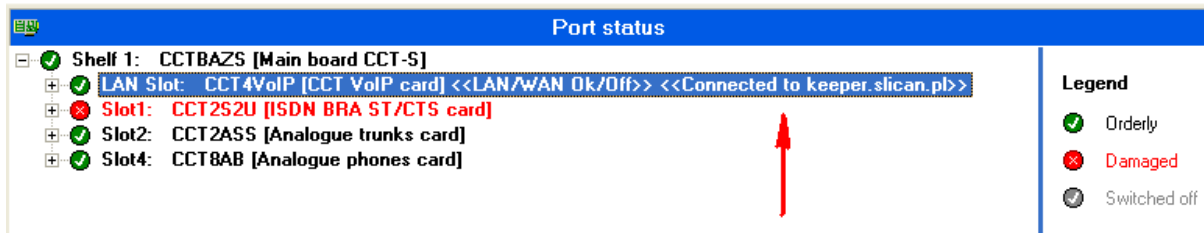
Fig. 1.13.: Information presentation about connection with keeper

Information about loss or reestablishing connection are collected in **PBX monitor**, in sheet **PABX events**.
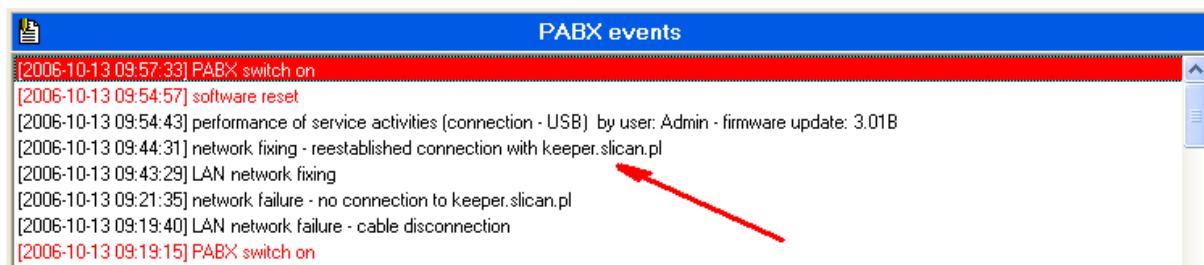
Fig. 1.14.: PBX events – information about connection with keeper

Configuring ConfigMAN for modem transmission

(in CCT with firmware >=5.11 purchase separate modem module is required)
Modem connections to the PBX can either be through external or internal lines. To enable modem communications in **ConfigMAN** the **Modem** option needs to be selected in the **PABX>> Connection configuration** dialog box. Additionally, if there's more than one modem installed in the system, the appropriate entry needs to be selected from the drop down box.
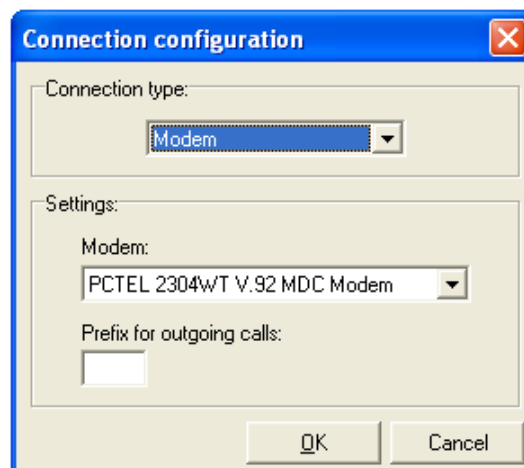
Fig. 1.15.: Selection of modem connection

If the system is connecting to the PBX via modem, then after selecting **Download configuration** (F8 key) option from the PBX menu, the system will ask for the number to dial into, and the method of dialing: direct, or using DISA announcement. When the PBX is dialed into using the DISA function, the delay for dialing the internal modem number can be adjusted.
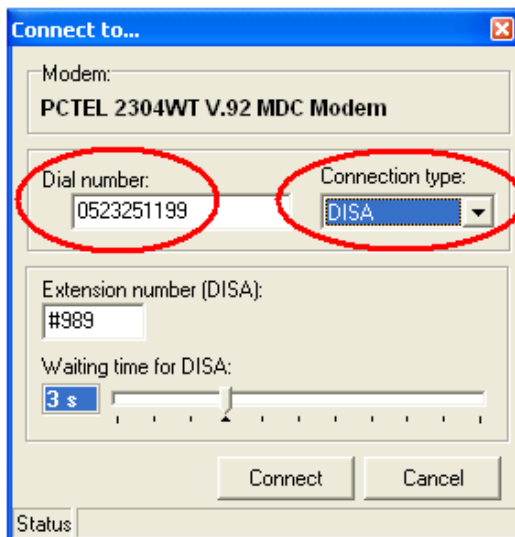
Fig. 1.16.: Configuring modem connection

If the modem has not been assigned an internal number, it can be reached using the *989 service while on the prompt, or while being directly connected to an internal (subscriber) line.

Attention: *To delay the digits being dialed, they can be separated by commas ",". The comma introduces a delay of 1s. To further increase the delay more comma symbols can be inserted.*

For direct modem call in sheet **Incoming calls/Settings/Main**, incoming call should be routed to system modem as shown on below figure.
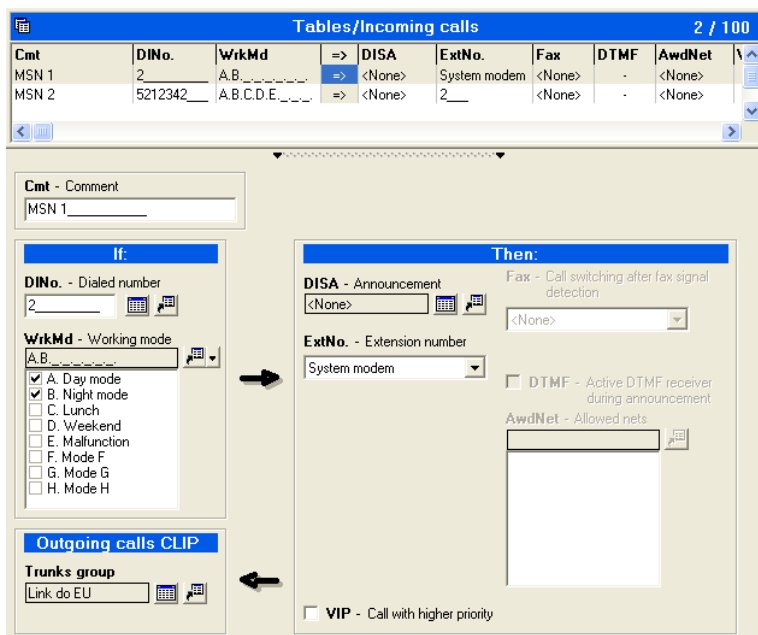
Fig. 1.17.:Settings for direct modem calls

If there are problems concerned modem connection between PC and PBX, additional initial modem commands should be used. These commands should have following form:

MM11   – for Motorola modems

$+MS=V32$ – for other modems

These commands should be entered in *Modem properties* widow, in tab *Advanced* as shown on below figure.
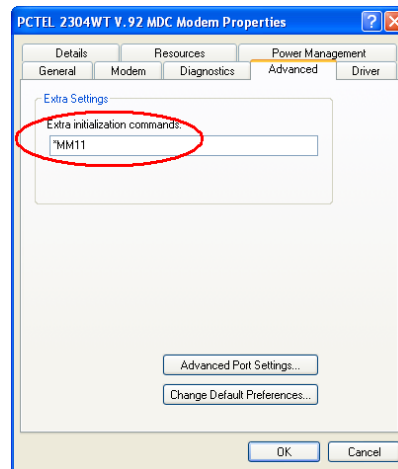
Fig. 1.18.: Advanced modem properties for Windows XP

Purpose of using initial mode commands is to force slower modem working speed. If despite of using additional modem initial commands, connection between PC and PBX modem cannot be established, please contact with Slican service department.

2 ConfigMAN software description

2.1 PBX details

To view the details of the PBX, you have to connect to Slican PBX and download it's configuration.

In the *PABX* sheet you can find current information about:

- PBX type,
- serial number,
- software version (firmware) of the PBX,
- software version (firmware) of LAN/VoIP/REC card (if installed).

The *Statistics* field shows the number of ports and maximum capacity.
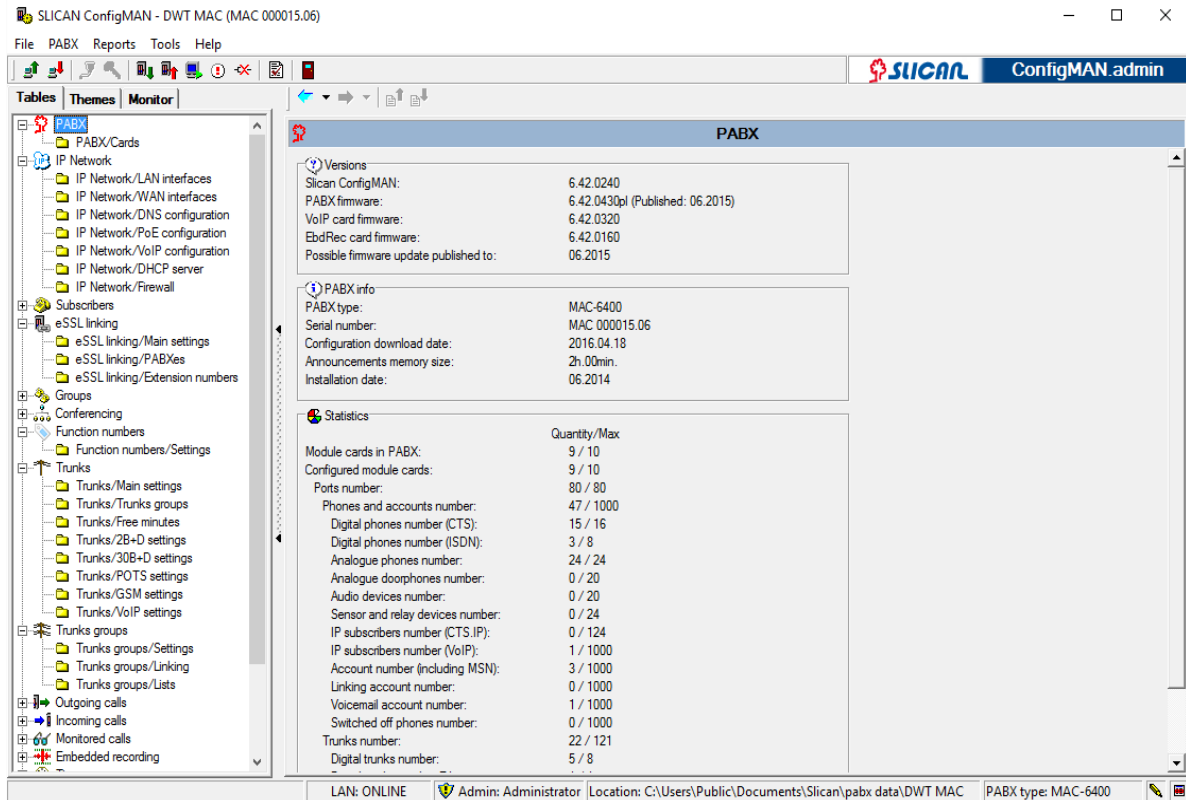
Fig. 2.1.: PBX details – most important information about the Slican PBX

Attention! *For CXS-0424 file with LAN/VoIP firmware is included in PBX firmware (both firmwares are together). Because of this appropriate message can be found in PBX details.*

The main branches of the themes tab show statistics for the various themes groups.
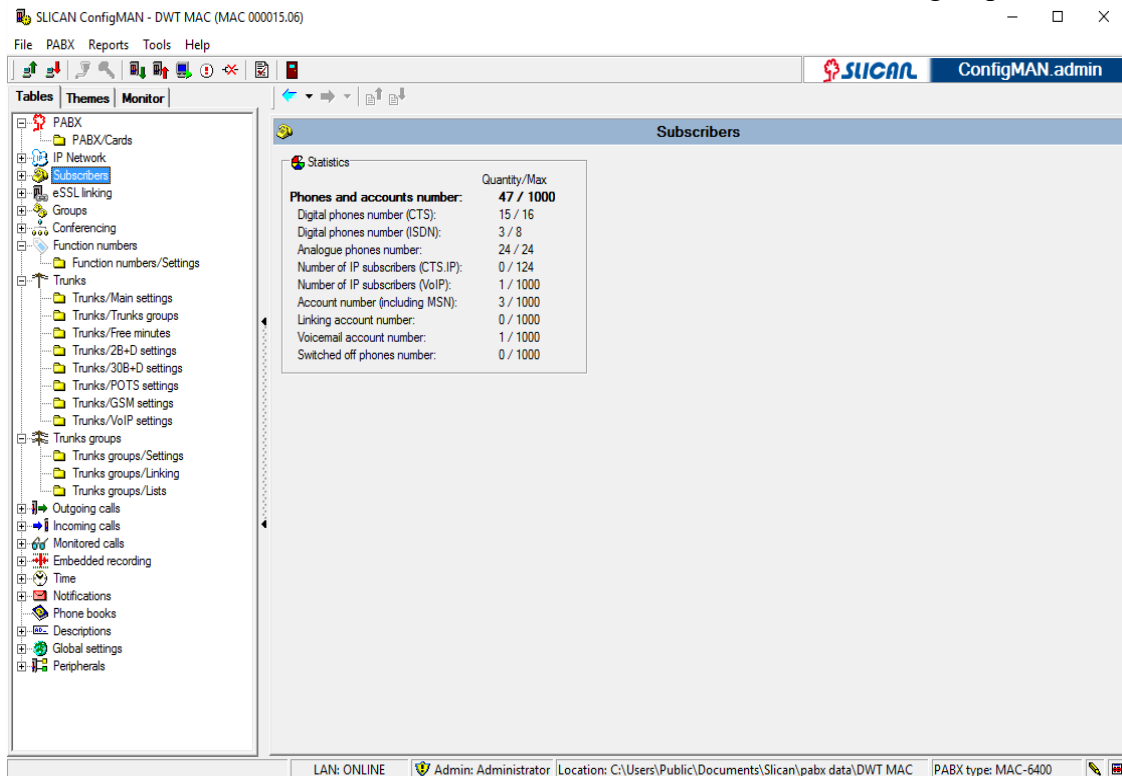
Fig. 2.2.: Statistics – Details for Subscribers

2.2 Interface description

The user interface of *ConfigMAN* is divided into three panes:
- tree,
- table,
- form.
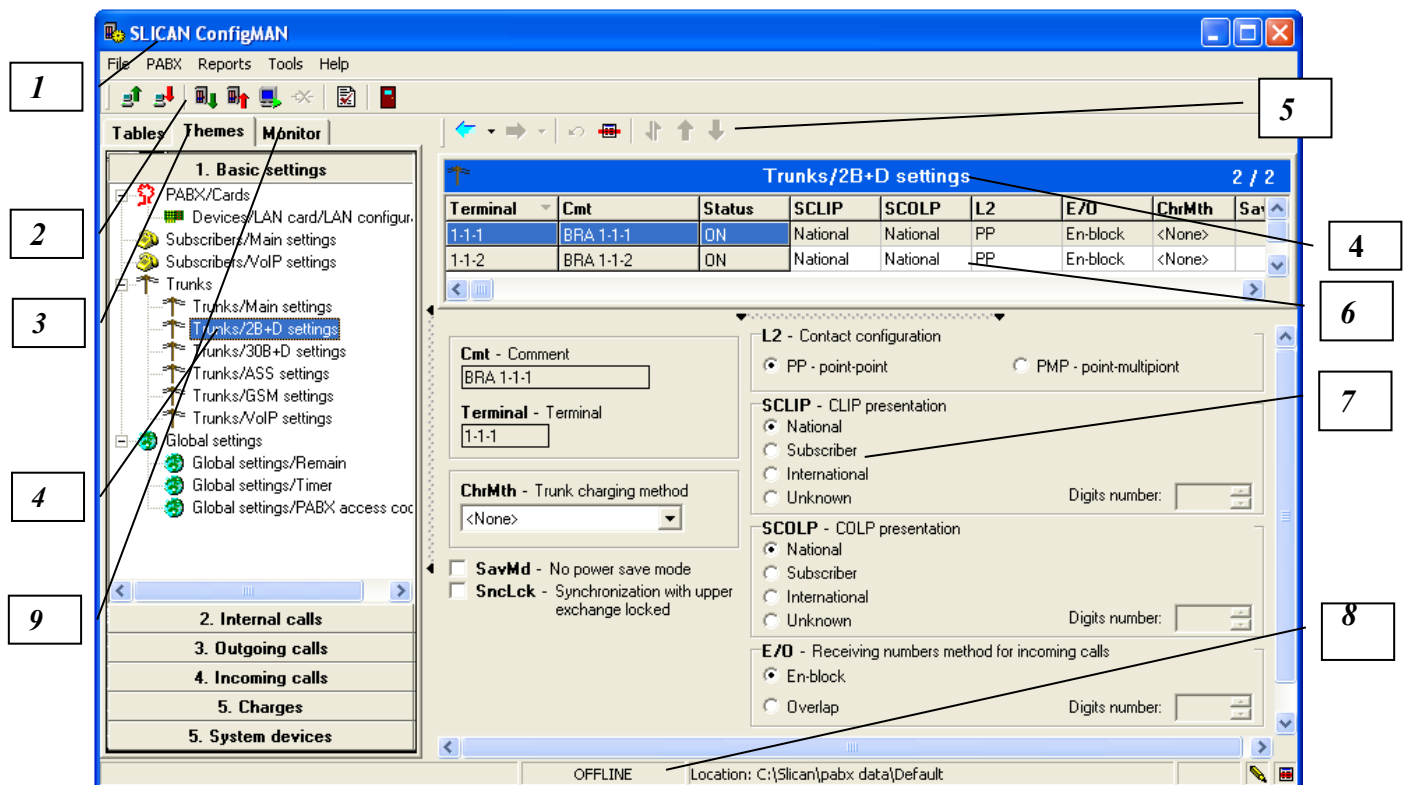
Fig. 2.3.: Description of ConfigMAN interface

1.	Main menu	*Main menu, containing the configuration options and built-in help*
2.	Toolbar	*Shortcuts to the most often used functions of the application*
3.	Tree view tabs	*Tabs for selecting either themes or table mode for the tree view window*
4.	Tree view	*List of sheets stored in the PBX*
5.	Table toolbar	*Contains tools for navigating and editing the tables*
6.	Table	*Detailed view of the selected sheet*
7.	Editing panel	*For editing individual fields in the sheet*
8.	Status bar	*Information about the current state of the PBX, as well as the location of the configuration file*
9.	Monitor	*Software module to monitor and diagnose the PBX*

2.3 Help system
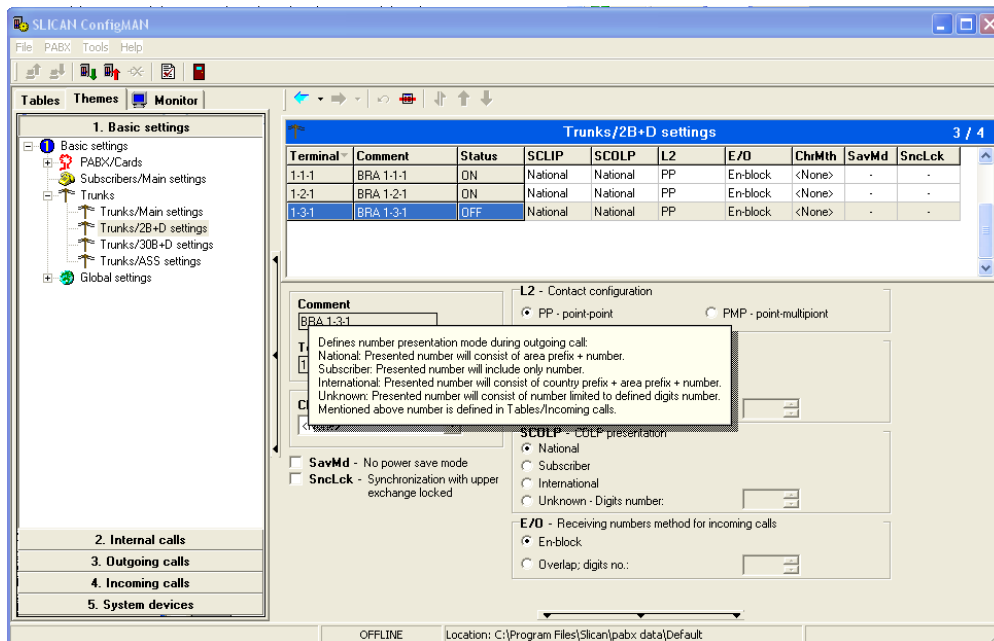
Fig. 2.4.: Displaying context help for fields available in specified window

ConfigMAN is equipped with an extensive hint-based help system on the ways of filling out fields in the sheets. The user can always select an extended description of a field in the form. While hovering the cursor over an interesting field, the cursor will change to: ⛽. An additional mouse click allows the user to view the extended description of any field.

2.4 Sheet views

Status fields in the tables are colour-coded:

- Editable fields in the table have a white background,
- Non-editable fields are marked grey.

Fig. 2.5.: Sheet editing possibilities

The currently selected row is highlighted in blue.
Additionally, the fields with a new value, which hasn't yet been upload to the PBX table, are highlighted in red.

2.5 Editing record

Editing a record in the table is accomplished using the editing panel. To edit a selected record, it has to be selected in the sheet panel first, and then, the value has to be entered in the appropriate field on the editing panel.

Fig. 2.6.: Record editing possibilities - editing panel for sheet Subscribers/Call forwarding

The **ConfigMAN** software has several ways of filling out record fields. One is the toggling of a field by selecting it in the window. The second method is the direct input of a field's value in an edit box.

If a field has an icon for selecting it's value from a different table, it's still possible to edit it directly, even if the field itself is greyed out.

If data in an associated table needs to be changed, the table can be accessed using the shortcut button next to the field which is to receive the value from the other table. If there are more than one associated table, it is possible to choose one table from list of all associated. It is showed on below figure. It is possible to switch to selected associated tables.

Fig. 2.7.: Record edition - switch to one of many tables

2.6 Multiple-editing

If multiple fields in the table need to have their value set, they can be changed using multiple-editing.

Changing the settings in several records at the same time is accomplished by holding down the SHIFT (selecting range of records) or CTRL (selecting separate records) key and clicking the records to be selected, highlighting them with blue color.
It possible to select all table records by pressing CTRL and A keys.

Fig. 2.8.: Editing many records during the same time

Modifying a field's value will cause every selected record to be updated with the entered value.

2.7 Editing table rows (records)

Add record

To add a new record, click the ***Insert new record*** button, or press the CTRL and INSERT keys simultaneously.

Fig. 2.9.: Adding new record (row) in table

Adding the record inserts an empty row, with all fields with default value set into these fields.

Copy record (Paste)

Copying records allows you to transfer the settings between records, without adding a new entry, so it can be noticed that functions *Copy* and *Paste* do not realize theirs tasks. Although, if for extension number selected in sheet **Subscribers / Main settings** is set - for example **UrgEn** (urgent call enable) and **OutFwd** (Permission for outgoing calls forwarding), located in sheet **Subscribers / Service settings** after copy record of specified extension number these setting will be copied, as well as other settings located in other sheets. Then while pasting, all copied setting will be inserted in record indicated before pasting.

To copy a record, click on the **Copy selected record** button, or press the SHIFT+CTRL+C keys combination.
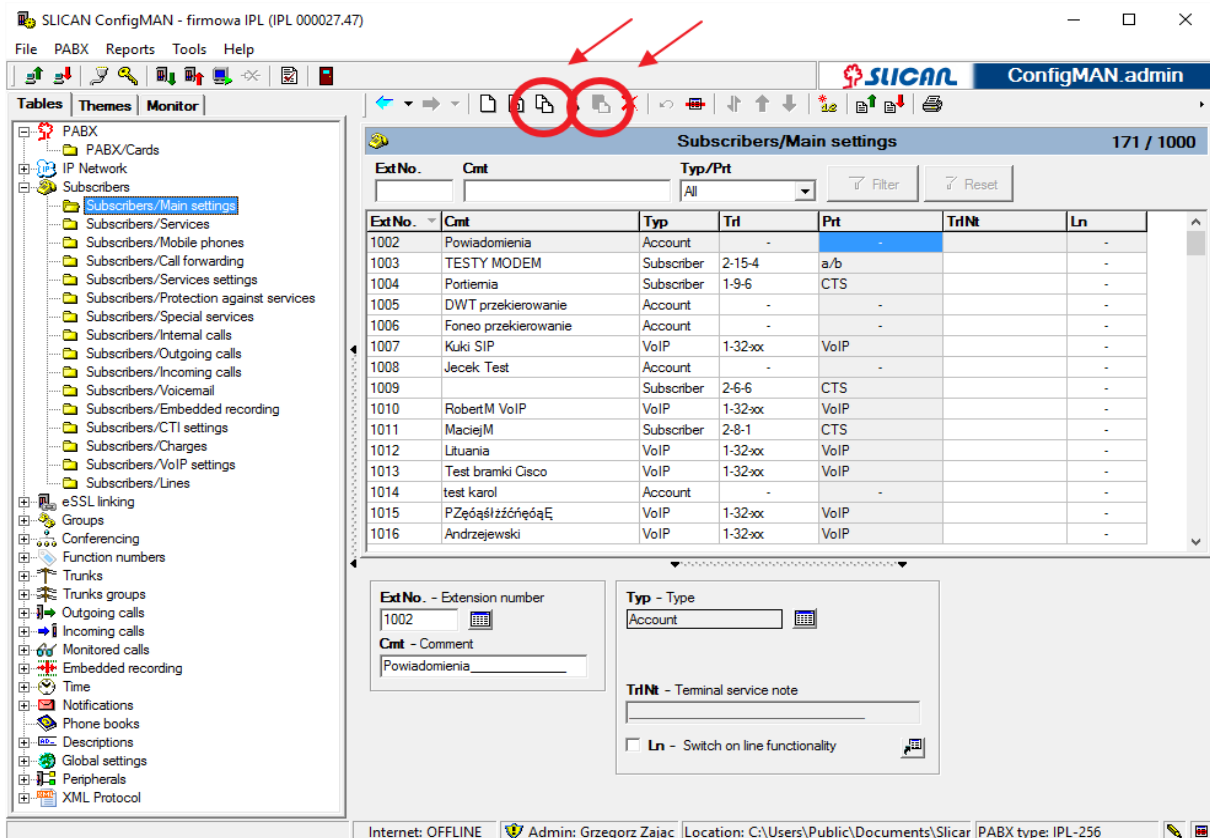
Fig. 2.10.: Copying records (rows)

Next, select the target record, and click the **Paste selected record** button, or press the SHIFT+CTRL+V keys combination.

Copy and *Paste* functions cannot be used while multi-editing.

Attention: Fields which must have a unique value cannot be duplicated or copied. (e.g.: ExtNum, Comment, Terminal)

Cut, delete record

The *Cut selected record* button, or the SHIFT+CTRL+X key combination works in similar way as the copy command i.e. stores the whole record in memory, but at the same removes it from the sheet.
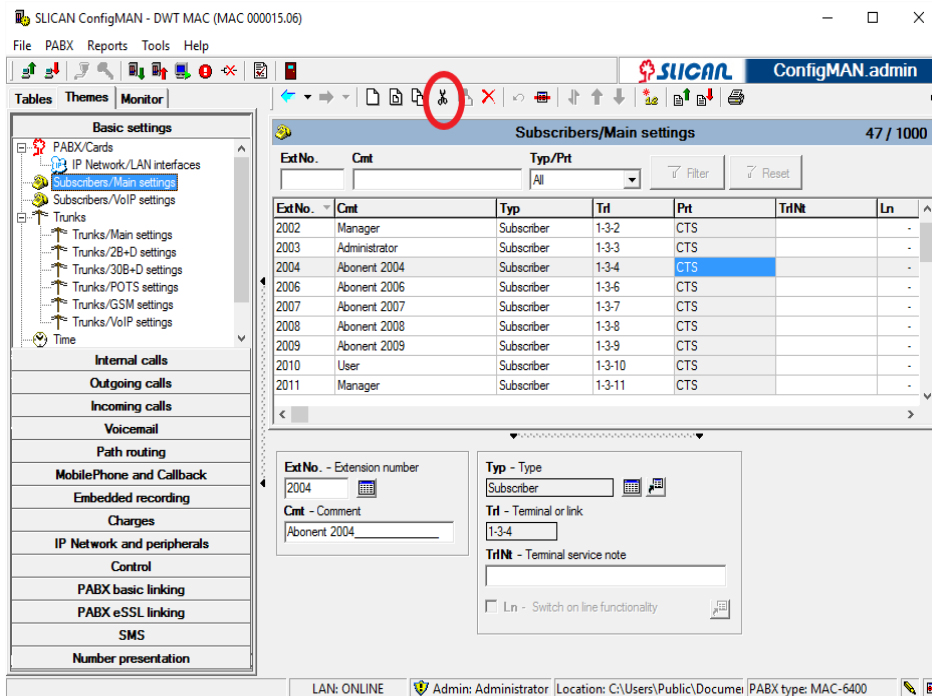
Fig. 2.11.: Cutting records (rows)

Option Delete selected records or the CTRL+DELETE key combination is used for deleting record (or records).
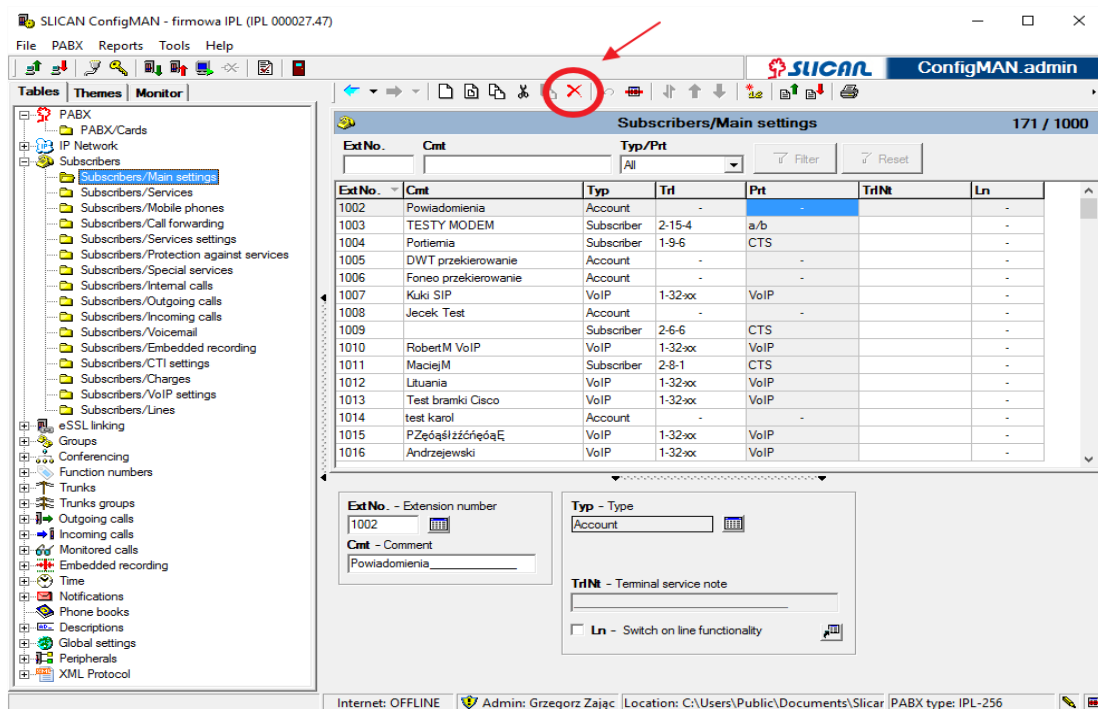
Fig. 2.12.: Deleting records (rows)

Duplicate

Duplicating records works by adding a new (additional) record to the sheet, filling it out with values from the selected (original) entry. To duplicate a record, select it, and then click **Duplicate selected record**, or use the CTRL+SHIFT+INSERT keys combination.

Fig. 2.13.: Duplicating records (rows)

Attention: *It is not possible to duplicate or copy fields which must be unique. (ex: ExtNo, Comment, Terminal)*

2.8 Keyboard shortcuts

ConfigMAN enables the user to move around the program without using a mouse. To do this, each important function in the program has a keyboard shortcut associated with it.

Mode	Shortcut	Function
Editing	Ctrl+Ins	Insert record
	Ctrl+Del	Delete record
	Ctrl+Z	Undo
	Ctrl+A	Select all
	Ctrl+F7	Verify configuration
	Shift+Ctrl+Ins	Duplicate record
	Shift+Ctrl+C	Copy record
	Shift+Ctrl+V	Paste record
	Shift+Ctrl+X	Cut record
	PgUp	Scrolling records up in table window
	PgDown	Scrolling records down in table window
Connecting to the PBX	Ctrl+F9	Login and password
	Ctrl+F5	PBX connection configuration
	F5	Launching system console
	F6	Launching ISDN measure
	F7	Upload to PBX
	F8	Download from PBX
	F9	Activating/Deactivating monitor
	F10	Analog phone simulation
	F11	Stop/restore monitoring
	Ctrl+BkSp	Disconnect
	F1	Help

	Ctrl+O	Open configuration
	Ctrl+S	Save configuration
	Ctrl+N	New configuration
	Alt+F4	Quit
View manipulation	*Ctrl+Alt+←*	Hide tree view panel
	Ctrl+Alt+→	Show tree view panel
	Ctrl+Alt+↓	Hide editing (form) panel
	Ctrl+Alt+↑	Show editing (form) panel

Moving around the different windows is shown in the diagram below.
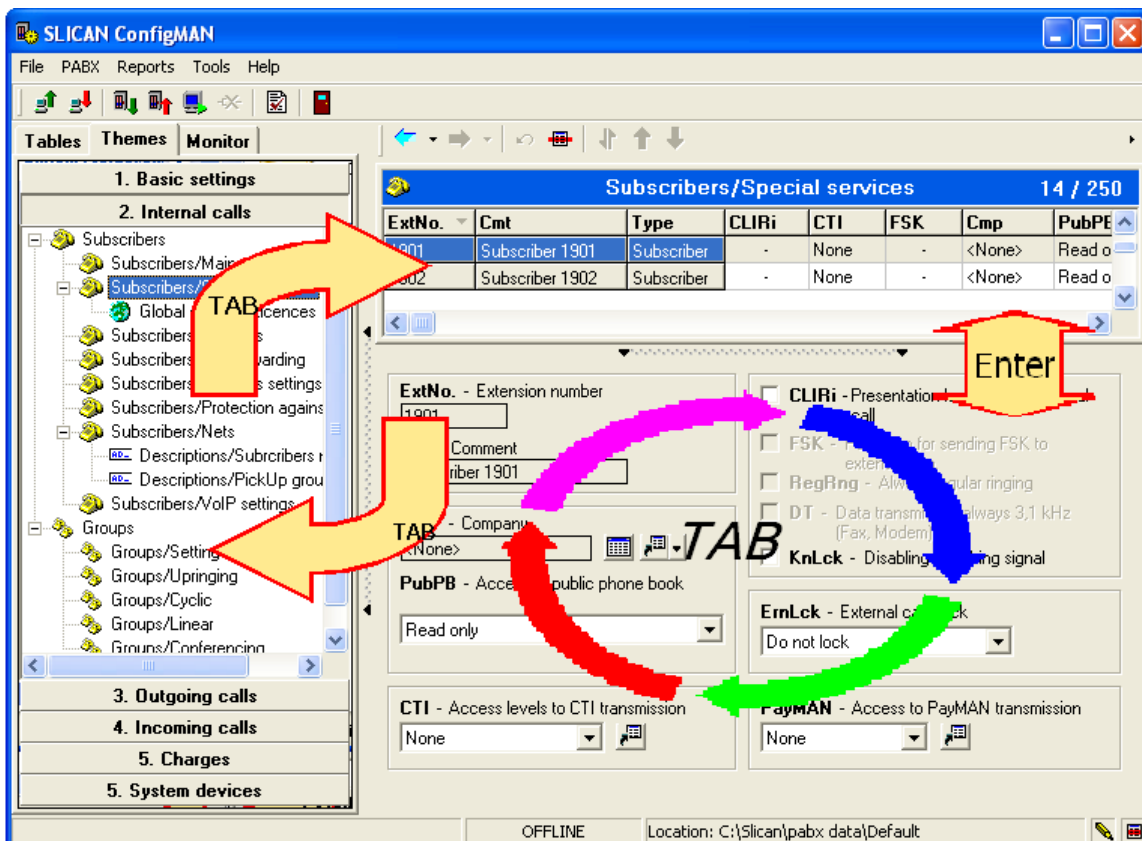
Fig. 2.14.: Navigation using keyboard

3 Backing up the configuration (PBX Back Up)

Creating a backup of the configuration

Before modifying the current PBX configuration it is recommended to make an up-to-date backup of the PBX.
The backup copy contains the following data:
- PBX configuration,
- data from public and private phonebooks,
- announcements uploaded by PBX administrator.

Backup files have a *.cpy extension.

To make a backup copy:
➢ connect to the PBX,
➢ from the *PABX* menu, select *Perform backup*,
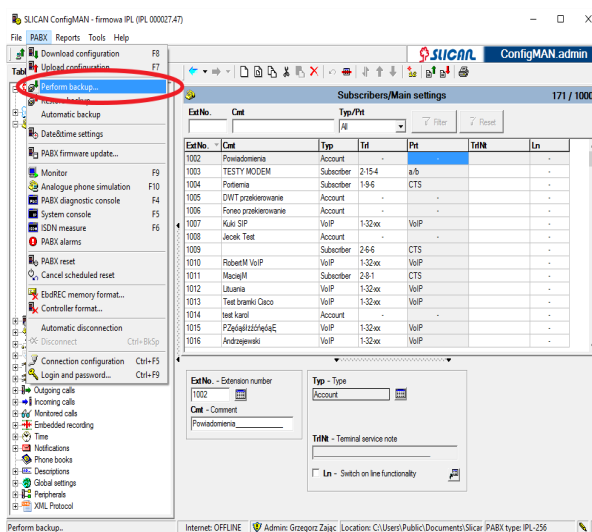
Fig 3.1.: Backup from PABX menu

➢ select a directory on the hard drive, and provide a file name,
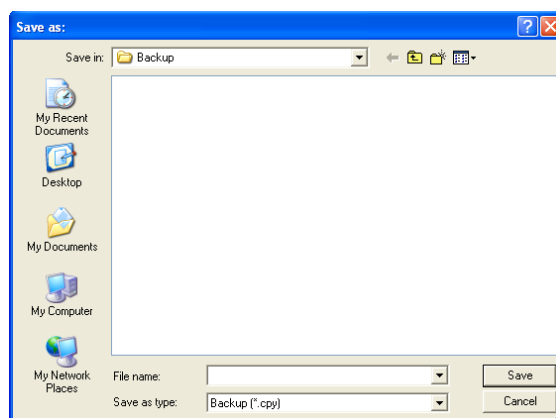
Fig 3.2.: Selecting location and name of backup file

After finishing backup, application informs about result of operation.
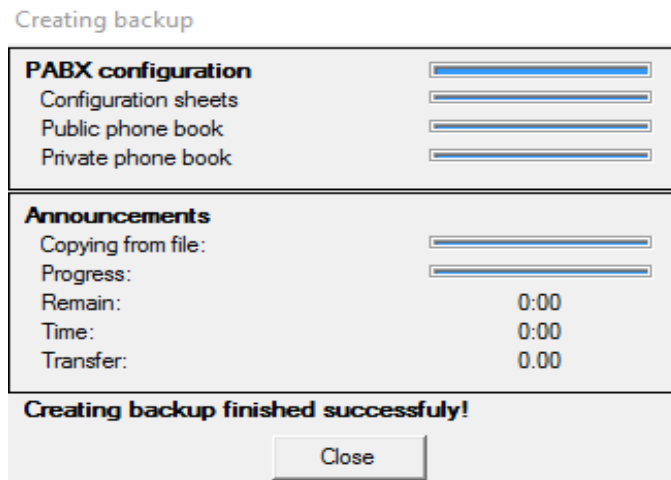
Fig. 3.3.: Backup process window

Automatic backup

If the option ***Automatic backup*** is selected in the ***PABX*** menu, the program, after each successful connection to the PBX, compares the date of the backup with the date of last modification of the configuration.
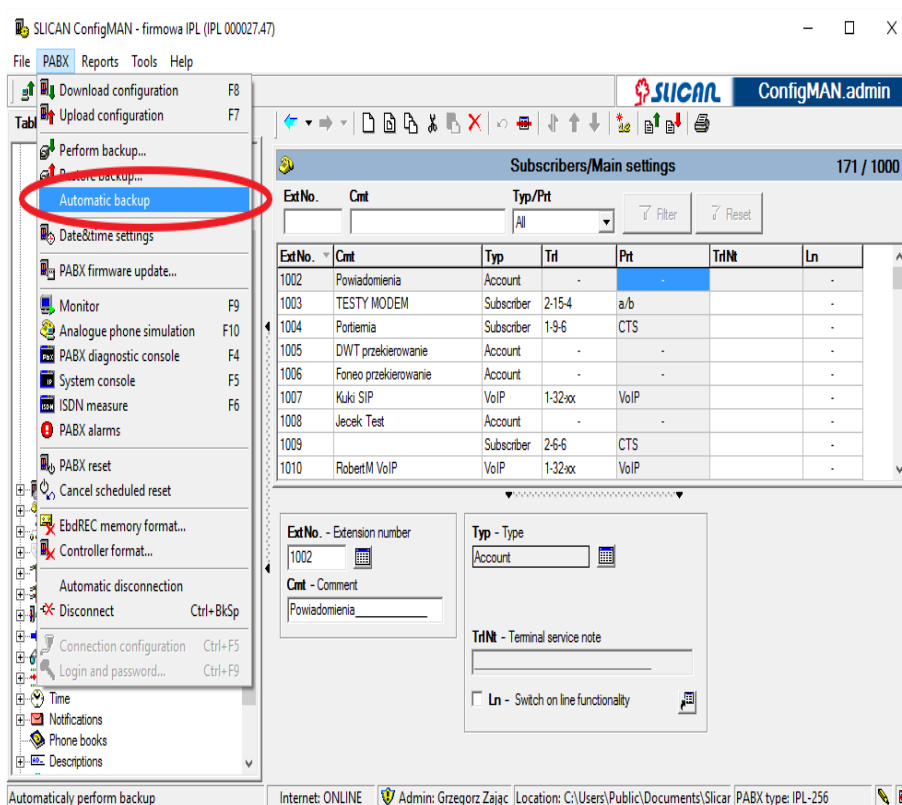
Fig. 3.4.: Selecting option – automatic backup

If the backup copy is older than one day, the program will automatically ask to update the backup copy on the computer's hard drive. There is an option in this window, permitting to switch off this reminder.
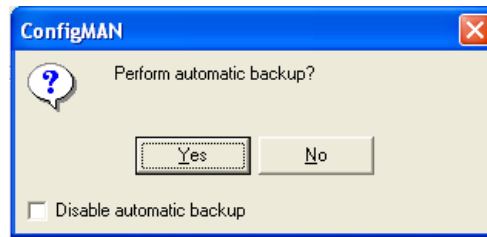
Fig. 3.5.: Perform automatic backup window

If the backup request is confirmed, the current backup copy will be saved as ***auto3_<no>.cpy***
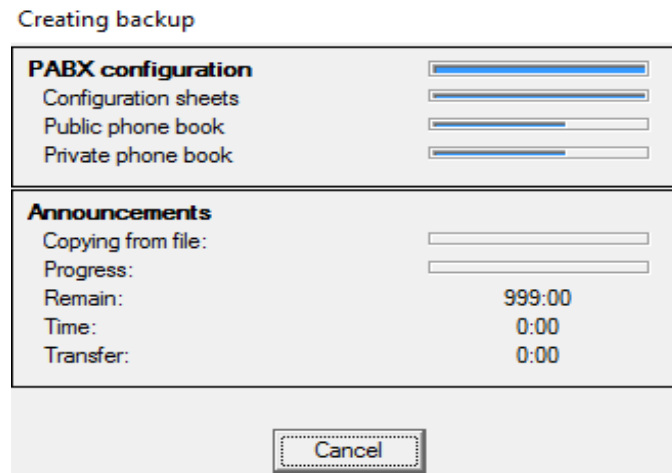
Fig. 3.6.: Backup process widow

To avoid creating too many files, the program creates only three backups. Each subsequent copy will overwrite the oldest existing file.
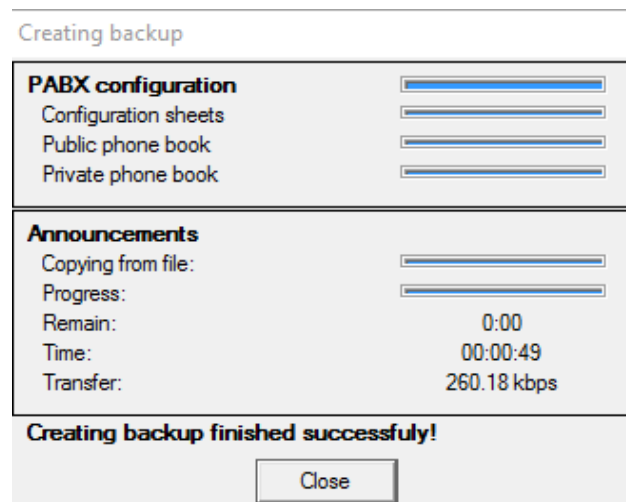
Fig. 3.7.: Backup process widow- backup finish

Restoring configuration backup

To restore a backup configuration, select *Restore backup* option from the *PABX* menu.
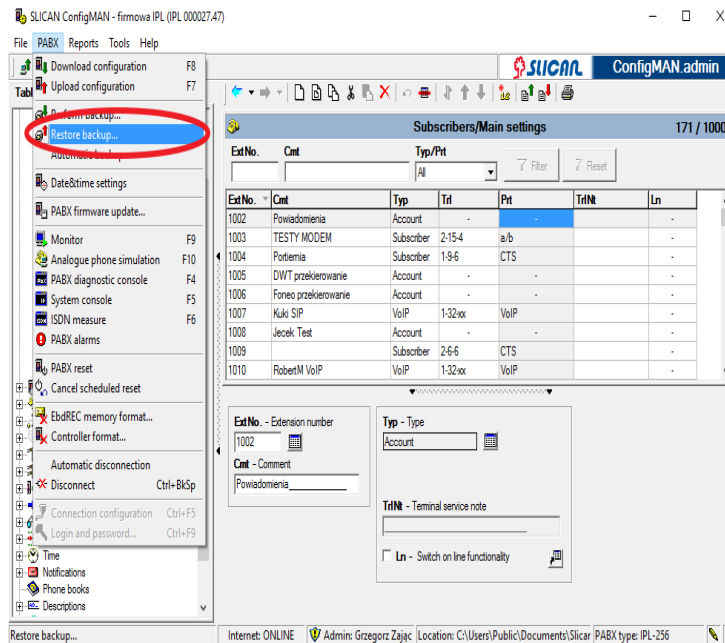
Fig. 3.8.: Selecting option - restore backup

Select the backup file you wish to upload to the PBX.
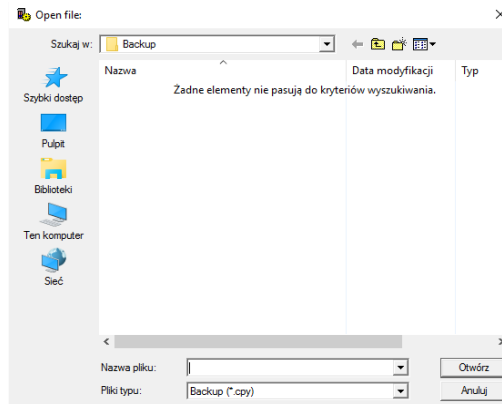
After completing the uploading operation, the program will display a success notification.
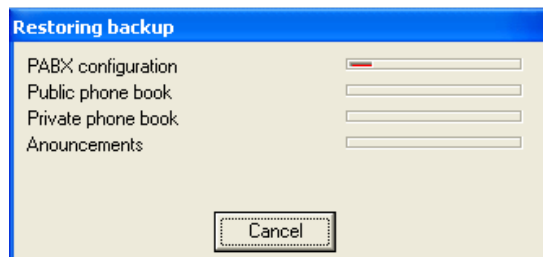
4 Programming the PBX

4.1 Programming a new PBX (Themes tab)

To program new PBX it is recommended to download empty configuration from PBX (to obtain information about installed hardware), next initial configuration should be created using **Hard reset wizard**.

> *Attention!*
> Whole functionality of Hard reset wizard is described in chapter 10.2Restoring PBX factory settings – Hard reset wizard.

After initial programming using **Hard reset wizard** PBX configuration should be personalized according individual user needs. Following actions should be performed:

- After connecting to the PBX, download the configuration. To do that, select *Download* from the *PABX* menu, or press *F8*.
- If the PBX is being configured for the first time, meaning the PBX is new, switch the tree-view to topic view. To do that, click the *Themes* tab.
- The window will show a list of keys – shortcuts to the various PBX programming themes groups.

In general, programming the PBX can be divided into several steps, shown on the flowchart below:
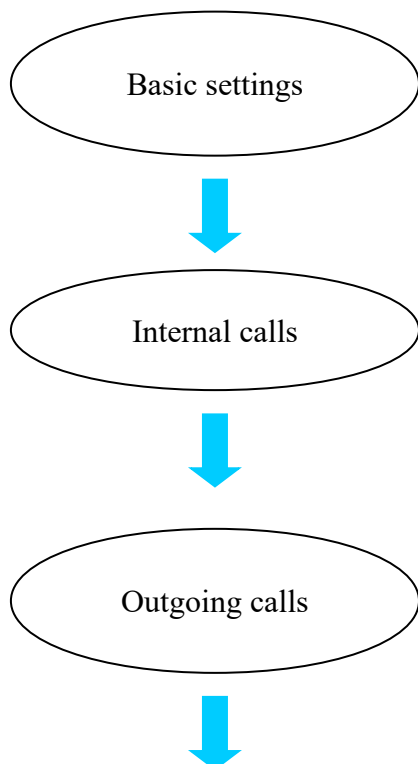
First programming step

Confirming the equipment installed in the PBX. Defining the parameters for the various internal and external ports, as well as the settings for the LAN card. Additionally, this step includes providing the subscribers with extension numbers, and setting the global parameters for the PBX.

For further information, see Chapter 4.2Basic settings.

Second programming step

Defining the rules of internal calls in the PBX. Assigning appropriate privileges and access classes to the extension numbers. Dividing the extension numbers into nets and groups. Creating groups. Defining the relations between nets and groups.

For further information, see Chapter 4.3Internal calls.

Third programming step

Defining access classes for the various users and groups. Defining the outgoing methods. Creating rules, defining the carrier selection (LCR). Grouping the trunks into groups and assigning privileges to nets.

For further information see Chapter 4.4Outgoing calls.

Incoming calls

Fourth programming step

Defining the rules for routing the various connections from PSTN network. Defining the rules for accepting calls depending on the time of day. Automatic routing of incoming connections. Relating DDI numbers with extension and group numbers in the PBX.

For further information, see Chapter 4.9Incoming calls.

Call path routing

Fifth programming step

Defining rules of dynamic and static path routing. Configuration of static path routing according CLIP and dialed number. Selection of trunk groups and LCR entries to dynamic path routing.
For further information, see Chapter 4.9Incoming calls.

Embedded recording

Sixth programming step

Setting for embedded recording for subscribers and/or trunk groups – if embedded recording system is installed in PBX.

For further information, see Chapter 5Embedded recording.

Charges

Seventh programming step

Outgoing calls charging method configuration. Defining limits and recharging amounts for specific subscribers.
For further information, see Chapter 6 Charges

System devices

Eight programming step

Setting up system phones, door-phones and audio devices.

For further information see Chapter 8System devices..

4.2 Basic settings

The *Basic settings* tables allow the user to provide *basic configuration* or the PBX modules. In the *PABX/Cards* sheet, equipment automatically detected as new installed or removed can be confirmed.

4.2.1 New cards – accepting in system

➢ Activating new equipment requires a download of the current PBX configuration. After installing the modules, and turning the PBX on, download the configuration, and the PBX will automatically activate the new equipment in it's database, which is presented on following figure.
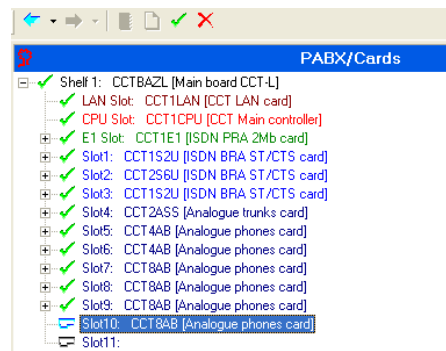
Fig. 4.1.: Confirming activated cards

➢ Confirming activated cards: to confirm a module, right-click on it, and select **Accept** from the drop down menu.

.
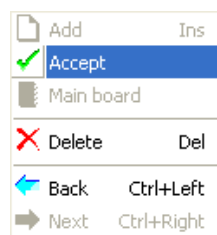
➢ After opening the **Cards tree**, make sure that the all newly installed modules have been activated, and that there's no equipment conflict in the PBX.
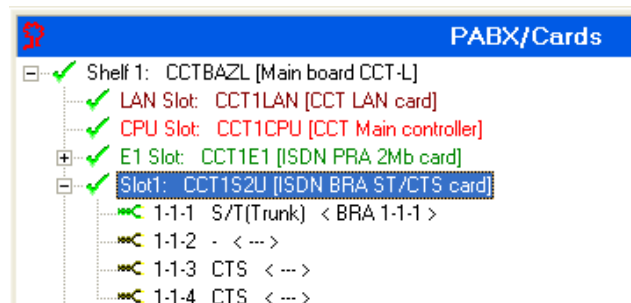
Fig. 4.2.: Promoting cards tree

ATTENTION!
If You insert CCT4S or CCT4S2U in *first slot,* third port on this card *(assigned as 1-1-3)* can work only as *subscriber.* In second and third slot cards CCT4S or CCT4S2U can work in any port configuration (subscriber/trunk). It means: maximal number of ISDN trunks in CCT-1668 PABXes is equal to 11, while number of ISDN subscribers is equal to 12.

IMPORTANT: Teletax modules are automatically activated in the PBX. There's no need to accept them. To confirm their installation, check the Trunks/**POTS settings** sheet (after selecting the proper analog port of a given card). After installing the TTX modules, new charging methods become available (*16kHz* and *16 kHz at start*).

4.2.2 Trunk configuration (external ports)

External ports of the PBX are created automatically during hard reset wizard according detected hardware. For every trunk one record is created in sheet **Trunks/Main** settings. The VoIP trunka are exceptions and should be created manually.

ATTENTION! Full functionality of VoIP card was described in chapters 19 and 20

4.2.2.1 Activating/deactivating trunk and configuring its own number

Field *OwnNo* finally identifies trunk in PBX. If specified trunk is switched on, it should has its own number (assigned by provider), followed by area number without zero (ex. 523251100). Trunk own number defined in field *OwnNo* is important for establishing calls defined in sheet Incoming calls/Settings/Main.

In field *Stu* trunk can be manually switched on or off.

Field *TrlNt* enables to create own notices concerned with specified trunk. Content of this field has no meaning for PBX configuration and can be changed freely.

4.2.2.2 ISDN BRA trunks (2B+D)

ISDN trunks require the selection of either PP or PMP - *L2* configuration. Presentation method for outgoing calls - *SCLIP* and for incoming calls - *SCOLP* should be also configured according information from provider.
DevTyp - Content of this field defines ISDN trunk operating method.
TE is regular trunk operating mode – operating with telecommunication provider, trunk only receives signaling from the other side (from upper exchange).
NT is trunk operating mode – trunk is master to the other side. This mode is often used while Slican operates with other PBXes. In this mode Slican PBX send to the slave PBX, power supply and trunk synchronization signal.

Attention!
  If you create link between two PBXes via ISDN trunk, in one linked PBX you should set NT option and in second – option TE.

L2 -   Defines terminal configuration for cooperating with provider.
        PP- Point - Point configuration used for DDI
        PMP- Point – Multipoint configuration used for MSN

Chn – This field enables number of trunk channels available for phone calls established by
        PBX. It can be used if remain channels are used by other devices connected before
        PBX.

ChrMth – Enables to define charging method of ISDN trunk:
        None – no charging is provided;
        by Charging Unit – call is charged according number of pulses received
        from provider; pulse price is defined in sheet *Trunks groups/Settings*
        by Currency – call is charged according information about call cost received from

provider.

SavMD – This option should be checked if provider doesn't set saving mode for this trunk. Checking this option enables to detect decay of 1st or 2nd layer.

SncLck – If, in a given situation, we don't want the trunk to be synchronized **with a upper exchange**, we can activate the synchronization lock. While connecting to many upper exchanges we can select exchange used to synchronization by synchronization lock for trunks connected to another exchanges.

FrcInb – If you check this field, PBX is forced to switch on acoustic directly after seizing specified trunk. This field may be useful while linking Slican PBX with PBX of other manufacturer, which doesn't properly send the signalization out of band.

E/O – Numbering receiving method for incoming calls via ISDN trunk
En-block – upper exchange send all digits together (in block)
Overlap – digits number, after receiving from upper exchange to dial number by PBX

SCLIP – defining CLIP method - number presentation method for outgoing calls
National – Presented number consist of area prefix and subscriber number;
Subscriber – Presented number includes only subscriber number;
International – Presented number consist of country prefix, area prefix and subscriber number;
Unknown – Presented number includes only subscriber number, this number is limited to specified digit number.

SCOLP – defining COLP method - number presentation method for incoming calls
National – Presented number consist of area prefix and subscriber number;
Subscriber – Presented number includes only subscriber number;
International – Presented number consist of country prefix, area prefix and subscriber number;
Unknown – Presented number includes only subscriber number, this number is limited to specified digit number.

ATTENTION: Edition of fields *E/O, SCLIP, SCOLP* is impossible if trunk is used for PBX linking.

CmtOut – Sending comments using UUS signaling.
UUS signaling is one of services offered by ISDN, usually this service is offered as default by telecommunication providers. Setting any option in this field enables to send additional information, ex. caller extension number or comment by public ISDN network. If Slican PBX with appropriate software version is target terminal, this information will be added to standard CLIP number. It concerns mainly CTS phones but this information is sended also to analog and ISDN phones. It can be also displayed by other devices if they can services UUS signaling.

CmtIn – Receiving comments sended by UUS signaling.
This field enables to configure if and what information transmitted by public network should be displayed on display of phones working in phone network.

4.2.2.3 E1 trunk – ISDN PRA (30B+D)

ISDN PRA (30B+D) trunk is configured identically as 2B+D trunk.

For 30B+D you can perform the same steps as for 2B+D trunk.

As a difference you should note the value of field **Chn.** Slider enables changing number of channels available for this trunk (ex. If you purchase trunk 15B+D)

4.2.2.4 POTS trunks

The selected trunk requires the following settings, received from the provider:
- dialing mode – field ***DlMd*** (tone or pulse),
- charging mode – field ***ChrMd***.

Trunk charging mode can be set as follow:
- *Poling* –if starting a call is signal by polarity change
- *By charging time* – charging time starts specified time after dialing a number ***ChrOfs***. This time is calculate starting from dialing last number digit or fulfilling condition concerning digit number, defined in sheet ***Tables/LCR / Settings.***
- *16 kHz at start* – starting a call is signal by first 16kHz pulse received from provider,
- *16kHz* **–** starting a call is signal by first 16kHz pulse, than charging is perform using subsequent 16kHz pulses, received from provider. Pulse unit cost is defined in sheet ***Trunks groups/Settings.***

Remain settings of POTS trunk:
- **FSK** – after switching on this option, on specified trunk presentation of calling number will be received, this presentation would be transferred to internal subscribers. Switching on this option increase time to DISA announcement.
- **DlDis** – after switching on this option trunk dialing tone detector is disabled.
- **BsyDis** – after switching on this option trunk busy signal detector is disabled.
- **RejEn** – after checking this option, caller via analog trunk can hear busy signal. When this field is unchecked, caller hears dialing signal.
- **FlsEn** – after checking this field caller on analog line is possible to use Flash service on this trunk (*85), Flash duration can be defined in field **FlsTi**.
- **FlsTi** – defines Flash duration in ms.

4.2.2.5 GSM trunks

For GSM trunks using it is required to define only one setting in sheet ***Trunks/GSM settings***. This required setting is ***PIN*** code for SIM card used by specified trunk. Additional (optional) setting is manual provider selection in field **Prd** and echo cancellation activation in field **EchCnc**. Earlier, trunk own number should be specified in sheet ***Trunks/Main settings***.
It is very important to fulfill the field named **ID** – *GSM gate identifier*, content of this field is an information and is fulfilled automatically by PBX. It is used during configuring SMS sending and receiving in TelefonCTI.plus application.

4.2.2.6 VoIP trunks

In Slican systems it is possible to set **up to 100** logical VoIP trunks. For such trunks, following parameters should be set up:

Chn – defines number of VoIP channels (simultaneous connections), which should be done via specified trunk. Defined here channel number could not be greater than channel number defined in sheet *PABX/Cards* using button *Channel configuration* located in editing panel for slot, where VoIP card is installed;

Ptl - type of VoIP protocol (SIP or eSSL can be selected),

Cdc – audio codec, which can be used in connections through specified VoIP trunk (GSM, G.711a, G.711u or G729 can be selected),

– **CnnTyp** – this field is important if VoIP trunk is used to linking PBX with other *Slican CCT-1668* or *MAC-6400* PBX using e*SSL* protocol – detailed information can be found in chapter 21.3. PBX linking using VoIP.

– *Adr* – provider server IP address, if using VoIP provider services or PBX used as server, if PBX is linked with another PBX. In this field instead of IP address, domain name should be also entered. If we define IP address, after semicolon we can also enter port number (the same is valid for entering domain name), if this port number is different than port defined in field **PrtSIP** in sheet *IP Network/VoIP configuration.* Default ports are:
5060 for SIP protocol

- *Dmn* – domain (set up <u>only for SIP protocol</u> if domain is different as server IP address, this info is supplied by VoIP provider), however some providers require in this field repetition of data entered in field *Adr*;

- *Usr and Pas* – login and password to access to VoIP provider or used while linking PBXes via VoIP (in this situation content of this field should be the same in both linked PBXes),

- *DTMF* – enables to set up *DTMF* transmition mode while using VoIP protocol. It is possible to choose one of three options depending on standard used by VoIP provider, another PBX or VoIP phones:
  - ➔ for SIP protocol:
    - *RTP (RFC 2833)*, *SIP (INFO)* [out band],
    - in band.
  - ➔ for eSSL protocol:
    - *eSSL (out band)*,
    - in band.

Attention!
Sometimes it can be noted, that human voice contains harmonics the same as DTMF tones. Voice of this type can be the reason of disturbances as a heared DTMF tones, if for VoIP trunk option „out band" is used. Solution in this situation is setting option „in band".

