

ConfigMAN 6.55

Programming manual

Slican IPS-08

Slican IPU-14

Slican IPM-032

Slican IPL-256

Slican CCT-1668

Slican MAC-6400

Slican MAC-ZERO

Release 1.8

for
firmware version 6.55



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1 Preparing the PBX for programming

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(x86)\Slican\ConfigMAN

The USB drivers are located in:

C:\Program Files(x86)\Slican\\Drivers\USB

Database directory for the MAN system (MultiMAN, ConfigMAN, BilingMAN):in

C:\Users\Public\Documents\Slican\pabx data\Default. It is visible within window status bar and can be changed from **Tools\Applications setup** tab.

Above mentioned paths are derived from Windows 10 OS, 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 are not supported..

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

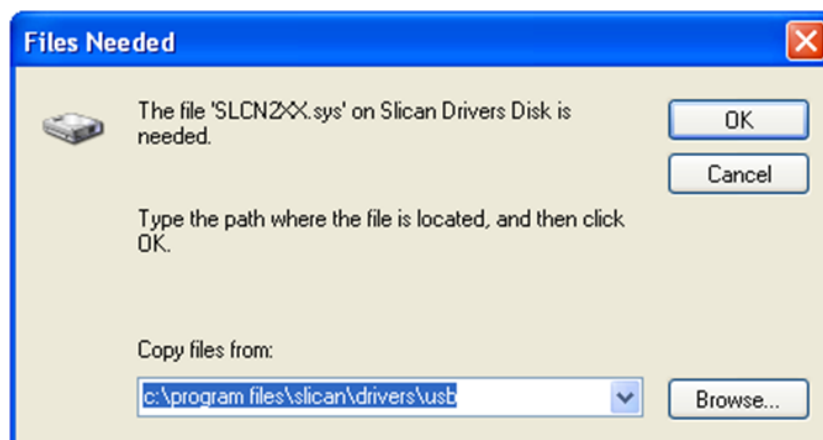
By default, after installing **ConfigMAN**, the USB drivers are located in: **C:\Program Files\Slican\Drivers\USB** or **C:\Program Files(x86)\Slican\Drivers\USB**.

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.: Windows 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**.



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**.

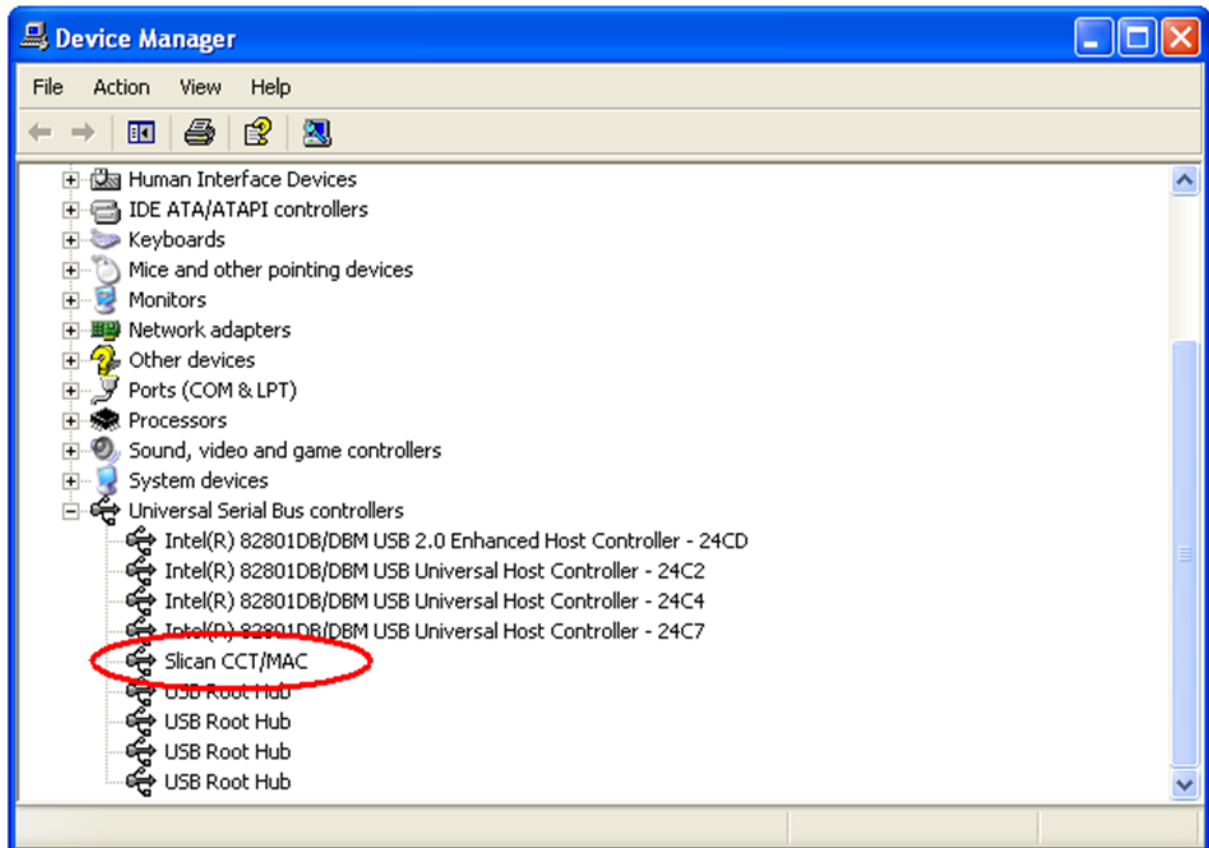


Fig 1.6: Windows device manager

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

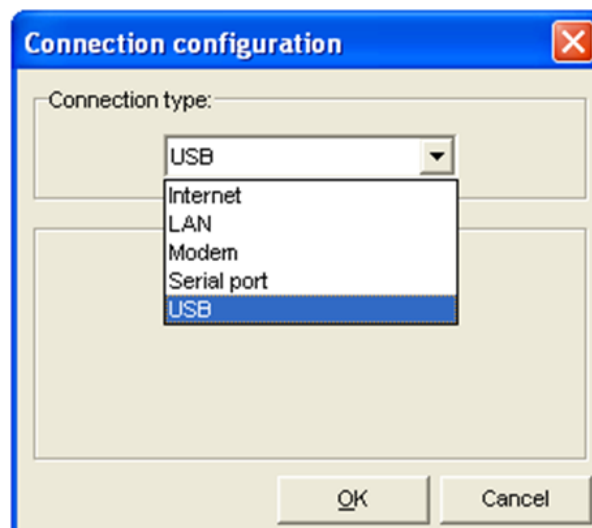


Fig 1.7.: Connection configuration window

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.

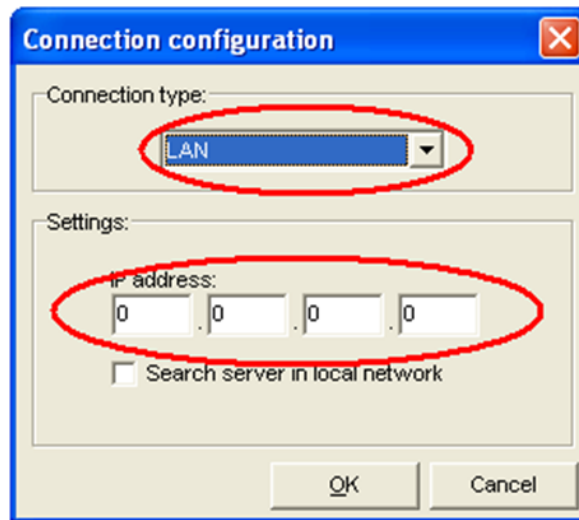


Fig 1.8.: Connection configuration via LAN

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.

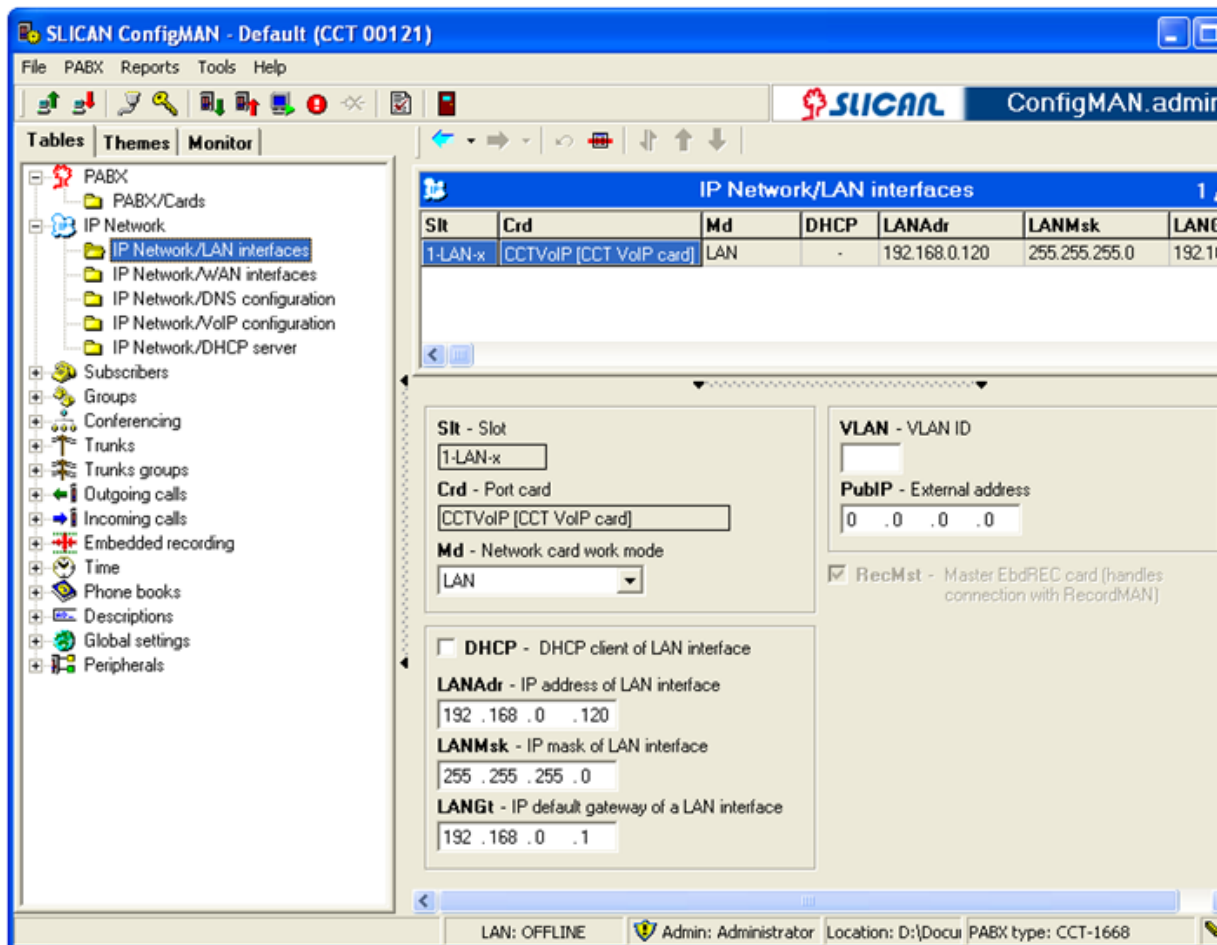


Fig 1.9.: LAN configuration

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.

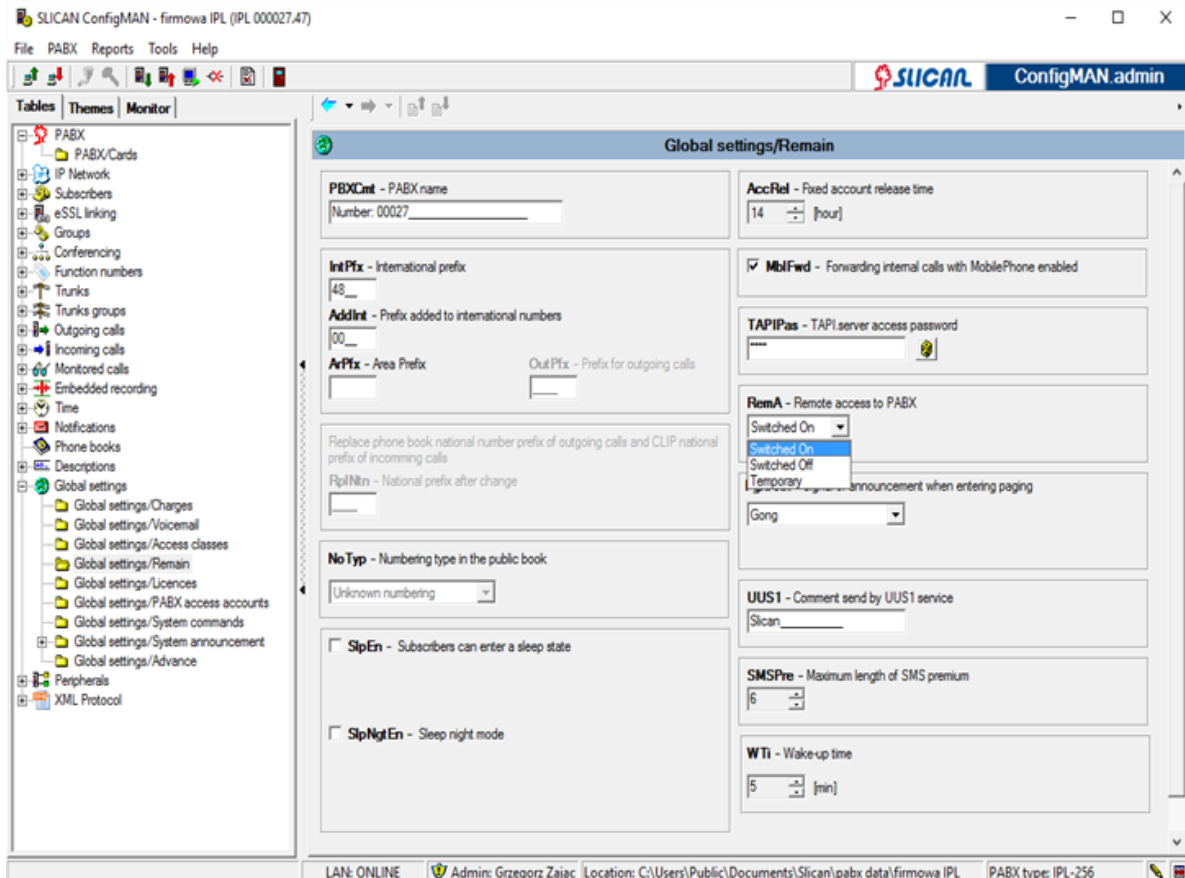


Fig 1.10.: Global settings / Remain – defining mode of remote PBX access

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.

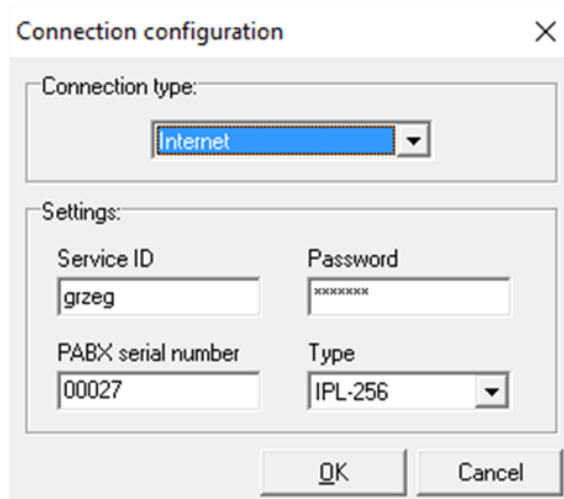


Fig 1.11.: Configuration for connection over Internet

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.

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 1.11.: Window for connecting with PABX over Internet – connection error identification

Information about Keeper connection can be found in *Monitor*, in *Port status* sheet.

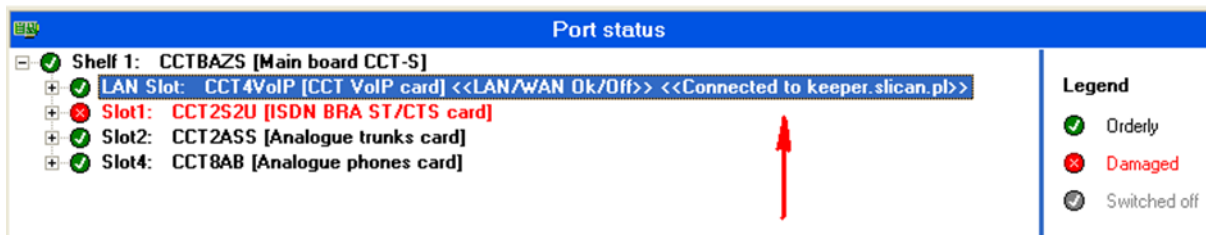


Fig 1.13.: Information presentation about connection with keeper

Information about loss or reestablishing connection are collected in **PBX monitor**, in sheet **PABX events**.

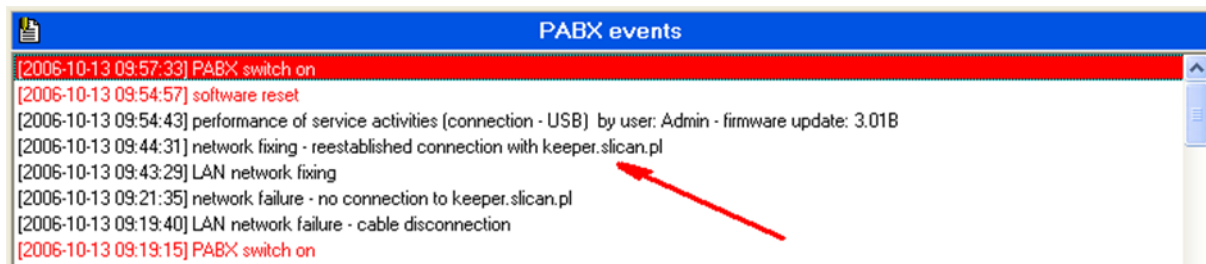


Fig 1.14.: PBX events – information about connection with keeper

Configuring ConfigMAN for modem transmission

(in CCT with firmware ≥ 5.11 , IPL, IPM 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.

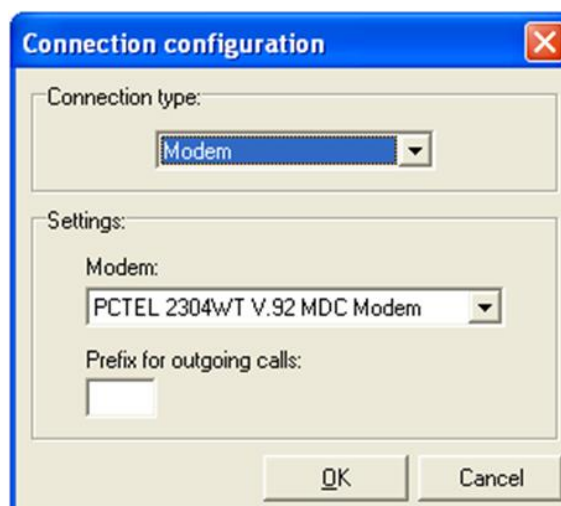


Fig 1.15.: Selection of modem connection

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.

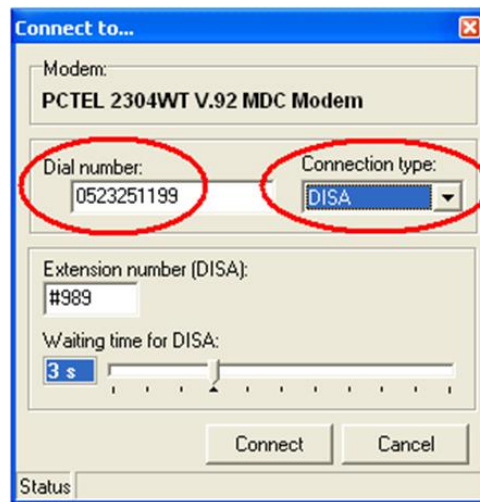


Fig 1.16.: Configuring modem connection

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.

Tables/Incoming calls									2 / 100
Cmt	DINo.	WrkMd	=>	DISA	ExtNo.	Fax	DTMF	AwdNet	
MSN 1	2	A.B._____	=>	<None>	System modem	<None>	-	<None>	
MSN 2	5212342	A.B.C.D.E.____	=>	<None>	2	<None>	-	<None>	

Cmt - Comment
 MSN 1

If:
DINo. - Dialed number
 2

WrkMd - Working mode
 A.B._____

- ☒ A. Day mode
- ☒ B. Night mode
- ☐ C. Lunch
- ☐ D. Weekend
- ☐ E. Malfunction
- ☐ F. Mode F
- ☐ G. Mode G
- ☐ H. Mode H

Outgoing calls CLIP
Trunks group
 Link do EU

Then:
DISA - Announcement
 <None>

ExtNo. - Extension number
 System modem

☐ **DTMF** - Active DTMF receiver during announcement

AwdNet - Allowed nets

☐ **VIP** - Call with higher priority

Fig 1.17: Settings for direct modem calls Incoming calss/Settings/Main

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.

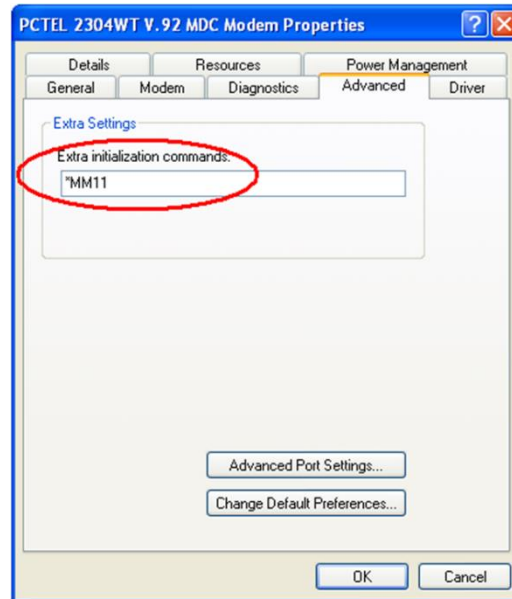


Fig 1.11.: Advanced modem properties for Windows

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

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

The **Statistics** field shows the number of ports and maximum capacity..

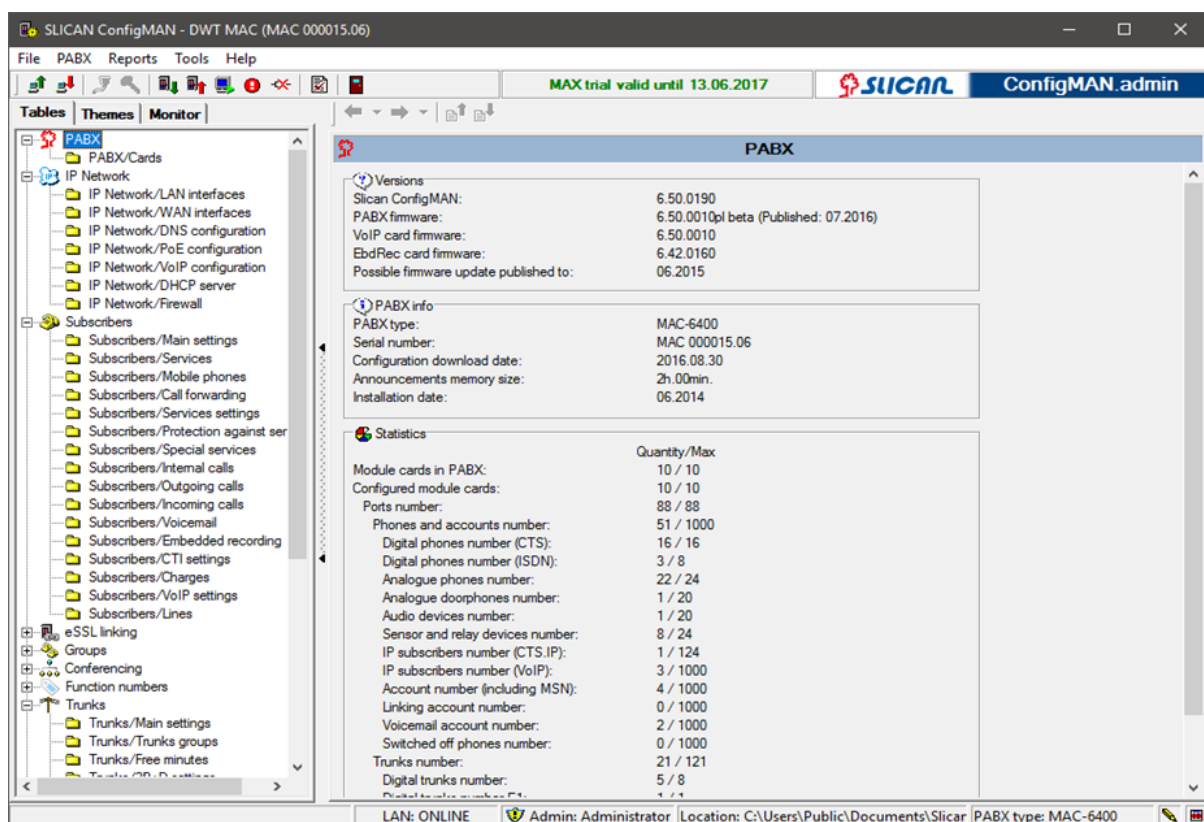


Fig 2.1: PBX details – most important information about the Slican PBX

Attention!

For IPL-256, IPM-032, IPU-14, IPS-08, CXS-0424 and CCT-1668 with fw. 5.11 or higher, file with LAN/VoIP/REC 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.

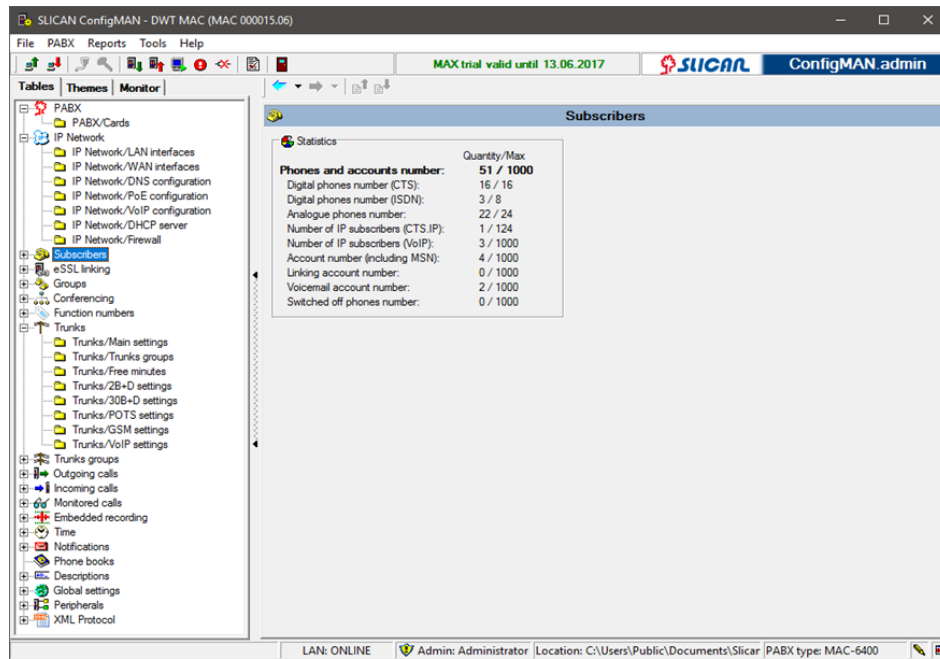


Fig 2.2: Statistics – Details for Subscribers

2.2 Interface description

The user interface of *ConfigMAN* is divided into three panes:

- tree,
- table,
- form.

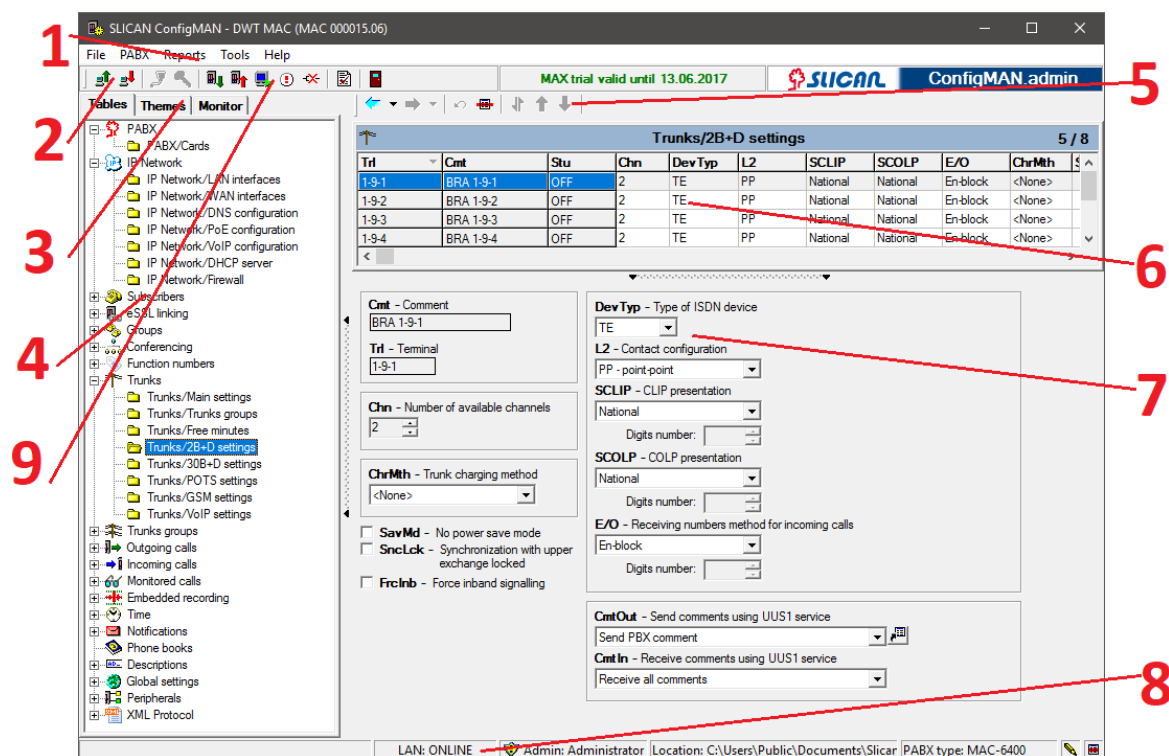


Fig 2.3: Description of ConfigMAN interface

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 and MaxTrial licence key expiration date
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

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.

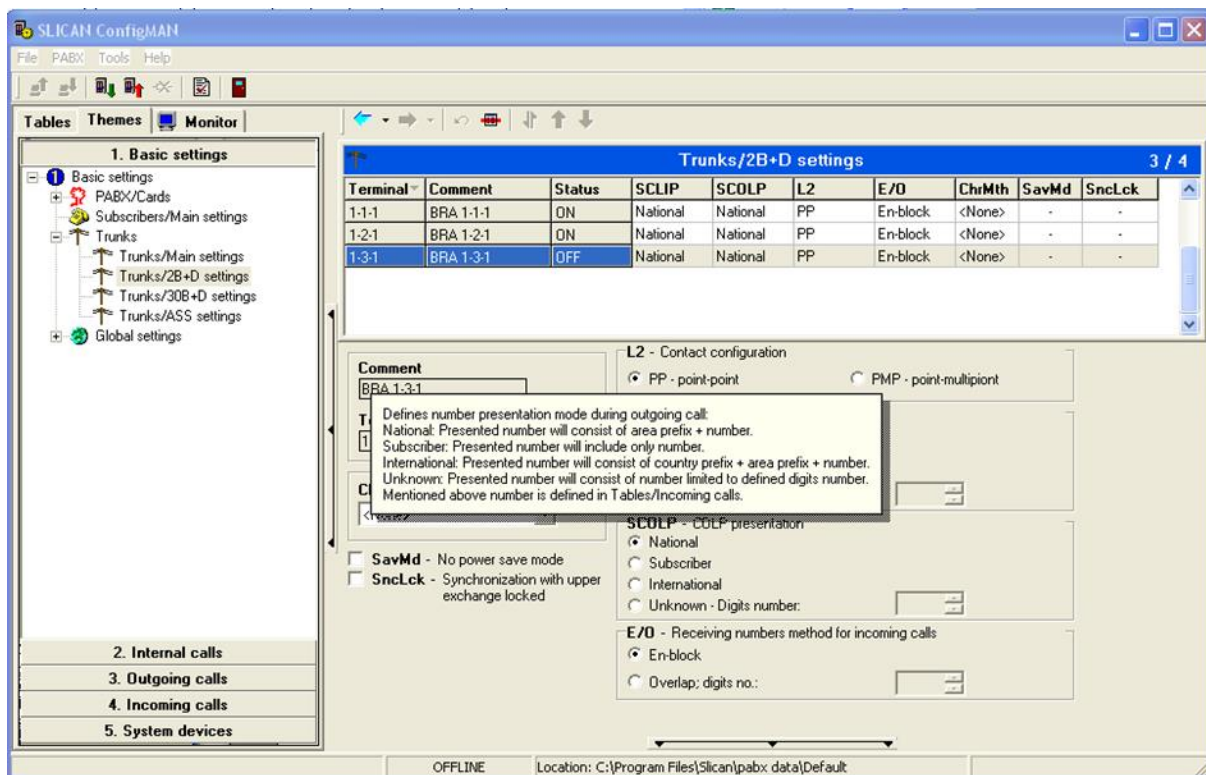


Fig 2.4: Displaying context help for fields available in specified window

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.

Subscribers/Services										
57 / 10										
ExtNo.	Cmt	Typ/Prt	Acc	HAcc	Lck	PIN	AlrC	AlrTi	GrpLmt	EnGrpH
2123	Abonent 2040	Subscriber	<None>	0:00	-	1111	-	<None>	Switched off	-
2055	cts ip 2055	Subscriber	<None>	0:00	-	2222	-	<None>	Switched off	-
2049	Abonent 2049	VoIP	<None>	0:00	-	-	-	<None>	Switched off	-
2032	Abonent 2032	VoIP	<None>	0:00	-	-	-	<None>	Switched off	-
2118	Lechu SIP 2118	VoIP	<None>	0:00	-	-	-	<None>	Switched off	-
2045	2045	VoIP	<None>	0:00	-	-	-	<None>	Switched off	-
2044	Powladomienia	Account	<None>	0:00	-	-	-	<None>	Switched off	-
2066	Artur test CTS IP	Subscriber	<None>	0:00	-	-	-	<None>	Switched off	-
120	120	Subscriber	<None>	0:00	-	-	-	<None>	Switched off	-
2037	Abonent 2037	Subscriber	<None>	3:00	-	-	✓	00:00	Switched off	-
2036	Abonent 2036	Subscriber	<None>	0:00	-	-	-	<None>	Switched off	-
2035	Abonent 2035	Subscriber	<None>	0:00	-	-	-	<None>	Switched off	-
2034	Abonent 2034	Subscriber	<None>	0:00	-	-	-	<None>	Switched off	-
2033	Abonent 2033	Subscriber	<None>	0:00	-	-	-	<None>	Switched off	-
2031	Abonent 2031	Subscriber	<None>	0:00	-	-	-	<None>	Switched off	-
1906	Abonent 1906	Subscriber	<None>	0:00	-	-	-	<None>	Switched off	-
1221	Rafal konto dla 1220	Account	<None>	0:00	-	****	-	<None>	Switched off	-
1220	Rafal testowy	VoIP	<None>	0:00	-	-	-	<None>	Switched off	-
2124	Konto Lechu 2124	Account	<None>	0:00	-	-	-	<None>	Switched off	-
1908	Abonent 1908	Subscriber	<None>	0:00	-	-	-	<None>	Switched off	-
1907	Abonent 1907	Subscriber	<None>	0:00	-	-	-	<None>	Switched off	-

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 its 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 is possible to select all table records by pressing CTRL and A keys.