– **CLIP –** in case of active DDI service on trunk – provider's SIP type presentation in nicoming and outgoing traffic:
  - ➔ Default
  - ➔ Remote-Party-Id
  - ➔ P-Asserted Identity

- **IDPfx** – enables for adding prefix to DDI number to make International format: ex. +48XX XXX XX XX
- **CmtOut** – sending comment in SIP signaling. Setting any option in this field, enables to send some additional information ex. Extension number or caller comment. If targer device is Slican PBX with appropriate firmware version, such information is added to presented CLIP number. It concerns mainly CTS phones, but it can be also realized by some analog and ISDN phones. It can be also displayed by other devices, but only if signaling maintenance to use with some VoIP providers, possibility of sending own comment was added.
- **CmtIN** – Receiving comments sent in SIP signaling. It enables configuring if and what information sended via public network should be displayed on phones used in telephone network.
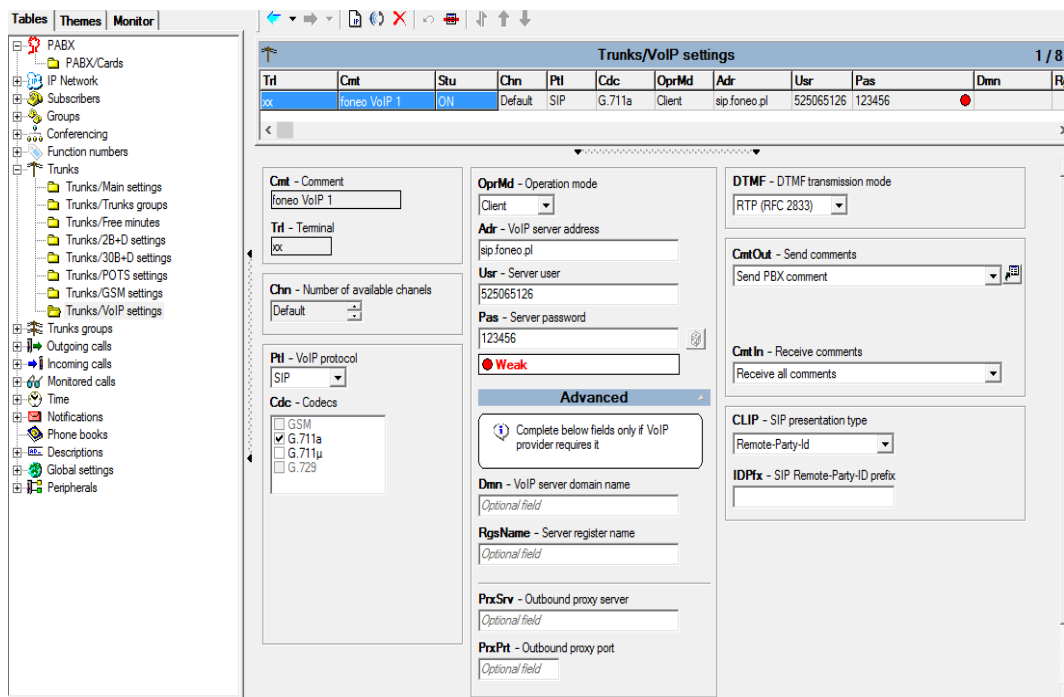
Fig. 4.3.: Trunks / VoIP settings sheet

Advanced – settings used only if VoIP provider requires it:
- **Dmn** – domain set for SIP protocol only, in case when its different than server IP address
- **RgsName** – filled if name for registration is different than in **Usr** field
- **PrxSrv** – proxy server address used by VoIP provider
- **PrxPrt** – ports used by VoIP provider's proxy server

VoIP trunks are used to maintain incoming and outgoing calls, except situation when fields: **Adr**, **Dmn**, **Usr**, **Pas** are not defined. In this configuration trunk can be used only for maintaining calls incoming from all anonymous users. It enables to attain extension number by any person equipped with any IP phone, operates with protocol defined for this trunk.

4.2.3 Global settings for PBX

The *Global settings/Remain* sheet allow the user to set the general operational parameters of the PBX. An *ArPfx* - Area prefix, has influence on phone numbers presentation format. Filling out the field will cause the local calls to be presented using 7 numbers, and the long-distance calls - 9 numbers. In case we would wish to have all calls presented using 9 numbers, the *Area prefix* field should remain empty.\

Field **SMSTrn** enables to define which trunk is used to sending SMSes to MobilePhone, about missed incoming calls.

Field **PgnAnn** enables to define announcement which will be played after establishing call with audio device working in *paging* mode, ex. connected to the paging system (setting of working mode can be done in field **Typ** – *Type* in sheet ***Peripherials / Audio*** – detailed information can be found in chapter 8.6 Audio devices configuration). Available settings are:

<None>: After establishing call audio channel is immediately switched on – caller voice can be hears in paging system loudspeaker immediately.

Gong: after establishing call, caller hears melody – in the same time in paging system loudspeakers triple-tone sound is played, after melody caller hears short beep and than he/she can speak any message.

Announcement: after establishing call, caller hears melody – in the same time in paging system loudspeaker selected announcement is played, after melody caller hears short beep and than he/she can speak any message.

4.2.4 Admin and user accounts

The PBX can support up to 25 users and admins. By default, one admin account is created, with full access to the PBX (so called "root").

IMPORTANT! *In new PBX, or after performing hard reset default admin account settings are as follows:*

- *log-in:* **admin***,*
- *password: four last digits of PBX serial number,*
- *code[1]: four last digits of PBX serial number.*

Creating users and additional admins is done by inserting new rows in the ***Global settings/PABX access codes*** sheet, defining login (field ***Log***) and password (field ***Pas***).
Login can consists of letters (small and capitals) and digits, login length should be from 4 to 16 characters. Password is limited in the same manner and can include special characters excluding "~" , ";" and <space>.

IMPORTANT! *After first run configuration, above data needs to be changed. Default login and password will result in critical error during uploading the configuration.*

---

[1] Used b

Fig. 4.6.: Sheet – Global settings/PABX access codes

Content of field **Cmt** has only information meaning – enables to identify persons with granted access to PBX and privileges of these persons. It is possible to define up to 25 access accounts, although it is required to define at least one user with admin privileges (privileges to using application *ConfigMAN* are set in field **CM - *ConfigMAN***). Entered here comment is used in PBX monitor to identify events and to define person who is currently connected to the PBX.

In PBX many users can be defined and their privileges to system managing are defined in field **CM** – *ConfigMAN*:
– *<None>* – user of such account has no privileges to PBX managing (can't change PBX configuration),
– *User* – has possibility to manage PBX but only using application *ConfigMAN.user* – limited access,
– *Admin* – has full possibility to managing PBX using application *ConfigMAN.admin* (if this user connect to PBX using application *ConfigMAN.user,* privileges will be limited – some sheets are not visible and some fields can't be changed). Admin doesn't see passwords (there are asterisks instead of letters), admin can of course change passwords Admin also can't create accounts for other admins and cant set the time fir temporal accounts.
– *Admin (root)*- full access and full privileges, including passwords review and creating next admin accounts.

Every person (admin or user) - depending on needs - following privileges could be granted (by checking appropriate field):
– ***BM*** – it enables using BilingMAN application and receive data from call buffer
– ***PM*** – it enables using PayMAN application
– ***Ann*** – it enables recording voice announcement by phone using service *92
– ***Sw*** – it enables changing PBX working mode using service *87 (this service can be recalled from internal phone and via DISA)
– **Lmt** – after checking this field you can define date and time – fields **LmtDt and LmtTi** – of account expiry (it require root privileges)

Fields **Ann** – *Announcement recording* and **Sw** – *Working mode switching* are available, if for this account 4-digit code (field ***Cd***) was previously set, this code authorize for using phone services. Unique code and unique login should be assigned to every person.

4.3 Internal calls

4.3.1 Extension numbers: creating and setting privileges

4.3.1.1 Creating new subscribers (physical extension numbers)

In Slican PBXes, a physical extension number (together with phone) and an account (virtual extension number) are treated as equally. All privileges, which can be assigned to a physical **extension number**, are also valid for the **account,** which doesn't have any equipment permanently assigned. Following types of extension numbers can be defined in system:
– *Subscriber* – extension number equipped in:
    analog phone,
    ISDN phone (with MSN main number),
    one of system phones – CTS series,

- *Account* – defines user without any equipment permanently assigned (without individual phone). Account may be recalled on any phone in PBX. If any user is equipped in individual phone, there is no necessary to create an account for this user, because it is possible to recall user's privileges on any other phone.
- *MSN* – user equipped in ISDN digital phone, to create MSN accounts at least one ISDN port should be defined as internal,
- *VoIP* – user equipped in any (except system) IP phone, or software (using protocol SIP).
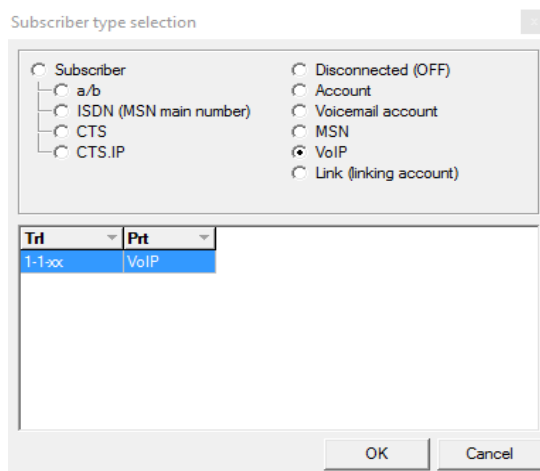
Fig. 4.5.: Adding new internal user in PBX – subscriber type selection

ATTENTION!
Full functionality of VoIP card was described in chapters 19 and 20
VoIP subscriber ports are subject of licensing. Information concerning licensing can be found in chapter 17.

Subscriber are created only when appropriate card is detected by system, accounts can be added manually in sheet **Subscribers / Main settings**. Manually can be also added subscribers of type **VoIP** or **Voicemail account**. Any type of subscriber, which is not used in specified moment can be switched off without removing it.

For any created subscriber – extension number – **ExtNo.** Can be changed, comment – **Cmt** can be added and type – **Typ** can be changed.
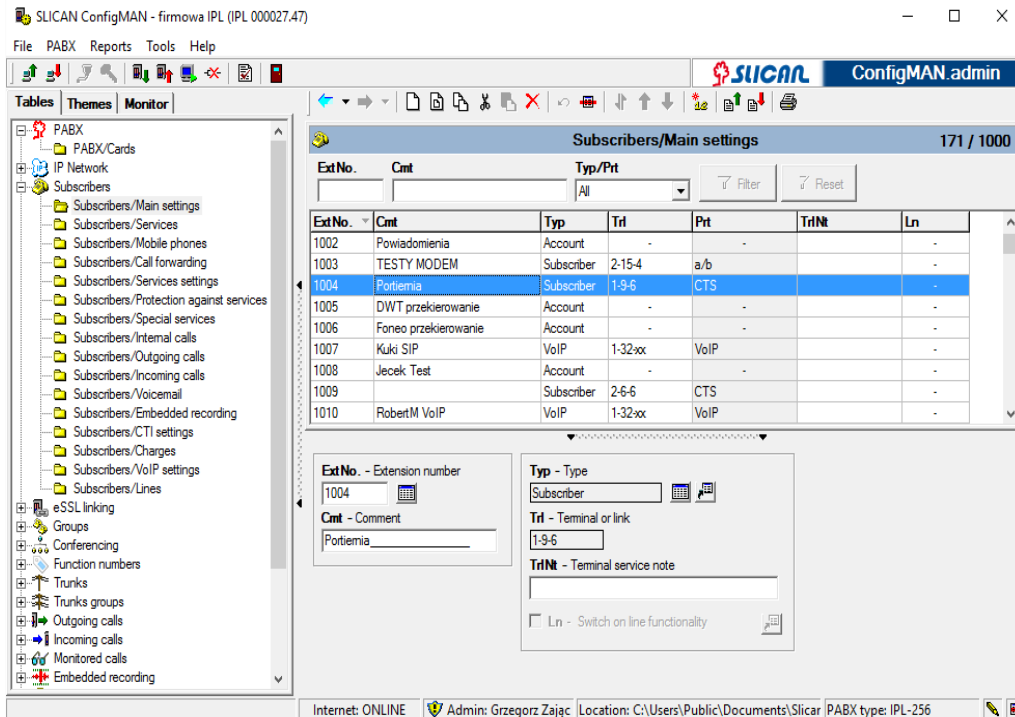
Fig. 4.6.: Sheet Subscribers/ Main settings

For VoIP subscribers special settings can be done in sheet **Subscribers / VoIP settings**.

ATTENTION! *Full functionality of VoIP card was described in chapters 19 and 20*

For **VoIP** subscribers following options can be set up:
- **Ptl** - used protocol type (SIP),
- **Cdc** - available audio codecs,
- **Pas** – password for log in to the server in PBX, it is possible to create password, using icon located next this field,
- **DTMF** – DTMF transmission mode (according standard serviced by user device), selection possibilities are described in chapter **VoIP trunks.**
- **DfnUlg** – checking this field for specified subscribers means, that PBX manages this subscribers phone as damaged,
- **PrmCnv** – checking this field enables to convert polish diacritical signs into similar signs from Latin alphabet.

4.3.1.2 Privileges to services

To every extension number (subscriber or account) privileges to PBX services can be assigned. Any extension number can be granted with privileges to activate some services. In **ConfigMAN** it can be done in sheet **Subscribers / Services settings**.
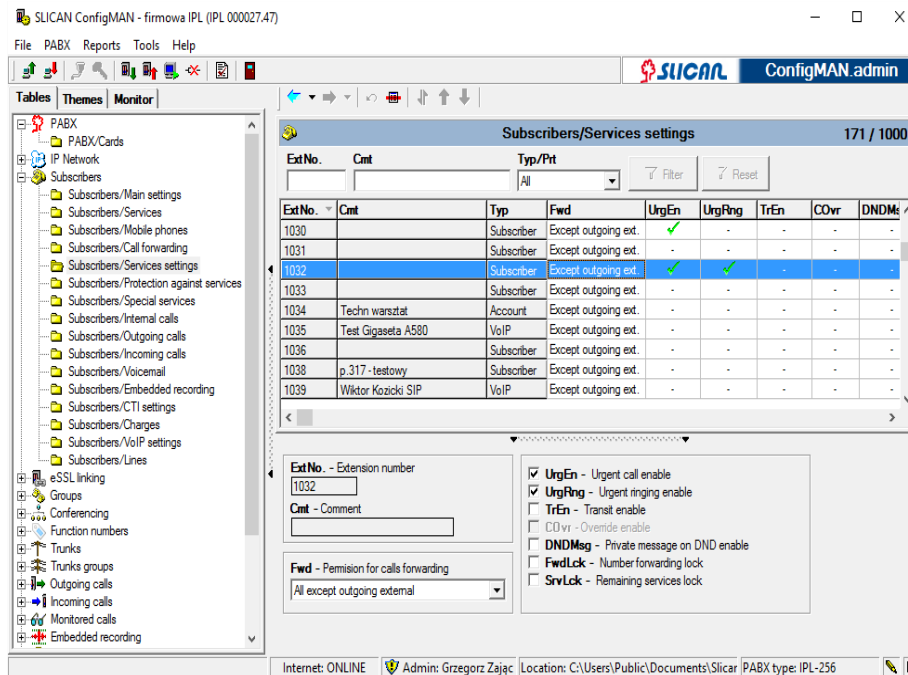
Fig. 4.7.: Sheet – Subscribers / Services settings

– **UrgEn** – urgent call enable – defines if from phone with this extension number *urgent calls* (*25) can be done

– **UrgRng** – privileges to urgent ringing - this settings defines if subscriber from this extension number will able to use *urgent ringing* service (*35). If this service is used while dialing extension number equipped with analog phone special ringing will be hear (other than standard ringing), for extension numbers equipped with system phones caller immediately will be hear in speakerphone. This service <u>is not realized</u> for ISDN phones. Service *urgent ringing* can be used only while dialing subscriber, which is member of the same PickUp group. Although, if extension number of subscriber using *urgent ringing* service is defined as VIP (sheet **Subscriber / Outgoing calls**), PickUp group is not checked and this subscriber is authorized to dial any extension number using service urgent ringing.

– **OutFwd** – this settings defines if from this extension number will be possible to forward outgoing calls. It concerns situation when outgoing call is established from extension number and then this connection is forwarded to other extension number. For marker **OutFwd**=0 (no authorization to outgoing calls forwarding) caller is not able to hold on an outgoing call to consult with anybody. If it is outgoing call forwarded by somebody, this marker is not valid and subscriber can hold such call as well as forward them. So **OutFwd** field enables to forwarding call by originator.

– **TrEn** – transit enable – this settings enables to establish connections of type external – external (to establish such connection field **OutFwd** for specified extension number should be checked) using services of forwarding to external numbers and establish outgoing calls from temporary external registered account. While establishing connections of type external - external <u>at least one trunk</u> must be <u>digital</u> trunk (ISDN, GSM or VoIP). Maximal duration time of transit call can be set up in field **TrTi** in sheet **Global settings / Outgoing calls.** Transit calls are distinguished from other call in billing reports created using *BillingMAN* application.

- **COvr** – call *override* – checking this option enables *override* another call if any part of this call has no marked option *Protection against override* in sheet **Subscribers / Protection against services.** Call override can be realized only from system phones.

- **DNDMsg** – enabling to record and use private DND message. Such message can be recorded (up to 30 seconds) using service *79. This message is played only when DND service is active on specified extension number. Checking this field is is equal to create new entry in sheet **Announcements** and reservation of time to record private message in system. If after enabling DND service private message or standard announcement is played depending on setting field **DNDMsg** in sheet **Subscribers / Call forwarding.** In this field private DND message service can be also switched off.

- **FwdLck** – number forwarding lock – this option defines if on this extension number forwarding services (*70, *71, *73, *74, *75, *76) could be enabled. Checking this option lock mentioned above services.

- **SrvLck** – remaining services lock – this option defines if from this extension number will be possible to using all remain services, do not individually restricted - only one exception is service *52 - activate account temporarily. This option must be checked if from specified extension number following services should be available: *21 – phone lock, *22 – DND switch on/off, *27 – private call, *31 – park/unpark a call, *36 – intercept ringing, *37 – intercept ringing in PickUp group, *53 – permanently account activate, *87 – manual working mode changing, *92 – playing and recording announcements, Flash*3 – establishing three party conference.

4.3.1.3 Call forwarding

In ConfigMAN application it is possible to define internal or external call forwarding number and type of forwarded calls. Forwarding settings can be realized in sheet **Subscribers/Call forwarding**.
It is possible to define following values for field **FwdStu:**
  - ➢ Unconditionally
  - ➢ While busy
  - ➢ When no answer
  - ➢ While DND

For above options there are possible following values for field **FwdTyp:**
  - ➢ All
  - ➢ Incoming direct
  - ➢ Internal
  - ➢ Incoming announced
  - ➢ Group calling only (Forwarding of this type are unavailable when extension number is busy).

If we want to realize call forwarding with set up option *When no answer,* additionally we can define number of ringing before forwarding. It could be done using field **FwdTi** in sheet **Subscribers / Call forwarding** or using service *71.

In the same sheet we can check status of **DND** - do not disturb service. This service can be activated from phone by selecting *221.
Additionally in field **DNDMsg** You can select one of following options:

- *Standard* – for internal calls voice message „Subscriber is temporarily inaccessible, please, call again later" is played and for incoming calls, caller hears busy signal,

- *Announcement* – one of DISA/Infolines announcements with number 0-9 from sheet **Announcements** is played,

- *Private message* – individually, using service *79, recorded message is played.

Even if DND service activated on specified phone, it is possible to activate on this phone service *57 – I am there. All calls for extension number with DND service activated are routed to extension number with activated service *57 (with this extension number defined as a parameter) even if besides DND service on this phone call forwarding service is also activated.

ATTENTION!
*For extension number with active DND service only calls from extension numbers with VIP privileges (sheet **Subscribers / Outgoing calls**) can be established if before dialing number, service *25 – Urgent call is selected (for CTS system phone this service can be assigned to programmable button or selected from phone menu).*

4.3.1.4 Internal calls

Nets in Slican PBX are kind of extension number set with appropriate privileges and restrictions for internal calls. Any subscriber or group can be assigned to one of eight predefined nets. Additionally to each subscriber privileges for calling to another nets can be assigned. Only one exception is group calling. Group agent can be an extension number assigned to net with no calling possibility. In this situation group privileges are more important than individual subscriber privileges.
In PBX can be maximal 8 nets assigned as A to H. Nets definitions can be found in sheet *Descriptions/Subscribers nets.*

4.3.1.5 MobilePhone

MobilePhone it is a function concerned with integration of fixed phone with GSM or DECT phone. This integration is different depending on type of mobile phone (additional or parallel).
- external number ex. cellular phone – PBX establish calls via appropriate trunk
- extension number ex. wireless phone - PBX establish calls between internal ports
Depending on configuration, calls can be routed to master and slave phone simultaneously or only to slave phone.

Depending on direction of call MobilePhone functionality is following:
- incoming call answered on MobilePhone can be transferred to any PBX extension number
- call from MobilePhone to PBX – privileges as for master phone are assigned to MobilePhone
- in both direction – during call with MobilePhone, master phone is busy

Description of sheet **Subscribers/MobilePhone**:
- **MblPh** – number of subscriber mobile phone
- **MblEn** – authorization of MobilePhone service using
- **RngTi –** time to up-ringing mobile phone

- **Rng –** after checking this field mobile phone will be up-ringed after time defined in field **RngTi**
- **RngDND** – after checking this field call will be immediately forwarded to mobile phone if DND service is active on master phone
- **MsdSMS** – in this field you can define if mobile phone (if it is cellular phone) should be informed via SMS about unanswered call. In SMS text there is information about data and hour of call, number of caller and information if call was unanswered or was routed to busy phone. SMSes are send via trunk defined in field **SMSTrn,** in sheet *Global settings/Remain*.
- **MsdMbl** – limits sending SMSes only to these phones which have active service MobilePhone.
- **ClbEn** – after checking this field it is possible to log in and callback service based on MP number. Details in chapter [4.3.1.6. CallBack service](#).
- **RcvSMS** – selection of SMS type informs about number of caller of received call
- **VmlSMS** – selection of SMS type informs about new message in voice mail box

If system phone is a master phone, than on its display caption MobilePhone is displayed when service is active. Additionally if mobile phone call to PBX extension number, master phone number is presented on target phone and if system phone is a target phone, caption MP appear in upper right corner of display.

Assumptions of MobilePhone service
- MobilePhone service is overriding service to any other call forwarding, but priority of this service is lover than "I am there..." service priority,
- calls to MobilePhone are established without checking of privileges and allowed prefixes,
- Mobile phone number can be entered and changed only from inside ConfigMAN application,
- connection between PBX and GSM mobile phone is established always according LCR rules,
- answering or rejecting call on master phone or mobile phone stops ringing on both phones,
- after answering call on mobile phone both phones (master and mobile) are busy for incoming calls until finishing call,
- during transferring back call from MobilePhone to PBX, master phone is busy,
- switching service on and off as well as its configuration can be done using master phone on mobile phone (during DISA announcement).
  - *77 0 – switching off mobile phone up ringing
  - *77 1 – switching on mobile phone up ringing
  - *77 2 – switching on mobile phone up ringing only when no answer
  - *77 3 – switching on mobile phone up ringing while DND (DND announcement isn't played)

Transferring back call answered on GSM mobile phone to PBX:
- if it is needed to transfer call, currently established on mobile phone call should be finished, from this moment there is 1 minute to transfer this call, during this time:
  - during 15 seconds there will be silence
  - during next 45 seconds route signal can be hear
- if during this time connection between mobile phone and PBX will not be established, waiting call will be interrupted (disconnected),

- during this 1 minute time number to connected with extension number to transferred to should be dialed (ex. DDI number or extension number dialed during DISA announcement), such call can be advised or not,
- your interlocutor will be hear standard waiting melody and voice message "please wait for connection", next ringing signal until answering call,
- now you can advise call and disconnect yourself or disconnect yourself during ringing signal,
- your interlocutor will be connected with dialed extension number immediately or - while dialed extension number busyness – will be wait until target phone will be free,
- transferred call will be disconnected if dialing time after forwarding will be exceeded. If call is transferred by mobile phone, retrieval of forwarded call is not possible.

ATTENTION!
MobilePhone service is subject of licensing and without purchasing the license code only one extension number in PBX can use this functionality. CallBack functionality is licensed separately.

Using MobilePhone as a MirrorPhone.

If as a number of MobilePhone, PBX extension number is defined (ex. DECT phone) extended system of MirrorPhone can be obtained, such system enables to:

- logical binding system phone CTS-202 or CTS-102 with analog phone. Due to different method of communicating between phone and PBX physical binding is not possible,

- present the same phone number (internal or external) for both phones.

- set second phone in busy status while from first one (any of phones) any call is established,

- cover (override) extension number call forwarding without deleting them. MobilePhone deactivation restores forwarding settings.

Example:

Subscriber wants to use wireless phone during work hours and after these hours all calls should be routed to mobile phone. If as a MobilePhone extension number of wireless phone will be entered and and call forwarding to mobile phone will be set, when MobilePhone service is activated call forwarding not working, but switching off MobilePhone service automatically activates call forwarding. There is no need to enter number and activate call forwarding service every time.

4.3.1.6 CallBack service

Callback is a service which enables to order call from PBX, cost of this call is assigned to caller extension number. CallBack service acts in following way:

- from phone defined in PBX as MobilePhone you should call to defined earlier number of trunk connected to PBX,

- PBX disconnects this incoming call,

- after short time PBX calls back to MobilePhone number. Call is realized according settings in LCR table.

- After answering call standard dial tone can be hear. During this signal target number (internal or external) can be dialed.

Below you can find examples of few benefits (not only financial) of using CallBack service:

- Transit function acting, establishing business calls is limited only to extension numbers or other phones defined in PBX data base. There is no danger of losing password and hacking system.

- As service user is not charged with any connection fee, it is possible to define private GSM numbers as MobilePhone, there is no need to cover license fee and continuous increasing number of business mobile phones.

- Call in both directions is established by PBX, so to target subscriber there is no send caller presentation. Target subscriber thinks that caller is located in company's premise. Due to this functionality, numbers of business and private GSM phones remain unknown.

Configuration of this function requires entering **MblPh** field according previous chapter and setting static path routing type as CallBack, it is described in chapter 4.5.1. PathFinder –Call routing

4.3.1.7 Call Waiting

Starting from 6.22 firmware version, new service of waiting call for individual number was introduced. This service operates as follows: incoming calls are entered into queue if dialed extension number is busy with another call. New call is announced by audio signal in handset and additionally on display of system phone.
Configuration of this service is performed in sheet **Subscribers/Incoming calls**

- ➢ **CWt** – defines **maximal number of incoming calls** waiting in queue.
- ➢ **WtMel –** enables to change melody on hold or defining announcement with private information
- ➢ **AnnBeg** – allows for Call Waiting announcement to be played from the beginning for every caller that is placed in queue
- ➢ **QAnn** – switches on the functionality which enables to play voice announcements with message about caller position in queue. For proper operation of this service it is necessarily to record additional announcements in sheet **Incoming calls/Announcements**. Checking this field generates information about possibility of automatic adding these annoumcements to PBX configuration.

4.3.1.8 Ringing interception group (PickUp group)

PickUp group it is set of extension numbers. In PickUp group it is possible to intercept ringing from other group members. Subscribers from different nets can be assigned to on PickUp group. Up to 99 PickUp groups can be defined in PBX (it can be done in sheet *Descriptions / PickUp groups*).

ATTENTION! Possibility of ringing interception in PickUp group is described in chapter 4.3.1.2.

4.3.1.9 Protection against services

In sheet *Subscribers / Protection against services* protection against ringing interception in pickup group - *PtcPck* - can be set up. It means, that extension numbers included in the same

PickUp group as specified extension number could not intercept ringing from this phone with service *37 (Intercept ringing in PickUp group).

In this sheet we can also set up protection against *override* by checking field *PtcOvr.* If one of call part has checked this field, it is enough to lock *override*.

4.3.1.10 Special services

Every extension number can has limitations set on the various PBX functions. It can be done in *Subscribers/Special services* sheet*.* Table in this sheet allows, for example, you to set access privileges for every extension number for using CTI applications or lock the external calls.

There are following parameters available in sheet *Subscribers / Special settings* :

– *Cmp* – enables to assign every extension number to one of fifteen companies, but word *company* is defined here slightly different (ex. when PBX is shared by few organizations it is in fact company but it may be also company's department). Assigning to one of *Cmp* is important while accesing to Public Phone Book and during charging using BillingMAN application.

– *PubPB* – Defines access to Public Phone Book. Following options are available for this field: *No access*, *Read only*, *Readout without privileges check* – subscriber can dial numbers included in Public Phone Book even if his/her privileges defined in fields **DayPrg** or **NgtPrg** in sheet **Subscribers/Outgoing calls** doesn't allow to do it - and *Read and write*. Last option is available only for CTS system phones. If subscriber is granted with privileges to using Phone Book he/she can dial numbers dialing ****ID** is a position in Phone Book. This dialing method is available for every subscribers.

– *CLIRi* – after checking this option extension number will not be presented, also comment assigned to this extension number will not be presented. Above mentioned setting concerns only internal calls.

– *RegRng* – defines ringing mode and PBX dial tone – if this field is checked subscriber hears always single ringing and continuous PBX dial tone, irrespective if any services (for instance DND) is set on this phone or not, otherwise if this field is unchecked ringing mode is determined by **ErnRng**, **ItlRng**, **DphRng and VIPRng** fields settings in sheet *Global settings / Incoming calls* and PBX dial tone depends on service settings on specified phone*.* If any service is activated on specified phone, dial tone is interrupted once, immediately after lifting handset or switching on speakerphone and then is continuous, otherwise it is continuous from the beginning. This option concerns only analog ports.

ATTENTION! For devices different as phones, ex. faxes or fax-modem connected to analog ports, it is recommended to check field *RegRng*.

– *DT* – this option should be checked if modem or fax - instead analog phone - is connected to the port. This settings is necessarily to establish connection of type *audio signals 3,1kHz* in outgoing calls routing through ISDN trunk.

– *KnLck* – enables to switch off knocking signal. *Knocking* signal can be hear on analog and system phones. This signal informs subscriber about waiting call by means of short beep (on system phone display also caption "Waiting..." is presented). Knocking signal can be hear for group incoming calls and for incoming call forwarded without announcing

(specified call "knocks" if earlier was included in queue). Knocking signal period can be defined in sheet *Incoming calls*, field **KnPer**.

– **ErnLck** – this setting defines limitation in external calls for specified extension number. Following options are available:

Do not lock – from this extension number all external calls can be established according its privileges (ex. using prefixes, access classes etc.).

Protection against services – after checking this option, external calls will not be transferred to this extension number (FLASH, call forwarding), also from this extension number there is not possible to access three party conference if at least one part of this conference is external. When external lock is realized by this option, it has no influence on direct external call routing via incoming and outgoing calls tables.

Full lock – there is no possibility to establish any call with this extension number if one part of such connection is external. This option is superior over all other settings concerning with outgoing calls.

– **CTI** – every extension number can access to: *TelephoneCTI* application – basic option or *TelephoneCTI.plus* with possibility of recording calls, sending SMSes and using recordings created by EbdRec system. How many extension number have access to CTI applications depends on license acquired. Appropriate license code should be entered in sheet **Global settings / Licences.** Additionally option *CTIP on RS-232 port* is available. This option enables to send CTIP protocol on RS-232 port, without log in (without using CTI application, it also doesn't require licensing). CTI application requires password to log in Subscriber. **PIN** defined in sheet **Subscribers / Services** is used as password.

– **PayMAN** – is not supported from version 6.20.

ATTENTION! TelephoneCTI and PayMAN applications are subjects of license. Detailed information on it could be find in *chapter 17*.

– **TAPI** – checking this field authorize subscriber to using TAPI interface which is needed for all *Microsoft* software, because *Microsoft* uses this protocol. Most popular application is using contact base from "Microsoft Outlook". To use TAPI protocol special plug-in for Microsoft Outlook was developed. After installing Microsoft Outlook plug-in, numbers from contact base can be dialed, presentation according contact base entry and editing contact is also supported. This option is not available for VoIP and ISDN subscribers.

ATTENTION! Using TAPI interface is subject of licensing.

4.3.1.11 Subscriber services – alarm call, locking phone, recalling fixed account, changing PIN code

Services available by phone: alarm call *28, locking phone *21, recalling fixed account *53, changing PIN *55 can be set also in ConfigMAN application. These settings can be set in sheet **Subscribers / Services**.

AlrC – this field defines if alarm call service is active for this extension number.

AlrTi – this field defines (with accuracy of 1 minute) time of performing alarm call for specified extension number. This field is active for extension number with active alarm call service (checked field *AlrC*).

Acc – recalling fixed account. After downloading configuration this field shows if on specified extension number account is recalled. Additionally this field enable to recall account from inside ConfigMAN application.

HAcc – defines deactivation hour of recalled fixed account. If *53 service is activated on phone with specified extension number, value in this field is inserted automatically according *AccRel* field settings in sheet *Global settings / Remain.* This value can be modified separately for each subscriber from inside ConfigMAN application. If You recall account from inside ConfigMAN application (field *Acc*), deactivation hour should be set manually.

PIN – this field enables to define four digits PIN code. This access code enables to perform services require PIN (ex. phone lock or recalling account). PIN is also used as a password for *TelephoneCTI* application and *TAPI* interface.

Lck – this field shows if phone with specified extension number is locked or not. It also enables to manually lock/unlock phone from inside ConfigMAN application. It can be done using phone by selecting service *21. Phone lock is active until it deactivation. From locked phone fixed and temporarily account can be recalled. Unlocking phone deactivate fixed account recalling.

4.3.1.12 Preferred hot line

Depending on settings in field **PrfHL** in sheet *Subscribers / Outgoing calls* it is possible to assign to every subscriber number number of another subscriber, group number, speed access code, which should be dialed after handset lifting or ID of external line which should be used as default to establish an outgoing call.

Fig. 4.9.: Sheet Subscribers / Outgoing calls

None – subscriber doesn't use preferred hot line.

Subscriber – if subscriber after lifting handset doesn't dial any number, after time defined in field **HLOfs** in sheet *Outgoing calls* subscriber number assigned in this field will be dialed.

Doorphone – if subscriber after lifting handset doesn't dial any number, after time defined in field **HLOfs** in sheet *Outgoing calls* subscriber number assigned in this field will be dialed.

Group - if subscriber after lifting handset doesn't dial any number, after time defined in field **HLOfs** in sheet *Outgoing calls* group number assigned in this field will be dialed.

Trunk – defines trunk which will be sized as default while establishing outgoing call. Outgoing call should be routed via trunk group or trunk list (including on first place trunk group) which include trunk defined as preferred hot line. If trunk defined as preferred hot line will be busy:

- if outgoing call is routed via trunk group, record of incoming call table will be omitted;
- if outgoing call is routed via trunk list, next list element will be selected.

ATTENTION!: Preferred trunk is take into account only if it (or trunk group including this trunk) is first list element.

Speed access code – if subscriber after lifting handset doesn't dial any number, after time defined in field **HLOfs** in sheet *Outgoing calls* speed access code will be dialed automatically, it means routing to specified entry in Outgoing Calls Table (sheet *Outgoing calls / Short numbers*). Next PBX behavior depends on settings in this entry. This option can be used to immediate or with specified delay (dependently on value set in field **HLOfs** – from the range 0÷9 seconds) establishing call with specified number.

4.3.2 Presentation for internal calls

In sheet *Subscribers/Outgoing calls* you can define what number should be presented for internal calls. Except subscriber's own number 2 options are available:
Another subscriber number – subscriber is presented with number of another subscriber.
Group number – subscriber is presented with group extension number.

After checking field **CLIRi** number for internal calls will not be presented.

4.3.3 Groups – creating, setting, adding subscribers

Subscribers can be assigned to specified group and reroute internal or incoming calls to this group. Maximal group number is equal to:
- 20 – in *CXS-0424*,
- 25 – in *CCT-1668*,
- 80 – in *MAC-6400, IPS-08, IPU-14, IPM-032, IPL-256* .

Additionally up to 8 ACD groups can be defined in any of mentioned above PBX.
Number of groups which can be created (depending on PBX firmware) can be examined in *Groups* sheet.

4.3.3.1 Upringing, cyclic and linear group

To create group appropriate sheet should be open:
- *upringing →* *Groups / Upringing / Settings*,
- *cyclic →* *Groups / Cyclic / Settings*,
- *linear →* *Groups / Linear / Settings*.

and add new group, by inserting new record. While creating group fields **ExtNo.** - extension number[2] and **Cmt –** group name should be defined.

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[2] Group without extension number can be used <u>only</u> for incoming calls.

For up-ringing, cyclic and linear group following settings are available:

– **Res** – extension number reservation time after group incoming call. It defines time of extension number unavailability for group calls after hang up the handset,

– **Net** – this settings defines specified group membership in one of existing in system nets - settings defines group privileges while internal calls,

– **Pck** – this settings enables to assign extension numbers to PickUp groups - ringing interception groups. Membership in such groups enables to using services *35 – urgent ringing in PickUp group, or *37 – intercept ringing in PickUp group,

– **QCap** – defines queue capacity - number of callers waiting for establishing connection with group agent. If queue will fill up (reaches number of waiting callers defined in field *QCap*), next caller will be hear busy signal. If caller number is granted with **VIP** mark (both, for internal callers – sheet **Subscribers**/**Outgoing calls** and external caller, with number defined in Public Phone Book – sheet **Phone Books/Public**), then while group number is busy, caller will be set as first in waiting call queue, even if queue is full.

– **MaxC** – content of this field defines number of incoming calls (how many incoming calls can established simultaneously) for specified group. When number of calling group agents is equal to content of field *MaxC*, incoming calls are set in queue of waiting calls.

– **MaxWt** – this parameter can be determined in the range of 1 to 30 minutes, it defines maximal waiting time for group calls. This time is valid both, while waiting for receiving call when the group is available as well as while caller is set in queue. This time is valid for calls from phones and from doorphones. Only one exception is situation when incoming call from outside is routed directly to group. In such situation, independently from group status (available or busy), upper exchange disconnects call after 3 minutes. For this reason, to extend waiting time it is recommended to route call using voice announcement.

– **KnLck** – enables to switch on the knocking signal, its inform group agents about waiting call. Knocking signal period is set in sheet **Global settings / Incoming calls** in field **KnPer**. By default this signal is switched on.

– **MaxFwd** – this field enables to set maximal number of simultaneous call forwarding from group (this parameter is available only for up-ringing group).

– **HidExt** – hide group extension number in phone book of system phones.

– **AnnQ** – announcements informing about position in queue – functionality which enables playing voice announcement informing about predicted waiting time and/or position in queue. For proper service operation it is necessarily to record additional voice announcements, it can be done in sheet **Incoming calls/Announcements**. In ConfigMAN application there are 39 predefined system announcements, which should be uploaded to the PBX. It is possible to change them by uploading .wav file.

Depending on selected option, following settings are possible:

- *waiting time + position in queue* - *waiting time* message is play once after 30 seconds of inserting into queue, next every 30 seconds announcement about *position in queue* is played. For this option it is possible to change two parameters **MaxT** – maximal number of time announcements and **MaxQ** – maximal number of announcements for persons waiting in queue (options described bellow)

- *waiting time* - *waiting time* message is play once after 30 seconds of inserting into queue. For this option it is possible to change parameter **MaxT** – maximal number of time announcements

- *position in queue – position in queue* message is played after every 30 seconds. For this option it is possible to change parameter MaxQ – maximal number of announcements for persons waiting in queue.

ATTENTION!
Queue waiting time is notified only if it is longer than 1 minute. Announcement about place in queue is never increased (ie. when VIP subscriber entered into queue).

- **MaxT** – enables to define maximal time when waiting time message is played. Following option are possible: 1, 2, 3, 5, 10, 15, 20 minutes. If waiting time exceeds 20 minutes an announcement "Expected waiting time exceeds 20 minutes. Please wait for connection"

  MaxQ – defines position in queue, to which announcements will be played. Whole range is 1 to 30 waiting callers. If number of waiting callers exceeds this number, a standard announcement will be played "Your call is waiting for connection. Please wait."

- **MaxFwd** – enables to set maximal number of simultaneous call forwarding from group (this parameter is available <u>only for upringing group</u>).

ATTENTION!
It is not recommended to use ISDN phones as group agent. Only one MSN for specified ISDN port can be group agent. If any phone connected to ISDN terminal is busy, whole terminal is maintained as busy for group calling.

Next for group defined as described above we can add group agents (subscribers) in sheets *Groups/Upringing/Agents, Groups/Cyclic/Agents, Groups/Linear/Agents* depending on type selected. Every group upringing, cyclic or linear can consist of up to 12 agents (subscribers). For specified subscriber upringing method can be defined.

Upringing Groups
In up-ringing groups, agents start ringing after time predefined in field **Ti.** This time is computed from the beginning of group calling. As first starts ringing agent with the shortest **Ti**. Group up-ringing continues until all agents ring or any agent pick the phone. If parameter **Ti** for all group agents is equal to 0 sec, all group agent start ringing immediately. Additionally checking field **Nbsy** is also possible. - group agent with checked this field is omitted when busy.

Cyclic groups – groups with regular call distribution
In cyclic group agents' phones rings after each other, so always rings phone of only one agent. As first rings phone of agent, who is next after agent lastly called as first. Using following figure, if lastly agent with extension number 104 was called first, during next call agent with extension number 107 will be call as first. Parameter *Ti* enables to define how long specified agent will be call before next one.

ATTENTION! In cyclic group call forwarding from group cannot be realized.

Linear groups
Linear group is very similar to cyclic group. Only one agent rings in linear group. As first rings agent defined as first in sheet *Groups/Linear/Agents.* According following figure, agent with extension number 104 will be ring as first. Then agents' phones ring respectively, according group agents list. Using **Ti** parameter ringing time of specified agent can be set up.

ATTENTION! In linear group call forwarding from group cannot be realized.

4.3.3.2 ACD group

ACD group is special group with automatic call distribution. In ACD group, one incoming call is routed to one and only one phone. If dialed subscriber doesn't pick-up the phone in specified time, ringing is automatically switched to another group agent, dependently on group type (*cyclic* or *rest time*). Maximal ACD group capacity (number of agents) is equal to 100.

Settings in ACD group

Up to 8 ACD groups can be created, up to 100 agents can be added to every ACD group. ACD group can be created by inserting new record in sheet *Groups/ACD/Settings.* ID number for each ACD group is assigned automatically. Above you can see figure with ACD group settings.

Field with group extension number **ExtNo**, identifies ACD group in PBX. Assigning this number is obvious if we want to dial this group also in internal calls.

Comment field **Cmt** enables to assign group name. If extension number is not assigned to ACD group, this field enables to identify specified group. Due to this reason ACD group names should be different.

In field **GrTyp** – group type, type of group should be defined. Following settings are available: *Switched OFF*, *Cyclic*, *Rest time*.

- *Switched OFF* – after setting this option ACD group is switched off.

- *Cyclic* – up-ringing schedule is the same as described in previous chapter, agents ringing - next after previous agent, only one phone ringing in the same time. As a first ringing next after previously ringing agent.

- *Rest time* – by this settings up-ringing order is defined by rest time of group agents since last established call.

In field **GrMng** you can define extension number of subscriber who manage ACD group. To ACD group manager routed are all information concerning improper acting of ACD group, for example when all agents are logged out. ACD group manager, in the same time, ACD group agent, in this situation **AutBRB** status is not valid for this subscriber. ACD group manager is authorized to log ion and log out subscribers to this group. It is possible with service *59 (more about services in Slican PBX CCT-1668 in instruction manual). ACD group manager can't be an account.

PIN field enables to enter four digits of access code. This code enables to log in subscriber to ACD group. Subscriber can log in by oneself to the ACD group or log out using service *58. If we do not enter PIN code, logging is possible only from inside ConfigMAN application or by ACD group manager.

After checking field **AutBRB** group agent is automatically logged out from ACD group if subscriber do not answer group call. After next lifting handset by this subscriber, he will be automatically logged in to ACD group.

In field **Net** we can define net where ACD group belongs. If extension number is assigned to ACD group, only subscribers from the same net can be agents of this group. This field is used to defining privileges for internal calls.

In field **MaxWt** you can define maximal expected waiting time in queue. This parameter determines ACD group busyness depending on expected waiting time. If expected waiting time is greater than defined in this field parameter value, ACD group is busy. It is not valid for forwarded calls and VIP calls – such calls are always inserted in queue. Expected waiting time is calculated according following formula – average time of 10 last calls * number of waiting calls/number of active (logged in and not **BRB** status) group agents. If all group agents are in **BRB** status this limitation doesn't work – ACD group is not busy and all calls are inserted in queue. Additionally information about it is send to ACD group manager.

Content of field **Rng** defines ringing time for every group agent. After this time next agent's phone is ringing. Defined here ringing time is valid independently on ACD group type.

Content of **Res** field defines group agent's reservation time after group incoming call. It defines time of group agent unavailability for next ACD group call after hang up the handset. During this time status of group agent is busy, but call routed directly to this extension number can be answered. Group agent can reduce reservation time by lifting and hanging up the handset. Any dialing or answering direct call doesn't reduce this time.

ACD group agents

Fields **ID**, **ExtNo.**, **Cmt** include values from sheet ***Groups / ACD / Settings***, entered while creating ACD group and here there is no possibility to edit them. Yo can only, using buttons **Add** and **Remove**, add or remove agents in specified ACD group. After pressing **Add** key following window can be visible:
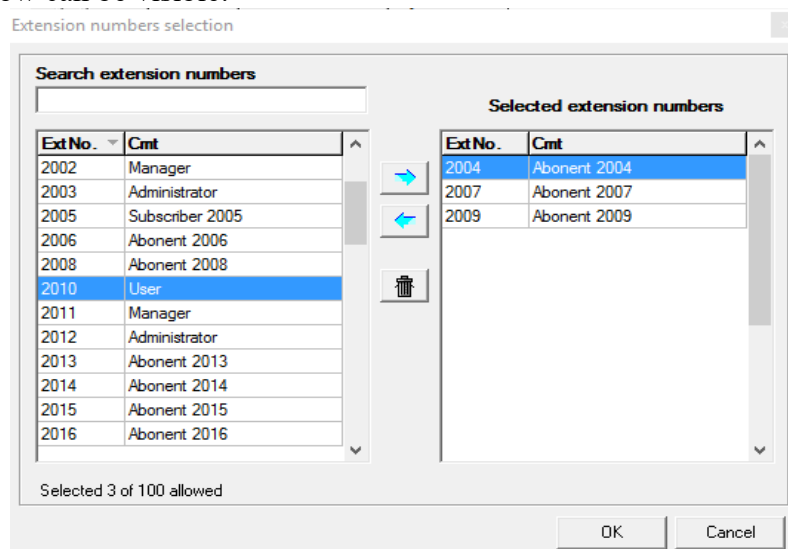
Figure 4.10: Window for selecting ACD group agents

You can add subscribers by selecting them in left part of window and clicking blue arrow directed to the right, confirming the selection. If we want to select few subscribers, we can use SHIFT key (selecting continuous range of records) or CTRL (selecting separate records). Removing records can be performed by selecting them in right part of widow and clicking arrow directed to the left.

4.3.3.3 Paging group

Paging group allows to simultaneos notificate form one, up to tens of PBX users. In case of upringing the group, on all target devices, the acoustic will turn on (speaker on system

phones), warning signal is played, then the iniciator has the possibility to announce required informations.

This functionality can be used in warning and alarm systems.

Configuration:

In ***Groups/Paging/Agents*** there are 8 groups available by default (quantitiy not changeable) to which user can add subscribers. Only system phones and analogue Audio devices, configured as Paging (more about configuring analogue port in chapter 8.5 Audio devices configuration). Agents assigned to ***Paging groups*** can also be a part of ohter groups. ***Paging groups*** is identified by ID tag, it does not have own ***Extension number***. So, to add it as a callable element, it has to be a part of a Function number which has its own number.

To call a Paiging group, Function number has to be dialed. This group is available both in internal and incoming calls – by proper record in *Incoming calls/Settings/Main – Prefix, Static path routing, Infolines and API Infolines.* Only subscribers with proper privileges can dial Paging groups – *Subscribers/Internal calls* field *Net.*
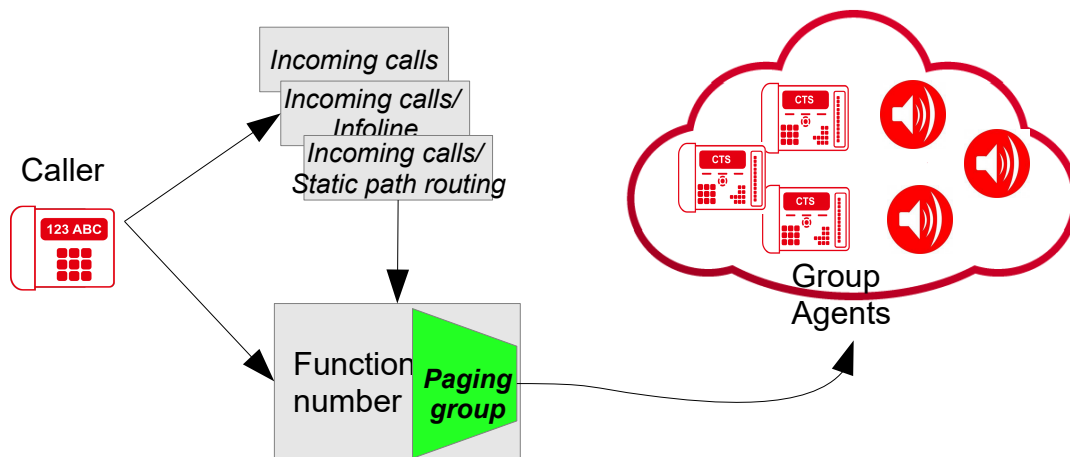
Figure 4.11: Paging group - operating principle

Operating principle:

Dialing a Function Number with assigned Paging Goup by caller, will upring the group. In this time, the caller hears a melody on hold, and on upringed devices two tone gong is played. After it ends, a beep sugnalize the caller, that his announcement is now heard by all devices.

Paging group has higher priority than DND, FWD, MobilePhone and reservation services. On upringed phone, Function Number Comment is displayed.

4.3.3.4 Conference group, conference room

Conference calls can be realized using three different methods:
- **Conference group** – up-ringing predefined participants,

- **Conference room** – conference between subscribers who dial specified number,
- **Private conference** – conference realized from any CTS system phone (excluding phone *CTS-102.HT*) using „Conference" service.

Conferences are subjects of following limitations:

PBX type	Maximal number of participants	Maximal number of simultaneous conferences
CXS-0424	30	10
CCT-1668	8	4
CCT-1668 (FW>=5.11)	12	10
MAC-6400	30	10
IPS-08	30	10
IPU-14	30	10
IPM-032	30	10
IPL-256	30	10

Table 4.1.: Limitations concerning conferences

Real number of conferences possible to establish depends on acquired license, although Slican guarantee, that always it is possible to establish 3 conferences in *MAC-6400* and *CXS-0424* as well as 2 conferences in *CCT-1668*.

Due to above limitations in ConfigMAN it is possible to define as follows
Conference group

Conference group enables to establish conference calls. Last participant in conference (8 in *CCT-1668* and 30 in *MAC-6400*) is conference originator (subscriber who initiate conference, any internal or external subscriber can be the conference originator). While defining conference group in sheet **Conferencing / Groups / Settings**, its extension number can be defined (in field **ExtNo.**). To establish conference this extension number should be dialed from any phone (if conference group has no extension number, it is possible only to reroute incoming call to such group – field **ExtNo**. in sheet **Tabeles / Incoming calls / Settings**). If subscriber equipped in any system phone *CTS-202* or *CTS-102*, is conference originator, it is possible to manage this conference from this phone:

before up-ringing group (dialing its extension number): if group is assigned to speed access key (see 8.4. Programmable buttons configuration in CTS system phones), then after pressing this key list of conference participants is downloaded and can be modified (add or remove participants),
after dialing group (call established): it is possible to add and remove participants, mute participants microphone, transferring custodian function, etc.

ATTENTION! *More information concerning conference call managing can be found in chapter „Conference" in system phone manual.*
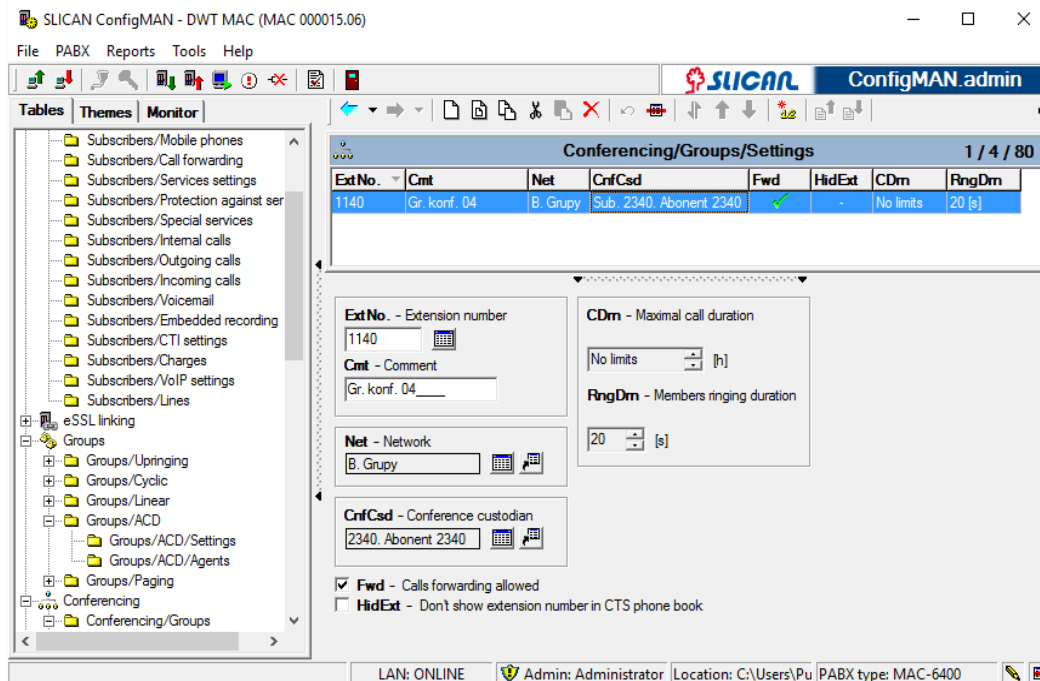
Figure 4.11: Sheet Conferencing / Groups / Settings

For conference group also following settings are available:

– **Cmt** – enables to enter group name. If group has no extension number, content of this field is used to conference group identification.

– **Net** – defines one of eight subscriber nets. It is used to define subscriber privileges gor incoming and outgoing calls.

– **CnfCsd** – in this field you should define conference custodian, only subscriber equipped in *any CTS-202 phone* can be conference custodian. This subscriber is not obliged to participate in conference, but is authorized to add/remove participants, monitor their status, switching on/off microphone to specified participant etc.

– **Fwd** – after checking this field, if "unconditional" (*74) or "when no answer" (*76) call forwarding for all or only internal calls is set for conference participant, number defined in call forwarding service is up-ringed (if it is no voice mail box DISA or infoline number). If this field is checked and one of mentioned above call forwarding is active for specified extension number, ex. to voice mail:

  – unconditional (*74) – extension number is not up-ringed and call forwarding to voice mail is not realized,

  – when no answer (*76) – extension number is up-ringed without call forwarding to voice mail.

If this field is not checked, call forwarding is not realized, despite of it is active (conference participant is ringing by the time defined in field **RngDrn**).

– **CDrn** – enables to limit conference call duration time.

– **RngDrn** – time, while every group participant is up-ringing. This settings is common for all participants.

After creating and configuring conference group, extension numbers which have to belong to this group should be pointed. It can be done in sheet ***Conferencing / Groups / Participants***, using appropriate buttons.

ConfigMAN
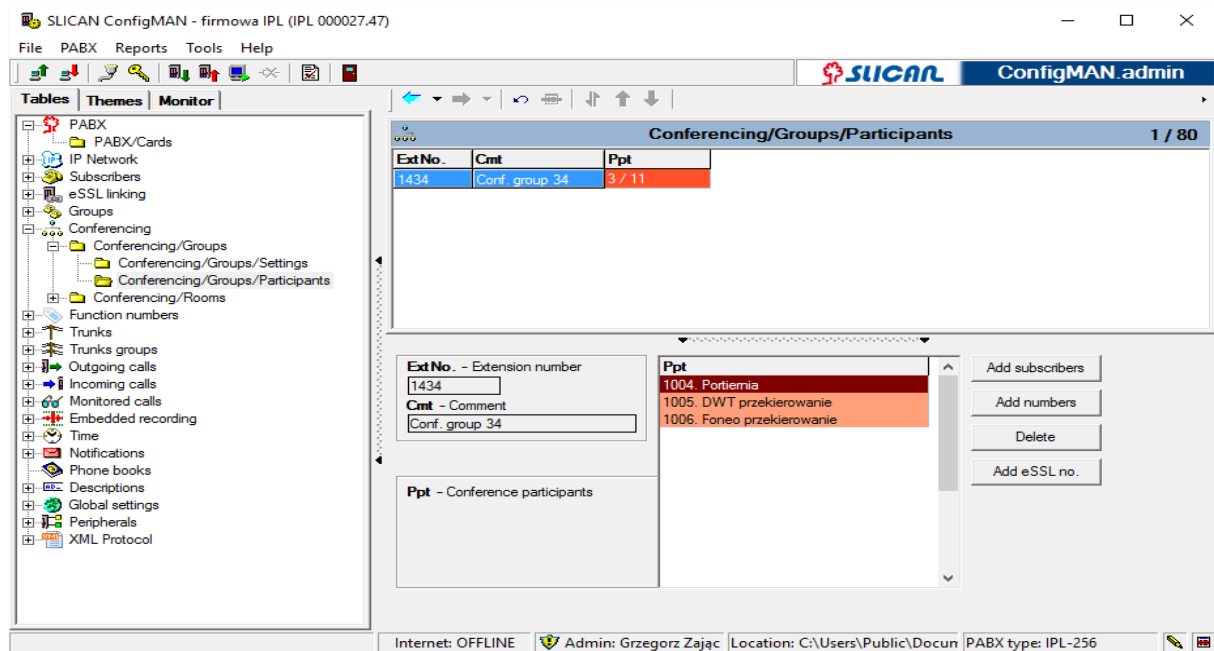
Fig. 4.12.: Sheet Conferencing / Groups / Participants

Conferencing group can be used as a paging group. If extension numbers equipped with system phones are participants of such group, dialing this group followed with service Urgent ringing (*35) causes switching on acoustic (speakerphone) on system phones.

Conference room

Such kind of conference is realized by dialing conference room extension number (entering to the conference room) by conference participants. Participants of such conference can be any subscriber (internal or external).

Example: There is Service Department in the company. Service manager has to hold meeting with servicemen, so inform all persons 'Today at 11:00 will take place a teleconference". Persons who are in company dial extension number and servicemen outside the company dial appropriate number to connect with conference room.

Fig 4.13.: Functionality of conference room

During creating conference room in sheet ***Conferencing / Rooms / Settings***, its extension number can be defined (in field **ExtNo.**). Dialing this number subscriber enter to this room and will be wait to next participants (if extension number is not defined, it is possible to route only incoming calls to this room – field **ExtNo.** in field ***Incoming calls / Settings / Main***).
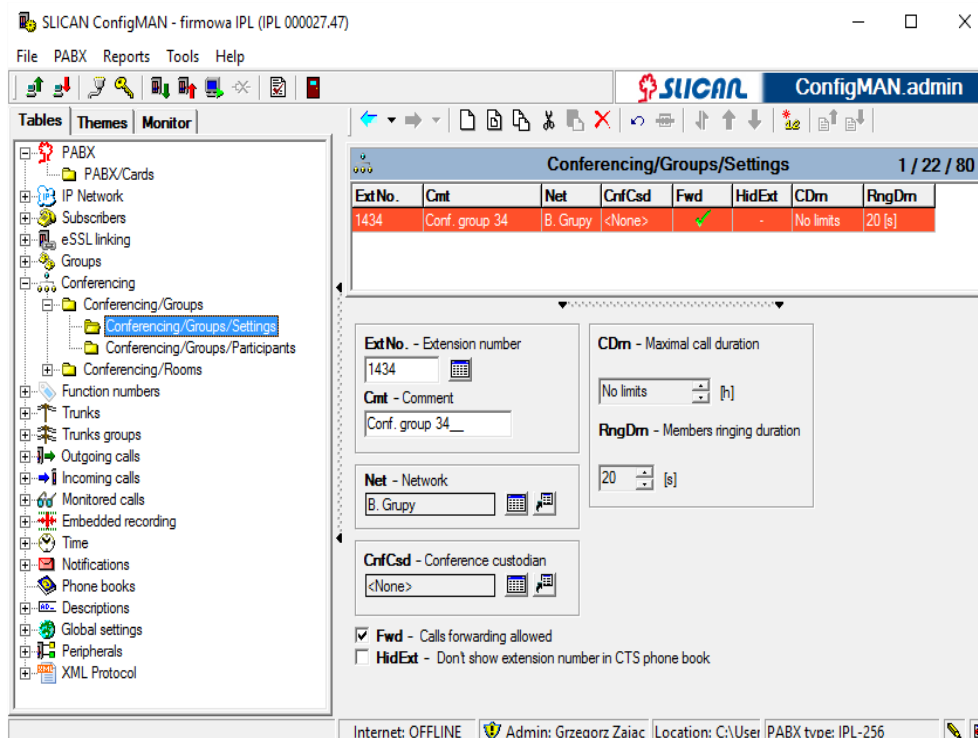
Fig. 4.14.: Sheet Conferencing / Rooms / Settings

Remain settings are:

- **Cmt** – enables to define conference room name. If extension number is not defined, this field enables its identification.

- **Net** – defines one of eight PBX nets assign to conference room. Content of this field enables to limit access for to conference room for subscribers in internal and incoming calls.

- **CnfCsd** – in this field conference custodian can be defined, conference custodian can be only subscriber equipped in any phone from family *CTS-202 or CTS-102.CL*. Conference custodian is not obliged to participate in conference, however conference custodian is authorized to add/remove conference participants, control their statuses, switching on/off microphone for selected person etc.

- **WtMel** – after checking this field subscriber who "enter" conference room, hears melody during waiting for custodian (lifting handset by custodian). This option is available if custodian is defined in field **CnfCsd**.

- **Fwd** – conference custodian defined in field **CnfCsd** is authorized to add to existing conference (consisting of subscribers who "enter" conference room) next participants. New participants can be defined in sheet ***Conferencing / Rooms / Participants*** or add using appropriate option accessible from his/her phone. After checking this field, if unconditional (*74) or "when no answer" (*76) call forwarding for all or only internal calls is active for new conference participant, call from conference custodian is routed to number defined in this service (if it is not number of voice mail, DISA or infoline).

    If thi field is not checked call forwarding is not realized, although it is active and participant's phone is up-ringing.

- **CDRn** – enables to limit conference call duration time.
- **PIN** – after defining 4-digits PIN code access to conference room is protected, after dialing conference room extension number, subscriber is prompted to enter this PIN code to enter the conference room. If conference room extension number is dialed by conference custodian or subscriber with extension number predefined in sheet *Conferencing / Rooms / Participants*, caller is not prompted to enter PIN code.

It is possible to add extension number which should be conference participants after creating and configuring conference room. It can be done in sheet *Conferencing / Rooms / Participants*, by using appropriate buttons. Beside, extension number equipped with any system phone series *CTS-202 or CTS-102.CL* is able to manage such conference room, as well as conference group:

- *before conference room up-ringing*: if conference room is assigned to speed access key (see 8.4 Programmable buttons configuration in CTS system phones), after pressing this key participant list is downloaded to phone and can be modified (add or remove conference participants),
- *after conference room up-ringing*: it is possible to add, remove participants, switching off/on microphones, transferring custodian function etc.

ATTENTION! More information concerning conference call management options can be found in chapter „Conference" of system phone instruction manual.
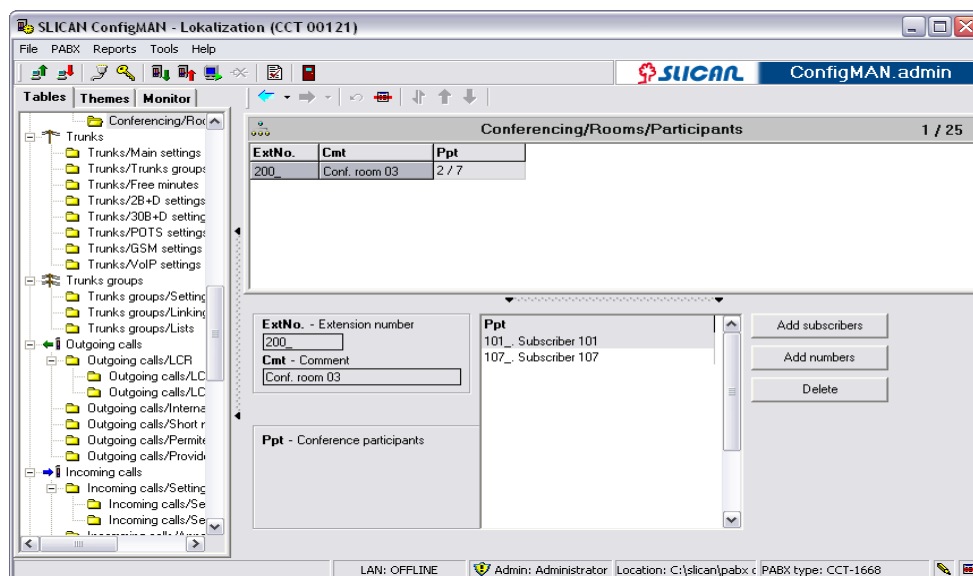
Fig. 4.15.: Sheet Conferencing / Rooms / Participants

4.4 Outgoing calls

Outgoing calls are all calls required establishing connections between Slican PBX and upper exchange.

To precise defining external calls routing, firstly we explain basics used to PBX programing:

Trunk – it is physical (or logical for VoIP calls) port in PBX which is used to connect PBX with telecommunication services provider.

Trunk group – it is defined trunks set of the same type and the same configuration (ex. ASS, ISDN in PP or PMP configuration), which for outgoing calls are occupied dependently on time of established calls (occupied is trunk for which in field **OutCnt** of sheet *Trunks / Free minutes* is the smallest value if free minutes are not used for this trunk group). It is possible to include in one trunk group, trunks of different types belonging to different providers, only one condition enabling to do this is following – dialing mode for these trunks must be identical.

Trunk list – it is a set of trunks and trunk groups which are occupied respectively (always starting from list beginning). **Trunks list** may consist of different types trunks groups and single trunks. Trunks list is used only for outgoing calls only. The same trunk or trunks group may be a member of many trunk lists.

Outgoing calls may be routed directly to trunk, trunks group or trunks list according PBX user requirements.

> **IMPORTANT:** *Any trunk may be a member of only one trunk group in the same time.*

4.4.1 Defining trunk group parameters

To create trunks group in PBX database we should add new trunks group in sheet *Trunks groups/Settings* by insert new record. Next we define limits concerning maximal number of incoming and outgoing calls for specified trunks group – *MaxInc* and *MaxOut*. These fields define maximal numbers of incoming and outgoing calls possible to establish via specified trunk group. It can be used for regular using trunk groups for incoming and outgoing calls or for assuring appropriate number of voice channels for outgoing calls. If trunk group is occupied for incoming calls, next caller incoming via trunk belonging to this trunk group hears busy signal. Only one exception are numbers defined as VIP in sheet *Public phone book/Public* and defined in sheet *Incoming calls/Settings/Additional*, for these numbers physical capacity of trunk group is only one limitation.

Attention: Incoming calls limitation is not valid for trunk groups including analogue trunks.

Fig. 4.16.: Sheet – Trunks groups/Settings

It is also possible to define pulse price - *Prc* - depending on provider.

After checking field *PthRt* paths of all *outgoing calls*, established via specified trunk group will be stored in PBX memory. After storing path in PBX memory all incoming calls routed via the same trunk group will be routed <u>directly</u> to the extension number which was the caller of above mentioned outgoing call. This procedure is described in details in chapter „Dynamic path routing".

Additionally each trunks group may be assigned to one of two *SPP* groups*.* This settings is used to Slican PBX cooperation with hotel application.
Assigning trunks to trunks group can be done in sheet *Trunks/Trunks groups*. For specified trunks, trunks group can be indicated.
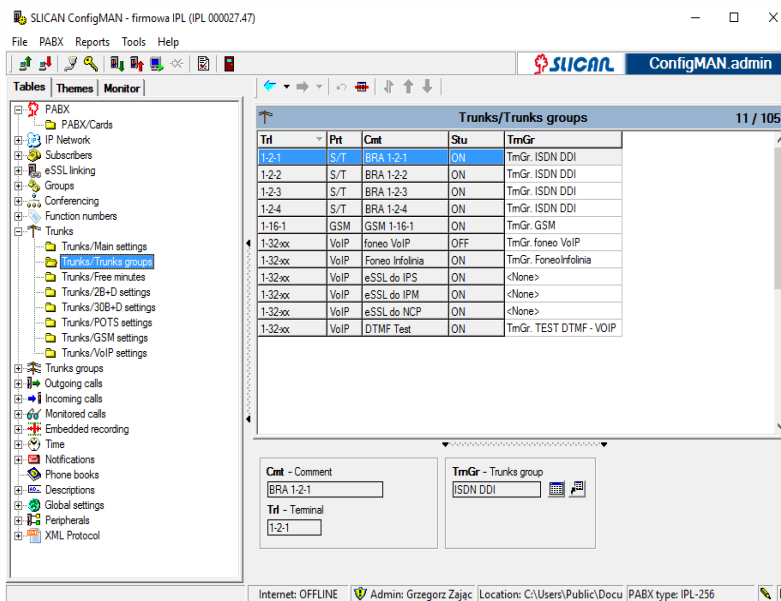
Fig. 4.17.: Sheet – Trunks / Trunks groups

4.4.1.1 Defining and setting parameters of trunk list
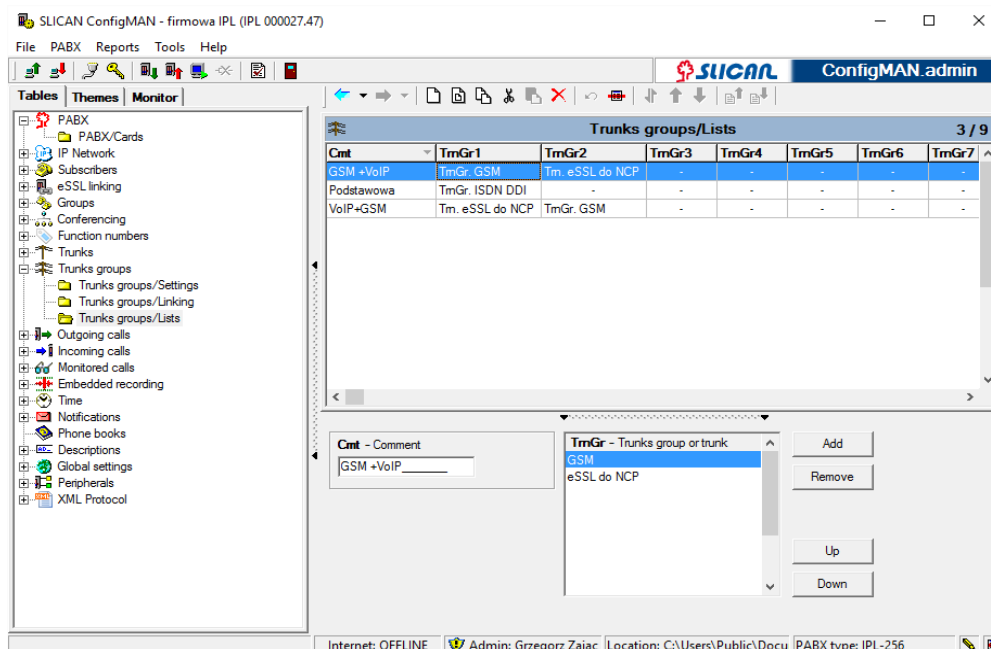
Fig. 4.18: Sheet – Trunk groups/Trunks lists

Trunks group and/or trunks can be members of trunks list. To create trunks list in PBX database we should add new trunks list in sheet **Trunks groups/Lists** by insert new record. After creating new trunks list, we should assign its name and add list's elements (eight maximal), using *Add* button. It is always possible to remove element from list by selecting it and pressing *Remove* button. To route outgoing calls according expectations, order of list elements should be observed. To define this order, buttons *Up* and *Down* can be used.

4.4.2 Privileges for outgoing calls

4.4.2.1 Privileges and privilege categories

During outgoing calls configuration extension numbers are granted privileges to dial specified external numbers. Extension numbers privileges are checked by PBX during establishing connection. If subscriber from specified extension number tries to dial number without appropriate privileges appears special message in form of voice announcement.
Up to eight privileges assigned as A to H can be used (**Descriptions/Subscribers privileges**).
In sheet **Outgoing calls/Permitted prefixes** defines privileges to using prefixes.
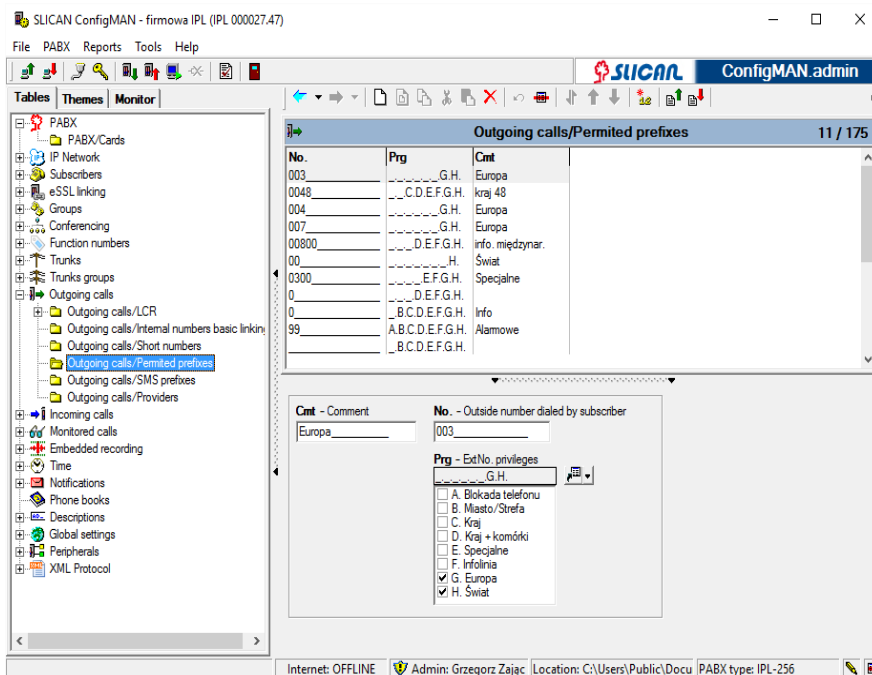
Fig. 4.19: Sheet – Outgoing calls / Permitted prefixes

ATTENTION!
A class privileges are always automatically assigned to subscriber with locked phone or subscriber with passed limit for calls.

ATTENTION!
After defining A class for specified prefix, established calls with this prefix are not charged. Locked subscribers or/and subscribers with A access class can dial only numbers with prefix assigned to A class.

4.4.2.2 Assigning access classes to subscribers and accounts for outgoing calls

Access class enables (or not) to use entries in tables after dialing specified prefix.

To realize outgoing calls as defined in sheet **Subscribers/Outgoing calls** to each extension number appropriate access class should be assigned.

Access class defines which trunk, trunk group or trunk list will be used to establish call with dialed number, it is done in sheet **Outgoing calls/LCR/Settings** or **Outgoing calls/Short numbers**.

Selecting some services before dialing number can change subscriber access class:
Private call (*27)- to subscriber extension number access class defined in field **PrvCls** is assigned,
Urgent call (*25) – to subscriber extension number access class **H** is assigned.
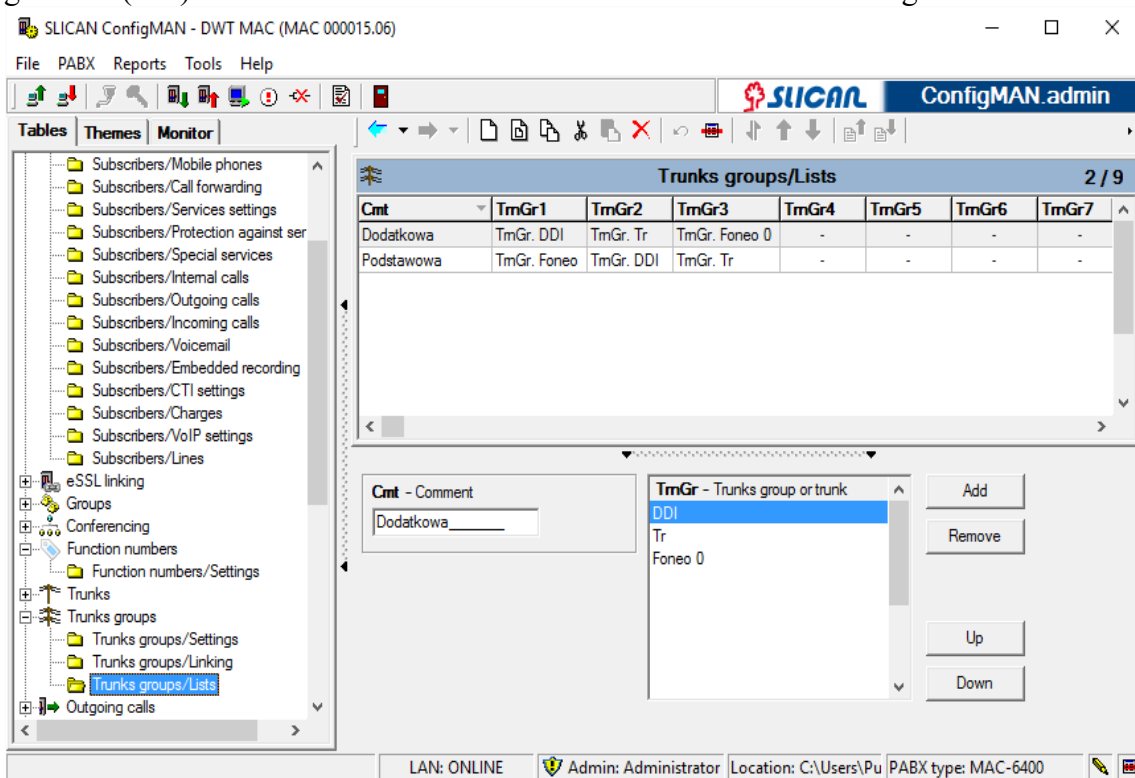
Fig. 4.20.: Sheet – Subscribers / Outgoing calls

4.4.2.3 Privileges to establishing urgent calls and transit

Access class H is treated as urgent calls class. It may be granted to extension number temporarily for one call if for this extension number field *UrgEn* in sheet *Subscribers/Service settings* was checked, as shown on figure below. This class is concerned with service **Urgent calls** (*25)**,** which enables to establish connection using lines predefined to this kind of calls.

Fig. 4.21.: Sheet – Subscribers / Services settings

4.4.3 Number presentation for outgoing calls

Another important condition checked by PBX is number presentation for outgoing calls.

IMPORTANT: Number presentation for outgoing calls is essential for digital trunks (ISDN) and VoIP.

During outgoing call, extension number can present itself by:
- own number,
- group number,
- ACD group number,
- conference number,
- other extension number,
- trunk own number.

Mentioned above options are available in sheet *Subscribers/Outgoing calls* in field **CLIP***:*

Trunk own number means value entered in field **OwnNo** in sheet *Trunks/Main settings*. If extension number should be presented itself by own number, group number or another extension number, appropriate entry in sheet *Incoming calls/Settings/Main* should be created.

ATTENTION!: Subscriber's number presentation for outgoing calls depends on content of **TrnGr** field in sheet *Incoming calls/Settings/Main*.
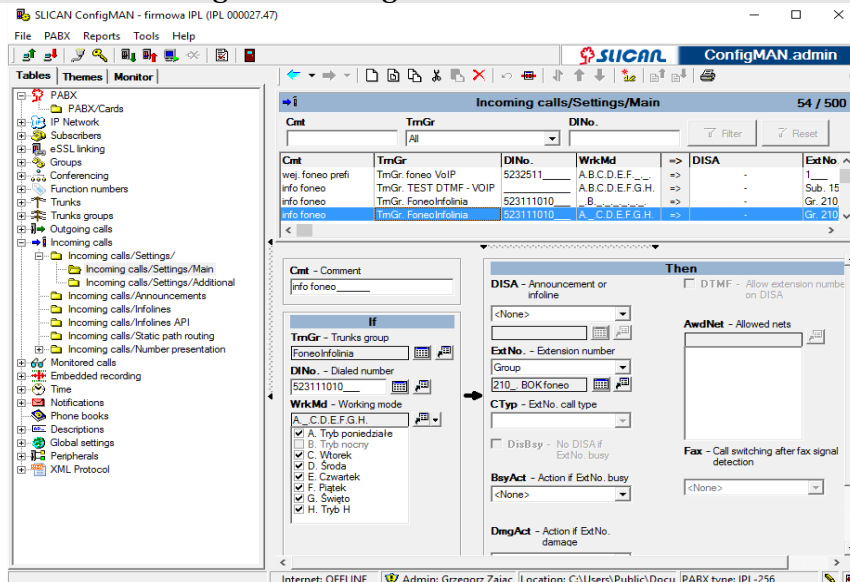
Fig. 4.22: Sheet – Incoming calls/ Settings/Main

Some practical remarks helpful by configuration of number presentation for outgoing calls:

1. Subscriber can presents himself only by number defined in table *Incoming calls/Settings/Main.* If specified number is not defined in this table, it is not possible to present with this number by any subscriber while establishing outgoing call.
   Field **TrnGr** can be additional filter for incoming calls and for presentation in outgoing calls. Additionally it should be noted, that subscriber can presents himself by any number included in table *Incoming calls/Settings/Main* if trunk group used to establish call is the same as trunk group defined in field **TrnGr** of this table.
2. Example settings. You want to obtain presentation 523251100 for outgoing call. If two numbers belonging to the same DDI range are routed to this subscriber, using field **TrnGr –** *Trunk group* (sheet *Incoming calls/Settings/Main*) you can define which number should be presented for outgoing call.

   Example 1:
   PROPER CONFIGURATION
   523201100 →   Extension number 1 →   no assigned trunks group (<Default> in field *TrnGr* – Trunk group)
   523201120 →   Extension number 1 →   DDI

   In above example for outgoing call from extension number 1 number 523201120 will be presented, because both DlNo and TrnGr fields are congruent, so second entry has higher priority.

   Example 2:
   IMPROPER CONFIGURATION

523201100 → Extension number 1 → DDI
523201120 → Extension number 1 → DDI

In this situation PBX cannot unequivocal determine number for presentation while outgoing call from extension number 1. So for outgoing calls number 523251100 will be presented (according first entry in sheet *Incoming calls/Settings/Main*).

3. If for this extension number option „Own extension number" (this parameter is defined in sheet **Subscribers/Outgoing calls**) is selected and there is no assigned own number, for outgoing calls trunk main number will be presented.

4. For two NT terminals (configuration point-multipoint) should be remember to create two trunks groups with different names (ex.: MSN1, MSN2) and assign every ISDN trunk to different trunk group. Symilarly if we use DDI numbers and MSN numbers. Otherwise number presentation for outgoing calls will be improper.

Example 3:
PROPER CONFIGURATION
523201100 → Extension number 1 → DDI in field *TrnGr* – Trunk group
523201120 → Extension number 1 → MSN in field *TrnGr* – Trunk group
In this situation subscriber will be always presented himself properly despite of outgoing trunk group is DDI or MSN.

5. While table of incoming calls configuration, if incoming call is routed to prefix it should be noted to define prefix included at last one digit.

Example 4:
5232511__ 1__ proper configuration
523251___ ___ improper configuration

While improper configuration wrong number presentation for outgoing calls occurs.

4.4.4 Outgoing calls configuration (Outgoing calls and LCR)

Rules determining outgoing calls are included in sheets *Outgoing calls/Short numbers* and *Outgoing calls/LCR/Settings.* In general conditions included in these tables are checked as defined in following drawing.

After analysis of all limitation concerning with extension number settings, PBX look for selected number in sheet *Outgoing calls/Short numbers*, and than decide about connection routing. Connection can be routed via trunk, trunk group or trunk list. If there is needed to choose provider, depending on day time further, more detailed number analysis is performed using sheet *Outgoing calls/LCR/Settings*.

4.4.4.1 Defining outgoing method

In CCT-1668 there is possible to realize outgoing calls in two ways: dialing appropriate outgoing prefix ex. "0" or using Hot Line with quick or slow number analysis. Outgoing method is determined by setting field "OutHL" in sheet ***Outgoing calls***.
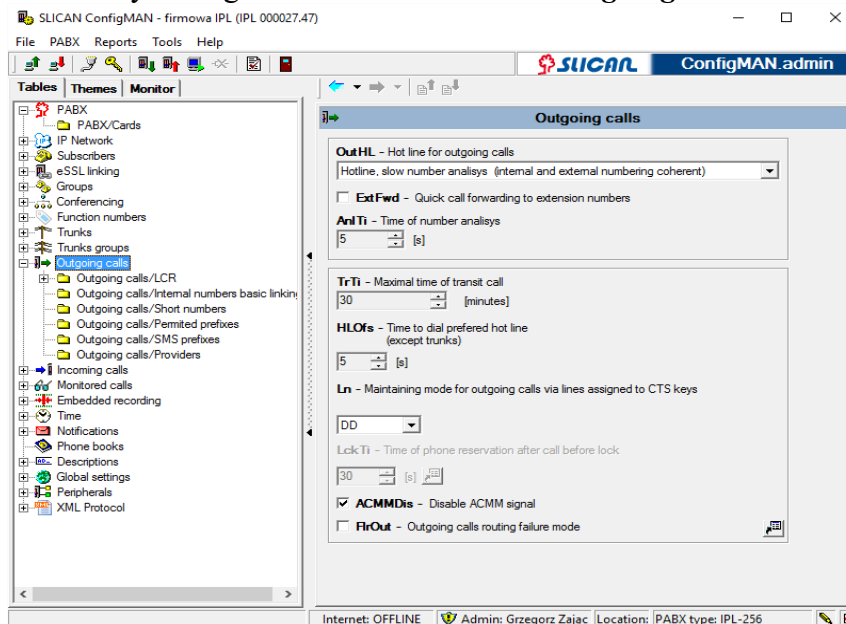
Fig. 4.23: Sheet – Outgoing calls

If <u>hot line with slow number analysis</u> is selected in configuration extension numbering can overlap with external numbering. After each digit dialing, PBX wait for the next digit for a time entered in field ***AnlTi***. If in this time no digit is dialed, PBX starts number analysis. This condition is valid only while dialing first four digits. If number of digits is greater, immediate number analysis is performed in sheets ***Outgoing calls/Short numbers*** or ***Outgoing calls/LCR/Settings***

<u>Quick number analysis</u> is used when extension numbering is separate from external numbering. If quick number analysis is active PBX analyzes dialed number as follows: extension numbers → group numbers → sheet ***Outgoing calls/Short numbers*** → sheet ***Outgoing calls/LCR/Settings***. If first digit is no from extension numbering range, table of outgoing calls is used. If there is no table entry starting with this digit, LCR table is used. Until PBX doesn't find appropriate entry, it waits 15 seconds for next digits (if this time expires and next digit is not dialed, subscriber hears busy signal). After identify any table entry, PBX waits for the rest of dialed digits. Waiting time is defined in field ***AnlTi***.

In the same sheet signal ACMM (upper exchange dialing signal) can be switched off. Until now after dialing zero trunk provider send ACMM signal. Usually CCT simulates such signal for internal subscriber, signals which can be hear by subscriber are the same as in home phone.

Additionally, outgoing method, in specified situations, can be defined using sheet Global settings/Subscriber access classes. In this sheet value of following fields can be set:

· field **UrgC** – defines access class assigned to subscriber after dialing *25, defines access class for urgent calls,

- field **TrnFlr** – defines access class assigned to subscriber, who trying make an outgoing call via damaged trunk,
- field **FlrC** – defines globally, which access class is assigned to subscriber's after selecting service *80 by PBX admin authorized to activate this service.

4.4.4.2 Defining rules of outgoing calls in sheet Outgoing calls/short numbers

In sheet *Outgoing calls/Short numbers* outgoing method is defined. Here connection route and establishing conditions are defined. Also dialed by PBX external number is determined.

In sheet *Outgoing calls/Short numbers* speed access codes are also defined - **SpAcd**. When speed access codes are selected, PBX automatically dial appropriate external number. After dialing short number next digit dialing is possible only in DTMF, it means that for external trunks – especially ISDN trunks – there is no possibility to dial next digits.

After checking **PrvC** field all calls realized by PBX and starting with prefix entered in field **ExtNo** are assigned as private. For such calls, privileges defined in field **PrvPrg,** in sheet *Subscribers*/*Outgoing calls* are checked.

After checking **HidExt** extension number is not included in extension number list in CTS phone.

Fig. 4.24: Sheet – Outgoing calls/Short numbers

In PBX different numbers can be defined as "outgoing digits" (field **ExtNo.**) and each "outgoing digit" could be granted to other access class - **ACls**. It enables to differentiate call routing with upper exchange for outgoing calls - **OutMth**.

IMPORTANT:

If in sheet *Outgoing calls/Short numbers* we can define outgoing digit in field **ExtNo** and if in field **OutMth** trunk, trunk group or trunk list is defined, seizing an external line takes place immediately after dialing an outgoing digit by subscriber. If trunk is ISDN trunk, sending digits to provider is overlapping (option Overlap).

It is always possible to route call to sheet *Outgoing calls/LCR/Settings* after dialing an outgoing digit. In this situation external line is not immediately seized. External line seizing takes Line will be seized after complete dialing number by subscriber, and for ISDN number will be send to provider in block (option In block).

4.4.4.3 Selection of least cost route (*Outgoing calls/LCR/Settings*)

Slican PBXes enable to route outgoing calls to different providers with least cost of connections. Outgoing calls can be routed to fixed or preselected providers according extension number privileges and day time. Additionally it is possible to limit establishing outgoing calls up to free minutes package granted by provider (detailed description is presented in chapter concerning free minutes package).

System checks if dialed number is coherent with prefix entered into field **No**. Access class **ACls** enables to define, if subscriber dialing outside is authorized to do this with specified entry (including defined in this entry an outgoing route: list, trunk group, trunk). After fulfiling all conditions defined in clause *If:*, outgoing method and number dialed by PBX is determined in clause *Then:*.
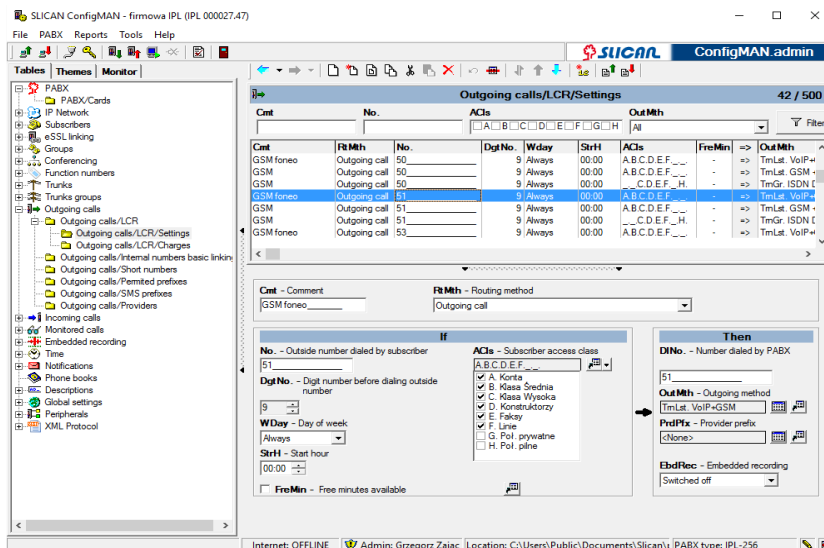
Fig. 4.25.: Sheet – Outgoing calls / LCR / Settings – external outgoing calls

Number which is sent by PBX to the upper exchange consist of provider prefix entered in field **PrdPfx** and number entered in field **DlNo**. Prefix from field *No* is replaced by prefix from field **DlNo**, a remain digits are send without changes.

Outgoing calls in Slican PBXes can be routed to Incoming calls table. Such situation takes place if You choose option *Back as incoming calls* in field **OutMth**. Number dialed by **LCR** table (field **DlNo.**) is checked by incoming calls table. This solution can be used to force full ten digits numbering plan for internal calls – due to internal calls charging.

4.4.4.4 Detailed rules of routing calls in LCR table

Dialed number analysis by PBX means check all conditions defined in LCR table. If system encountered first consistent entry, check if trunk is busy. If trunk is busy, system search next coherent entry, which can be realized.

If while number analysis in LCR table appropriate entry cannot be found, busy signal is send to the caller.

If all calls are analyzed in LCR table, entry defined PBX reaction on any other dialed number should be added. It is so called empty entry, defining rules for any others numbers dialed.

Cmt	RtMth	No.	DgtNo.	Wday	StrH	ACIs	Net	FreMin	=>	OutMth	PrdPfx	DINo.	EbdRec
Orange	Outgoing call	050___	10	Always	00:00	_.B.C.D.E.F.G.H.		-	=>	TrnLst. Basic	<None>	50	-
Era	Outgoing call	060___	10	Always	00:00	_.B.C.D.E.F.G.H.	-	-	=>	TrnLst. Basic	<None>	60	
National	Outgoing call	0___	10	Always	00:00	_.B.C.D.E.F.G.H.	-	-	=>	TrnLst. Basic	<None>		-

Fig. 4.26.: Sheet – Outgoing calls / LCR / Settings

If there is obvious to assure continuous calls establishing, for example while damaged or busyness of whole trunks group, additional entry can be inserted, with the same connecting conditions but with another outgoing method. In this second entry another trunk or trunks group should be determined. This trunk or trunk group will be occupied if previous will be busy. It is presented on following figure.

Cmt	RtMth	No.	DgtNo.	Wday	StrH	ACIs	Net	FreMin	=>	OutMth	PrdPfx	DINo.	EbdR
Orange 1	Outgoing call	050___	10	Always	00:00	_.B.C.D.E.F.G.H.	-	-	=>	Trn. VoIP Anon SIP	<None>	50	-
Orange 2	Outgoing call	050___	10	Always	00:00	_.B.C.D.E.F.G.H.	-	-	=>	TrnLst. Basic	<None>	50	
Era	Outgoing call	060___	10	Always	00:00	_.B.C.D.E.F.G.H.	-	-	=>	TrnLst. Basic	<None>	60	
National	Outgoing call	0___	10	Always	00:00	_.B.C.D.E.F.G.H.	-	-	=>	TrnLst. Basic	<None>		-

Fig. 4.27.: Sheet – Tables / LCR / Settings

Basic LCR entries are included in PBX database after first launch, but new entries can be add according user requirements. If there is need to add large number of entries, tool "Add in multiple LCR prefixes..." can be used. This tool can be accessible above LCR table.

Another available tool is „Multiple prefix modification wizard", it enable to modify all entries in LCR table ex. deleting 0 from all entries or adding any prefix.

4.4.4.5 Free minutes packages

Lastly providers of fixed or GSM trunks offer in subscription free minutes packages. To answer this offer and to meet customer expectations in Slican PBXes mechanism of using free minutes packages was implemented. Configuration of this functionality should be started from sheet ***Trunks / Free minutes***.

Figure 4.28.: Sheet Trunks / Free minutes

In this sheet You can see following fields:

– ***OutCnt*** – this field shows total time of connections established using specified trunk since last minute counter reset. Automatic (according settings in field **RstDay**) or manual resetting (using icon ✖) is possible.

ATTENTION!

While manual resting of counters **OutCnt** and **FreCnt** it should be observed that after clicking icon counter are resetting even if You are not connected currently to the PBX. After performing this operation, configuration should be uploaded to the PBX.

– **FreMin** – in this field number of free minutes should be entered. If „0" is entered (no free minutes package) for all trunks belonging to trunk group defined in this LCR entry, trunks are occupied evenly according time of established outgoing calls (occupied is trunk with currently lowest value of **OutCnt** field).
  If in LCR table entry outgoing calls are routed via list, elements of this list occupation is done as usually, always from the list beginning. If we have no free minutes packages, **FreMin** check-box (in LCR table) should not be checked. Otherwise – if this field is different than „0", free minutes package is granted, two options are possible:
  - if in LCR table, in specified entry, **FreMin** field is checked this trunk belonging to specified trunk group is occupied, for which number of unused free minutes is greatest (difference between numbers in fields **FreMin** and **FreCnt** is checked);
  - otherwise, if in LCR table, in specified entry, **FreMin** field is not checked, this trunk belonging to specified trunk group is occupied, for which number of unused free minutes is lowest, so using free minutes is so quick as possible.
– **FreCnt** – this is a number of already used free minutes (additionally expressed in percent), since last counter resetting. Resetting methods are the same as for field **OutCnt**.

- **RstDay** – defines day of month when all mentioned above counters are reset (it always take place at 23:59:59 hour of defined day). Once counters are reset, time of outgoing calls is calculated from the beginning.

Check-box **FreMin** determines method of using free minutes; this check-box is set for specified LCR table entry as it is show on following figure.
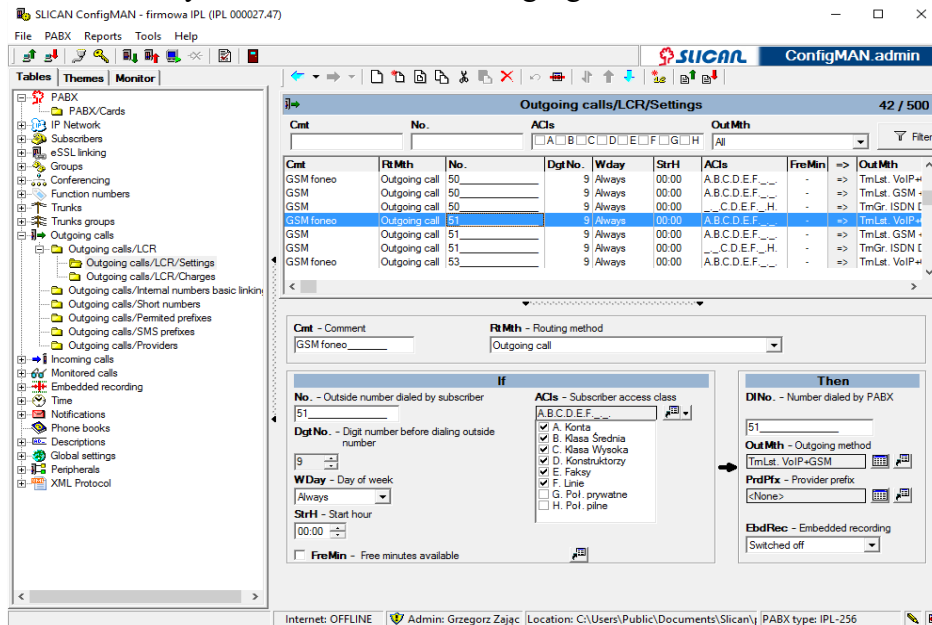
Figure 4.29: Check-box determines if for specified LCR table entry free minutes are used

4.4.4.6 Automatic phone locking

In sheet *Subscribers/Outgoing calls,* in field **PhLck** phone behavior after working mode change or switching on or off automatic phone locking can be defined (See *Time/PABX working modes*), following settings are possible:

1. *Switched off* – PBX working mode change has no influence on phone locking.
2. *Lock* – phone is locked if in current working mode has automatic phone lock field checked. Lock can be deactivated using service *20.
3. *Lock and unlock* – automatic locking and unlocking phones depends on current PBX working mode (manual phone unlocking is not necessarily).
4. *Automatically* - automatic locking and unlocking phones depends on current PBX working mode (manual phone locking and unlocking is not possible).
5. *Lock after time period* – if this option is set phone is locked after finishing call and time between finishing call and locking phone is defined in field **LckTi** (*Global settings/Outgoing calls*).
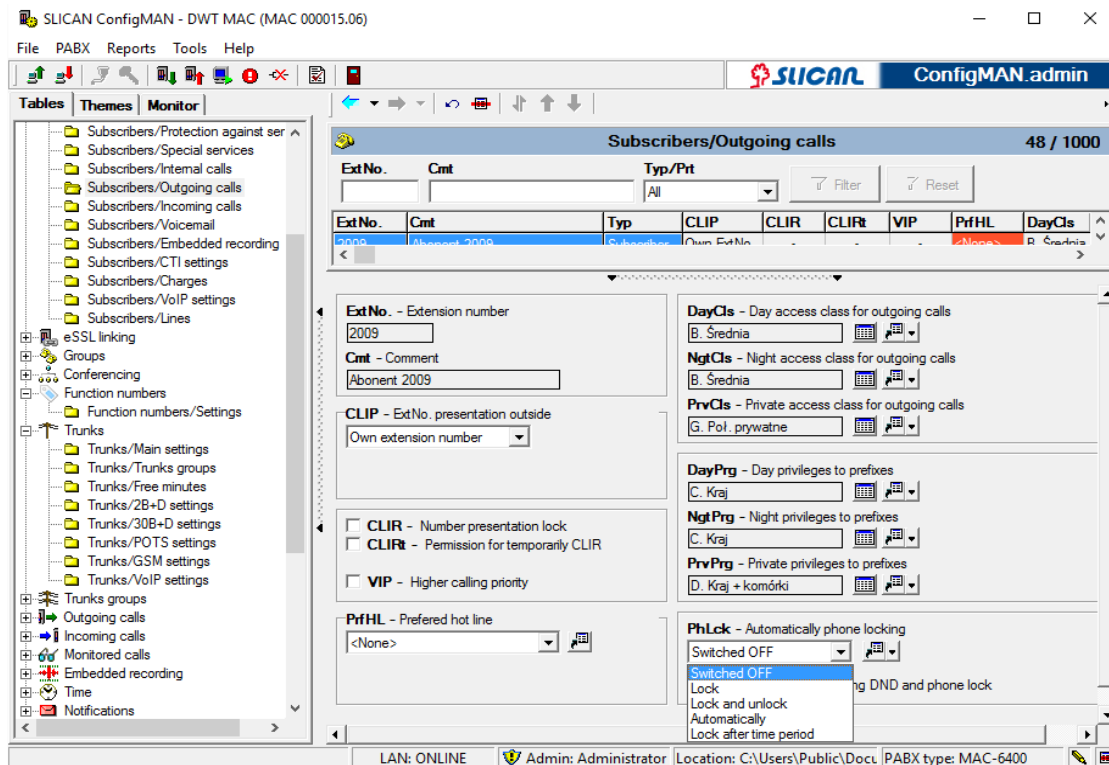
Figure 4.30.: Sheet Subscriber / Outgoing calls

Field **DNDLck** define DND service status while phone is locked (it concerns phone lock service *21, as well as automatic lock). If this field is checked after phone lock DND service is activated for specified extension number.

4.4.4.7 Provider table

With LCR table provider table is correlated. In provider table data concerning provider type and prefix are stored. These data are automatically added to dialed number before sending number to upper exchange.
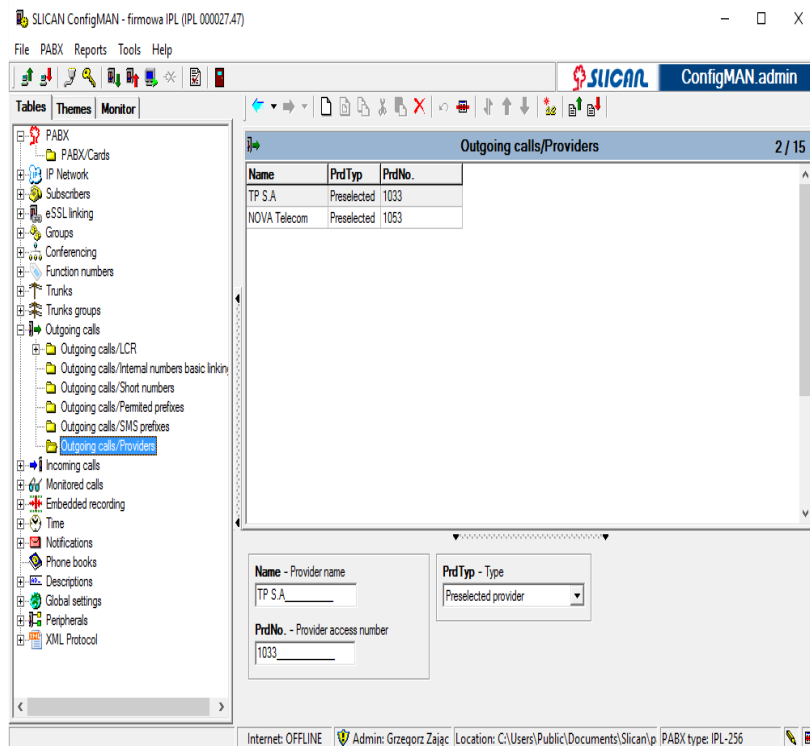
Fig. 4.31: Sheet – Outgoing calls / Providers

In this sheet we can define one of two provider type – figure above:

➢ **Preselected provider** – provider, who enables to establish connections with other providers after dial appropriate prefix,

➢ **DTMF gateway** – ringing in provider, PBX dial provider access code and after establishing connection with DTMF gateway ultimate number is dialed.

4.5 SMS service

Set of PBX settings which enable to send and receive SMS messages using Slican computer software:

- Slican PhoneCTI (available form version 6.XX or never)
- Slican TelefonCTI (application for older firmware versions)
- Third party applications

Applications PhoneCTI and TelefonCTI are subject of licesing. Please note, that SMS functionality for TelefonCTI application requires additional license – CTI.Manager (earlier named CTI.plus).

SMS for third party applications is also subject of licensing (sheet: **Global settings/Licenses** field **SMS**).

4.5.1 SMS service launching

To launch SMS functionality it is neccessarily to make following settings:

- GSM trunks(sheet: **Trunks/GSM settings**, in field **SMSStu=Switched ON**) which will send SMSes from subscribers (sheet: **Subscribers/CTI Settings** field **AwdSMS**)
- GSM trunks for which SMS functionality will be control by third party application (sheet: **Trunks/GSM settings**, in field **SMSStu=CTIP/XML**)
- GSM trunks, which will be receive and route SMS messages to specified subscribers or groups in sheet: **Trunks/GSM Settings** field **SMSSta.**

4.5.2 Restictions and charging cost for sending SMSes

- **Global Settings/Remain** field **SMSPre** – set the maximal length of SMS Premium (SMS with higher fee)
- **Outgoing calls/SMS Prefixes** – restriction table – defines required subscriber privileges **Prg**, allowed prefixes and digit number in field **No,** as well as SMS price **SMSPrc**.

Into this table you can enter two SMS types:

- SMS premium (with higher fee) – number format without sign "+" at the beginning, length of this SMS doesn't exceed number enteredinto field **SMSPre** located in sheet **Global Settings / Remain**.

- Standard SMS with sign "+" at the beginning, can be longer then number entered into field **SMSPre**

Fee is charged according field **SMSPrc** and added to field **Blnc** in sheet **Subscribers/Charges**.

If length or prefix are not compatible with entered data - SMS is not send.
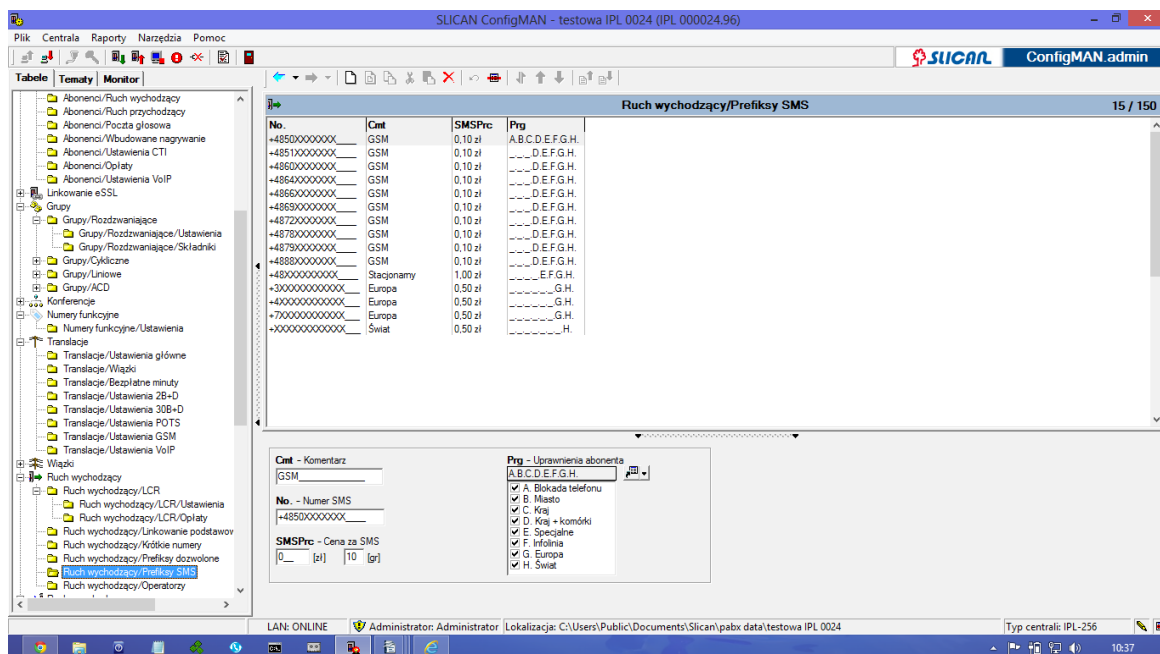
Fig. 4.4.: Incoming calls/ SMS Prefixes

4.5.3 Incoming SMS routing

SMS message receiving by PBX can be realized in three possible ways:

- if SMS text is personalized (extension number at the beginning) – SMS is routed to specified extension number
- if SMS text is not personalized – SMS receiving is done by appointing appropriate subscriber or group in field **SMSSta** (static routing).
- If SMS text is not personalized and field **SMSDyn** is checked (dynamic routing) – SMS is routed to subscriber who earlier sent SMS to this number. Dynamic routing has higher priority then static routing but is limited by time entered in field **PtRth**, in sheet **Incoming calls**

4.6 Function numbers

Function numbers were designed mainly for automatic distribution of internal calls, depending on PBX working mode. While dialing such number, ringing will be routed to subscriber, group, announcement, infoline or paging group.

This functionality may be entered as connection goal in:

- **Subscribers/Incoming calls - BckExt** – Backup subscriber,
- **Subscribers/Outgoing calls - PrfHl** – Preferred hot line,
- **Incoming calls – OphNo –** Orphan calls number,
- **Incoming calls/Infolines** – in fields **DgtAct** – digit dialed while infoline announcement after pressing appropriate button, **DefAct** as a default action if nothing is pressed, **BsyAct** – action if subscriber or group are busy,

For function numbers following settings are available:

- **ExtNo.** – enables to assign extension number,
- **DefFN** – defines default routing after dialing function number (Subscriber, Group, Announcement, Infoline, Paging group)
- **FN** – if field **DefFN** is defined, enables to appoint exceptions of default calls routing in PBX; if field **DefFN** is not defined, when you dial function number, call will be routed always according current content of field defining PBX working mode,
- **Cmt** – comment entered in this field has only information purposes,
- **Net** – by assigning specified net, you can limit access to this service (by default it is net A. Subscribers),
- **AwdNet –** defines nets, to which can access caller if he/she is routed to specified function number (by default it is net A. Subscribers, B.Groups), ex. PBX subscriber in internal calls can access nets A to F (dfefined in field **AwdNet**), If subscriber A dial extension number of subscriber B and subscriber B routes this call to function number which connect subscriber A to infoline, from this moment Subscriber A has access only to nets defined in field **AwdNet** of function number;
- **CLIRi** – after checking this field, function number will be not visible subscriber list on CTS system phone.

4.7 Unreleased numbers

Unreleased numbers are numbers, that can be reached only locally in one PBX. Voicemail number is a unrealeased number by default. Unrealeased number can be created by adding new **Function number** and selecting field **UrsNo.** Those kind of elements can not be dialed by subscribers from other linked PBXes (with eSSL protocol).

4.8 „Line" service in Slican PBX

Main idea while designing *line* functionality was creation of comfortable tool which enables to handle large incoming calls by consultants. This service enables to organize, monitoring and and maintenance of incoming, internal and outgoing calls. It enables to control calls similarly to operator console with multiple external phone lines connected. Due to its flexibility of "line" service, it is possible to use this functionality in large institutions as offices, banks, hospitals, dispatch centers, as well as small consultation offices. To support this service an application **ConsoleCTI** was developed. It is integrated computer-phone system, installed on computer used by subscriber of PBX extension number. Application visually presents all statuses of phone assigned to this application. It enables to maintain call by touch screen or computer mouse, directly on screen instead of buttons and phone keyboard. This application has many additional functions which simplify cooperation with phone, ie. call maintenance visible on monitor, so all status are very clear and intuitive, with many additional information; configurable contact list; call history; visible on screen phone keyboard and phone book.

– Basic task realized by this service is grouping many calls routed by internal or incoming calls mechanisms (ex. by incoming calls table, static path routing) to specified number – so called *line* with specified channel number (channel number defines number of possible simultaneous calls). All calls can be simultaneously observed and maintain by several consultants with Slican system phones (so called 'familiar phones'). Every consultant has possibility of view caller number and select who should be answered first. Several *lines* should be create and freely assigned to specified phones. It enables to create groups maintaining different problems. Currently maintained calls can be temporarily parked or transfer to another person.

– For outgoing calls consultants can realize calls using specified *line*, in this situation presentation is made using *line* number. Due to hide subscriber own number, all call back will be routed to *line* number, not to consultant own number.

– Using *line* service is possible after appropriate configuration and its functioning is based on BLF lamps as well as associated with these lamps buttons on system phone. Programming buttons maintaining *lines* can be done only from inside **ConfigMAN** application, it cannot be done from phone menu. To switch on the service, multichannel virtual line should be created in PBX configuration, such created line should be assigned to 'familiar phones'. Line functionality is not subject of licensing. Maintanace of this functionality is possible using Slican system phones CTS-330, CTS-202 and CTS-203. Service was introduced in firmware version 6.22

– *Lines* can be observed by single- or multichannel lamps or used simultaneously by both lamp types:
- single-channel lamp – shows status of one voice channel belonging to specified *line*, it represents one one call for incoming or outgoing calls. It is possible to configure several single-channel lamps for specified *line*, in such situation every lamp represents one call status. This status is visible on all phones (phones of all consultants), which have lamp for this line configured.
– Behavior of single-channel lamp and status of channel assigned to it:
  – lamp is blinking – line incoming call; preview of call of it answering on any 'familiar phone' is possible;
  – lamp is shining – call is on; it is possible to park this call from phone with call is established to, 'familiar phones' can only view status of this call

–   lamp is flashing – call is parked; it is possible to unpark this call on any 'familiar phone' by pressing button and lifting handset;
–   lamp is switched off – there is no call on line; it is possible to take line for outgoing call, by any 'familiar phone'

Number of configured single channel lamps decides about quantity of observed calls on this line. If number of channels is greater than number of configured lamps, there will be no possibility to preview status of all channels. If all lamps are busy (ringing, call, hold) there is no possibility to establish outgoing call.

- multichannel lamp – enables to observe incoming calls and establishing outgoing calls, using one button. It shows status of all channels assigned to specified line.

–   Behavior of multichannel lamp and status of channel assigned to it:
–   lamp is blinking – there are line incoming or parked calls; preview of calls or answering selected call is possible;
–   lamp is switched off – there is no incoming or parked calls
–   lamp never shine

Multichannel lamp presents only incoming and parked calls.

4.8.1 Service configuration

PBX global configuration

Launching *line* functionality:

1. in sheet **Subscribers/Main settings** checkbox **Ln** should be checked – it define object – an account.

2. **Subscribers/Incoming calls** – setting of additional configuration options (number of calls waiting on line, music on hold, announcement):
    ➢   **CWt** – defines *maximal number of calls* managed by line for incoming calls.
    ➢   **WtMel –** enables to change melody on hold or defining an announcement with desired information
    ➢   **QAnn** – enables to inform caller about position in queue

3. configuration of *line* parameters in sheet **Subscribers/Lines**:
    ➢   **ExtNo –** extension number – editable in sheet **Subscribers/Main Settings**
    ➢   Cmt – comment – editable in sheet **Subscribers/Main Settings**
    ➢   **Lna** – setting of call maintenance type:
        - outgoing and incoming calls with taking first free channel – on system phone it is visible by shine on the first from the top lamp which is switched off, it concerns lamps assigned to line channel.
        - outgoing and incoming calls with taking channel which was used earliest – every call shine on phone lamp which is assigned to this channel.
    ➢   **WtchC** – by default transfer call immediately free line channel. If this option is checked, channel if free only after answering made bu dialed subscriber. If dialed subscriber doesn't answer, it is possible to return back to this call by pressing line button and lifting handset.
    ➢   **MaxChn** – defines *maximal channel number* managed by line and assigned to it number of simultaneous established calls (incoming and outgoing). Informs about default number of single-channel lamps, which should be configured on 'familiar phones'. Available values are from range 0-99, by default 5.
    ➢   **RezOut** – reserves channels *only* for outgoing calls. It decreases number of channel for incoming calls. This option is important maintenance of calls,

which should be routed outside. Values of this field can be **from 0 to MaxChn-1**.

Configuration of line maintenance and BLF lamps/buttons for system phone.

Configuration is performed separately for every phone, which should maintain line functionality

- ➢ **Peripherials/CTS/Settings:**
  - ◦ **LnRng** – setting ringing type used for line functionality. Line ringing signal can be different then set on phone for direct incoming calls.
  - ◦ **LnAns** – checking this check-box enables to answer call, which wait for longest time, directly by lifting handset (without using button assigned to this call)
  - ◦ **LnOut** – defining line take on automatically for outgoing calls.

- ➢ Configuration of line type in sheet **Peripherals/CTS/Buttons**

- ➢ In this sheet it is possible to configure phone buttons, which will be assigned to the lines. After pressing such button opens window in which field **Ln – line** can be checked. Selection of specified line force defining maintenance method for BLF/phone buttons. By default line if configured to work in single-channel mode, with channel number equals to content of field **MaxChn** in sheet **Subscribers/Lines**. Settings acceptation assign next phone function button to maintenance of this line. If channel number is greater then number of buttons in system phone, ConfigMan application proposes assignment of less number of channels. Number of managed channels and number of configured BLF lamps can be changed using slider. If for specified phone you set less number of channel then application default, maintenance of some calls will be not available for this phone.

  If you select check-box multichannel line, one lamp will be use to all channels maintenance.

4.8.2 Calls maintenance

4.8.2.1 Incoming calls

Calls incoming to the *line* are visible on CTS phone using BLF lamp assigned to the *line*. Additionally acoustic signal according settings of field ***LnRng*** in sheet **Peripherals/CTS/Settings** is also possible – individually for every 'familiar phone'. In phones where call is established to, acoustic signal is limited to simple 'beep' signal.

For ***single-channel*** mode displaying caller presentation (number or comment) takes place after pressing phone button assigned to appropriate channel when button lamp blinking. Presentation time is equal to 5 seconds, when during this time someone lift phone handset call will be established and ringing on another 'familiar phones' will be finished. Another method of call answering is lifting handset without pointing ringing line. In this method call will be established with caller who wait for the longest time. After pressing right kontext button – Reject during caller presentation, number is not longer presented, acoustic signal is muted and standby screen is displayed. Rejecting, concerns only this phone, ringing is still present on another phones. Pressing „SPK OFF" button during incoming call – mutes acoustic signal for all calls and enables another actions on phone, ex. reviewing phone book or establishing outgoing call.

For multichannel line display of caller presentation is done by repeatedly pressing button assigned to this line. It enables to review all currently ringing calls. After displaying last call, next pressing this button displaying presentation of first caller once more. Apart from mentioned above notice, establishing call and phone maintenance is exactly the same as for single-channel line.

4.8.2.2 Outgoing calls

In standby mode, when there is no call waiting on line, phone behaves as no line service was configured. For *single-channel* mode – during standby or for incoming calls, longer pressing button of free channel reserves this channel for outgoing call. Channel name will be displayed on phone screen and lamp is shining on all 'familiar phones', which informs about it reservation. Dialing number is possible during next 30 seconds, otherwise reservation expires. For *multi-channel* mode reservation is done during longer pressing button, in this period line channels change and numbers of channels are presented on phone screen. After release button when specified channel is presented, channel will be reserved for outgoing call. Call can be established as for single-channel line.

For outgoing calls, *line* number is presented instead caller number. Privileges are the same as for caller privileges.

4.8.2.3 Park and transfer calls

Active call, bothy incoming and outgoing can be parked with context menu button, situated on right side of phone screen. After pressing such channel by another line user information about subscriber who parked call is displayed. Call can be unpark by any of 'familiar phone' if pressing appropriate channel button and lifting hanset.

Call transfer is realized as usual for system phone: pressing „flash" button and dialing subscriber number or by selecting configured function key. If for outgoing calls you want to present line number, after pressing „flash" button, you should reserve line by pressing button assigned to free channel and dial requested number. Transfer calls is possible with announcing or without it. Call transferred to the line never returns to subscriber who transfer it. If in line configuration option for monitoring call **WtchC** is active, subscriber who transferred call can return back to this call, if requested number doesn't answer.

Maximal number of calls possible to maintain in line functionality:
–   outgoing calls = **MaxChn**
–   incoming calls = **Cwt**, with limitation, simultaneously established calls = **MaxChn** – **RezOut**. Remain calls don't reserve channels because there are waiting in queue and there are not signal on phones in any way. If any channel is free, call call waiting in queue is established using this channel.

4.9 Incoming calls

Before incoming call is routed to specified extension number it is a subject of detailed analysis. Before this analysis proper CLIP formatting is executed.
First rules included in sheet *Incoming calls/CLIP categories* are executed to standardize CLIP format. Three options are possible: *International*, *National* and *Other*. After number classifying appropriate prefix is added to this number. Prefixes are defined in sheet *Incoming calls*, field **IntPfx** for *International* format and **NtnPfx** for *National* format. Due to incoming number standardize, number presented to internal subscriber enables to ring back to this number without any modification.

4.9.1 PathFinder – Call routing

To every incoming call 3 parameters are assigned:
1. Caller number – so called CLIP (no caller number, so called CLIR – is also an information);
2. Number dialed by caller;
3. Target in PBX – often it is extension number, it may be also an announcement or infoline.

Due to these parameters, incoming calls are firstly routed to PathFinder – static and dynamic – if there are no appropriate rules for PathFinder – call is routed directly to sheet *Incoming calls/Settings/Main*.

PathFinder – Incoming calls/Static path routing

If above mentioned call parameters are coherent with any entry in sheet *Incoming calls/ Static path routing*, action defined in this entry will be executed. Otherwise *dynamic path routing register* will be checked.
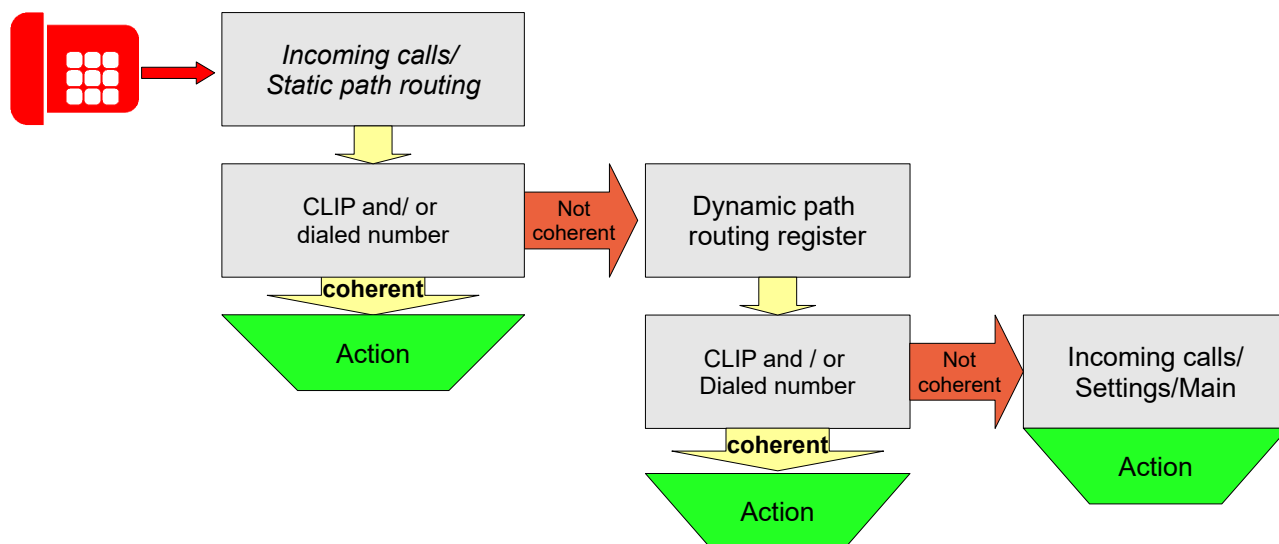
Depending on option selected in field ***CLIP:***
- ***Prefix –*** appropriate entry will be applicable for all incoming calls, originated from numbers begins with digits defined in this field (it is possible to enter whole caller number in international format).
- ***Unknown -*** appropriate entry will be applicable for all incoming calls without number presentation or with restricted caller number (CLIR).

If field **DlNo.** stay empty, it means, that entry will be applicable for calls incoming to every trunk number connected to the PBX.

Attention: Field **Prefix** in sheet *Incoming calls/Static path routing* should starts with CC (Country Code), ex. **48** for Poland!

In sheet *Incoming calls/Static path routing* few types of routing can be realized, routing type can be individual for every single entry.
PBX behavior is selected in field ***RtTyp:***
1. ***No path routing*** – this option enables exceptions for static path routing;
2. ***No dynamic path routing*** - this option enables exceptions for dynamic path routing and routes calls to analyze in sheet *Incoming calls/Settings/Main;*

3. ***Connect with*** – such type of path routing enables to route call according content of field ***Act***. Call can be routed directly to specified subscriber or group, doorphone, to infoline, to system modem, to table of Outgoing Calls (and outside via this table), access class checking is omitted and number from table of Outgoing Calls is automatically dialed. In this situation ***Dynamic Calls Register*** isn't checked;

4. ***Change dialed number*** – according CLIP number, dialed by caller number is changed into number defined in field ***No.*** and call is analyzed in table of Incoming Calls without analyzing in register of dynamic path routing;

5. ***Account login*** – in this situation, after checking CLIP number and dialed number account defined in field Act is recalled and caller can use PBX as any other PBX subscriber;

6. ***CallBack*** – an account is logged for which caller CLIP is equal to content of **MblPh**, in sheet ***Subscribers/MobilePhone***. To enable regular costless from Mobile Phone to PBX extension numbers caller disconnect should take place inside 10 seconds, otherwise call is analyzed in incoming calls table and routed according rules defined in this table. This option is subject of licensing and activated by checking field **CallBack** in sheet ***Subscribers***/***MobilePhone***.