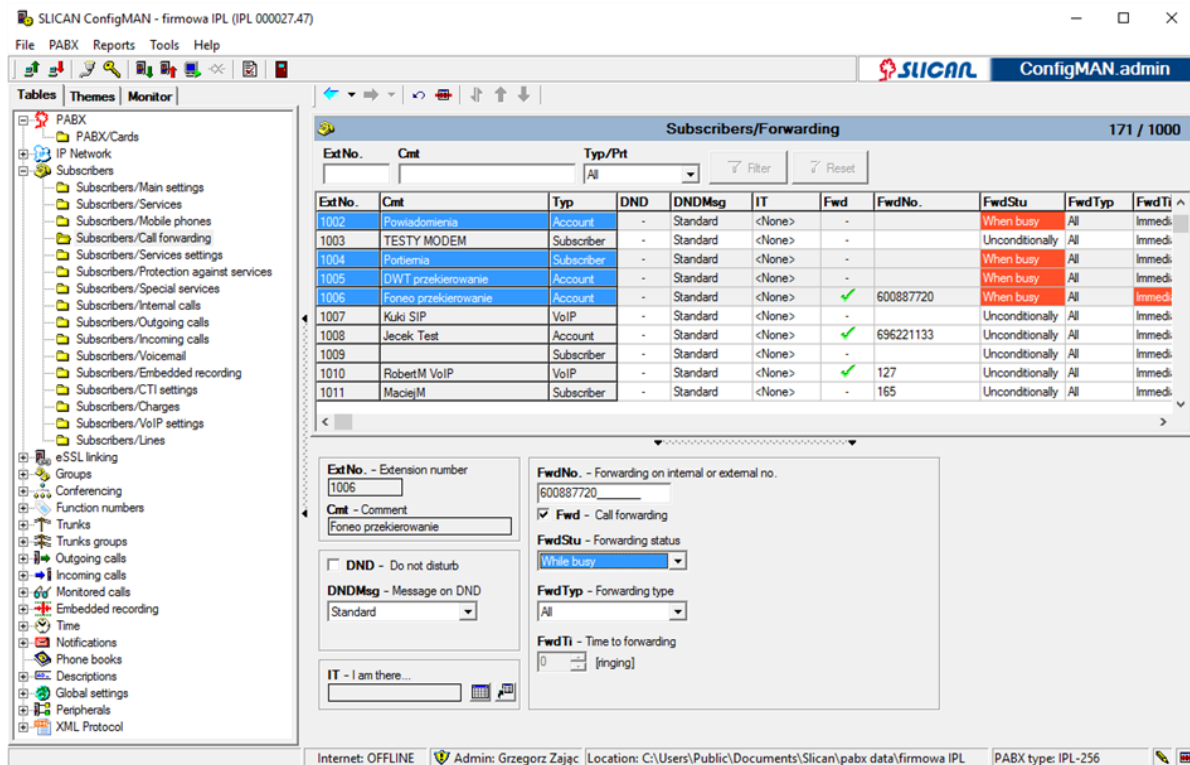


Fig 2.8: Editing man records during same time

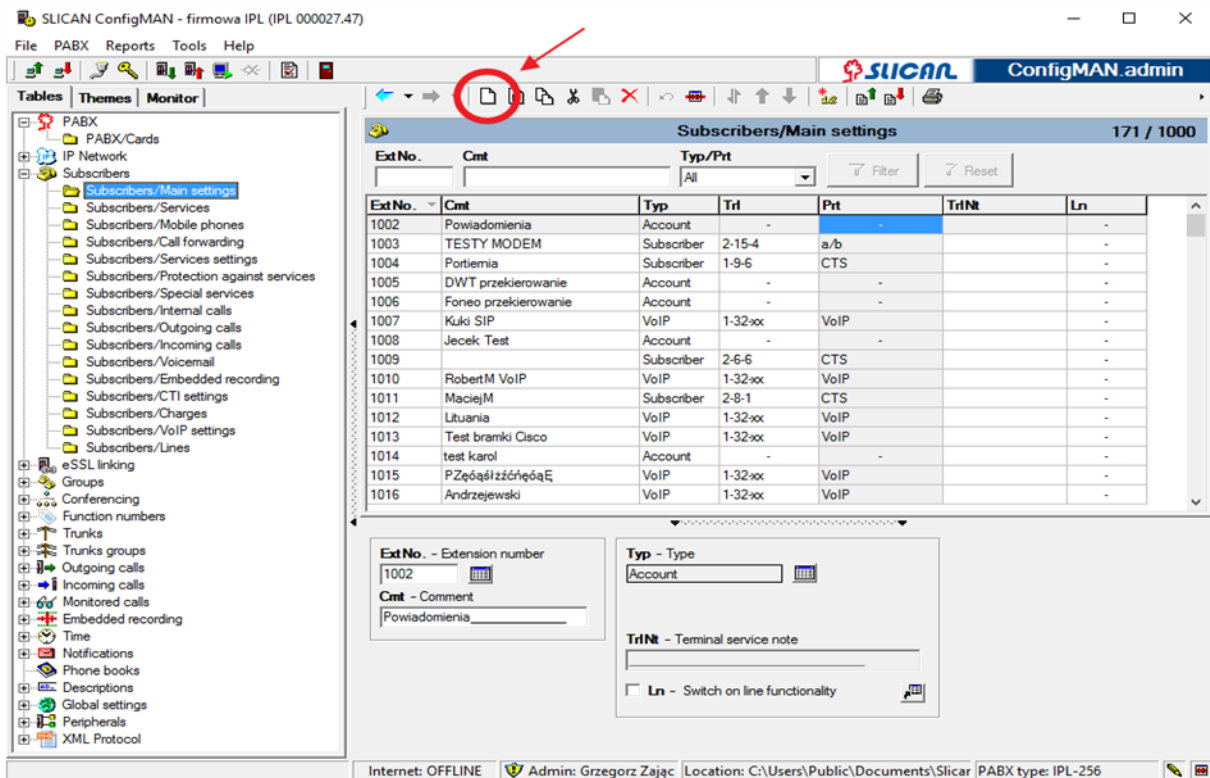
Modifying a field's value will cause every selected record to be updated with the entered value.

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.

Adding 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 their 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.

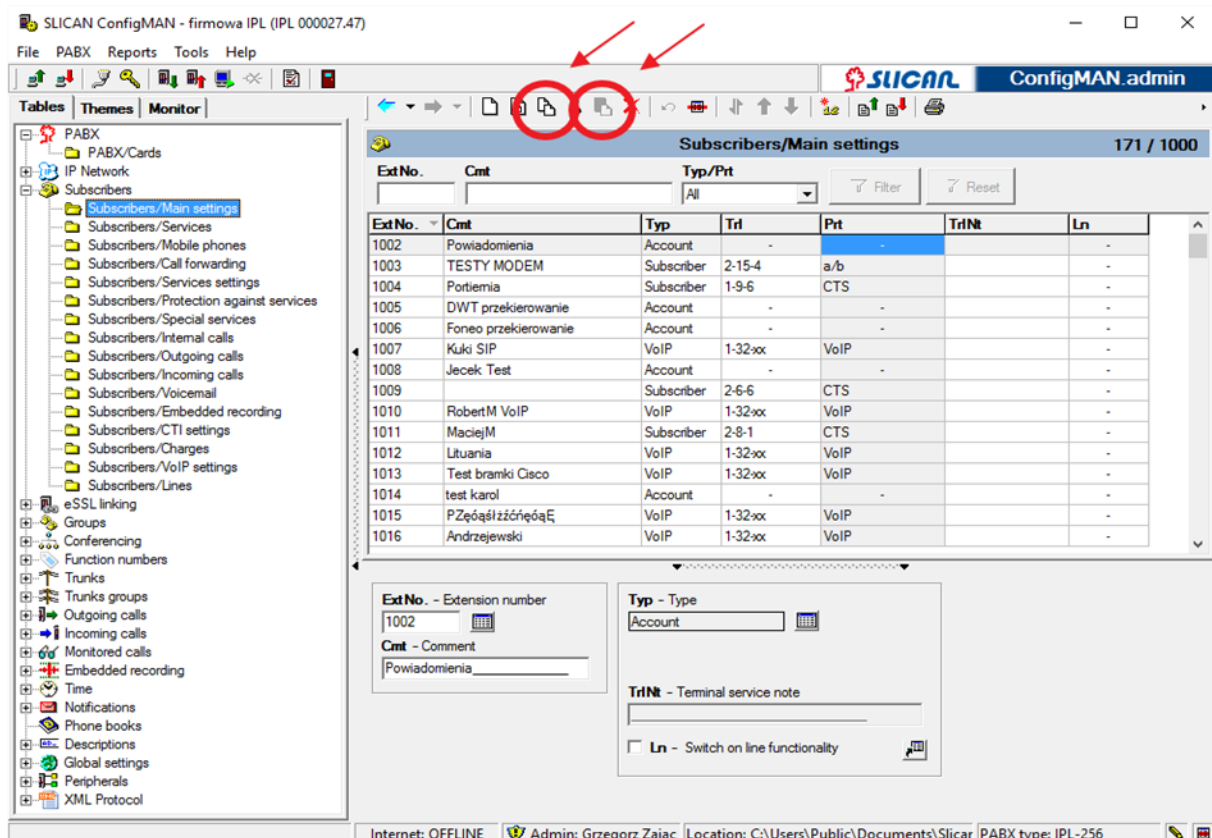


Fig 2.10: Copying records (rows)

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

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

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.

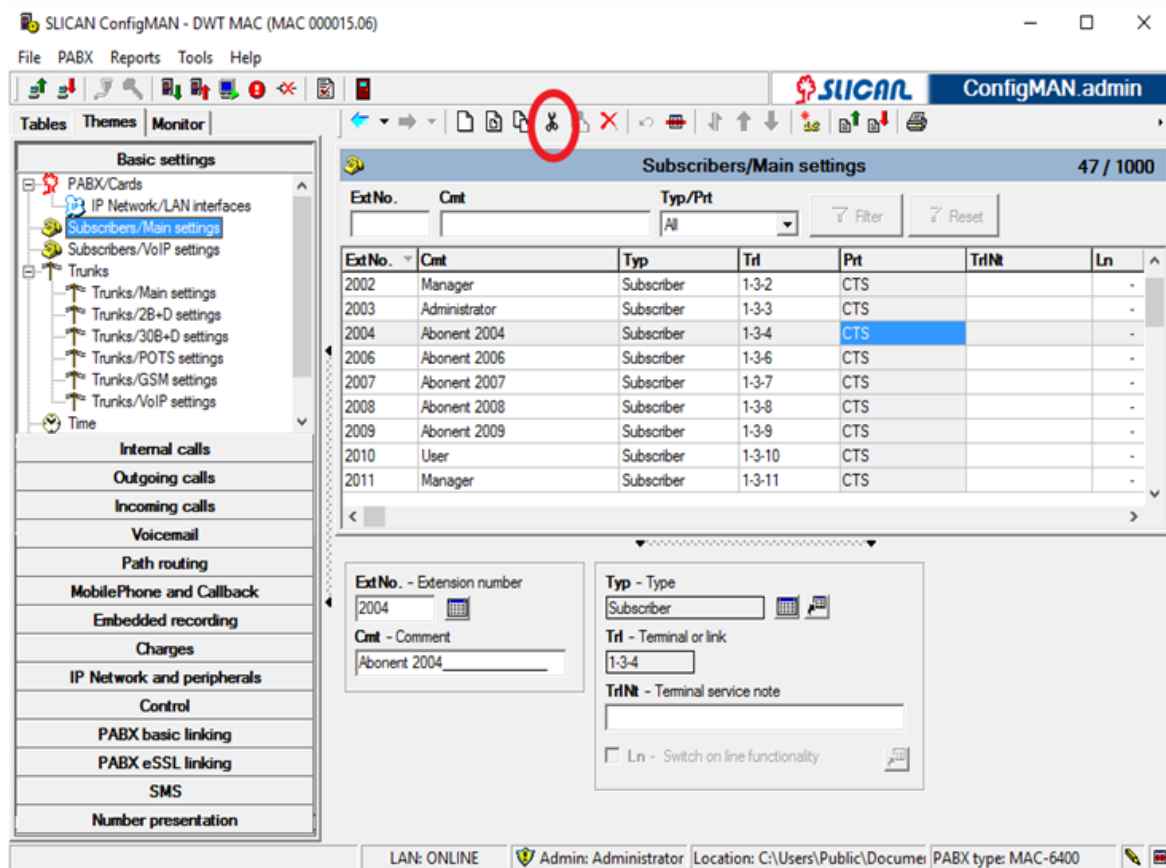


Fig 2.11: Cutting records (rows)

Option Delete selected records or the CTRL+DELETE key combination is used for deleting record (or records).

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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.

<i>Mode</i>	<i>Shortcut</i>	<i>Function</i>
Editing	<i>Ctrl+Ins</i>	Insert record
	<i>Ctrl+Del</i>	Delete record
	<i>Ctrl+Z</i>	Undo
	<i>Ctrl+A</i>	Select all
	<i>Ctrl+F7</i>	Verify configuration
	<i>Shift+Ctrl+Ins</i>	Duplicate record
	<i>Shift+Ctrl+C</i>	Copy record
	<i>Shift+Ctrl+V</i>	Paste record
	<i>Shift+Ctrl+X</i>	Cut record
	<i>PgUp</i>	Scrolling records up in table window
	<i>PgDown</i>	Scrolling records down in table window
Connecting to the PBX	<i>Ctrl+F9</i>	Login and password
	<i>Ctrl+F5</i>	PBX connection configuration
	<i>F4</i>	Launching diagnostic console
	<i>F5</i>	Launching system console
	<i>F6</i>	Launching ISDN measure
	<i>F7</i>	Upload to PBX
	<i>F8</i>	Download from PBX
	<i>F9</i>	Activating/Deactivating monitor
	<i>F10</i>	Analog phone simulation
	<i>F11</i>	Stop/restore monitoring
	<i>Ctrl+BckSp</i>	Disconnect
	<i>F1</i>	Help

<i>Mode</i>	<i>Shortcut</i>	<i>Function</i>
View manipulation	Ctrl+O	Open configuration
	Ctrl+S	Save configuration
	Ctrl+N	New configuration
	Alt+F4	Quit
	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.

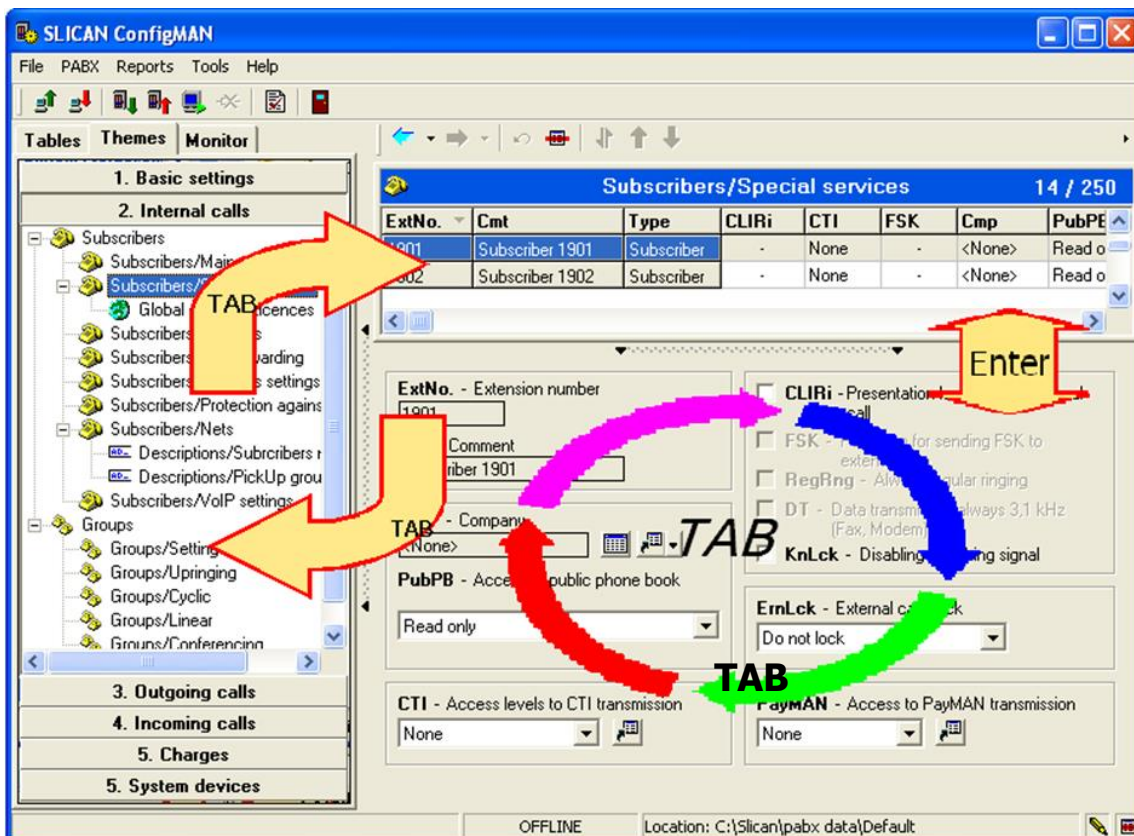


Fig 2.14: Navigation using keyboard

2.9 Opening few different config files within one location

ConfigMAN application allows to run only single PBX config file. Sometimes there is need for viewing different config files at same time. This option is provided by MultiMAN application, which saves each location and separate entry and its own data folder.

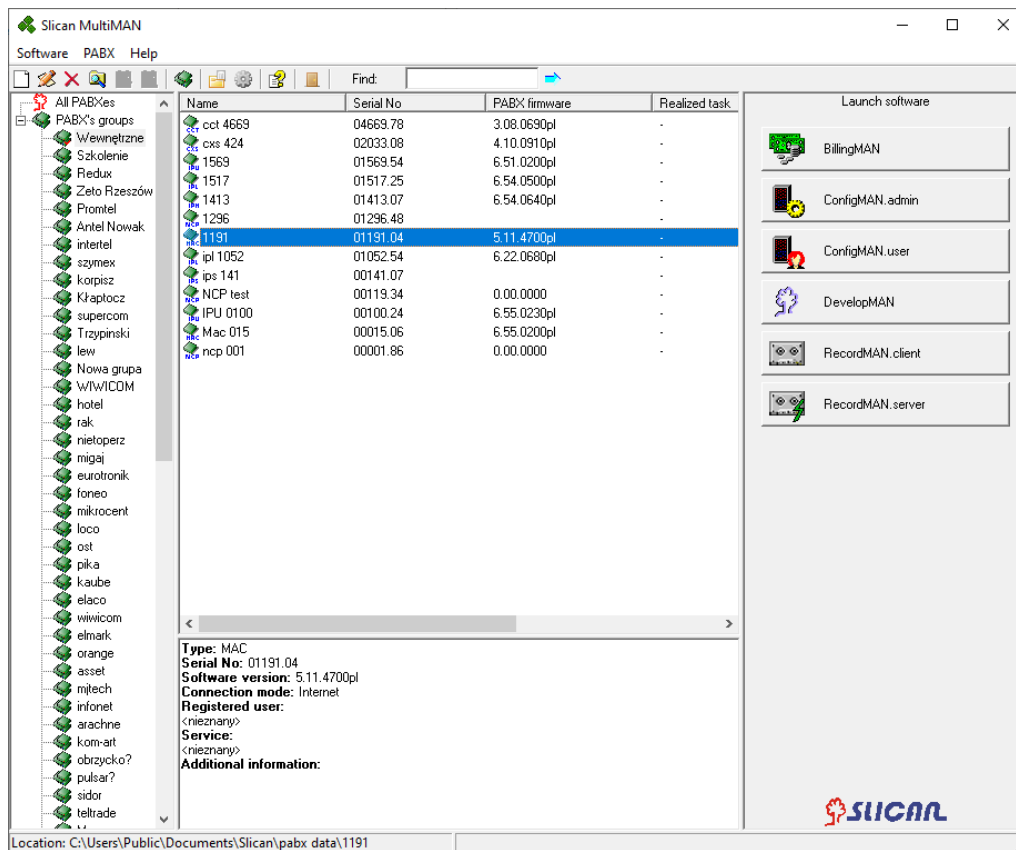


Fig 2.15: Opening different locations within MultiMAN

Each location can open separate PBX config with different sessions and switch between them. It allows for simultaneous connection with each PBX (multiple access) if connection through keeper server, assigned PBX IP address, or using modem or USB are used. Additionally, each supporting software for PBX can be accessed from the panel – BillingMAN and RecordMAN. is used, using assigned.

All options regarding sending and downloading files from, to PBX are active.

Attention!

Each PBX may have only single connection from ConfigMAN application at the same time. Restriction was introduced to secure from multiple people trying to simultaneously configure PBX.

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 (for system phones),
- announcements uploaded by PBX administrator.

Backup files have a *.cpy extension.

To make a backup copy:

- using PC connect to the PBX,
- from the **PABX** menu, select **Perform backup**,
- select a directory on the hard drive, and provide a file name,

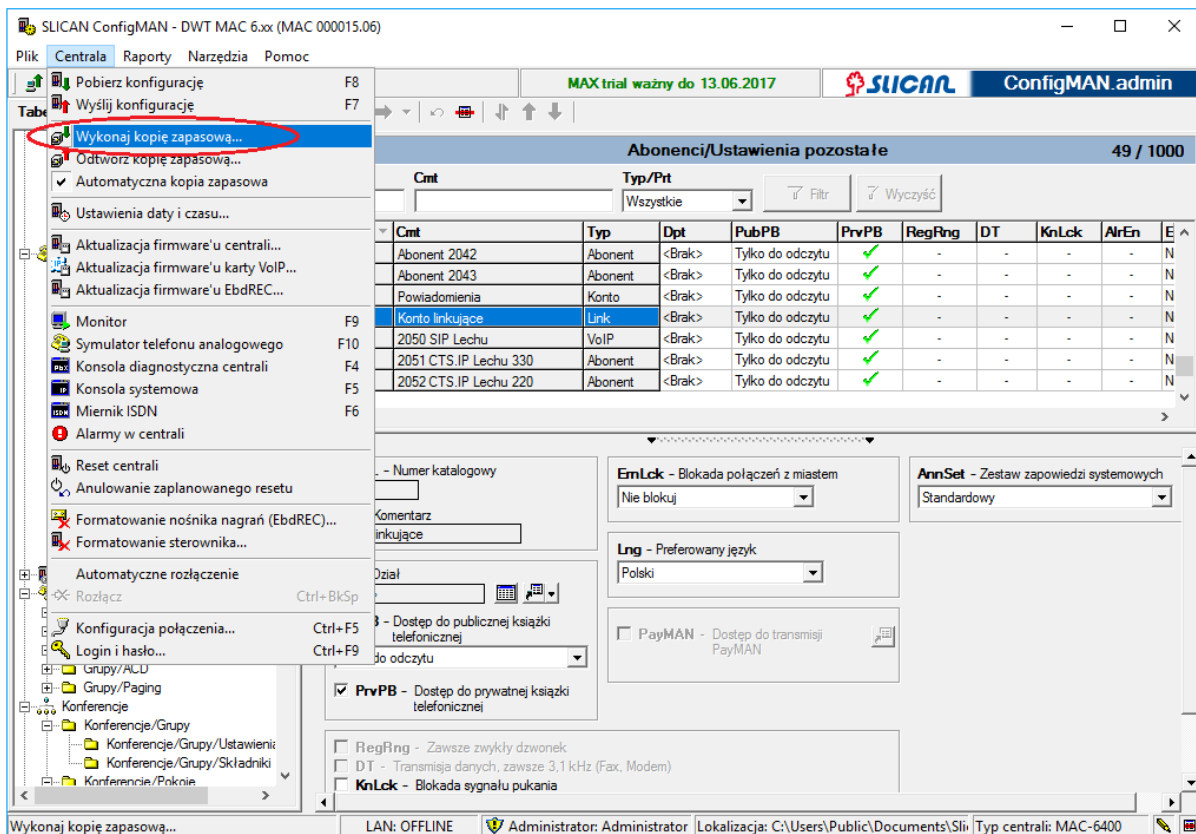


Fig 3.1: Backup from PABX menu

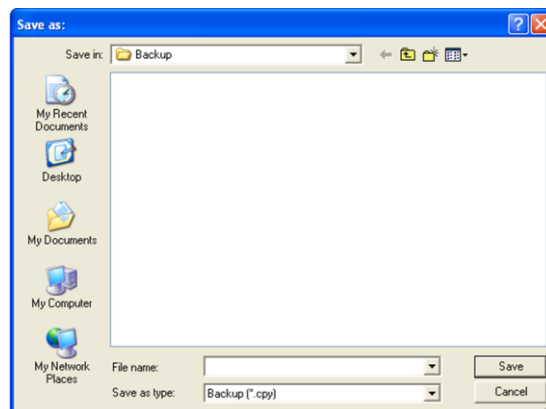


Fig 3.2: Selecting location and name of backup file

After finishing backup, application informs about result of operation.

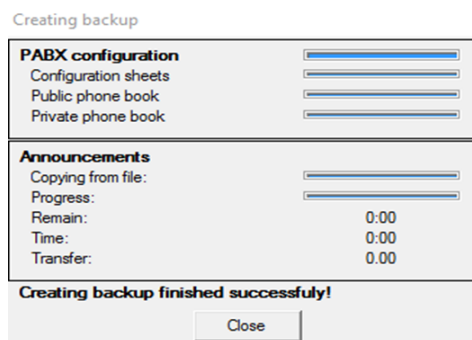


Fig 3.3: Backup process window

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.

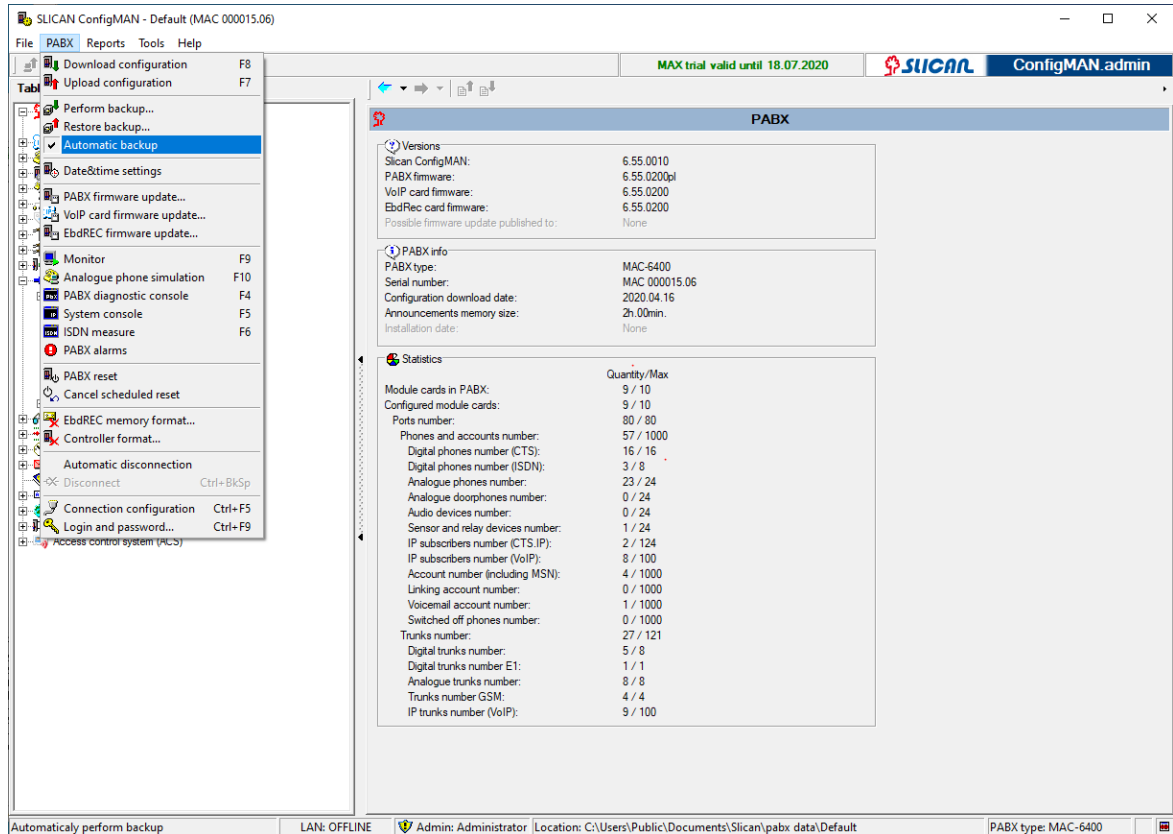


Fig 3.4: Selecting option – automatic backup

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.

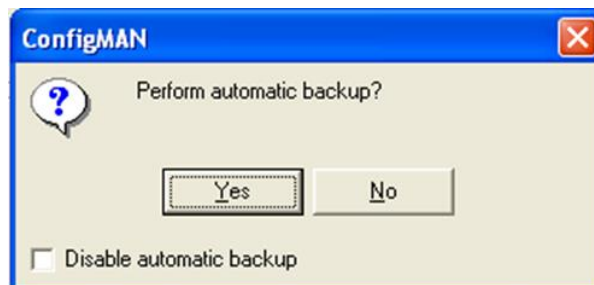


Fig 3.5: Perform automatic backup window

If backup request is confirmed, the current backup will be saved as *Version FW_RRRR-MM-DD_N.cpy*

where:

- **Version FW** – full software version of current PBX firmware

- **RRRR** – year

- **MM** – month

- **DD** – day

- **N** – number of copy from this day

Example: 6.54.0700_2020-02-22_1.cpy

Backup copy contains PBX configuration, phonebook and uploaded announcements.

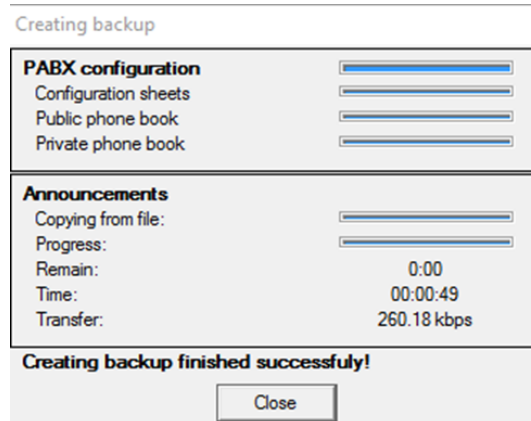


Fig 3.6: Backup process window

Restoring configuration backup

To restore a backup configuration, select **Restore backup** option from the **PABX** menu.

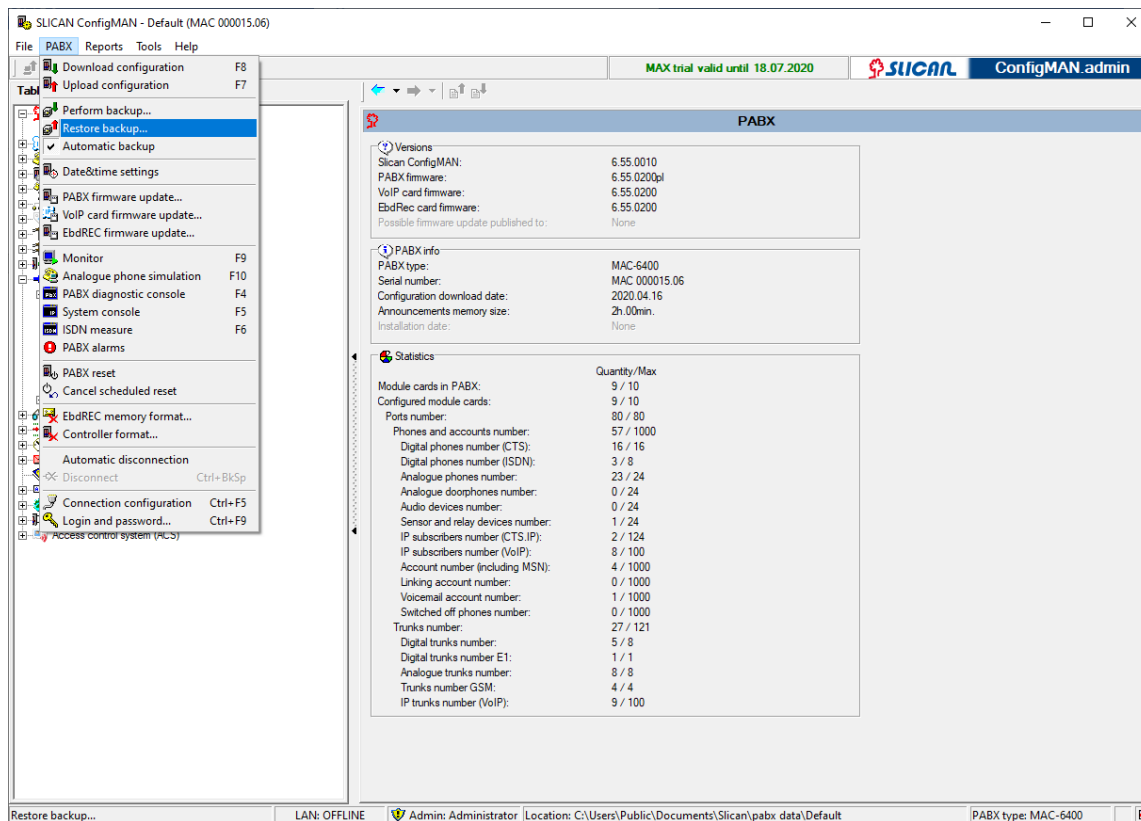


Fig 3.7: Selecting option - restore backup

Select which file will be sent to PBX.

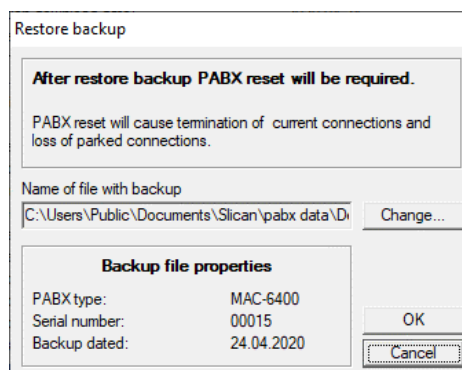


Fig 3.8: Restore backup window

After completing the uploading operation, the program will display a success notification.

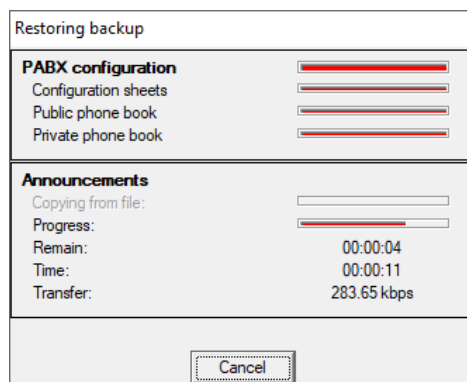


Fig 3.9: Restoring backup window

4 Programming the PBX

4.1 Programming a new PBX (Themes tab)

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

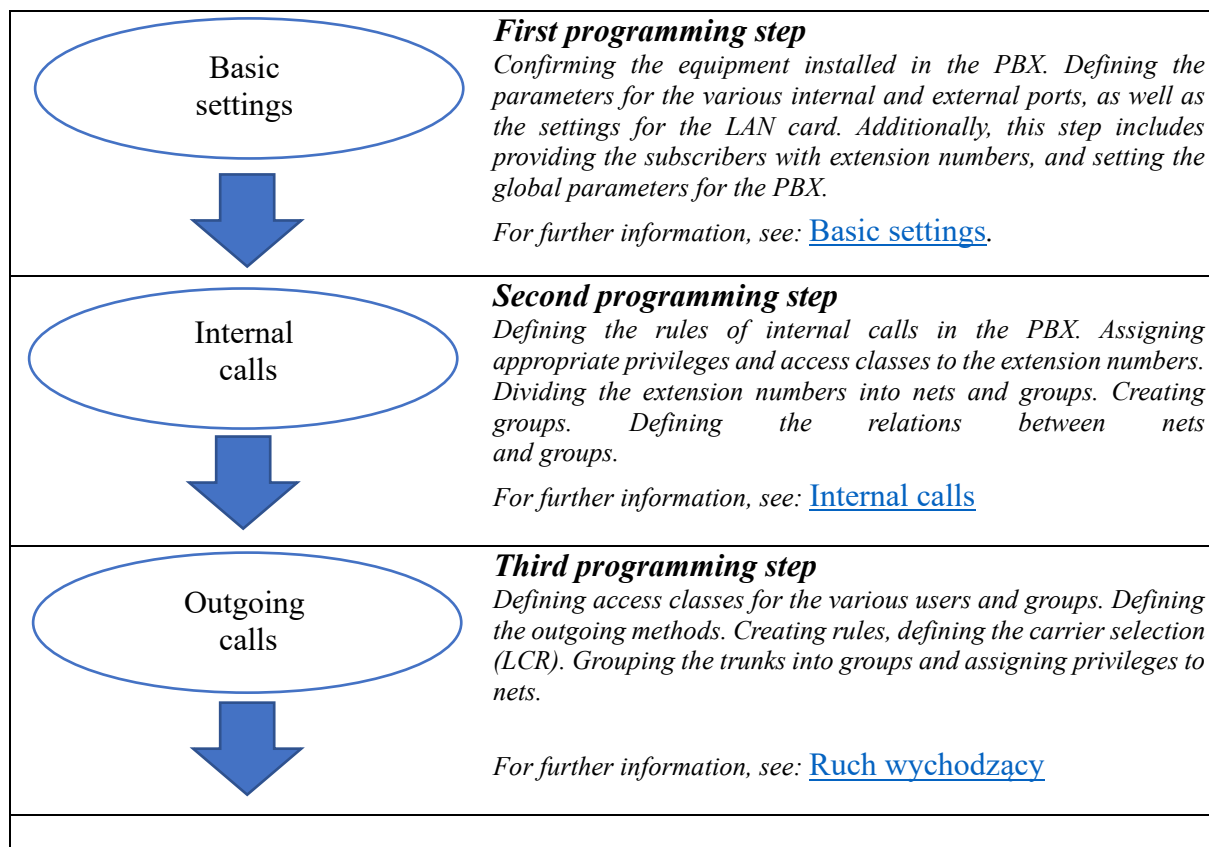
Attention!

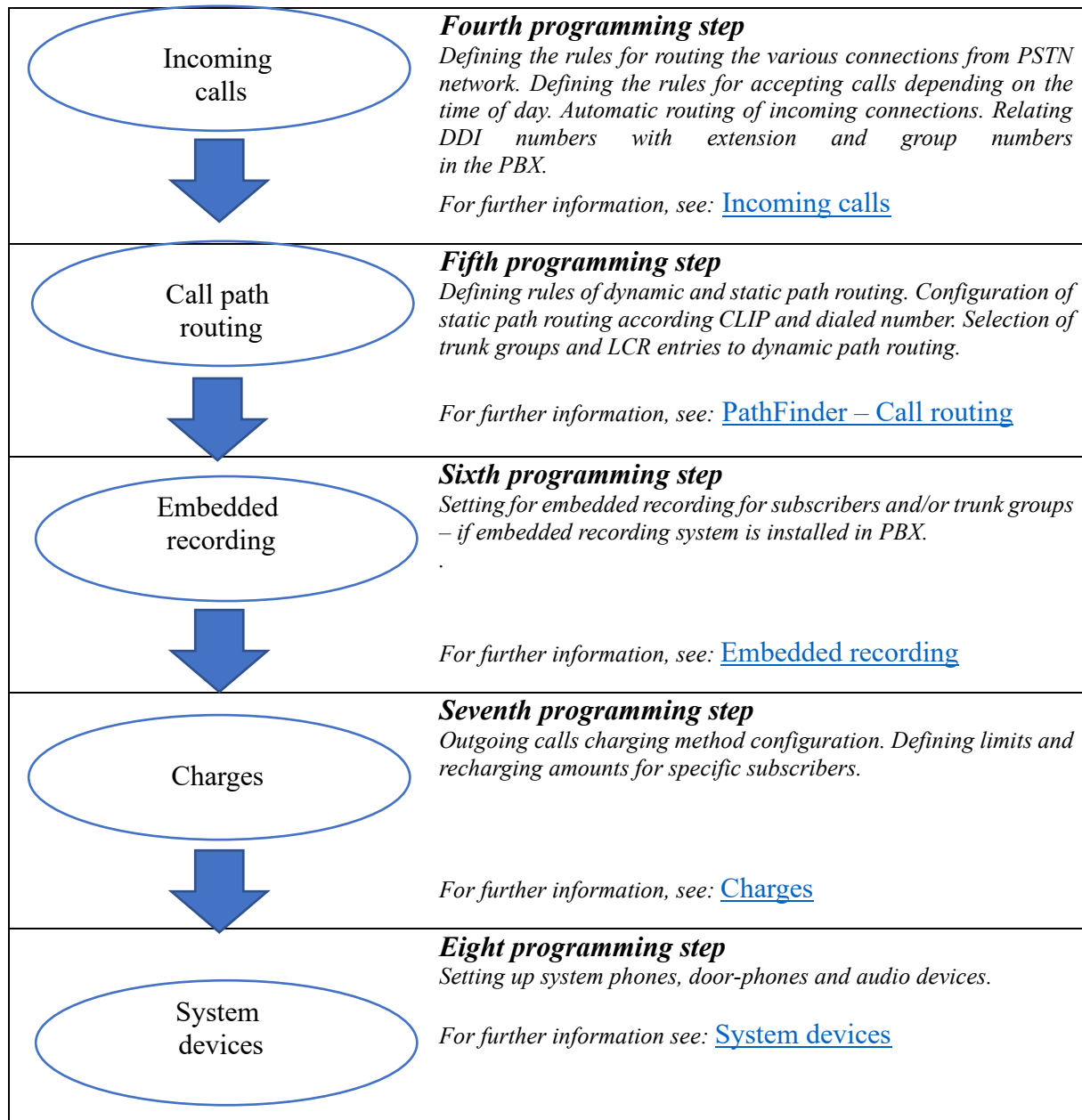
Whole functionality of Hard reset wizard is described in chapter [Hard reset wizard](#)

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:





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 on **PABX/Cards**.

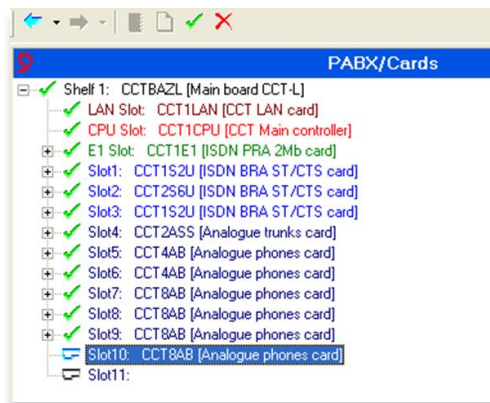
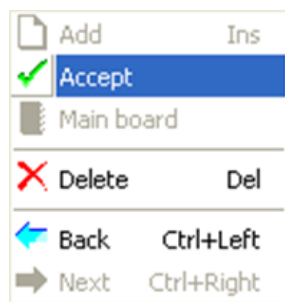


Fig 4.1: Confirming activated cards

- 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 (Ctrl+F7).

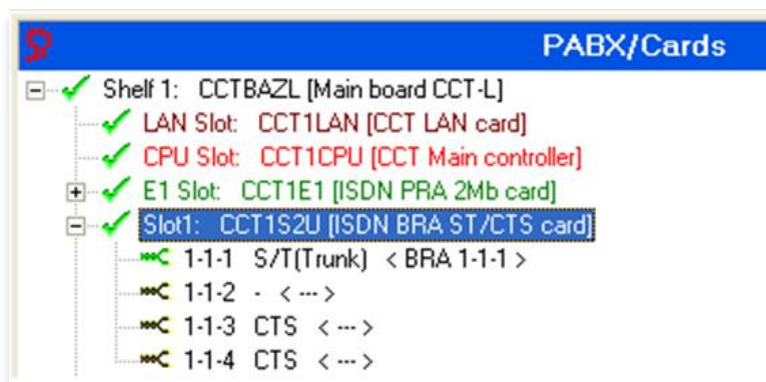


Fig 4.2: Promoting card tree

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 PBXes is equal to 11, while number of ISDN subscribers is equal to 12.

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, wizard will allow to create one, the rest and should be created manually later using ConfigMAN.

UWAGA!

Full functionality of VoIP card was described in chapters [LAN and VoIP cards in Slican PBXes](#) and [VoIP functionality in Slican systems](#)

4.2.2.1 Activating/deactivating trunk and configuring its own number

Each field meaning:

- **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**.
- **Stu** - manually switching trunk on or off.
- **TrInt** - 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. **National** is default setting.

- **DevTyp** - Content of this field defines ISDN trunk operating method..
 - TE - is regular trunk operating mode – operating with telecommunication provider, trunk only receives signaling and power 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 ISDN DDI
 - PMP - Point – Multipoint configuration used for ISDN MSN

- **Chn** – This field enables number of trunk channels available for phone calls established by PBX. It can be used if remaining channels are used by other devices connected before PBX eg. ISDN phone.
- **ChrMth** – Enables to define charging method of ISDN trunk:
 - *None* – no charging is provided
 - *Charging Unit* – call is charged according to number of pulses received from provider; pulse price is defined in **Trunk groups / Settings, Prc** field;
 - *Currency* – call is charged according to 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.
- **Frclnb** – 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.
- **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. This is number visible to person that was called, if their phone supports this:
 - *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
- **E/O** – Numbering receiving method for incoming calls via ISDN trunk:
 - *En-block* – upper exchange send all digits together (in block)
 - *Overlap* – digits numbers, after receiving from upper exchange to dial numbers by PBX

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 sent 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

The selected trunk requires the following settings, received from the provider:

- field **DiMd** – dialing mode – tone or pulse,
- field **ChrMd** - charging mode.

POTS 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**,

Remaining 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 and delay when phones will ring up.
- **DiEn** – after switching on this option trunk dialing tone detector is enabled.
- **BsyEn** – after switching on this option trunk busy signal detector is enabled.
- **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.
- **EchEn** – after checking this option echo cancellation during connection is enabled (doesn't apply to calls routed to fax from sheet Incoming calls/Settings/Main **Fax** field)

4.2.2.5 GSM trunks

For GSM trunks configuration is set up from setting in sheet **Trunks/GSM settings**. Basic setting is **PIN** code for SIM card used by specified trunk.

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.

Additional settings:

- automatic or manual choice of GSM provider. **Prd** field can be set as:
 - Auto – automatic choice of GSM provider,
 - Manual – allows manual choice of GSM provider. This setting is helpful while using provider with weak or changing signal and provider allows roaming with other providers eg. Play in Poland or various MVNO providers.
- **EchCnc** - enables echo cancellation for all calls or only transits through analog trunk.
- **CLIR** –enables CLIP blocking for outgoing calls using GSM trunk when operator allows this functionality.
- **WrkMD** – field use to set mode of GSM port (2G, 3G) when using 3G sim card. Available only for IPL and IPM PBXes.

Other fields change settings regarding applications allow to send or receive SMSes:

- **ID** – *GSM gate identifier* – is used for informing and is automatically filled by PBX. It allows to configure sending and receiving SMSes *SenderSMS* i *TelefonCTI.plus* applications.
- **SMSStu** – enables proper permissions for sending SMSes using gateway:
 - **Enabled** – with [MessengerCTI.Desktop](#) *ap* (according to *AwdSMS* field *Subscribers/CTI settings sheet*)
 - **SenderSMS** –with Slican SenderSMS application
 - **External applications** – with applications created by external companies or *Slican TelefonCTI.plus* applications
- **SMSSta** – enables static routing of incoming SMSes to chosen subscribers or subscribers group using [MessengerCTI.Desktop](#) application
- **SMSDyn** – enables dynamic answers to SMS (return) to original sender (subscriber using [MessengerCTI.Desktop](#) application to send SMS). Also allows for direct call back.

4.2.2.6 VoIP trunks

ATTENTION!

Description of full VoIP functionality can be found in chapters [Lan and VoIP cards in Slican PBXes](#) and [VoIP functionality in Slican PBXes](#)

In Slican systems it is possible to set **up to maximum:**

- **100** logical VoIP trunks for PBXes IPL 256, IPM-032 (except Low Cost version), MAC-6400, CCT-1668, CXS-0424
- **16** logical VoIP trunks for IPU-14
- **8** logical trunks for IPM-032 Low Cost and IPS-08

VoIP trunks must be configured with those fields:

- **Chn** – defines number of VoIP channels (simultaneous connections), which should be done via specified trunk. Channel number set up here can not be greater then number defined for whole VoIP trunk. It is accessible using *PBX/Cards* sheet after clicking on Controller *slot* (CCT and IPx) or VoIP Card slot (MAC). Using *Configuration* buttons assigned to **VoIPChn** field VoIP Channels properties windows can be opened. Informations there correspond with physical DS_p modules, physical channels and licensing restrictions, codecs usage and three sliders that allow to manually change number of channels for phones,

CTS.IP and Sip Trunks. For trunk channel number is default value according to hardware, licenses and above restrictions.

- **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 depending on hardware)
- **CnnTyp** – this field is important if VoIP trunk is used to linking PBX with other *Slican CCT-1668* or *MAC-6400* PBX using eSSL protocol – detailed information can be found in chapter: [PBX linking with eSSL](#)
- **OprMd** – Setting trunk connection mode – client or server. When connecting to VoIP provider client is default setting, when connecting two PBXes, one must be set as server, second as client.
- **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 eg, 195.162.16.201:5060), if this port number is different than port defined in field **PrtSIP** in sheet *IP Network/VoIP configuration*. Default port is 5060 for SIP.
- **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** transmission 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 of 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 heard DTMF tones, if for VoIP trunk option „out band” is used. Solution in this situation is setting option „in band”.

- **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 target 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 sent via public network should be displayed on phones used in telephone network

- **CLIP** – in case of active DDI service on trunk – provider's SIP type presentation in incoming 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

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.

- **PBXCmt** – PBX name and ID used during linking PBXes
- **IntPfx** – contains international prefix of country in which PBX is located
- **AddInt** – for incoming traffic, adding prefix which allows to directly call back international numbers, eg. 00,
- **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(recommended).
- **NoTyp** – numbering typ of pbx public phone book – *National numbering* lub *Unknown numbering*. Setting up mode of saving numbers in phone book and calls history. Allows to simplify and organize call back method for different settings of outgoing calls config. Option active only when outgoing calls go using „city calls number „(**Outgoing calls OutHl** field – **Brak**, **Outgoing calls/Short numbers** – chosen digit for city calls). For detailed info look into chapter [Phone book modes](#)
- **SlpEn (SlpTi)** – set up time of inactivity – when it passes phones and CTI apps go into night sleep mode (dimmed display, „sleep” state for app). Option not related to PBX working mode
- **SlpNgtEn (SlpNgtTi)** – set up time of inactivity during night mode - when it passes phones and CTI apps go into night sleep mode (dimmed display, „sleep” state for app)
- **AccRel** –set up after how many hours fixed account will be released. Can be set from 0 to 23 hours.
- **MblFwd** – global field allowing MobilePhone users to forward incoming internal calls to another PBX user (option available since version 6.42)
- **TAPIPas** – password for TAPI server version – TAPI.server.
- **RemA** – settings of remote PBX access turned on for privileged user, three modes are available: switched on, switched off, temporary (30 min).
- **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 **Peripherals / Audio** – detailed information can be found in chapter [Audio devices configuration](#)). Available settings are:
 - *<None>*: After establishing call audio channel is immediately switched on – caller voice can be heard in paging system loudspeaker immediately).
 - *Two-tone gong*: after establishing call, caller hears melody – in the same time in paging system loudspeakers two-tone sound is played, after melody caller hears short beep and then he/she can speak any message.
 - *Zapowiedź. 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 then he/she can speak any message).
- **UUS1** – setting up comment sent in ISDN 2B+D, 30B+D or VoIP trunk signalling. Trunk configuration must enable sending comments.
- **SMSPre** – setting up max number length of SMS Premium (higher cost). Sending SMS to number with exact or higher number of digits won't be available.
- **WTi** –setting maximum time of wake-up repeating, after which service will be counted as not realised.
- **MaxTi** – setting up default call duration limit for phones working in [Auditorium](#)

- **ANo** –access number which will set up GSM calls to app *MessengerCTI.Mobile*

Firmware 6.54 introduced timed restriction for some services:

- **FwdLmt** – opcja wyłączenia usługi przekierowania bezwarunkowego dla tych abonentów centrali, u których w zakładce setting of turning off unconditional forwarding for PBX subscribers which in tab *Subscribers/Call Forwarding*, **FwdLmt** field with *"limit by global settings"*
- **DNDLmt** – setting of turning off DND for subscribers, which in tab *Subscribers /Call Forwarding*, field **DNDLmt** is set *"limit by global settings"*
- **GrpLmt** – setting login off subscribers from groups, if their tab pcja wylogowania z grup tych abonentów centrali, u których w zakładce *Subscribers /Services*, field **GrpLmt** is set *"limit by global settings"*

Wyłączenie każdej z funkcjonalności może posiadać kryterium czasowe (ilość godzin do wyłączenia) lub godzinowe (o wyznaczonej godzinie) Turning off each feature can have time criteria (amount of hours that have to pass to turn of) or hourly (during set hour).

4.2.4 Admin and user account

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").

ATTENTION!

In new PBX, or after performing hard reset (using [Hard Reset Wizard](#)) default admin account settings are as follow:

- login: **admin**,
- password: *four last digits of PBX serial number*,
- code¹: *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>.

¹ Used by service *87 – Manual working mode changing and service *92 – Announcement recording.

Cmt	Log	Pss	CM	BM	Cd	Ann	Sw	Lmt
Key 2	user1	*****	Admin	✓	****	✓	-	-
Administrator	admin	*****	Admin (Root)	✓	****	✓	✓	-

Cmt - Comment
Key 2

Log - Login
user1

Pss - Password
Change password
Delete login and password

CM - ConfigMAN
Admin

☒ **BM - BillingMAN**

Cd - Four digits code
Change code
Delete code

Privileges to applications:
☐ PM - PayMAN

Privileges to phone services:
☒ Ann - Announcement recording
☒ Sw - Working mode switching

☐ **Lmt - Account expires after declared time**

Fig. 4.4: Sheet Global settings / PABX access accounts

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 for 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 (support stopped from fw 4.11/5.11)
- **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 (codes and logins can't repeat).

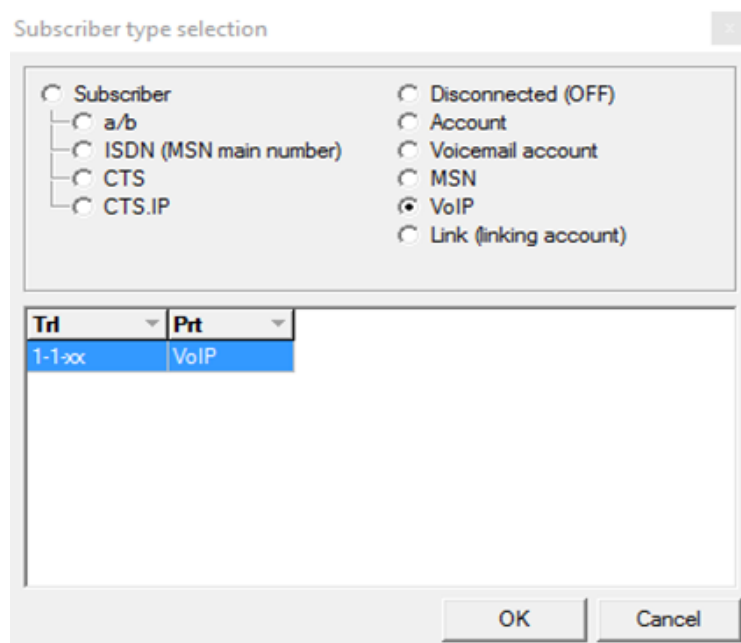
4.3 Internal calls

4.3.1 Extensions 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 SIP protocol).



The dialog box titled "Subscriber type selection" contains two columns of radio button options. The left column includes "Subscriber" (selected), "a/b", "ISDN (MSN main number)", "CTS", and "CTS.IP". The right column includes "Disconnected (OFF)", "Account", "Voicemail account", "MSN", "VoIP" (indicated by a dot in the center of the circle), and "Link (linking account)". Below the options is a table with two columns, "Trl" and "Prt". The first row has "1-1xx" under "Trl" and "VoIP" under "Prt". At the bottom right are "OK" and "Cancel" buttons.

Trl	Prt
1-1xx	VoIP

Fig. 4.5: Adding new internal user in PBX – subscriber type selection

ATTENTION!

Full functionality of VoIP card was described in chapters [LAN and VoIP Cards in Slican PBXes](#) oraz [VoIP functionality in Slican systems](#)

VoIP subscriber ports are subject of licensing. Information concerning licensing can be found in chapter [Licenses](#)

Subscribers 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** or **Linking account**. Any type of subscriber, which is not used in specified moment can be switched off without removing it

For any created subscriber those parameters:

- extension number can be changed – **ExtNo.**,
- comment can be added – **Cmt**,
- type can be changed – **Typ**
- terminal service note can be added – **TrINt**

Virtual account can have line functionality enabled - **Ln**

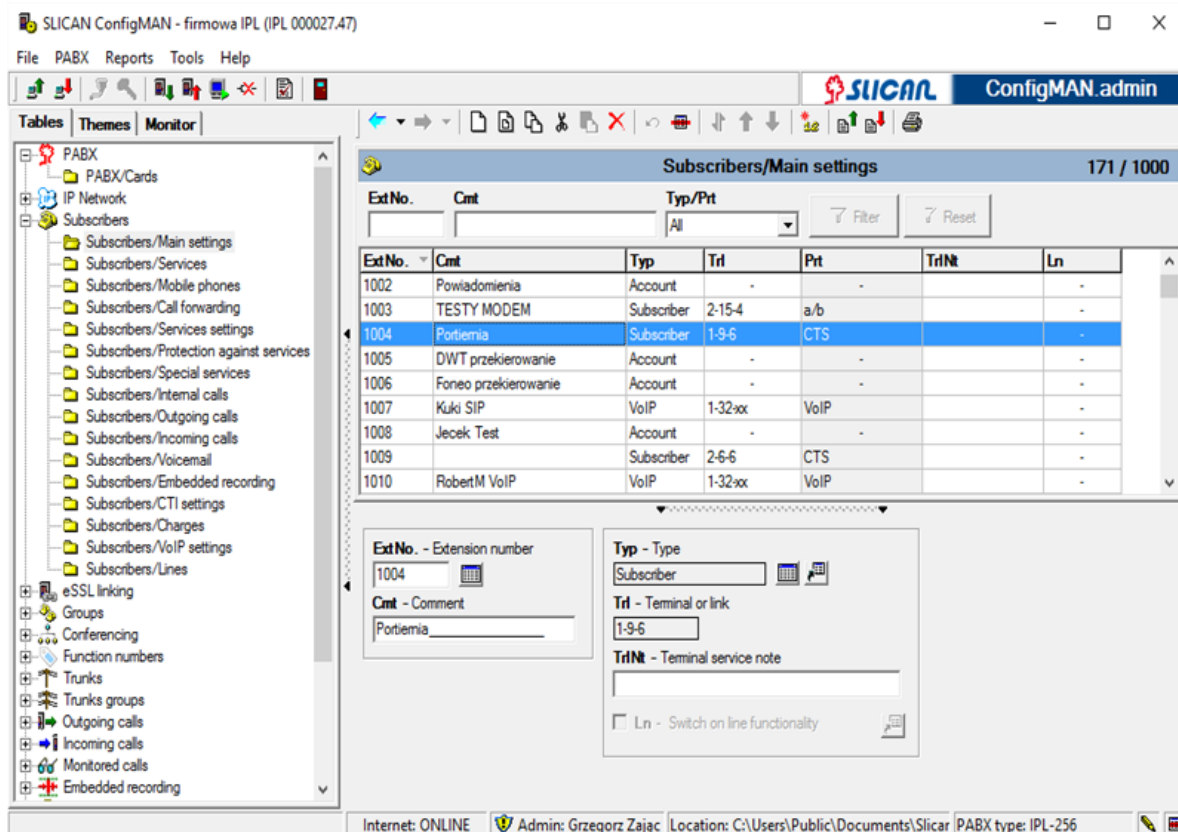


Fig. 4.6: Sheet Subscribers / Main settings

Special cases include:

- **VoIP** subscribers with additional settings, located in *Subscribers/ VoIP Settings sheet*
- Accounts with enabled line functionality, additional settings are located within *Subscribers/Lines* sheet

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  allows to generate secure password for subscriber (recommended),
- **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, how PBX manages this subscribers phone when unlogged. „damaged”, „disconnected (off)” or „do not disturb” can be set.
- **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**.

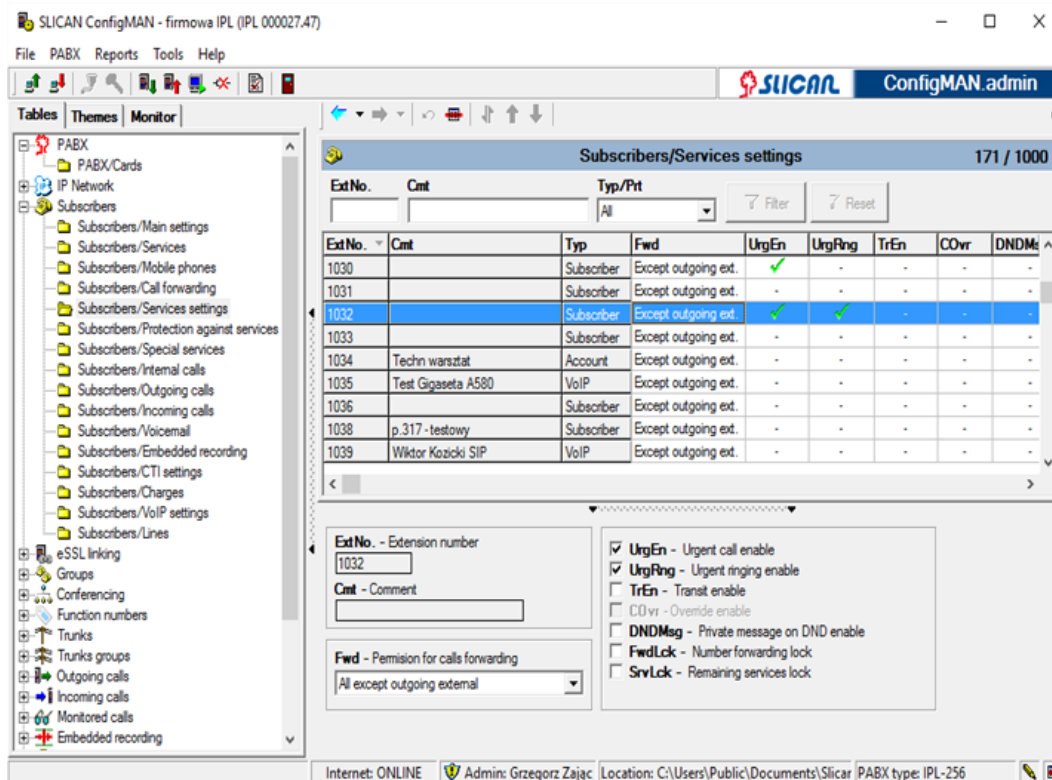


Fig. 4.7: Sheet **Subscribers / Services settings**

- **Fwd** – this field enables ability for subscribers to forward calls.

Available settings:

- None – subscriber can't forward any calls – PBX won't react to use of Transfer/flash phone button. Also call can't be suspended to make contact another subscriber.
 - Except outgoing external – subscriber can forward internal and incoming external calls but only to internal subscriber,
 - All – subscriber can forward all incoming internal and external calls, also to another external numbers
- **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 be able to use *urgent ringing* service (*35). If this service is used while dialing extension number equipped with analog phone special ringing will be heard (other than standard ringing), for extension numbers equipped with system phones caller immediately will be heard in speakerphone. This service is not realized 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.

- **TrEn** – transit enable – this settings enables permissions:
 - For subscriber logged from outside PBX to account (when using DISA, static path routing) allows to dial external number,
 - Subscriber can enable forwarding to external number with active transit (external number -> subscriber number with forwarding- >external number)
 - Allows to connect calls between two external numbers (subscriber call --> external number 1, hold call 1, subscriber call --> external number 2, connect external number 1 and 2; field **Fwd** in same sheet must be set to **All**).

When connecting transit calls at least single trunk must be digital trunk(ISDN, GSM or VoIP). Maximal time of transit call can be set up in field **TrTi** sheet **Outgoing calls**. Due to lack of proper signalling of analog trunks and risk with lack of proper disconnect in some cases, there is option of enabling transit on these trunks with [system command FXO Transit](#).

Transit calls are differentiated by PBX system, which are visible in charged billings reports from charging software *BillingMAN*.

- **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 *78 and set up using service *790. This message is played only when DND service is active on specified extension number. Checking this field is equal to create new entry in sheet **Incoming call/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. Subscriber can do this using proper service codes or [WebCTI](#) application.

It is possible to define following values for field **FwdStu** using **ConfigMAN**:

- Unconditionally,
- While busy,
- When no answer,
- While DND is on.
- unregistered account – available only for virtual accounts

For above options (can be setup using CM or service *72 on the phone) there are possible following five forwarding types and 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 regarding incoming calls to subscriber:

- **Standard** – for internal calls routed through disa voice message „Subscriber is temporarily inaccessible, please, call again later” is played and for direct external incoming calls, caller hears busy signal,
- **Announcement** – one of DISA/Infoclines announcements with number 0-9 from sheet **Announcements** is played,
- **Private message** – individually, using service *78, 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.

Detailed info regarding forwarding services using phone is available from article [Services description](#)

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).*

Firmware version 6.54 introduced timed option of temporary disabling forwarding and DND:

- **FwdLmt** – disable unconditional forwardign service
- **DNDLmt** – disable DND service

Disabling these functionalities can be set individually for each subscriber depending on one of three criteria:

- *limit by global setting* – according to **Globalne settings/Remain** tab
- *limit by own setting(single)* – at set hour service will be turned off
- *limit by own setting (cyclic)* – each day, at set hour service will be turned off

4.3.1.4 Internal calls

Nets in Slican PBX (**Subscribers/Internal Calls** tab, **Net** field) are kind of extension number set with appropriate privileges and restrictions for internal calls(doorphones, audio devices, groups, other types). 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. By default subscribers are assigned to net A and have access to every other net. By changing subscriber net or changing access to specific nets we can create separate groups which can or can't call to each other. Using this within PBX we can separate eg. Different companies or division working within single pbx, which shouldn't have access to each other or secretary-director setups, where only secretary can access director and director can access all other subscribers.

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 another phone, GSM or DECT phone. This integration is different depending on configuration and can simultaneously ring up main and extra phone or only extra phone. Functionality depends on type of used additional phone, if internal phone logged to pbx was chosen or external one outside of PBX:

- external number ex. cellular phone – PBX establish calls via appropriate trunk;if PBX is installed with GSM gateways, SMS notifications about caller number can be send (picked and not picked up)
- extension number ex. Wired or wireless DECT phone - PBX establish calls between internal ports

Depending on direction of call MobilePhone functionality is following:

- incoming call answered on MobilePhone can establish and 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** – **MblPh** – number of subscriber mobile phone,
- **MblEn** – authorization of MobilePhone service using,
- **MblClb** – calling back from MobilePhone number to PBX external number which was displayed during incoming call ringing, will route callback to caller of incoming call, even if this was external number. Time of routing path is set in **PthRt** field, *Incoming calls* sheet,
- **MblFwd** – allows Mobilephone user to forward incoming external call to another PBX internal subscriber,
- **Rng** – after checking this field mobile phone will be up-ringed after time defined in field **RngTi**,
- **RngTi** – time to up-ringing mobile phone,
- **RngDND** – after checking this field call will be immediately forwarded to mobile phone if DND service is active on master phone,
- **ClbEn** – after checking this field it is possible to log in and callback service based on MP number. Details in chapter [CallBack](#),
- **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 Centrum SMS trunk,
- **MsdFsh** – allows to choose type of SMS send to GSM phone(SMS or Flash SMS) – information with CLIP of unanswered caller,
- **Msd (dl, dB, DB, TF, G)** – set of filters to send SMS notifications depending on incoming unanswered call type – internal, external, direct, forwarded external, group forwarded or unanswered,
- **RcvSMS** – allows to choose type of SMS send to GSM phone(SMS or Flash SMS) – information with CLIP of answered caller,
- **Rcv (l, D, DB, TF, G)** – set of filters to send SMS notifications depending on incoming unanswered call type – internal, external, direct, forwarded external, group forwarded ,
- **VmlSMS** – allows to choose type of SMS send to GSM phone(SMS or Flash SMS) – information about voicemail,
- **SncNo** – enables synchronization of MobilePhone number in sheet *Subscribers/MobilePhone* with PBX phone book. Number can be visible or restricted.

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 lower 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. Any MobilePhone (non internal number) call disconnect won't cause ringing of master phone to stop. It is due PBX not being able to control state of external number. Operators use different methods of informing eg. announcement or disconnect when subscriber is busy, damager or outside of GSM network range
- SMS notification with CLIP of caller happens after few calls, one after another from same number is send only once within 30 minutes. Each call from different number to this mobile phone will reset counter and allow to instantly send notification.
- 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 on mobile phone up ringing
 - *77 1 – switching on mobile phone up ringing, if don't answer or DND
 - *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 45 seconds to transfer this call, during this time:
 - during 15 seconds there will be silence,
 - during next 30 seconds route signal can be heard,
- if during this time connection between mobile phone and PBX will not be established, waiting call will be interrupted (disconnected)
- if we cal from GSM Mobile phone: to internal subscriber number to whom we want to transfer or eg. external PBX number with DISA and then choose internal:
 - if we want to announce call then wait as target subscriber will pic kup, inform him about transfer and disconnect
 - if announcing isn't needed, disconnect after hearing call signal
- after our GSM Mobilephone will connect call to PBX our original call partner will hear MoH and announcement „please wait for connection”, after that call signal until picking up.
- waiting subscriber will be connected with target subscriber instantly, if he is busy, after his disconnect.
- 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. *Callback is connected with Mobilephone, extra license isn't necessary (since fw. 6.50). More info is available in chapter [Licenses](#)*

Using MobilePhone as 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 phones 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) when 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 GSM phone. If MobilePhone is set as wireless phone (internal extension) and call forwarding to GSM number then::

- when MobilePhone is active – calls are routed to extra MobilePhone (wireless), standard forwarding isn't working,
- when MobilePhone is not active – standard forwarding is working (to GSM phone).

Service activation or deactivation is set by phone (main or additional one) with service codes described above. There is no need to change forwarded number each time.

4.3.1.6 Callback

Callback is service related, interjoined with MobilePhone. This service enables to order call from PBX, giving call signal and allow to call from MobilePhone main extension. Cost of this call is assigned to caller extension number and while calling original extension CLIP is shown.

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 (use static path routing to Callback service)
- 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 heard. During this signal target number (internal or external) can be dialed.

Single static path routing entry can be used by every Callback service user. PBX will compare number of calling user to number of Mobilephone user. Jeden wpis statycznego trasowania może być wykorzystywany przez wszystkich użytkowników usługi CallBack. Centrala uruchamia regułę przez porównanie numeru dzwoniącego z numerem MobilePhone użytkownika.

ATTENTION!

*If call external numbers, field **TrEn** – transit enable must be set, within **Subscribers/Services Settings** sheet for this specific subscriber.*

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 **MbIPh** field according previous chapter [MobilePhone](#) and setting static path routing type as CallBack, it is described in chapter w [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. Service works with internal and external traffic. Through Incoming Calls table traffic can be routed via DISA and also directly to subscriber.

How service works:

- First caller chooses number, hears call signal until target subscriber will pick up
- Next caller – after calling number that depends on settings – call signal/standard MoH/ specific announcement – with information about queue position according to calling order
- Callers which try to call but go over queue limit will get disconnected

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 custom information
- **AnnBeg** - allows for Call Waiting announcement to be played from the beginning for every caller that is placed in queue, if **WtMel** is set with custom announcement,
- **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 announcements to PBX configuration,
- **CWen** – for VoIP and ISDN phones CW can be disabled within PBX to allow using dedicated functionality of the phones .

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 [Protection against services](#).

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*:

- **Dpt** – enables to assign every extension number to one of 99 departments (added in *Descriptions/Dzialy* sheet). As **Department** term we mean logically seperated part of company, division or seperate (another) company. Assigning subscribers to departments allows :
 - Creating raports according to company structure with **BillingMAN** application
 - As manager of [WebCTI](#) application – set up some of subscribers services (assigning MobilePhone number, forwarding, DND, I am here, blocking phone, ussing account, alarm clock, phone book settings, recording files access)
- **PubPB** – Defines access to Public Phone Book. Following options are available for this field:
 - **No access**
 - **Read only**
 - **Read 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
 - **Read and write** - . Last option is available only for CTS system phones and [MessengerCTI.Desktop/ WebCTI](#) application.
- **PrvPB** – allows to access private phone book.

Allowing subscribers to access phone book allows him to choose numbers that belong to in using short numbers, by using ****ID** (ID – three digit number from 001 to 999). This method of reaching numbers from public phone book is available to all subscribers.

- **RegRng** *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.
- **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.

UWAGA!

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

- **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**.
- **AlrEn** – enabling activates notification about damaged subscribers ports
- **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. Subscriber won’t be able to use services related to forwarding, if there is external number on hold (*32, *36) . 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.
- **Lng** – changes language settings of CTS display for specific subscriber. 3 languages are implemented: Polish, English and German.
- **AnnSet** – changing system announcements settings for specific subscriber. Allows to use standard system announcements built into PBX firmware or one of 3 additional sets. Useful option for PBXes that handle traffic of subscribers using different languages or PBX owner wants to set up his own, customised announcements. Custom announcements should be

recorded and uploaded to PBX (according to chapter [Announcements](#)) and set up in sheet i ustawić w odpowiednim zestawie w zakładce **Global Settings/System announcement/Announcement sets**. If specific set doesn't have custom announcement in some spots, those will play default announcement instead.

- **EnAt** – enables [Auditorium](#) mode for this phone.

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**

The screenshot shows the ConfigMAN application window. The left sidebar contains a tree view with the following structure:

- PABX
 - PABX/Cards
 - IP Network
 - Subscribers
 - Subscribers/Main settings
 - Subscribers/Services
 - Subscribers/Mobile phones
 - Subscribers/Call forwarding
 - Subscribers/Services settings
 - Subscribers/Protection against services
 - Subscribers/Special services
 - Subscribers/Internal calls
 - Subscribers/Incoming calls
 - Subscribers/Voicemail
 - Subscribers/Embedded recording
 - Subscribers/CTI settings
 - Subscribers/Charges
 - Subscribers/VoIP settings
 - Subscribers/Lines
 - eSSL linking
 - eSSL linking/Main settings
 - eSSL linking/PABXes
 - eSSL linking/Extension numbers
 - Groups
 - Conferencing
 - Function numbers
 - Trunks
 - Trunks groups
 - Outgoing calls
 - Incoming calls
 - Incoming calls/Settings/
 - Incoming calls/Settings/Main
 - Incoming calls/Settings/Additional
 - Incoming calls/Infolines
 - Incoming calls/Infolines API
 - Incoming calls/Static path routing
 - Incoming calls/Number presentation
 - Monitored calls
 - Embedded recording
 - Time
 - Notifications
 - Phone books
 - Descriptions
 - Global settings

The main window displays the 'Subscribers/Services' tab with a table of 13 subscribers. The table columns are: Ext No., Cmt, Typ/Prt, Acc, HAcc, Lck, PIN, AirC, AirTi, GrpLmt, and EnGrpH. The table data is as follows:

Ext No.	Cmt	Typ/Prt	Acc	HAcc	Lck	PIN	AirC	AirTi	GrpLmt	EnGrpH
120	120	Subscriber	<None>	0:00	-	-	-	<None>	Switched off	-
121	Abonent 120	Subscriber	<None>	0:00	-	-	-	<None>	Switched off	-
1220	Rafal testowy	VoIP	<None>	0:00	-	-	-	<None>	Switched off	-
1221	Rafal konto dla 1220	Account	<None>	0:00	-	****	-	<None>	Switched off	-
1906	Abonent 1906	Subscriber	<None>	0:00	-	-	-	<None>	Switched off	-
1907	Abonent 1907	Subscriber	<None>	0:00	-	-	-	<None>	Switched off	-
1908	Abonent 1908	Subscriber	<None>	0:00	-	-	-	<None>	Switched off	-
2000	Abonent 1220	Voice mail account	<None>	0:00	-	-	-	<None>	Switched off	-
2001	Nowy Abonent 2001	Subscriber	<None>	0:00	-	****	-	<None>	Switched off	-
2002	Abonent 2002	Subscriber	<None>	0:00	-	-	-	<None>	Switched off	-
2003	Abonent 2003	Subscriber	<None>	0:00	-	-	-	<None>	Switched off	-
2004	Abonent 2004	Subscriber	<None>	0:00	-	-	-	<None>	Switched off	-
2005	Abonent 2005	Subscriber	<None>	0:00	-	-	-	<None>	Switched off	-
2006	Abonent 2006	Subscriber	<None>	0:00	-	-	-	<None>	Switched off	-
2007	Abonent 2007	Subscriber	<None>	0:00	-	-	-	<None>	Switched off	-
2008	Abonent 2008	Subscriber	<None>	0:00	-	-	-	<None>	Switched off	-
2009	Abonent 2009	Subscriber	<None>	0:00	-	-	-	<None>	Switched off	-
2010	Abonent 2010	Subscriber	<None>	0:00	-	-	-	<None>	Switched off	-
2011	Abonent 2011	Subscriber	<None>	0:00	-	-	-	<None>	Switched off	-
2012	Abonent 2012	Subscriber	<None>	0:00	-	-	-	<None>	Switched off	-
2013	Abonent 2013	Subscriber	<None>	0:00	-	-	-	<None>	Switched off	-

Below the table, there are configuration fields for the selected subscriber (Ext No. 120):

- Ext No. - Extension number:** 120
- Cmt - Comment:** 120
- PIN - PIN-Code:** (empty field)
- Lck - Phone lock:** ☐
- Acc - Recall of fixed account:** (empty field)
- HAcc - Account deactivation hour:** 0
- AirC - Enable alarm call:** ☐
- AirTi - Alarm call at:** 00:00
- GrpLmt - Time limit for login in group:** switched off

Fig 4.8: sheet Subscribers / Services

Settings available in SERVICES tab:

- **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 its deactivation. From locked phone fixed and temporarily account can be recalled. Unlocking phone deactivate fixed account recalling.
- **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. Additional field **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.

- *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*).

Firmware 6.42 added option of setting up maximum time of repeating alarm clock. It may be setup as 1, 3, 5, 7, 9, 11, 13 or 15 minutes. Default time of ringing is 40 second after which is 80 second pause. Setting up maximum time of ringing is related to notifications about realising alarm clock within **HotelP** protocol. If within set time wake up won't be realised – there will be busy signal or subscriber won't pick up and set – information about not realised wake up will be send. There is option to change alarm clock time (global for all subscribers) within tab *Global settings/Remain* *WTi* field.