7. ***Open doorphone*** – if conditions of static path routing are fulfilled – doorphone will be opened without establishing call;

8. ***Trigger relay*** - if conditions of static path routing are fulfilled – specified relay will be trigger;

9. ***Reject*** – in this situation caller hears busy signal;

10. ***No answer*** – caller hears ring-back tone and after three minutes disconnection takes place.

PathFinder – Dynamic path routing

Dynamic path routing enables to store call connection paths to using they in the future to establish connection, directly, without infoline, group or other subscribers.

If in register of dynamic path routing specified call can be found, call will be routed to target subscriber included in this entry. If there is no appropriate entry in dynamic path routing register, incoming call will be routed via table of incoming calls included in sheet ***Incoming calls/Settings/Main***.

I. Dynamic path routing <u>initiated by incoming calls.</u>
A.Incoming call with **CLIP** calling: 22 3333333. According setting for incoming calls (***Incoming calls / Settings / Additional*** field **PthRt.**) - call should be routed to Infoline, **Dialed number**: 52 3251111, after playing announcement call is routed to subscriber 142 – target subscriber **(1)**. Call was finished.
(2). PBX stored call path:

1. Caller **CLIP** =223333333
2. **Number** called by **caller** 523251111
3. Extension number of target subscriber **Sub**=142.
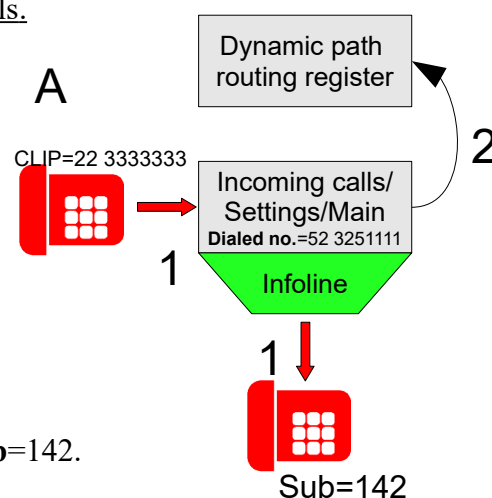
B. After some time the same caller (with CLIP=223333333) dial once more the same number **(1)**, he/she will be immediately routed with the same extension number as previous – the same target subscriber omitting Infoline. Routing path in register will be refreshed **(2)**.
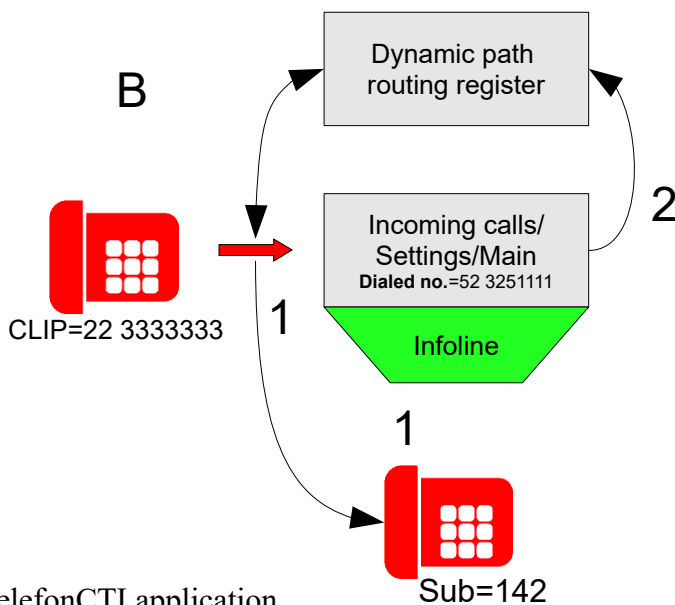
Mentioned above routing path.: caller number, number dialed by caller and target subscriber will be automatically deleted when:

– Sub switch on DND,
– Sub is busy,
– Sub do not answer the call,
– Sub rejects call from CTS phone or TelefonCTI application,
– expires time of path validity – this time is defined in field **PthRt** in sheet *Incoming calls,*

and call will be routed to Infoline as it was at a first time (according settings for incoming calls).

Incoming call path routing takes place only if net defined in sheet *Incoming calls/Settings/Main* is the same as target subscriber net.

Attention! Incoming calls are not routed if:

– call was answered by subscriber who is not a member of any PickUp group,
– call was transferred to subscriber who is a member of other PickUp group,
– in sheet *Incoming calls/Static path routing* the exceptions can be set – by setting option *No path routing*.
– there is no presentation of incoming call number (CLIR).

Dynamic path routing of incoming calls, as presented in example I, can be switch on in sheet *Incoming calls/Settings/Additional,* in field **PthRt**. Calls will be routed (their paths will be stored), only for specified entries in table of incoming calls.

II. – Dynamic path routing initiated by outgoing calls.

A. If PBX subscriber SubA dials GSM no. then (it is not important if call is established or no) after finishing this call PBX stores path of this call in its memory.
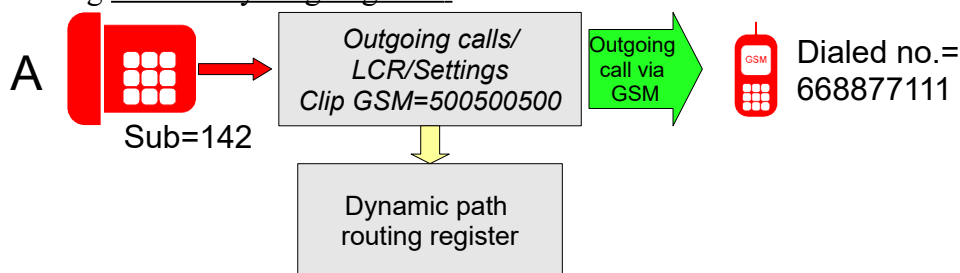
B. Due to this feature GSM user can call back to the no. which was
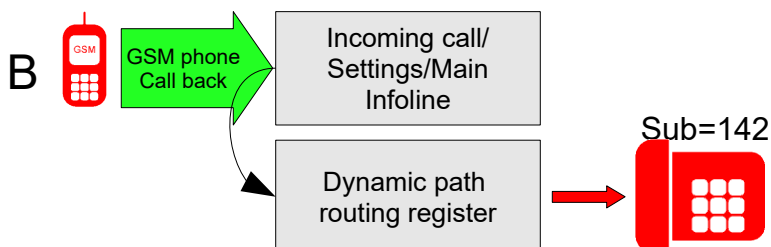
presented (500500500)
and establish call
directly with SubA.

Attention:
Routing path stored in dynamic path routing register will be erased if GSM user calls back and call
– will be rejected;
– will not be answered;
– target subscriber will be in DND status;
– time of path validity will expire – this time can be defined in sheet *Incoming calls,* in field **PthRt**, .

If during call back GSM user do not want to establish call with SubA, but routed according incoming call settings, GSM number presentation should be hidden. For mobile phones dialed number should be followed by code *31*.
Dynamic path routing of outgoing calls, <u>as presented in example II</u>, can be swith on in sheet Trunks groups/Settings**,** in field **PthRt**. Calls will be routed (their paths will be stored), only for specified trunk groups.

Attention! *PBX reset erases dynamic path routing register.*

4.9.2 Managing incoming calls via incoming calls table

In this place you should defuine basic conditions, which must be fulfilled to establish call with specified extension number.

Defining rules of incoming calls routing using DDI and MSN numbering consist of:

*   defining, which number is dialed by caller,
*   via which trunk group call is established (defining trunk group is important if extension number should present itself with trunk group number),
*   PBX working mode when routing rule is valid.
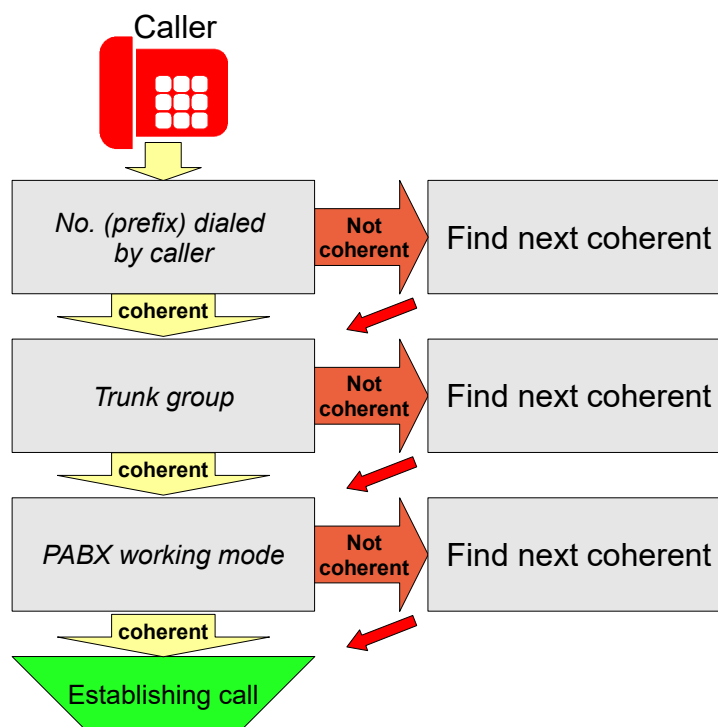
4.9.2.1 Routing mode

Target, where the call is routed (field **ExtNo.**) can be achieved directly or by (DISA/Infoline announcement - field **DISA**. Options available during direct routing:

- Action when (target) **ExtNo.** is busy – this functionality enables to alternative routing call - field **BsyAct.**
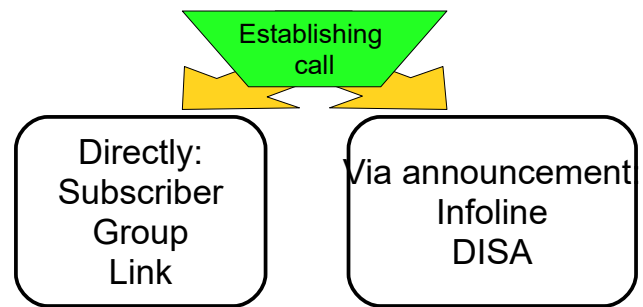
Options available during routing via announcement:

- Action when (target) **ExtNo.** is busy – this functionality enables to alternative routing call - field **BsyAct.**
- If call is routed to specified target (via DISA or Infoline announcement) by extension number dialing made by caller using DTMF – such possibility should be sometimes limited (ex. direct dialing CEO extension number). This type of limitation can be realized by net selection in sheet *Incoming calls/Settings/Main* – field **AwdNet**.
- DTMF receiver can be switched off during DISA announcement, it disable possibility of dialing any extension numbers by callers, after playing announcement call is routed to extension number defined in field **ExtNo**.
- Automatic fax detection. Automatic detection of fax calls is possible during announcement. To enable fax signal detection – 1100Hz signal, in field **Fax** in sheet *Incoming call/Settings/Main* should be defined, if fax call should be routed to specified subscriber or group. To proper fax detection, fax should be configured to send fax signal (1100Hz) after establishing call. If fax signal detection isn't realized, it may be concerned to lack of fax signal at the calling device or manual working mode.
- **DisBsy** – this functionality is dedicated to analogue trunks – if call target is busy, DISA announcement is not switched on, to doesn't expose caller on extra costs.
- *Incoming calls/Settings/Main* field **Ctyp.** During incoming call in the same time starts playing DISA announcement and ringing on target subscriber or group with following remarks:

  - If call is answered before finishing announcement playing – announcement is interrupted.
  - Possibility of dialing DTMF digits during announcement is switched of.

4.9.2.2 Possibility of selecting other ext.no on DISA announcement – function *

Slican PBXes enable – while incoming call on DISA announcement, withdraw of previously dialed extension number to dial another extension number is possible. This can be made using function *.

After dialing number on DISA announcement, if target number:
- is busy, damaged or unlogged we can withdraw dialed number in 10 sec.
- no answer, we can withdraw dialed number to the end of ringing signal
- has enabled DND service, we can withdraw dialed number in 10 sec. after finishing announcement

There is no possible to use this service if while DISA announcement wrong number is dialed (there is no such extension number, there is no access to dialed number net). Using * function is not possible while selecting service.

4.9.2.3 Incoming calls/Settings/Additional

VIP – After checking this field for specified entry, incoming call routed via this entry is not limited according rules concerning trunk group capacity for incoming calls. This option is useful while creating „incoming numbers" which shouldn't be busy.

MaxInc – maximal number of calls which can be realized in the same time in the same time.

Cdrn – maximal duration time of call routed by this entry. It is useful if you use free numbers.

EbdRec – priority of recording specified trunk/trunk group/trunk list. If other value then Switched off, calls routed via this entry are recorded.

PthRt – if this field is checked, path of call routed via this entry is stored in dynamic path routing register.

WtMel – this field contains name of announcement played to caller during waiting in queue, if routing to this queue is caused via appropriate entry in incoming calls table.

RngMel - this field contains name of announcement played to caller during up-ringing of target extension number.

HldMel - this field contains name of announcement played to caller during holding.

4.9.3 Infolines (Incoming calls/Infolines)

Infolines in Slican PBXes are simple IVR solution, which enable caller to select route using DTMF dialing, and in this way obtain next information or connect to subscriber or group. Infolines can be merged in sophisticated announcement system.

Functionality
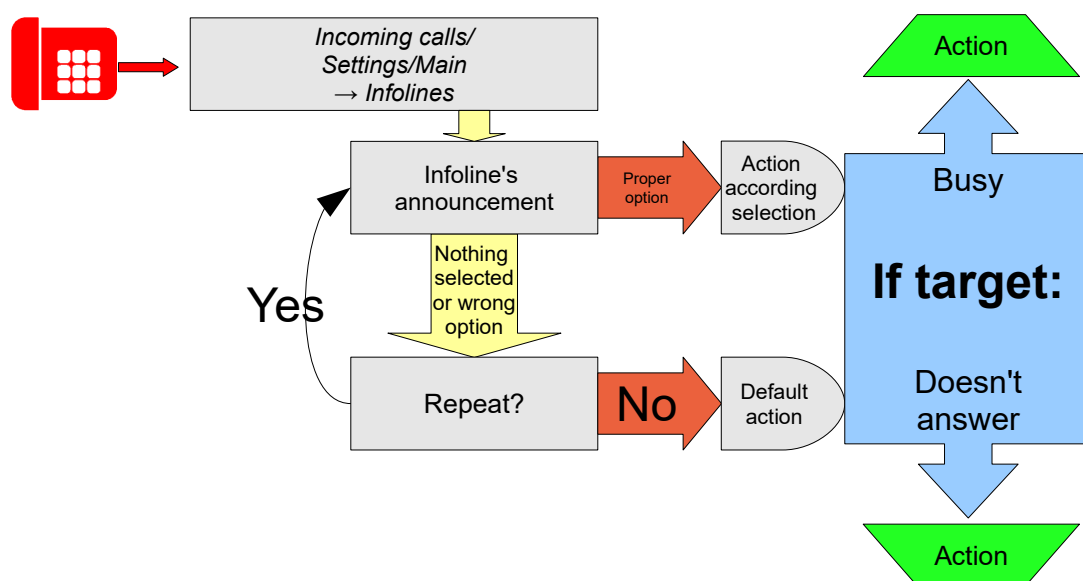
1. Incoming call is routed to infoline.
2. Caller hears voice announcement defined in sheet ***Incoming calls/Infolines***, in field

Ann.

3. During this announcement and during time defined in sheet ***Incoming calls/Infolines***, in field **WtTi** after its playing caller can select next step using phone keypad. To define action digits 1 to 9, 0 and signs * and * are used – action is defined in sheet ***Incoming calls/Infolines***, in field **DgtAct**. If caller doesn't make any selection action defined in sheet ***Incoming calls/Infolines***, in field **DefAct**. After checking field **ExtNo** in sheet ***Incoming calls/Infolines***, extension number can be dialed during playing announcement.

4. If nothing is selected or selection is wrong, it can be defined how many times announcement ***Ann*** is repeated, it depends on settings in field ***RptTi***.

5. If Subscriber or Group is dialed after perform appropriate action, You can decide what should be done in two different situations:
   - target is <u>busy</u> – action defined in field **BsyAct** is performed,
   - target <u>doesn't answer</u> – action defined in field **AnsAct**.

6. If caller connected to Infoline (during playing announcement defined in field ***Ann*** and during time defined in field   ***WtTi*** after this announcement) doesn't perform any action, and the same takes place as many time of playing announcement as defined in field ***RptTi***, action defined in sheet ***Incoming calls/Infolines***, in field **DefAct** is performed.

Attention! Infolines are subject of licensing. By default – without license 1 Infoline is available.

Attention! If in field **AnsAct** there is an extension number of subscriber belonging to group accessible from infoline, this subscriber is not up-ringed.

Infoline parameters and limitations

Maximal numbers of infolines in system: 99.
Maximal time of connecting to Infoline for one call is equal to <u>15 minutes.</u>
Maximal number of calls which are connected to Infolines in the same moment are equal to:
   - for Slican MAC-6400 – 10*number of shelves
   - for Slican CCT-1668 – 4
   - for Slican CXS-0424 – 10

If maximal number of calls is achieved, next calls wait for PBX resources release (Caller hears ring-back signal until resources released).
Infoline is busy starting from begin of announcement playing until caller establish connection with PBX extension number or enter the group queue.
Resource assignment can be done in sheet ***Incoming calls***, field **InfChn**.

4.9.4 API Infolines

API Infolines it is a mechanism a little bit more sophisticated, than Infolines described in previous chapter. Configuration is very similar to standard infoline, there are some more configuration fields, ie. **SrvAdr** which should contain server address together with name of script which analyze data entered by caller and according these data manages the call. More detailed description of Infoline API functionality and example of PHP script structure can be found after registration on http://sdk.slican.pl

Attention! **API Infolines are subject of licensing as standard Infolines.**

4.9.5 Number presentation for incoming calls – only ISDN

Obtained number presentation (COLP)

Target subscriber can itself present to caller (from system phone or ISDN phone) depending on settings.

Can presents itself by:

- own extension number,
- according settings for outgoing calls (field **CLIP** sheet **Subscribers/Outgoing calls**),
- trunks group own number,
- lock presentation for incoming calls.

Setting **COLR** field in sheet **Subscribers/Incoming calls** decides about extension number presentation for caller. We can select one from four options:

- presentation with number dialed by caller
- presentation according **CLIP** field settings
- own number presentation
- no presentation – lock presentation of obtained number

4.9.5.1 Presentation of received number (doesn't concerns system phones)

For subscribers equipped with analogue phones (with Clip-FSK) or VoIP phones (softphones lub hardware phones except CTS-202.IP) it is possible to set caller presentation mode – it can be done in sheet **Subscribers /Incoming calls**, field **CLIPRcv**.
Following options are available:

- **OFF** – number presentation is switched off;
- **CLIP** – caller presentation is sended as number and assigned comment;
- **CLIP (without comment)** – only number presentation. This option is useful when connected phone improperly interprets full CLIP (some old phone models with CLIP function);
- **Dialed number (without comment)** – instead of caller presentation number dialed by caller without comment is sended. This option can be used for phones where calls from many lines are received and it is important which line is calling.

ATTENTION! *For some phone models CLIP presentation is not received when double ringing is set. In this situation option* **RegRng** *in sheet* **Subsribers/Special services**. *This option concerns only analogue ports*.

4.9.5.2 Presentation of area names

In sheet **Phone books/Area names** you can define comments/area descriptions which depend on **prefix** of the caller presentation number. This comment is displayed on system phones, analogue phones with CLIP-FSK and for VoIP users.

Another using method of Area Names sheet:
If Company X uses numbering range ex. 22 333 33 xx, (and appropriate entry is located in the table) then any call incoming from number wit above mentioned prefix is presented as Company X.

4.9.5.3 Phonebook modes

To clarify and organize redial from call history and PBX phonebook in case of different outgoing methods, in 6.30, two phonebook modes was implemented:

- regional number phonebook

- unknown number phonebook

Choosing possible phonebook mode is possible in **Global settings/Remain** field **NoTyp.**

Operating principle:

Work mode: "Regional number phonebook"

Designed for PBX's working in **Hotline or with one additional outgoing number:**

- **PBX phonebook –** numbers are stored without additional outgoing number or in International form eg. +48523251100
- **Outgoing traffic –** number from PBX phonebook are dialed directly from LCR.
- **Incoming traffic –** if PBX is working with „additional outgoing number" and in Incoming calls/Presentation/CLIP categories this additional number will be inserted, then this number will appear on analogue, ISDN and VoIP phones. Number will not be added in PhoneCTI, ConsoleCTI, RecordMAN, BillingMAN, XML protocol anf call history.
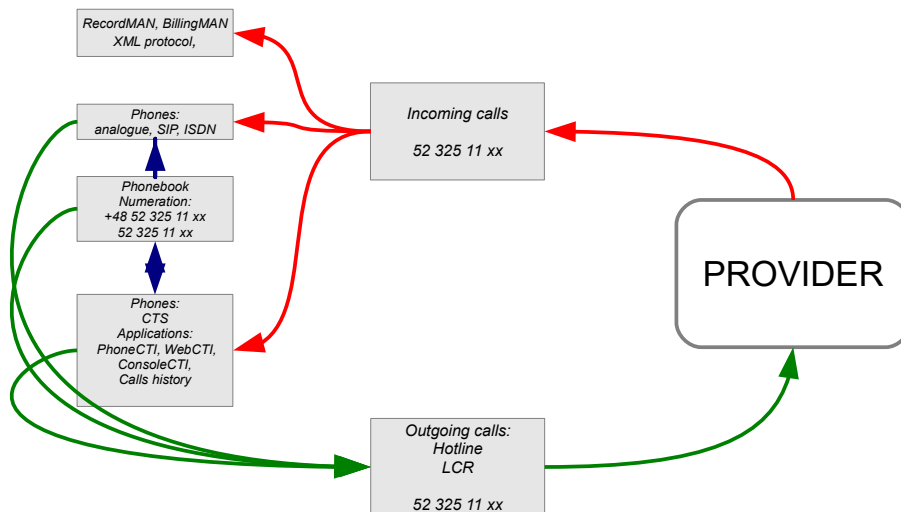
Figure 4.5: Regional number phonebook - Hotline
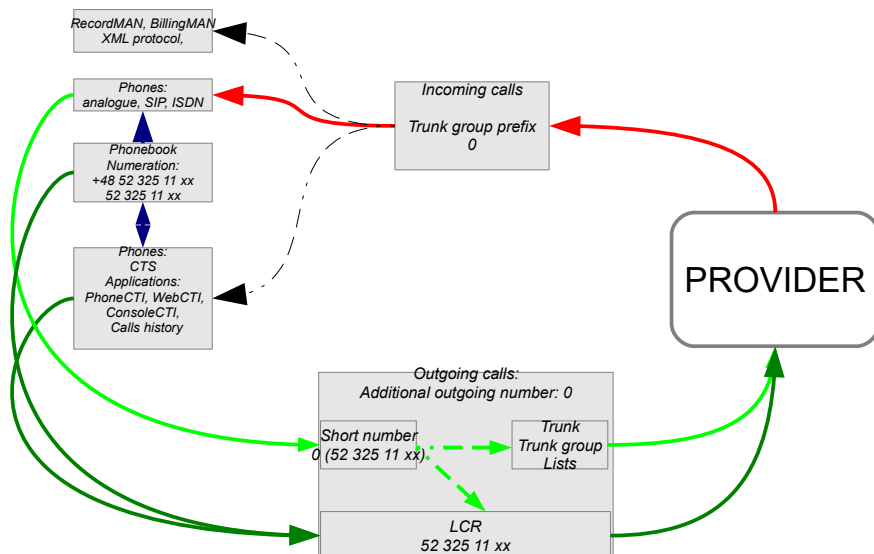
Figure 4.6: Regional number phonebook - additional outgoing number

94

Work mode: "Unknown number phonebook"
Designed for PBX's wit **one or more additional outgoing numbers"**

- **PBX phonebook –** records with „additional outgoing number" (set as added number to CLIP in Incoming call/Number presentation/Trunk group prefix) or as +48523251100
- **Outgoing traffic –** number from PBX phonebook in form +48523251100 are dialed directly from LCR. If PBX is configured to use different trunks with different short numbers, when calling using history or Redial, added number „0" (Incoming call/Number presentation/Trunk group prefix) will be added to outgoing call.
- **Incoming traffic -** if in Incoming call/Number presentation/Trunk group prefix a number is added to trunks, incoming call will be displayed with this number. This number will be added to all kind of phones, call history and applications.
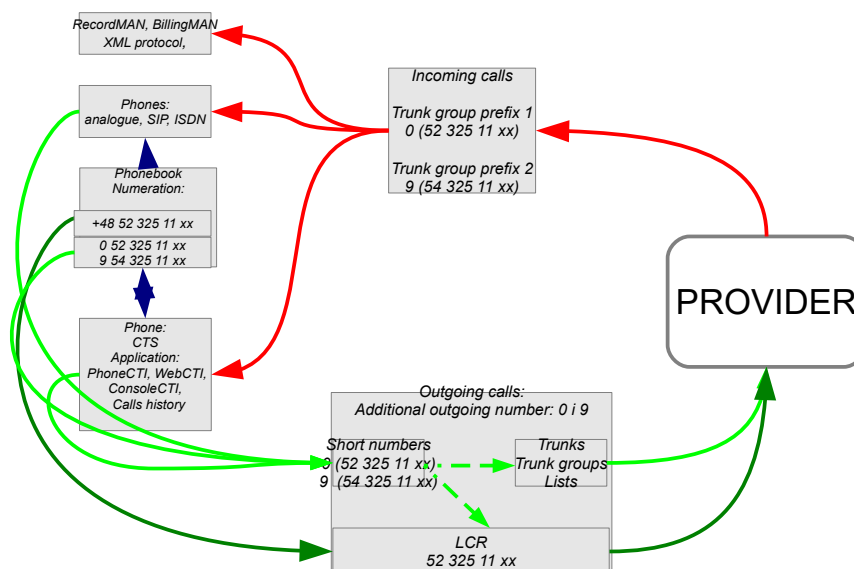
Figure 4.7: Unknown number phonebook - additional outgoing number

4.9.6 PBX working mode

There are 8 working modes in PBX (***Time/PABX working modes –*** mode's names can be edited). PBX can works only in one mode at the same time. When each working mode is obligatory is decided by entries in sheet ***Time/Working modes time-table***. Additionally we are able to managing working modes.

Managing working mode can be defined in sheet ***Time***, field **MngMd.** There are following possibilities:

- *Automatic* – PBX switch on and off working modes according ***Time/Working modes time-table***
- *Temporary manual* – In PBX we can force any working mode defining how long this working mode must be obligatory. After this time PBX returns to automatic mode.
- *Manual* – PABX will work in manually specified working mode until change defined in ***Time/Working mode time-table***. Additionally it is possible to switch off any automatic I field **NoAut** – after checking this field working mode change will be possible only manually.

Working mode time table (Time/Working mode schedule)

Entries in this table define time of changing PBX working mode – it is one week calendar and holidays. Working mode changing time is set in field **MdTi**.
Field **Cpt** defines text of caption, which will be displayed on system phone displays.
Records can not be repeated.

Holidays (Time/Holidays)

Holidays table enables to freely defining free days and entering PBX working mode assigned to these days – it is so called eight day of week ("Holiday"). Defined holidays can be movable or repeat every year in the same day.

To add "Holiday" working mode managing, separate entry should be added in sheet *Time/Working modes schedule* and in field **WDay** of this entry option *Holiday* should be selected. Holiday working mode is valid whole day and night.

Local, manual working mode changing (service *87)

Manual working mode changing can be done using service *87. Extension number used to switching manual working mode changing should be authorized with appropriate privileges (*Global Settings/ PABX access accounts* field **Sw**).
Working mode changing using analogue phone:
*87 PIN Working mode no. (1…8) Hour (HHMM) – setting validation of this mode until specified hour – after this hour PBX returns to automatic mode changing according working modes time table (*Manual, temporary mode*).
*87 PIN Working mode no. (1…8) * - setting validation of this mode until next manual change (*Manual mode*).
*87 * - setting automating working mode changing.
*87 * - checking if manual changing mode is switched on.
*87 PIN Working mode no. (1…8) 9999 – setting specified working mode without automatic change. Change takes place after next manual working mode changing.

Captions on system phones and working mode

Captions displayed on system phone depend on PBX working mode:

- automatic working mode – comment included in sheet *Time/Working modes time table* – field **Cpt**,
- manual working mode – comment in sheet *Time/Working modes time table*– field **Cmt**.

Emergency working mode while lack of 230V~ power supply

When lack power supply (230V) occurs, PBX can automatically change working mode to emergency working mode, which can be defined in sheet *Incoming calls* - field **EmMd**.
Power supply failure is signals by flashing **POWER** LED (flashing period 0,25s) and audio beep once a minute.

Attention! *Emergency working mode can be activated only if PBX working in automatic working mode switching.*
If PBX is working in emergency mode (no 230V), switching to manual changing mode is possible. Quitting manual switching mode and enter in automatic switching mode doesn't force changing to emergency working mode even if there is no 230V power supply.

Extension number privileges depending on working mode

There are 2 possible methods of limiting privileges of extension numbers depending on working mode:

1. Phone lock: For which working mode phone lock should be active can be set in sheet *Time/Working mode* field **PhLck**. Additionally extension number for which above mentioned lock should be active can be selected in sheet *Subscribers/Outgoing calls* field **PhLck**.
    For this method of locking phones, subscriber can manually unlock phone using service ***20PIN** (this service is not active if option *Only automatic* is set in field **PhLck**, in sheet *Subscribers/Outgoing calls*) **.**

2. Downgrading privileges and access classes. After checking field **NgtMd** in sheet *Time/Working mode* in specified working mode night access classes and night privileges will be valid – they are defined in sheet *Subscribers/Outgoing calls,* in fields **NgtCls** and **NgtPrg**. This settings is automatic and controlled only by working modes.

4.9.7 Call forwarding to operator (orphan calls)

In sheet *Incoming calls*, field **OphTi**, there is a time defined, after this time unsnsered call will be assumed as orphan call and forwarded to operator (global settings).

Following definitions of operator are possible:

1. None: service is deactivated,

2. Subscriber: if incoming call is not answered, call is forwarded to subscriber with defined extension number (including group mailbox),

3. Group: if incoming call is not answered, call is forwarded to group with defined extension number,

4. Announcement: if incoming call is not answered, defined announcement is played, after playing announcement call is disconnected.

Attention! *This service concerns incoming calls routed to individual extension number – not a group.*

4.9.8 Remote services authorization

Setting of field **AccTrn** in sheet *Incoming calls* enables or not remote services activation during DISA/Infoline announcement.
Following options are possible:

–   No limits – account can be recalled/working mode changed without any limitation during DISA announcement,
–   Only with CLIP – account can be recalled/working mode changed during DISA announcement only if caller's CLIP isn't switched off (CLIR service isn't activated),
–   Not allowed – remote services can't be realized.

4.9.9 Global settings for incoming calls

Settings in field **KnPer** in sheet *Incoming calls* enables to define knocking signal period. This signal informs subscriber about waiting calls (on system phones display message „Waiting.." is presented).
Knocking signal can be hear for incoming calls in following situations:
  • call is routed to the group

- call is forwarded without <u>announcing</u>

4.9.10 Melody on Hold (MoH) options

4.9.10.1 Melody on hold selection

This settings is done in field **MelTyp**, sheet *Incoming calls*. Following options are available:
- standard,

- recorded announcement,

- external source (ex. Audio device MAB-1101).

4.9.10.2 Announcement during melody on hold - „Please wait for the connection"

During playing melody on hold, every 30 sec. there is message „Please wait for the connection" played.

If playing this message is not necessarily (ex. Because melody on hold includes similarly text) it can be switched off by checking field **VceDis** in sheet *Incoming calls*.

4.9.10.3 Melody on hold instead of ringing signal

If it is necessarily to „cover" ringing signal by melody on hold, during waiting for establishing call, field **MelRng** in sheet *Incoming calls* should be checked.

4.9.11 Voice announcements

4.9.11.1 Introduction

Information about announcement memory capacity and announcements duration can be found in sheet *Incoming calls/Announcements*.
1. Announcements *1...9 type DND/DISA/Infolinie* can be recorded from phone or using ConfigMAN,
2. Remain announcements <u>10...99</u> can be recorded <u>only</u> using ConfigMAN (including ConfigMAN.user).

3. <u>Private DND/Voice mail message</u> can be recorded from phone connected to authorized extension number or using ConfigMAN.

4.9.11.2 Recording announcement DND/DISA/INFOLINE from phone (*92)

Announcement in PBX can be recorded only by user, who is authorized to do this with appropriate PIN code (*Global settings/PABX access accounts*, field *Ann*). Voice announcement text can be recorded from any phone (it is recommended to use system phone, including system IP phones).
> *Recording from phone (*92) is described in instruction manual*

4.9.11.3 Private DND/Voice mail message (*79)

PBX subscribers – if they are authorized to do this (sheet ***Subscribers/Services*** settings, field **DNDMsg***) –* record and set private DND messages using service ***79**. Much more about recording private DND/Voice mail messages you can find in instruction manual.

4.9.11.4 Announcements informing about position in queue

To use queue announcements functionality it is necessarily to upload to the PBX appropriate voice announcements. After marking check-box in field **AnnQ** message about possibility to create default announcements. After confirmation 39 default announcement will be uploaded, including:

- 30 – about location in queue from 1 to 30
- 1 – information about exceeding maximal declared number of calls waiting in queue
- 7 – about waiting time 1, 2, 3, 5, 10, 15 or 20 minutes
- 1 – information about exceeding maximal declared waiting time

After executing mentioned above activities, upload PBX with database once more

> ***ATTENTION!***
> **All announcements can be replaced with individual by import from file *.wav or recording new from inside ConfigMAN application. It is not necessarily to upload PBX with all announcements, it depends on PBX configuration.**

4.9.11.5 Recording voice announcement from inside ConfigMAN application

Organizing and recording voice announcement can be done in sheet ***Incoming calls/Announcements***.

To realize announcement in PBX you can do one o the following:

- *import* previously prepared **.wav type file (button **Import**);
- *record* an announcement using embedded sound recorder (button **Record**, signal source should be selected, also **Signal level** should be set to include between green lines);
- *buy* recordings from voice synthesizer (button **Buy –** next actions according information in browser) and import recordings.
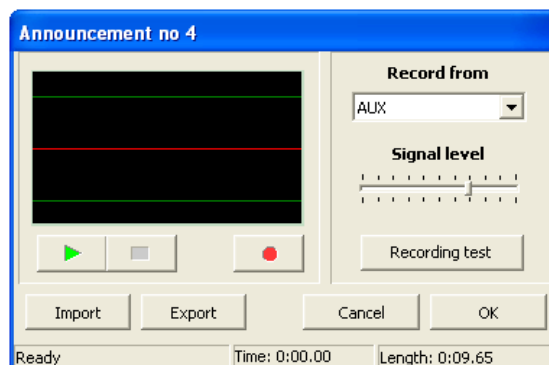
Fig. 4.32: Embedded sound recorder

Notes concerning preparing voice files in external applications:

- sampling frequency **8kHz** and sample resolution **16 bits.**,
- if recording voice and music, voice cannot be much louder than music.

Files not fulfilling these requirements may loss quality after uploading to PBX.

It is recommended to precisely describe title of announcement in sheet ***Incoming calls/Announcements***, field **Cmt** (into this field can be entered up to 32 signs).

Details concerning memory occupation and recordings length can be found in legend displayed on the right side of ***Incoming calls/Announcements*** window**.**

Status:
- „empty" means, that announcement is saved in PABX but there is no downloaded on computer disc (for this announcement content of field **Ti** is greater than 0 sec).
- „new" announcement is recorded on computer but there is no uploaded to PBX (new document icon).
- „v" - announcement if it is identical in PBX and PC and can be listen on PC.

*IMPORTANT***:**

1. After recording or import an announcement it is recommended to switch on „automatic volume adjustment" in sound recorder window.

*2. Announcements assigned with status „new (to send)" can be send only together with PABX configuration (**F7**).*

3. Voice announcements are not downloaded automatically together with PBX configuration. Announcements registration can be done using option PABX → Perform backup.

4.9.11.6 Replacement of system announcements

W firmware version 6.20 possibility of defining system announcements was introduced. This functionality can be usef for instance by companies, where workers and customers using different languages. Due to this functionality it is possible to define system announcement sets in different languages. So using one PBX, every person using this PBX will be hear voice messages in proper language.

By default in Slican PBXes there is one set of system announcement. After uploading to the PBX individual announcement set, there is possibility to create up to three additional announcement sets. They can be used during PBX operation, there is possibility to assign every announcement set to subscriber or doorphone, as well as for every numbering area, where call come from.

System announcements can be upload to the PBX in two different way:
- ➢ if they are stored in .wav files, they should be uploaded using sheet *Incoming calls/Announcements* button *Import*
- ➢ they can be record using phone with service ***92**.

After uploading new announcements to PBX, it is possible to define three new announcement sets. It can be done in sheet *Global settings/System announcements/Sets*, in fields:
- ➢ **SysAnn –** standard system announcement text, this field is not editable.

- ➢ **Set1**/**Set2**/**Set3 –** in these fields it is possible to select announcement, which in specified set correspond to standard announcement with text visible in field **SysAnn**.

After defining announcement sets it is possible to define which set will be used:
- ➢ globally in PBX – in sheet *Global settings/System announcements*
- ➢ individually – in appropriate sheets in field **AnnSet:**
  - – for specified subscriber – *Subscribers/Special settings*
  - – for specified doorphone - *Peripherals/Doorphones*
  - – for specified numbering area – *Incoming calls/Area names*

4.9.12 Voice mail

4.9.12.1 Introduction

There are two types of voice mail:

- *individual* voice mail – assigned to specified extension number and managed by subscriber with this extension number,
- *group* voice mail – can be assigned to few extension numbers, so messages for these extension numbers are stored in collective file, which can be managed by any PBX subscriber.

Maximal capacity of voice mail box is limited by capacity of memory assigned to voice announcements and one duration of one left voice message is defined by content of field **MaxCap**.

Summary time, which can be used for storing voice mail messages:

- 60 minutes for CCT-1668 and CXS-0424
- 120 minutes for MAC-6400.

This memory is used for voice mail messages and for announcements (DISA/Info-lines+DND messages).

4.9.12.2 Individual voice mail

To enable extension number to use voice mail, appropriate memory space should be reserved for voice mail messages for this extension number - field **MaxCap** – *Voice mail capacity* in sheet **Subscribers / Voicemail**.

In the same sheet you can find field **MsgCt** – *Voice mail messages count* – this field shows number of voice mail messages stored for specified subscriber while downloading PBX configuration.

Default settings of voice mail box number is <u>1000</u> and to change it, content of field **ExtNo** in sheet **Global settings/Voicemail** should be changed.

If access to private messages should be protected, field **EnPIN** – *Enable voice mail logging by PIN* in this sheet should be checked – after dialing voice mail box number, entering subscriber PIN (defined in field **PIN** – *PIN-Code* in sheet **Subscribers / Services**).

Incoming call is routed to specified subscriber voice mail box, if forwarding to previously defined voice mail number is active for this subscriber. This forwarding can be set by PBX admin (see chapter <u>4.3.1.3Call forwarding</u>) or by subscriber, if subscriber is authorized to forwarding calls (unchecked field **FwdLck** – *Forwarding lock* in sheet **Subscribers / Services settings**).

An announcement played to caller and inform about recording start can be standard or self recorded. About new message in voice mail box subscriber is informed automatically. Announcing method about voice mail message on analogue phone can be defined by status of field **FSK** in sheet ***Subscribers/Voicemail***.

- FSK unchecked – system realize an announcement by calling back to extension number and presenting itself with number 1000 and comment Voice mail.
- FSK checked – system realize an announcement by switching on new message LED or displaying message "New message" (if analogue phone can manage such functionality ex. Slican XL-2023ID)

Detailed information concerning announcing can be found in PBX Instruction manual.

Using ❌ icon, located near **MsgCt** field it is possible to delete stored messages.

ATTENTION!

While erasing all messages (field **MsgCt**) you should remember, that after clicking icon messages are not erased from memory even if you are connected to the PBX. It is required to upload configuration tom PBX.

4.9.12.3 Group voice mail (virtual voice mail box)

To create group voice mail functionality, voice mail box should be created in sheet **Subscribers/Main settings** ➔ **New** ➔ **Type**=**Voicemail account**, and next assign appropriate space for storing voice messages – sheet **Subscribers*/Voicemail***, field **MaxCap**.

Routing call to group voice mail box is possible in following situations:

- call forwarding to voice mail box is active,
- appropriate entry is in incoming calls table,
- appropriate entry is in static path routing table,
- appropriate routing is made in infoline.

Group voice mail can be defined as:
- backup subscriber,
- orphan calls target,
- voice mail box is agent of up-ringing or linear group.

Welcome message of voice mail recording can be set similarly as for individual voice mail box. Subscriber pointed in field **IT** – *I am there* - sheet **Subscribers/Call forwarding** (the same can be done with service - *57 with group voice mail box number.

If access to messages recorded in group voice mail box should be protected, field **PIN** – *PIN-code* in sheet **Subscribers/Services** should be fulfilled. Then any subscriber after dialing group voice mail box number is asked about PIN code to managing messages stored in voice mail box. Detailed information concerning announcing new voice mail messages and listening stored messages can be found in PBX „Instruction manual".

4.10 Monitored calls

Slican has implemented a new tool for monitoring aquality of work ofgents (employees). It allows us to listen to selected calls or providing consultation (support) during a call. This functionality is available for subscribers privileged by PBX administrator, set for defined numbers or call type.

If a call, meeting the conditions of a monitored call is made, PBX is automatically connecting with defined earlier internal or external number, allowing for listening or consultation.
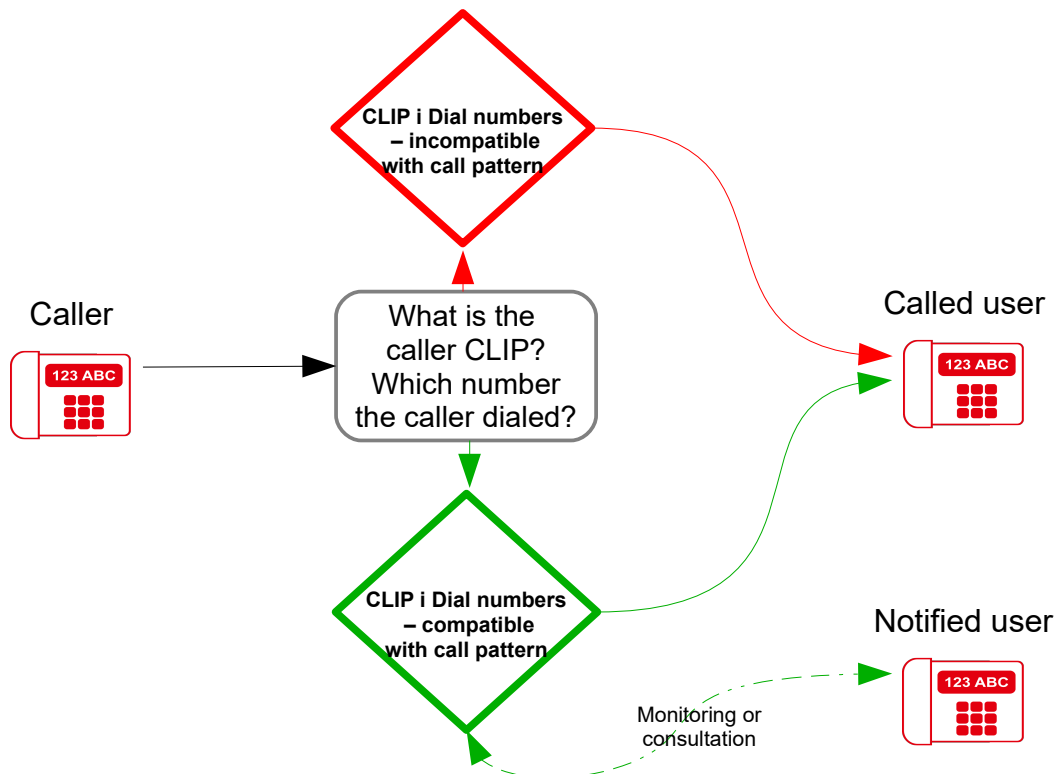
Figure 4-9.1: Monitored call - call process

To add new monitored call, in table **Monitored calls/Settings** create new record. Using the input data, we determine which calls will be monitored.
Available settings:
CnnTyp – choosing available monitored call type (internal, incoming, outgoing)
CLIP – number, that the caller is identified
DINo – number dialed by the caller
NtfNo – number notified about monitored call
ChrAcc – number that is a payer for calls
3PTY – three way conversation instead of listened call

Depending on the type of call ***CnnTyp*** fields ***CLIP*** and ***DINo*** need proper internal or external numbers typed:

   1. *Internal calls:*
CLIP – caller subscriber number; DINo – called subscriber number
   2. ***Outgoing external calls***
CLIP – caller subscriber number; DINo – external number (or part of it, eg. prefix)
   3. ***Incoming external calls***
CLIP – external number or part of it (international format eg. 48xxxxxxxxx); DINo – internal number, to whicha incoming call id directed (subscriber, group)

If CLIP and DINo of a call matches typed fields, the PBX will make a additional call to NtfNo, who can listen to or participate in conversation (depending on 3PTY).

When a call is made, number of caller will be displayed on notified phone.

5 **Embedded recording**

Embedded recording enables to incoming, outgoing and internal calls recording.
For CCT-1668 before inserting storage media in appropriate slot on recording card, updating of PBX software should be <u>obviously</u> performed, following procedures should be done:
1. <u>Controller firmware updating</u> (to version 3.05 or later).
2. <u>VoIP card firmware updating</u> (to version 1.21 or later).

5.1 Storage media

Calls are recorded as files in *.wav format according mentioned below standards on following storage media:

Approximate times of recordings presents following table:

Storage media capacity CF/HDD [GB]	CCITT G.711 A-law 64kb/s, 8kHz, mono [hours]	GSM 6.10 13kb/s, 8kHz, mono [hours]
1	34	160
4	130	630
40	1300	6300
100	3300	15000
160	5300	25000

Table 5.1.: Approximate times of recordings for different storage media

PBX	Storage media min 512MB
CXS-0424	SD
CCT-1668 firmware <5.11	CF
CCT-1668 firmware >=5.11	SD or HDD
MAC-6400	HDD
IPL-256	SD or HDD
IPM-032	SD
IPU-14, IPS-08	microSD

Table 5.2.: Storage media type for specified PBX

Due to data security and recording efficiency in recording module special file system is used. To create this file system on storage media installed in PBX option **EbdRec memory format...** in menu **PABX** should be selected or in system console command **rec format** should be executed.
After selecting mentioned above option, window appears on display and request of formatting storage media should be confirmed. For formatting from inside system console all information will be visible also in console window. PBX informs about finishing formatting and include appropriate event in PBX monitor.

ATTENTION! *Formatting process can continue through several minutes. During this time recording and access to recorded calls is unavailable.*

5.2 Recording calls configuration

Call recording channels are subject of license (see chapter concerning licensing). Without license only demo version is active, it means that only 1 recording channel is available and only 60 seconds of each call can be recorded (after this time call recording stops but call is still established).

It possible to record following calls:

- outgoing calls established via specified trunk group,
- calls incoming via specified trunk group,
- established by specified extension number (external and internal calls),
- calls with doorphone.

5.2.1 Recording calls established via specified trunk group

Settings concerning calls established via specified trunk group can be found in sheet *Trunks groups / Settings*.
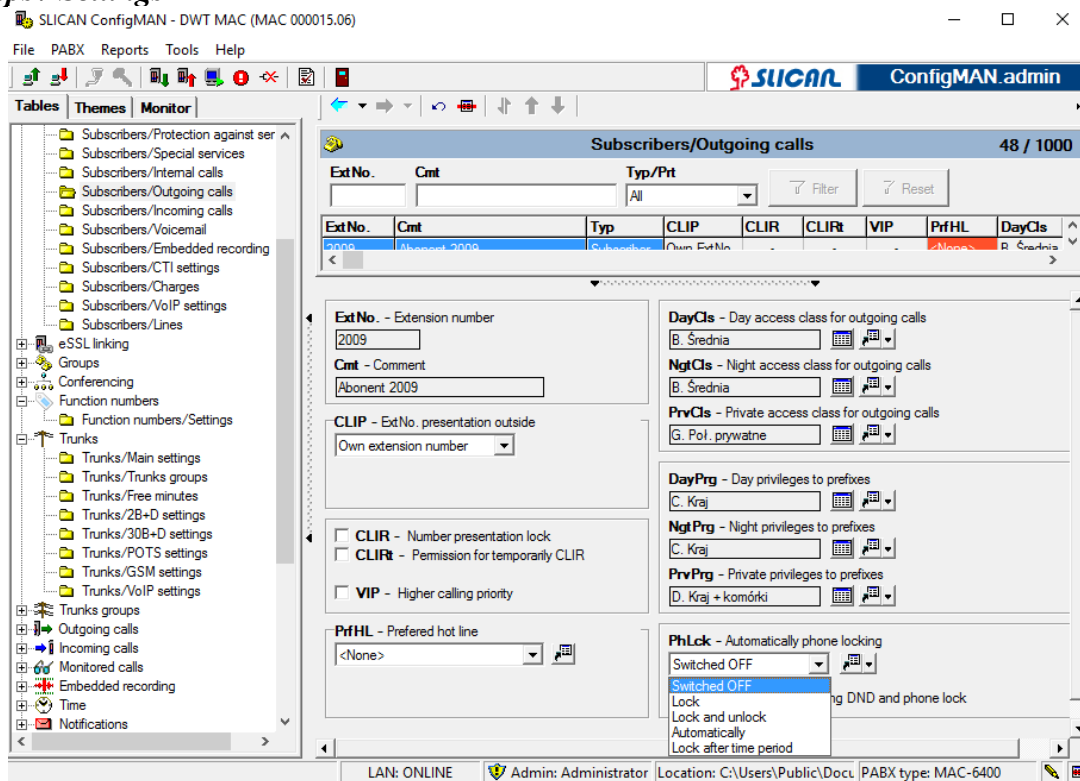
Fig. 5.1.: Call recording settings for trunk group

Following settings are available:

- **RecInc** – following options can be selected in this field:

  - *Switched off* – Calls incoming via this trunk group will not be recorded.

If options *low priority* or *high priority*, than calls incoming via this trunk will be recorded, if extension number where call is routed has no option *Prohibited* set in field **EbdRec** in sheet *Subscribers / Embedded recording*. The difference is as follow:

  - *Low priority* – with this settings it should be remembered, that recording will be interrupted, if all recording channels are busy and new call which should be unconditionally recorded (*high priority* is set for this call recording). Interrupted

recording is marked appropriately in *RecordMAN.client* application. Recording can be resume only if call is transferred to another number and any recording channel is free during call transfer – it means, when new recording file is created.

- *High priority* – recording with such defined priority will be interrupt recordings with option *low priority*, if all recording channels are busy.

– **RecOut** – following options can be selected in this field:

- *Switched off* – Calls outgoing via this trunk group will not be recorded

If options *low priority* or *high priority*, than calls outgoing via this trunk will be recorded, if extension number, which establish outgoing call via this trunk group has no option *Prohibited* set in field **EbdRec** in sheet *Subscribers / Embedded recording*. The difference is as follow:

- *Low priority* – with this settings it should be remembered, that recording will be interrupted, if all recording channels are busy and new call which should be unconditionally recorded (*high priority* is set for this call recording). Interrupted recording is marked appropriately in *RecordMAN.client* application. Recording can be resume only if call is transferred to another number and any recording channel is free during call transfer – it means, when new recording file is created.
- *High priority* – recording with such defined priority will be interrupt recordings with option *low priority*, if all recording channels are busy.

5.2.2 Recording specified subscriber calls

Settings concerning automatic recording calls established with specified extension number can be found in sheet *Subscribers / Embedded recording*.

Following options can be selected for specified subscriber, in field *EbdRec*:
- **Prohibited** – no calls of subscriber with this extension number will be recorded. Example: for extension number A, option *Prohibited* is selected, if call recording is checked in sheet *Trunks groups definition* or for extension number B, which is second part of call, option *All calls* is selected, call will be not recorded. An exception is situation when subscriber A uses call recording service, which is accessible from any *CTS-202* phone. In such situation, subscriber A by pressing key with programmed recording service (programing method is described in system phone manual) can records call.
- **Switched off** – calls of this extension number will not be recorded on its side but will be recorded on opposite side, of course if on opposite side recording is permitted. Example: if extension number A has embedded recording *switched off* and somebody is calling from extension number B, which has option *All calls* set in field **EbdRec** or from outside via trunk group, with **RecInc** checked – call will be recorded. If both call participants have selected option *Switched off*, it is possible to record call „on demand" by pressing speed access key (read above).
- **Only internal** – by this settings only internal calls of this extension number will be recorded.
- **Only external** – by this settings only external calls (incoming and outgoing) of this extension number will be recorded.
- **All calls** – both, internal and external calls of this extension number will be recorded.

Fig. 5.2.: Settings concerning recording calls of specified subscriber

For every subscriber with recorded calls additionally following fields are available:

- **HiRec** – after checking this field call recording for this subscriber is assigned with *higher priority* , it means that if in system are recorded call with lower priority and all recording channels are busy, one call with lower priority is interrupted to record new call of specified subscriber.

- **AccLvl** – in this field belonging to specified access level can be defined, content of this field decides about recording visibility in *RecordMAN.client* application.

5.2.3 Recording calls with doorphone

Also it is possible to record doorphone calls. To do this field **EbdRec** in sheet *Devices / Doorphones* (figure below) should be checked. In this situation if doorphone call is routed to extension number with recording calls prohibited, such call will be not recorded.

Following options are available:

- *Switched off* – calls of this doorphone will not be recorded, if on opposite side recording is permitted.

- *Low priority* – with this settings it should be remembered, that recording will be interrupted, if all recording channels are busy and new call which should be unconditionally recorded (*high priority* is set for this call recording). Interrupted recording is marked appropriately in *RecordMAN.client* application. Recording can be resume only if call is transferred to another number and any recording channel is free during call transfer – it means, when new recording file is created.

- *High priority* – recording with such defined priority will be interrupt recordings with option *low priority*, if all recording channels are busy.

Fig. 5.3.: Settings of doorphone calls recording

After switch on recording, in field **EbdRec** recording access level should be defined (field **AccLvl**), it decides about recording visibility in *RecordMAN.client* application.

5.3 Recording results, files *.wav

Audio files (*.wav), with call data included are stored in daily folders, ex. **2009-05-19**.

> ***ATTENTION AUTOMATIC ERASING!***
> *Call recording card automatically erases oldest folders. If there is only current folder on recording card (folder with current date), sequentially oldest files are erased. It can happens if free memory on CF/SD card is smaller then 10MB.*

Apart from daily folders, there is also TEMP folder on CF card. In this folder stored are files concerning currently established calls (do not finished yet), before moving these files to current day folder after finishing and file *info.txt*. Contents of this field can be as follows:

```
MAC-6400 00314 114464MB HDD 2
```

where:
`MAC-6400` – PBX type (CCT-1668, MAC-6400, CXS-0424),
`00314` – serial number,
`114464MB` – capacity of storage media (always in megabytes),
`HDD` – storage media type (CF, HDD, SD),
`2` – version of file name format *.wav

File name format is identical for both types of file and can be presented as follows:

```
xxxxxxx zz yyyymmdd hhnnss iiiRMNF ClipNo ClipName DialNo DialName
                ColpNo ColpName rr time A_.wav
```

where:

xxxxxxx – call identifier [0000001..9999999],

zz – counter of performed transfers [00..99],

yyyymmdd – call date:

       yyyy - year [ex. 2007],

       mm - month [01..12],

       dd - day [01..31],

hhnnss – time of starting call:

       hh - hour [00..23],

       nn - minute [00..59],

       ss - second [00..59],

iiiRMNF – call type (small letters are always, capitals depends on what happend during
       call):

       e (external call) – external call,

       i (internal call) – internal call,

       N (no recording channels) - no recording channels,

       D (demo version) – only one minute recording (wersja demo),

       B (recording break) – recording interrupted by call with higher priority,

       P (recording prohibited) – recording continuation prohibited
          (call with recorded DISA announcement is routed to subscriber with recording
    prohibited),

       M (malicious call) - malicious call,

       S (manual recording start) - manual recording start,

       E (manual recording stop) - manual recording stop,

       T (transferred call) - transferred call,

       F (forwarded call) - forwarded call,

       R (resume call) - resumed call,

       H (hold call) - hold call,

       C (conference call) - conference call,

       K1÷K999 – adding subscriber to conference with specified identifier,

       K0 – adding subscriber to conference which is not recorded,

       V (VIP type call) - VIP type call,

       O (Voice recording) – Voice mail box recording,

       G (Group) – group call,

       A (Account) – account is recalled by caller or target subscriber,

       I (Intercept) – call pick-up,

       X (VOX - Voice Operated eXchange) – recording using speech detector
          (VOX type audio device),

       Q (Queue) – call established via queue,

       L (Limit time) – recording limit was overflow (call is still going on),

       Y (3PTY) – Three party call,

       Z (call override) – call override,

ClipNo – caller number,

ClipName – caller name (for external numbers, comment from area names table or public
phone book),

DialNo – dialed number,

DialName – name of dialed number,

`ColpNo` – target subscriber name,
`ColpName` – name of target subscriber (or external numbers, comment from area names table or public phone book),
`rr` – ringing time in target subscriber or waiting time for group calling [in seconds],
`time` – call duration [in seconds],
`A_` - access level of caller and target subscriber defined in field **AccLvl** in sheet ***Subscribers / Embedded recording*** [A..H, _ - access level not assigned]

Notices concerning files wav and txt creating:
- If there is no target subscriber (ex. call was finished on infoline or DISA announcement), no one file is created (there was no call). However such call can be recorded, if field **DISARec** in field ***Global settings / Embedded recording*** is checked.
- After transfer of any call, new file *.wav is generated.
- If caller, dialed or target subscriber name there is sign: „ ”(space), *, /, \, ”, ?, :, |, < or >, this sigh is replaced by sign „_”.
- If there is no extension number or comment, sign „_” is entered.
- If number is restricted or there is no number presentation, sign „_” is entered.
- Maximal time of call duration (wave file) is limited to 4 hours.

5.4 *Access to recorded calls – applications and configuration*

In first version of recording system server (storing and serving recordings) and client (recording browser) functions were included in one application - *RecordMAN*. Such solution has many advantages, but also many disadvantages. These disadvantages were specially visible in large telecommunication systems based on servers *MAC-6400*.

As a rule of operating first version of embedded recording system, firstly recording were synchronized, in fact recordings were copied from storage media in PBX (used as buffer) to hard drive of PC where application was launched. In PBX MAC-6400 as storage media is used hard drive with capacity of 120GB (installed on REC card), in this situation synchronization may continue several hours. Also after longer time of not using application mentioned above process continue very long, due to this disadvantage there was no speed access to stored recordings. Increasing number of recordings channels in version 4.02 due to possibility of installing up to 8 REC cards in PBX (MAC-6400), recording storage capacity may be increased up to 960GB. In this situation synchronization time may be very long, additionally total capacity of storage media in PBX may be greater than capacity of archive disc in PC.

Another disadvantage was lack of access levels to recordings. In *RecordMAN* every one, who knew password to this application can access to all recordings. To eliminate above mentioned disadvantages in second version of recording system functions of server (*RecordMAN.server*) and browser (*RecordMAN.client*) were separated.
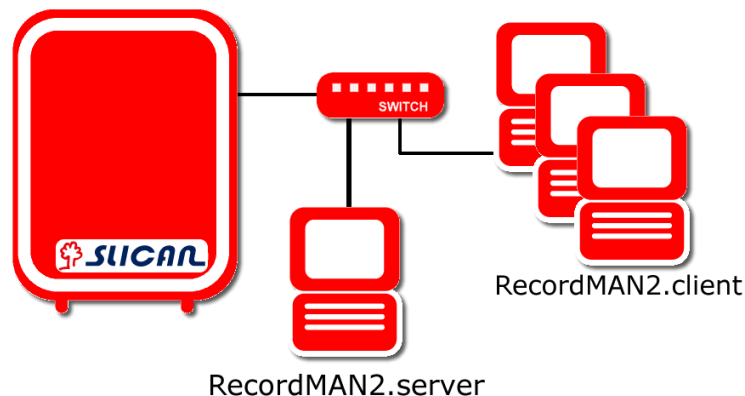
Figure 5.4.: Embedded recordings 2 – dedicated PC as recordings server

5.4.1 RecordMAN.server

Main task of RecordMAN.server application is downloading recordings from storage media installed in PBX and copying they on hard drive of PC in LAN network. It is specially important for *CCT-1668* PBX, where *CompactFlash* card is used as a storage media – this card operates as a buffer, so while large calls number and limited card capacity oldest recording may be erased before copying. For *MAC-6400* PBX main purposes of using RecordMAN.server is storing data on hard drive with larger capacity or/and preparing a backup copies. Beside *RecordMAN.server* can serve recordings (also archive) for *RecordMAN.client* application. While recordings serving user privileges (granted in sheet **Global settings / RecordMAN.client accounts**) are checked, so privileges change results in access to recordings stored in PBX and archive recordings.

ATTENTION! To use RecordMAN.server, appropriate license code should be acquired. This application has no possibility to play recordings.