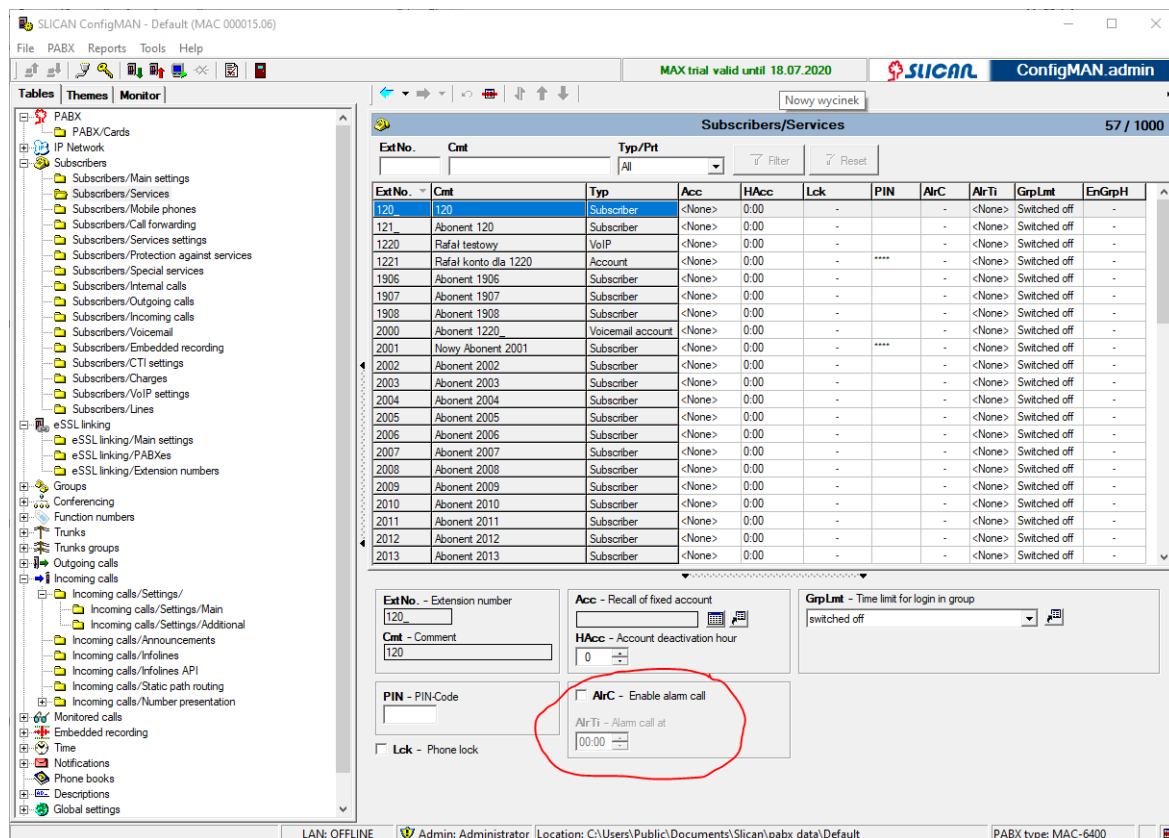


Fig 4.9: Alarm clock settings

Firmware version 6.54 presented option of timed logging out from group for subscribers - **GrpLmt**

Logging out of group can be set individually for each subscriber according to one from these three criteria::

- *limit by global settings* – setting from tab *Global settings/Remain* will be used

- *limit by own settings (single)* – at set hour subscriber will be logged out from all groups he was part of, this will happen only one time
- *limit by own settings (cyclic)* – each day, at set hour subscriber will be logged out from all groups he was part of

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.

ExtNo.	Cmt	Typ	CLIP	CLIR	CLIRt	VIP	PrfHL	DayCls	NgtCls
2007	Abonent 2007	Subscriber	Own ExtNo.	-	-	-	<None>	B. Średnia	B. Średnia
2008	Abonent 2008	Subscriber	Own ExtNo.	-	-	-	<None>	B. Średnia	B. Średnia
2009	Abonent 2009	Subscriber	Own ExtNo.	-	-	-	<None>	B. Średnia	B. Średnia

ExtNo. - Extension number: 2009
Cmt - Comment: Abonent 2009
CLIP - ExtNo. presentation outside: Own extension number
☐ **CLIR** - Number presentation lock
☐ **CLIRt** - Permission for temporarily CLIR
☐ **VIP** - Higher calling priority
PrfHL - Preferred hot line: <None>
DayCls - Day access class for outgoing calls: B. Średnia
NgtCls - Night access class for outgoing calls: B. Średnia
PrvCls - Private access class for outgoing calls: G. Pol. prywatne
DayPrg - Day privileges to prefixes: C. Kraj
NgtPrg - Night privileges to prefixes: C. Kraj
PrvPrg - Private privileges to prefixes: D. Kraj + komórki
PhLck - Automatically phone locking: Switched OFF
☐ **DNDLck** - Synchronizing DND and phone lock

n: Administrator Location: C:\Users\Public\Documents\SI PABX type: MAC-6400

Fig 4.10: Sheet **Subscribers / Outgoing calls**

- **None** – subscriber doesn't use preferred hot line.
- **Abonent** – jeżeli abonent po podniesieniu słuchawki nie wybierze jakiegokolwiek numeru, to po czasie określonym w polu **HLOfs** arkusza **Ruch wychodzący** nastąpi automatyczne wybranie numeru abonenta przypisanego w polu **PrfHL**.
- **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 field **PrfHL** 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 field **PrfHL** 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.

1.1.1 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 these options are available:

- Another subscriber number – subscriber is presented with number of another subscriber,
- Group number – subscriber is presented with group extension number,
- Function number - subscriber is presented with function extension number
- eSSL number – subscriber is presented with linked PBX number

Additional settings regarding presentation:

- **CLIRi** – after checking this field, subscriber won't be showing presentation of his phone number or comment when call other subscribers with CTS, analog (with FSK) and VoIP phones
- **HidExt** – allows to hide phone extension number and comments for phone book of CTS phone and CTI applications ([MessengerCTI](#)/ [WebCTI](#)/ [ConsoleCTI](#)),

4.3.2 Groups – creating, settings, adding subscribers

To expand organising PBX traffic – subscribers (also those from linked PBX with eSSL) can be assigned to chosen groups and organise internal and incoming traffic

Types of groups within PBX:

- upringing group
- cyclic group
- linear group
- ACD group
- paging group
- conferencing

Maximum, total number of groups :upringing, cyclic, linear and conferences for each PBX:

- 20 –IPS-08 and *CXS-0424*,
- 25 – *CCT-1668*,
- 80 – *MAC-6400*, *IPL-256*, *IPM-032*, *IPU-14*.

For remaining groups, their maximum amount for different PBX is:

- ACD - 8
- paging - 8

Depending on firmware version, amount of available groups to create may differ. This amount can be checked by clicking **Groups** sheet within **ConfigMAN** tree.

4.3.2.1 Upringing, cyclic and linear groups

Upringing groups

In up-ring 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-ring 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.

Example: there are four group agents – 2001, 2002, 2003, 2004. If during call connection within group agent 2001 was ringing up, next group call will ring up agent with extension 2002 (if it isn't busy). 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**.

Example: there are four group agents – 2001, 2002, 2003, 2004. If during call connection within group agent 2001 was ringing up, next group call will ring up same agent (if it isn't busy). Parameter **Ti** enables to define how long specified agent will be call before next one.

ATTENTION!

In linear group call forwarding from group cannot be realized.

Creaint groups: upringing, cyclic, linear

To create group, go to proper sheet:

- *upringing* → **Groups / Upringing / Settings**,
- *cyclic* → **Groups / Cyclic / Settings**,
- *linear* → **Groups / Linear / Settings**

then add new group instering new record. When creating group, **ExtNo.** – extension number and **Cmt** – name.

ATTENTION!

When **Group** doesn't have extension number – it can be reached from incoming/internal traffic only by **Function Numbers**

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

- **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.
- **Res** – 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 90 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 (no announcement/disa). 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.
- **MaxFwd** – this field enables to set maximal number of simultaneous call forwarding from group (this parameter is available only for up-ringing group).
- **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.
- **HidExt** – hide group extension number in phone book of CTS system phones.

- **CNtf** – enables zaznaczenie pola powoduje wyłączenie składnikom powiadomień o nieodebranych połączeniach na grupę (parametr dostępny tylko dla grupy rozdzwaniania).
- **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 necessary 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.

Functionality works for external and internal calls if in Incoming Calls table user announcement or DISA is set up. Depending on selected option, following settings are possible:

- ***waiting time + position in queue*** - ***waiting time*** message is play once after 15 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 below)
- ***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 15 seconds then 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."

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 / Uprising / Agents*** or
- ***Groups / Cyclic / Agents*** or
- ***Groups / Linear / Agents***.

. Every group upringing, cyclic or linear can consist of up to 12 agents (subscribers). For specified subscriber upringing method can be defined.

4.3.2.2 ACD group

ACD group is special group with automatic call distribution.

Zasada działania:

- 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.
- If first call is still ringing or subscriber agent is already during call, next agents will ring up until all of them are busy..
- If all agents are busy – next call will be in queue, with announcements informing about queue places and waiting time
- Depending on chosen ACD group type – *cyclic* or *rest time spoczynku* – different ringing strategy will occur (described below).

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..

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. If group doesn't have extension number, it can be reached in internal/external calls using **Function Numbers**.

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.

Available settings:

- In field **GrTyp** – group type, type of group should be defined:
 - **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 depending on time set by administrator, 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. Agent that didn't talk for longest time will be first to ring up.
- In field **GrMng** you can define extension number of subscriber who manage ACD group. If manager have CTS-2/3xx he can observe group statistics on phone screen. Full information is in manual for above phones. 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 in 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 or for incoming calls chosen with digits during DISA.
- 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.
- fields **AnnQ**, **MaxT** oraz **MaxQ** were described above, within description of groups: upringing, cyclic i linear

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. You can only, using buttons **Add** and **Remove**, add or remove agents in specified ACD group. After pressing **Add** key following window can be visible:

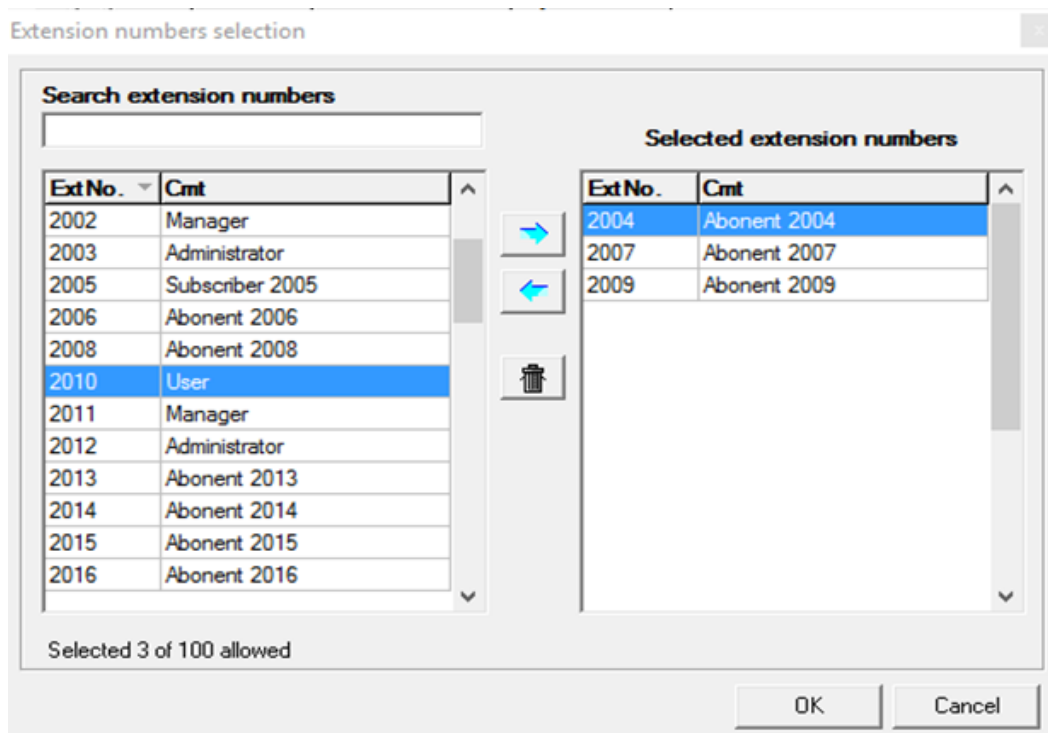


Fig 4.11: Window for selecting ACD group agents

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), CTRL + A (all records). Removing records can be performed by selecting them in right part of widow and clicking arrow directed to the left..

4.3.2.3 Paging group

Grupa pagingowa służy do **Paging group** allows to simultaneous notificat 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 (gong), then the initiator 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 (A ... H) to which user can add subscribers. Only system phones and analogue Audio devices, configured as Paging (more about configuring analogue port in chapter [Audio devices configuration](#)). Agents assigned to **Paging groups** can also be a part of other 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 Paging 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**.

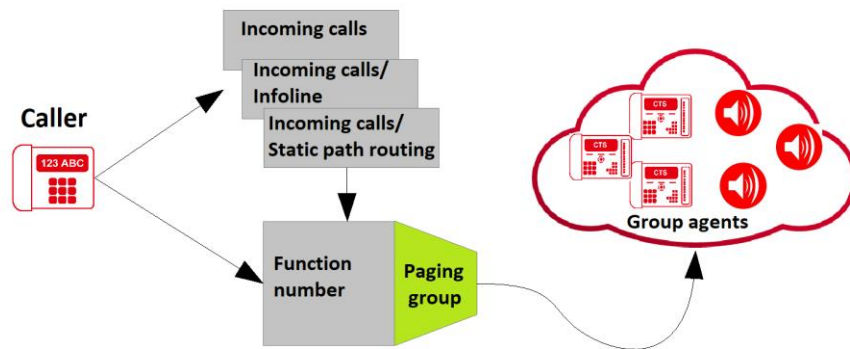


Fig 4.12: Paging Group – Operating principle

Operating principle:

Initiating caller will hear melody while calling. In meantime group agents (System phones and audio speakers) will hear two twone gong and CTS phone will show comment of proper function number. After gong sound will be played, initiator will hear short signal after which he can provide his message (his voice can be heard by group agents). After finishing message, initiator hangs up handset and gong tone is played again.

Paging group has higher priority than DND, FWD, MobilePhone and reservation services.

4.3.2.4 Conference group, group 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”.

Actual number of conferences that can be created depends on bought licenses.

Due to limited amount of maximum users connected to conference bridge within PBX, within **ConfigMAN** from sheet **Conferencing /Group /Participants** i **Conferencing /Rooms /Participants**, we can set up to:

Limitations regarding conferencing:

<i>PBX type</i>	<i>Maximal number of participants (sum of them in all conferences)</i>	<i>Maximal number of simultaneous conferences</i>
MAC-6400, CXS-0424	30	10
IPS-08, CCT-1668 (FW <=4.11)	8	4
IPL-256, IPM-032, IPU-14, CCT-1668 (FW >=5.11)	12	4

Fig 1: Limitations concerning conferences**Conference group**

Conference group enables to establish conference calls. Last participant in conference (12 in *IPL-256*, *IPM-032*, *IPU-14*, *CCT-1668*, 8 in *IPS-08*, 30 in *MAC-6400* and *CXS-0424*) 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:

- Internal calls – with Function Number
- infoline
- incoming calls – through incoming calls table - field **ExtNo.** sheet **Incoming calls/Settings/Main**,
- static path routing

If subscriber equipped in any system phone *CTS* (with exception of *CTS-102.HT*) 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 [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.

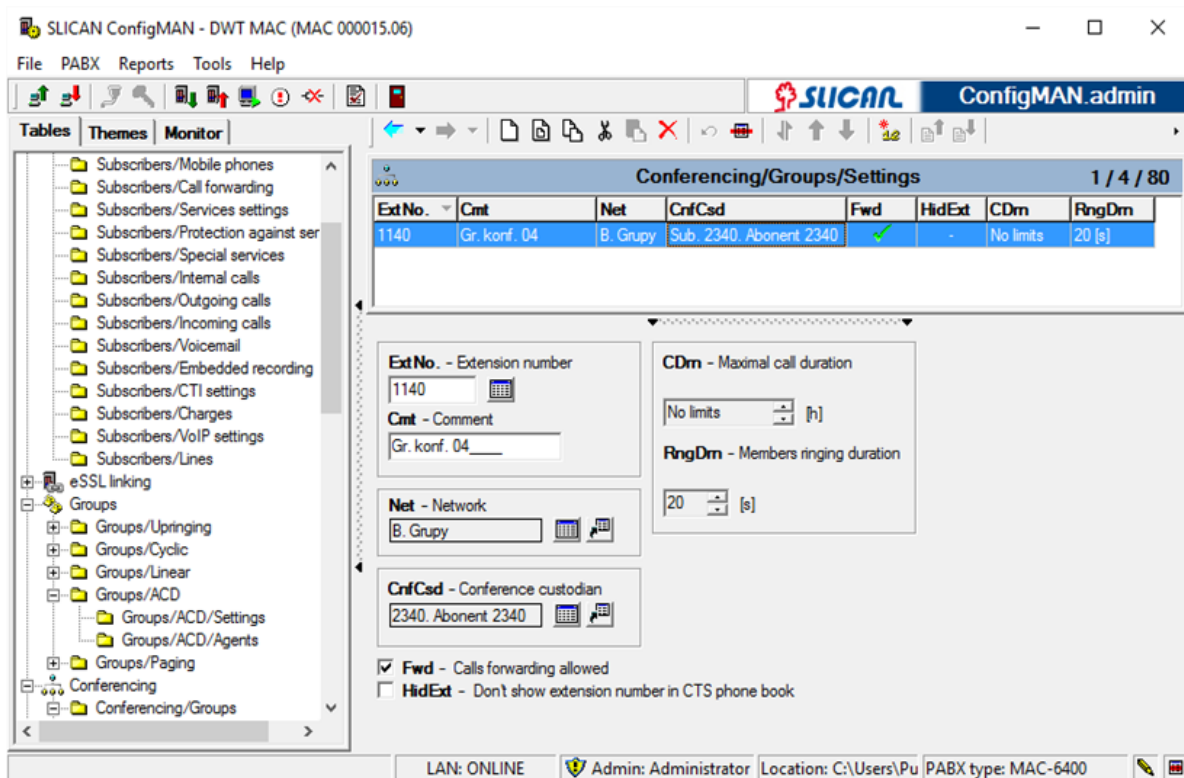


Fig 4.13: sheet Conferencing / Groups / Settings

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** – set to which net groups are assigned. With those permissions for calls from subscribers and group regarding internal and incoming calls can be set.
- **CnfCsd** – in this field you should define conference custodian, only subscriber equipped in *any CTS phone(expect CTS-102.HT)* 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 **RngDrm**).

- **HidExt** – conference internal number won't show up in phone book of CTS and applications [MessengerCTI/](#) [WebCTI](#)

- **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.

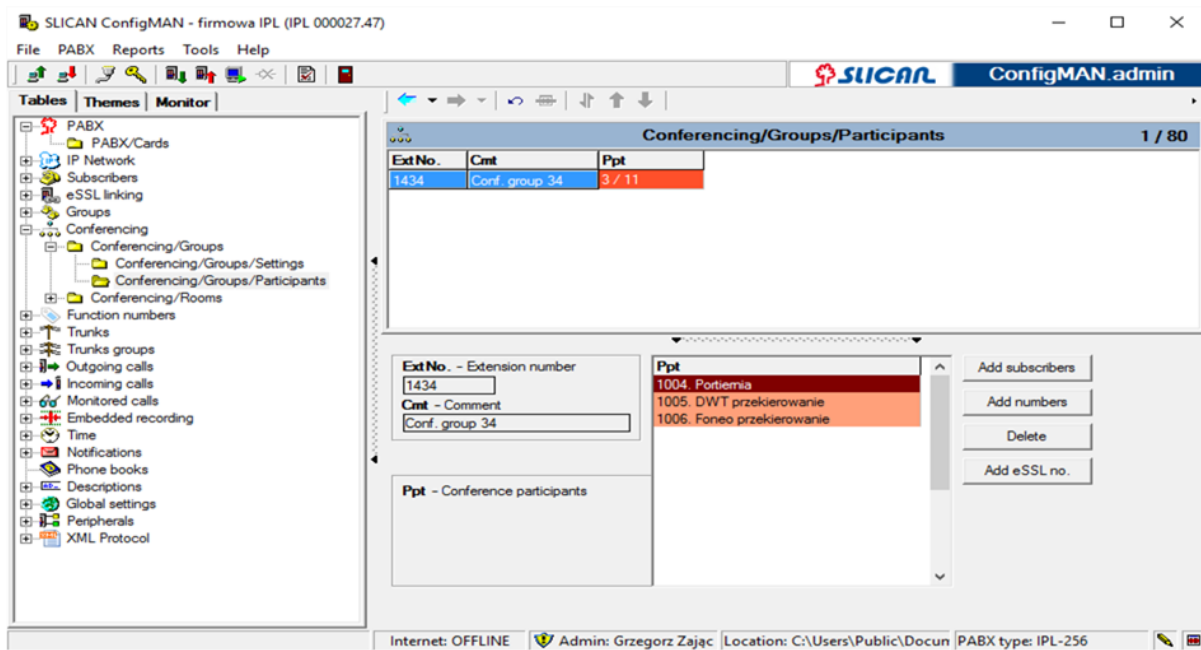


Fig 4.14: sheet *Conferencing / Group / Participants*

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 4.15: Functionality of conferencing 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**).

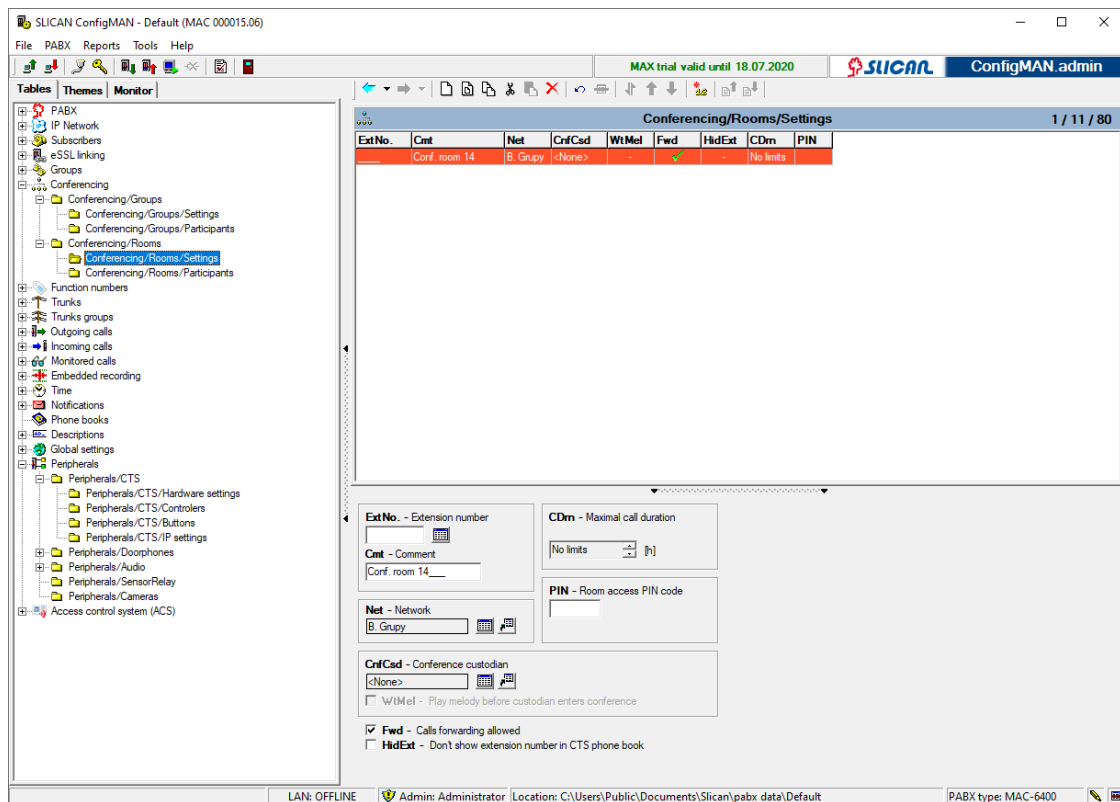


Fig 4.16: sheet Conferencing / Rooms / Settings

Remaining 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 phone (except CTS-102.HT)*. 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** – 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). After field is checked and subscribed have forwarding active as above eg. for voicemail:
 - unconditional (*74) – no ringup and no forwarding to voicemail
 - when no answer (*76) – subscriber will ringup without forwarding to voicemail.

If the field isn’t checked, call won’t be forwarded despite being active([hone will call for time set in field **RngDrn**).

- **HidExt** – conference internal number won't show up in phone book of CTS and applications [MessengerCTI/ WebCTI](#)
- **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 phone (except CTS-102.HT)* 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 [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.

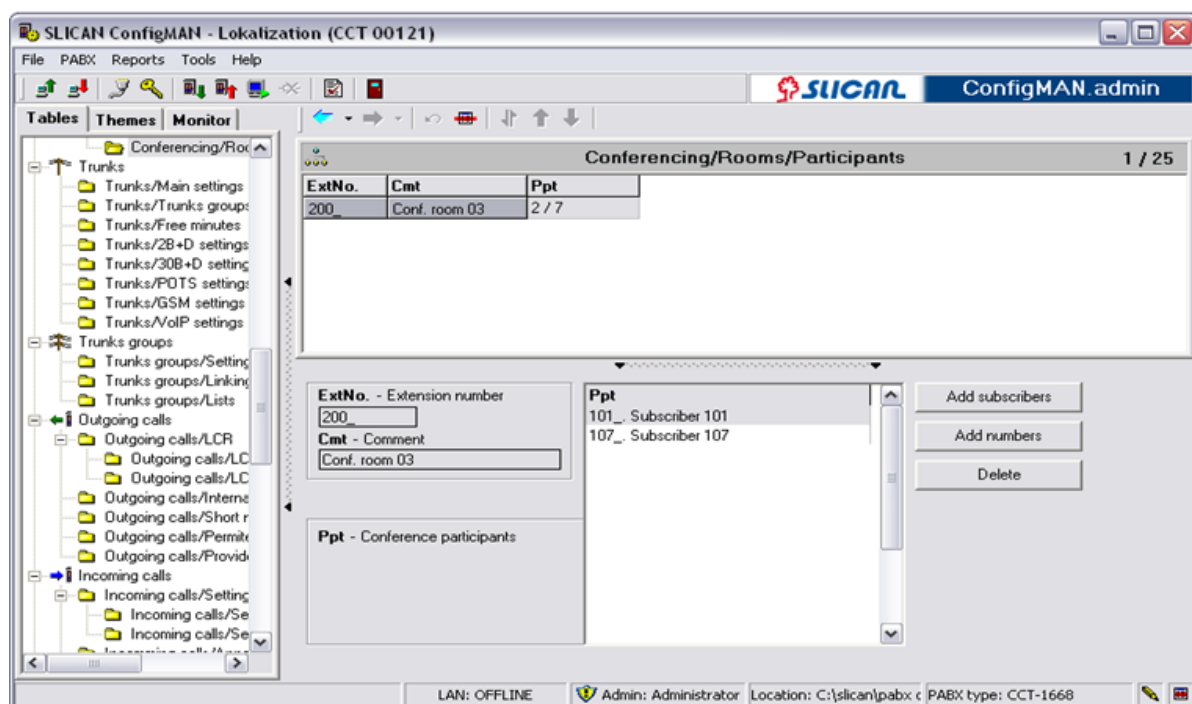


fig 4.17: sheet **Conferencing / Rooms / Participants**

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.
- **Trunks group** – it is defined trunks set of the same type and the same configuration (ex. POTS, 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 Trunk without trunk list

If functionality related to assinging trunk to trunk list isn't needed (restricting call channels, recording calls from trunk group, dynamic call routing, call tolls, itd.), trunk can be created without assigning to trunk group in case of linking trunk group. Above method don't apply to linking over ISDN.

4.4.2 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..

Since fw. 6.42 there is option of managing traffic withink trunk group - dynamic channel reservation. Due to rigd seperation of incoming and outgoing traffic withink trunk group isn't enough – fluid option of reserving single channel was added. With this setting PBX will keep track to leave one trunk group channel as reserved. Reserve doesn't apply to VoIP type subscriber (breaks reserve).

Field **DynRes** - set how dynamic channel reservation works:

- None – dynamic channel reservation doesn't work
- For outgoing calls
- For incoming calls

Example:

- PBX have 3 ISDN trunks (6 channels)
- High incoming calls traffic to PBX
- Set up field **DynRes** for trunk group and outgoing traffic
- Trunk group will always (dynamicaly) will reserve 1 channel needed for outgoing call

IMPORTANT!

Any trunk may be a member of only one trunk group in the same time.

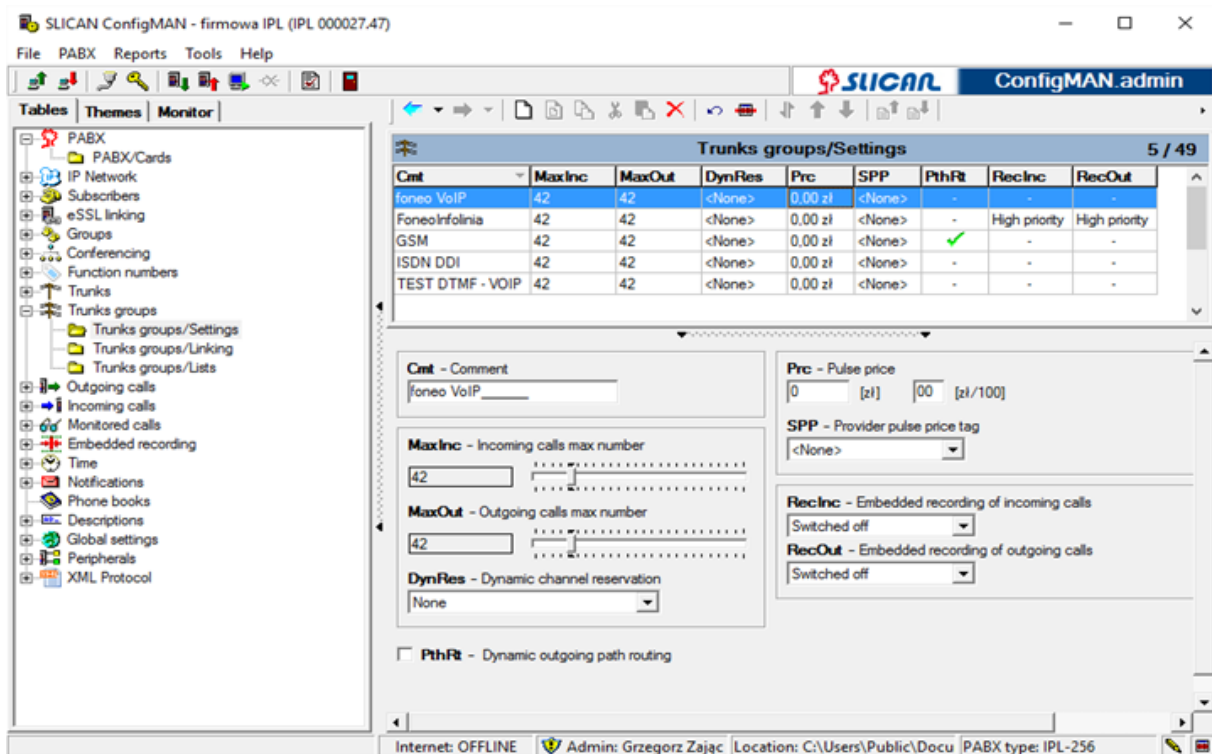


Fig 4.18: Sheet Trunk groups / Settings

Prc field allows to set up price tag for pulse during call. This is available for operator using tariffication pulses over ISDN. 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 directly to the extension number which was the caller of above mentioned outgoing call. This procedure is described in details in chapter [PathFinder – 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 and set up price per pulse within this software.

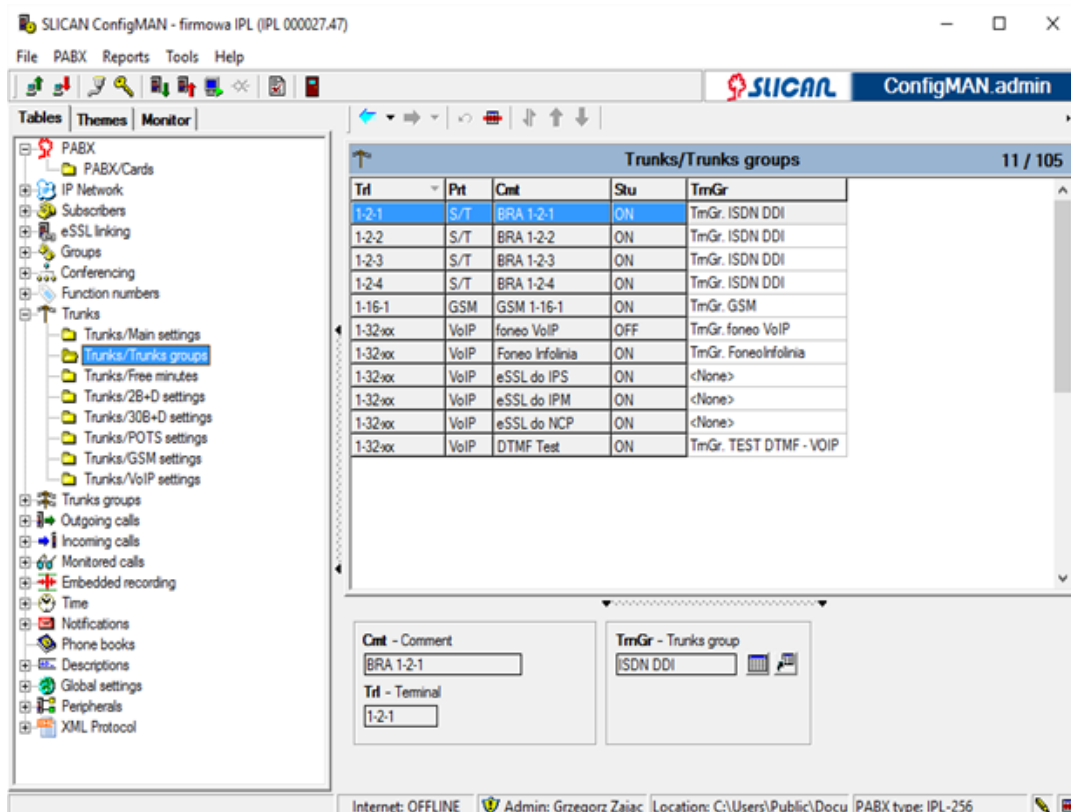
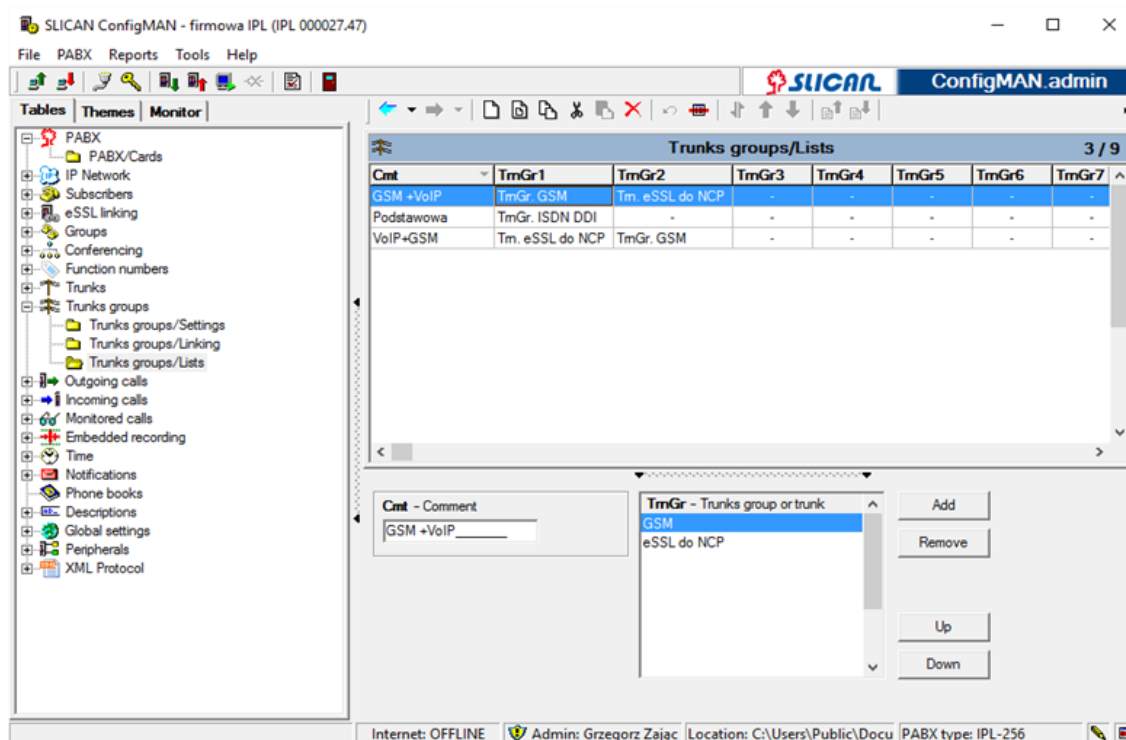


Fig 4.19: Sheet Trunks groups / Settings

Assigning trunks to trunks group can be done in sheet **Trunks/Trunks groups**. For specified trunks, trunks group can be indicated.

4.4.2.1 Defining and setting parameters of trunk list

Fig 4.20: Sheet *Trunk groups / Trunks lists*

Trunks list sort trunk groups and/or translation according to set order. This allows to set outgoing calls strategy for user according to only single entry. Strategy of using channels depends on records order on the list (also depends on trunk being busy or trunk failure). Channels will be used from trunk listed in **TrnGr1**, then next entries. 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. List can contain different trunk groups and trunks, from different operators if their method of choosing (reaching) target number is consistent.