To enable automatic recordings downloading by *RecordMAN.server*, appropriate settings should be entered to configuration of PBX with embedded recording system, PBX should cooperate with *RecordMAN.server* application via LAN network:

1. Set all parameters concerning with LAN interface of VoIP/Rec card (in CCT-1668) or MACREC (in MAC-6400) in sheet **IP Networks / LAN Interfaces**. About defined IP address user of *RecordMAN.server* application should be informed (for recording system consist of few MACREC cards – address of master card – card with checked **RecMst** field in mentioned above sheet).
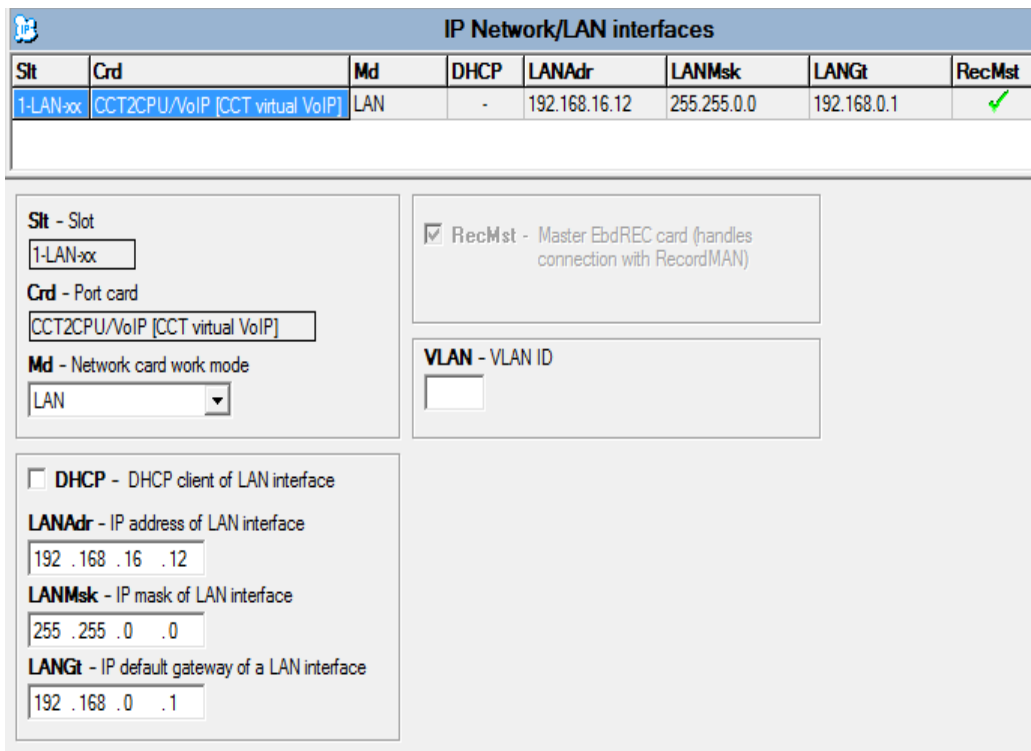
Figure 5.5: LAN interface configuration for cooperation of CCT-1668 with RecordMAN application
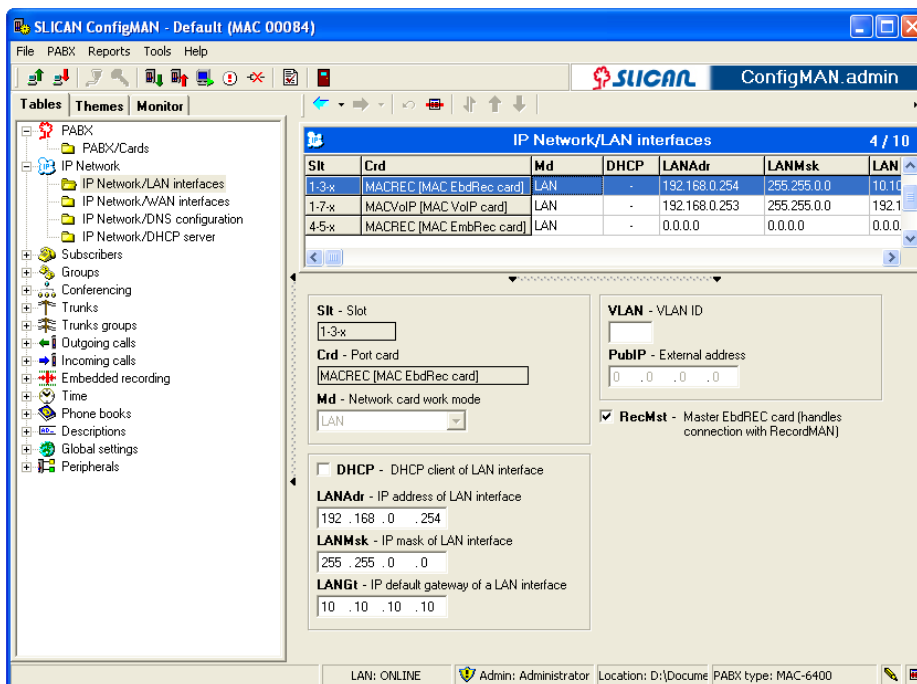
Figure 5.6.: LAN interface configuration for cooperation of MAC-6400 with RecordMAN application

2. Enter license code which include *RecordMAN.server* account, in field **LicEbdRec** in sheet ***Global settings / Licences***.

3. In field **HardKey** in sheet ***Embedded recording*** enter hardware key code, which can be read in appropriate settings tab of *RecordMAN.server* application (if license mentioned *in point 2* is not proper, this field is grayed – there is no possibility to fulfill this field).
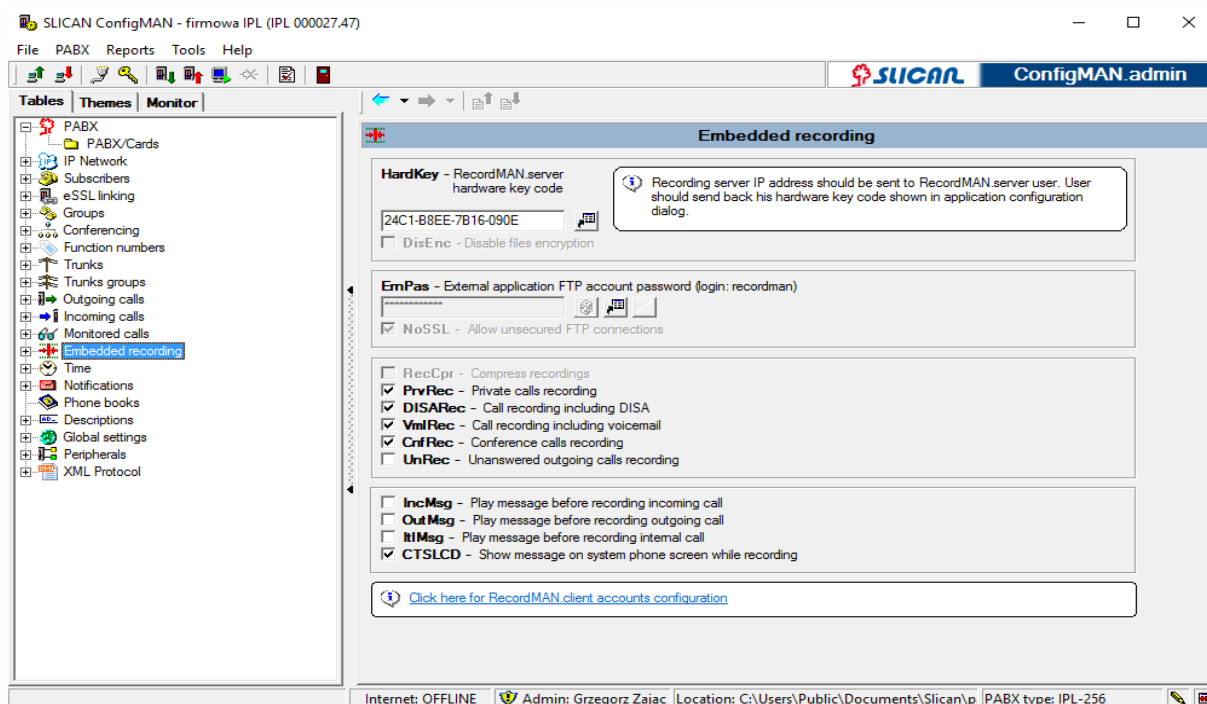
Figure 5.7.: Entering HardKey in sheet Embedded recordings

5.4.2 RecordMAN.client

It is core application for recordings managing. This application doesn't download recordings on local disc (hard drive of computer, where application is launched) after connecting with server (computer, where *RecordMAN.server* is installed or call recording card in PBX), so synchronization of recordings directories is quite quick. Only after ordering to play specified recording appropriate file is downloaded on to the disc and played. It is possible to save such recordings on disc, if user of *RecordMAN.client* is granted, by PBX admin, with appropriate privilege.

*ATTENTION! Due to connection limit defined for PBX (FTP server) simultaneously up to 10 clients[4] (10 PCs with **RecordMAN.client** application, working in LAN network), but application can be installed on more workstations, dependently on acquired license code (maximally 30).*

To enable using *RecordMAN.client* application, PBX with embedded recording system should be configured to cooperation with application via LAN network:

1. Set all parameters concerning with LAN interface of VoIP/Rec card (in CCT-1668) or MACREC (in MAC-6400) in sheet ***IP Networks / LAN Interfaces***. About defined IP address user of *RecordMAN.client* application should be informed (for recording system consist of few MACREC cards – address of master card – card with checked **RecMst** field in mentioned above sheet).
2. If application will be used on more than one computer, license code for required number of *RecordMAN.client* accounts should be entered in field **LicEbdRec** in sheet ***Global settings / Licences***.
3. Create and configure accounts for access to recordings – it can be done in sheet ***Gl***obalne settings / RecordMAN.client accounts. Bellow, you can find all parameters which can be defined for every account.

Available settings are:

- **Log** – used to authorization of *RecordMAN.client* application user.
- **Pss** - used to authorization of any FTP client software, different than *RecondMAN*2
- **HardKey –** used also for authorization of *RecordMAN.client* application user. This code is unique for every computer, where *RecordMAN.client* application is installed.
- **Cpy** – after set check-mark in this field, user identified with specified for this account login and hard key will be able to copy *.wav file on local disc from *RecordMAN.client* application.
- **AccTyp** – in this field you can define, for specified account, user privileges to play recordings. Following options are available:
  - *No access* – for specified user it is no possible to play any recordings,
  - *Restricted access* – user can play only recordings which meet requirements defined in fields **AccLvl**, **FtrStr** and **AccTi**,
  - *Full access* – user can play any stored recording.
- **AccLvl** – recordings can be assigned to specified access level, ex. In sheet ***Subscribers / Embedded recording*** or ***Peripherial / Doorphones***. In this field you define what access levels are available for specified user.
- **FtrStr** – this field enable to set for any account privileges to play/store calls of specified extension number. Filter parameters can be defined by prefix, ex. For numbering range from *100* to *400*:

  – after entering *2,* user will be able to play/store recordings concerned with all subscribers with extension number starting from digit 2,
  – after entering *26* user will be able to play/store recordings concerned with all subscribers with extension numbers from *260* to *269*.

Filtering recordings according subscriber comment is also possible.

- **AccTi** – this field enables to limit access to play/store recordings only from specified number of days back. In this field following options are possible:
  - *No restrictions* – user is able to play/store recordings independently on day of recording creation,
  - *Days back since today* – by this option number of days back should be entered. If entered number is equal to 30, only recordings created 30 days back from today and newer are available.

5.4.3 FTP/SFTP client

Access to recordings is also possible using any FTP client (ex. Embedded in *Total Commander* application) or playing recordings system developed by another manufacturer. In this situation recordings are protected by fixed login *recordman* (can't be changed) and FTP server password, this server is PBX with embedded recording system. Password is defined in field **ErnPas** in sheet ***Global settings / Embedded recordings*** should contain at least 4 characters, it could be entered manually or generated automatically using icon 🎲. Access to recordings from application different than RecordMAN is subject of licensed. Details concerning license code can be found in chapter 16.7.License for embedded recording. If there is no license code, there is no possible to assign login and password and there will be not possible to connect to „FTP" (unchecked **NoSSL**) or to „SFTP" (codded connection).

5.5 *Optimization of system resources needed for call recording*

In PBX monitor statistics are entered, which inform about using recording channels. Purpose of these statistics is answer the question if using system resources (DSP sub modules; 1 SM.DSP = 6 recording channels) and number of acquired license codes for recording channels enable appropriate quality of embedded recording calls system.

ATTENTION! Mentioned above text do not concerns *CXS-0424. For this PBX, after installing LAN or VoIP and, always 8 recording channels are available. Additionally for CCT with FW >=5.11 there is also fixed number of recording channels = 36.*

Following parameters are registered since last PBX reset:
- *Maximal requisition for recording channels* – defines maximal number of needed recording channels,
- *How many times recording channels didn't allocate* – specifies how many times call recording can't be done because of lack of resources.
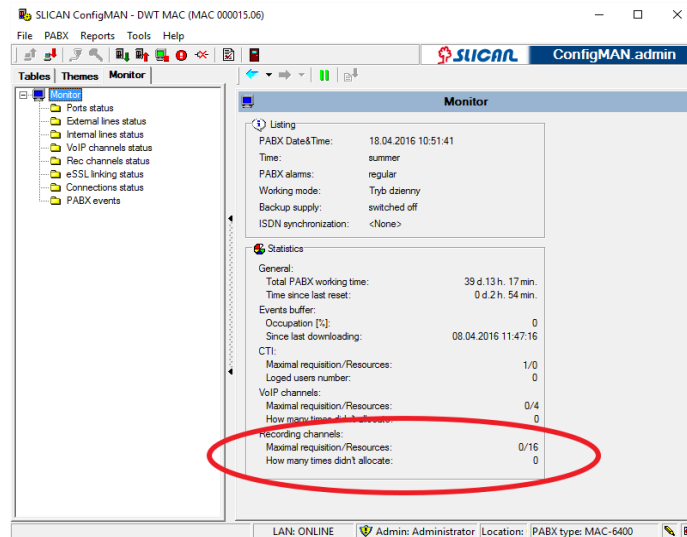
Fig. 5.8.: Statistics in PABX monitor

Number of available recording channels resulting from installed hardware resources can be verified in sheet **PABX/Cards**. After clicking in LAN slot, in bottom part of window field **RecChn** – *Number of available REC channels* will be visible. By this field button **Show** can be clicked. After pressing this button following window is shown:
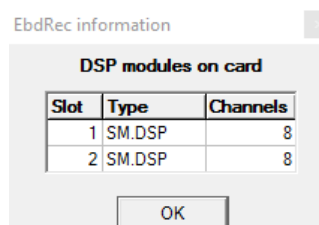
Fig. 5.9.: Info about available REC channel

6 Charges

In Slican PBXes it is possible to define connections charges. It can be realized in   to ways:
– directly from inside *ConfigMAN* with cooperation with additional applications: hotel applications, *PayMAN*
– using specialized charging application - *BillingMAN*

Charging directly from *ConfigMAN* application is used when:
– we using limits for calls from extension numbers,
– we want to display current call cost on CTS system phones display,
in other situations we using other applications – hotel applications, *PayMAN* or we using charging application *BillingMAN*.

6.1 Setting parameters for calls charging

ConfigMAN application is able to charge outgoing calls. If outgoing calls are routed using LCR table it is possible to define charging method according dialed prefix and trunk.

Fig 6.1.: Sheet – Outgoing calls / LCR / Charges

To proper charging, following parameters should be defined in ConfigMAN: number of initial pulses, price for one pulse and charging frequency.
3 types of charges are distinguished:
➢ by period,
➢ by minute price,
➢ by trunks group settings.
In first and second situation charging is performed by settings defined by service. We use charging according trunks group settings if:
➢ calls are routed via **Outgoing calls** table**,** charging is possible only according parameters received from provider

➢ calls are routed via **LCR,** call prices are set according parameters defined in ConfigMAN.

6.2 Setting call limits for subscribers

If from extension number outgoing digit is dialed and than number assigned in **Permited prefixes table** to A privileges (less privileges), such connection there is no charging (field **Blnc** is no incremented) because this connection is treated as internal.

Similarly if from extension number prefix from **LCR** is dialed and than the same number as above (with A privileges defined in **Permited prefixes** table) such connection also **isn't** charged.

Call charge is calculated not often than once per 10 seconds. After exceeding limit A privileges are assigned to extension number.

Fig. 6.2.: Sheet – Subscribers / Charges

Cost limits actualization depending on **LmtTyp** field settings. It is possible to select following options: No limit, No recharging, Every day recharging, Every Monday recharging, Every first day of month recharging or limit in hotel or PayMAN application.

ATTENTION! If subscriber has privileges to using public phone book with active option *Read without privileges check* (field **PubPB** in sheet **Subscribers / Special services**), costs of connection with number from public phone book are not subtracted from call costs balance.

6.3 Event registration options

In sheet *Global settings/Charges* there are several fields, which enable registration of calls in PBX buffer or registration lock.
Slican PBX's allows for recording of events:
- incoming and outgoing external calls
- internal calls
- services

After checking field **EvtRgs** it is possible to register every calls in PBX. Next five fields enable to lock specified types of call. These fields are active only after checking field **EvtRgs**. Checking field **AlrBuf** in-flues to BUFFER LED in CCT-1668, BUF LED in MAC-6400 and STATUS LED in CXS-0424. After checking this field, if buffer is fulfilled in 85%, PBX performs following activities:
- Diode BUFFER/BUF/STATUS – depending on PBX type – start shining with steady light,
- Sends message "BUFFER 85%" to CTS phones – this message will be displayed only on phones which checked field **LCDErr** in sheet *Peripherals/CTS/Settings*.
- Sends SMS with information about buffer fulfilling to number defined in field defined in field **AlrSMS**, in sheet *Global settings/Remain* – conditions of sending such SMS are following: GSM card is installed in PBX and field **AlrSMS** is fulfilled.

If buffer fulfilling continues, if data in buffer will be overwritten, diode BUFFER/BUF/STATUS – depending on PBX type – start blinking. This diode can be switched off by downloading calls with BillingMAN application with selected option Display reset.
If field **AlrBuf** is not checked, alarm about buffer overfilling is not generated. By default this field is not checked.
Next field in this sheet – **VAT** – defines VAT rate added to call cost.

7 Automatic control (Sensor/Relay)

PBX can be used to automatic control. In PBX CXS-0424 to realize this function special card IORS (2 x sensor and 2 x relay) is used. In PBXes MAC-6400 and CCT-1668 cards MAC4RL4SN and CCT4RL4SN (4 x sensor i 4 x relay) can be used. Optionally Slican MSB-1102 Multi Switch Box can be used as interface between PBX and controlled device. This device is connected to PBX subscriber port and can operate as sensor or relay. Realized function depends on jumper settings inside the device (details in MSB Instruction manual) and value inserted in field **PrtCfg, in sheet *PABX / Cards***.
Rest of configuration process should be done in sheet *Peripherials/SensorRelay*
- **ExtNo** – extension number to be dialed to ask sensor status or change relay status
- **Cmt** – comment, which is send to phones with CLIP FSK
- **Trl** – personal notice (displayed only in ConfigMAN)
- **Net** – assigning to subscriber net, used to limit privileges for incoming calls
- **PIN** – security code, which prevents unauthorized access
- **Stu** – switch on/off specified port
- **ShwExt** – after checking this field, extension number of this device is included i system phones extension number list

7.1 Sensor function

This function enables to ask status of sensor connected to PBX subscriber port. Sensor can be in two different status:

- open
  - open loop (CCS mode)
  - no voltage (DCI mode)
- closed
  - open loop (CCS mode)
  - voltage between loop terminals (DCI mode)

Detailed description and connection diagram in every mode can be found in instruction manual.

To ask this status number included in field **ExtNo**, sheet *Peripherials/SensorRelay* should be dialed. If PIN code is entered in configuration, this code should be also dialed during establishing call. After establishing call an announcement assigned in field **OnAnn** or **OffAnn** is played, depending on sensor status (as default message „Service active/inactive"is played). With parameter **NrmSnr** default status of sensor is defined (close/open).

7.2 Relay

Relay status change can be forced with two methods:

- by pulse – for MSB-1102 device pulse duration is equal to 1s, for IORS card this time can be adjusted by parameter **PlsDrn** in sheet *Peripherial/SensorRelay.*
- by voltage level – only for IORS card, relay status is remembered after PBX reset

To trigger relay, call should be established with number entered in field **ExtNo.** In sheet *Peripheral/SensorRelay*. If **PIN** code was entered during configuration, it is necessarily to dial this PIN code during call, next:

- for IORS card (CXS-0424)
  - pressing „1" – relay is switched on
  - pressing „0" – relay is switched off
- for MAB-1102
  - pressing 1 – relay status change

Relay status change can be also trigger via static path routing table or according Control schedule defined in PBX. An announcement defined in field **OnAnn**. In field **CmbSns** relay can be bind with sensor, so system phone button enables to relay control and button LED indicates relay status.

7.3 Control schedule

Relays can be trigger cyclically according predefine schedule. Terms (up to 50) dcan be defined in sheet *Time/Control schedule*:

- **Cmt** – comment
- **WDay** – week day
- **Ti** – Time (hh:mm)
- **RelAct** – relay action
- **Rel –** selected relay

8 __System devices__

To Slican PBXes following number of dedicated system phones series CTS can be connected:

PBX	CTS-102.CL / CTS-102.HT, CTS-202.CL / CTS-202.BT (U terminal)	CTS-102.IP, CTS-202.IP / CTS-203.IP	Total
CXS-0424	12	18	24
CCT-1668 firmware <5.11	18	18	36
CCT-1668 firmware >=5.11	18	60	78
MAC-6400	124	124	124

System phone enable to use two phone books – *private* and *public*. Both phone books are stored in PBX.

8.1 Private phone book edition

Edition of this phone book from inside ConfigMAN application is limited to adding/removing prefixes ex. zero. For preserve personality, numbers are not visible totally. Instead of last three digits there are „*", and instead name of contact there is subscriber name.

8.2 Public phone book edition

Edition of public phone book is possible only from inside new applications Slican WebCTI/PhoneCTI or using CTS system phone. Appropriate privileges gran ted to extension number are only condition which enable to edit public phone book:
*   for WebCTI(admin) application **Subscribers/CTI Settings** fiels **WebCTI** (this functionality is not subject of licensing) or
*   for CTS system phone and WebCTI(user) application: **Subscriber/USpecial settings** field **PubPB** = Write and readout.
*   Numbers can be also dialed by index **XXX – this index can be edit from inside WebCTI/PhoneCTI applications.

Slican WebCTI instruction manual can be found on pubwiki-en.slican.com
Slican PhoneCTI instruction manual can be found on pubwiki-en.slican.com

__ATTENTION!__
*About possibility of using index dialing numbers from public phone book by index **xxx decides setting of parameter __PubPB__ in sheet __Subscribers/Special settings__.*

__ATTENTION!__
*Correct dialing number from public phone book using shortcut **xxx is possible only if configuration of PBX enables routing calls via __LCR__.*

8.3 Programmable buttons configuration in CTS system phones

PBX admin can remote program buttons for every CTS phone. Buttons programming is performed using sheet *Peripherals/CTS/Buttons*. Buttons can be also programmed directly from CTS phone keypad.

On this sheet we can select if we programming phone buttons or buttons on console which is connected to phone. As default up to two system console can be connected to system phone CTS-202, although to four system phones in PBX up to five system consoles CTS-232 can be connected, it can be done by checking field *CnsSrv – Five console service* in sheet *Peripherals/CTS/Buttons*. It enables for increasing number of programmed buttons up to 162. Each button can be programmed as:

- subscriber,
- group,
- conference (conference group or room),
- doorphone,
- audio device,
- sensor
- relay
- trunk group,
- line,
- number from public phone book,
- service (including call recording[3]),
- function,
- working mode.
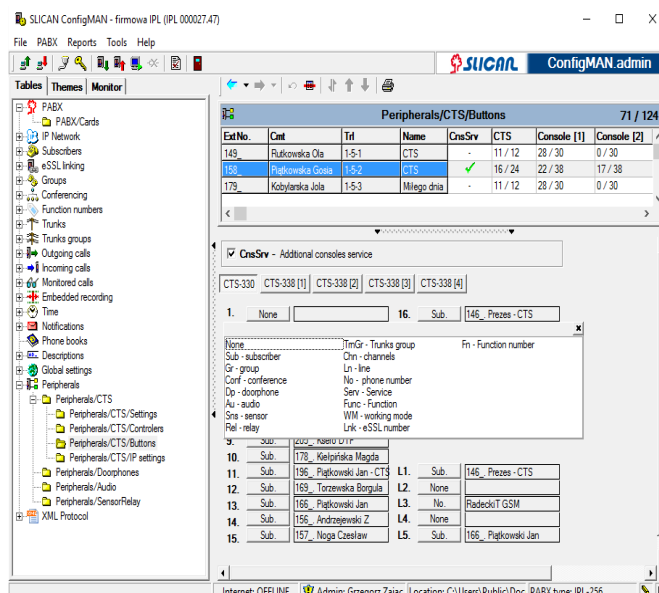
Fig. 8.4.: Sheet – Devices / CTS / Buttons

From inside *ConfigMAN* application etiquette printout is possible. To do this option *CTS button etiquette* from menu *Reports* should be selected.

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[3] This service can be programmed only from inside ConfigMAN
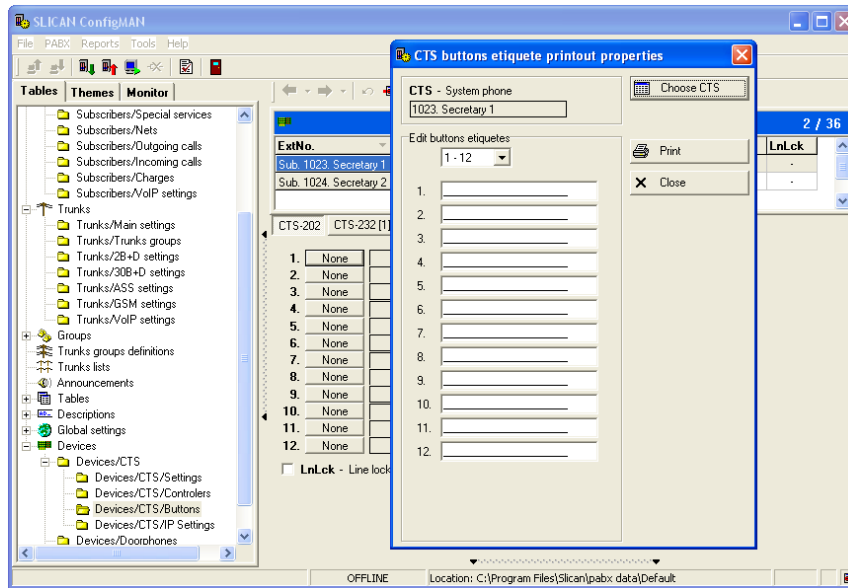
Fig 8.5.: Button etiquette for CTS-202 edition

Names to be printed are assigned by default according function realized by keys. There is possibility to individual edition of every single etiquette (then new line sign can be inserted, by pressing button ⏎), but text justification buttons act for all etiquettes. By pressing **Print** button, printout preview is displayed on screen.
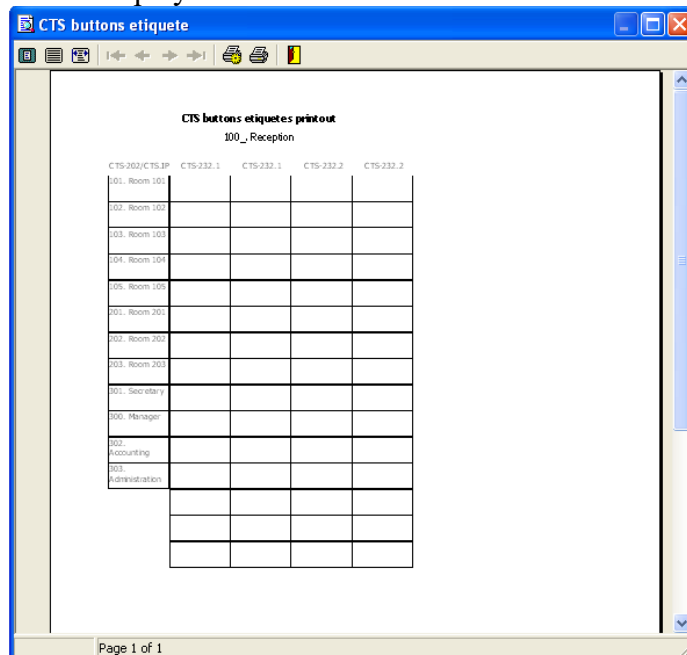
Fig 8.6.: CTS-202 etiquette preview

8.4 System phones CTS IP configuration

There are CTS-102.IP, CTS-202.IP and CTS-203.IP settings in sheet **Devices/CTS/Settings IP**. These settings enable their cooperation Slican PBXes.

Most important is *MAC address* of phone. *MAC address* it is hardware, unique address of Ethernet LAN card included in CTS IP phone. This address is assigned to phone by manufacturer while production process. *MAC address* of every system phone can be find on label at the bottom of phone or in phone menu **Menu / VoIP / Status**.
Only phones with their *MAC addresses* entered in field **MAC** of described sheet will be able to working with PBX.
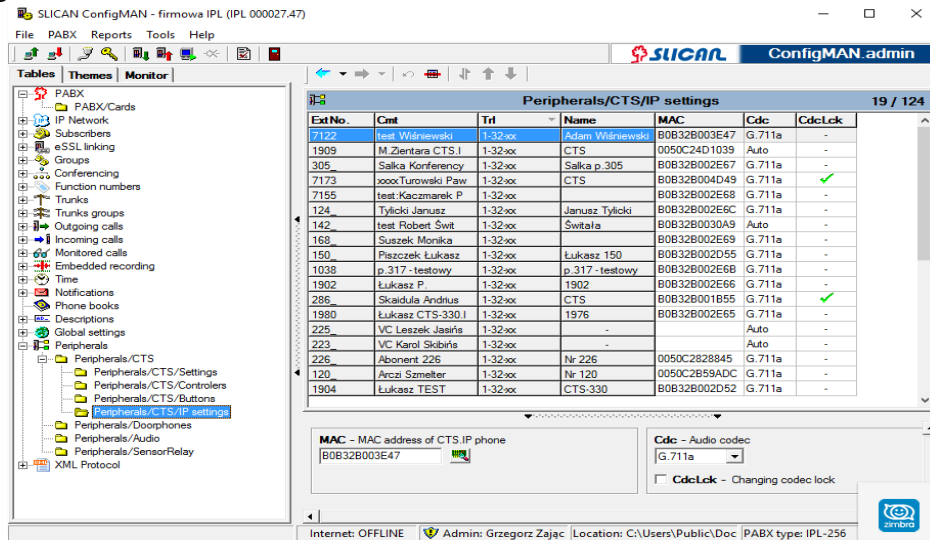
Fig. 8.7.: Sheet – Peripherals / CTS / IP Settings

To authorize CTS IP system phone in PBX following should be done:
– phone configuration (according system phone manual)
– connect it to the network – properly configured phone log to PBX
– there is an icon right on field **MAC** in ConfigMAN application. After pressing this icon all system phones CTS IP waiting for authorization will be displayed.
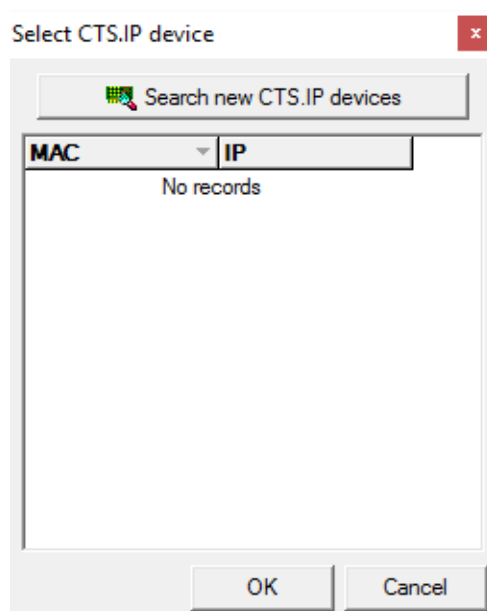
Fig. 8.8.: Window with detected CTS-203.IP phones waiting for authorization

In this window phones (*MAC addresses*) with proper VoIP server IP address will be visible. Also remain IP setting should be properly entered. Phone MAC addres*s* can be also entered manually.

Additionally in this sheet audio codec can be defined. Using this codec all calls from this system phone will be established – field **Cdc**:
– **Auto** – codec will be selected by PBX; if phone CTS-203.IP works in PBX LAN network, codec G.711 will be used, otherwise – if phone works in WAN network, codec GSM will be used
– **G.711a** – all calls will be established using this codec,
– **GSM** – all calls will be established using this codec.
Used codec can be changed from phone. After checking field **CdcLck** there will be no possibility to change codec from phone.

8.5 *Audio devices configuration*

To Slican PBX audio devices can be connected. To use all features of PBX it is recommended to use *Slican MAB-1101*, which can be configured differently – according your needs. Every audio device is connected to one analogue port. To configure port as audio device, in sheet *PABX/Cards* appropriate settings should be done.
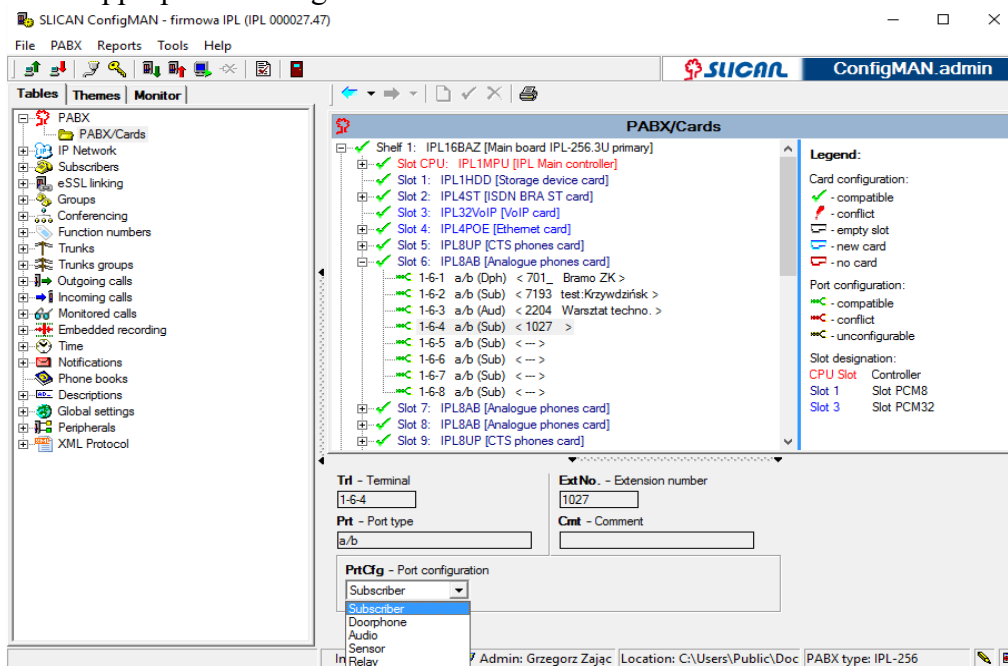
Fig. 8.9.: Sheet – PABX / Cards

Next type of audio device should be defined in sheet *Peripherals/Audio* – field *Typ*.:
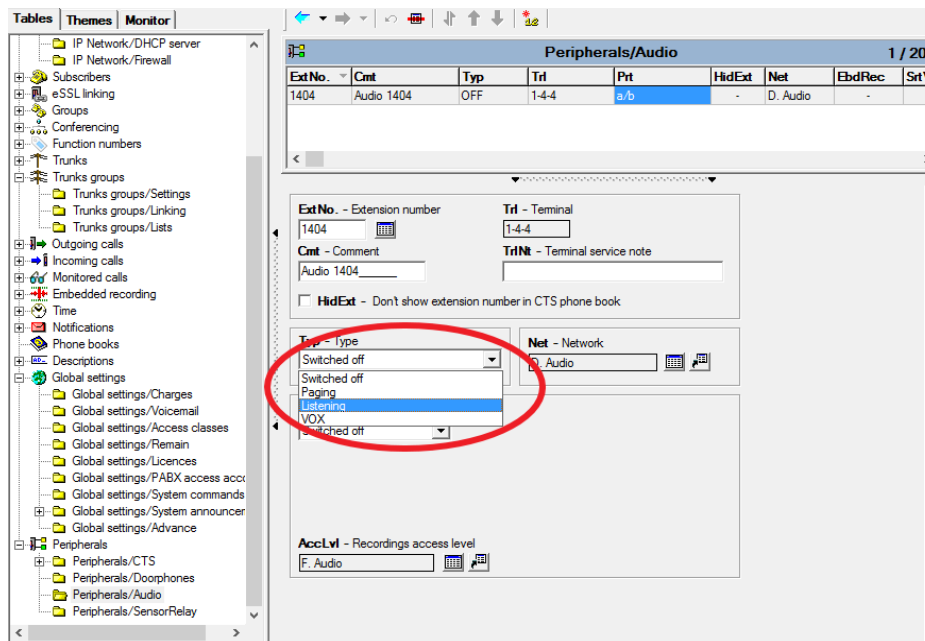
Fig. 8.10.: Sheet – Peripherals/ Audio

Switched OFF – enables to switch off an audio device.

Paging – enables to transmit audio signal from phone line to amplifier, for instance amplifier included in paging system. In this situation it is possible to defining signal which should be played in system loudspeaker before establishing call between subscriber and audio device (it can be done in field **PgnAnn** in sheet *Global settings / Remain*).

Listening– enables only for one direction communication, ex. listening room, it can be also used for connecting external signal source, ex. CD player,

VOX[4] – this option enables automatic switching on/off recording dependently on level of signal received by audio device. Such solution enables to switch on recording when someone is speaking (signal level above threshold level) and switch off when there is a silence (signal level below the threshold level), it save space on recording storage media.

---

[4] Eng. *Voice Operated eXchange*
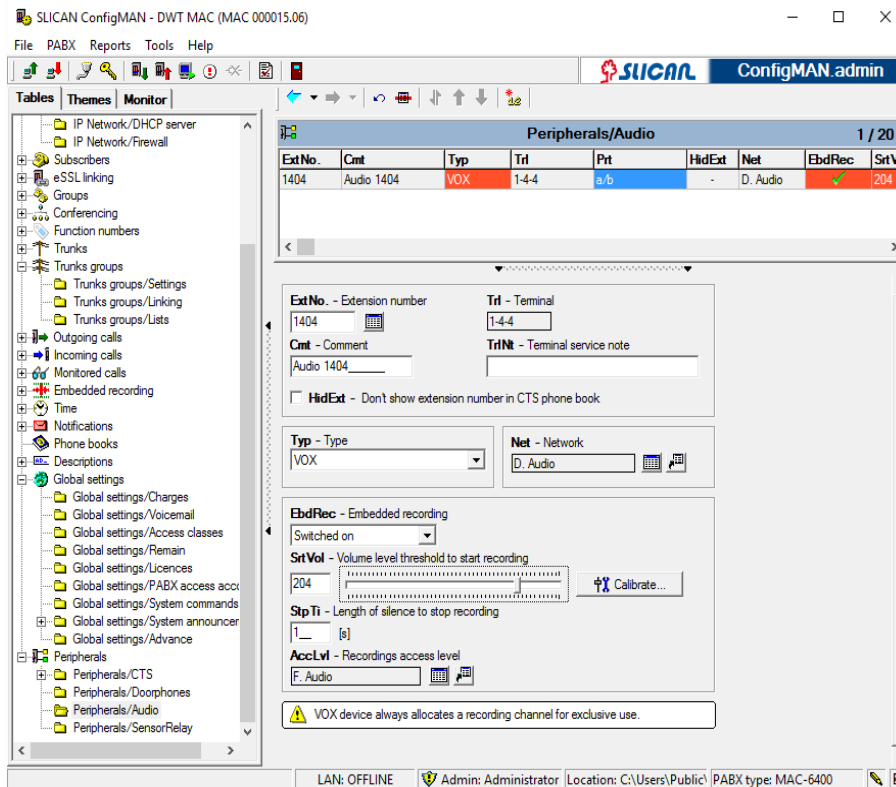
Figure 8.11.: Sheet Peripherals / Audio for audio device of VOX type

ATTENTION! *An audio device working in* **VOX** *mode occupy always one recording channel.*

Setting of threshold level which automatically switch on/off recording can be done in field **SrtVol** – *Volume level threshold to start recording* (to set this field button **Calibration**, available here, can be used). Content of field **StpTi** is used to define silence time before recording stop, if volume level is below defined in field **SrtVol** for the time defined in this field, recording stops. Appropriate settings protect against switching off recording during pauses in speaking (many *.wav files for one call can be generated).

8.6 **Doorphone configuration**

The SLICAN PBXes are compatible with Slican doorphones (excluding BRS). Every doorphone occupied one analogue port. To configure the device for working with doorphones, set the device configuration in the *PABX/Cards* sheet as a doorphone.
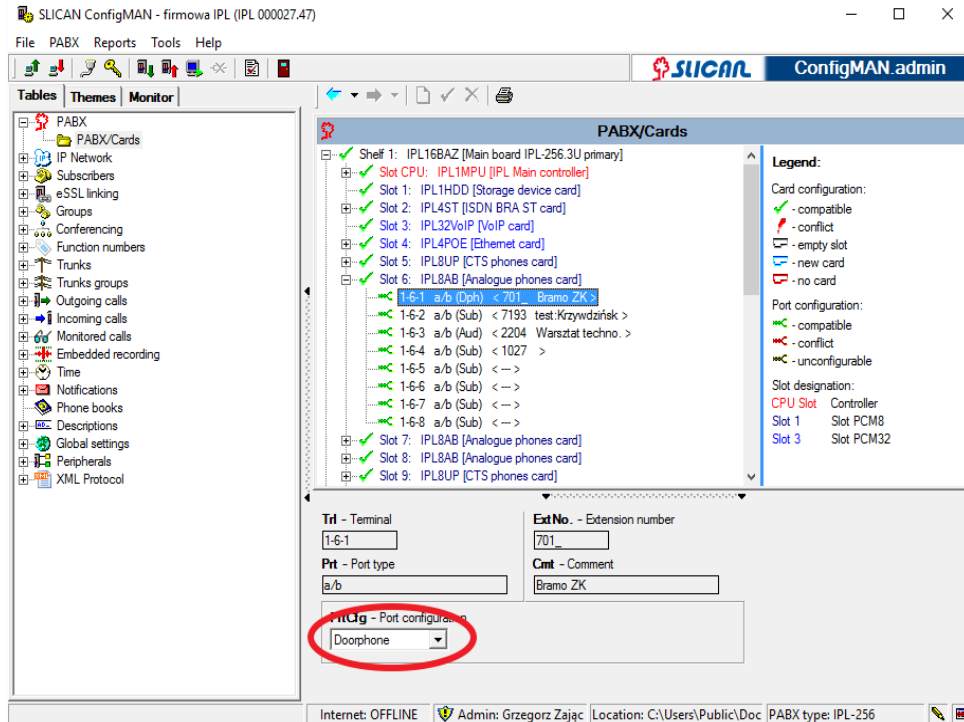
Fig. 8.12.: Sheet – PABX / Cards
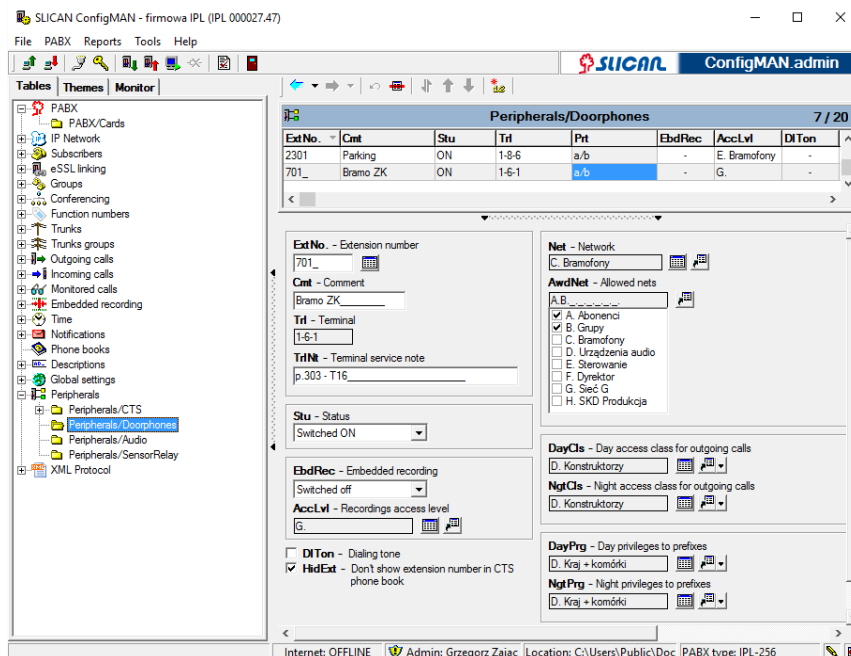
Fig. 8.13.: Sheet – Peripherals / Doorphones

For doorphone following settings are available:

- **ExtNo.** – enables to assign an extension number to the doorphone,
- **Cmt** – comment entered in this field has only information purpose,
- **Stu** – defines device status. Two options are possible:
  - *Switched ON* – enables to switch on the doorphone,
  - *Switched OFF* – enables to switch off the doorphone.
- **EbdRec**, **AccLvl** – described in chapter 5.2.3Recording calls with doorphone,
- **Net** – by assigning doorphone to specified net, access to this doorphone can be limited,
- **AwdNet** – this field defines nets to call can be established while dialing from doorphone internally (if check-mark is set for specified net, it is possible to dial any subscriber belonging to this net),
- **DlTon** – if this field is not checked, after dialing extension number, an announcement defined in field **MelTyp** in sheet *Global settings / Incoming calls* is played. Depending on settings in field **VceDis** (located in the same sheet) this announcement is interrupted with voice message „Please wait for connection". After checking this field standard signal is generated.
- **DayCls** – defines day access class for outgoing calls
- **NgtCls** – defines night access class for outgoing calls
- **DayProg** – defines day privileges to prefixes
- **NgtPrg** – defines night privileges to prefixes

9 **PBX settings verification**

After entering any settings verification should be performed. To do that, select the **Configuration verification** icon, also it is possible to use keyboard shortcut Ctrl+F7 (see chapter 2.8 Keyboard shortcuts).
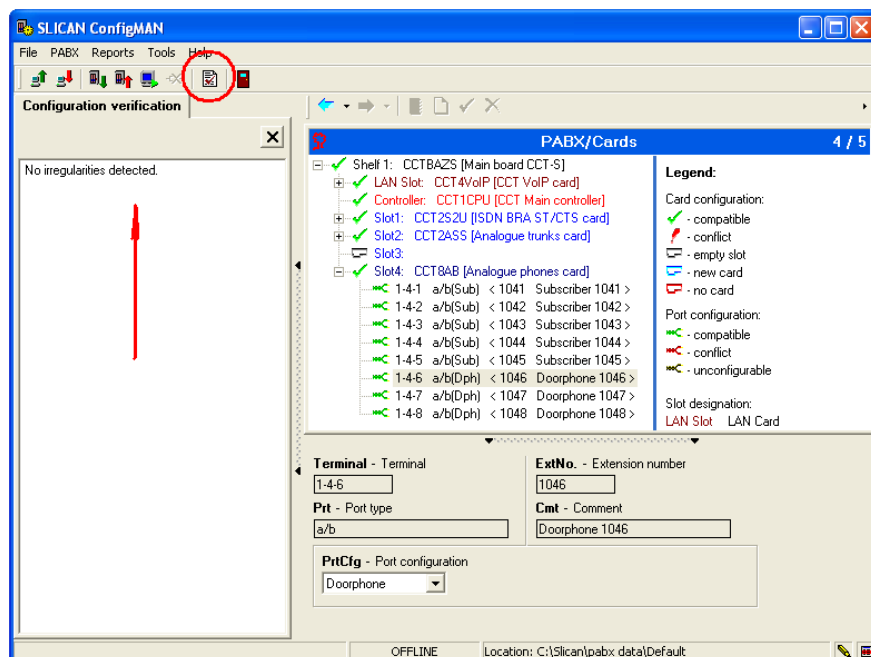
Fig. 9.1.: Configuration verification

If any error is encountered in **ConfigMAN** settings all errors are displayed in tab *Configuration verification*. On below figure you can see irregularity concerning with assigning the same extension number to two different subscribers.

Fig. 9.2.: Conflicts in extension numbers

ConfigMAN

10 PBX reset

10.1 PBX reset
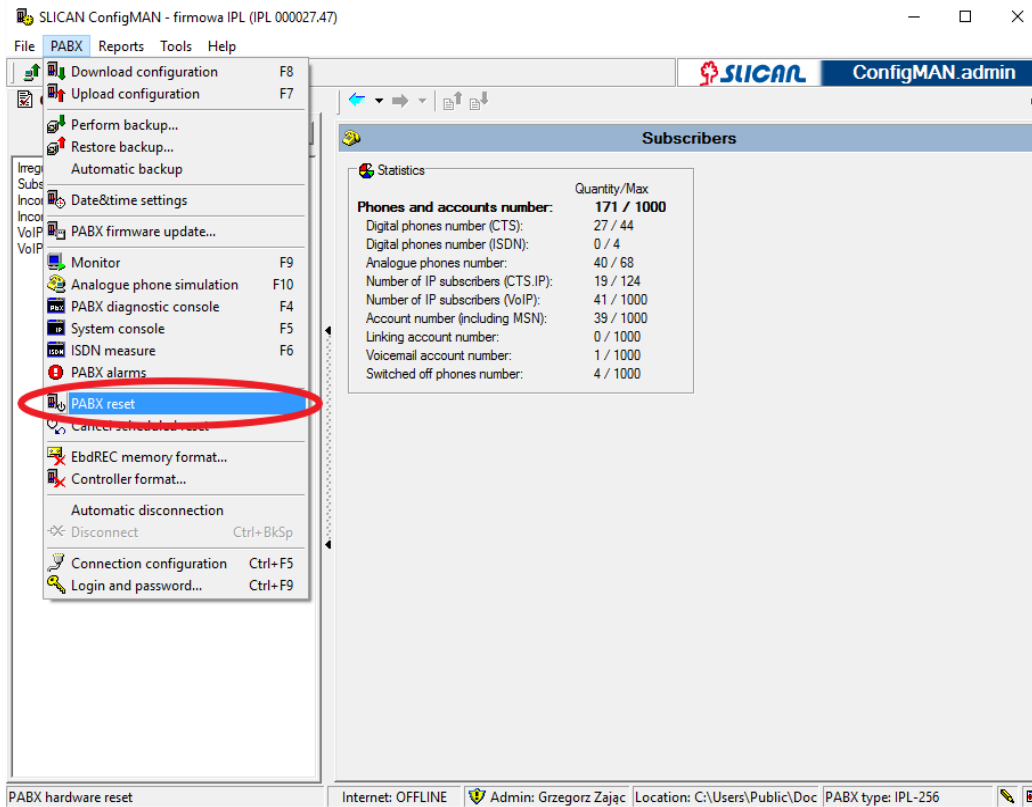
Fig. 10.1.: PABX reset

After selecting PBX reset – visible on above figure – reset mode selecting window is displayed on screen. It is possible to select one from two options:

- *Reset after closing all connections* – controller wait until no calls will be established in PBX
- *Reset immediately* – stops work, cancel all current and parked calls.
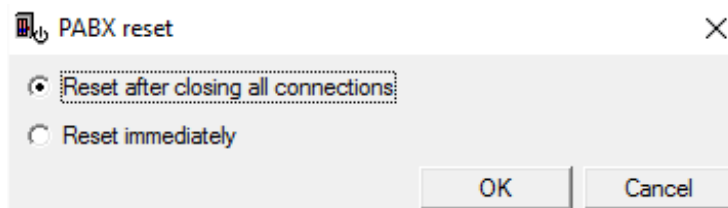
Fig. 10.2.: Reset mode selecting window

ATTENTION!
It is possible to cancel scheduled PBX reset procedure in any time by selecting option **Cancel scheduled reset** *from menu PABX.*

130

10.2 Restoring PBX factory settings – Hard reset wizard

If we want to introduce the factory settings database into the PBX, select the **Hard reset wizard** from the **Tools** menu.
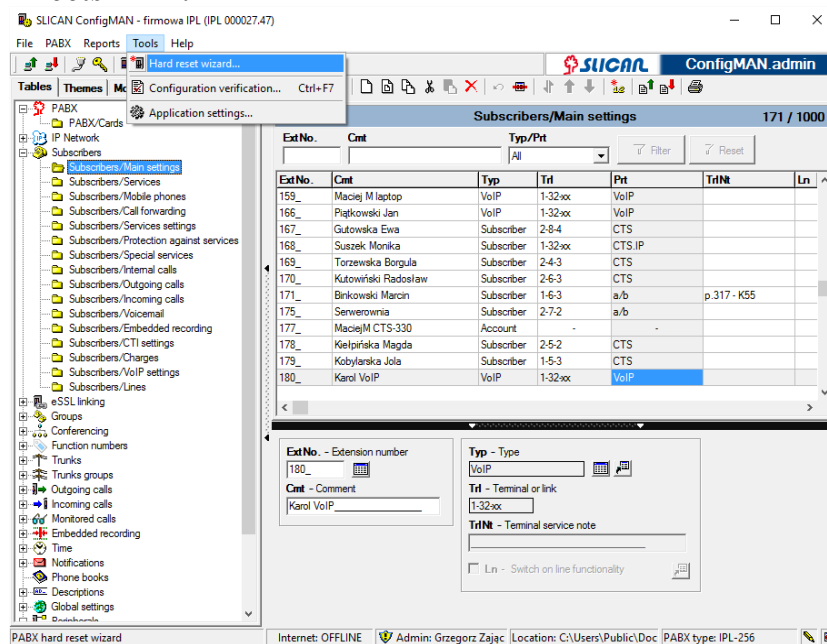
Fig. 10.3.: Basic PBX configuration wizard

After starting the **Hard reset wizard** the following message will be displayed:
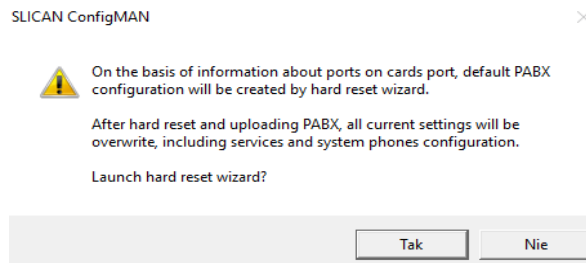
Fig. 10.4.: PABX hard reset wizard message

Next, the system will request the entry of own numbers and the number of DDI numbers for ISDN trunks. It is crucial to set which trunks are to be working in PMP configuration.

If the system detects analogue ASS or GSM trunks, it will request their own numbers. Additionally for GSM trunks SIM card PIN number. In the next step hard reset wizard detects VoIP card (if it is installed), logical VoIP trunks and their own number should be added.

Also it is important to provide the correct area code, for the PBX's location. Described above steps are presented on following figures.
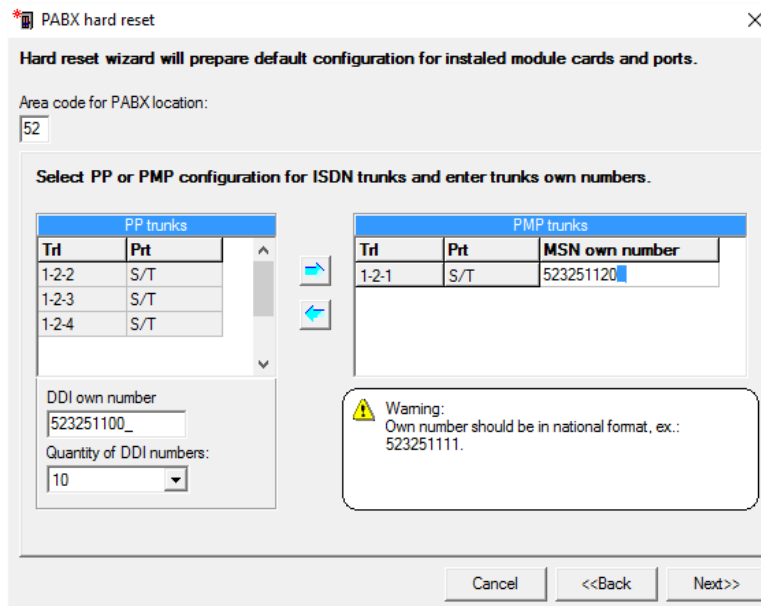
Fig. 10.5.: Hard reset wizard – ISDN trunks
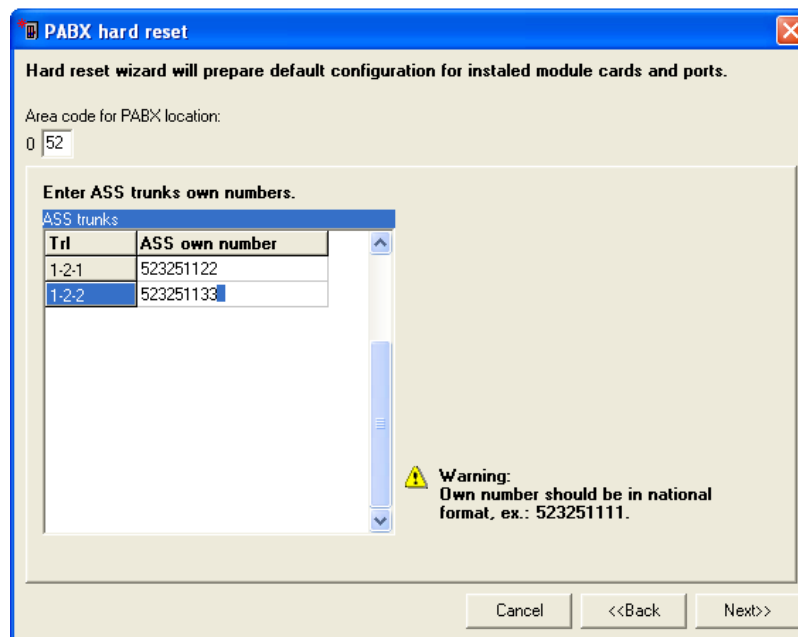
Fig. 10.6.: Hard reset wizard – POTS trunks

Fig. 10.7.: Hard reset wizard – GSM trunks

ATTENTION!

*In hard reset wizard we should add an anonymous VoIP trunks (one for every transmitting protocol: SIP, IAX). Its own number should be assigned. Other trunks, depending on needs, should be added in sheet **Trunks/Main settings** and **Trunks/VoIP settings.** See also chapter concerning VoIP functionality.*

Fig. 10.8.:LAN interface configuration

Fig. 10.9.: Hard reset – VoIP trunks

After defining trunk in next step (next figure) outgoing method or type of number should be defined as well as extension numbering rule.

Fig. 10.10.: Hard reset wizard – extension numbering and outgoing method

Next step is setting parameters of PBX admin account. <u>Default settings</u> for this account are:
- login: **admin**,
- password: *four last digits of PBX serial number*,
- code[5]: *four last digits of PBX serial number*.

Here it is possible to change default admin settings. After checking field **Hide password on screen using * characters** entered signs are not visible on screen.

---

[5] Used by services *87 – Manual working mode change and *92 – Announcement recording.

Figure 10.11.: Hard reset wizard – admin account

After setting all parameters needed to PBX working, automatic PBX configuration according manufacturer settings is performed.

Fig. 10.12.: Hard reset wizard – result

IMPORTANT: *After performing the hard reset, the configuration needs to uploaded to the PBX.*

After performing hard reset other PBX settings can be defined according user needs.

REMEMBER: *Hard reset causes the old access accounts to be deleted, and replaced by new one **admin account**. If you change default parameters **login** and **password** – they should be UNCONDITIONALY remember (if you loose these data there will be no possibility to connect with PBX using ConfigMAN and BillingMAN applications).*

11 Controller format

Controller formatting is exclusively a service function. **Going through the complete procedure, causes all the information in the PBX to be lost!**
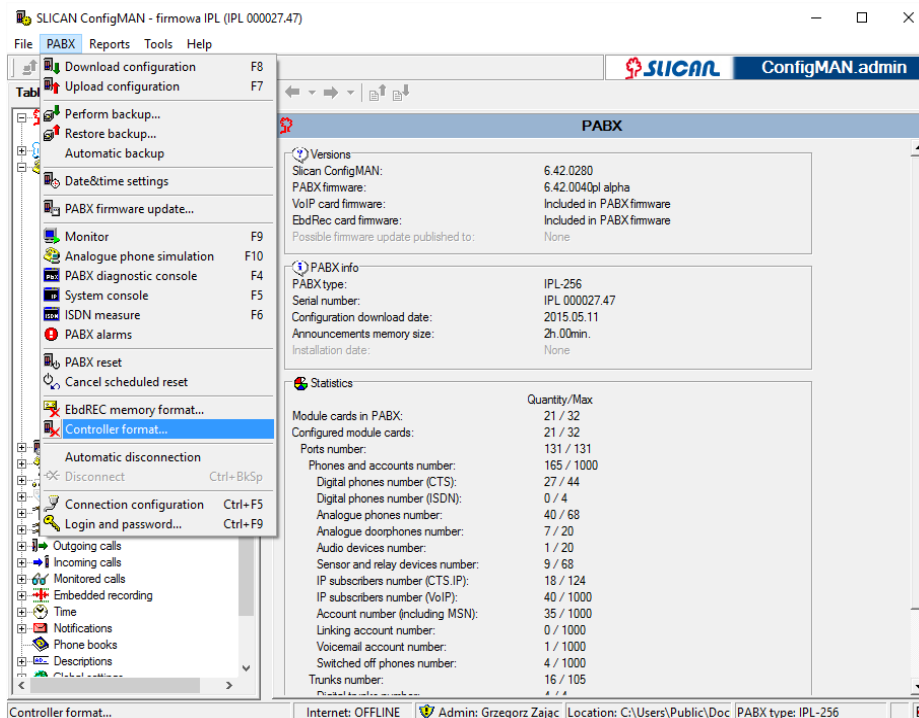
Fig. 11.1.: Controller format

After selecting option *Controller format* following window appear.
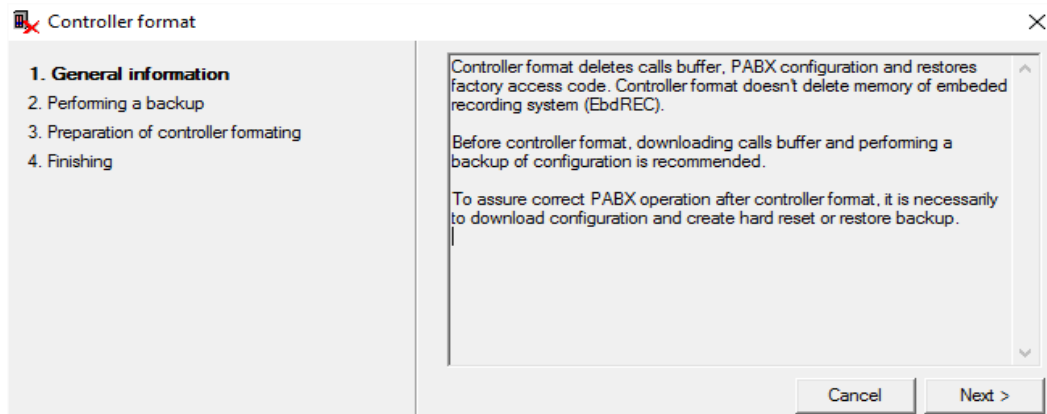
Fig. 11.2.: Controller format - general information

ATTENTION!
Before using the command, it is recommended to download the call buffer and make a configuration backup copy.
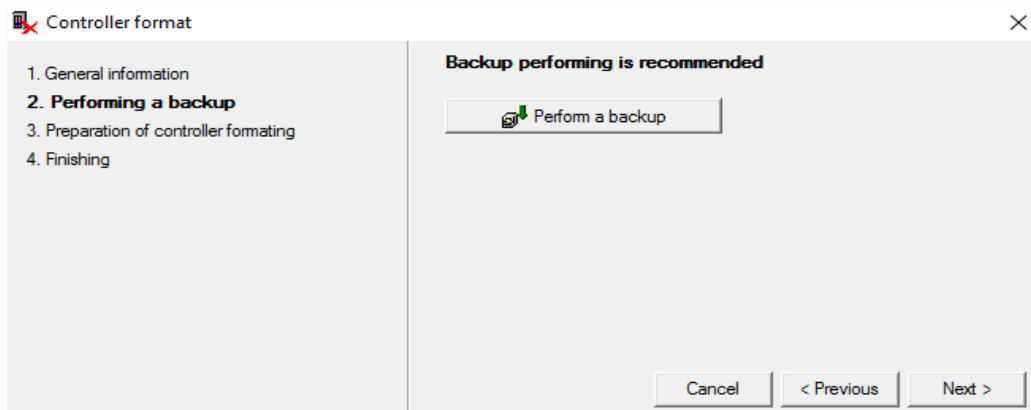
Fig. 11.3.: Controller format – performing backup

Window which appears in the next step depends on firmware version in PBX:
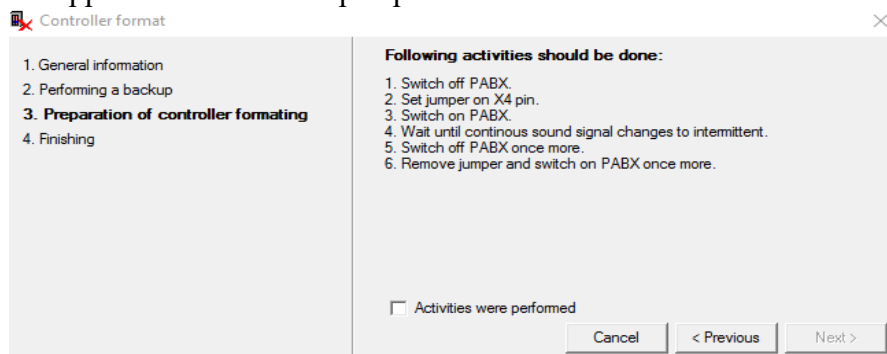
After performing all activities described in above window, you should check field „Activities were performed" and press the button Next, then follow appearing messages.