4.4.3 Privileges for outgoing calls

4.4.3.1 Privileges and privileges 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.

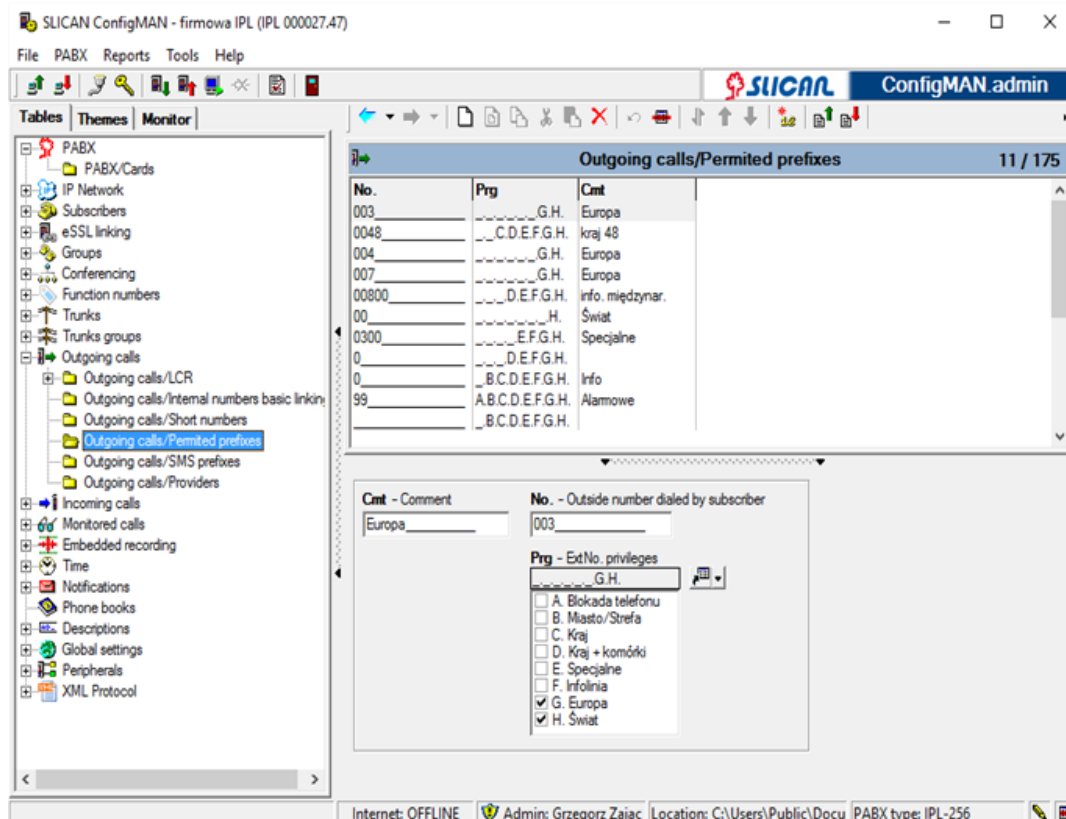


Fig 4.21: Sheet Outgoing calls / Permitted prefixes

ATTENTION!

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

After defining A class for specified prefix, established calls with this prefix are not charged.

4.4.3.2 Assigning access classes to subscribers and accounts for outgoing calls

Access class enables (or not) to use entries in tables (use trunk, trunk group or list) 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

The screenshot shows the SLICAN ConfigMAN interface for MAC 000015.06. The main window displays the 'Subscribers/Outgoing calls' configuration page, showing a table of subscribers and a detailed configuration panel for a selected subscriber.

Subscribers/Outgoing calls table:

ExtNo.	Cmt	Typ/Prt	CLIP	CLIR	CLIRt	VIP	PrvHL	DayCls	NgtCls	Pr
121	Abonent 120	Subscriber	Own ExtNo.	-	-	-	<None>	C. Wysoka	C. Wysoka	B.
1220	Rafal testowy	VoIP	Own ExtNo.	-	-	-	<None>	B. Srednia	B. Srednia	B.
1221	Rafal konto dla 1220	Account	Own ExtNo.	-	-	-	<None>	<Full>	<Full>	<F
1906	Abonent 1906	Subscriber	Own ExtNo.	-	-	-	<None>	B. Srednia	B. Srednia	B.
1907	Abonent 1907	Subscriber	Own ExtNo.	-	-	-	<None>	B. Srednia	B. Srednia	B.
1908	Abonent 1908	Subscriber	Own ExtNo.	-	-	-	<None>	B. Srednia	B. Srednia	B.
2000	Abonent 1220	Voicemail account	Own ExtNo.	-	-	-	<None>	B. Srednia	B. Srednia	B.
2001	Nowy Abonent 2001	Subscriber	Own ExtNo.	-	-	-	<None>	B. Srednia	B. Srednia	B.
2002	Abonent 2002	Subscriber	Own ExtNo.	-	-	-	<None>	B. Srednia	B. Srednia	B.
2003	Abonent 2003	Subscriber	Own ExtNo.	-	-	-	<None>	D. Klasa dostępu D	D. Klasa dostępu D	B.
2004	Abonent 2004	Subscriber	Own ExtNo.	-	-	-	<None>	B. Srednia	B. Srednia	B.
2005	Abonent 2005	Subscriber	Own ExtNo.	-	-	-	<None>	B. Srednia	B. Srednia	B.

Configuration panel for ExtNo. 121:

- ExtNo.:** 121
- Cmt:** Abonent 120
- CLIP:** Own extension number
- CLIR:** ☐ Number presentation lock
- CLIRt:** ☐ Permission for temporarily CLIR
- VIP:** ☐ Higher calling priority
- PrvHL:** Preferred hot line: <None>
- DayCls:** Day access class for outgoing calls: C. Wysoka
- NgtCls:** Night access class for outgoing calls: C. Wysoka
- PrvCls:** Private access class for outgoing calls: B. Srednia
- DayPrv:** Day privileges to prefixes: D. Kraj + komorki
- NgtPrv:** Night privileges to prefixes: D. Kraj + komorki
- PrvPrv:** Private privileges to prefixes: D. Kraj + komorki
- PhLck:** Automatically phone locking: Switched OFF
- DNDLck:** ☐ Synchronizing DND and phone lock

Fig 4.22: Sheet *Subscribers / Outgoing calls*

4.4.3.3 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.

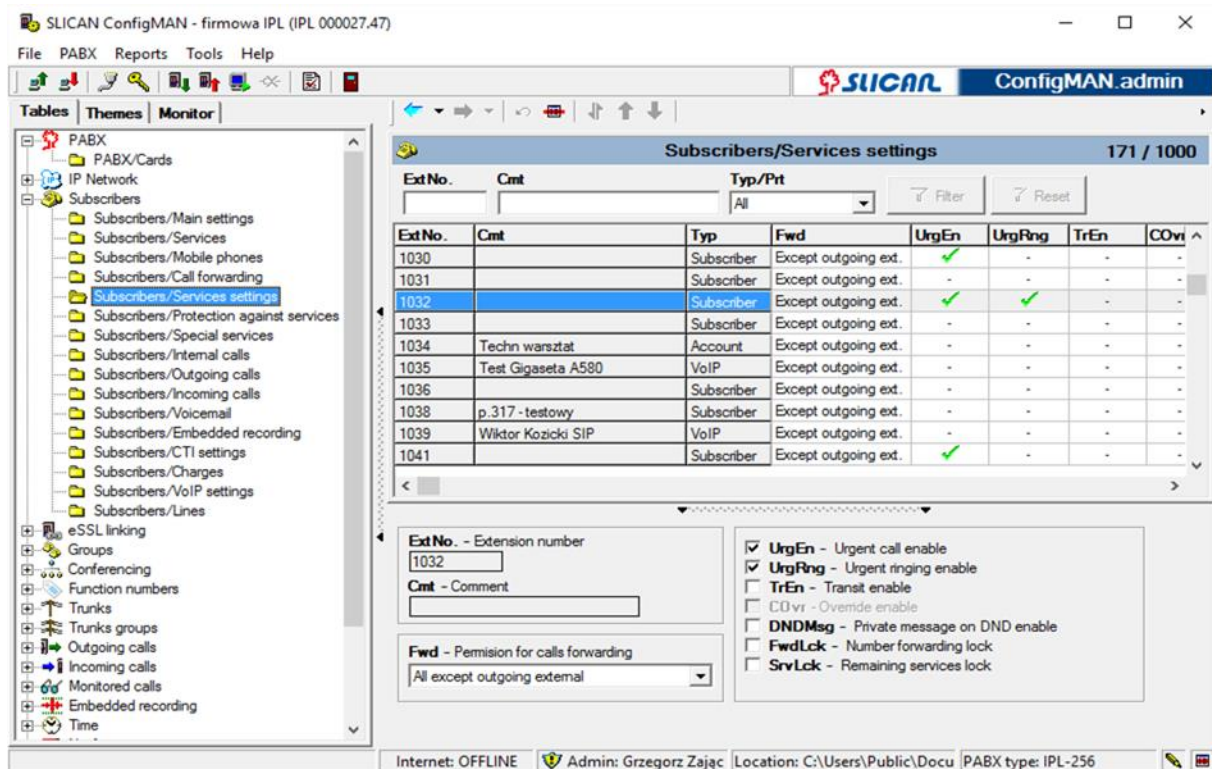


Fig 4.23: Sheet *Subscribers / Services settings*

4.4.4 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
- other extension number,
- group number,
- conference number,
- infoline number,
- trunk own number,
- number defined by user.

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 (routing external number from thinks trunk group to subscriber).

ATTENTION:

Subscriber's number presentation for outgoing calls depends on content of **TrnGr** - *trunk* and **DINo.** – *dialed number* field in sheet **Incoming calls/Settings/Main**.

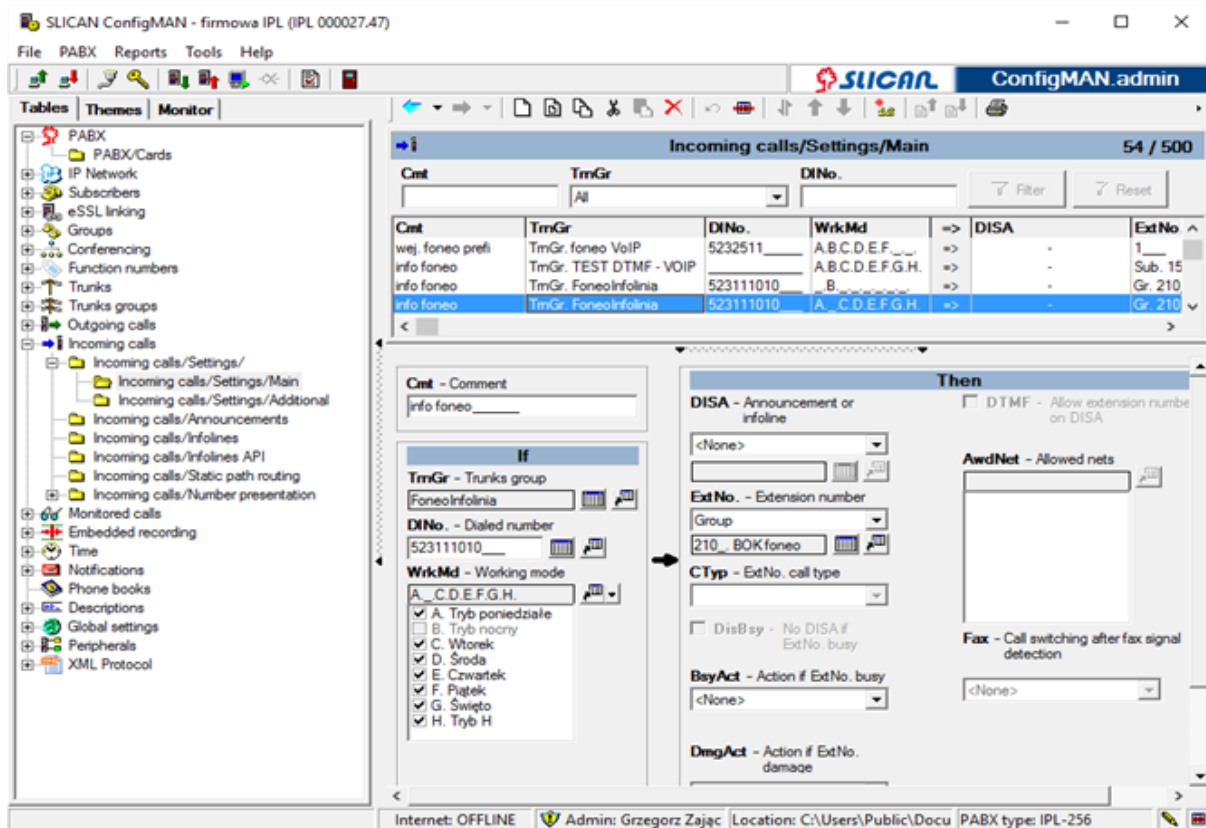


Fig 4.24: Sheet **Incoming calls/Settings/Main**

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

Subscriber can presents himself only by number defined in table **Incoming calls/Settings/Main**. For record, in field **DINo.**, there must be full national number, when number is routed directly (subscriber, DISA, infoline, group, Conference etc.), or partial number if oncoming calls are routed to prefix so group of DDI extensions. Field **TrnGr** can be additional filter for incoming calls and for presentation in outgoing calls. Entered trunk group must be consistent with trunk digits from field **DINo.** (according to tab **Trunks/Trunks Groups**) or set as any. Not setting number in Incoming Calls table results in blocking option of presenting with this number by any subscriber during external calls. One exception is using option *Defined Number* within subscriber settings (**Subscribers/Outgoing calls** field **CLIP**). Notice must be taken that this number must be consistent with numbers among trunk used for external calls, in case of inconsistency main trunk number will be used.

- Example settings. You want to obtain presentation 523251120 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

Within table **Incoming calls/Settings/Main** defined

523251100 → Extension number 1 → no assigned trunks group (<Default> in field **TrnGr** – Trunk group)

523251120 → Extension number 1 → DDI trunk

In above example for outgoing call from extension number 1 number 523251120 will be presented, because both **DINo.** and **TrnGr** fields are congruent, so second entry has higher priority (will be higher in Incoming calls table).

Example 2:

IMPROPER CONFIGURATION

523251100 → Extension number 1 → DDI trunk

523251120 → Extension number 1 → DDI trunk

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*).

- 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.
- 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. Similarly if we use DDI numbers and MSN numbers. Otherwise number presentation for outgoing calls will be improper.

Example 3:

PROPER CONFIGURATION

Within table *Incoming calls/Settings/Main* defined:

523251100 → Extension number 1 → DDI trunk in field **TrnGr**

527390000 → Extension number 1 → MSN trunk in field **TrnGr**

In this situation subscriber will be always presented himself properly depending if outgoing trunk group is DDI or MSN.

- While configuring table of incoming calls, if incoming call is routed to prefix it should be noted to define prefix included at last one shared digit. This entry allows to route calls to numbers range (DDI), when with minimum amount of entires we can route calls to biggest amount of subscribers. National numbers have 9 digits.

Example 4:

PROPER CONFIGURATION

DINo. 5232511__ ExtNo. Prefiks 1__ - when DDI numbers range is consistent with PBX internal numbers range (7 digits DINo., Prefix 1 – when for entries digits 1 overlap, 3 digit numbering range within PBX starting with digit 1)

PROPER CONFIGURATION

5278956__ 11__ - when DDI numbers range is inconsistent with PBX internal numbers range (7 digits DINo., Prefix 11 when for entries digits 56 and 11 overlap, 4 digit numbering range within PBX starting with digits 11)

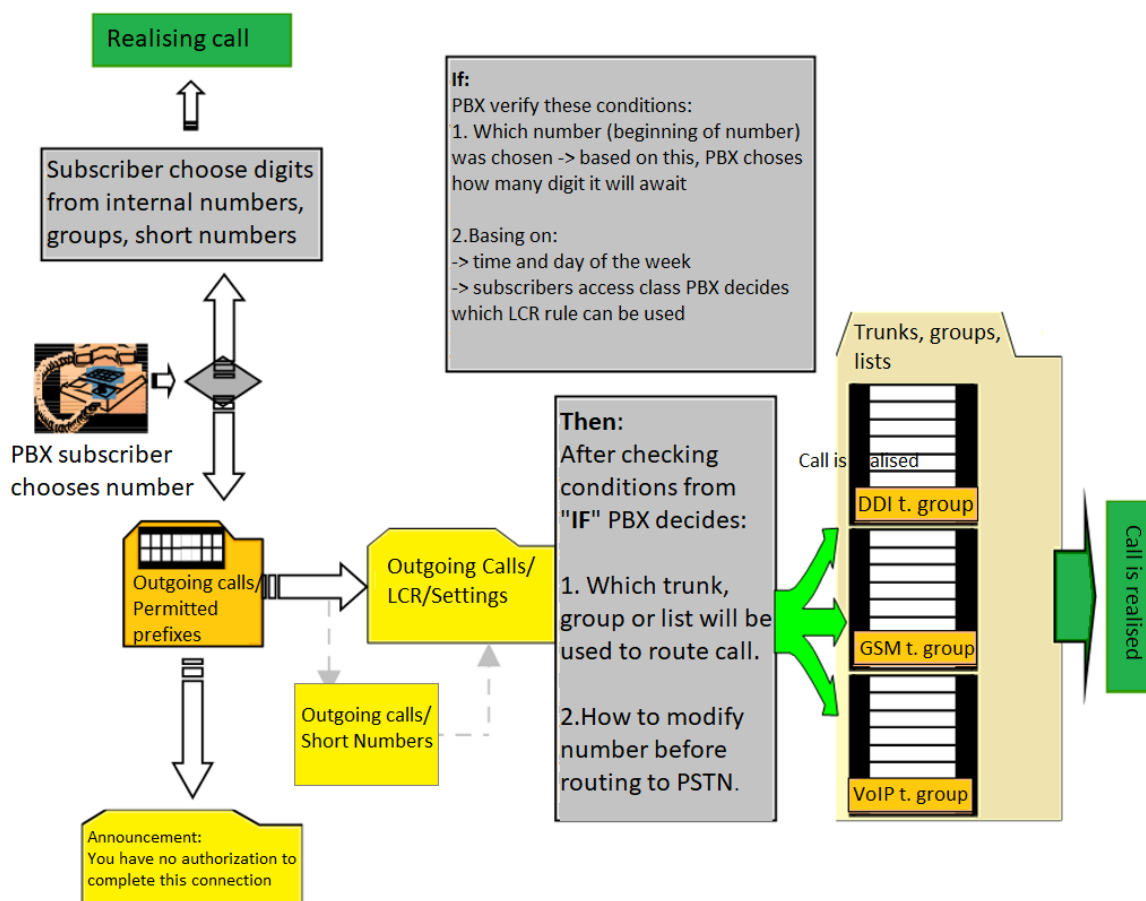
IMPROPER CONFIGURATION

523251__ __

W przypadku nieprawidłowej konfiguracji wystąpi błędna prezentacja abonenta w ruchu wychodzącym.

4.4.5 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**, and then 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**.

4.4.5.1 Defining outgoing method

In PBX 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**.

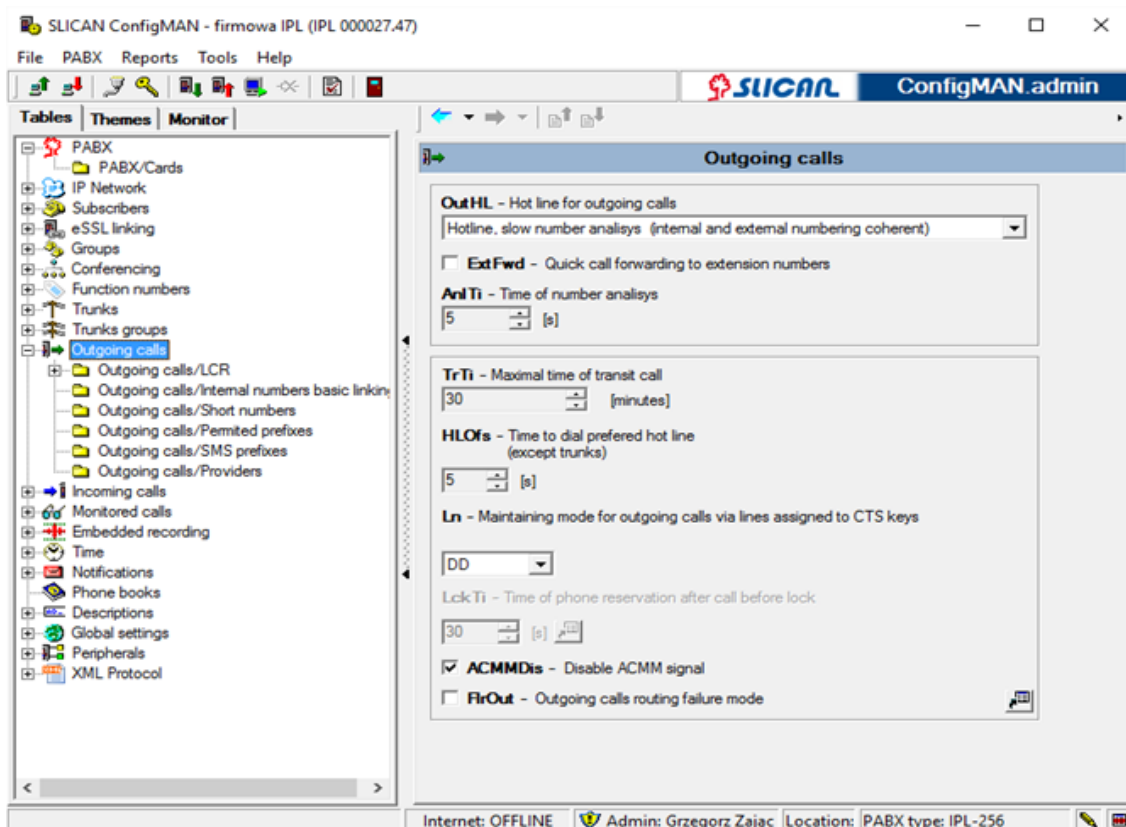


Fig 4.25: Sheet Outgoing Calls

If **hot line with slow number analysis** 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**.

Quick number analysis- CURRENTLY ISN'T RECOMENDED TO SERVICE TRAFFIC BETWEEN DIFFERENT CITIES due to external numbers range overlapping with internal subscribers extensions numbers.

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.

Specified outgoing method – additionally, outgoing method, in specified situations, can be defined using sheet **Global settings/Subscriber access classes**. In this sheet value of following three 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. Service is used during emergency and traffic is routed through emergency trunks.

4.4.5.2 Defining rules of outgoing calls in sheet *Outgoing calls/short numbers*

In sheet ***Outgoing calls/Short numbers*** outgoing method is defined after choosing number eg. external calls digit. Here connection route and establishing conditions are defined. Also dialed external number by PBX 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 and CTI application.

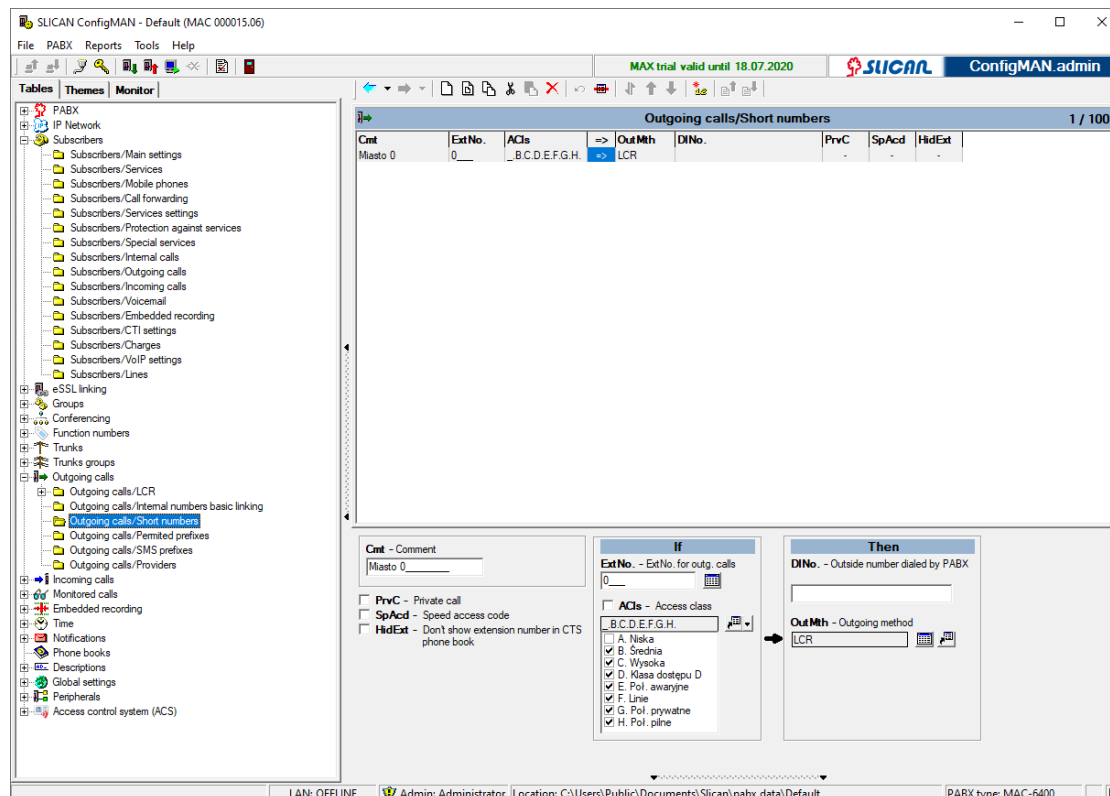


Fig 4.26: Sheet *Outgoing calss/ Short numbers*

In PBX different numbers can be defined as "outgoing digits" (field **ExtNo.**) and each "outgoing digit" could be granted to other access class – **ACls** (subscriber configuration is in sheet **Subscribers/Outgoing Calls**, field **DayCls** and **Nightcls**). 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**).

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

Slisan 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 **ACIs** 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 fulfilling all conditions defined in clause **If:**, outgoing method and number dialed by PBX is determined in clause **Then:**

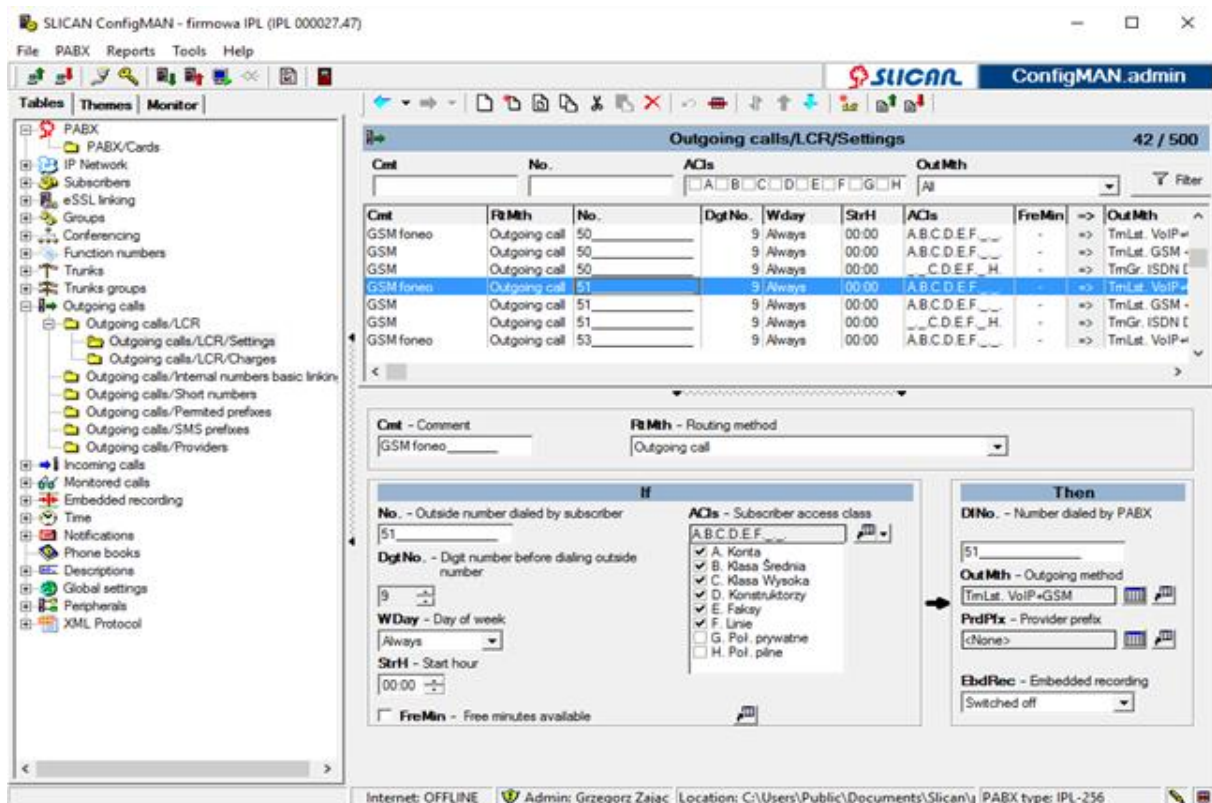


Fig 4.27: Sheet Outgoing Calls / LCR/ Settings – external outgoing calls

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

When from field Outgoing calls in Slisan 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 **DINo.**) 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, simulating outgoing calls which terminates within same PBX.

4.4.5.4 Detailed rules of routing calls in LCR table

Dialed number analysis by PBX means check all conditions defined in **Outgoing calls/LCR/Settings table**. If system encountered first consistent entry, check if trunk(group) 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.

Outgoing calls/LCR/Settings													3 / 500
Cmt	RtMth	No.	DgtNo.	Wday	StrH	ACls	Net	FreMin	=>	OutMth	PrdPfx	DINo.	EbdRec
Orange	Outgoing call	050	10	Always	00:00	_B.C.D.E.F.G.H.	-	-	=>	TmLst. Basic	<None>	50	-
Era	Outgoing call	060	10	Always	00:00	_B.C.D.E.F.G.H.	-	-	=>	TmLst. Basic	<None>	60	-
National	Outgoing call	0	10	Always	00:00	_B.C.D.E.F.G.H.	-	-	=>	TmLst. Basic	<None>		-

Fig 4.28: 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.

Outgoing calls/LCR/Settings													4 / 500
Cmt	RtMth	No.	DgtNo.	Wday	StrH	ACls	Net	FreMin	=>	OutMth	PrdPfx	DINo.	EbdR
Orange 1	Outgoing call	050	10	Always	00:00	_B.C.D.E.F.G.H.	-	-	=>	Tm. VolP Anon SIP	<None>	50	-
Orange 2	Outgoing call	050	10	Always	00:00	_B.C.D.E.F.G.H.	-	-	=>	TmLst. Basic	<None>	50	-
Era	Outgoing call	060	10	Always	00:00	_B.C.D.E.F.G.H.	-	-	=>	TmLst. Basic	<None>	60	-
National	Outgoing call	0	10	Always	00:00	_B.C.D.E.F.G.H.	-	-	=>	TmLst. Basic	<None>		-

Fig 4.29: Sheet Tables/LCR/Settings

After Hard Reset Wizard Basic LCR entries are included in PBX database, 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.

4.4.5.5 Emergency working mode for outgoing calls

In case of emergency preventing from proper handling outgoing calls by PBX eg. poor quality of VoIP calls, trunk failures – which can’t be properly interpreted by PBX mechanisms and can’t be realised by LCR outgoing calls rules – PBX administrator or privileged person can react in that case. Option to configure separate trunks which realise calls during emergency mode, which during normal operation can be used (but don’t have to). When mode is needed, it can be enabled by manually turning on emergency mode. After that subscriber automatically have proper access class allowing for outgoing calls through emergency trunks..

Emergency mode can be turned by:

- Administrator/PBX manager – with **ConfigMAN** application – field **FlrOut** sheet **Outgoing calls**
- Subscriber with privileges to use *80 service
- Subscriber with programmed CTS phone button

E – failure calls is default access class for emergency calls. It is defined with **FlrC** field tab **Global Settings/ Access classes**. Default class for those types of calls and description can be changed in **ConfigMAN** application settings.

ATTENTION!

In case of turning on emergency mode:

- For outgoing calls all traffic is routed according to rules for E access class for this subscriber.
- Calls to other Slican PBX – linked by eSSL – won't be routed with direct link but with emergency trunks as alternative call maintenance (just like during link failure – see chapter [Alternative routing settings](#))

4.4.5.6 Free minutes packages

Functionality allows to set limits for outgoing calls during month. Configuration of this functionality should be started from sheet **Trunks / Free minutes**.

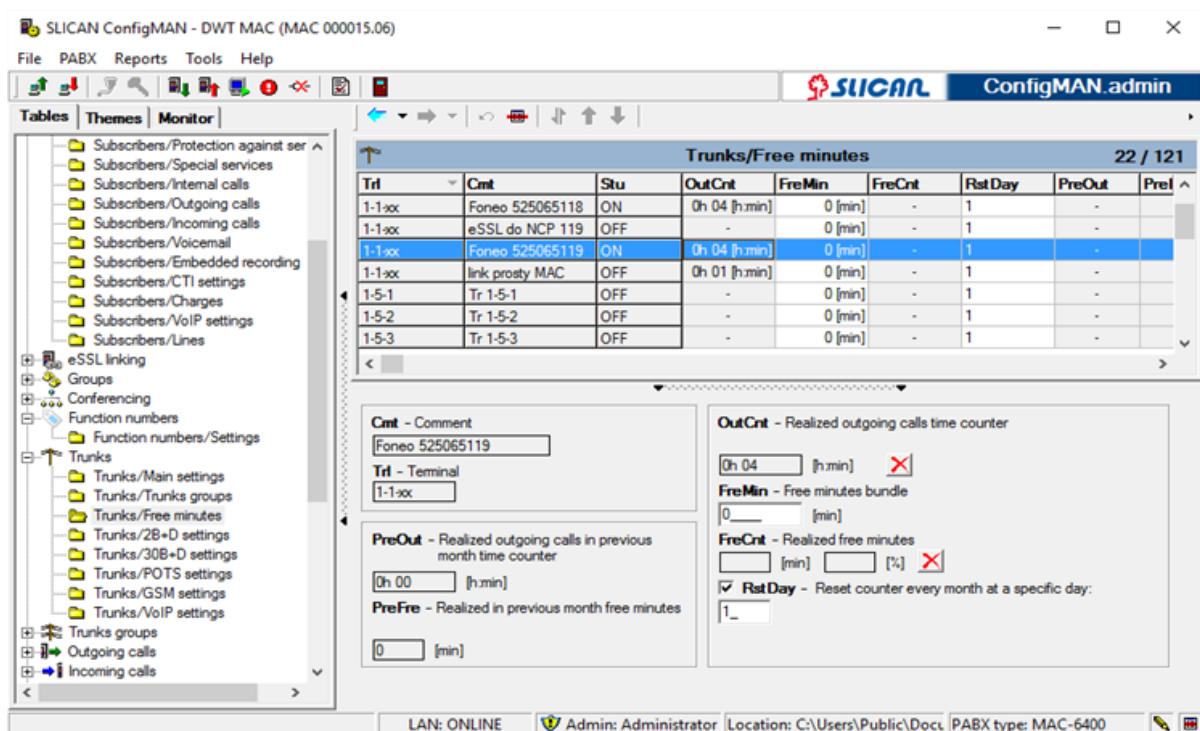


Fig 4.30: Sheet Trunks / Free minutes

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 aren't resetting even if You are 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** – **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** – **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.

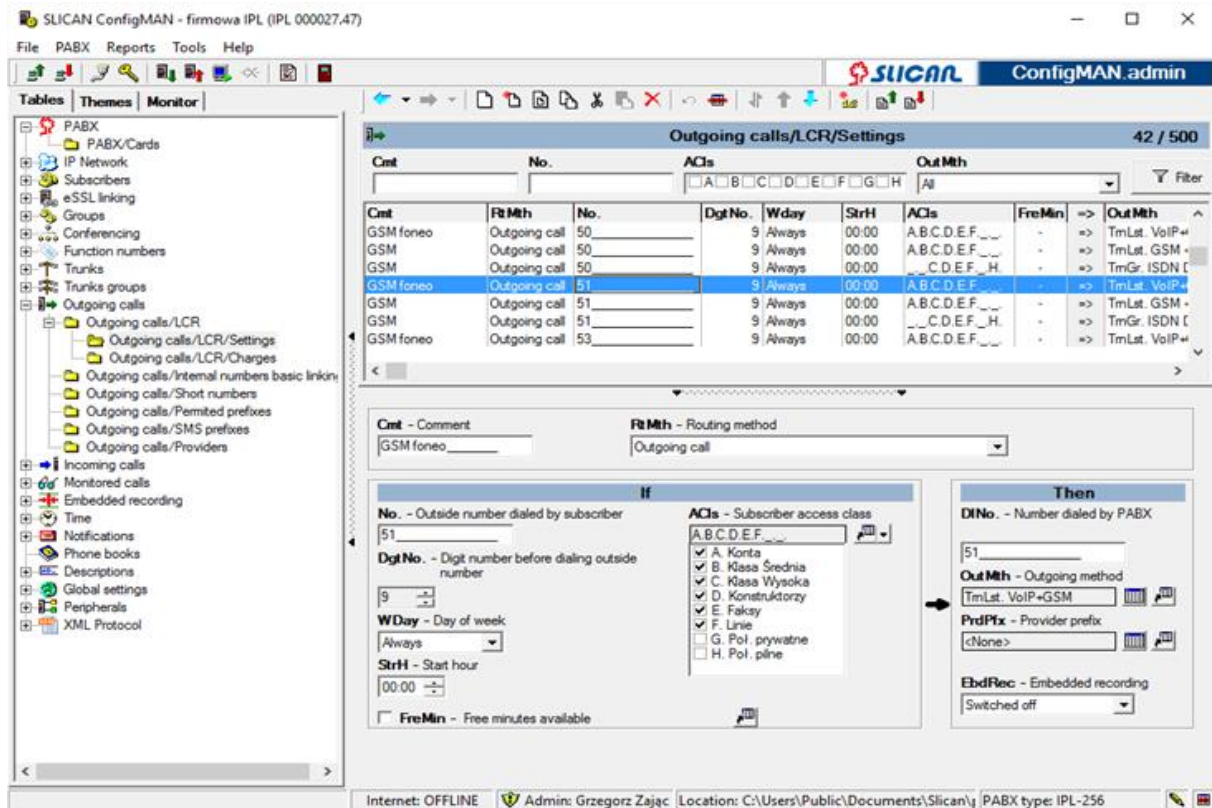


Fig 4.31: Check-box determines if for specified LCR table entry free minutes are used

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.

4.4.5.7 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:

- *OFF* - w *Switched off* – PBX working mode change has no influence on phone locking.
- *Lock* - phone is locked if in current working mode has automatic phone lock field checked. Lock can be deactivated using service *20.
- *Lock and unlock* – automatic locking and unlocking phones depends on current PBX working mode (manual phone unlocking is not necessarily).
- *Automatically* - automatic locking and unlocking phones depends on current PBX working mode (manual phone locking and unlocking with *20/*21 is not possible).
- *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**).