Controller format finish will be signal as follows:
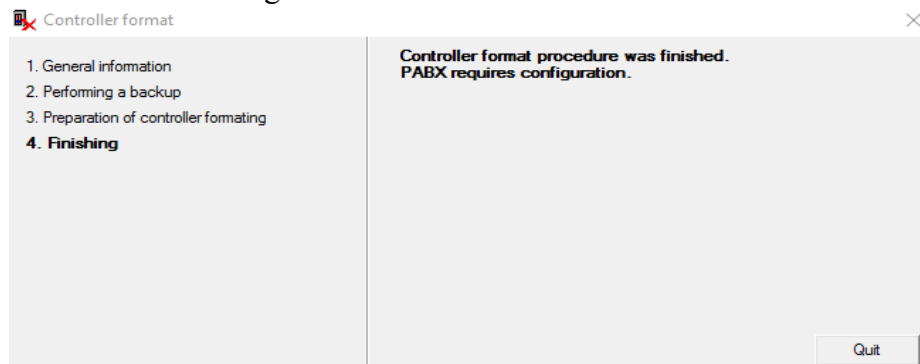
Figure 11.4.: Controller format - finish

12 Uploading PBX firmware

12.1 Uploading controller firmware

In *MAC-6400* uploading PBX firmware causes also updating LAN firmware, because LAN module is integrated with controller.

Before starting the firmware replacement procedure in Slican PBX familiarize yourself with the following information.

IMPORTANT: *Firmware update should be done during periods of low traffic, to allow a time margin for non-conflicting repairs in case of any problems.*

Remember: *Before replacing the firmware you need to* **create backup** *and, using* **BillingMAN** *download* **billing records** *from the PBX. It is also important to check the web page for information on the changes introduced by the new firmware.*

Attention: *After uploading configuration PBX reset should be performed.*

Attention: *Restart procedure may continuous a little bit longer (ca. 5 minutes) as usual.*

IMPORTANT: *During PBX restart and firmware uploading power supply cannot be switched off.*

Firmware uploading can be performed via any interface. However, due to the long time required to transfer the firmware, and the low connection quality, it is not recommended to use a modem for that task.

To replace the firmware, use the ***Firmware update*** option in the ***PABX*** menu.
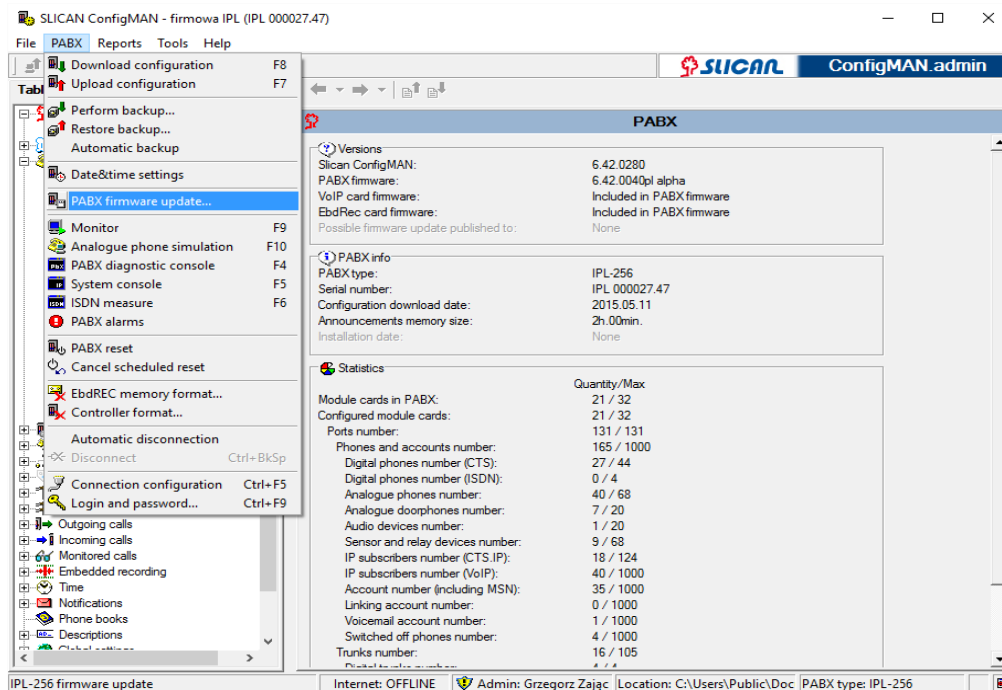
Fig. 12.1.: Update PABX firmware

Selecting this option will start the automatic firmware replacement wizard.

First message about requirement of downloading configuration before firmware uploading appears on screen:
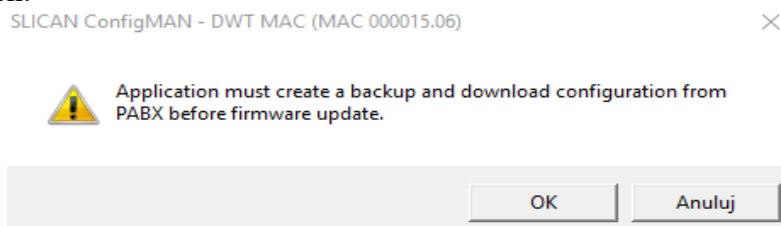
Fig. 12.2.: Firmware updating – download configuration request

After clicking OK button system will request the location of the current PBX firmware.
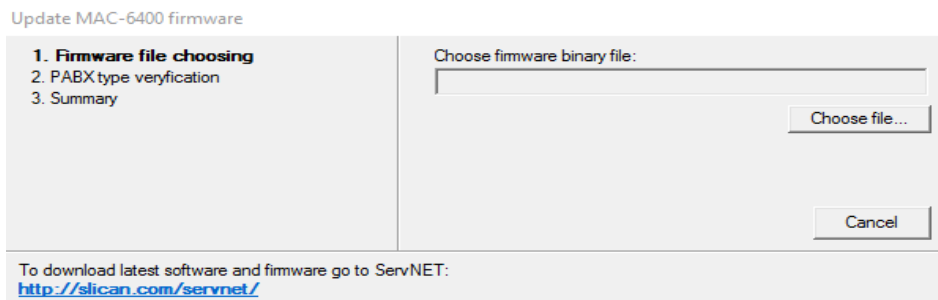
Fig. 12.3.: Firmware update – displaying first window

In some circumstances after selecting file message about necessity of acquiring license for firmware can be visible on screen. Detailed information can be found in chapter 17.1Licenses for PBX firmware.

After acquiring the file to be transmitted, the dialog box will display the firmware version, and the PBX families, this firmware is designed for.
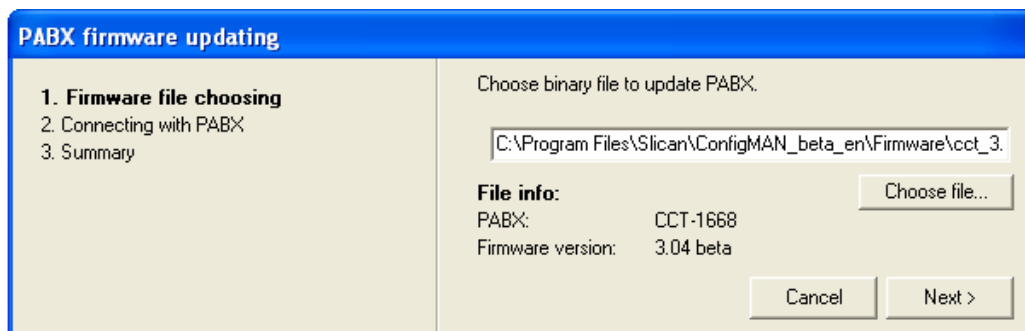
Fig. 12.4.: Firmware update – selecting file including appropriate firmware version

After confirming the firmware replacement process, the system will warn, that a reset of the PBX will be needed.
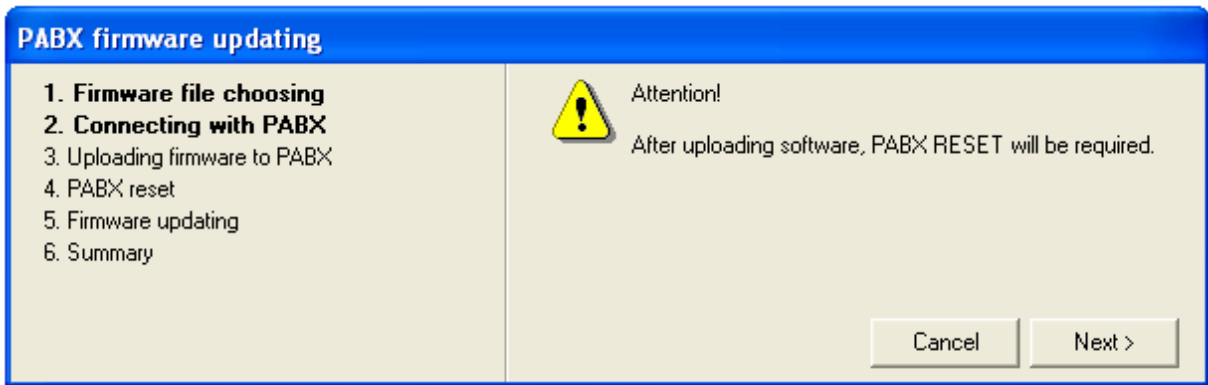
Fig. 12.5.: Firmware update – starting updating process

Selecting *Next* button starts firmware uploading.
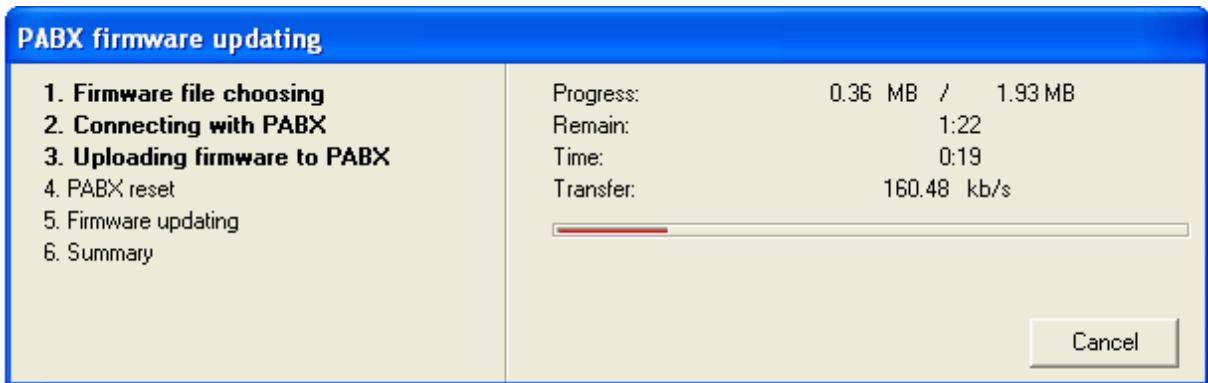
Fig. 12.6.: Firmware update – downloading/uploading firmware from/to PABX

After finishing uploading, the system will ask PBX reset. PBX reset is necessarily to proper functioning new firmware.
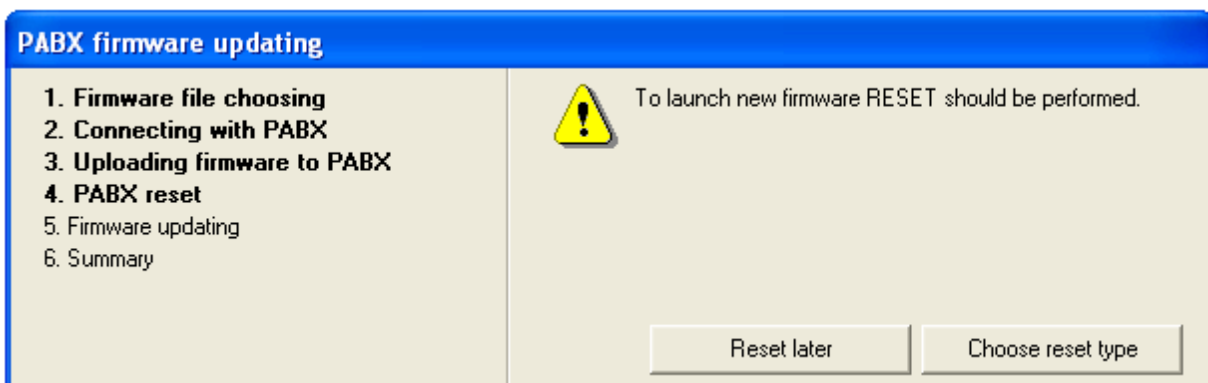
Fig. 12.7.: Firmware update – performing PABX reset

After finishing the process, the system will display a summary, or any possible causes of a failure.
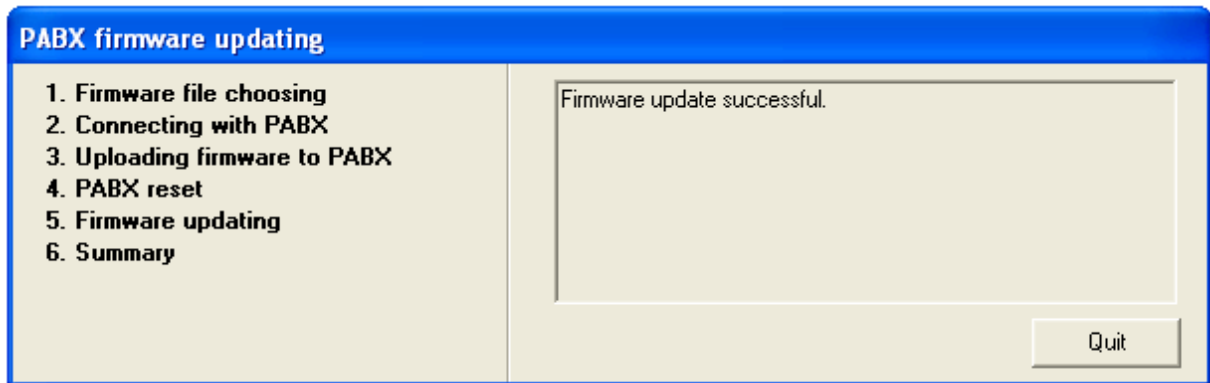
Fig. 12.9.: Firmware update – summary

12.2 VoIP card firmware changing

ATTENTION!
During VoIP card firmware changing power supply cannot be switched off. During LAN or VoIP card firmware changing computer and PBX cannot be disconnected.
In CXS-0424 there is no separate file with VoIP card firmware. This firmware is included in PBX firmware.

VoIP card firmware change can be done using following transmission media:
- *CCT-1668* – LAN network or (for older versions including 3.08 only via LAN network),
- *MAC-6400* – all available transmission media but USB port is recommended.
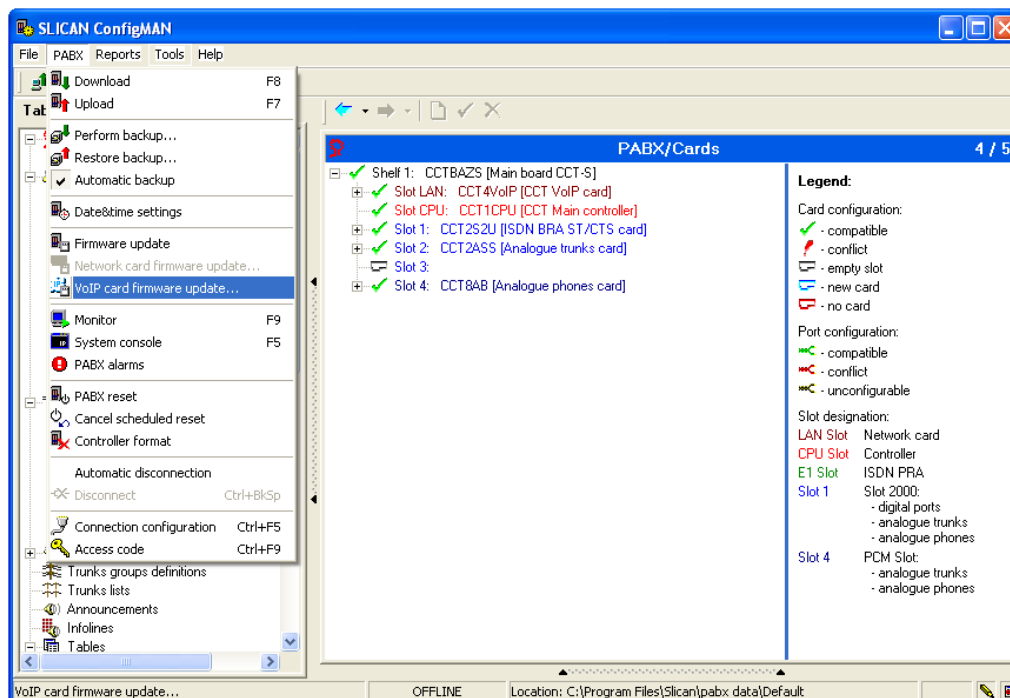
Fig. 12.10.: VoIP card firmware update – select option

After connecting with PBX, you should select **VoIP card firmware update** from **PABX** menu.

Next we select location of new firmware using button *Choose file...* and click *Next.*
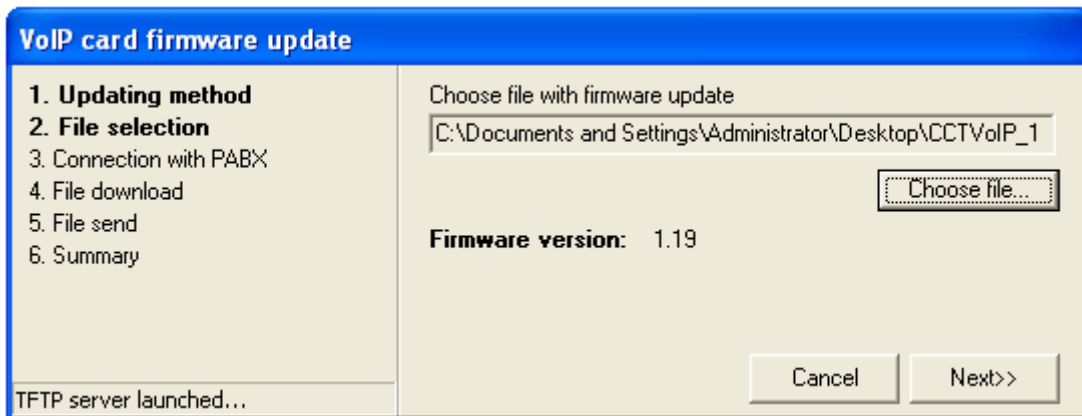
Fig. 12.13.: VoIP card firmware update – firmware file was selected

Remember! After clicking *Next,* PBX cannot be reset during firmware updating.
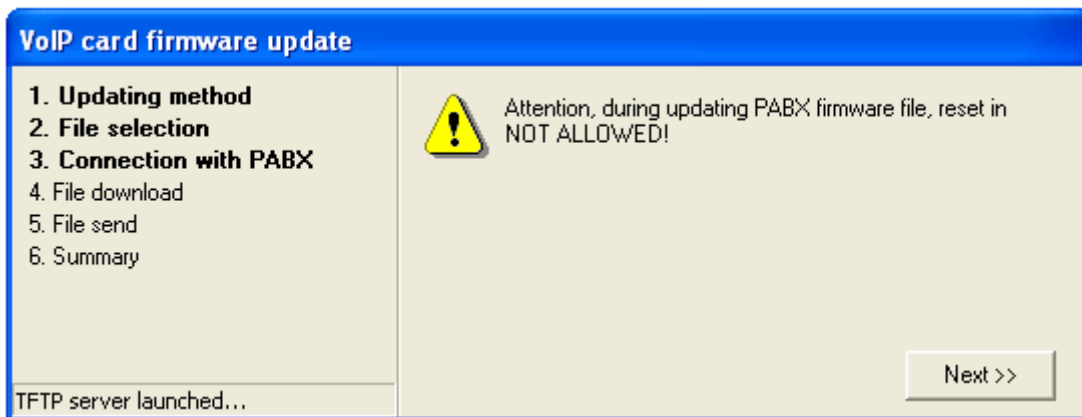
Fig. 12.14.: VoIP card firmware update – warning about reset

After pressing *Next* key, new firmware is uploaded to the card.
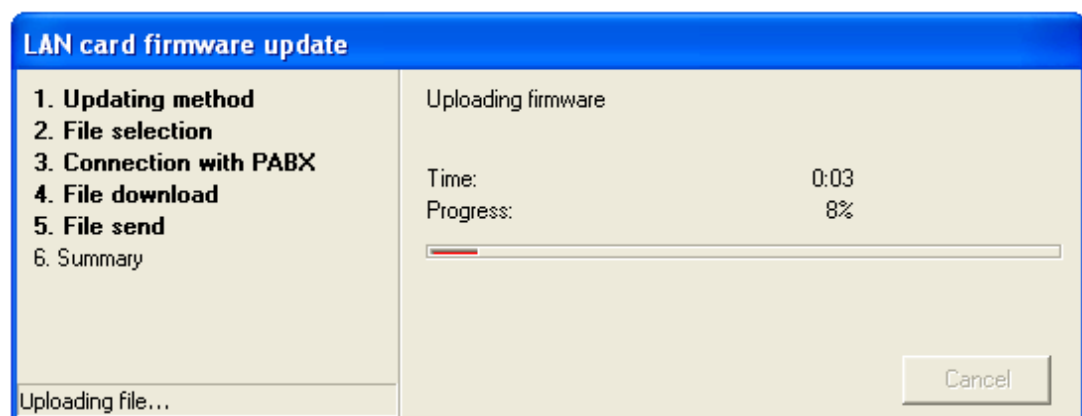
Fig. 12.15.: VoIP card firmware update – firmware uploading

After successful finishing updating application informs as follows.
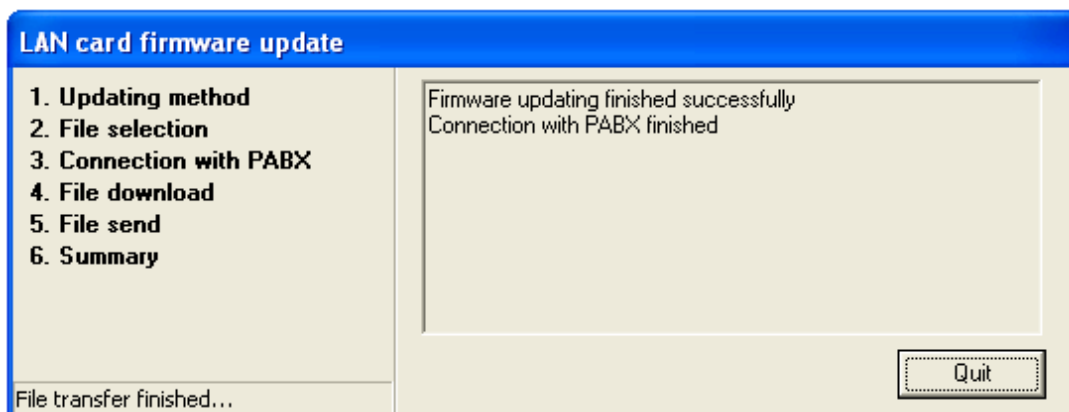
Fig. 12.16.: VoIP card firmware update- summary

Information about firmware updating can be found in **PABX monitor**, in sheet **PABX events**.

If interrupting connection system informs about it. Perform all described above activities once more. If it is unsuccessful, contact with service.

13 PBX alarms

A useful function of the software is possibility of tracing all alarms, notifying us of any possible irregularities.
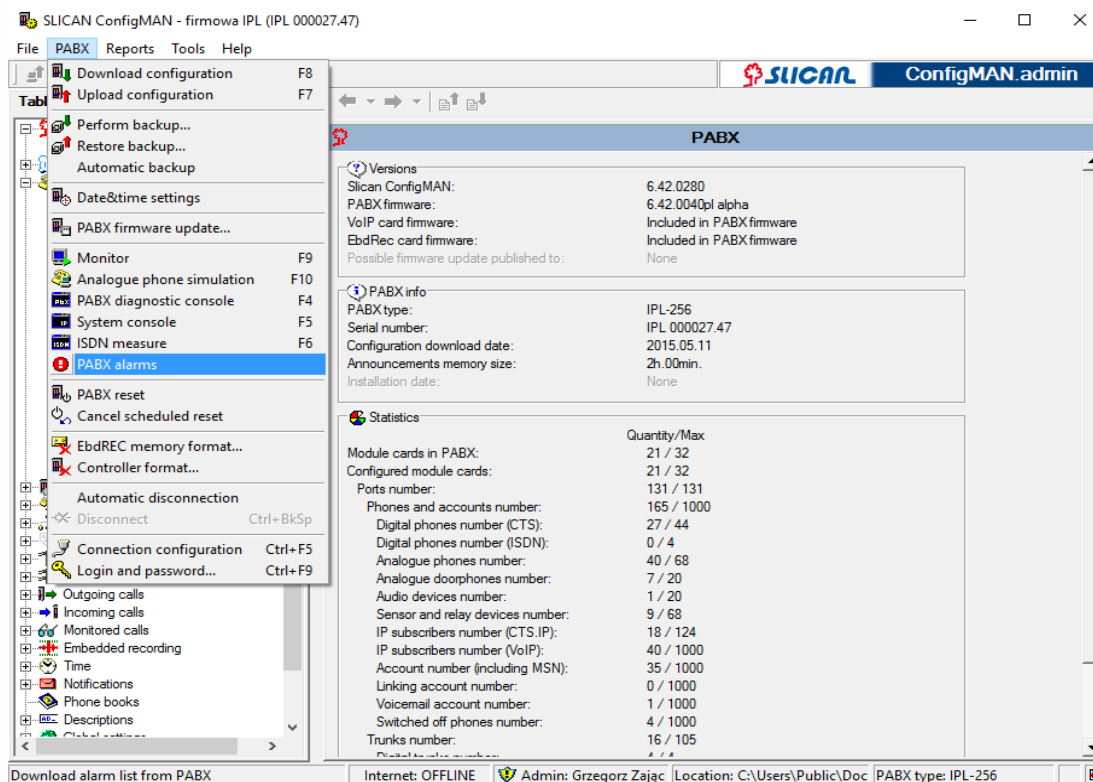
.Fig. 13.1.: PABX alarms

Examples of critical errors:
  – card conflict,
  – port conflict,

- – hardware damaged,
- – improper firmware,
- – invalid value of battery voltage,
- – no PBX serial number,
- – no supply,
- – call recording card from another PBX,
- – embedded recording system do not works.

If a critical or a non-critical error is encountered (e.g.: damaged trunk), a flashing exclamation mark icon is displayed in the toolbar. After clicking it, a dialog box is displayed, containing a description of error, and a list of actions needed to rectify it.
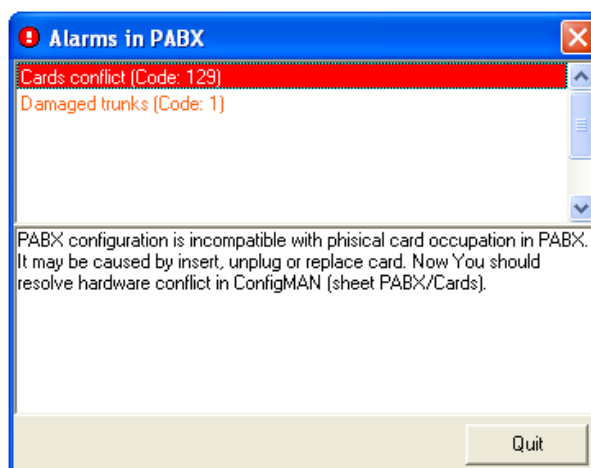
Fig. 13.2.: PABX alarm – example

14 Notifications

Current PBX monitoring can be easier due to notification functionality. Due to this functionality, SMSes are send to previously defined GSM numbers, these SMSes inform about events of some types. Basic requirement of this functionality is installing GSM card in PBX and configure two parameters in sheet *Notifications*:

- • **SMSTrn** – in this field, GSM trunk to send notifications should be defined;
- • **ChrAcc** – in this field notification SMS cost payer should be defined.

Additionally in sheet *Notifications/Messages,* in field **PBXCmt** PBX name can be entered, this name is send in SMS text together with alerts and events. In field **LogCmt** additionally description can be added, this description is used during failed log in to PBX from inside ConfigMAN, BillingMAN applications.

14.1 Service alerts

In this sheet up to 8 numbers (field **No.**) can be defined, to these numbers SMS notification, informing about specified alerts are send. Notification about following alerts can be switched on:

- • **SysFlr** – system failure e.g.:
  - ◦ ringer current failure
  - ◦ main/shelf power supply failure
  - ◦ controller overheating/standard controller temperature (plus shelf number)
  - ◦ PBX reset caused by error

- ◦ VoIP card reset
- ◦ recording card failure/fixing
- ◦ remove/insert port card (also slot number)
- ◦ no communication with shelf
- **PrtFlr –** port failure accessible only if field **SysFlr** is checked. Ex:
  - ◦ trunk failure/fixing
  - ◦ subscriber port failure/fixing
- **Cfg** – uploading new software to the PBX:
  - ◦ configuration change
  - ◦ firmware update
  - ◦ announcement upload
  - ◦ LAN/VoIP/EREC card firmware update
  - ◦ memory storage format
- **EvtLog** – fulfilling event buffer in 85% or its overload
- **RecFlr** – recording system failure
- **Rst –** PBX switch off/on
- **LogLck** – lock subscriber account by entering wrong PIN code three times or lock service access to PBX after three failed log-in attempts
- **ErnPwr** – switch to emergency power supply after lack of normal power supply or restore normal power supply
- **RemA** – activation temporarily PBX remote access
- **AccLmt** – expiration of temporarily PBX remote access

Additionally to number defined in field **No.** a SMS will be send if configuration of notification for this number will change.

14.2 Events

In this sheet up to 50 numbers (field **No.**) can be defined, to these numbers SMS notification, informing about specified events are send. Notification about following events can be selected in field **EvtTyp**:

- manual PBX working mode change
- utilization of free minutes for specified trunk (trunk selection)
- sensor trigger (sensor selection)

Additionally for events text of notification SMS can be edited. During edition special marker **1** can be used, in SMS this marker is replaced by comment of sensor/trunk/working mode, depending on event type.

15 **Monitor**

Monitor is a *ConfigMAN* function, designed to supervise PBX operations:

- PBX port status,
- external line status,
- internal line status,
- VoIP channels status,
- recordings channel status
- status of calls made by PBX,
- PBX events.

The PBX can be monitored using any available connection method.

*ATTENTION: For the **Monitor** o operate correctly, the current PBX configuration needs to be downloaded.*

The Monitor contains useful functions to simplify operations, amongst others: buttons for activating/deactivating (keyboard shortcut: F9) . 🖥. It also allows the connections to be exported to a file 📄.
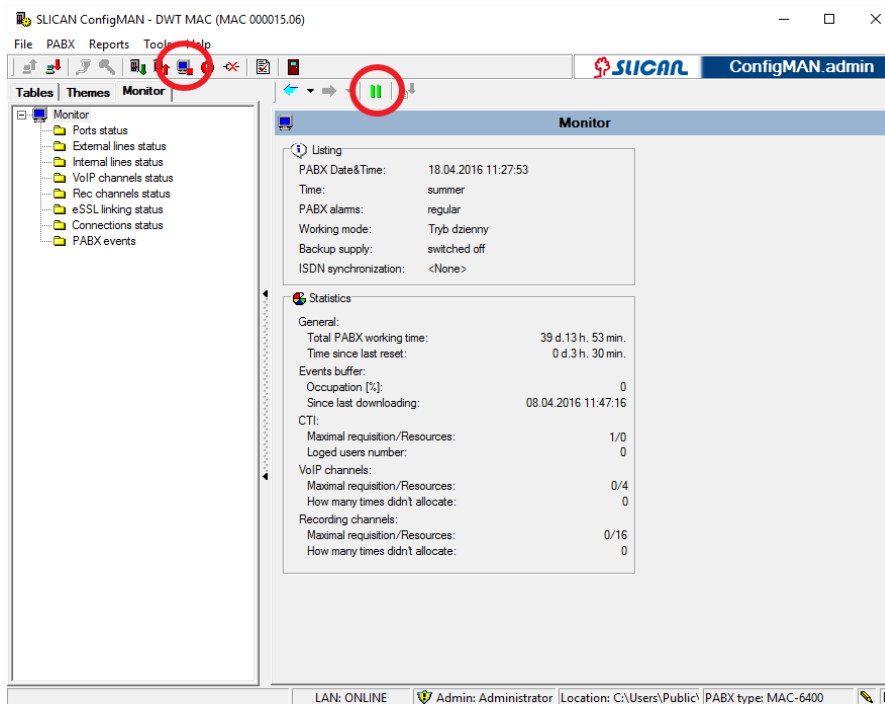
Fig. 15.1.: PBX monitor

Attention: *In ConfigMAN application there is possible to download/upload configuration without disconnecting Monitor. It is also possible to realize different functions from menu "PBX" during Monitor work.*

A very useful functionality of the software is the ability to freeze monitoring output on the screen (keyboard shortcut F11). It's introduction allows us to keep interesting information on screed, making them easier to analyze in detail. Clicking again the button ⏸ or F11 refreshes the status of monitored lines and ports.

15.1 Monitoring port status

Port status monitor provides information on the current status of all ports. Ports which are working correctly, are marked with a green tick. Faulty ports, on the other hand, are shown in red. Also, for digital lines, the cause of the alarm is shown. It is recommended to turn of unused ports in the configuration, to avoid their state in the Monitor to be in any doubt. For VoIP card status of following ports is presented:

– VoIP subscribers (logical terminal 1-LAN-1),
– CTS-202.IP subscribers (logical terminals from 1-LAN-2 to 1-LAN-19),
– VoIP trunks (logical terminal 1-LAN-1).

Additionally information about status of LAN/WAN ports and connection to keeper.slican.pl server are presented.
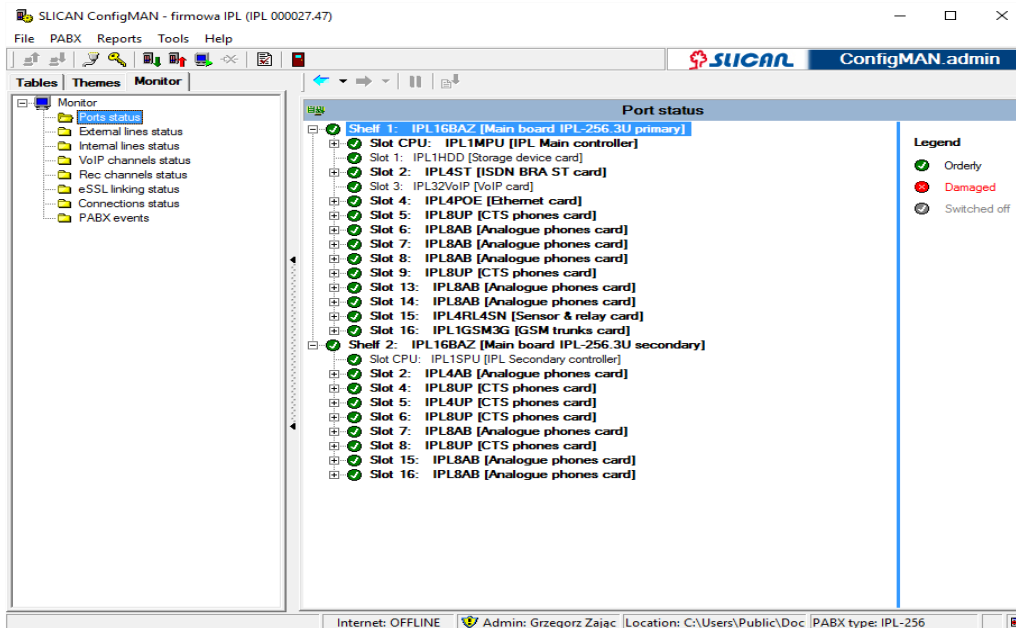
Fig. 15.2.: PBX Monitor – Port status

Fig 15.3.: Port status for VoIP subscriber after switching on PBX

Fig. 15.4.: Port status for VoIP subscriber after log in – used by subscriber IP address is visible

Fig. 15.5.: Port status for VoIP subscriber after log out – last login IP address is visible.

For GSM trunks it is possible to monitor current status selected GSM port (ex. information about log in to network or SIM card installation) as well as signal level information which can be used while aerial installation.
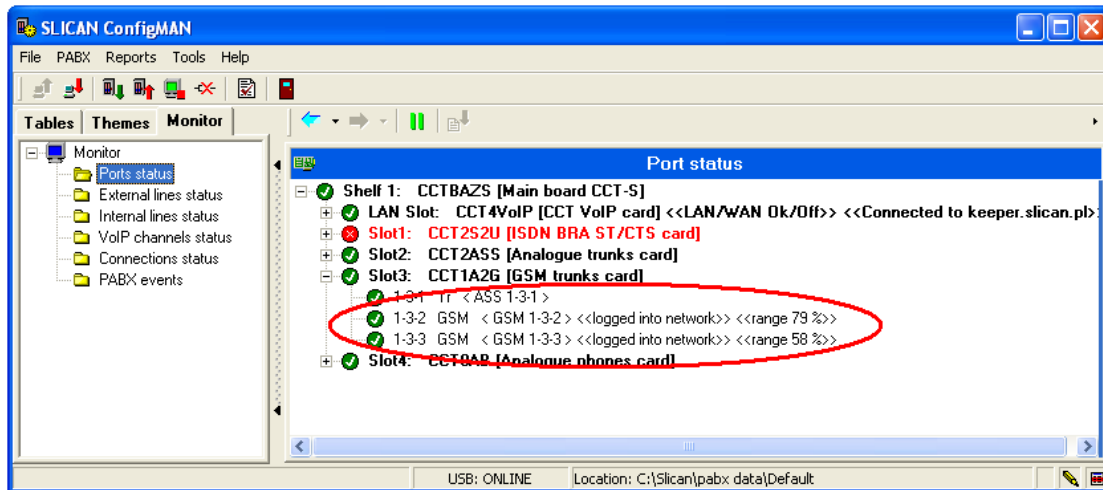
Fig. 15.6: PABX Monitor – Port status – GSM ports

15.2 Monitoring of external lines

Line state monitoring enables you to supervise trunks. To selectively monitor a line, you can select individual trunks to be watched (toggling them in the glasses column).
After selecting a filter, only the toggled trunks will be displayed. Additionally, the applied filter allows you to watch connections on the selected lines in the **„Connection status",** described bellow.
View filters:
In the *Show* field, you can set one of four available filters:

- *free* – unused trunks,
- *in use* – trunks currently in use,
- *damaged* – trunks marked by the PBX as damaged,
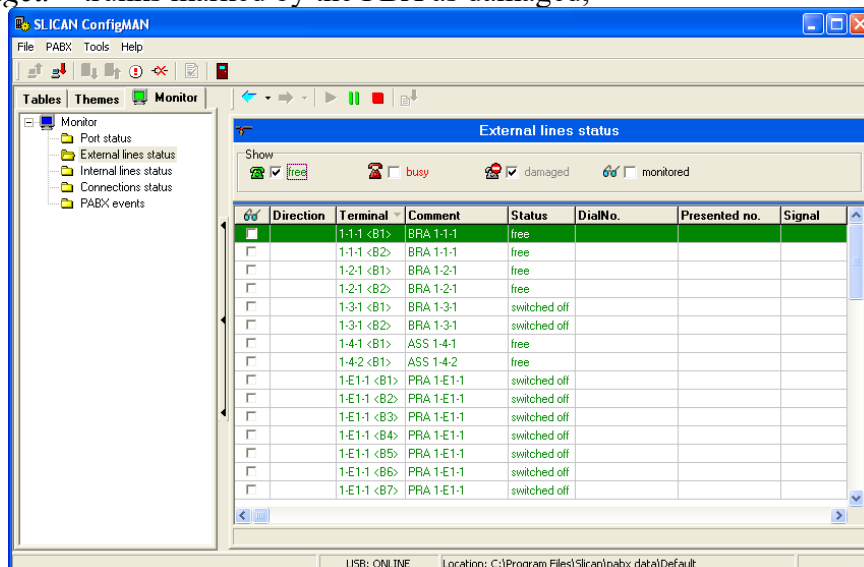
Fig 15.7: Monitor – External lines status

- *watched* – only lines marked as „monitored".

15.3 *Monitoring of internal line status*

Monitoring internal lines shows, in real time, all subscriber status (internal ports) in the PBX. If a subscriber is making, or receiving a call, the number dialed is displayed.
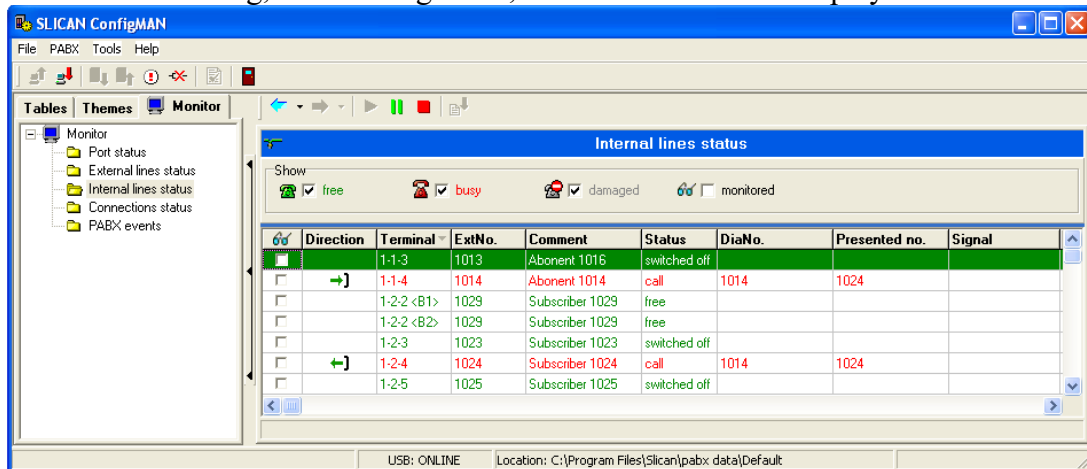
Fig. 15.8.: Monitor – Internal lines status

Displayed bellow *figure* presents, in Monitor window, one connection between extension numbers. In this situation from extension number 1901 number 123 is dialed , this number is group number and in group on extension number 1041 receive call. It should be noted that in rows representing caller and ultimate subscriber fields "**DlNo**" and "**Presented No**" include the same values. It is because in presented situation connection is internal.
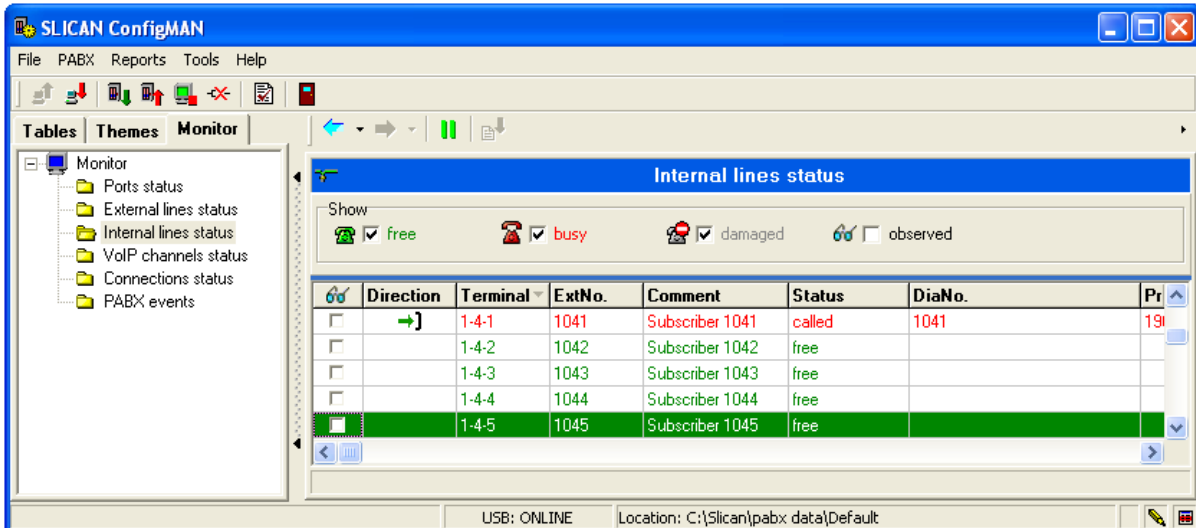
Fig 15.9.: Monitor – Internal lines status – Internal connections

15.4 *VoIP channels status*

If in PBX VoIP card is installed, monitor enables to monitor channel occupation. Bellow figure present situation when one of f available VoIP channel is busy. In this situation we have outgoing call (column *Direction*) from VoIP caller with name 1901 Grzegorz VoIP (column *Comment*), who dial number 100 (column *DialNo*). Caller present with number 1901 (column *Presented no.*). Channel status is defined in column *State.* Because established connection occupy only one VoIP channel, it means that is established between VoIP and regular internal extension number (analogue phone, ISDN, CTS-202 or CTS-202.Plus) or external number in PSTN network.
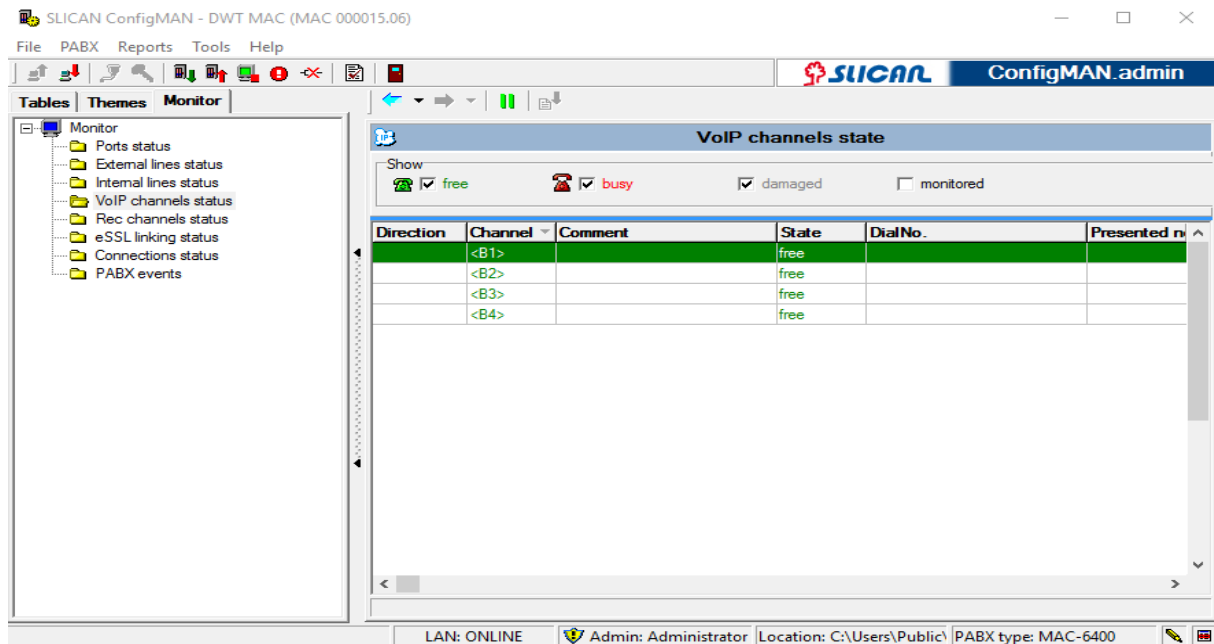
Fig. 15.10: Monitor – VoIP channel status

15.5 *Recording channel status*

If in PBX VoIP card with initialized embedded call recording system is installed, due to this functionality it is possible to watching busyness/availability of recording channels. Presented below figure sows situation when no channel is busy. In column *Comment* there is a name of recording channels presented (trunk or subscriber), while in column *ID* there is daily number of recording is presented, this number can be used in *RecordMAN* to find specified call recordings.
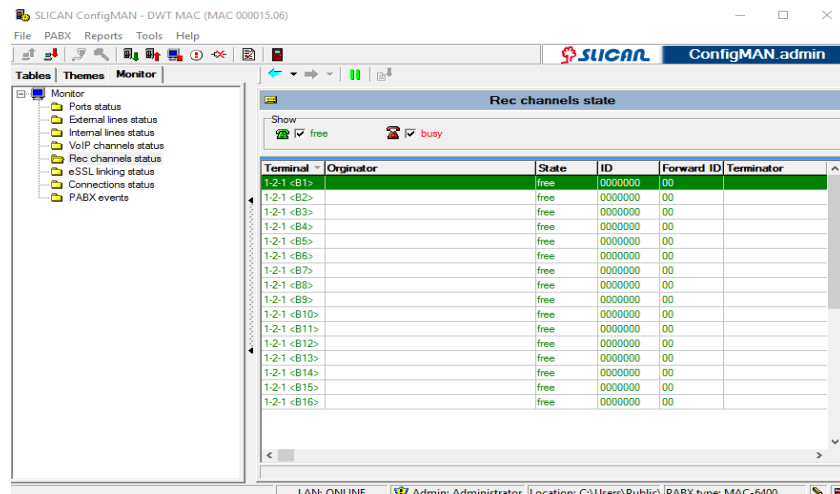
Fig. 15.11.: Monitor – Recording channels status

15.6 Monitoring connection status

Monitoring connection status allows the user to watch calls coming through the PBX. It is a fast way to diagnose connection errors. **ConfigMAN** enables he connections to be checked for availability in real-time, as well as quickly analyzing the recipients of incoming calls.
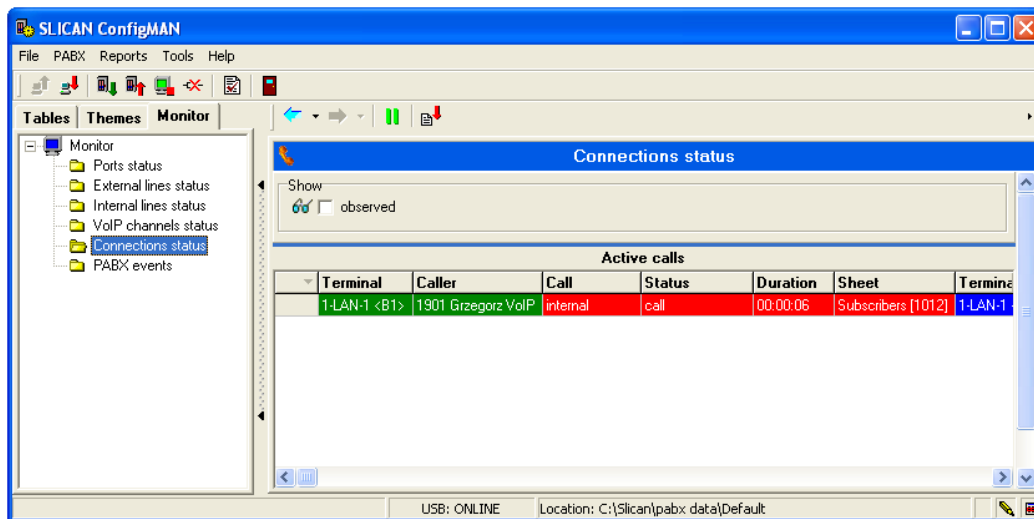
Fig. 15.12.: Monitor – Connection status

Watching calls is a practical way to verify the calls configuration in a PBX. The connection monitor provides, among others, the following information:
  - ➢ connection type - **Connection column:** incoming, outgoing, internal,
  - ➢ The sheet and the exact row used to direct the traffic and control call execution . Right-clicking on the **Sheet** field, allows you to "jump" to the row in the sheet used for routing this connection.

„Jumps" are also possible to:

External line status – if the clicked field is a part of external call.

Internal line status – if the clicked field is a part of internal call.

The **Active calls** window contains currently active calls. The **Finished calls** widow contains all watched and completed calls.
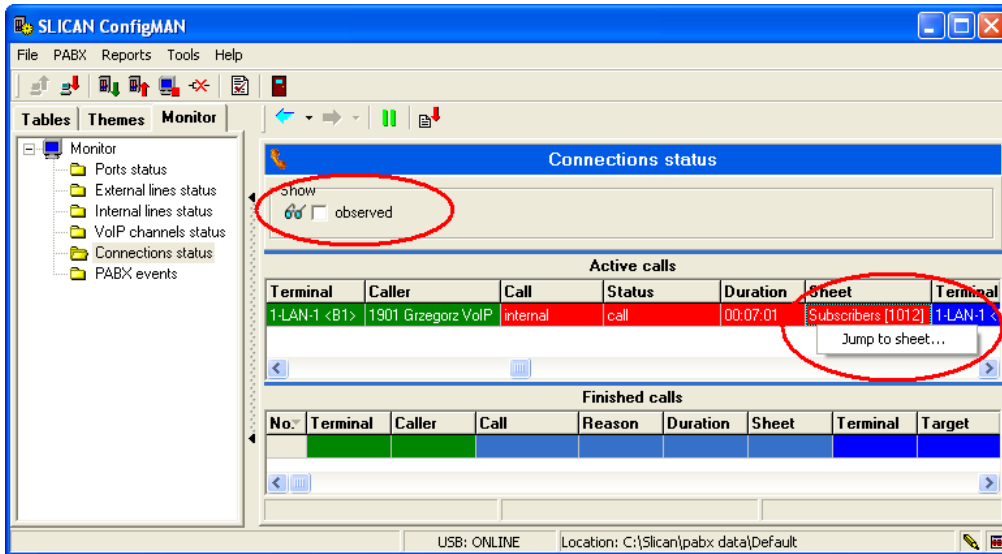
Fig. 15.13.: Monitor – Connection status window

Additionally, you can apply the **observed** filter to limit the number of records displayed.

Window *Active calls* contains few hide information, which are displayed after placing mouse cursor over some fields. You can see following information:

– placing mouse cursor over field *Caller* causes displaying box with information about dialed number (*Fig. 15.14*)

– placing mouse cursor over field *Target* causes displaying box with information about presented number (*Fig. 15.15*)

– placing mouse cursor over field *Duration* causes displaying box with information about time of establishing call (*Fig. 15.16*)
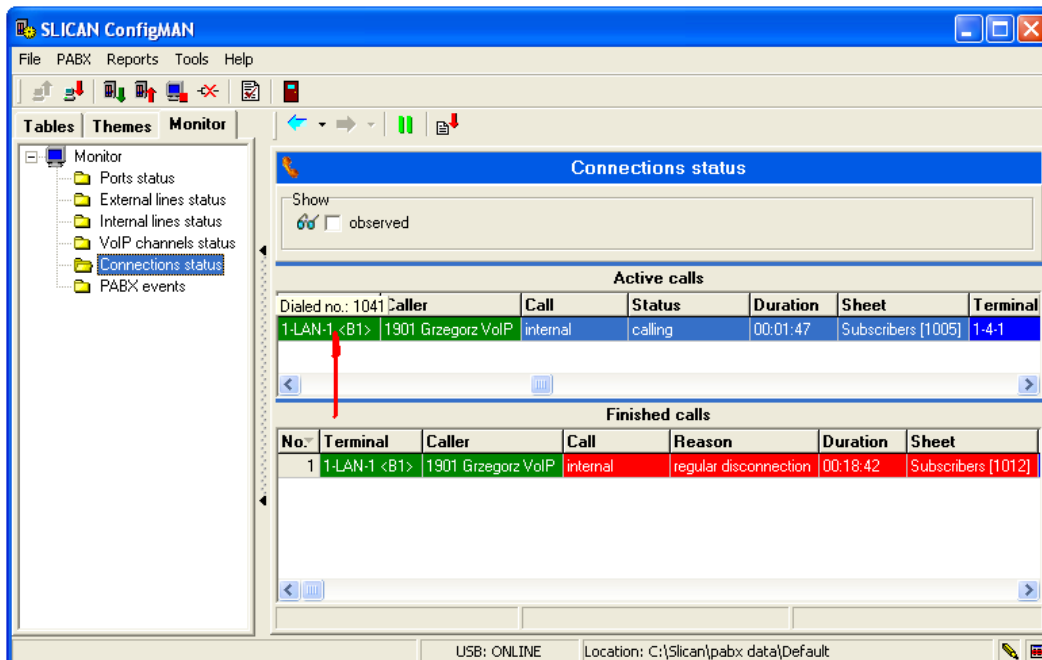
Fig. 15.14.: Monitor – Connection status – Information about dialed number
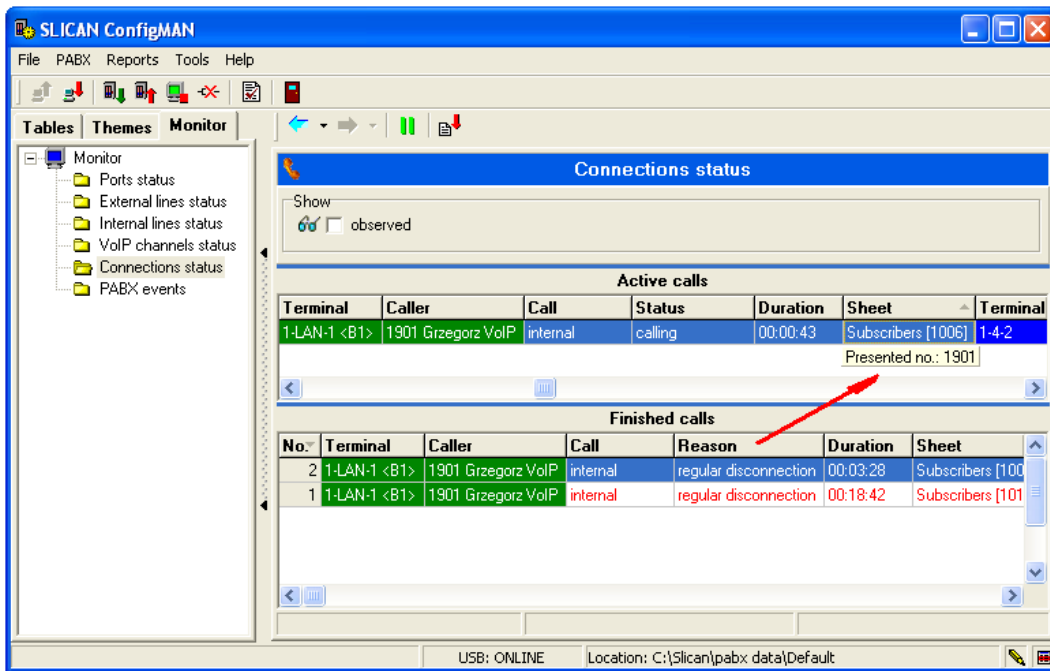
Fig. 15.15.: Monitor – Connection status – Information about presented number
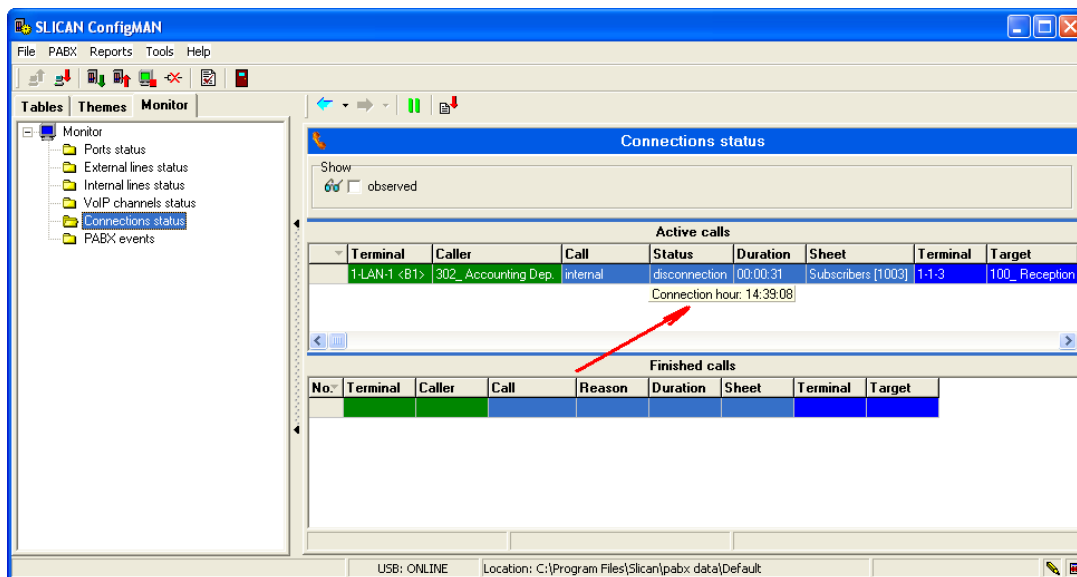
Fig. 15.16.: Monitor – Connection status – Information about call starting hour

15.7 Monitoring PBX events

In the event buffer stored are all events which occurred in PBX including:

- – switching on/off PBX,

- – activating off emergency power supply,

- – trunk damaging/fixing,

- – subscriber damaging,

- – call recording card damaging,

- – performance of service activities (configuration change, PBX firmware updating, LAN/VoIP/EbdRec card firmware updating),

- – reset free minutes assigned to the trunk.