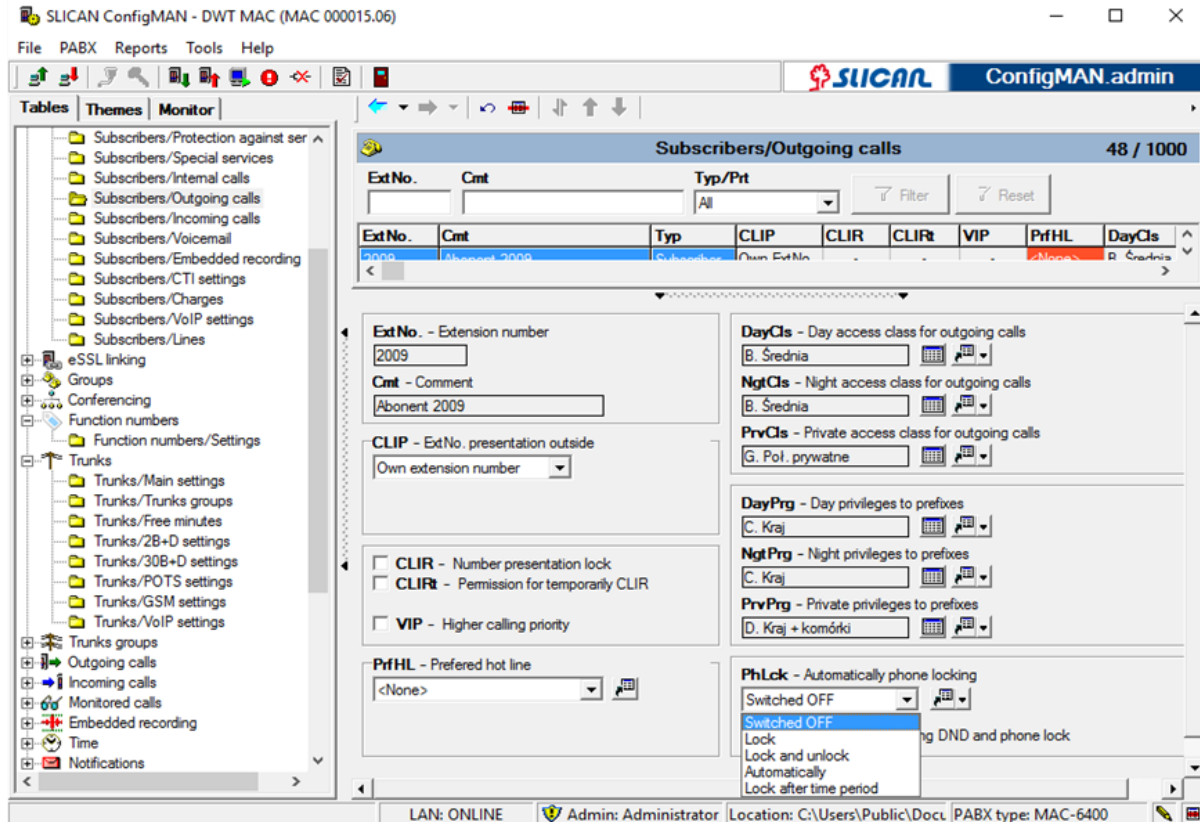
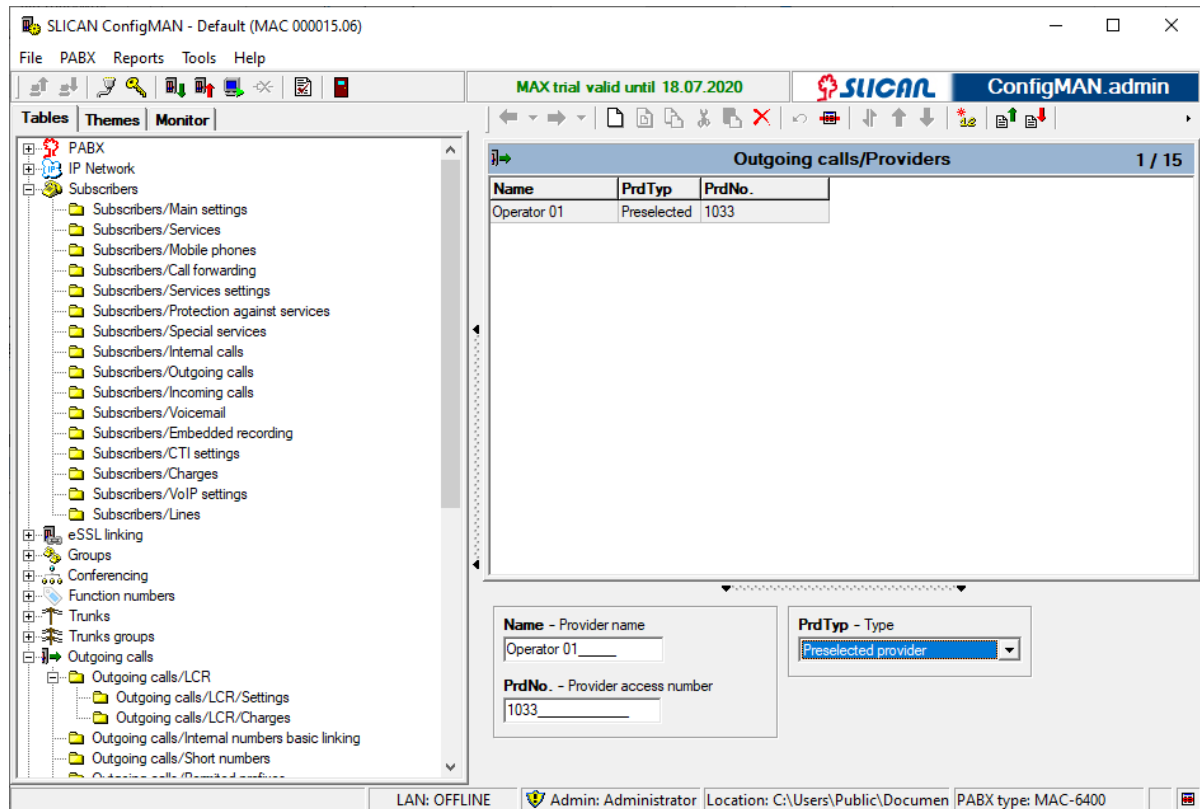


Figure 4.32: Sheet Subscriber / Outgoing calls

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.

4.4.5.8 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.



Figur 4.33: 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 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 international/national prefix is added to this number. Prefixes are defined in table ***Global Settings/Remain***, field **IntPfx** for *International* format and **NtnPfx** for *National* format. Due to incoming number standarization, number presented to internal subscriber enables to ring back to this number without any modification.

4.5.1 PathFinder – Call routing

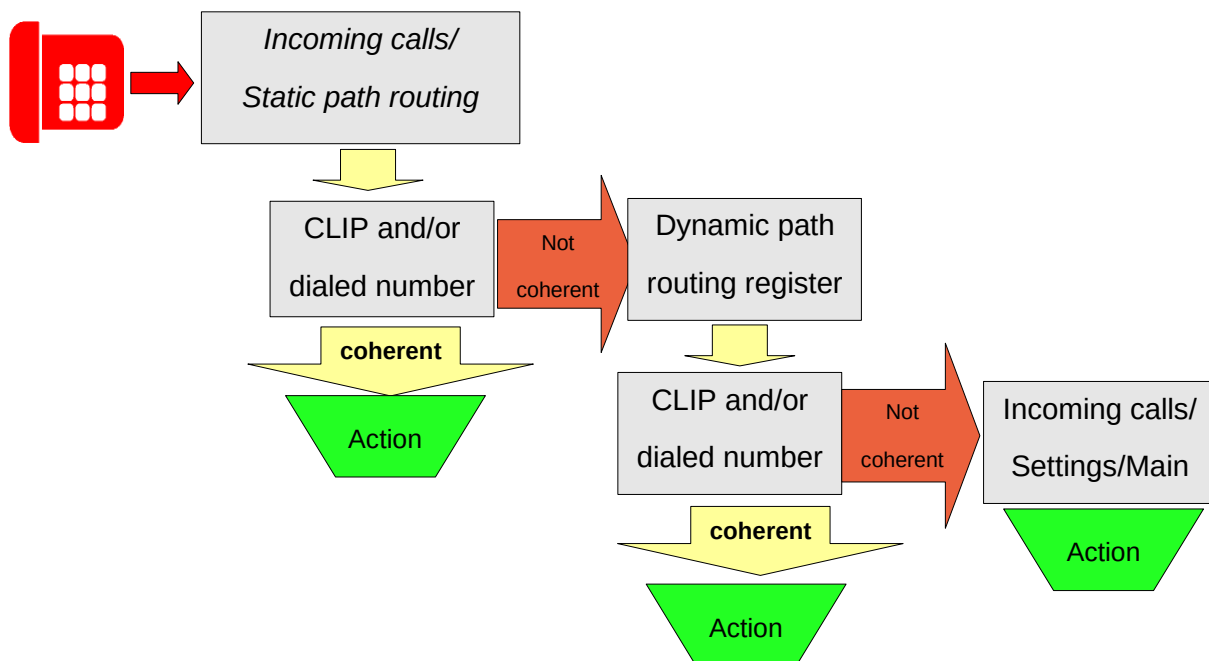
To every incoming call 3 parameters are assigned:

1. Caller number – so called CLIP (lack of 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 forming are firstly routed to PathFinder – static then dynamic – if there are no appropriate rules for PathFinder – call is routed directly to Incoming Calls table (sheet ***Incoming calls/Settings/Main***).

4.5.1.1 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 **DI~~No.~~** is empty, it means, that entry will be applicable for calls incoming to every trunk number connected to the PBX.

ATTENTION!

Incoming calls/Static path 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 from the table.

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**. This also avoids checking **Dynamic Calls Register**.
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(along with CLIP);
6. **CallBack** – an account is logged for which caller CLIP is equal to content of **MbIPh**, in sheet **Subscribers/MobilePhone**. To enable regular costless calls from Mobile Phone to PBX extension numbers or external 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. **Put relay on stand-by** – if conditions of static path routing are fulfilled – specified relay will on stand-by;
10. **Open DPH.IP doorphone from MobilePhone** – Mobile phone calling from GSM to chosen PBX number will open door/gate by activating EZ electrolock, STA relay or both at once, without establishing call.
11. **Reject** – in this situation caller hears busy signal;
12. **No answer** – caller hears ring-back tone and after three minutes disconnection takes place.

4.5.1.2 PathFinder – Dynamic Path Routing

Dynamic path routing enables to store call connection paths to using them in the future to establish connection, directly, without infoline, group or other subscribers. It works for incoming calls (connecting call again to same internal extension) and outgoing (option of calling back extension).

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 initiated by incoming calls.

A. Incoming call with CLIP calling: 22 3333333. According to 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.

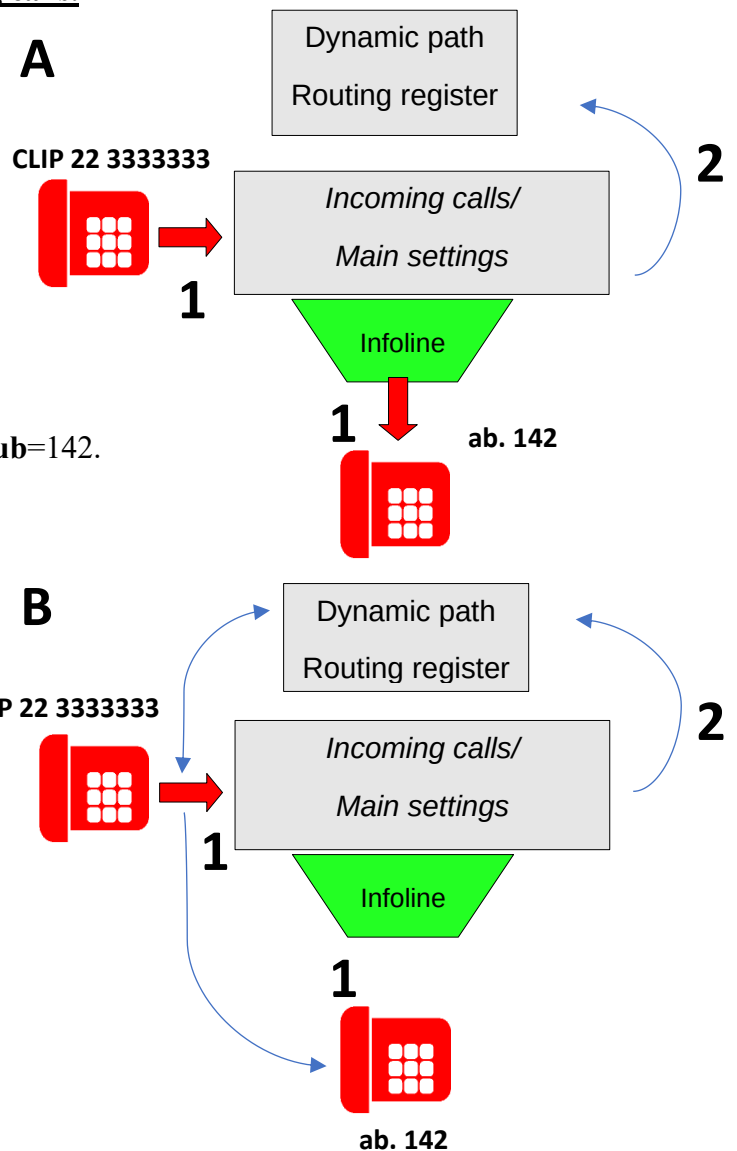
(2). PBX stored call path.:

1. CLIP Caller = 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 app [MessengerCTI.Desktop](#)
- 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 to 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!

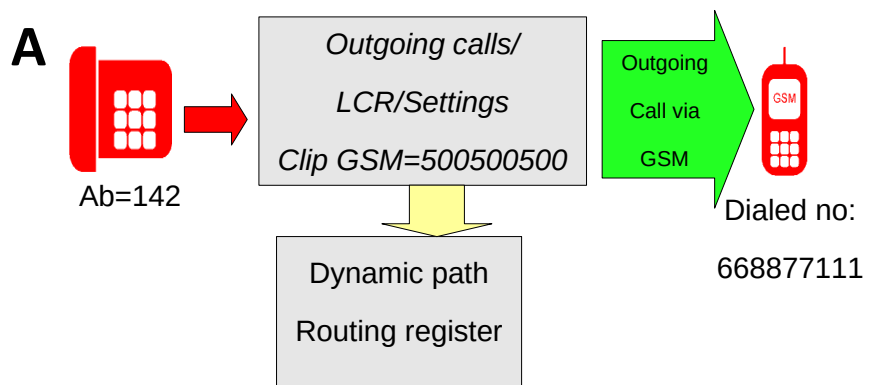
Incoming calls are not routed if:

- *połączenie odebrał abonent nie będący członkiem żadnej grupy PickUp,*
- *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).*

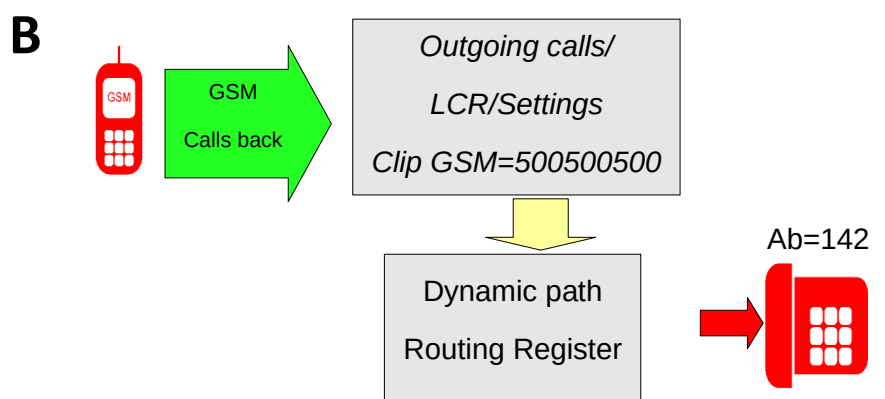
Dynamic path routing of incoming calls, as presented in example I, can be switch on in sheet **Incoming calls/Settings/Additional**, in field **PthRt**. Calls will be routed (their paths will be stored), only for specified entries in table of incoming calls.

II. Dynamic path routing initiated by outgoing calls.

A. If PBX subscriber SubA 142 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 142.



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, as presented in example II, can be switch on in sheet **Trunks groups/Settings**, in field **PthRt**. Calls will be routed (their paths will be stored), only for specified trunk groups.

ATTENTION!

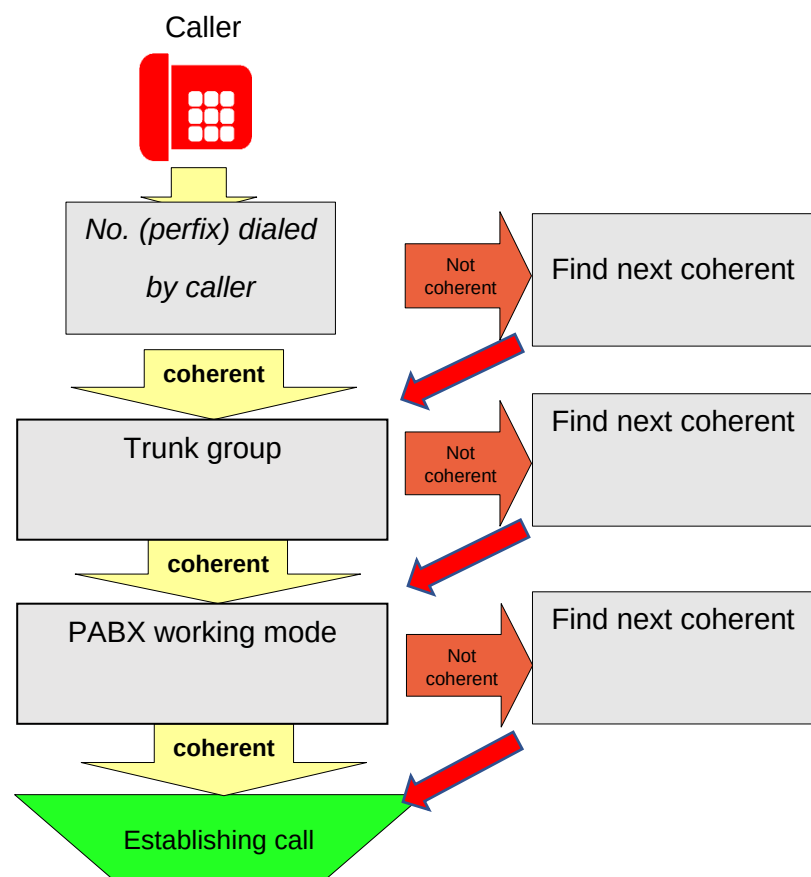
PBX reset erases dynamic path routing register.

4.5.2 Managing incoming calls via incoming calls table

In this place you should define basic conditions, which must be fulfilled to establish call with specified extension number – unless they are picked up.

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

- defining, which number is dialed by caller,
- via which trunk group call is established,
- PBX working mode when routing rule is valid.



4.5.2.1 Routing mode.

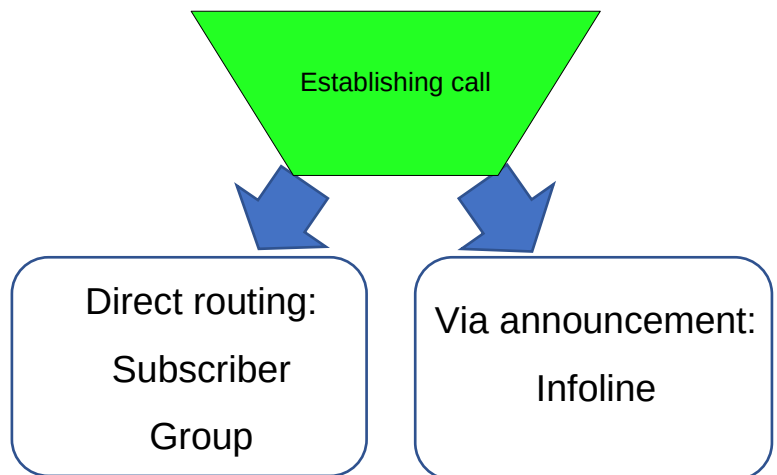
Target, where the call is routed (field **ExtNo.**) can be achieved directly or by (DISA/Infoline announcement - field **DISA**).

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**. (used for eg. restricting routing directly to director number).
- 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. When subscriber isn't busy announcement is played.
- Calling to extensions by PBX - sheet **Incoming calls/Settings/Main** field **Ctyp** =:
 - After DISA – cal will happen after playing announcement
 - on DISA - 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 off.



4.5.2.2 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(if it is bus, doesn't answer etc). 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.

4.5.2.3 Incoming calls/Settings/Additional.

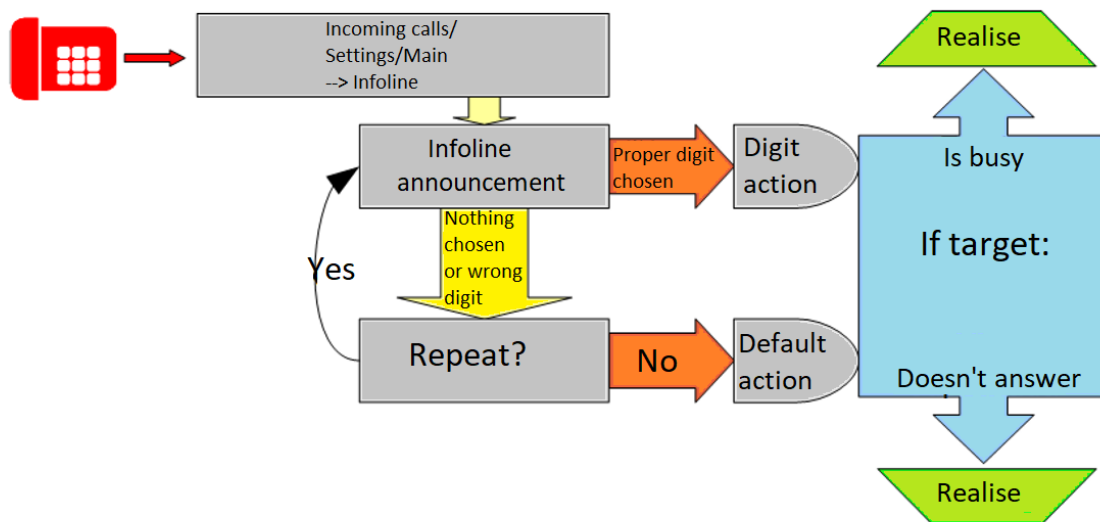
Within these sheet there are parameters regarding Incoming Calls table entries records:

- **MaxInc** – maximum number of calls which can be realized in the same time by this entry.
- **Cdrn** – maximal duration time of call routed by this entry. It is useful if you use free numbers eg. 800.
- **EbdRec** – priority of recording specified trunk/trunk group/trunk list. If other value then Switched off, calls routed via this entry are recorded.
- **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.
- **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.

4.5.3 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 **An**.
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 are in those state after being dialed:
 - busy – action defined in field **BsyAct** is performed,
 - doesn't answer – action defined in field **AnsAct**. This field can't have subscriber/group in field **DgtAct**. When connecting to subscriber – action will be realised after time in field **DIFre** sheet *Incoming calls* will pass. If call was dialed to group – it will be realised after time in field **MaxWt** sheet *Groups/ (Uprising, Cyclic, Linear, ACD)/ Settings* will pass.
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.

Be default for **DefAct**, **BsyAct** i **AnsAct** available actions are:

- disconnect
- forward – to subscriber, group, conference, announcement, another infoline, linked pbx number or function number,
- back to forwarding (to subscriber that forwarded to infoline)

Additionally for **DefAct** traffic can be router to short number (**Outgoing call/Short Numbers**) or after playing infoline announcement – for each number from DDI range – again to incoming calls table (according to [artykułem Wiki](#))

UWAGA!

Since fw. 6.50 infolines are subject of licensing.

Up to firmware 6.42 without license 1 infoline is available.

UWAGA!

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

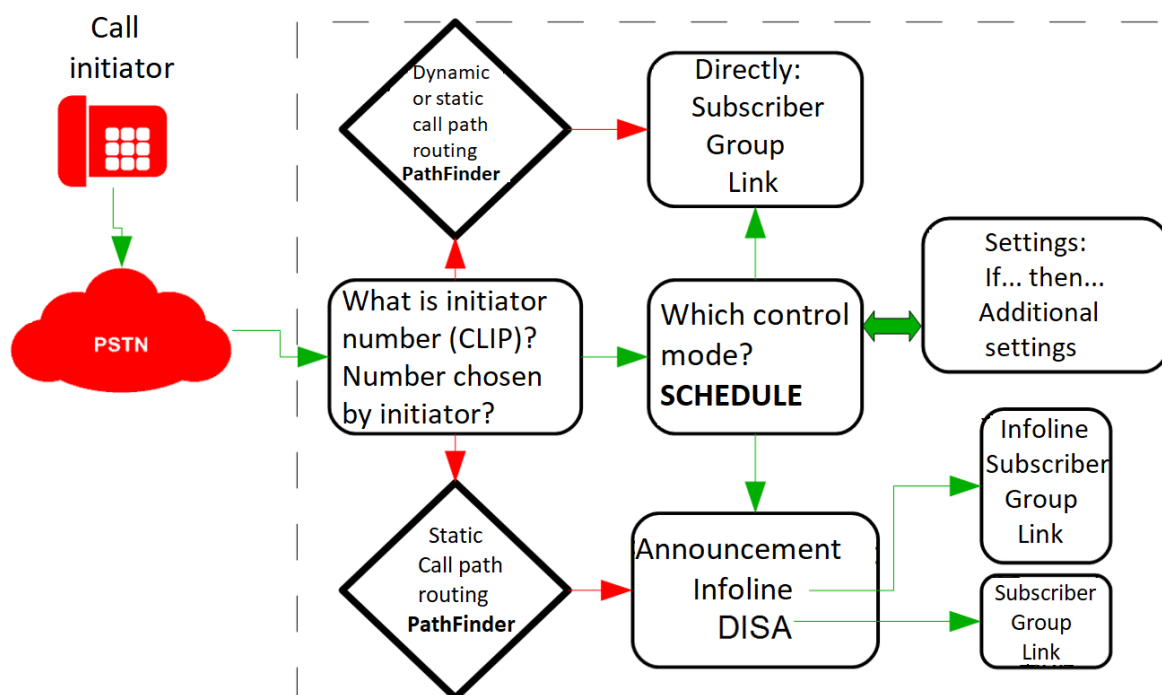
Infoline parameters and limitations

Maximum numbers of infolines in system: 99.

Maximum time of connecting to Infoline for one call is equal to 15 minutes.

Maximal number of calls which are connected to Infolines in the same moment are equal to:

- IPL-256, IPM-032, IPU-14, IPS-08 – 32
- MAC-6400 – 10* number of shelves
- CCT-1668 sterownik ver.2 – 32
- CCT-1668 sterownik ver.1 – 4
- 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**.

4.5.4 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 eg..

- **SrvAdr** - **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>
- **SrvTmo** – maximum time of waiting for server reply.

UWAGA!

API Infolines are subject of licensing as standard Infolines.

4.5.5 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:

- According to field **COLR** from sheet *Subscribers/Incoming calls*:
 - Numer chosen by initiator,
 - According to field **CLIP** from sheet *Subscribers/Incoming calls*,
 - Main number,
 - no presentation – lock presentation of obtained number.

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

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

Following options are available:

- **OFF** – number presentation is switched off;
- **CLIP** - caller presentation is send 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 send. 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 analogue phone models CLIP presentation is not received when double ringing is set(during incoming external calls). In this situation option **RegRng** in sheet **Subscribers/Special services** – Always regular ringing.*

4.5.5.2 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.

Zastosowania:

- This comment is displayed on system phones, analogue phones with CLIP-FSK and for VoIP users. By default hard reset wizard creates list of numbering area according to PNK/Telephone Numbering Plan.
- 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 with above mentioned prefix is presented as “Company X”.
- If we want to assign international prefix of caller with playing system announcements with his language – create announcements for this prefix within field **AnnSet** (1..3).

4.5.6 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 **MessengerCTI**, **ConsoleCTI**, **RecordMAN**, **BillingMAN**, , XML protocol and call history.

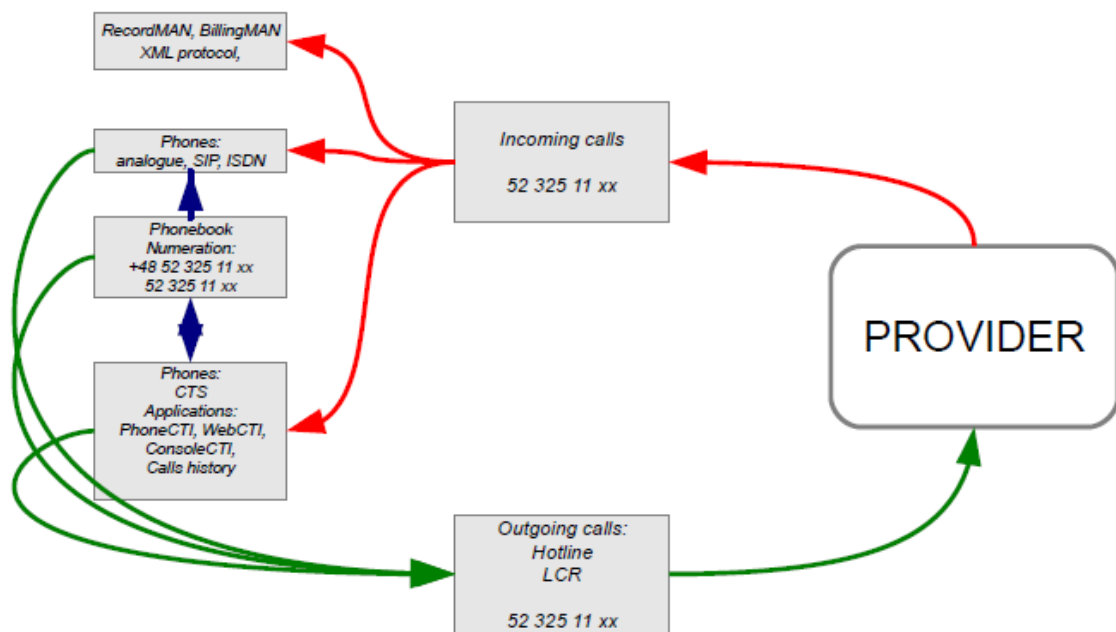


Figure 4.34: Regional number phonebook - Hotline

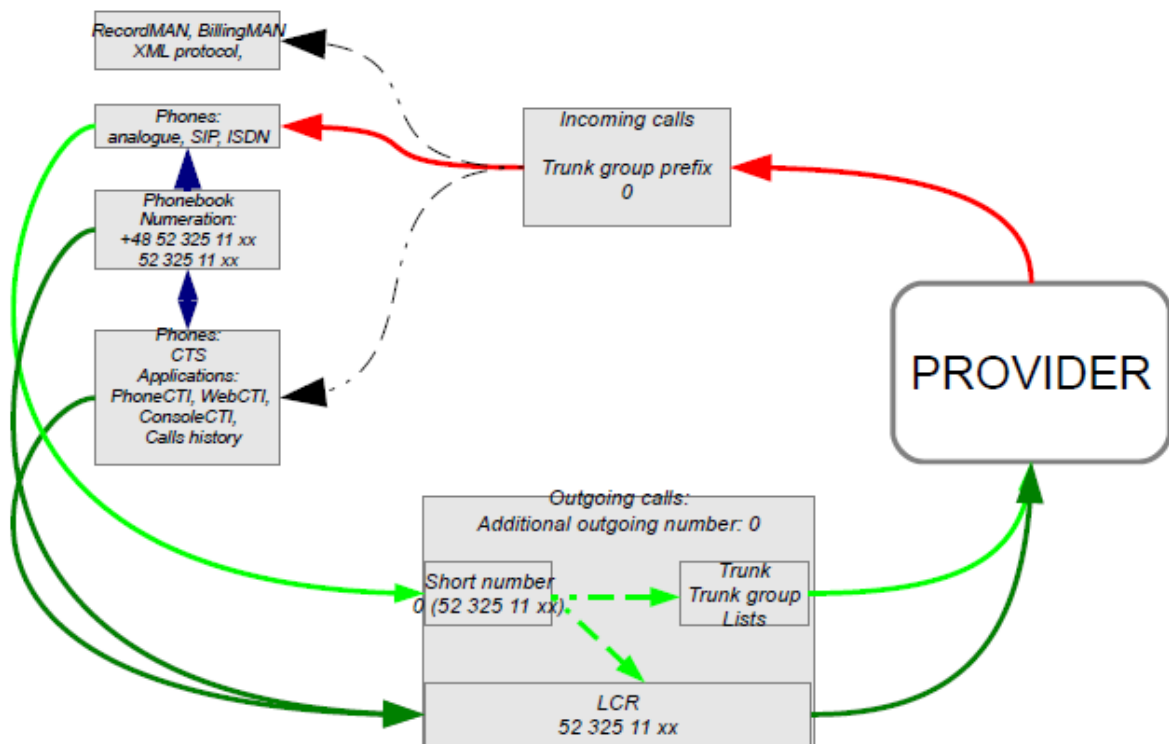


Figure 4.35: Regional number phonebook - additional outgoing number

Work mode: "Unknown number phonebook"

Designed for PBX's with **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 with field **AddPfx**(prefix added to CLIP), incoming call will be displayed with this number. This number will be added to all kind of phones, call history and applications like *RecordMAN*, *BillingMAN*.

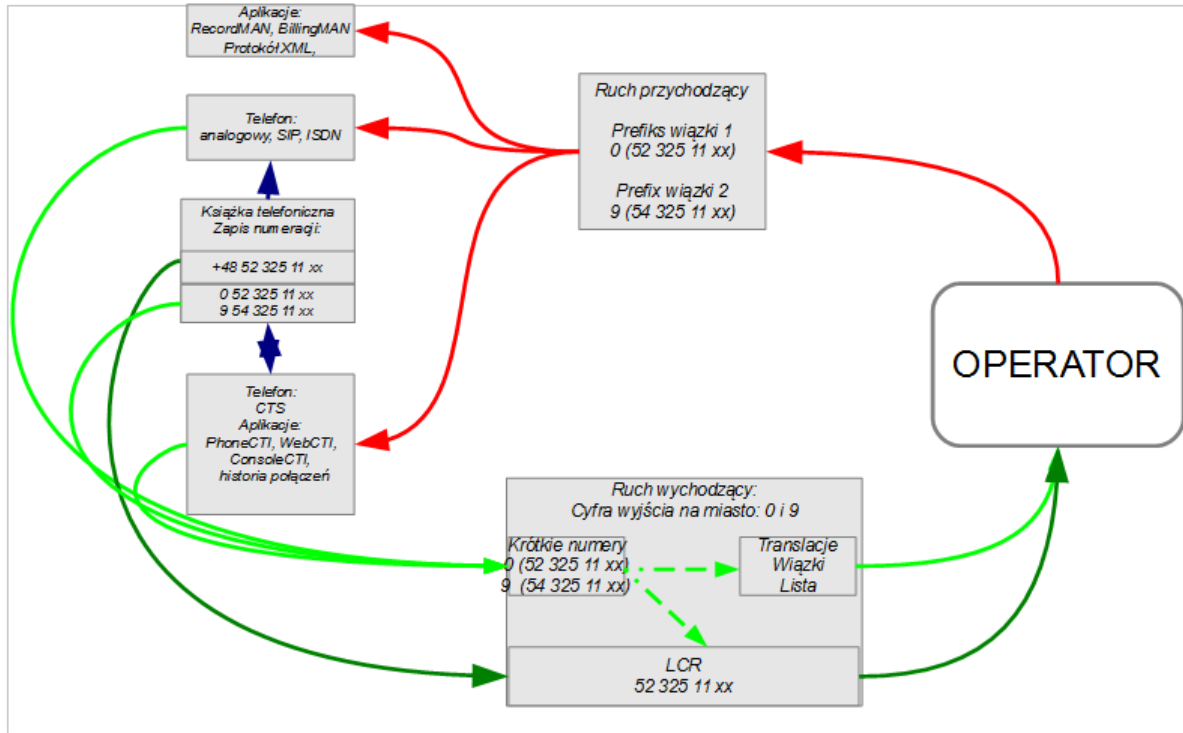


Fig 4.36: Unknown number phonebook - additional outgoing number

4.5.7 PBX working mode

There are 8 working modes in PBX (**Time/PABX working modes** – mode's names can be edited). PBX can work 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 **Ti**. 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(repeating each year like New Year, Christmas) 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**).

Changing Working mode 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 PIN *** - 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 by user with permissions.

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 lacking 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.

4.5.8 Call forwarding to operator (orphan calls)

If there is requirement that unanswered calls incoming to pbx won't be lost and be serviced on chosen phones, service drop to operator must be used. For subscribers which will have drop enabled, must be set with field **OphTyp** within sheet **Subscribers/Incoming Calls**.

Following definitions of operator are possible:

1. by global configuration: set up target (**Incoming calls** field **OphNo.**), to whom incoming calls will be routed if it wasn't picked by subscriber
2. by working mode: set up target (**Time/PABX working mode** field **OphNo.**) where we can set up different operators for each working mode .
3. by company: set up target (**Descriptions/Departments** field **OphNo.**) for subscribers which in sheet **Subscribers/Special services** field **Dpt** have set up one of the departmenys.

Time after which call is forwarded is set up in sheet **Incoming calls** field **OphTi**.

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.
5. Infoline: forward to infoline
6. Function number: forward to function number funkyjny

ATTENTION!

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

4.5.9 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,
- 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.

4.5.10 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 heard for incoming calls in following situations:

- call is routed to the group
- call is forwarded without announcing

4.5.11 Melody on Hold (MoH) options

4.5.11.1 Melody on hold selection

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

- standard (Slican system announcement),
- recorded announcement
- external source (ex. Audio device MAB-1101)
- ringing signal

4.5.11.2 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*.

4.5.11.3 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.

4.5.12 Voice announcements

4.5.12.1 Introduction

If it is necessary to „cover” ringing signal by melody on hold, during waiting for establishing call, field **MeIRng** in sheet **Incoming calls** should be checked. Without extra license user can set up only up to 20 minutes of announcement, to get over limit license is required – up to 2 hours.

1. Announcements 1...9 type DND/DISA/Infolinie can be recorded from phone or using **ConfigMAN**,
2. Remain announcements 10...99 can be recorded only using **ConfigMAN** (including **ConfigMAN.user**).
3. Private DND/Voice mail message can be recorded from phone connected to authorized extension number or using **ConfigMAN**.

4.5.12.2 Recording announcement DND/DISA/INFOLINE from phone,

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.

4.5.12.3 Private DND/Voice mail message

PBX subscribers – if they are authorized to do this (sheet **Subscribers/Services** settings, field **DNDMsg**) – record and set private DND messages using service ***79**, ***78**. Much more about recording private DND/Voice mail messages you can find in instruction manual.

4.5.12.4 Announcements informing about position in queue

To use **queue announcements** functionality from sheet **Groups / Upgringing (Cyclic, Linear, ACD)/Settings** it is necessary 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 position in queue from 1 to 30
- 1 – information about exceeding maximal declared number of calls waiting in queue
- 7 - o czasie oczekiwania - 1, 2, 3, 5, 10, 15 lub 20 minut,
- 1 – information about exceeding maximal declared waiting time

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

ATTENTION!

For incoming calls to pbx, announcement will work only when routed through DISA or Infoline – so during active call with operator (Connect).