The event buffer has capacity limited to 50 000 in *CCT-1668,* 60 000 in *CXS-0424* and 200 000 in *MAC-6400* (every next event overwrite the oldest one). Event list can be write in text (*.txt) or *.csv file.
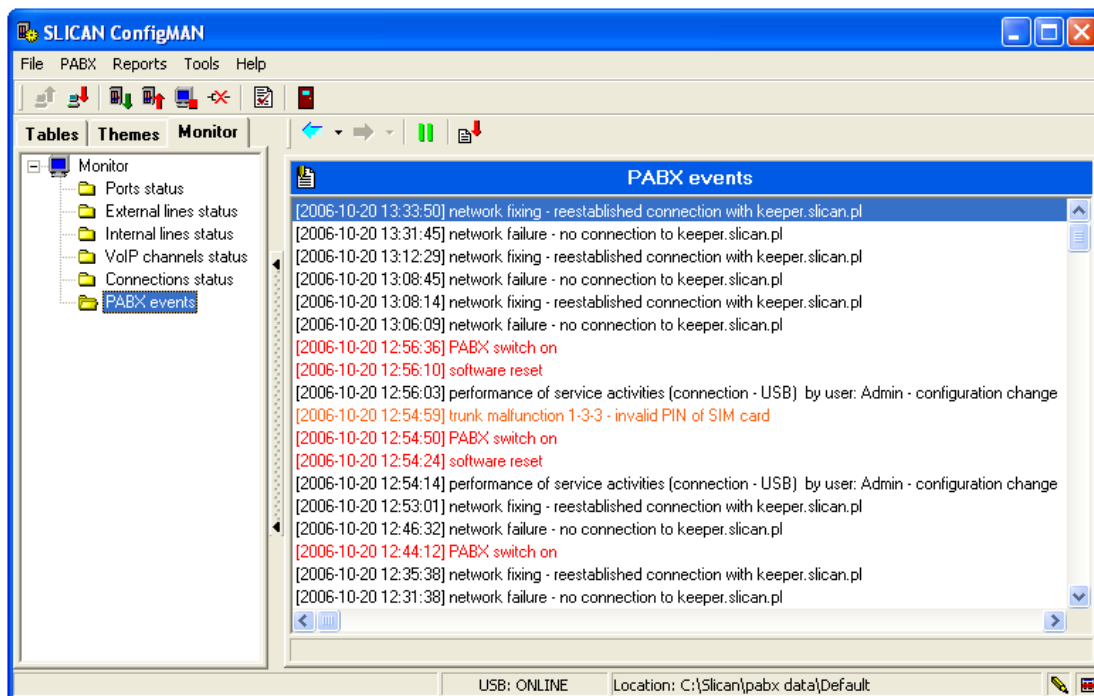
Fig. 15.17.: Events in PBX

16 Acoustic test services

In Slican PBXes two acoustic test services are available:
- Echo test - *981
- Melody test - *982

These services can be selected from any phone working with PBX as well as for incoming call on DISA announcement. For echo test „loop" is realized and tester will be hear in handset his own voice. Melody played by PBX while waiting will be hear in handset for melody test.

Incoming calls can be routed on selected service by entering appropriate entry in **Incoming calls** table.
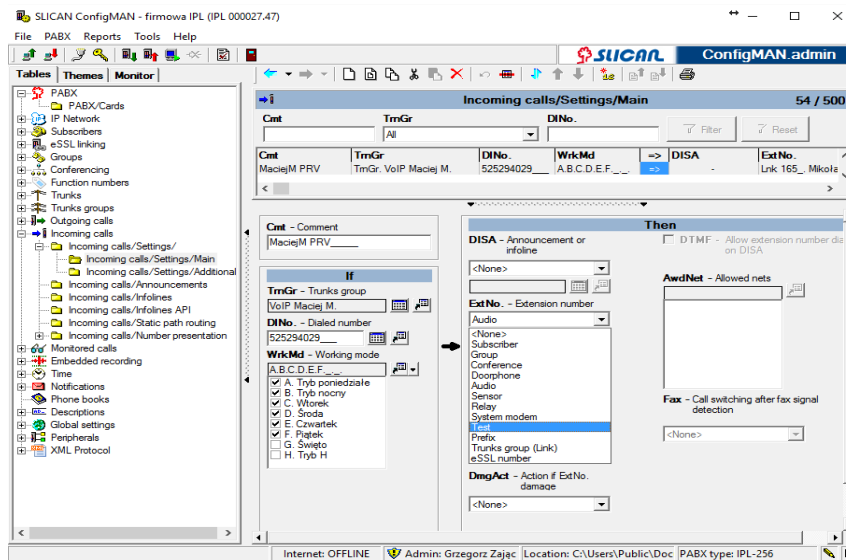
Fig. 16.1.: Setting acoustic test service for incoming calls
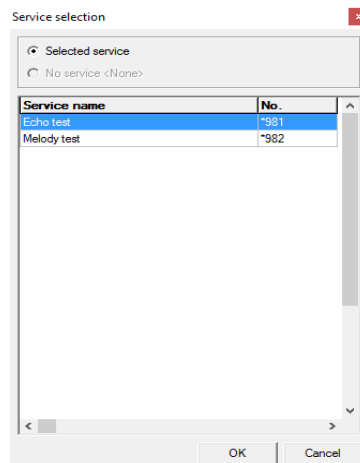
Fig. 16.2.: Test type selection

These functions are specially useful while remote PBX managing, ex. for remote evaluation of VoIP calls quality. This evaluation enables decide ex. about codec change to improve quality of calls.

Company Slican offer two phone numbers for using described above functions:
- (4852)3251101 – echo test
- (4852)3251102 – melody test.

17 __Licenses__

In PBX licenses are required for:
– PBX firmware
– *TelephoneCTI* and *TelephoneCTI.plus* software
– sending CTIP protocol to RS-232 interface (included in license for *TelephoneCTI*),
– PayMAN software
– TAPI workstation
– VoIP subscribers ports
– VoIP channels
– embedded recording system (license includes number of recording channel, *RecordMAN.server* and *RecordMAN.client* accounts),
– Infolines (number of infolines and maximal time of all recorded announcements)
– number of conference participants
– MobilePhone service
– CallBack service
– TouchCTI software
– Number of simultaneous calls via XML.IVR

17.1 Licenses for PBX firmware

Entering license code for firmware is required if you updating firmware from version 3.08 (or earlier) to version 4.xx/5.xx. Full table of relations between firmware version and serial number is available on site: http://wiki.slican.pl/index.php/Numery_seryjne_dla_wersji_FW

License code is issued for specified model, ex. Model S of CCT-1668 PBX or 2SH (shelves number) of MAC-6400 PBX. Missing of license code doesn't lock PBX operation but some limitations occurred. In fact the number presentation (CLIP) in internal and external call doesn't work. Instead of number, message "no license" and number "0000" is displayed on system phones C*TS-202*, and number "0000" is presented on analogue phones.

License names

Firm.LIC-XX.verYY.ZZZ – license for firmware in version YY for model XX of PBX ZZZ

, where:
   XX = PBX model, ex. S, L, EU (for CCT-*1668*), 1÷4SH (number of shelves for *MAC-6400*),
   YY = PBX firmware version,
   ZZZ= CCT (for *CCT-1668*), CXS (for *CXS-0424*) or MAC (for *MAC-6400*).

17.2 CTI software

License code is entered in field ***LicCTI*** in sheet ***Global settings/Licenses.*** Entering license code causes unlocking PBX to using application according aquired license code. Information

about maximal number of PhoneCTI, WebCTI, TelefonCTI and TelefonCTI.plus application, according acquired license code is displayed after entering code.

ATTENTION: When buying PBX with firmware 6.20 or later, license concerns total number of users of PhoneCTI and TelefonCTI applications. When you upgrade PBX firmware from version 4.xx/5.xx to version 6.20 or later, license concerns only TelefonCTI application. If you want to use new PhoneCTI application it is necessarily too buy new license.

If license code for CTI application was entered, you should switch on the privilege of using CTI applications for specified subscribers (in the scope of the license), it can be done in sheet *Subscribers/CTI Settings*.

Application doesn't allow to assign privileges to more subscribers then acquired license. PhoneCTI/WebCTI applications to log in requires password entered in sheet *Subscribers/CTI Settings*.

License names:
TelefonCTI.LIC-XXSUB.YYY – License for XX TelephoneCTI workstations
TelefonCTI.plusLIC-XXSUB.YYY - License for XX TelephoneCTI.plus workstations

where:
        XX = number of workstations,
        YYY = CCT (for *CCT-1668*), CXS (for *CXS-0424*), MAC (for *MAC-6400*), IPL (for IPL-256), IPM (for IPM-032) or IPS (for IPS-08).

17.3 ConsoleCTI

After entering license code possibilities of using application according this license is unlocked. Information about maximal number of **ConsoleCTI** users, which results from acquired license, is show in **ConfigMAN** application after entering license code. If license code is entered, for specified subscriber you should switch on the privileges to using application (in scope of license), it can be done by entering checkmark next to the ConsoleCTI field in sheet *Subscribers/CTI Settings*. This functionality is entered in firmware version 6.22.

17.4 TouchCTI and TouchCTI.VoIP software

This functionality is not supported from version 6.20.

17.5 PayMan software

This functionality is not supported from version 6.20.

17.6 TAPI workstations

Starting from firmware version 4.02 interface **TAPI** (to all *Microsoft* applications using this protocol) is available. The common use of **TAPI** protocol is using it to integrate PBX with *Microsoft Outlook* contact base. Special plug-in was developed by Slican to use TAPI protocol with this application. After installing it, numbers from this base should be dialed and number included in contact base can be presented with assigned to it name, contact can be

also edited. <u>Without license code</u> only <u>one workstation</u> can be used (one subscriber can use **TAPI** interface).

After entering license code in field **LicTAPI,** in sheet *Global settings / Licences* subscriber who use this interface should be assigned. It can be done in field **TAPI** – *Access to TAPI* in sheet *Subscribers / Special services*.

> **ATTENTION!**
> *This option is not available (field **TAPI** is greyed) for VoIP and ISDN subscribers.*

If subscriber use TAPI protocol and plug-in to *Microsoft Outlook* application, to log into the PBX, PIN code should be entered – it can be define in sheet *Subscriber / Services*.

Licence names:

> **TAPI.LIC-XXSUB.YYY** - License for XX TAPI workstations

where:

> XX = number of TAPI workstation (number of subscribers using TAPI interface),
> YYY = CCT (for *CCT-1668*), CXS (for *CXS-0424*) or MAC (for *MAC-6400*).

Starting from version 4.07 in PBX there is accessible also TAPI for server , password to this application should be defined in field **TAPIPas**, sheet *Global settings/Remain*.

17.7 VoIP card channels and subscriber ports

Number of possible VoIP subscribers and used VoIP channels is defined using license code.

<u>Without license</u> only demo version is active, in this version you have:
– 1 VoIP subscriber port (CTS-202.IP or VoIP – type of subscriber is defined by user and can be changed in any time),
– 1 VoIP channel,
– call time is limited to 60 seconds.

In PBX CCT-1668 and CXS-0424 up to 100 VoIP subscriber ports and up to 18 CTS-102.IP or CTS-202.IP ports can be created.
In PBX MAC-6400 up to 1000 VoIP subscriber ports and up to 18 CTS-102.IP or CTS-202.IP ports can be created.

To increase number of available internal VoIP ports You should receive appropriate license from Trade Department and enter license code in fields **LicVoIP** in sheet *Global settings / Licences*. It is possible to buy license for any ports number (in scope of PBX capacity) in any time.

As mentioned above, number of VoIP channel is also subject o licensing. Depending on license code and type of VoIP card following VoIP channels can be used:
– CCT - from 2 to 8,
– MAC - from 4 to 32,
– CXS - from 2 to 8.

If you use, ex. MAC16VoIP card (enables to use maximally 16 channels) with license for 4 channels, in any time you can increase number channels by purchasing in Trade Department license for next channels. License code can be increased by 2 channels or its multiplicity.

License names:

VoIP.LIC-XXSUB.ZZZ – license for XX VoIP subscribers (SIP or CTS.IP)
VoIP.LIC-YYCH.ZZZ – license for YY VoIP channels

where:

XX = number of VoIP subscribers,
YY = number of VoIP channels,
ZZZ = CCT (for *CCT-1668*), CXS (for *CXS-0424*) or MAC (for *MAC-6400*).

ATTENTION: In monitor statistics there is possibility to check how many times there were to little VoIP channels (how many times VoIP channels were not been assigned).

17.8 *License for embedded recording*

Subjects of licensing are:
– recording channels on call recording cards,
– account of *RecordMAN.server*,
– accounts of *RecordMAN.client,*
– access to recordings from external applications FTP/SFTP

Without license only demo version is active, in this version you have:
 – 1 recording channel,
 – 1 account of *RecordMAN.client* (*RecordMAN.client* application can be installed on one computer in LAN network),
 – recording time is limited to 60 seconds.
Recordings made in DEMO mode are specially signed in *RecordMAN.client* application.

Maximal number of recording channels (number of simultaneously recorded calls) is:
– *CXS-0424:* **8** (conscious number of channels after installing any LAN or VoIP card**)**
– *CCT-1668*: **12** (module *CCT.xx.M-8VoIP12Rec*[6] – 0VoIP0Rec card with installed 2 sub modules SM.DSP),
– *MAC-6400*: **128** (8 *MAC.M-16REC* modules installed).

If you use, ex *MAC.M-16REC* card *(*it enables to use maximally 16 channels) with license to use 4 channels, in any time you can increase number channels by purchasing in Trade Department appropriate license.
Licence names:

EbdRec.LIC-XXCH.ZZZ – license for XX recording channels
RecordMAN.server – license for using *RecordMAN.server* application
RecordMAN.client – license for 1 account of *RecordMAN.client*, maximally 30

where:

XX = number of recording channels,
ZZZ = CCT (for *CCT-1668*), CXS (for *CXS-0424*) or MAC (for *MAC-6400*).

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[6] xx – defines PABX model, ex. S, L, EU.

ATTENTION: In monitor statistics there is possibility to check how many calls were no recorded, because lack of available recording channels – see chapter6.25.5Optimization of system resources needed for call recording.

17.9 Infolines and announcement license

It is possible to define maximal 99 info-lines. Number of possible info-lines depends on acquired licenses. Without license only 1 infoline is available. Maximal time of all voice announcements is also licensed. Without any license only 20 minutes of recording space is available. Maximal time of all announcement in currently manufactured PBXes can be expanded up to 120 minutes (precise time depends on PBX controller version). It means possibility of acquiring license expanding announcements time from 20 up to 120 minutes. License number is entered in sheet **Global settings / Licenses**.

Totally PBX enable to record up to 99 announcements (both DISA and private DND messages).

License names:

InfoLines.LIC-XXINF.ZZZ – license for XX info-lines (XX = <2..99>)

InfoLines.LIC-YYTIME.ZZZ - license for YY minutes (YY = 30..120 every 10 minutes)

where:

XX = number of info-lines,

YY = duration time of all announcements,

ZZZ = CCT (for *CCT-1668*), CXS (for *CXS-0424*) or MAC (for *MAC-6400*).

17.10 Number of conference participants

Conferences can be realized using 3 methods:

- **Conference group** – up-ringing of predefined participants,

- **Conference room** – conference between subscribers who dial the specified extension number,

- **Private conference** – conference created from any *CTS-202* phone, using „Conference" service.

Mentioned above conference calls have different functionality, but it has no influence on licensing. Participant of conference room, private conference or conference group uses the same number of PBX resources and requires the same license. License limited only number of participants (currently talking) in every conference, it is presented in following table:

PBX type	Maximal number of participants (currently talking) in one conference		Maximal number of simultaneous conferences
	Without license	License	
CCT-1668	4	8	4
CXS-0424	4	30	10
MAC-6400	4	30	10

Table 17.1.: Hardware limitation concerning conferences

According above table, **without license** in any PBX only conference may consist of originator and tree participants, it is totally four participants or originator can establish conference with four participants and doesn't take a part in this conference themselves (doesn't talk). Similarly in conference room, with one little difference – in conference room there is no requirement to exist originator or custodian, so four participants may use conference room. No license doesn't limit the number of conferences, it means if in field **LicCnf** – License *code for conferencing* in sheet ***Global settings / Licenses*** is empty (license was not purchased), it is possible to establish:

– in C*CT-1668* PBX: 4 conferences with 4 participants in every conference, which requires 16 time slots (additional 7 time slots is available, but in this PBX only 4 conferences can be established),

– in CXS-*0424* PBX: 7 conferences with 4 participants in every conference, which requires 28 time slots,

– in *MAC-6400* PBX: 7 conferences with 4 participants in every conference and one 3PDY – it requires 31 time slots [in this situation all available in PBX resources are used].

License names:

      Conf.LIC-XXCP.ZZZ – license for XX conference participants

where:

      XX = maximal number of one conference participants,

      ZZZ = CCT (for *CCT-1668*), CXS (for *CXS-0424*) or MAC (for *MAC-6400*).

17.11 MobilePhone license

This service is accessible starting from firmware 4.07. This functionality for one subscriber is free of charge, for the next subscribers license purchasing is required. More information concerning this functionality can be found in chapter 4.3.1.5MobilePhone.

17.12 CallBack license

This service is accessible starting from firmware 4.07. This functionality for one subscriber is free of charge, for the next subscribers license purchasing is required. More information concerning this functionality can be found in chapter 4.3.1.6CallBack service.

17.13 IVR license

Service available from firmware (5.11 for CCT). It defines number of calls maintained simultaneously by XML.IVR protocol. Details concerning this solution are available for registered users on web page http://sdk.slican.pl

17.14 SMS license

This service is available from firmware version 6.21. By enter appropriate license code into field **LicSMS** you can define how many GSM trunks, installed in PBX, which can be used for sending and receiving SMSes <u>using third party applications</u>. It concerns trunks in sheet *Trrunks/GSM Settings.*

17.15 eSSL licence

Feature available from 6.40 firmware version. All PBXes that are linked with eSSL protocol, require **LicESSL** licence key. Licence key value must be minimum equal to the quantity of number of subscribers and accounts present in PBX. Lack of this licence (or quantity value lower than actual subscriber/account in PBX) will result in critical error. If Trial licence key will expire during system existence, CLIP presentation in calls between PBXes will be deactivated (similar as in firmware licence key expiration).

17.16 Time license – TRIAL

PBX user can tests all licensed functionality before final purchase licenses for functionality interesting him. To do this "TRIAL" license code should be entered into field **LicMax** - sheet *Global settings/License.* "TRIAL" type license is issued for specified time period (usually 1 month) and entering this license code unlock all licensed functionality with maximal values defined by hardware PBX configuration. As mentioned above "TRIAL" license is limited to the specified time period and after entering this license code into field **LicMax,** validity period is displayed on the right side of the screen. After expiration of "TRIAL" license PBX returns to the values determined by remain licenses.

17.17 Hardware PBX device number

Hardware device number means physical module installed on PBX controller card. In memory of this module, device serial number is stored. Licenses described in previous chapters are issued according PBX serial number. So they are active only for PBX equipped with sub module includes serial number used to generate license code. If PBX is equipped with other hardware device number or there is no equipped with any hardware device number, licenses are inactive. It means, that there is no possible to use CTI applications (CTIP protocol), PayMAN application, recording calls, as well there is no possible to establish VoIP calls.

Attention: *PBX is **locked** after removing sub module during operation. After installing hardware device number during PBX operation alarms are not deleted, until reset. PBX reset verify hardware device number.*

18 Replacing cards in PBX – activating (deactivating) cards

18.1 Accepting a new card

- turn the PBX off,
- insert card in an appropriate slot,
- turn the PBX on,

- download the configuration,
- accept the card in the **PABX/Cards** sheet, *(select(highlight) the new card. Click the right mouse button and open a menu, from which select **Accept** to add the card to the database),*
- provide the necessary settings for the card being activated,
- upload the configuration to the PBX.

ATTENTION!:
Due to dynamic card identification, MACVoIP card added while PBX is switched on visibly is equipped only in 4 channels. This will be updated after accepting card, uploading configuration and download configuration once more.

18.2 Removing card

- turn the PBX off and disconnect the power,
- remove the card,
- turn the PBX on,
- download the configuration,
- remove the card from the **PABX/Cards** sheet,*(in the sheet, select (highlight) the removed card. Click the right mouse button and open a menu, from which select **Remove** to remove the card),*
- upload the configuration to the PBX.

19 LAN and VoIP cards in Slican PBXes

1. **CCT1668 – PBX firmware <5.11**

PBXes CCT-1668 can be equipped in one of three types LAN card:
1. LAN module (card CCT1LAN + distribution frame in models S, L or panel in model EU) – LAN network interface enables local (LAN network) or remote (from internet) for PBX configuration from ConfigMAN and using CTI application.
2. VoIP 4 module (card CCT4VoIP + distribution frame in models S, L or panel in model EU) – enables 4 VoIP channels (4 simultaneous calls), WAN interface and all described above LAN module functionality
3. VoIP 8 module (card CCT8VoIP + distribution frame in models S, L or panel in model EU) – enables 4 VoIP channels (4 simultaneous calls), WAN interface and all described above. LAN module functionality

Main task of VoIP modules is establishing phone connections (using PBX CCT-1668) via local and internet network. This functionality is detailed described in next chapter.

VoIP modules LAN cards assures establishing phone calls using SIP protocol. User can decide if more important is call quality or occupied bandwidth, so LAN card are equipped in two audio codecs: G.711, GSM, G.729, while availability of two last codecs depends on types of installed DSP sub modules. Additionally VoIP modules can support functionality of call recording – availability of this functionality depends on type of installed DSP sub modules, it is presented by following table.

Sub-module		VoIP	VoIP codecs			REC
SM.DSP	SM.DSP-V	channels	G.711	GSM 6.10FR	G.729	channels
1	-	4	+	+	-	6
2	-	8	+	+	-	12
-	1	4	+	-	+	-
-	2	8	+	-	+	-
1	1	8	+	4	4	6

G.711 codec is international standard of audio coding in channels with 64kbps bandwidth, it concerning only audio coding. It is codec similar to standard and assures the same quality. In practice it should be assumed that total bandwidth for establishing single call (to occupy one channel on card) equals **110kbps** in each direction.

 G.729 codec is implementation of voice compression method based on 10ms voice samples. Using this codec fax signal cannot be transmitted reliable, so to do this codec G.711 or other transmission methods are recommended. This standard is used mainly for VoIP technology , because of small requirements concerning bandwidth (it requires only 8 Kb/s). Although for solution implemented in PBX it should be assumed, that total bandwidth needed for establishing one call (one channel bandwidth) is equal to **40 Kb/s** in every direction.

Bandwidth needed to VoIP functionality depends on kind of used audio codecs.

To VoIP cards three device types can be connected:
– system phones CTS.IP
– IP phones – any VoIP phones using protocols SIP or IAX
– computer software – so called soft-phone

ATTENTION!
All PBX Slican services are available from system phones CTS-202.IP. While using IP phones of other manufacturer or soft phones some cervices may be realized improper, ex. call transfer (FLASH).

2. CCT1668 – PBX firmware =>5.11
PBXes with firmware => 5.11 don't use VoIP-REC card. VoIP and call recording functionalities are realized by PBX controller. System supports 8 VoIP channels and by default 8 G.711a codecs. Due to increase functionality, it is possible to instal additional sub-module - SM.DSP-2V including 8 G.711μ and G.729 codecs. Available interfaces - LAN and WAN.

3. MAC-6400

MAC-6400 can be equipped in following types of cards with network interface:

➢ PBX controller – LAN interface enables local (from LAN network) or remote (from Internet) PBX configuration from inside *ConfigMAN* application as well as cooperation with CTI applications

> VoIP module (0VoIP card) – assures realization of voice calls in IP network. Number of channels and codec types depends on number of installed DSP modules (1DSP = 4 channels). Maximal number of sub-modules is equal to 4. It is possible to install sub-modules of type SM.DSP and SM.DSP-V. Available interfaces - LAN and WAN

> VoIP module (card 16VoIP, card 32VoIP) - 16 or 32 VoIP channels, codecs G.711 and G.729. Available interfaces - LAN and WAN. Cards are not upgradable.

> Call recording module EbdRec (card 8REC, card 16REC) – enables to register in PBX any calls. Available interface - LAN

For *MAC-6400* system, VoIP and EbdRec cards can be installed in any slot (assigned as SLOT 1 to SLOT 10) located on main board of master or slave shelf. In PBX (despite of its type) can be installed only one VoIP card.

4. IPL-265 and IPM-032

VoIP functionality can be realized by PBX controller or by specialized cards. If increasing channel number is required and resources located on controller are not enough or codec G.729 should be used PBX should be equipped in card 16, 32 or card tandem 32VoIP Master and 32VoIP Slave. After card installation number of VoIP is not a sum of available channels on controller and card, channels on controller are switched off.

> PBX controller – embedded VoIP and bdRec functionality. System provides 8'th VoIP channels and by default 8'th G.711a codecs. Network interfaces enable to local (LAN network) or remote (Internet) PBX configuration from inside *ConfigMAN* application and cooperation with CTI applications. Available interfaces - LAN and WAN

> VoIP module (card 16VoIP, card 32VoIP, tandem 2x32VoIP) - 16, 32 or 64 VoIP channels, codecs G.711 and G.729. Available interfaces - LAN and WAN. Cards are not upgradeable.

> PoE module – for IPM and IPL PBX family there is designed new hardware module - 4 ports PoE card (Power over Ethernet). This card was designed to simultaneous connection of PBX and system phones CTS.IP - manufactured by Slican – to the customer LAN network, another network devices compatible with this technology can be also connected (only one condition is no network devices between PBX and IP phone). In IPL system you can install maximally two cards, in IPM one card in slot 2 or 5.

IP resources configuration in PBXes

ConfigMAN software includes few sheets to configuration mentioned above modules:
– *IP Network / LAN Interfaces* – for configuration of all mentioned above modules,
– *IP Network / WAN Interfaces* – for WAN configuration,
– *IP Network / DNS Configuration* – for DNS configuration,
– *IP Network / DNS Configuration* – for PoE configuration,
– *IP Network / VoIP Configuration* – for VoIP parameters configuration,
– *IP Network*/*Router* – enables to router configuration.

Additionally sheet *IP Network / LAN Interfaces* is located in group *Basic settings*.

PoE card configuration

In sheet **IP Network/PoE Configuration** appropriate settings should be entered in following fields:
- **Cmt** – in this field comment for specified port should be entered
- **Pwr** – port power supply should be switched on or off (with limitation to 5W)
- **NetPrt** – in this field port working mode should be entered. This field depends on settings in sheet **IP Network/LAN Interfaces.** If in field **Md** LAN mode is set, default interface in field **NetPrt** is also LAN. When in field **Md** LAN+WAN (Router) is set, in field **NetPrt** interfaces LAN or WAN are possible to selection.

20 <u>VoIP functionality in Slican systems</u>

20.1 Basic information

Fig. 20.1.: Slican PBX with VoIP functionality – general view

VoIP subscribers – in PBX call established by these subscribers are routed in the same way as calls established by regular subscribers.

VoIP extension numbering -this numbering is identical as for regular extension numbering. VoIP extension number is treated by PSTN or GSM network as regular extension number, with the same number format.

PBX services - all PBX services are available from system phones CTS-202.IP.

Number of VoIP extension number – in Slican PBXes it is possible <u>to create two typed of VoIP extension numbers:</u>
- **extension numbers equipped with CTS.IP** ports destined **only** for system VoIP phones manufactured by Slican (system phone use it own communication protocol),
- **extension numbers equipped with regular VoIP phones (also softphones) -** ports destined for IP phones of other manufacturers and softphones, which uses SIP protocol.

<u>**VoIP trunks**</u> – it can be created <u>up to 100 logical VoIP trunks</u>. VoIP trunks are used for:
- serve incoming calls (anonymous - from any IP network user),
- serve calls in cooperation with VoIP providers,
- PBXes networking.

Configuring PBX in company's network – PBXes CCT-1668 equiped in VoIP card can work in one of following mode (defined in field **Md** – *Network card work mode* in sheet *IP Network / LAN interfaces)*:

1. LAN mode: In this mode PBX is regular node of corporate LAN network. To force PBX to works in such configuration, appropriate settings should be made in sheet *IP network/LAN Interfaces* and in router. These settings can defined as fixed or download-able from DHCP server. It is recommended to manual set network parameters.
2. LAN+WAN (Router) mode: In this mode there is possibility to establish VoIP connections.
   a) *NAT switched off* – no check-mark in field **NAT** – *Network addresses translation* in sheet *IP Network / WAN Interfaces*:

PBX works in LAN and WAN network, using router to packet prioritization. In this mode LAN and WAN interfaces are separated. No one packet from LAN network cannot appear on WAN interface and vice verse, no one packet from WAN interface cannot be addressed on LAN interface. Port forwarding function is unavailable due to no NAT operation, as well as there is no communication between LAN and WAN interfaces. Function of static interface and PBX services routing function is available. Among these services following can be enumerate:

- **Keeper** – commands addressed to Keeper are routing to LAN or WAN interface

- **VoIP** – calls to VoIP provider are routed to LAN or WAN interface

Available are also ACLs for services which require launching server:

- **CTI** – permission for connecting with opened port 5524 on LAN or WAN interface.

b) *NAT switched on* – set check-mark in field **NAT** – *Network addresses translation* in sheet *IP Network / WAN Interfaces*:

In this situation PBX is used also as router to Internet for local network connected to PBX [in this situation WAN port is connected to DSL modem, while LAN to switch with hosts connected (computers, IP phones)]. Available are all router functions.

- port forwarding - sheet *IP Network / Router / Port Forwarding*:

While ports forwarding, entries sequence is important, first matching rule is realized. It simplify port configuration because at the beginning of table there are entries concerning port ranges and exceptions are at the end of this table. Every port of WAN interface can be forwarded to the specified port of specified LAN device, the same for port range. Port range from WAN interface can be forwarded to the specified port range of LAN network device with specified IP address.

- static routing - sheet *IP Network / Router / Static routing*:

In static routing, packet arriving from any source (LAN/WAN) is checked (its source IP address) and routed to appropriate interface – LAN or WAN, or to device with appropriate IP address (gateway).

- firewall - sheet *IP Network / Router / Firewall*:

Firewall define what packets are accepted an what packets are rejected. Packets can be rejected according IP addresses or port numbers. Default rules for packets arriving from WAN interface are as follows:

- TCP ports 21-23 are rejected

- TCP port 5524 is accepted

- TCP ports 5523-5530 are rejected

- UDP ports 5523-5530 are rejected

Independently on working mode **VoIP** card can be also **DHCP** server for devices in LAN network. To use this functionality, check-mark should be entered in field **DHCPEn** – *Enable DHCP server* in sheet *IP Network / DHCP Server,* and next define range of addresses in field *AdrRg*.

ATTENTION!
In every card working mode to preserve QoS, marker TOS of sended packets is set to 16 (0x10) to obtain minimal possible delay of packets. Of course it is true if network devices serve marker TOS.

20.2 *PBX configuration in company's network – LAN mode*

In LAN mode Slican PBX is an element of local network in company and uses only physical LAN interface. To establish VoIP connections while connecting to LAN network, appropriate PBX configurable settings should be done in sheet **Devices/LAN card/LAN configuration** and in network settings (router).
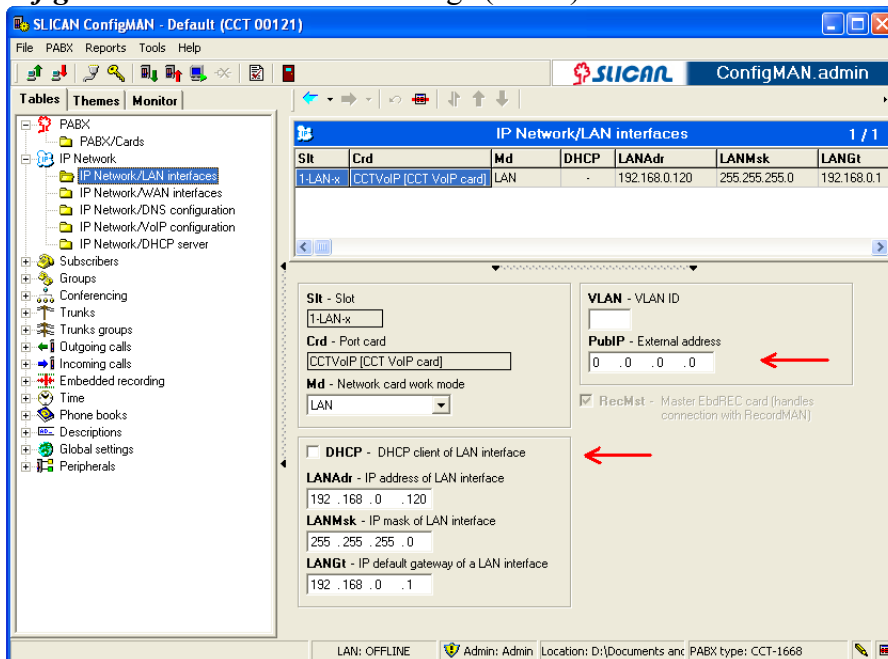
Fig. 20.2.: Sheet Devices/LAN card/LAN configuration

Basic settings in this mode are:
- **LANAdr** – IP address in local network,
- **LANMsk** - subnet mask,
- **LANGt** – default gateway address.

These settings can be defined as fixed or received automatically from DHCP server. In this second situation field **DHCP** should be checked. It is recommended to use manual network settings.

ATTENTION!
By default, after LAN card installation in field "LANAdr" value 192.168.0.248 is entered. This address enables to communicate with PBX via LAN network. It is enough to set IP address as 192.168.0.xxx an define subnet mask as 255.255.255.0 to further PBX configuration via LAN network.

Performing above activities enable log PBX to LAN network and using CTI application.
In the next step fields **DNS1** and **DNS2** should be entered. These fields defines addresses of DNS servers, preferred and alternative, which enable to connect to VoIP providers servers if in sheet *Trunks/VoIP settings*, in field **Adr** domain name is entered. Attempting to connect with provider, PBX firstly uses address defined in field **DNS1**. If server with this address is unavailable, PBX using address entered in field **DNS2**. After LAN card installation DNS by default are set to addresses DNS servers belonging to Polish National provider TPSA.

Fig. 21.3.: Working chart of PBX CCT-1668 in LAN network - with VoIP calls establishing possibility
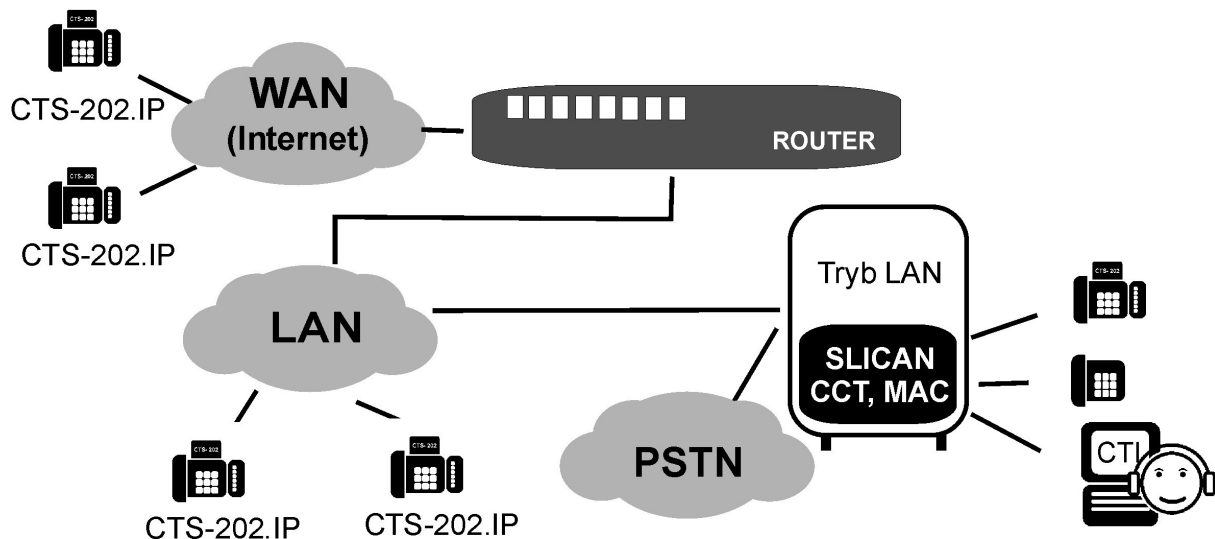
Next settings is external (public) address of network where PBX working. This address is obtained from computer network admin should be entered in field **PubIP**. Entering this address is necessarily to proper VoIP calls establishing. If this address changes in field **PubIP** 0.0.0.0 should be entered. In this situation possibility of PBX connection via 5525 port to keeper.slican.pl server should be observed. This server (managed by Slican company) is used to remote PBX managing.

PBX working in LAN mode is located behind NAT, according this – to enable VoIP transmission some UDP ports must be rerouted. For particular protocols following ports are used as default:
–   SIP and eSSL – 5060, 8100-8200
–   for system phones CTS-202.IP – 5522

Additionally it is important to assure QoS on PBX VoIP routers, especially transmission band, which depends on used audio codecs.
If there are VLAN used in local area network, where PBX is working, You can assign PBX to one of VLANs by entering VLAN identifier into VLAN field, according information from admin.

20.3 PBX configuration in company's network – LAN + WAN (Router) mode

In LAN+WAN mode PBX uses both ports LAN and WAN. It simultaneously connected to devices of internet provider (ex. DSL modem) as well as local company's. Connection to LAN network enables to using CTI application, managing PBX via LAN network and establish calls with VoIP users in LAN. This mode requires separate (assigned to PBX) public IP address. Connection to Internet provider devices can be realized in three ways:

1.   PBX works in LAN+WAN mode using the same Internet interface

Using above configuration PBX has no influence on packet sending in DSL modem. It is so, connection quality depends on available bandwidth and number of packets sending from LAN

Fig. 20.4.: Internet interface in LAN+WAN mode
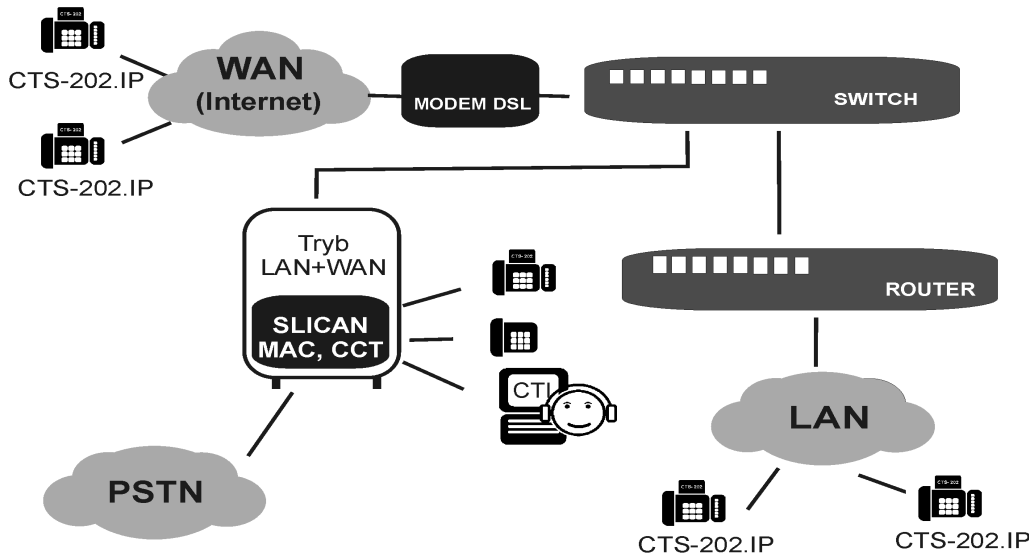
network. To eliminate reasons of poor calls quality, separate Internet connection should be used to establish VoIP calls in Slican PBX.

2. PBX using separated Internet interface for VoIP calls
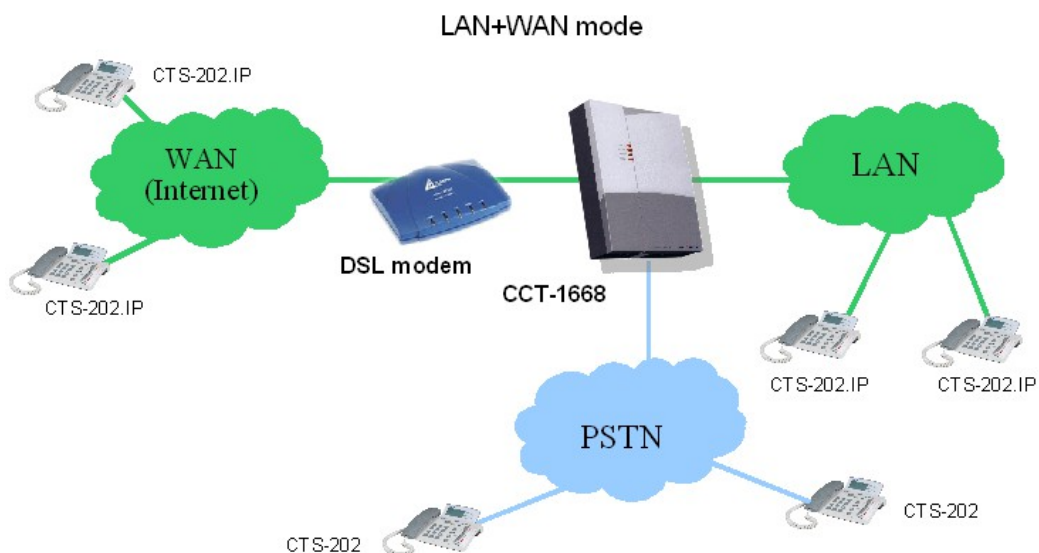
Fig. 20.5.: Using separate Internet connection to establish VoIP calls

To provide the best quality of service it is recommended to dedicate Internet connection to VoIP calls establishing. Company's local computer network should used separate Internet connection.

3. PBX as a router

PBX is a router for LAN network connected to it. Additionally is used as NAT (if checkmark is set in field NAT – *Network addresses translation* (masquerade) in sheet **IP Network / WAN interfaces**) for local network clients and assures QoS for VoIP calls if Internet connection is shared with another devices in LAN network. Working scheme is presented on following figure:
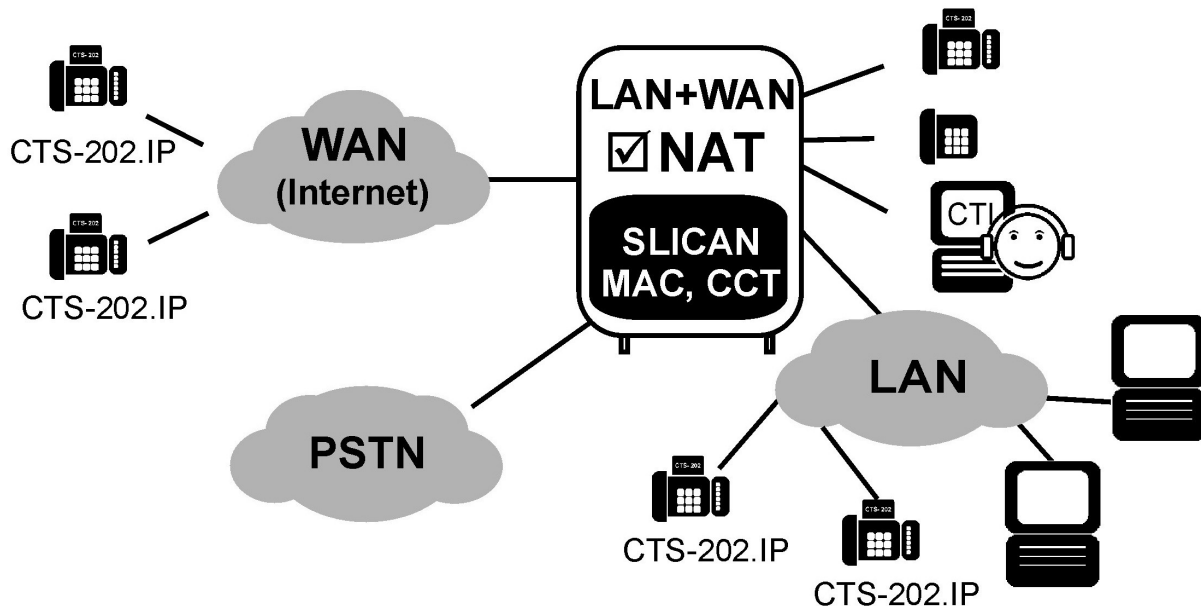
Fig. 20.6.: Working scheme for Slican PBX used as router

Appropriate settings for LAN+WAN (Router) mode can be done in following sheets:
– **IP Network/LAN Interfaces** – for LAN port (settings are identical as described above LAN mode)
– **IP Network/WAN Interfaces** – for WAN port
Basic settings for WAN port are as follows:
– **WANAdr** – IP address in WAN network,
– **WANMsk** - subnet mask,
– **WANGt** – default gateway address.
After checking **DHCP** field above mentioned settings are downloaded from DHCP server.

Figure 20.7.: Sheet IP Network / WAN Interfaces – LAN+WAN (Router) mode

Additionally in field **Bwth** – maximal bandwidth should be entered – possible to use for calls routed to Internet provider (upload). Setting this parameter is necessarily to assure QoS by PBX. If PBX works in network, where typical DSL service is used, ex. 512 kbps (download 512 kbps, upload 128 Kb/s) in field **Bwth** 128 kbps should be entered.
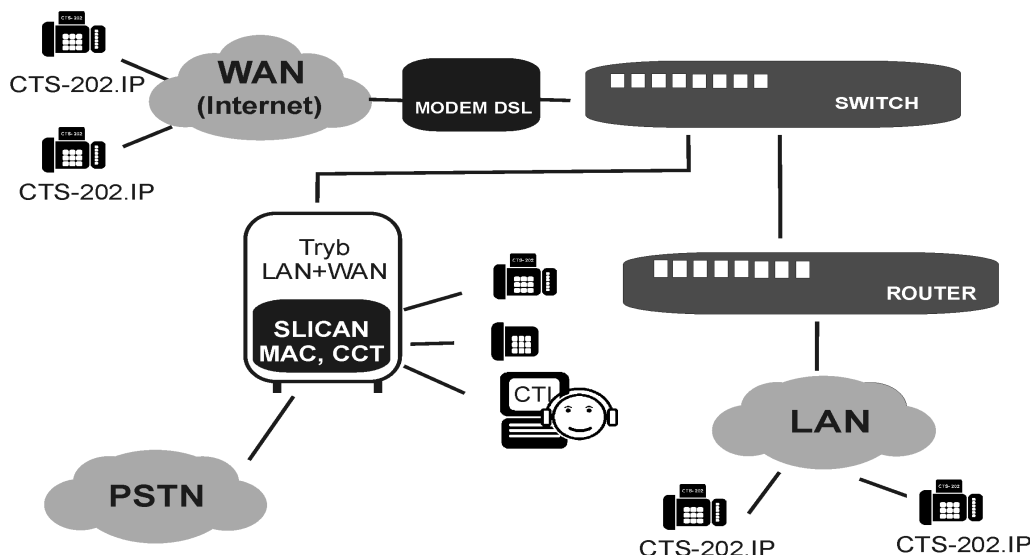
If there are used VLANs in LAN where PBX working, PBX can be assigned to one o them by entering appropriate number in field **VLAN,** according LAN admin suggestions.

20.4 Basic methods of using VoIP functionality in Slican PBXes

20.4.1 Operation with VoIP extension number

There is possible to establish following calls for VoIP extension numbers in Slican PBXes:
- internal – between themselves and with regular extension numbers (analogue, ISDN, CTS-202 and CTS-202.Plus),
- outgoing – to PSTN and GSM networks
- incoming – from PSTN and GSM networks

In Slican PBX:
1. Instal and accept new network card. Next, after clicking **Channel configuration** button in sheet **PABX / Cards** define how many channels – from available number of channels (depending on card type and acquired licenses) – can be occupied by:
   - subscribers, who use system phones CTS-202.IP
   - VoIP subscribers (in example it is subscriber using X-Lite software but it can be any IP phone maintaining SIP or IAX protocol).
   - VoIP trunks (in example VoIP trunks there are not used)
2. Select VoIP card working mode (LAN, LAN+WAN)
3. In sheet **Subscribers / Main settings** You can create new subscribers of VoIP type (number of subscribers possible to creation depends on licenses acquired)
4. Perform detailed configuration of VoIP subscribers in sheet ***Subscribers / VoIP settings***
5. In sheet **Devices / CTS / IP settings** configure PBX to cooperation with system phones CTS-202.IP.
6. Plan and configure settings and service privileges for every subscriber.
7. Configure incoming and outgoing (LCR table) calls and subscriber privileges for incoming and outgoing calls, for new subscribers (mentioned above activities should be done in the same way as for all other PBX subscribers – ex. equipped in analogue phones).

In CTS.IP phones you should:
1. Configure communication method with VoIP server (PBX).
2. Configure phone settings for local network.

ATTENTION!
Information concerning PBX configuration can be found in system phones installation and manual.

To use softphone as VoIP subscriber device following activities should be done:
1. Download from http://www.counterpath.net/ freeware version of X-Lite software (available for both Windows and Linux); there is only an example of available softphones.
2. Instal X-Lite software on PC.
3. Configure softphone to cooperate with PBX.
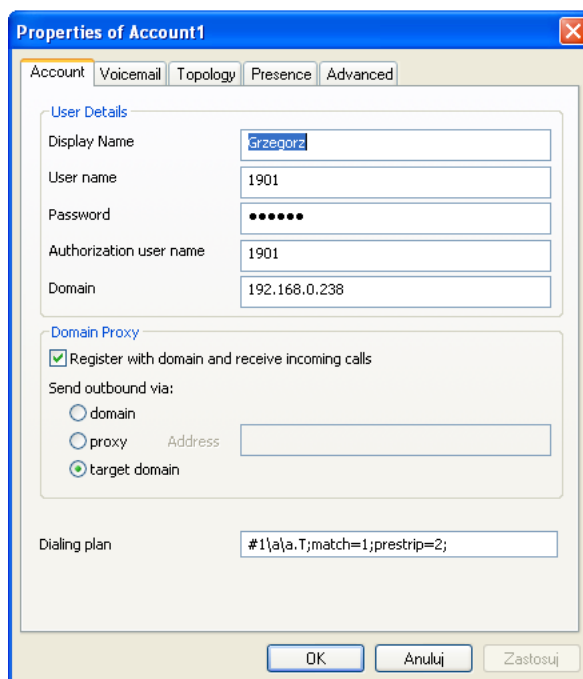
Fig. 20.8.: X-Lite configuration for operating with Slican PABX

To X-Lite configuration following data are needed:
- *Display Name* – displayed user name
- *User name* – extension number of VoIP subscriber
- *Password* – access code for VoIP server (PBX), defined in ConfigMAN in sheet **Subscribers / VoIP settings**
- *Authorization user name* - VoIP extension number
- *Domain* – public IP address of VoIP server (PBX)
- field *Register with domain and receive incoming calls* – should be checked

ATTENTION!
X-Lite software is possible free software with SIP protocol. Below table present possible software which can be used with Slican PBXes.

Software	Used protocol	Operating system	Web page address
X-Lite	SIP	Windows/Linux	http://www.counterpath.net/
Twinkle	SIP	Linux	http://www.twinklephone.com/
Zoiper	SIP, IAX	Windows/iOS /Android	http://www.zoiper.com/

20.4.2 Incoming calls from IP network, from anonymous users

Slican PBXes enable to receive incoming calls from any users of IP network, users who are not PBX subscribers (have no extension numbers). This solution enables to establish call between any external VoIP phone and any internal subscriber or group. Connection can be routed to DISA announcement, so caller can recall account and dial any external number if account exists in PBX and appropriate privileges are granted to this account.
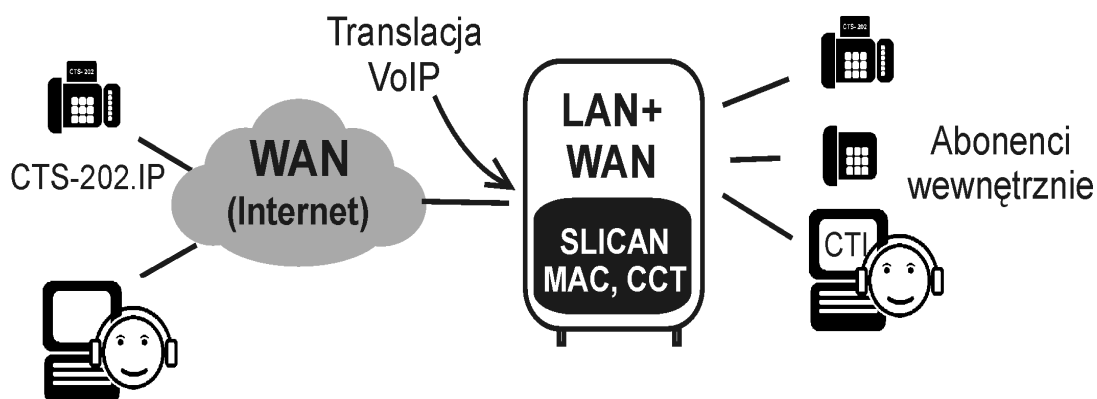
Fig. 20.9.: Calls from anonymous callers, routed via VoIP trunk

Following activities should be done to enable communication between anonymous subscribers:

1. In PBX create VoIP trunk for incoming calls an assign own number to this trunk. It can be the same number as already used for analogue or ISDN trunk.
   To set up VoIP trunk:
   - define numbers of channels in sheet **PABX / Cards** (menu **Channel configuration**),
   which should be used for calls
   - select transmission protocol (SIP or eSSL)
   - select possible audio codecs
   and
   - DTMF transmission mode (only for SIP protocol).

2. Program incoming calls routing for this trunk. If You use the same own number if for ISDN or analogue trunk, incoming calls are routed identically as for calls from PSTN and there is no necessarily to add new entry in Incoming Calls Table.

3. Inform company's customers about trunk own number (number to be dialed by customer), PBX IP address, selected transmission protocol, available audio codecs.

20.4.3 Cooperation with VoIP provider

By creating logical VoIP trunk and using services of VoIP provider we can establish calls to telephone networks all over the world (PSTN and GSM networks). This solution – presented on below figure – enables to drop down costs of calls (specially international and mobile calls costs).
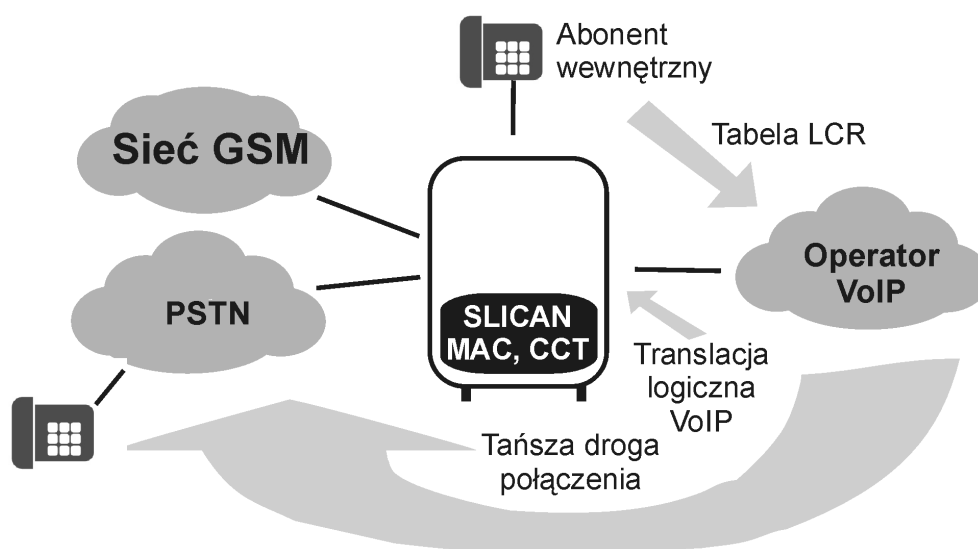
Fig. 20.10.: Cooperation of Slican PBX with VoIP providers

To establishing such calls VoIP trunk should be created ad set appropriate fields according information received from VoIP provider. Next table of outgoing calls should be programmed according needs. If from any extension number international number is dialed and appropriate entries in LCR table enable to dial this number, VoIP provider establish appropriate connection to PSTN network.

To enable PBX cooperation with VoIP provider and establish VoIP connections, in PBX following activities should be done:

1. Obtain configuration data from VoIP provider(IP address, domain – if it is different than IP address, login, access code and trunk own number)

2. Create VoIP trunk in PBX and assign to it own number obtained from VoIP provider

3. Configure VoIP trunk according data obtained from VoIP provider and define - in sheet **PABX / Cards** (menu **Channel configuration**) - how many channels should be used for calls

4. Program outgoing calls routing for this trunk (LCR table) and eventually incoming calls to route all international and mobile calls to VoIP provider (created VoIP trunk).

21 **PBX linking with eSSL protocol**

21.1 General information

Linking of Slican PBXes can be executed using VoIP or ISDN trunk. Linking using ISDN trunk can be executed via BRA or PRA interface, although linking range is limited while ISDN linking. For E1 trunk it is 300 meters and for 2B+D trunk maximal linking range is equal to 1000 meters.

Protocols for linking:

- eSSL (ISDN or VoIP) – full integration, obvious condition: all PBXes should use firmware version 6.20 or later.

- Simple linking (ISDN or VoIP) – older Slican PBXes or PBXes from third party manufacturers

21.1.1 eSSL system features

- **Charging**

External calls, both outgoing and incoming are registered in call buffer of this PBX where there is trunk 'participating' in external calls. Due to this functionality all call in eSSL network are transit to PSTN by one PBX. To charging cost of calls one BillingMAN database can be used.

- **Configuration**

It is simple, it is enough to properly calculate linking trunk group and all information necessarily to call configuration will be transferred between PBXes automatically.

- **Integration**

Subscriber statuses from PBXes included in eSSL system can be presented on other system phone buttons assigned to subscriber and in PhoneCTI/VoiceCTI/WebCTI applications. Specified subscribers in eSSL system can belong to one group.
Privileges (ex.: Access class, privileges to prefixes, call type, accessed nets etc) of specified subscribers from different PBXes are transferred in both directions during establishing call.

21.1.2 Assumptions

•eSSL (**e**xtended **S**lican **S**mart **L**ink) linking enables to link **Slican servers with firmware starting from 6.20**, after linking all servers are visible as one system with common extension numbering.

•In whole eSSL network extension number in PBXes cannot overlapping.
•Call realized via VoIP is realized using protocol SIP 2.0, so in boundary routers, port forwarding is required as in **ConfigMAN** settings (**IP Network/VoIP configuration**). By default it is port 5060(UDP)- for SIP signaling and 8100-8200(UDP) for RTP voice samples.
•In eSSL network **cannot be different type PBXes with the same serial number**.
•Total capacity of eSSL system: 3999 linked numbers + extension numbers from PBX. It should be analyzed individually for every PBX in system.
•Maximal number of linked PBXes: 15.
•Time required for transfer statuses of all subscribers after establishing link in no longer then 60 seconds
•Time required to transfer comments from all PBXes is no longer then 10 minutes (1 comment per 1 second)

21.2 eSSL network topology

21.2.1 Serial/Cassade
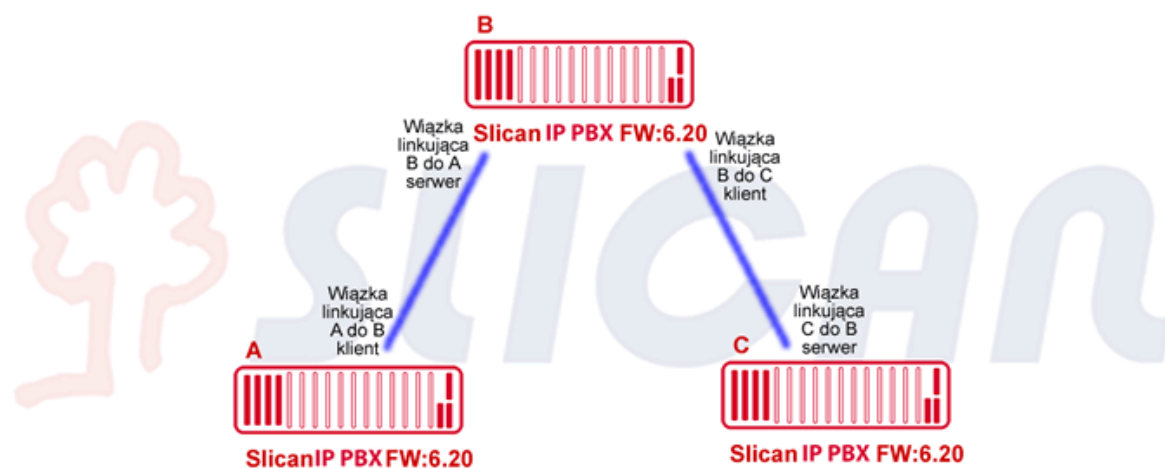
21.2.2 Each other topology
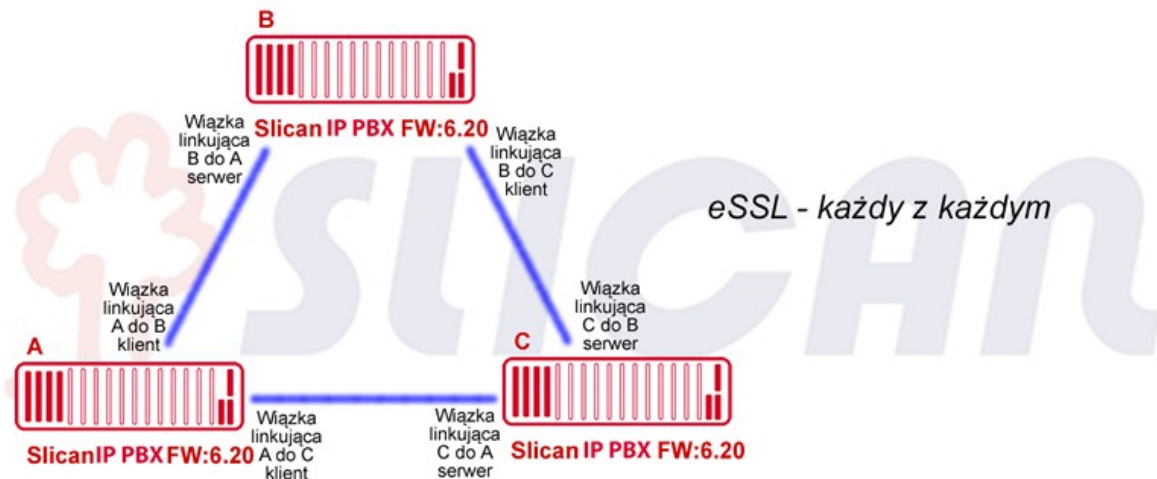
21.2.3 Mixed topology
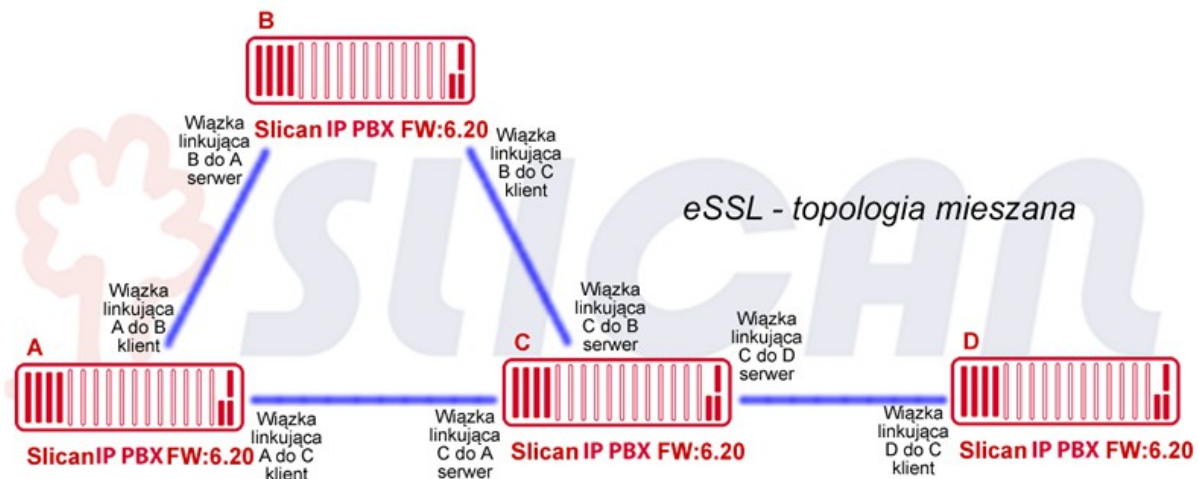
21.3 eSSL configuration

1. In sheet **Global settings/Advanced** switch on the field **LnkESSLEn –** to show option concerning **eSSL** system linking. This field should be checked in all PBXes included in **eSSL** system.
2. Create linking trunk group in sheet **Trunk groups/Linking** and define its type as **LnkTyp→eSSL**. According good practice it is recommended to assign unequivocal comment.

21.3.1 Connection viaVoIP

1. Create linking trunk (in sheet **Trunks/Main settings)**. According good practice it is recommended to assign it unequivocal comment. Field **OwnNo.** remain empty.
2. Configure trunk **Trunks/VoIP settings** in field **ConnTyp** should be selected, which link side trunk should be as well as complete fields **Adr/Log/Pas** by client side and fields **Log/Pas** on server side. It should be remembered, that authorization data on client and server side must be the same. In field **Ptl ->** eSSL should be selected.
3. In sheet **Trunks/Trunk groups** created trunk should be assigned to created earlier linking trunk groups.

From this moment eSSL linking trunk group is ready to operation.

21.3.2 Connection via ISDN

Fig. 21.1.: In every relation for eSSL (ISDN) – one PBX is NT device and second TE.

1.Select ISDN port/ports, which will be operate as linking trunks (**Trunks/Main settings)**. According good practice, it is recommended to assign it unequivocal comment. Field **OwnNo.** Should stay empty.

2.Trunks should be configured in sheet **Trunks/2B+D Settings/30B+D Settings** and select in field **DevTyp**, which connection side should be trunk (NT→TE). It should be remember, if one PBX is connected to public network via ISDN, this PBX should be NT for link. For linking via 2B+D trunk in sheet **Trunks/2B+D settings** option **SavMd** should be checked.

3.In sheet **Trunks/Trunk groups** trunk should be assigned to earlier created linking trunk group.

From this moment linking trunk eSSL is ready to operation.

21.3.3 Information synchronization in PBX network

After sending configuration to PBX a snchronization between then start automatically.

Synchronization start automatically after properly configuration of eSSL linking trunks group in all relations.

In **eSSL** there are 2 types of synchronized data:

1.**eSSL number base** (including extension number, number and type of PBX, number type, comments) – such synchronization is realized during first connection via eSSL or after change of configuration concerned with mentioned above data. This synchronization can be forced by erasing all content of table in sheet **eSSL Linking/Extension numbers** and uploading configuration to PBX. In summary data are synchronized if it is required.

2.**Port status** – it is done automatically and refresh together with any change of linked port status.

Synchronization status can be checked in sheet **Monitor/eSSL linking status.**

After number base synchronization in eSSL network configuration should be once more downloaded from PBX to actualization database in sheet **eSSL Linking/Extension numbers**. In monitor, in sheet **eSSL linking status** for every PBX in eSSL system following data are presented:

- PBX comments in field **PBXCmt**, in sheet **Global settings/Remain**
- PBX serial number
- Correctness status of linking trunk group **Stu**

- **Prg –** progress of eSSL number database synchronization.

Attention: Depending on PBX capacity eSSL PBX network, database synchronization progress may take several minutes.

21.3.4 Settings for alternative call maintenance

Example:

- Caller in A PBX dial number from C PBX. After dialing number eSSL system finds, that there is no possibility to call C PBX in scope of eSSL, because trunk group between A and C PBX is damaged.
- Automatically call will be established via PSTN.
- Such calls are possible only in scope of eSSL.

Fig. 21.2.: Alternative call maintenance , if connection with linking PBX is not possible.

If it is foreseen to use alternative call maintenance in eSSL PBX system, field **OutNo** in sheet **eSSL Linking/Main settings** should be fulfiled with number used to establish call if there are some problems with eSSL link. Two types of alternative call maintenance are possible:

- dialing with announcement – one number is dialed, then operator transfer this call to appropriate extension number,
- after dialing number, DISA announcement is played, then extension number is dialed (DTMF) **DISASrv**= On.

21.3.5 Automatic baypass call maintenance

Example:

- According below figure caller in A PBX dial extension number in C PBX. By default call will be established via trunk group between A an C PBXes.
- If direct link A-C is damaged, A PBX will find bypass route via B PBX.
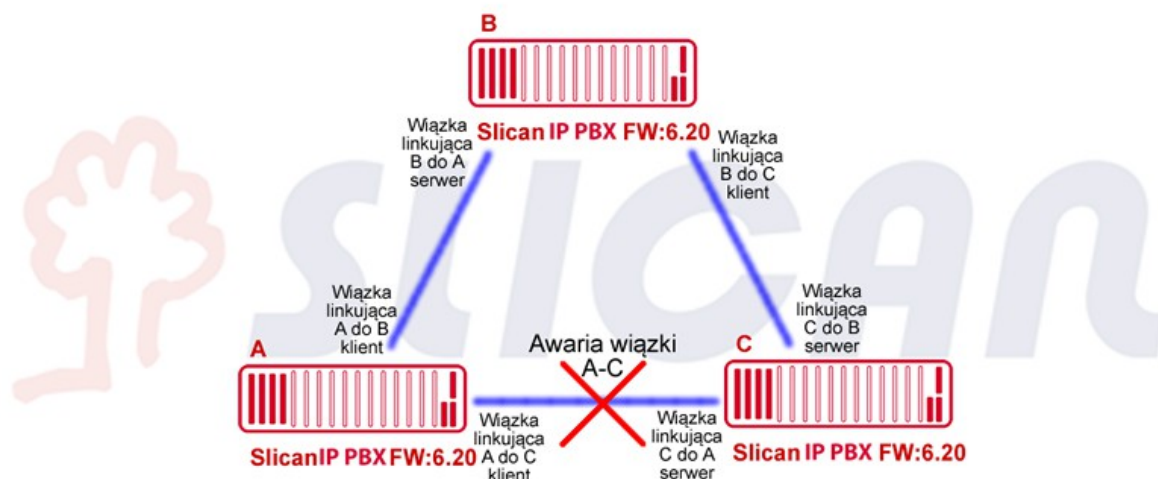
Function of automatic bypass call maintenance shouldn't be configured – it works automatically.

22 PBX linking with eSSLv2 protocol

22.1 Basic informations

eSSLv2 system is a expanded version of eSSL linking feature. Due to separation of synchronization elements (public phonebook, chat, SMS) and calls protocol, additional configuration steps are required to set up this system.

22.1.1 eSSLv2 system features

- **Charging**

External calls, both outgoing and incoming are registered in call buffer of this PBX where there is trunk 'participating' in external calls. Due to this functionality all call in eSSL network are transit to PSTN by one PBX. To charging cost of calls one BillingMAN database can be used.

- **Configuration**

It is simple, it is enough to properly configure linking trunk group and all information necessarily to call configuration will be transferred between PBXes automatically. Additionally, Internet connection should be active in all system elements (PBXes), where at synchronization server side, correct ports will be forwarded.

- **Integration**

Subscriber statuses from PBXes included in eSSLv2 system can be presented on other system phone buttons assigned to subscriber and in PhoneCTI/WebCTI applications.
Specified subscribers in eSSLv2 system can belong to one group.
Privileges (ex.: Access class, privileges to prefixes, call type, accessed nets etc) of specified subscribers from different PBXes are transferred in both directions during establishing call.
Client PBXes will use one, common public phonebook, placed in server PBX. It is possible to exchange messages between PhoneCTI applications inlinked PBXes. Static and dynamic SMS messages routing is also available in eSSLv2 linking.

22.1.2 Assumptions

- eSSLv2 (**e**xtended **S**lican **S**mart **L**ink version 2) linking enables to link **Slican servers with firmware starting from 6.40**, after linking all servers are visible as one system with common extension numbering and public phonebook.

- In whole eSSLv2 system, extension numbers in PBXes cannot overlap.

- In eSSLv2 system, one PBX should be choosed as server. It will contain system's public phonebook, available for client-PBXes.

- TCP 5529 port must be forwarded to PBX-server controller card IP address.

- Call realized via VoIP is realized using protocol SIP 2.0, so in boundary routers, port forwarding is required as in **ConfigMAN** settings (**IP Network/VoIP configuration**). By default it is port 5060(UDP)- for SIP signaling and 8100-8200(UDP) for RTP voice samples.

- In eSSLv2 system, different PBX models with similar serial number cannot be used.

- Time required for transfer statuses of all subscribers after establishing link in no longer then 60 seconds.
- Time required for exchanging comments between PBXes – up to 10 minutes (1 commect per second).
- Time required for exchanging public phonebook's records depends on its size and can take (during first synchronization) up to one hour.

22.2 *ESSLv2 network topology*

22.2.1 Required connections
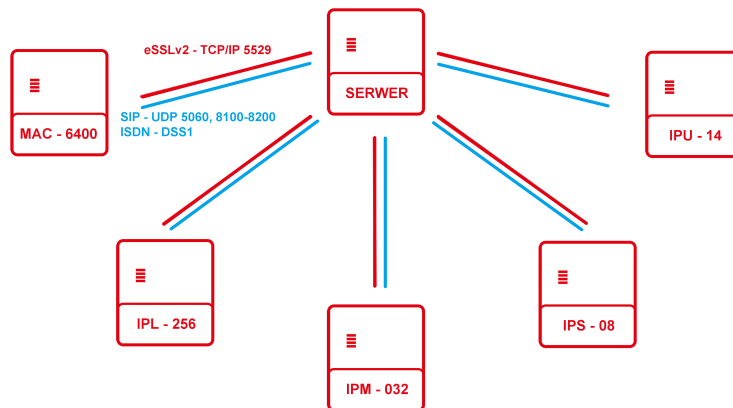
Above picture presents required connection between PBXes, for eSSLv2 system to work. Each PBX-client must be connected to PBX-server with TCP and UDP protocols. TCP protocol allows for synchronizatioon exchange and UDP, for handling calls.

22.2.2 Cross connections
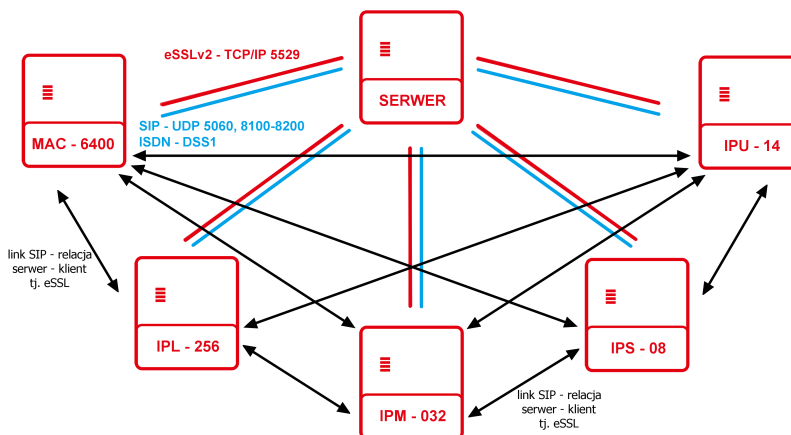
Above example is not required. Althrough, it secures PBXes against connection problems with PBX-server. In this example, each linking trunk should be assigned to separate trunks group. Each trunk group will have different **PBXNo.** Used.
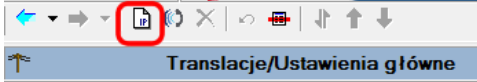
22.3 eSSLv2 configuration

1. In sheet **Global settings/Advanced** switch on the field **LnkESSLEn –** to show option concerning **eSSL** system linking. This field should be checked in all PBXes included in **eSSL** system.
2. In **eSSL linking/Main settings,** field **Ptl** needs to be set as Version 2.
3. **OprMd**. Defines, if configured PBX will serve as synchronization server for whole system.
4. For server – we choose which elements will be synchronized in linking system.
5. For client – set servers controller card IP address.
6. Create linking trunk group in sheet **Trunk groups/Linking** and define its type as **LnkTyp→eSSL**. According good practice it is recommended to assign unequivocal comment.
7. For PBX-server. In **PBXNo.** Field on selected trunk group, type serial number of the linked PBX (e.g. IPL000232.29), **SrvCnn** should not be checked.
8. In all cross connections between client PBXes, correct serial number "of the other side" is also required (**PBXNo.** field in Trunks groups/Linking).

22.3.1 Connection viaVoIP

It is necessarily to create pair of trunks (client-server) as on above figure:

1. Create linking trunk (in sheet **Trunks/Main settings)**. According good practice it is recommended to assign it unequivocal comment. Field **OwnNo.** remain empty.

2. Configure trunk **Trunks/VoIP settings** in field **ConnTyp** should be selected, which link side trunk should be as well as complete fields **Adr/Log/Pas** by client side and fields **Log/Pas** on server side. It should be remembered, that authorization data on client and server side must be the same. In field **Ptl -> eSSL** should be selected.

3. In sheet **Trunks/Trunk groups** created trunk should be assigned to created earlier linking trunk groups.
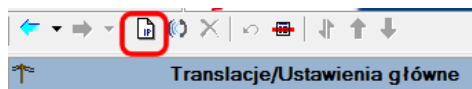
From this moment eSSLv2 linking trunk group is ready to operation.
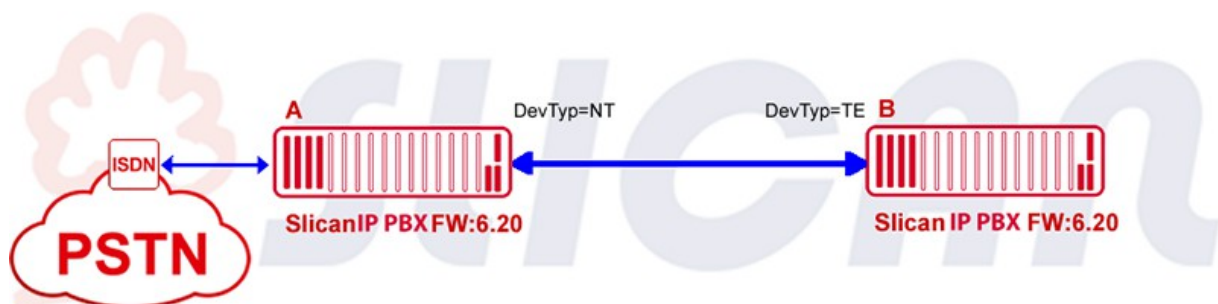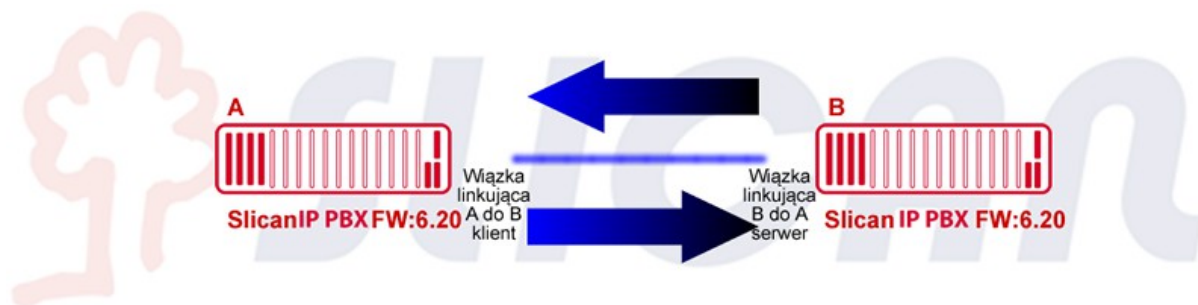
22.3.2 Connection via ISDN

1. Select ISDN port/ports, which will be operate as linking trunks (**Trunks/Main settings**). According good practice, it is recommended to assign it unequivocal comment. Field **OwnNo.** Should stay empty.
2. Trunks should be configured in sheet **Trunks/2B+D Settings/30B+D Settings** and select in field **DevTyp**, which connection side should be trunk (NT→TE). It should be

remember, if one PBX is connected to public network via ISDN, this PBX should be NT for link. For linking via 2B+D trunk in sheet **Trunks/2B+D settings** option **SavMd** should be checked.

3. In sheet **Trunks/Trunk groups** trunk should be assigned to earlier created linking trunk group.

From this moment linking trunk eSSLv2 is ready to operation.

22.4 eSSL trunks without trunk groups

Warning!
Described feature does not apply to ISDN linikng.

If features related to trunk groups does not interest us (call quantity limitations, trunks group recording, dynamic call routing, call charge), we can reate VoIP trunk without assigned trunk group. Fields, that idetify linked PBXes will be available in **Trunks/VoIP settings** .

22.5 Information synchronization in PBX network

After sending configuration to PBX a snchronization between then start automatically. Synchronization start automatically after properly configuration of eSSL linking trunks group in all relations.

In **eSSL** there are 2 types of synchronized data:

1.**eSSL number base** (including extension number, number and type of PBX, number type, comments) – such synchronization is realized during first connection via eSSL or after change of configuration concerned with mentioned above data. This synchronization can be forced by erasing all content of table in sheet **eSSL Linking/Extension numbers** and uploading configuration to PBX. In summary data are synchronized if it is required.

2.**Port status –** it is done automatically and refresh together with any change of linked port status.

Synchronization status can be checked in sheet **Monitor/eSSL linking status.**

After number base synchronization in eSSL network configuration should be once more downloaded from PBX to actualization database in sheet **eSSL Linking/Extension numbers**.

In monitor, in sheet **eSSL linking status** for every PBX in eSSL system following data are presented:

- PBX comments in field **PBXCmt**, in sheet **Global settings/Remain**
- PBX serial number
- Correctness status of linking trunk group **Stu**
- **Prg –** progress of eSSL number database synchronization.

Attention: Depending on PBX capacity eSSL PBX network, database synchronization progress may take several minutes.

22.5.1 Settings for alternative call maintenance

Example:

- Caller in A PBX dial number from C PBX. After dialing number eSSL system finds, that there is no possibility to call C PBX in scope of eSSL, because trunk group between A and C PBX is damaged.
- Automatically call will be established via PSTN.
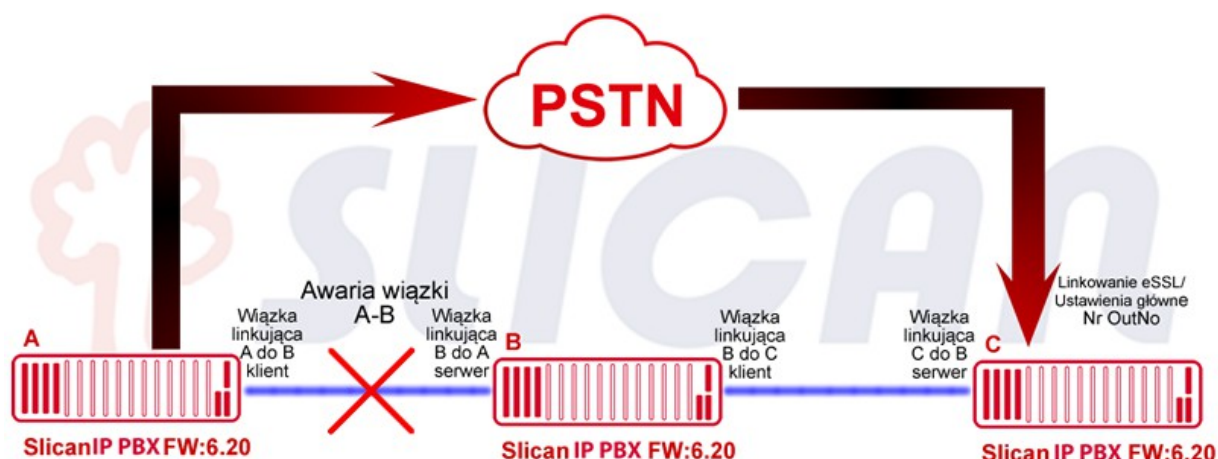- Such calls are possible only in scope of eSSL.

    If it is foreseen to use alternative call maintenance in eSSL PBX system, field **OutNo** in sheet **eSSL Linking/Main settings** should be fulfiled with number used to establish call if there are some problems with eSSL link. Two types of alternative call maintenance are possible:
- dialing with announcement – one number is dialed, then operator transfer this call to appropriate extension number,
- after dialing number, DISA announcement is played, then extension number is dialed (DTMF) **DISASrv**= On.

22.5.2 Automatic baypass call maintenance

Example:

- According below figure caller in A PBX dial extension number in C PBX. By default call will be established via trunk group between A an C PBXes.
- If direct link A-C is damaged, A PBX will find bypass route via B PBX.

Function of automatic bypass call maintenance shouldn't be configured – it works automatically.

23 **Basic linking**

While linking old PBXes (NCT-1248 or CCA-2720) with ISDN ports or PBXes from third party manufacturers via ISDN (QSIG compatible with ETS 300 172 protocol – basic dialing control) or VoIP (SIP) *Basic linking method* should be used.

23.1 Configuration of basic linking method

1. In sheet **Global settings/Advanced** switch on the field **LnkEn –** to show options concerning with basic linking.
2. Create main linking account in sheet **Subscribers/Settings** in field **Typ→Link.** (Linking account number should be outside of PBX numbering plan). This account represents privileges and registers calls of all subscribers from linked PBX.
3. Create linking trunk group in sheet **Trunk groups/Linking** and assign its type **LnkTyp→Basic**. According good practice it is recommended to assign it unequivocal comment.
4. Assign main linking account to create earlier trunk group in sheet **Trunk groups/Linking** field **LnkAcc.**
5. Define interpreting method for call incoming from linked PBX – field **LnkInc**:

    - **Route to extension number or LCR** – all calls from linked PBX are treated as internal or outgoing and routed to LCR table. They are not maintained by incoming rules.

    - **Route to incoming call table** – all calls from linked PBX are treated as external and are routed to incoming calls table.

    - **Route according CLIP** – PBX will be routed calls according number of digits in CLIP, some call will be treated as internal, some as external,
    - If it is necessarily to different presentation of calls to linked PBX, field **LnkPfx** should be fulfilled.
6. Assign prepared linking trunk (SIP or ISDN) to trunk group in sheet **Trunks/Trunk groups**, field **TrnGr.**

7. If there is necessity to individual differentiating privileges of subscribers from linked PBX, it is possible to create individual linking accounts. Extension number of individual linking account must be coherent with subscriber number in linked PBX. Additionally subscribers from linked PBX will be directly visible in billing system of main PBX, otherwise for one linking account – call cost of all subscribers will be charged for one account. Individual account can be created as main account with slightly difference – in sheet **Subscribers/Main settings** field **Trl** should be fulfilled with created earlier linking trunk group.
8. To route calls to linked PBX, in sheet **Outgoing calls/Extension number linking** following entries should be added:
    - **No.** - start of numbering plan from linking PBX, ex. 3 for numbering plan 3xx

    - **DgtNo.** - number of digits, after dialing which connection will be established, for above example it is 3.

    - **OutMth** linking trunk group linked with PBX
    - **DgtDl** – number of digits, which should be send to linked PBX (starting from the right).

Attention: Despite of transmission medium, it is recommended to use eSSL (extended Slican Smart Link) protocol to linking PBXes, it is protocol developed in Slican company. This protocol enables to obtain full functionality of linking.

24 **Sheet list in application**

Users more familiar with Slican PBX can define particular settings in sheets. To select specified sheet, thematically grouped „sheets tree" is used. There are sheet concerning the same theme in each sheet group. Below is presented view of all sheets in group.

PABX

PABX/CARDS

IP NETWORK

IP NETWORK/LAN INTERFACES
IP NETWORK/WAN INTERFACES
IP NETWORK/DNS CONFIGURATION
IP NETWORK/VOIP CONFIGURATION
IP NETWORK/DHCP SERVER
IP NETWORK/ROUTER
> IP NETWORK/ROUTER/PORT FORWARDING
> IP NETWORK/ROUTER/STATIC ROUTING
> IP NETWORK/ROUTER/FIREWALL

SUBSCRIBERS

SUBSCRIBERS/MAIN SETTINGS
SUBSCRIBERS/SERVICES
SUBSCRIBERS/MOBILE PHONES
SUBSCRIBERS/CALL FORWARDING
SUBSCRIBERS/SERVICE SETTINGS
SUBSCRIBERS/PROTECTION AGAINST SERVICES
SUBSCRIBERS/SPECIAL SERVICES
SUBSCRIBERS/INTERNAL CALLS
SUBSCRIBERS/OUTGOING CALLS
SUBSCRIBERS/INCOMING CALLS
SUBSCRIBERS/VOICEMAIL
SUBSCRIBERS/EMBEDDED RECORDING
SUBSCRIBERS/CHARGES
SUBSCRIBERS/VOIP SETTINGS

GROUPS

GROUPS/UPRINGING
> GROUPS/UPRINGING/SETTINGS
> GROUPS/UPRINGING/AGENTS
GROUPS/CYCLIC
> GROUPS/CYCLIC/SETTINGS
> GROUPS/CYCLIC/AGENTS
GROUPS/LINEAR
> GROUPS/LINEAR/SETTINGS
> GROUPS/LINEAR/AGENTS

GROUPS/ACD
  GROUPS/ACD/SETTINGS
  GROUPS/ACD/AGENTS

CONFERENCING

CONFERENCING/GROUPS
  CONFERENCING/GROUPS/SETTINGS
  CONFERENCING/GROUPS/PARTICIPANTS
CONFERENCING/ROOMS
  CONFERENCING/ROOMS/SETTINGS
  CONFERENCING/ROOMS/PARTICIPANTS

TRUNKS

TRUNKS/MAIN SETTINGS
TRUNKS/TRUNKS GROUPS
TRUNKS/FREE MINUTES
TRUNKS/2B+D SETTINGS
TRUNKS/30B+D SETTINGS
TRUNKS/POTS SETTINGS
TRUNKS/GSM SETTINGS
TRUNKS/VoIP SETTINGS

TRUNKS GROUPS

TRUNKS GROUPS/SETTINGS

TRUNKS GROUPS/LINKING

TRUNKS GROUPS/TRUNKS LISTS

OUTGOING CALLS

OUTGOING CALLS/LCR
  OUTGOING CALLS/LCR/SETTINGS
  OUTGOING CALLS/LCR/CHARGES
OUTGOING CALLS/INTERNAL NUMBER LINKING
OUTGOING CALLS/SHORT NUMBERS
OUTGOING CALLS/PERMITED PREFIXES
OUTGOING CALLS/PROVIDERS

INCOMING CALLS

INCOMING CALLS/SETTINGS
  INCOMING CALLS/SETTINGS/MAIN
  INCOMING CALLS/SETTINGS/ADDITIONAL
INCOMING CALLS/ANNOUNCEMENTS
INCOMING CALLS/INFOLINES
INCOMING CALLS/API INFOLINES
INCOMING CALLS/STATIC PATH ROUTING
INCOMING CALLS/CLIP CATEGORIES

EMBEDDED

RECORDING

EMBEDDED RECORDING/RECORDMAN.CLIENT ACCOUNTS
EMBEDDED RECORDING/ACCESS LEVELS

TIME

TIME/PABX WORKING MODES
TIME/WORKING MODES TIME-TABLE
TIME/CONTROL SCHEDULE
TIME/HOLIDAYS

NOTIFICATIONS

NOTIFICATIONS/MESSAGES
NOTIFICATIONS/SERVICE ALERTS
NOTIFICATIONS/EVENTS

PHONE BOOKS

PHONE BOOKS/PUBLIC
PHONE BOOKS/PRIVATE
PHONE BOOKS/AREAS NAMES

DESCRIPTIONS

DESCRIPTIONS/SUBSCRIBERS NETS
DESCRIPTIONS/SUBSCRIBERS ACCESS CLASSES
DESCRIPTIONS/SUBSCRIBERS PRIVILEGES
DESCRIPTIONS/PICKUP GROUPS
DESCRIPTIONS/COMPANIES

GLOBAL SETTINGS

GLOBAL SETTINGS/CHARGES
GLOBAL SETTINGS/VOICEMAIL
GLOBAL SETTINGS/ACCESS CLASSES
GLOBAL SETTINGS/REMAIN
GLOBAL SETTINGS/LICENSES
GLOBAL SETTINGS/PABX ACCESS ACCOUNTS
GLOBAL SETTINGS/SYSTEM COMMANDS

PERIPHERALS

PERIPHERALS/ CTS
    PERIPHERALS/CTS/SETTINGS
    PERIPHERALS/CTS/ CONTROLLERS
    PERIPHERALS/CTS/BUTTONS
    PERIPHERALS/CTS/IP SETTINGS
PERIPHERALS/DOORPHONES
PERIPHERALS/AUDIO
PERIPHERALS/SENSORRELAY

25 Acronyms used in application

In ConfigMAN application to field description there are acronyms used. These acronyms are created according first letters of field name described functionality of this field. In below table there are all acronyms together with sheet name where such field is located.

Acronym	Name	Sheet
BrdStu	Board Status	PABX/Cards
Cmt	Comment	PABX/Cards
Crd	Card	PABX/Cards
CrdStu	Card Status	PABX/Cards
ExtNo.	Extension number	PABX/Cards
MnBrd	Main Board	PABX/Cards
Prt	Port	PABX/Cards
PrtCfg	Port Configuration	PABX/Cards
RecChn	Recording Channels	PABX/Cards
Shf	Shelf	PABX/Cards
Slt	Slot	PABX/Cards
Trl	Terminal	PABX/Cards
Trn	Trunk	PABX/Cards
VoIPChn	VoIP Channels	PABX/Cards
Ver	Firmware version	PABX/Cards
Slt	Slot	IP Network/LAN interfaces
Crd	Card	IP Network/LAN interfaces
Md	Mode	IP Network/LAN interfaces
DHCP	DHCP	IP Network/LAN interfaces
LANAdr	LAN address	IP Network/LAN interfaces
LANMsk	LAN mask	IP Network/LAN interfaces
LANGt	LAN gateway	IP Network/LAN interfaces
VLAN	VLAN	IP Network/LAN interfaces
PubIP	Public IP	IP Network/LAN interfaces
RecMst	Recording master	IP Network/LAN interfaces
Slt	Slot	IP Network/WAN interfaces
Crd	Card	IP Network/WAN interfaces
Md	Mode	IP Network/WAN interfaces
DHCP	DHCP	IP Network/WAN interfaces
WANAdr	WAN address	IP Network/WAN interfaces
WANMsk	WAN mask	IP Network/WAN interfaces
WANGt	WAN gateway	IP Network/WAN interfaces
VLAN	VLAN	IP Network/WAN interfaces
Bwth	Bandwidth	IP Network/WAN interfaces
NAT	NAT	IP Network/WAN interfaces
EnPtl	Protocol enable	IP Network/WAN interfaces
Slt	Slot	IP Network/DNS configuration
Crd	Card	IP Network/DNS configuration
AutDNS	Automatic DNS	IP Network/DNS configuration

DNS1	DNS1	IP Network/DNS configuration
DNS2	DNS2	IP Network/DNS configuration
Slt	Slot	IP Network/VoIP configuration
Crd	Card	IP Network/VoIP configuration
PrtSIP	Port SIP	IP Network/VoIP configuration
Rlm	Realm	IP Network/VoIP configuration
PrtIAX	Port IAX	IP Network/VoIP configuration
PrtRTP	Port RTP	IP Network/VoIP configuration
Slt	Slot	IP Network/DHCP server
Crd	Card	IP Network/DHCP server
DHCPEn	DHCP enable	IP Network/DHCP server
AdrRg	Address register	IP Network/DHCP server
Cmt	Comment	IP Network/Router/Port forwarding
Ptl	Protocol	IP Network/Router/Port forwarding
WANPrt	WAN port	IP Network/Router/Port forwarding
LANAdr	LAN address	IP Network/Router/Port forwarding
LANPrt	LAN port	IP Network/Router/Port forwarding
Cmt	Comment	IP Network/Router/Static routing
NetAdr	Network address	IP Network/Router/Static routing
NetMsk	Network mask	IP Network/Router/Static routing
RtTo	Route to	IP Network/Router/Static routing
GtAdr	Gateway address	IP Network/Router/Static routing
Cmt	Comment	IP Network/Router/Firewall
Ptl	Protocol	IP Network/Router/Firewall
SrcAdr	Source address	IP Network/Router/Firewall
SrcMsk	Source mask	IP Network/Router/Firewall
SrcPrt	Source port	IP Network/Router/Firewall
DstAdr	Destination address	IP Network/Router/Firewall
DstMsk	Destination mask	IP Network/Router/Firewall
DstPrt	Destination port	IP Network/Router/Firewall
Act	Action	IP Network/Router/Firewall
Cmt	Comment	Subscribers/Main settings
ExtNo.	Extension number	Subscribers/Main settings
Trl	Terminal	Subscribers/Main settings
Typ	Type	Subscribers/Main settings
TrlNt	Terminal note	Subscribers/Main settings
Cmt	Comment	Subscribers/Services
ExtNo.	Extension number	Subscribers/Services
PIN	PIN	Subscribers/Services
Lck	Lock	Subscribers/Services

Acc	Account	Subscribers/Services
HAcc	Account activation hour	Subscribers/Services
AlrC	Alarm Call	Subscribers/Services
AlrTi	Alarm Time	Subscribers/Services
Cmt	Comment	Subscribers/Mobile Phone
ExtNo.	Extension number	Subscribers/Mobile Phone
MblPh	Mobile Phone	Subscribers/Mobile Phone
MsdSMS	Message SMS	Subscribers/Mobile Phone
MsdMbl	Messabe Mobile	Subscribers/Mobile Phone
ClbEn	BallBack Enable	Subscribers/Mobile Phone
MblEn	Mobile Phone Enable	Subscribers/Mobile Phone
Rng	Ringing	Subscribers/Mobile Phone
RngTi	Ringing Time	Subscribers/Mobile Phone
RngDND	Ringing DND	Subscribers/Mobile Phone
Cmt	Comment	Subscribers/Call forwarding
ExtNo.	Extension number	Subscribers/Call forwarding
DND	Do Not Disturb	Subscribers/Call forwarding
DNDMsg	DND Message	Subscribers/Call forwarding
IT	I'm there	Subscribers/Call forwarding
FwdNo.	Forwarding number	Subscribers/Call forwarding
Fwd	Forwarding	Subscribers/Call forwarding
FwdTyp	Forwarding type	Subscribers/Call forwarding
FwdStu	Forwarding status	Subscribers/Call forwarding
FwdTi	Forwarding time	Subscribers/Call forwarding
Cmt	Comment	Subscribers/Services settings
ExtNo.	Extension number	Subscribers/Services settings
UrgEn	Urgent enable	Subscribers/Services settings
UrgRng	Urgent ringing	Subscribers/Services settings
OutFwd	Outgoing forwarding	Subscribers/Services settings
TrEn	Transit enable	Subscribers/Services settings
COvr	Call override	Subscribers/Services settings
DNDMsg	DND Message	Subscribers/Services settings
FwdLck	Forwarding Lock	Subscribers/Services settings
SrvLck	Service Lock	Subscribers/Services settings
Cmt	Comment	Subscribers/Protection against services
ExtNo.	Extension number	Subscribers/Protection against services
PtcPck	Protection PickUp	Subscribers/Protection against services
PtcOvr	Protection Override	Subscribers/Protection against services
Cmt	Comment	Subscribers/Special services
ExtNo.	Extension number	Subscribers/Special services

Cmp	Company	Subscribers/Special services
PubPB	Public phone book	Subscribers/Special services
CLIRi	CLIR internally	Subscribers/Special services
RegRng	Regular ringing	Subscribers/Special services
DT	Data transmision	Subscribers/Special services
KnLck	Knocking Lock	Subscribers/Special services
ErnLck	External Lock	Subscribers/Special services
CTI	CTI	Subscribers/Special services
TouchCTI	TouchCTI	Subscribers/Special services
PayMAN	PayMAN	Subscribers/Special services
TAPI	TAPI	Subscribers/Special services
Cmt	Comment	Subscribers/Internal calls
ExtNo.	Extension number	Subscribers/Internal calls
PckGr	PickUp group	Subscribers/Internal calls
CLIPi	CLIP internal	Subscribers/Internal calls
CLIRi	CLIR internal	Subscribers/Internal calls
Net	Network	Subscribers/Internal calls
AwdNet	Allowed nets	Subscribers/Internal calls
Cmt	Comment	Subscribers/Outgoing calls
ExtNo.	Extension number	Subscribers/Outgoing calls
CLIP	CLIP	Subscribers/Outgoing calls
PrfHL	Preferred Hotline	Subscribers/Outgoing calls
PhLck	Phone lock	Subscribers/Outgoing calls
DNDLck	DND lock	Subscribers/Outgoing calls
DayCls	Day access class	Subscribers/Outgoing calls
NgtCls	Night access class	Subscribers/Outgoing calls
PrvCls	Private access class	Subscribers/Outgoing calls
DayPrg	Day privilege	Subscribers/Outgoing calls
NgtPrg	Night privilege	Subscribers/Outgoing calls
PrvPrg	Private privilege	Subscribers/Outgoing calls
CLIR	CLIR	Subscribers/Outgoing calls
CLIRt	CLIR temporary	Subscribers/Outgoing calls
VIP	VIP	Subscribers/Outgoing calls
Cmt	Comment	Subscribers/Incoming calls
ExtNo.	Extension number	Subscribers/Incoming calls
CLIPRcv	CLIP receive	Subscribers/Incoming calls
OphTyp	Orphan type	Subscribers/Incoming calls
COLR	COLR	Subscribers/Incoming calls
BckExt	Backup Extension	Subscribers/Incoming calls
ShwPrx	Show proxy	Subscribers/Incoming calls

ExtNo.	Extension number	Subscribers/Voicemail
Cmt	Comment	Subscribers/Voicemail
ClrOld	Clear Old	Subscribers/Voicemail
FSK	FSK	Subscribers/Voicemail
MaxCap	Maximal capacity	Subscribers/Voicemail
MsgCt	Message count	Subscribers/Voicemail
ExtNo.	Extension number	Subscribers/Embedded recording
Cmt	Comment	Subscribers/Embedded recording
EbdRec	Embedded recording	Subscribers/Embedded recording
HiRec	High recording priority	Subscribers/Embedded recording
AccLvl	Axxess level	Subscribers/Embedded recording
Cmt	Comment	Subscribers/Charges
ExtNo.	Extension number	Subscribers/Charges
LmtTyp	Limit type	Subscribers/Charges
Rchrg	Recharge	Subscribers/Charges
Lmt	Limit	Subscribers/Charges
Blnc	Balance	Subscribers/Charges
Cmt	Comment	Subscribers/VoIP settings
ExtNo.	Extension number	Subscribers/VoIP settings
Typ	Type	Subscribers/VoIP settings
Ptl	Protocol	Subscribers/VoIP settings
Cdc	Codec	Subscribers/VoIP settings
Pas	Password	Subscribers/VoIP settings
DTMF	DTMF	Subscribers/VoIP settings
DfnUlg	Define unlogged	Subscribers/VoIP settings
PrmCnv	Permission to convert	Subscribers/VoIP settings
ExtNo.	Extension number	Groups/Upringing/Settings
Cmt	Comment	Groups/Upringing/Settings
Res	Reservation	Groups/Upringing/Settings
GrTyp	Group type	Groups/Upringing/Settings
Net	Network	Groups/Upringing/Settings
Pck	Pickup	Groups/Upringing/Settings
KnLck	Knocking Lock	Groups/Upringing/Settings
QCap	Queue capacity	Groups/Upringing/Settings
MaxC	Maximal call	Groups/Upringing/Settings
MaxFwd	Maximal forwarding	Groups/Upringing/Settings
MaxWt	Maximal waiting	Groups/Upringing/Settings
ExtNo.	Extension number	Groups/Upringing/Agents
Cmt	Comment	Groups/Upringing/Agents
Sub	Subscriber	Groups/Upringing/Agents

Ti	Time	Groups/Upringing/Agents
NBsy	Next while busy	Groups/Upringing/Agents
ExtNo.	Extension number	Groups/Cyclic/Settings
Cmt	Comment	Groups/Cyclic/Settings
Res	Reservation	Groups/Cyclic/Settings
Net	Network	Groups/Cyclic/Settings
Pck	Pickup	Groups/Cyclic/Settings
KnLck	Knocking Lock	Groups/Cyclic/Settings
QCap	Queue capacity	Groups/Cyclic/Settings
MaxC	Maximal call	Groups/Cyclic/Settings
MaxFwd	Maximal forwarding	Groups/Cyclic/Settings
MaxWt	Maximal waiting	Groups/Cyclic/Settings
ExtNo.	Extension number	Groups/Cyclic/Agents
Cmt	Comment	Groups/Cyclic/Agents
Sub	Subscriber	Groups/Cyclic/Agents
Ti	Time	Groups/Cyclic/Agents
ExtNo.	Extension number	Groups/Linear/Settings
Cmt	Comment	Groups/Linear/Settings
Res	Reservation	Groups/Linear/Settings
Net	Network	Groups/Linear/Settings
Pck	Pickup	Groups/Linear/Settings
KnLck	Knocking Lock	Groups/Linear/Settings
QCap	Queue capacity	Groups/Linear/Settings
MaxC	Maximal call	Groups/Linear/Settings
MaxWt	Maximal waiting	Groups/Linear/Settings
ExtNo.	Extension number	Groups/Linear/Agents
Cmt	Comment	Groups/Linear/Agents
Sub	Subscriber	Groups/Linear/Agents
Ti	Time	Groups/Linear/Agents
ID	ID	Groups/ACD/Settings
ExtNo.	Extension number	Groups/ACD/Settings
Cmt	Comment	Groups/ACD/Settings
GrTyp	Group type	Groups/ACD/Settings
GrMng	Group manager	Groups/ACD/Settings
PIN	PIN	Groups/ACD/Settings
AutBRB	Automatic be right back	Groups/ACD/Settings
Net	Network	Groups/ACD/Settings
MaxWt	Maimal witing time	Groups/ACD/Settings
Res	Reservation	Groups/ACD/Settings
Rng	Ringing	Groups/ACD/Settings

ID	ID	Groups/ACD/Agents
ExtNo.	Extension number	Groups/ACD/Agents
Cmt	Comment	Groups/ACD/Agents
ExtNo.	Extension number	Conferencing/Groups/Settings
Cmt	Comment	Conferencing/Groups/Settings
Net	Net	Conferencing/Groups/Settings
CnfCsd	Conference custodian	Conferencing/Groups/Settings
Fwd	Forwarding	Conferencing/Groups/Settings
CDrn	Call duration	Conferencing/Groups/Settings
RngDrn	Ringing duration	Conferencing/Groups/Settings
ExtNo.	Extension number	Conferencing/Groups/Participants
Cmt	Comment	Conferencing/Groups/Participants
Ppt	Participant	Conferencing/Groups/Participants
ExtNo.	Extension number	Conferencing/Rooms/Settings
Cmt	Comment	Conferencing/Rooms/Settings
Net	Net	Conferencing/Rooms/Settings
CnfCsd	Conference custodian	Conferencing/Rooms/Settings
WtMel	Waiting melody	Conferencing/Rooms/Settings
Fwd	Forwarding	Conferencing/Rooms/Settings
CDrn	Call duration	Conferencing/Rooms/Settings
PIN	PIN	Conferencing/Rooms/Participants
ExtNo.	Extension number	Conferencing/Rooms/Participants
Cmt	Comment	Conferencing/Rooms/Participants
Ppt	Participant	Conferencing/Rooms/Participants
Cmt	Comment	Trunks/Man settings
OwnNo.	Own number	Trunks/Man settings
Trl	Terminal	Trunks/Man settings
TrlNt	Terminal note	Trunks/Man settings
Stu	Status	Trunks/Man settings
Cmt	Comment	Trunks/Trunks groups
Trl	Terminal	Trunks/Trunks groups
TrnGr	Trunk group	Trunks/Trunks groups
Cmt	Comment	Trunks/Free minutes
Trl	Terminal	Trunks/Free minutes
PreOut	Previous outgoing	Trunks/Free minutes
PreFre	Previous free	Trunks/Free minutes
OutCnt	Outgoing counter	Trunks/Free minutes
FreMin	Free minutes	Trunks/Free minutes
FreCnt	Free counter	Trunks/Free minutes
RstDay	Reset day	Trunks/Free minutes

Cmt	Comment	Trunks/2B+D settings
Trl	Terminal	Trunks/2B+D settings
Chn	Channel	Trunks/2B+D settings
ChrMth	Charging method	Trunks/2B+D settings
SavMd	Saving mode	Trunks/2B+D settings
SncLck	Synchronization lock	Trunks/2B+D settings
FrcInb	Force inband	Trunks/2B+D settings
DevTyp	Device type	Trunks/2B+D settings
L2	Level 2	Trunks/2B+D settings
SCLIP	SCLIP	Trunks/2B+D settings
SCOLP	SCOLP	Trunks/2B+D settings
E/O	En-block/overlap	Trunks/2B+D settings
CmtOut	Comment outgoing	Trunks/2B+D settings
CmtIn	Comment incoming	Trunks/2B+D settings
Cmt	Comment	Trunks/30B+D settings
Trl	Terminal	Trunks/30B+D settings
Chn	Channel	Trunks/30B+D settings
ChrMth	Charging method	Trunks/30B+D settings
SncLck	Synchronization lock	Trunks/30B+D settings
L2	Level 2	Trunks/30B+D settings
SCLIP	SCLIP	Trunks/30B+D settings
SCOLP	SCOLP	Trunks/30B+D settings
E/O	En-block/overlap	Trunks/30B+D settings
FrcInb	Force inband	Trunks/30B+D settings
CmtOut	Comment outgoing	Trunks/30B+D settings
CmtIn	Comment incoming	Trunks/30B+D settings
Cmt	Comment	Trunks/POTS settings
Trl	Terminal	Trunks/ POTS settings
DlMd	Dialing mode	Trunks/ POTS settings
TrnRel	Trunk Release	Trunks/ POTS settings
ChrMd	Charging mode	Trunks/ POTS settings
ChrOfs	Charging offset	Trunks/ POTS settings
FSK	FSK	Trunks/ POTS settings
DlDis	Dialing disable	Trunks/ POTS settings
BsyDis	Busy disable	Trunks/ POTS settings
FlsEn	Flash enable	Trunks/ POTS settings
ID	ID	Trunks/GSM settings
Cmt	Comment	Trunks/GSM settings
Trl	Terminal	Trunks/GSM settings
Stu	Status	Trunks/GSM settings

PIN	PIN	Trunks/GSM settings
Cmt	Comment	Trunks/VoIP settings
Trl	Terminal	Trunks/VoIP settings
Chn	Channel	Trunks/VoIP settings
Ptl	Protocol	Trunks/VoIP settings
Cdc	Codec	Trunks/VoIP settings
Adr	Address	Trunks/VoIP settings
Dmn	Domain	Trunks/VoIP settings
Log	Login	Trunks/VoIP settings
Pas	Password	Trunks/VoIP settings
DTMF	DTMF	Trunks/VoIP settings
CmtOut	Comment outgoing	Trunks/VoIP settings
CmtIn	Comment incoming	Trunks/VoIP settings
DDI	DDI	Trunks/VoIP settings
Cmt	Comment	Trunks groups/Settings
Prc	Price	Trunks groups/Settings
PthRt	Path routing	Trunks groups/Settings
MaxInc	Maximal incoming	Trunks groups/Settings
MaxOut	Maximal outgoing	Trunks groups/Settings
SPP	SPP	Trunks groups/Settings
RecInc	Record incoming	Trunks groups/Settings
RecOut	Record outgoing	Trunks groups/Settings
Cmt	Comment	Trunks groups/Linking
LnkTyp	Linking type	Trunks groups/Linking
LnkAss	Linking account	Trunks groups/Linking
LnkInc	Linking incoming	Trunks groups/Linking
LnkPfx	Linking prefix	Trunks groups/Linking
Cmt	Comment	Trunks groups/Lists
TrnGr	Trunk group	Trunks groups/Lists
OutHL	Outgoing Hotline	Outgoing calls
AnlTi	Analisys time	Outgoing calls
TrTi	Transit time	Outgoing calls
HLOfs	Hotline offset	Outgoing calls
Ln	Line	Outgoing calls
LckTi	Lock time	Outgoing calls
ACMMDis	ACMM disable	Outgoing calls
Cmt	Comment	Outgoing calls/LCR/Settings
RtMth	Routin method	Outgoing calls/LCR/Settings
No.	Number	Outgoing calls/LCR/Settings
DgtNo.	Digit number	Outgoing calls/LCR/Settings

Wday	Week Day	Outgoing calls/LCR/Settings
StrH	Start hour	Outgoing calls/LCR/Settings
ACls	Access class	Outgoing calls/LCR/Settings
DlNo.	Dialed number	Outgoing calls/LCR/Settings
FreMin	Free minutes	Outgoing calls/LCR/Settings
EbdRec	Embedded recording	Outgoing calls/LCR/Settings
Cmt	Comment	Outgoing calls/LCR/Charges
No.	Number	Outgoing calls/LCR/Charges
FreMin	Free minutes	Outgoing calls/LCR/Charges
CDrn	Call duration	Outgoing calls/LCR/Charges
ChrTyp	Charging type	Outgoing calls/LCR/Charges
ChrIni	Charging init	Outgoing calls/LCR/Charges
ChrPrc	Charging price	Outgoing calls/LCR/Charges
ChrPer	Charging period	Outgoing calls/LCR/Charges
Cmt	Comment	Outgoing calls/Internal number linking
No.	Number	Outgoing calls/Internal number linking
DgtNo.	Digit Number	Outgoing calls/Internal number linking
OutMth	Outgoing method	Outgoing calls/Internal number linking
DgtDl	Digit to dial	Outgoing calls/Internal number linking
Cmt	Comment	Outgoing calls/Short numbers
PrvC	Private call	Outgoing calls/Short numbers
SpAcd	Speed access call	Outgoing calls/Short numbers
ExtNo.	Extension number	Outgoing calls/Short numbers
ACls	Access class	Outgoing calls/Short numbers
DlNo.	Dialed number	Outgoing calls/Short numbers
OutMth	Outgoing method	Outgoing calls/Short numbers
Cmt	Comment	Outgoing calls/Permited prefixes
No.	Number	Outgoing calls/Permited prefixes
Prg	Privileges	Outgoing calls/Permited prefixes
Name	Name	Outgoing calls/Providers
PrdNo.	Provider number	Outgoing calls/Providers
PrdTyp	Provider type	Outgoing calls/Providers
OphNo.	Orphan number	Incoming calls
OphTi	Orphan time	Incoming calls
EmMd	Emergency mode	Incoming calls
AccTrn	Account trunk	Incoming calls
PthRt	Path routing	Incoming calls
MelRng	Melody ringing	Incoming calls
DlFre	Dialing free	Incoming calls
DlBsy	Dialing busy	Incoming calls

KnPer	Knocking period	Incoming calls
InfChn	Infoline channels	Incoming calls
MelTyp	Melody type	Incoming calls
VceDis	Voice disable	Incoming calls
ErnRng	External ringing	Incoming calls
ItlRng	Internal ringing	Incoming calls
DphRng	Doorphone ringing	Incoming calls
VIPRng	Vip ringing	Incoming calls
Cmt	Comment	Incoming calls/Settings/Main
DlNo.	Dialed number	Incoming calls/Settings/Main
WrkMd	Working mode	Incoming calls/Settings/Main
TrnGr	Trunk group	Incoming calls/Settings/Main
DISA	DISA	Incoming calls/Settings/Main
ExtNo.	Extension number	Incoming calls/Settings/Main
CTyp	Call type	Incoming calls/Settings/Main
DisBsy	Disable when busy	Incoming calls/Settings/Main
BsyAct	Busy action	Incoming calls/Settings/Main
Fax	Fax	Incoming calls/Settings/Main
DTMF	DTMF	Incoming calls/Settings/Main
AwdNet	Allowed network	Incoming calls/Settings/Main
Cmt	Comment	Incoming calls/Settings/Additional
TrnGr	Trunk group	Incoming calls/Settings/Additional
DlNo.	Dialed number	Incoming calls/Settings/Additional
WrkMd	Wprking mode	Incoming calls/Settings/Additional
MaxInc	Maximal incoming	Incoming calls/Settings/Additional
CdDrn	Call duration	Incoming calls/Settings/Additional
EbdRec	Embedded recording	Incoming calls/Settings/Additional
VIP	VIP	Incoming calls/Settings/Additional
PthRt	Path routing	Incoming calls/Settings/Additional
ID	ID	Incoming calls/Announcements
Cmt	Comment	Incoming calls/Announcements
Ti	Time	Incoming calls/Announcements
Stu	Status	Incoming calls/Announcements
Cnl	Control	Incoming calls/Announcements
Cmt	Comment	Incoming calls/Infolines
Ann	Announcement	Incoming calls/Infolines
WtTi	Waiting time	Incoming calls/Infolines
RptTi	Repeating time	Incoming calls/Infolines
DefAct	Default action	Incoming calls/Infolines
DgtAct	Digit action	Incoming calls/Infolines

ExtNo.	Extension number	Incoming calls/Infolines
BsyAct	Busy action	Incoming calls/Infolines
AnsAct	Answer action	Incoming calls/Infolines
Cmt	Comment	Incoming calls/API Infolines
Ann	Announcement	Incoming calls/API Infolines
FTmo	Finishing time	Incoming calls/API Infolines
RptTi	Repeating time	Incoming calls/API Infolines
WtMel	Waiting melody	Incoming calls/API Infolines
RngMel	Ringing melody	Incoming calls/API Infolines
DgtNo	Digit number	Incoming calls/API Infolines
AwdHsh	Hash allowed	Incoming calls/API Infolines
NTmo	Next time	Incoming calls/API Infolines
SrvAdr	Server sddress	Incoming calls/API Infolines
SrvTmo	Server time	Incoming calls/API Infolines
ActID	Action ID	Incoming calls/API Infolines
DefAct	Default action	Incoming calls/API Infolines
BsyAct	Busy action	Incoming calls/API Infolines
AnsAct	Answer action	Incoming calls/API Infolines
Cmt	Comment	Incoming calls/Static path routing
RtTyp	Routing type	Incoming calls/Static path routing
CLIP	CLIP	Incoming calls/Static path routing
DlNo.	Dialed number	Incoming calls/Static path routing
Act	Action	Incoming calls/Static path routing
No.	Number	Incoming calls/Static path routing
EbdRec	Embedded recording	Incoming calls/Static path routing
HardKey	Hardware key	Embedded recording
ErnPas	External Password	Embedded recording
RecCpr	Recording compression	Embedded recording
PrvRec	Private call rerecording	Embedded recording
DISARec	DISA recording	Embedded recording
VmlRec	Voice mail recording	Embedded recording
CnfRec	Conference recording	Embedded recording
IncMsg	Incoming message	Embedded recording
OutMsg	Outgoing message	Embedded recording
ItlMsg	Internal message	Embedded recording
CTSLCD	CTS LCD	Embedded recording
DisEnc	Disable encryption	Embedded recording
Log	Login	Embedded recording/RecordMAN.client accounts
HardKey	Hardware key	Embedded recording/RecordMAN.client accounts

Cpy	Copy	Embedded recording/RecordMAN.client accounts
AccTyp	Access type	Embedded recording/RecordMAN.client accounts
AccLvl	Access level	Embedded recording/RecordMAN.client accounts
FtrStr	Filter dtring	Embedded recording/RecordMAN.client accounts
AccTi	Access time	Embedded recording/RecordMAN.client accounts
ID	ID	Embedded recording/Access levels
Cmt	Comment	Embedded recording/Access levels
SncRTC	Real time clock synchronization	Time
MngMd	Managing mode	Time
WrkMd	Working mode	Time
EndMnu	Manual end	Time
NoAut	No automatic	Time
ID	ID	Time/PABX working modes
Cmt	Comment	Time/PABX working modes
PhLck	Phone lock	Time/PABX working modes
NgtMd	Night mode	Time/PABX working modes
OphNo.	Orphan number	Time/PABX working modes
Wday	Week Day	Time/Work mode schedule
MdTi	Mode time	Time/Work mode schedule
WrkMd	Working mode	Time/Work mode schedule
Cpt	Caption	Time/Work mode schedule
Cmt	Comment	Time/Control schedule
Wday	Day of week	Time/Control schedule
Ti	Time	Time/Control schedule
RlyAct	Relay action	Time/Control schedule
Rly	Relay	Time/Control schedule
Cmt	Comment	Time/Holidays
DMY	Day, month, year	Time/Holidays
SMSTrn	SMS trunk	Notifications
ChrAcc	Charged account	Notifications
PBXCmt	PBX Name	Notifications/Messages
LogCmt	Log comment	Notifications/Messages
Cmt	Comment	Notifications/Service alerts
No.	Number	Notifications/Service alerts
NtfTyp	Notification type	Notifications/Service alerts
SysFlr	System failure	Notifications/Service alerts

PrtFlr	Port failure	Notifications/Service alerts
Cfg	Configuration	Notifications/Service alerts
EvtLog	Event log	Notifications/Service alerts
RecFlr	Recording failure	Notifications/Service alerts
Rst	Reset	Notifications/Service alerts
LogLck	Login lock	Notifications/Service alerts
EmPwr	Emergency power	Notifications/Service alerts
RemA	Remote access	Notifications/Service alerts
AccLmt	Account limit	Notifications/Service alerts
Cmt	Comments	Notifications/Events
EvtTyp	Event type	Notifications/Events
EvtSrc	Event source	Notifications/Events
No.	Number	Notifications/Events
NtfTyp	Notification type	Notifications/Events
SMS	SMS text	Notifications/Events
ID	ID	Phone books/Public
Name	Name	Phone books/Public
No.	Number	Phone books/Public
Cmp	Company	Phone books/Public
EdLck	Edition lock	Phone books/Public
VIP	VIP	Phone books/Public
Pfx	Prefix	Phone books/Area names
Name	Name	Phone books/Area names
ID	ID	Descriptions/Subscribers nets
Cmt	Comment	Descriptions/Subscribers nets
ID	ID	Descriptions/Subscribers access classes
Cmt	Comment	Descriptions/Subscribers access classes
ID	ID	Descriptions/Subscribers privileges
Cmt	Comment	Descriptions/Subscribers privileges
ID	ID	Descriptions/PickUp groups
Cmt	Comment	Descriptions/PickUp groups
ID	ID	Descriptions/Companies
Cmt	Comment	Descriptions/Companies
EvtRgs	Evrents registering	Global settings/Charges
DisInE	Disable incoming established	Global settings/Charges
DisInM	Disable incoming missed	Global settings/Charges
DisOutE	Disable outgoing established	Global settings/Charges
DisOutM	Disable outgoing missed	Global settings/Charges

DisItl	Disable internal	Global settings/Charges
AlrBuf	Alarm buffer	Global settings/Charges
VAT	VAT	Global settings/Charges
ExtNo.	Extension number	Global settings/Voicemail
EnPIN	Enable PIN	Global settings/Voicemail
IntPfx	International prefix	Global settings/Remain
ArPfx	Area prefix	Global settings/Remain
AlrSMS	Alarm SMS	Global settings/Remain
AccRel	Account release	Global settings/Remain
SlpEn	Sleep enable	Global settings/Remain
SlpTi	Sleep time	Global settings/Remain
TAPIPas	TAPI password	Global settings/Remain
RemA	Remote access	Global settings/Remain
PgnAnn	Paging announcement	Global settings/Remain
LnkEn	Linking enable	Global settings/Remain
CmdEn	Command enable	Global settings/Remain
XMLEn	XML enable	Global settings/Remain
LicFrm	Licence firmware	Global settings/Licences
LicVoIP	Licence VoIP	Global settings/Licences
LicEbdRec	Licence embedded recording	Global settings/Licences
LicInf	Licence infolines	Global settings/Licences
LicCnf	Licence conference	Global settings/Licences
LicCTI	Licence CTI	Global settings/Licences
LicTAPI	Licence TAPI	Global settings/Licences
LicPayMAN	Licence PayMAN	Global settings/Licences
LicMax	Licecce maximal	Global settings/Licences
Cmt	Comment	Global settings/PABX access accounts
Log	Login	Global settings/PABX access accounts
Pss	Password	Global settings/PABX access accounts
CM	ConfigMAN	Global settings/PABX access accounts
BM	BillingMAN	Global settings/PABX access accounts
Cd	Code	Global settings/PABX access accounts
PM	PayMAN	Global settings/PABX access accounts
Ann	Announcement	Global settings/PABX access accounts
Sw	Switching	Global settings/PABX access accounts
PBXCmt	PABX comment	Global settings/Notices
LogCmt	Login comment	Global settings/Notices
Cmd	Command	Global settings/System commands
ExtNo.	Extension number	Peripherials/CTS/Settings
Trl	Terminal	Peripherals/CTS/Settings

Cmt	Comment	Peripherals/CTS/Settings
DevTyp	Device type	Peripherals/CTS/Settings
Name	Name	Peripherals/CTS/Settings
OutPfx	Outgoing prefix	Peripherals/CTS/Settings
Lng	Language	Peripherals/CTS/Settings
Hph	Headphone	Peripherals/CTS/Settings
AutAns	Auto Answer	Peripherals/CTS/Settings
AutDl	Auto Dial	Peripherals/CTS/Settings
LCDStu	LCD status	Peripherals/CTS/Settings
LCDErr	LCD error	Peripherals/CTS/Settings
LnlLck	Line lock	Peripherals/CTS/Settings
Cnt	Contrast	Peripherals/CTS/Controlers
RngTon	Ringing tone	Peripherals/CTS/Controlers
RngVol	Ringing volume	Peripherals/CTS/Controlers
SpkLsp	Speakerphone loudspeaker	Peripherals/CTS/Controlers
HdsLsp	Handset laudseaker	Peripherals/CTS/Controlers
HphMic	Headphones microphone	Peripherals/CTS/Controlers
CnsSrv	Console service	Peripherals/CTS/Buttons
MAC	MAC	Peripherals/CTS/IP settings
Cdc	Codec	Peripherals/CTS/IP settings
CdcLck	Codec lock	Peripherals/CTS/IP settings
ExtNo.	Extension number	Peripherals/Doorphones
Cmt	Comment	Peripherals/Doorphones
Trl	Terminal	Peripherals/Doorphones
TrlNt	Terminal note	Peripherals/Doorphones
Stu	Status	Peripherals/Doorphones
Net	Network	Peripherals/Doorphones
EbdRec	Embedded recording	Peripherals/Doorphones
AccLvl	Access level	Peripherals/Doorphones
AwdNet	Allowed nets	Peripherals/Doorphones
DlTon	Dialing tone	Peripherals/Doorphones
ExtNo.	Extension number	Peripherals/Audio
Cmt	Comment	Peripherals/Audio
Trl	Terminal	Peripherals/Audio
TrlNt	Terminal note	Peripherals/Audio
Typ	Type	Peripherals/Audio
Net	Network	Peripherals/Audio
EbdRec	Embedded recording	Peripherals/Audio
AccLvl	Access level	Peripherals/Audio
Cmt	Comment	Protocol XML/Accounts

Log	Login	Protocol XML/Accounts
Pss	Password	Protocol XML/Accounts
IVR	IVR	Protocol XML/Accounts
Cmt	Comment	Protocol XML/IVR
Log	Login	Protocol XML/IVR
Acc	Account	Protocol XML/IVR
EbdRec	Embedded recordings	Protocol XML/IVR
Chn	Channel	Protocol XML/IVR