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

4.5.12.5 Recording voice announcement from inside ConfigMAN application,

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

To realize announcement in PBX you can do one of 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);

Regarding recording voice:

- 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 lose quality after uploading to PBX.

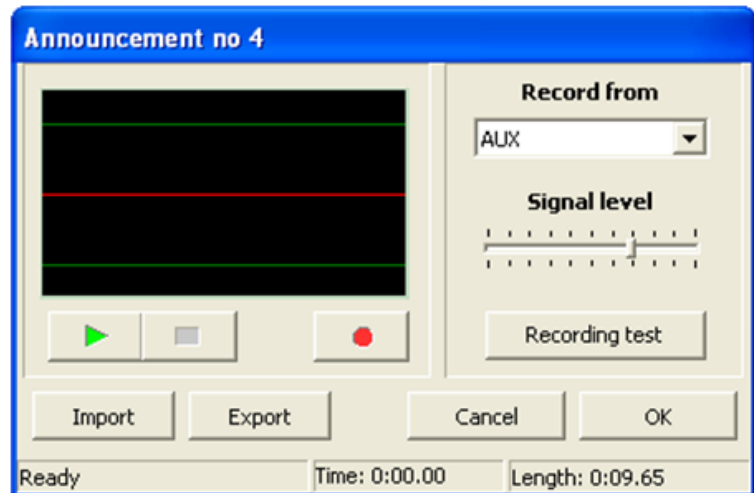


Fig 4.37: Embedded sound recorder

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:

- „V” – current – means that this announcement is saved on PBX, for this announcement field **Ti** – *duration time* is higher than 0,
- „→” - to be sent – announcement is recorded on computer but there isn't uploaded to PBX yet,
- „X” – announcement is empty.

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 archivisation can be done using option **PABX → Perform backup**.

4.5.12.6 Replacement of system announcements

W firmware version 6.20 possibility of defining system announcements was introduced. This functionality can be used 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***
- recorded with embedded sound recorder
- 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 /Main Settings***,
 - for specified numbering area – ***Incoming calls/Number Presentation/Area names***,

4.5.13 Voice mail

4.5.13.1 Introduction

There are two types of voice mail:

- **Individual** – assigned to specified extension number and managed by subscriber with this extension number,
- **group (virtual)** – 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(that have permissions).

Maximum capacity of voice mail box is limited by capacity of memory assigned to voice announcements. Duration of one left voice message can't go over time set in **MsgTi** field in sheet ***Global Settings/Voicemail*** (0,5 to 2 min), and capacity of all messages in single voicemail box is defined by content of field **MaxCap** from sheet ***Subscribers/Voicemail***.

Summary time, which can be used for storing voice mail messages is 120 minutes for IPL-256, IPM-032, IPU-14, IPS-08, MAC-6400, CCT-1668, CXS-0424.

This memory is shared for voice mail messages and for announcements (DISA/Infoclines+DND messages).

4.5.13.2 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 1000 and to change it, can be changed from sheet **Global settings/Voicemail**. 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 [Call forwarding](#)) 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.

Powiadomienia o nowej wiadomości na poczcie głosowej abonent otrzyma automatycznie. 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. Additionally information about new message can be send to subscriber GSM phone (details in chapter [MobilePhone](#)).

Messages can be removed:

- By subscriber when using phone, when listening to voicemail,
- By PBX admin, using „X” button near field **MsgCt** sheet **Subscribers/Voicemail**.

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 to PBX.*

4.5.13.2.1 Group voice mail (virtual voice mail box)

To create group voice mail functionality, voice mail box should be created just like new subscribers in sheet **Subscribers/Main settings** powołać ją podobnie jak zwykłego abonenta -

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,
- With function number.
-

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”.

4.6 Monitored calls

Slican has implemented a new tool for monitoring aquality of work of agents (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. Service also works when picking up ringing call from another subscriber, call to number with active forwarding, call transfer and Mobilephone.

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.

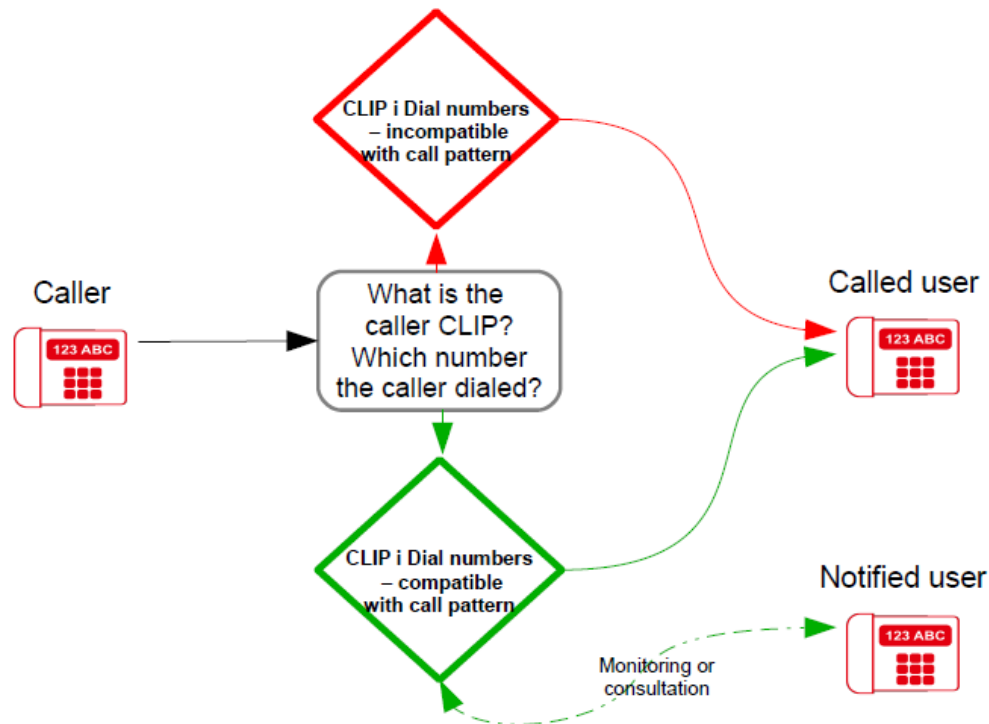


Figure 4.38: Monitored call – call process

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 paying 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. 48xxxxxxxxxx); **DINo** – internal number, to which incoming call id directed (subscriber, group)

Fig 4.39: Monitored calls - settings

Monitored calls/Settings					
CnnTyp	CLIP	DINo.	NtfNo.	ChrAcc	3PTY
Internal	117	118	189	Sub. 2070. Linia 2070	-
Outgoing	302	5232511117	523251118	<None>	-
Incoming	48503030981	302		<None>	✓

Fig 4.40: Monitored calls - records

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 on original PBX or linked by eSSL, number of caller will be displayed on notified phone. Monitored calls (supervised) are registered for tarifications only if they are external.

4.7 Supervisor service

Supervisor is defined for CTS system phone button. It allows to assign service to system phone button only using **ConfigMAN**.

Service allows:

- Monitoring chosen call (silent)
- Consultation with agent
- Three-way call
- Manual setting of monitored call recording

If Supervisor have programmed RECORD button, it's status reflects monitored call. He can turn on/off recording of monitored call.

4.7.1 Permissions and calls limitations of Supervisor service

Supervisor must have CTS phone with programmed service on button. To monitor calls, Supervisor chooses extension number or programmed button of Subscriber or Doorphone. musi posiadać telefon CTS z zaprogramowaną lampką usługi.

For call to connect, these conditions must be met:

- Target Subscriber or Doorphone must be making a phone call
- Target Subscriber or Doorphone can't be during call to 3 way call or conference.

- Supervisor must have access to Subscriber or Doorphone Net.
- Target Subscriber must be member of same wybrany PickUp group as Supervisor.
- Target Subscriber or his call partner can't have blockade WN3 turned on (join as 3/third).

4.7.2 Using service with CTS phone

Button with Supervisor service

4.7.2.1 Idle state

- Pressing button turns on light, service is active until turned off
- Next button press turns off light – service is inactive

4.7.2.2 Connecting to busy subscriber or doorphone with active Supervisor service

4.7.2.3 Monitoring state – light on, lampka świeci, listening to current call

- Pressing button with programmed Supervisor service – going into Consulting state
- Press context button WN3 – going into WN3 state – 3 way call
- Press context button Disconnect – disconnect monitored call
- Pressing turned off light RECORD – turns on call recording
- Pressing turned on light RECORD – turns off call recording

4.7.2.4 Consulting state – blinking light, consulting with target number, target partner doesn't hear conversation

- Pressing button with programmed Supervisor service – return to Monitoring state, ending consultation
- Press context button WN3 - going into WN3 state – 3 way call
- Press context button Disconnect – disconnect monitored call

4.7.2.5 WN3 state

- Pressing button with programmed Supervisor service – going into Monitoring state
- Press context button Disconnect – disconnect monitored call

Hanging up handset during each state causes to exit service.

4.7.3 Connecting to talking subscriber with inactive Supervisor button

Initiator will hear busy signal, after pressing Supervisor button will enter **Monitoring** state

4.7.4 Connecting to free subscriber with active Supervisor button

Reject call – service isn't available

4.8 Auditorium

PBX have additional option of setting up call time limit for specific phones. Functionality was prepared for prison, to control length of telephone calls from inside prison. It is related to prisoner right for specific number of calls and their length. Default time limit (1 to 99 minutes) is configured within PBX and same for every phone in auditorium mode. There is option of extending call time limit by 1 minute by supervisor, for individual auditorium phone.

Principle of operation:

To start functionality – PBX phones must be chosen for auditorium functionality. During normal mode they will be locked out of incoming and outgoing calls, also after picking up handset announcement will be heard.

On phone or CTS phone – each auditorium must have programmed separate button allowing to unblock and monitor it. For prisoner to be able to use specific auditorium, person using assigned CTS must hold service button until it will light up. For CTS user it is info that auditorium is unlocked and have usable call time. Additionally on screen info will be displayed, indicating call time limit and amount of used time. CTS user with single, short button press can briefly check state of auditorium used time, next button press will add time to the limit. Longer button press with light on will instantly delete limit and disconnect call.

30 seconds before limit ends, prisoner is informed with sound signal about incoming disconnect. Using up limit will disconnect call and block phone. CTS user during any moment extend call limit for 1 minute or end call earlier.

To configure service:

- From sheet **Subscribers/Special Services** in **EnAt** field choose auditorium phones
- From sheet **Global Settings/Remain** in **MaxTi** field set up maximal limit for all PBX auditoriums
- From sheet **Peripheral/CTS/Button** for chosen user CTS phones program separate buttons for each auditorium with service *Ab – Auditorium*.

4.9 My subscriber account

since 6.50 PBX firmware version, a new non-licensed feature is added. Basic subscribers settings can be changed via website, without additional CTI licence key.

In **Subscribers/CTI settings** tab, field **Pss** needs to be filled. Subscriber connecting to PBX IP using web browser, can log using phone number and password from **Pss** field have access to:

- information about number (access level to phonebook, active services, language settings, call costs)
- services configuration (phone lock, account recall, alarm setting)
- Forwarding, DND and I am here settings,
- application language settings,
- download application [MessengerCTI.Desktop](#)

Acquisition of CTI licence key, will allow to full WebCTI feature access:

- [WebCTI](http://pubwiki.slcan.com/index.php/WebCTI) - <http://pubwiki.slcan.com/index.php/WebCTI>

4.10 Parallel phone feature – MessengerCTI application,

Feature implemented in 6.50 PBX firmware version. Based on integration physical subscriber phone equipment or VoIP subscriber (base phone) and application [MessengerCTI](#) (desktop – Windows or mobile – Android since fw. 6.53).

Feature allow:

- MessengerCTI application on desktop with VoIP off – being able to control system phone (initiating and picking up calls) and for any base phone – additional features (call history, phone book, etc)
- MessengerCTI application on desktop with VoIP off – simultaneous or independent ringing up of application and/or base phone (eg. only application, when base phone is damaged or disconnected)
- MessengerCTI application on mobile (android smartphone) - simultaneous (phone + mobile application + desktop application) or independent ringing up application and or desk phone (depending on which system part is turned off)

MessengerCTI.Desktop is PC software for Windows system, which have functionalities: controlling phone, access to phonebook, calls history, monitoring status of other subscribers, chat, sending SMS and additionality – making phone calls with built in SIP client. Application isn't typical softphone klient so it can't be used with PBX (servers) of other vendors. For logging and making phone calls access to PC with application that can reach PBX, with properly configured computer network or internet is necessary.

MessengerCTI.Mobile – application for Android smartphones which have functionalities: controlling smartphone – making calls with GSM or SIP, access to phonebook, calls history, monitoring status of other subscribers. For logging and making phone calls access to android smartphone with our application that can reach PBX, with properly configured computer network or internet is necessary.

Functionality of parallel phone is available by installing, running on phone and/or PC application *MessengerCTI*, after that login at same time to subscriber account. All terminals (smartphone + applications) use same extension number.

Parallel phone features allows for telephone contact from different places eg.:

- local (workplace/house) – using base phone, PC with application, phone with application,
- mobile (travel) – using PC or phone with application

For ease of usage during phone calls while using PC, wired or wireless headphones can be used..

Answering and making calls is possible while using normal phoned and logged applications. During ringing of incoming calls, phone and PC, phones with applications will ring up. Answering call with phone stops ringing up on other devices and vice versa, answering from device with app stops ringing up of phone and other applications.

Making outgoing calls is possible from phone and application, but after starting call using one of the devices, will block others from making outgoing calls.

Application usage depends on current version of CTI license. Detailed info is in chapter [Licenses](#).

Full application description is available on websites:

- [MessengerCTI.Desktop](http://pubwiki.slican.com/index.php/MessengerCTI_Manual) –
http://pubwiki.slican.com/index.php/MessengerCTI_Manual
- [MessengerCTI.Mobile](http://pubwiki.slican.com/index.php/MessengerCTI.Mobile_Manual) –
http://pubwiki.slican.com/index.php/MessengerCTI.Mobile_Manual

Required PBX resources:

- hardware VoIP channels (VoIP card, controller card with VoIP resources and VoIP channels licence – if we are using *MessengerCTI* app with VoIP or as base phone – VoIP ports
- VoIP ports – when using phone as base VoIP account
- proper CTI license *MessengerCTI* application

ATTENTION!

MessengerCTI application can work without making VoIP voice calls, then VoIP channel licenses aren't required.

PBX and application configuration:

1. Create base phone in sheet **Subscribers/Main settings** (physical subscriber port – analog, system or VoIP)
2. within **Subscribers/CTI Settings** for created subscriber set up:
 - field **CTI** – access level **CTI.user** lub **CTI.user.Plus**
 - field **Pss** – login password to application
 - for *MessengerCTI.Mobile* - fields **AwdMbl**, **AwdGSM**, **IncC**, **MnNo**, **SecNo** regarding access and handling calls by applications
3. in tab **Global Settings/Remain** for *MessengerCTI.Mobile*:
 - for field **Ano** – access number to PBX, with it application can realise calls in GSM mode
 - for field **Adr** – PBX IP adress or host name necessary for realising phone calls in VOIP
4. install *MessengerCTI* application on PC or smartphone
5. login application to PBX according to manual:
 - *MessengerCTI.Desktop*:
http://pubwiki.slican.com/index.php/MessengerCTI_Manual#Connection
 - *MessengerCTI.Mobile*:
http://pubwiki.slican.com/index.php/MessengerCTI.Mobile_Manual

4.11 In SMS service

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

- [MessengerCTI.Desktop](#)
- [PhoneCTI.voip](#) (dostępny od wersji 6.xx lub w nowszej)
- [TelefonCTI](#) (aplikacja znana z wersji wcześniejszych)
- [SenderSMS](#) (od wersji fw. 6.22)
- Third party applications

Depending on owned application and way of sending SMS licenses are required:

- for using *MessengerCTI.Desktop* application – sending and receiving SMS from application is included
- for using *TelefonCTI* - sending and receiving SMS requires additional licenses:
 - CTI.Manager (old name of CTI.plus)
 - License for sending SMS from third party application (since fw. 6.20)
- sending SMS from *SenderSMS* application
- sending SMS from third party application – other vendors

4.11.1 SMS service launching

For software to use SMS function, GSM modules must be configured properly (**Trunks/GSM settings**):

- for *MessengerCTI* application – field **SMSStu=Enabled**
- *TelefonCTI* and third party applications - field **SMSStu=External applications**
- *SenderSMS* application - field **SMSStu=SenderSMS**
- GSM trunks that will receive and route SMS text to particular subscribers or groups - fields **SMSSta** i **MSMDyn**.
- Additionally – from *MessengerCTI.Desktop* application SMS can be send if particular subscribers in sheet **Subscribers/CTI settings** fields **CTI** and **AwdSMS** are properly set up.

ATTENTION!

SMS was described in Centrum SMS article..

ATTENTION!

Enabling SenderSMS or External applications will cause to stop functionality of sending service notifications via SMS and sending notification about calls via MobilePhone.

4.11.2 Restrictions and charging cost for sending SMSes

Sending SMS generate costs. They are counted for subscriber costs limit in sheet **Subscribers/Charges**. According to this, in familiar manner to outgoing calls, charges table and limit for sending SMSes was introduced. PBX have default configuration of charges and limits, which are configured in sheets:

- **Global Settings/Remain** field **SMSPre** – set up maximum length of SMS Premium (SMS with higher costs)
- **Outgoing calls/SMS prefixes** – restriction table which sets up:
 - **Prg** – minimum permission of subscribers to send SMS,
 - **No.** – number of allowed prefixes and their digit amount,
 - **SMSPrc** – SMS cost for each prefix.

In table we can set up two types of SMS:

- SMS premium – with higher fee – SMS forma number without „+” before number, which length don't go over field in sheet **Global settings /Remain** in field **SMSPre**

- SMS standard - with „+” at beginning – higher number of digit then set in field **SMSPre**

SMS fee is counted – according to cost from field **SMSPre** and added to cost limit of subscriber in field **Blnc** sheet *Subscribers / Charges*.

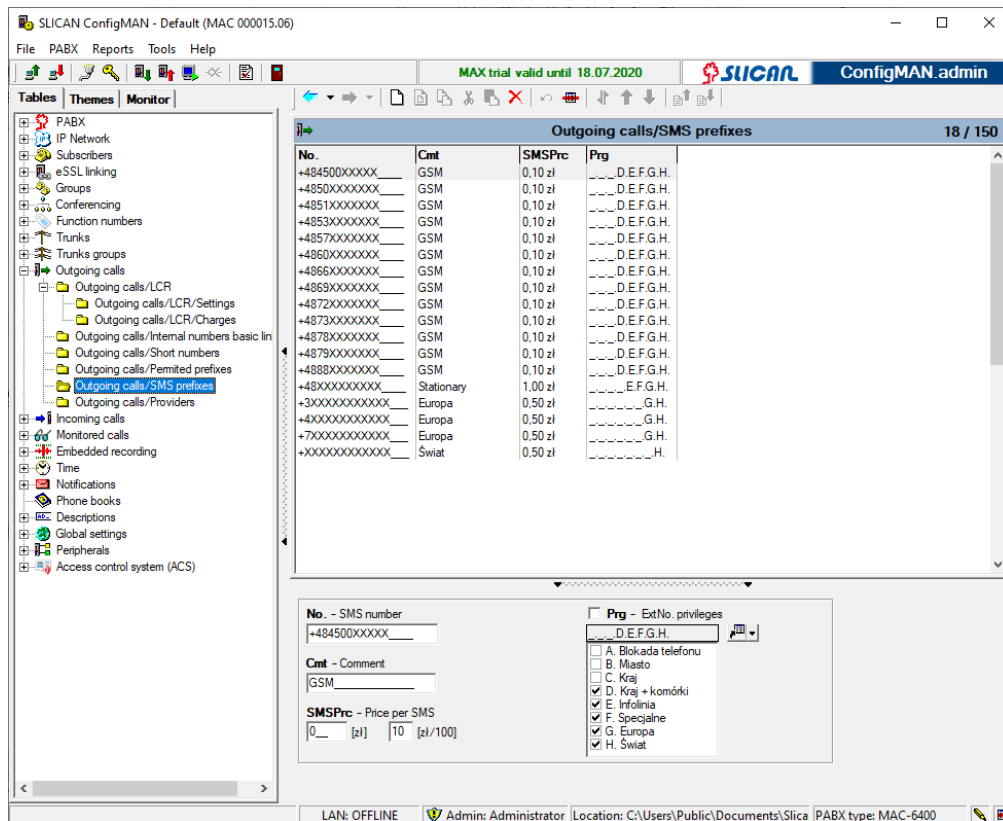


Fig 4.41: Sheet *Outgoing calls / SMS prefixes*

If length or prefix are not compatible with entered data - SMS is not send.

4.11.3 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 PtRth**.

4.12 Function numbers

Function numbers were designed mainly for automatic distribution of internal calls, depending on PBX working mode. It is familiar to rules in incoming calls but assigned to

internal calls. While dialing such number, ringing will be routed (depending on current working mode) to subscriber, group, announcement, Infoline, short number, modem, relay, service or paging group. Unique feature of functions numbers allow to route call to extensions which don't have extensions number eg. infoline, announcements, paging groups.

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 / Infoline** - 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, short number, modem, relay, service)
- **DefAct** – after choosing **DefFN** – Relay, set up how relay will act after choosing extension number
- **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. Subscriber),
- **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 (defined 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 on subscriber list on CTS system phone.
- **HidExt** - - after checking this field, function number will be not visible on subscriber list on CTS system phone.
- **UrsNo.** – after checking this field, for other linked PBX with eSSL, number won't be visible and can be doubled.

4.13 Unreleased numbers

Unreleased numbers are numbers, that can be reached only locally in one PBX, where it was created (it isn't sent to linked PBX). Voicemail number is a unreleased number by default..

Unreleased number can be created by adding new **Function number** (sheet **Functions number/Settings**) and selecting fields **DefFN** and/or **FN**, also set up field **UrsNo.** Those kind of elements can not be dialed by subscribers from other linked PBXes (with eSSL protocol).

4.14 „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 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. Maintenance of this functionality is possible using Slican system phones CTS-330, CTS-220, 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.

4.14.1 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;
 - **AnnBeg** – allows to always play announcement from beginning;
 - **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 oldest busy – every call shine on phone lamp which was free for longest time from PBX perspective.
- **WtchC** – by default transfer call immediately frees line channel. If this option is checked, channel is made free only after answering made by 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** – **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.
- **ResOut** – 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.

1. Peripherals/CTS/Hardware Settings:

- **LnRng** – setting ringing type used for line functionality. Line ringing signal can be different then set on phone for direct incoming calls.
- **DisRng** – disable phone bell when application [*ConsoleCTI*](#) is logged.
- **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.

2. 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 acceptance 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.

4.14.2 Calls maintenance

4.14.2.1 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 context 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.

4.14.2.2 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.

4.14.2.3 Park and transfer calls

Active call, both 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 handset.

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** sheet **Subscribers/Lines**;
- incoming calls = **Cwt** sheet **Subscribers/Incoming calls**, with limitation, simultaneously established calls = **MaxChn – RezOut (Abonenci/Linie)**.
- 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 waiting in queue is established using this channel.

4.15 Registering missed calls

IPx, CCT and MAC PBXes allow to manage saving history and notifications about missed calls. Functionality works for CTS system phones and MessengerCTI application..

Depending on chosen fields from sheet **Subscribers/Incoming calls** subscriber history won't register or be notifying on CTS phone and CTI applications :

- **DisM** – disable missed calls register

- **DisBsy** – disable calls register if subscriber was:
 - busy
 - caller resigned or maxium waiting time on transfer passed
 - for caller, passed CallWaiting time for target subscriber
- **DisFwd** – disable missed calls register if forwarded

5 Embedded recording

Embedded recording enables recording incoming, outgoing and internal calls recording.

Availability of calls recording depending on PBX version:

- IPL, IPM, IPU, IPS and CXS - any FW version.
- CCT-1668 - since 3.05 controller version and 1.21 VoIP cards version,
- MAC-6400 - since 3.05 controller version and 2.01 recording card version.

5.1 Storage media

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

<i>Storage media capacity CF/SD/HDD [GB]</i>	<i>CCITT G.711 A-law 64kb/s, 8kHz, mono [hours] IPX/CCT/MAC/ CXS</i>	<i>GSM 6.10 13kb/s, 8kHz, mono [hours] only MAC-6400</i>
1	34	160
4	130	630
8	270	1200
40	1300	6300
100	3300	15000
160	5300	25000

Fig 2: 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 lub HDD
MAC-6400	HDD
IPL-256	SD lub HDD
IPM-032	SD
IPU-014, IPS	mikro SD

Tabela 3: 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(of storage) can continue through several minutes(depending on storage capacity). 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).

ATTENTION!

Since 6.50 version – no demo version, recording isn't possible without license.

Up to 6.42 version – 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.
- calls realised with chosen LCR entry
- realised with chosen entry of Incoming calls table (*Incoming Calls/Settings/Additional*)

After enabling recording in field **EbdRec** also permissions of recording access levels must be set (field **AccLvl**), which changes access to them only for user for proper permissions within **RecordMAN** application.

Basic settings regarding recording calls can be configured form sheet **Embedded Recording**. Settings that can be changed are:

- method of working with applications archiving recordings: *RecordMAN*. *Server* (Hardware key, encryption, checking application connection state) *and FTP* (password, encryption)
- method and recording profiles
- informing caller about recording
- maximal amount of time of saving recordings on pbx

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**.

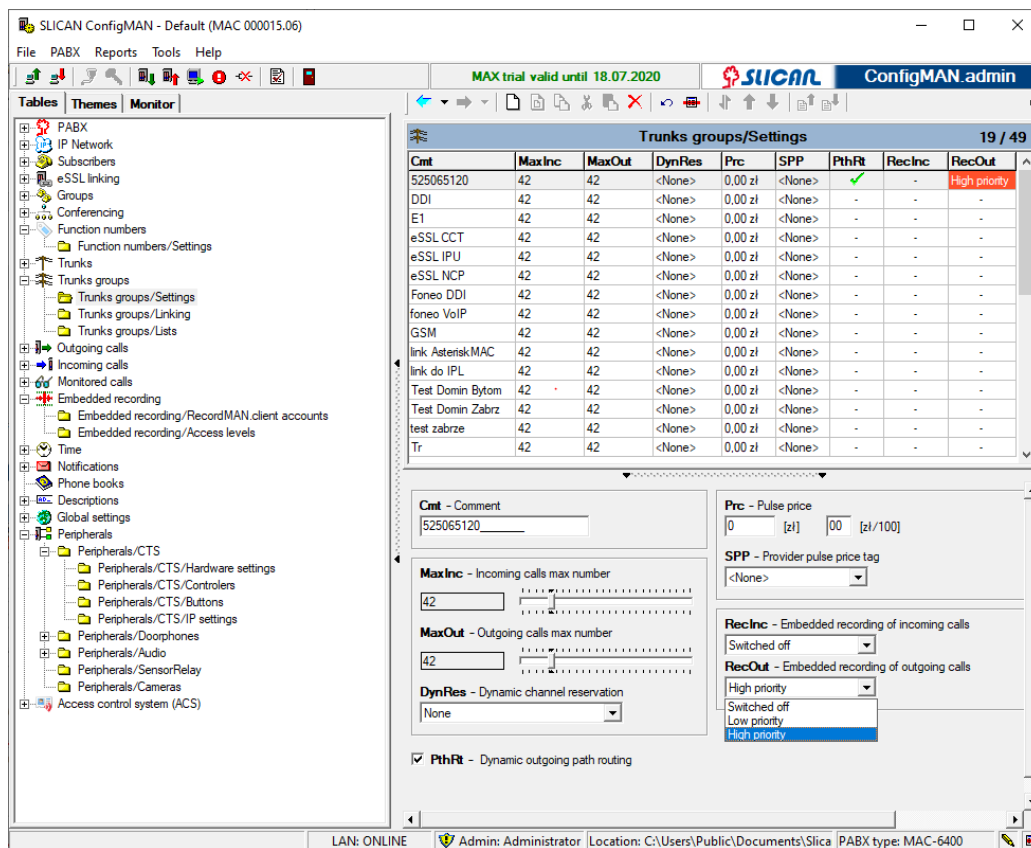


Fig 5.1: Call recording settings for trunk group

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 interrupt recordings with option *low priority*, if all recording channels are busy.

5.2.2 Recording calls through LCR rules

Settings concerning automatic recording calls established through LCR rules can be found in sheet **Outgoing calls/LCR/Settings**.

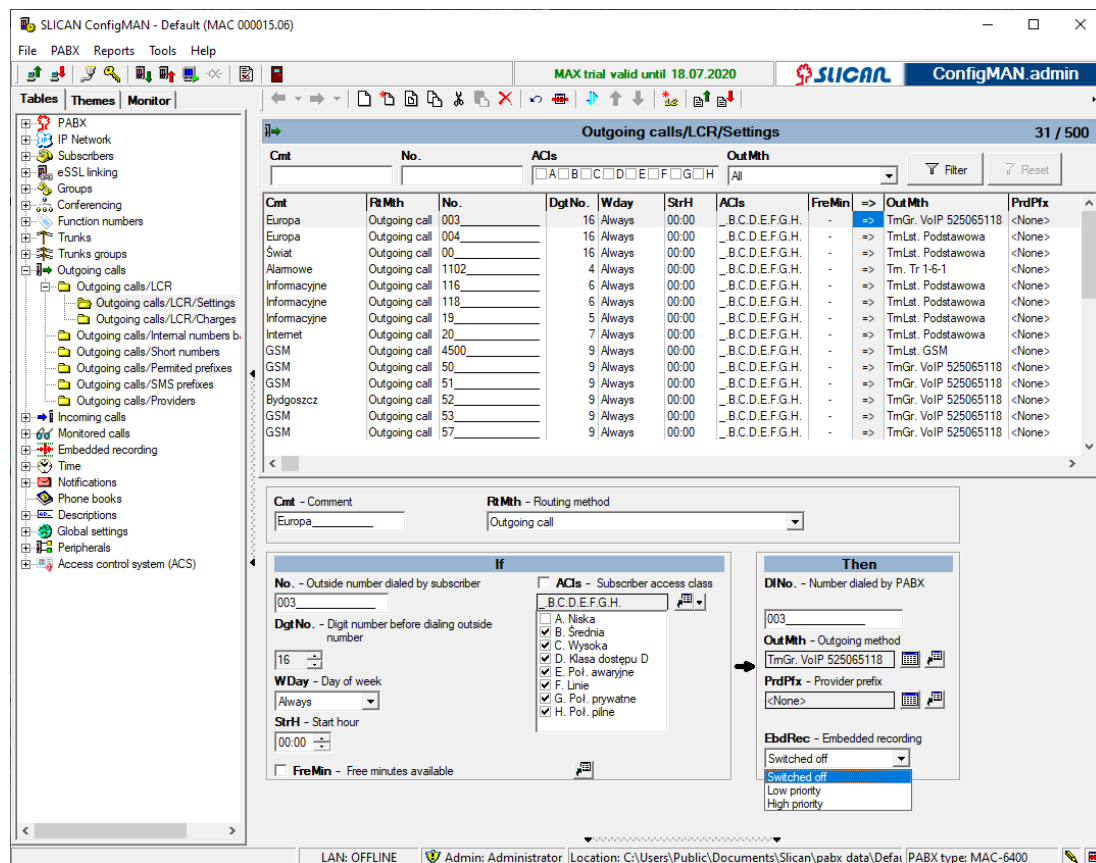


Fig 5.2: Call recording settings for LCR

For EbdRec field there are following settings:

- *Switched off* – calls with this rule won't be recorded.
- *Low priority* – calls will be recorded if calling extensions doesn't have enabled option *Prohibited* for field **EbdRec** of sheet **Subscribers / Embedded recording**. 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 resumed. 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** - calls will be recorded if calling extensions doesn't have enabled option **Prohibited** for filed **EbdRec** of sheet **Subscribers / Embedded recording**. recording with such defined priority will be interrupt recordings with option **low priority**, if all recording channels are busy.

5.2.3 Recording calls through Incoming calls table rules

Settings concerning automatic recording calls established through Incoming calls table rules can be found in sheet **Incoming calls /Settings /Additional**.

The screenshot displays the SLICAN ConfigMAN interface. The main window shows a table titled 'Incoming calls/Settings/Additional' with 14 rows and 10 columns. The columns are: Cmt, TrmGr, DIno, WlkMd, MaxInc, CDm, EbdRec, VIP, PthRt, and WtMel. The 'EbdRec' column is highlighted in red for the first row, indicating 'High priority'.

Cmt	TrmGr	DIno	WlkMd	MaxInc	CDm	EbdRec	VIP	PthRt	WtMel
E1	TrmGr: E1	521234567	A.B	No limits	<None>	High priority	-	-	Standan
525065117	TrmGr: VoIP 525065117	525065117	A.B	No limits	<None>	-	-	-	Standan
684518430	TrmGr: Pawlikowski	684518430	A.B.C.D.E.F.G.H	No limits	<None>	-	-	-	Standan
684518490	TrmGr: Pawlikowski	684518490	A.B.C.D.E.F.G.H	No limits	<None>	-	-	-	Standan
684518640	TrmGr: Pawlikowski	684518640	A.B.C.D.E.F.G.H	No limits	<None>	-	-	-	Standan
684520122	TrmGr: Pawlikowski	684520122	A.B.C.D.E.F.G.H	No limits	<None>	-	-	-	Standan
684594627	TrmGr: Pawlikowski	684594627	A.B.C.D.E.F.G.H	No limits	<None>	-	-	-	Standan
tr2	TrmGr: tr2	A.B	A.B	No limits	<None>	-	-	-	Standan
Wejście 7	TrmGr: VoIP 525065119	525065119	A.B	No limits	<None>	-	-	-	Standan
525065118	TrmGr: VoIP 525065118	525065118	A.B	No limits	<None>	-	-	-	Standan
GSM 2	TrmGr: GSM	A.B	A.B	No limits	<None>	-	-	-	Standan
tr	TrmGr: Tr	A.B	A.B	No limits	<None>	-	-	-	Standan
DO NCP	Out.	529992001	A.B.C.D.E.F.G.H	No limits	<None>	-	-	-	Standan
Pozostałe	<Any>	A.B	A.B	No limits	<None>	-	-	-	Standan

Below the table, there is a configuration section for the 'If' condition. It includes fields for 'Cmt - Comment' (E1), 'TrmGr - Trunks group' (E1), 'DIno - Dialed number' (521234567), and 'WlkMd - Working mode' (A.B). The 'Then' section includes settings for 'MaxInc' (Maximum simultaneous calls), 'CDm' (Maximal call duration), 'EbdRec' (Embedded recording), 'WtMel' (Melody on waiting), 'RngMel' (Melody on ringing), and 'HldMel' (Melody on hold). The 'EbdRec' field is set to 'High priority'.

Fig 5.3: Call recording settings for Incoming calls table

For EbdRec field there are following settings:

- **Low priority** – calls will be recorded if calling extensions doesn't have enabled option **Prohibited** for filed **EbdRec** of sheet **Subscribers / Embedded recording**. 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 resumed. 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** - calls will be recorded if calling extensions doesn't have enabled option **Prohibited** for filed **EbdRec** of sheet **Subscribers / Embedded recording**. recording

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

5.2.4 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***:

1. ***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 CTSphone. In such situation, subscriber A by pressing key with programmed recording service (programing method is described [Programmable buttons configuration in CTS system phones](#)) can records call.
2. ***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).
3. ***Only internal*** – by this settings only internal calls of this extension number will be recorded.
4. ***Only external*** – by this settings only external calls (incoming and outgoing) of this extension number will be recorded.
5. ***All calls*** – both, internal and external calls of this extension number will be recorded.

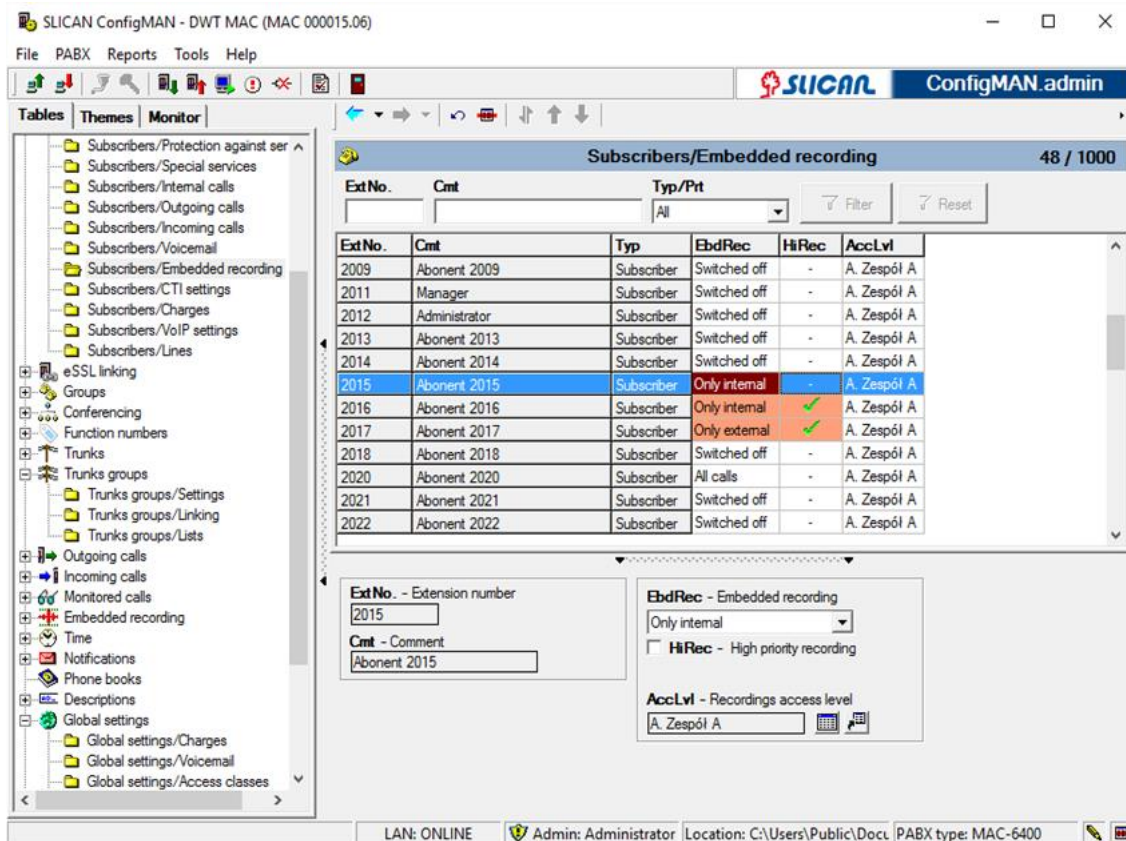


Fig 5.4: Settings concerning recording calls of specified subscriber

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

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

5.2.5 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 opposite side recording is permitted, it will record.
- **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.

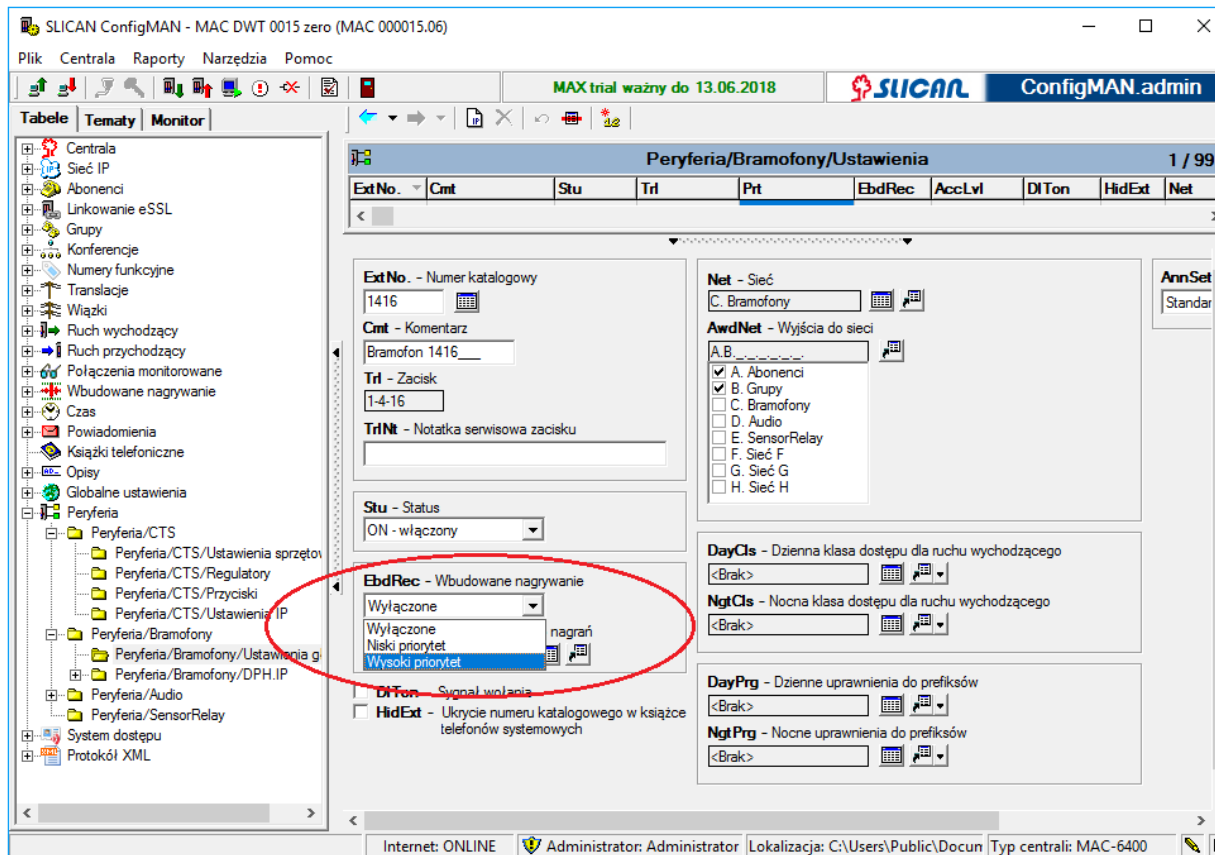


Fig 5.5: Settings of doorphone calls recording

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!

Karta nagrywania rozmów automatycznie kasuje najstarsze katalogi. Jeżeli na karcie jest tylko bieżący katalog (z bieżącą datą), to usuwane są kolejno najstarsze pliki. Ma to miejsce wtedy, gdy wolny obszar na dysku twardym (karcie CF/SD) jest poniżej 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),

00314 - serial number,

114464MB - capacity of storage media (always in megabytes),

HDD - storage media type (CF, HDD),

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
```

, gdzie:

xxxxxxx - call identifier [0000001..9999999],

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

yyymmdd - call date:

yyyy - year [np. 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 happened 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 (demo version),

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

P (recording prohibited) - recording continuation prohibited

(call with recorded disa DISA is routed to subscriber with prohibited recording)

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) - resume 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]

Notes concerning creating wav files:

- If there is no target subscriber (ex. call was finished on infoline or DISA announcement), 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 sign is replaced by underscore_.
- If there is no extension number or comment, sign „_” is entered.
- If number is restricted or there is no number presentation, sign „_”.
- 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 this situation synchronization may take 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. 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.

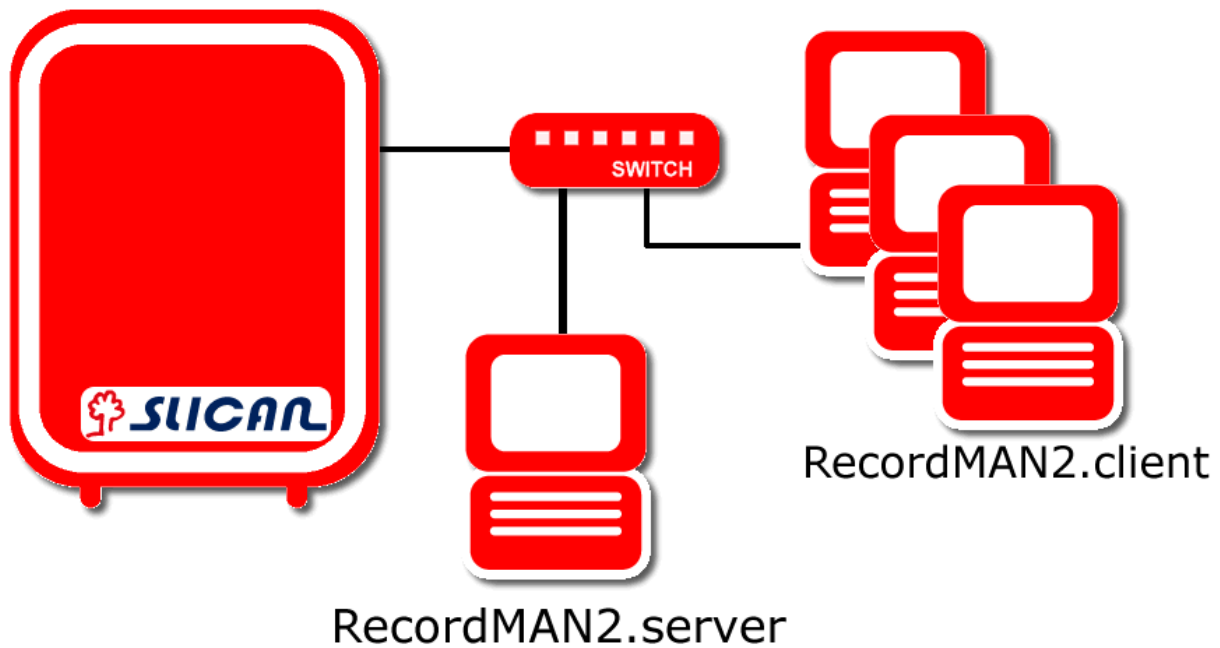


Figure 5.6: Embedded recordings 2 – dedicated PC as recordings server

5.4.1 RecordMAN.server

Main task of **RecordMAN.server** application is downloading recordings from storage media installed in PBX and copying them on hard drive of PC in LAN network. Transmission between PBX and application is encrypted. Copy is used to secure recording files. It is specially important for PBXes, that use memory card storage (*CompactFlash* or *SD card*) – this card operates as a buffer, so while large calls number and limited card capacity oldest recording may be erased before copying. For PBXes with HDD card, main purpose 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 or WAN for IPx or, CCT or LAN MACREC card (in MAC-6400) in sheet *IP Networks / LAN Interfaces* according to chapter [PBX configuration in company's network – LAN mode](#). 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).

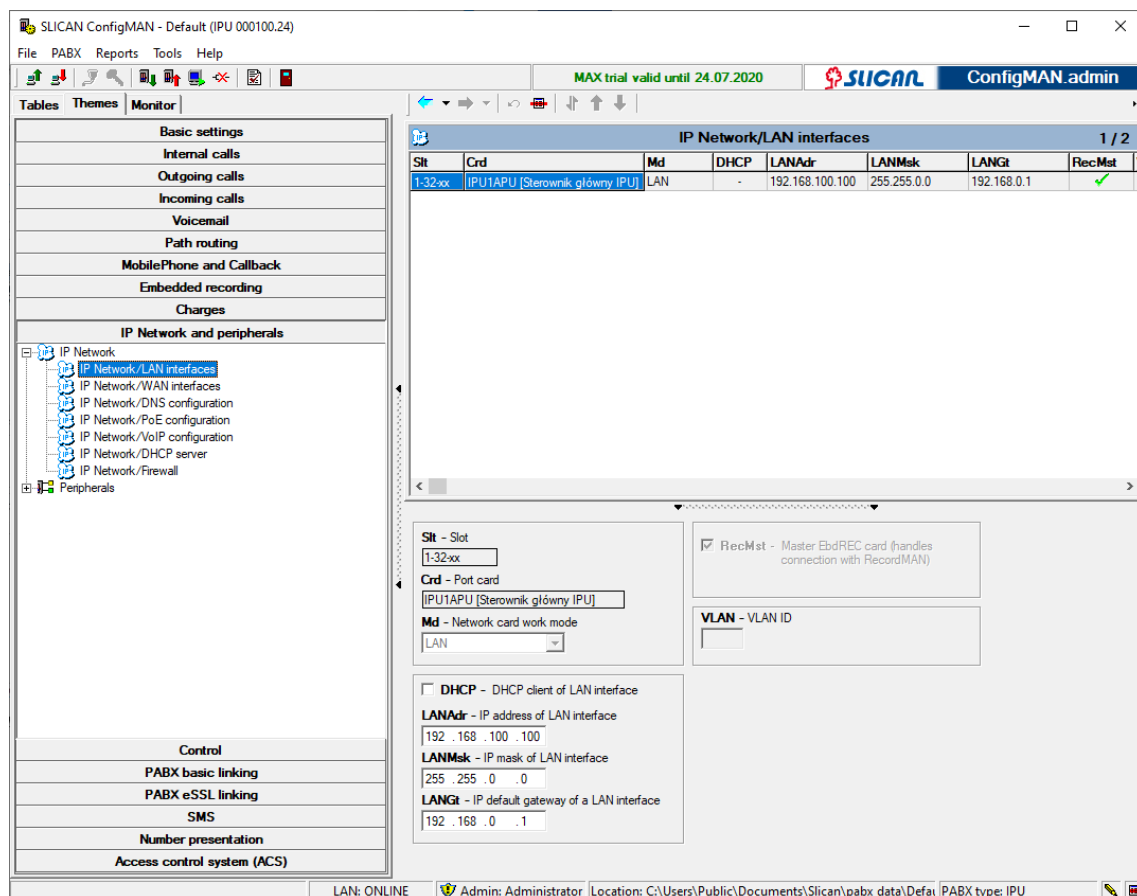


Figure 5.7: LAN interface configuration for cooperation of IPX and CCT with RecordMAN application

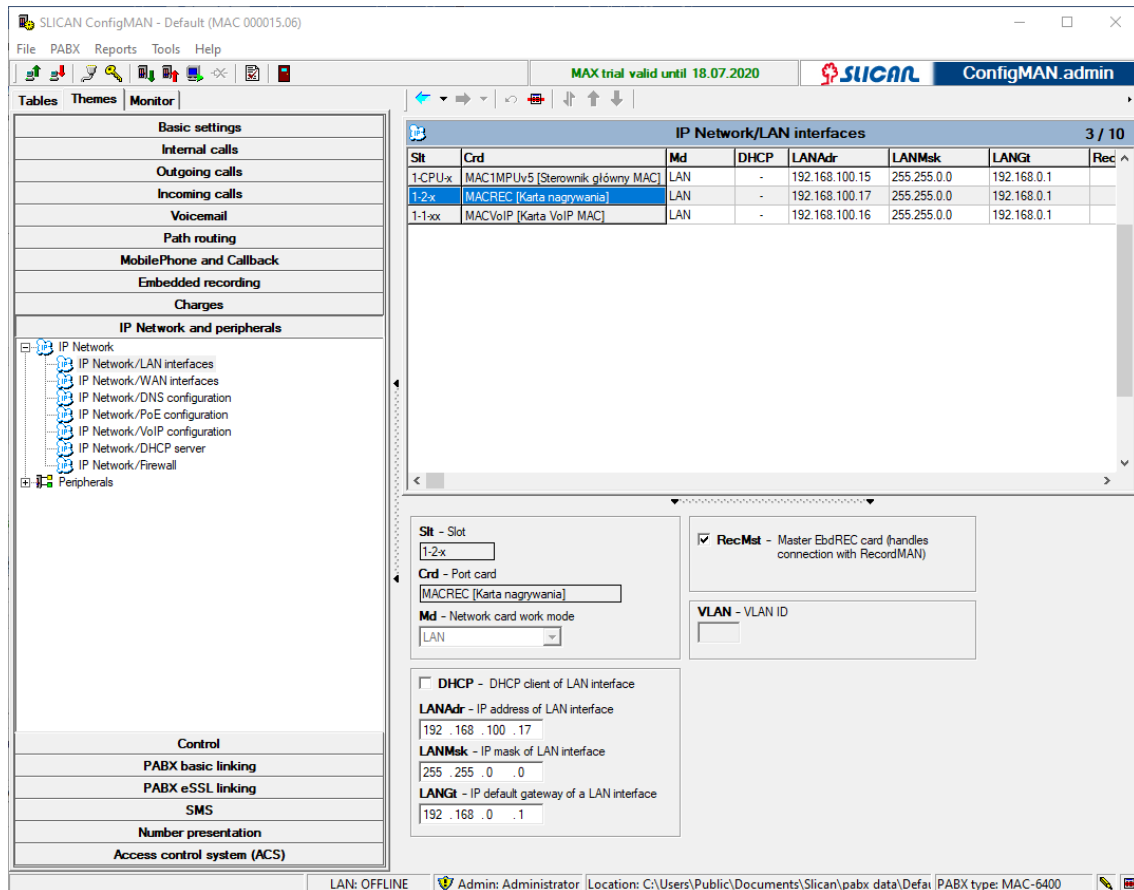


Figure 5.8: LAN interface configuration for cooperation of MAC-6400 with RecordMAN application

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).

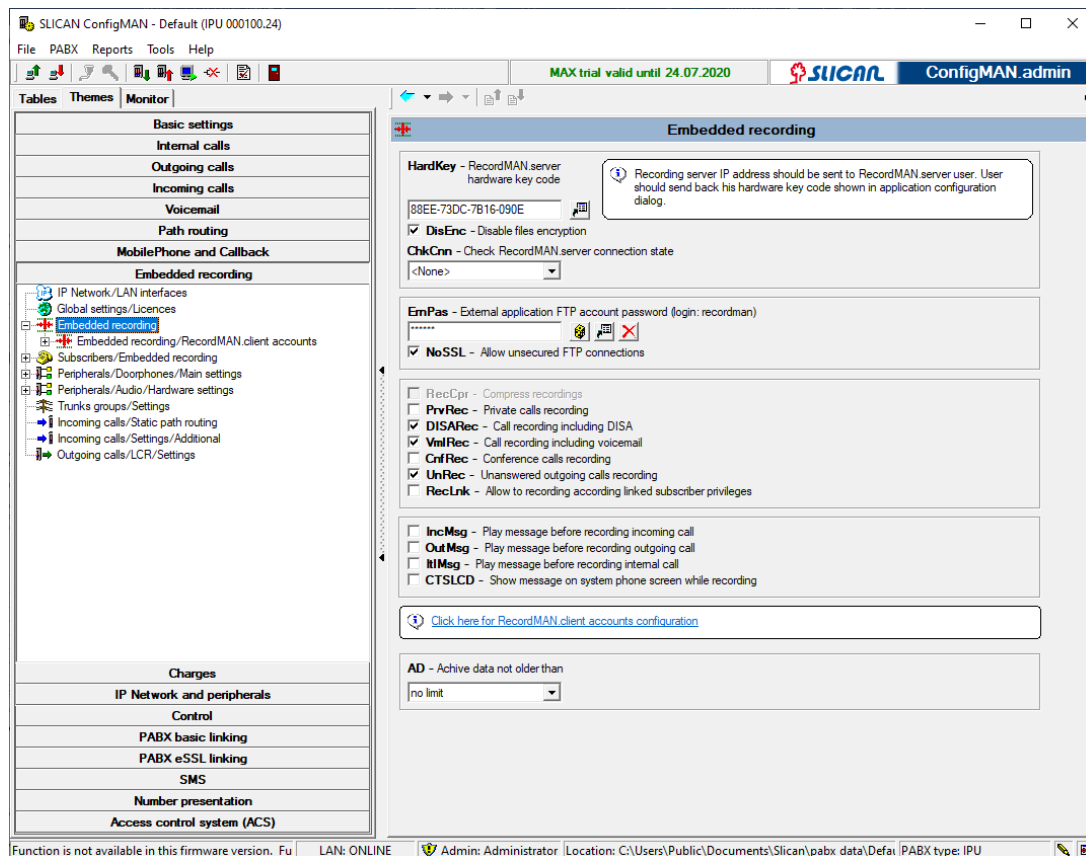


Figure 5.9: Entering HardKey in sheet Embedded recordings

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. Transmission is encrypted. 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 clients² (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 IPx and CCT-1668) or MACREC (in MAC-6400) in sheet **IP Networks / LAN Interfaces** according to chapter [PBX configuration in company's network – LAN mode](#). 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 **Globalne 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** – password,
- **HardKey** - used also for user authorization user. This code is unique for every computer, where 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 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 **Peripheral / 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 [Licenses - 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” (encrypted connection).

5.4.4 RecordMAN.server connection status

To increase the safety of call recordings stored on additional drive, 6.51 PBX firmware version introduces new monitoring system for **RecordMAN.server** application connection with user's PC. Application will monitor PBX for recording files that aren't archived yet and will download them. In **Embedded recording** tab, field **ChkCnn**, **PBX** administrator can choose how often the connection status will be checked.

Available settings:

- None – PBX will not check connection status;
- every 10 minutes – PBX will register an event, when application will not respond through 10 minutes (setting for servers working 24/7)
- co 24 godziny - centrala zarejestruje zdarzenie, jeśli wystąpi brak komunikacji 24 hours – PBX will register an event, when application will not respond through 24 hours (settings for servers turned off every day – e.g. user's PC)

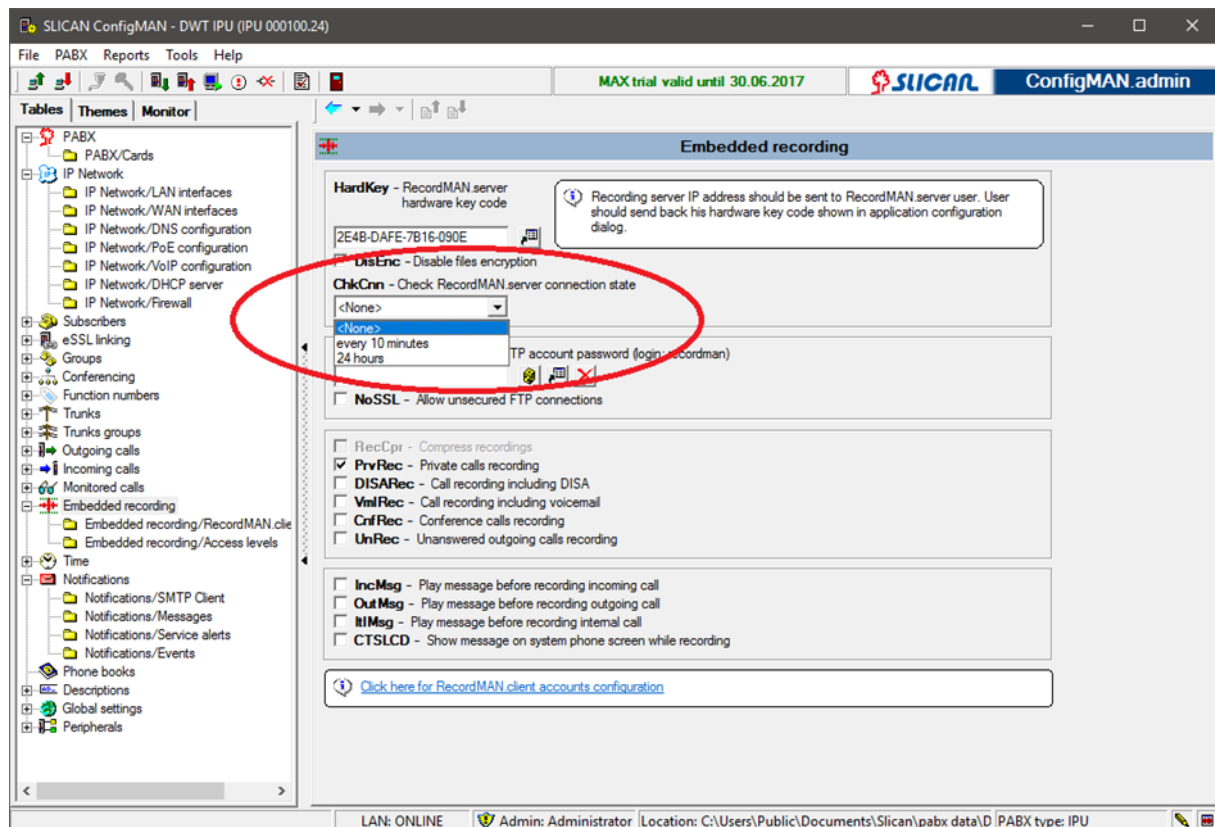


Fig 5.10: Monitoring RecordMAN

Events are registered in PBX's memory and they are visible in PBX events in PBX monitor. Additionally, those events can be sent by a notification in SMS or e-mail message, by configuring them in **Notifications/Service alerts** tab in **RecFlr** field.

WARNING!

Each PBX reboot will start the clock of sending notifications from zero due to timer working from volatile memory. This information is crucial for settings "24 hours", because of notification long time interval.

5.5 Optimization of system resources needed for call recording

In **PBX monitor** statistics are introduced, which inform about using recording channels. Purpose of these statistics is answer the question if using system resources and number of acquired license codes for recording channels enable appropriate quality of embedded recording calls system.

Following parameters are registered since last PBX reset or turning on:

- *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.

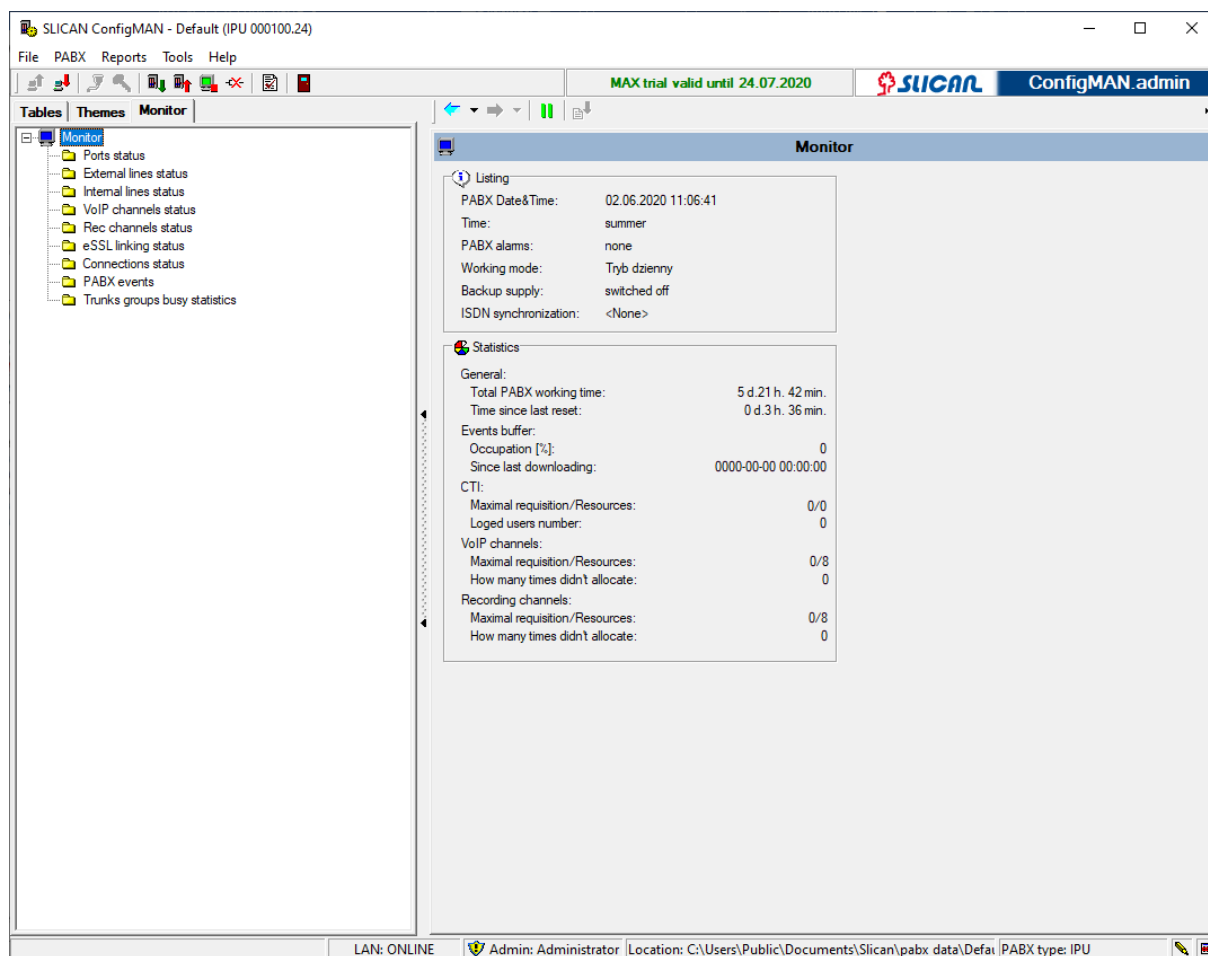


Fig 5.11: Statistics in PABX monitor

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:

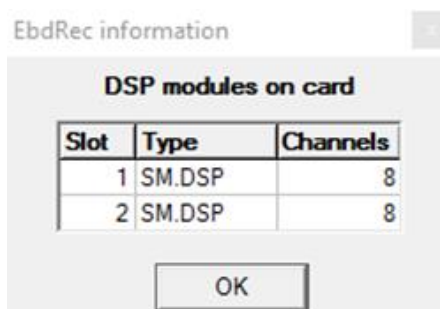


Fig 5.12: Info about available REC channel

6 Charges

In Slican PBXes it is possible to define connections charges. It can be realized in two ways:

- directly from inside *ConfigMAN* with cooperation with additional applications: hotel applications,
- using specialized charging application – *BillingMAN*.

Charging directly from *PBX* application is used when:

- we using limits for calls from extension numbers,
- chcemy wyświetlać aktualny koszt połączenia na aparatach systemowych CTS.

In other situations we using other applications – hotel applications, or 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 to dialed prefix and trunk.

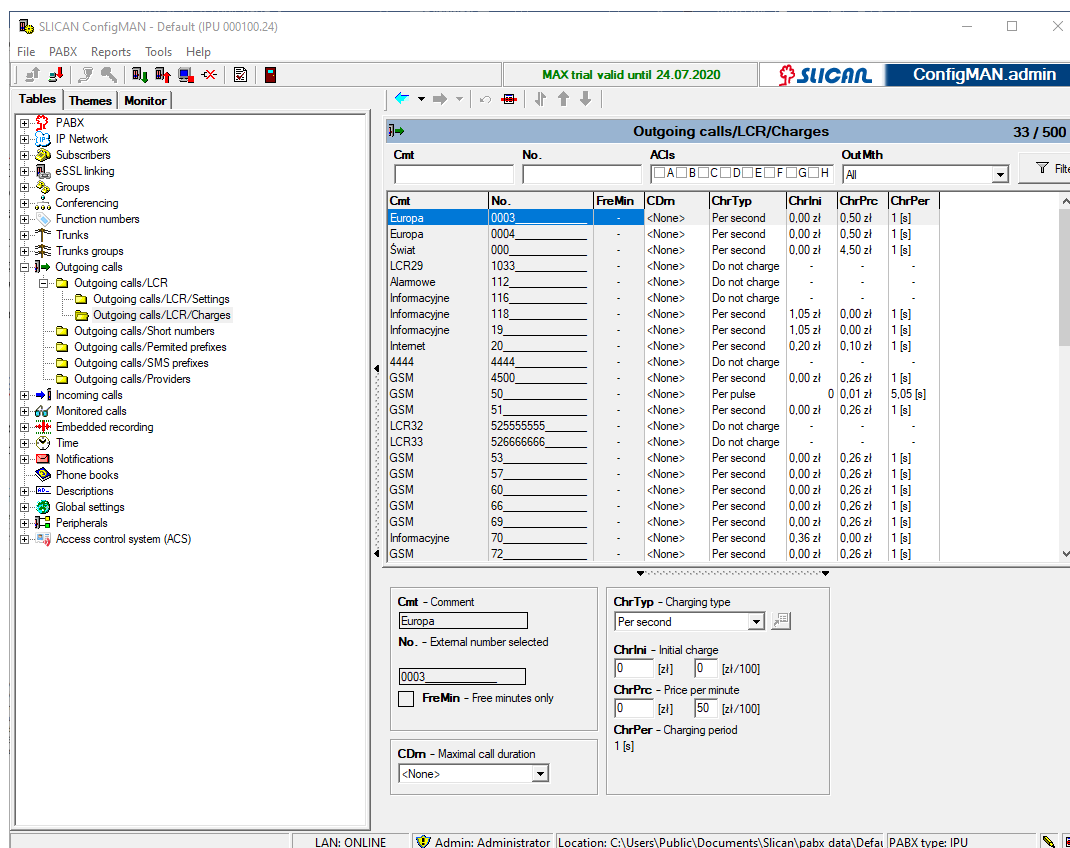


Fig 6.1: Sheet – Outgoing calls / LCR / Charges

3 types of charges are distinguished:

- Per pulse – rarely used
- Per second – commonly used
- Use trunk group settings – operator provides charge prices – currently rarely used

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/Short Numbers**, charging is possible only according parameters received from provider,
- calls are routed via **LCR**, call prices are set according parameters defined in **Outgoing Calls/LCR/Charges**.

6.2 Setting call limits for subscribers

If from extension number outgoing digit is dialed and than number assigned in **Permitted prefixes table** to A privileges (less privileges), such connection there is no charging (field **Blnc** in **Subscribers/Charges**) isn't incremented) because this connection is treated as internal(no charges). Similarly if from extension number prefix from **LCR** is dialed and than the same number as above (with A privileges defined in **Permitted 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.

The screenshot shows the SLICAN ConfigMAN interface. The left sidebar contains a tree view with categories like PABX, IP Network, Subscribers, and eSSL linking. The main window displays the 'Subscribers/Charges' table with 21/100 entries. Below the table, there is a configuration panel for a selected extension (3001).

ExtNo.	Cmt	Typ	Lmt Typ	Rchrg	Limit	Blnc	BlncRat
3001	Abonent 3001	Subscriber	None	0.00 zł	0.00 zł	3.77 zł	
3002	Rafal test	Subscriber	None	0.00 zł	0.00 zł	0.00 zł	
3003	Abonent 3003	Subscriber	None	0.00 zł	0.00 zł	0.00 zł	
3004	Abonent 3004	Subscriber	None	0.00 zł	0.00 zł	0.00 zł	
3006	Abonent 3006	Account	None	0.00 zł	0.00 zł	0.00 zł	
3007	Powiadomienia	Account	None	0.00 zł	0.00 zł	0.00 zł	
3008	Abonent 3008	VoIP	None	0.00 zł	0.00 zł	42.96 zł	
3009	Abonent 3009	VoIP	None	0.00 zł	0.00 zł	15.72 zł	
3010	Abonent 3010	Subscriber	None	0.00 zł	0.00 zł	0.88 zł	
3011	Abonent 3011	Voicemail account	None	0.00 zł	0.00 zł	0.00 zł	
3012	Abonent 3012	Account	None	0.00 zł	0.00 zł	0.00 zł	
3013	Abonent 3013	Voicemail account	None	0.00 zł	0.00 zł	0.00 zł	
3014	konto przekierow	Account	None	0.00 zł	0.00 zł	0.18 zł	
3015	skrynykat 3015	Voicemail account	None	0.00 zł	0.00 zł	0.00 zł	
3016	Abonent 3016	<None>	None	0.00 zł	0.00 zł	94.15 zł	
3017		<None>	None	0.00 zł	0.00 zł	9.76 zł	
3018	Abonent 3018	<None>	None	0.00 zł	0.00 zł	12.84 zł	

Configuration details for ExtNo. 3001:

- ExtNo.: 3001
- Cmt: Abonent 3001
- Lmt Typ: No limit
- Rchrg: Automatic recharging amount (0 zł)
- Limit: Available limit for calls (0 zł)
- Blnc: Charging costs balance (3.77 zł)
- BlncRat: Balance resetting

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 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
- SMS
- internal calls,
- services
- access control system

After checking field **EvtRgs** it is possible to block registering of some events, outgoing, incoming calls and SMS messages.

Below, next fields influence registering calls within internal calls (realised and not realised)Poniżej, kolejne pola wpływają na rejestrację połączeń w ruchu wewnętrznym (zarówno zrealizowanych jak i niezrealizowanych).

Next fields allow to register services, relay events and sensor state. Services which won't be registered: picking up call, urgent ring, CLIR, blocking phone, checking limit.

Below, there are options to turning registering events of Slican Access Control System (ACS), based on doorphones and DPH.IP RF readers.

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 /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/Hardware Settings**.
- Sends SMS with information about filling buffer to number defined in field defined in field **Rcp**, in sheet **Notifications/Service Alerts** – conditions of sending such SMS are following: GSM card is installed in PBX and record with event **EvtLog** was created.

If buffer fulfilling continues, if data in buffer will be overwritten, diode BUF/STATUS – depending on PBX type – will start blinking. This diode can be switched off by downloading calls with **BillingMAN** application with selected option *Display reset*. If field **AlrBuf** in sheet **Global Settings/Charges** 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. Each PBX uses:

- IPL - 2 relay and 2 sensor card or 4 relay and 4 sensor
- IPM - 2 relay and 2 sensor card
- IPU – port on controller card 1 relay and 1 sensor (IPU-14.101, IPU-14.102), 2 relay and 2 sensor card
- IPS-08.101, IPS-08.104 - port on controller card 1 relay and 1 sensor
- CXS-0424 - dedicated IORS card - 2 przekaźników i 2 sensorów
- MAC-6400 – 4 relay and 4 sensor card
- CCT-1668 - 4 relay and 4 sensor card
- Optionally Slican MSB-1102 Multi Switch Box can be used as interface between PBX and controlled device. Multi Switch Box is connected to PBX subscriber port and can operate as sensor or relay 2 x relay and 2 x sensor or 1 x relay and 1 x sensor).

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 **Peripherals/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** – physical port number
- **TrlNt** – terminal service noty (displayed only in **ConfigMAN**)
- **Net** – assigning to subscriber net, used to limit privileges for incoming internal calls
- **PIN** – security code, which prevents unauthorized access
- **Stu** – switch on/off specified port
- **Cmb** – combine relay with sensor, allowing to display current status of relay on CTS button
- **OnAnn** – announcement played after triggering relay or sensor
- **OffAnn** - announcement played after stand-by state of sensor or relay in bistable mode
- **ShwExt** – after checking this field, extension number of this device is included in system phones extension number list

Additional sensor settings:

- **NrmSns** – normal sensor state
- **IntTi** – sensor integration time, which must pass with sensor inactive state to return to normal state
- **EvgRgs** – which sensor events will be registered
- **Act** – action that will be done after sensor is triggered

Additional settings for sensor working as bistable:

- **RelSt** – set up how to treat idle state (unlatched, latched)

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)
 - parametric loop open (EOL mode) – only IPL-256 and IPM-032
- closed
 - open loop (CCS mode)
 - voltage between loop terminals (DCI mode)
 - parametric loop open (EOL mode) – only IPL-256 and IPM-032

Detailed description and connection diagram in every mode for each type of PBX can be found in instruction manual and article [Automation and notification in Slican Servers](#). To check sensor status number included in field **ExtNo**, sheet *Peripherals/SensorRelay* should be dialed. If PIN code is entered in configuration, this code should be also dialed during establishing call, confirming with #. 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). Another way of monitoring sensor state is assigning it to button of CTS system phone or configure SMS notification about changed sensor state. With parameter **NrmSnr** default status of sensor is defined (close/open).

Additionally there is option of setting up **Act** action during triggering sensor, which is assigned to relay within field **Cmb**. In case of trigger sensor, we can set up action:

- None
- Impulse or switch on
- Switch off (only for sensor in bistable mode)
- State change (only for sensor in bistable mode)

MSB-1102 isn't required if we want to check loop state open/closed (doesn't apply to voltage). Just connect device to PBX subscriber port, just remember that MSB-1102 I guarantee higher security due to electric separation PBX circuit and device circuit.

Functionality and configuration of sensor with MSB-1102 and RL/SN card is identical.

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 *Peripheral/SensorRelay*.
- by voltage level – only for RL/SN or 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 confirming with #.

Next:

- for RL/SN or IORS card:
 - pressing digit „1” – monostable mode – trigger relay or for bistable mode – switch on relay
 - pressing digit „0” - bistable mode – turn off relay
- for MSB-1102:

- pressing digit 1 – monostable mode - trigger relay

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.

Changing relay state can be triggered by:

- externally – call coming through [Static routing table](#)
- based [Control schedule](#) defined for PBX.
- Call to function number

Sheet **Function Numbers/ Settings** as default action for field **DefFN**, or proper working mode **FN** set up Relay. Field **DefAct** or **Act** assigns relay action:

- None
- Impulse or switch on
- Switch off (only for relay with bistable mode)
- State change (only for relay with bistable mode).

Above steps allow to trigger relay action without additional use of phone buttons **0** or **1** to choose digits.

Functionality and configuration of relay with device MSB-1102 and RL/SN card differ. Relay with MSB works only as monostable and is trigger only once, constant time – 1 second. Relay from RL/SN card can work either as monostable with regulated trigger time up to 60 seconds or as bistable.

7.3 Control schedule;

Relays, doorphones and restarting PBX can be triggered cyclically according to predefined schedule. Terms (up to 50) can be defined in sheet **Time/Control schedule**.

Options available for schedule:

- **Cmt** – comment
- **WDay** – week day for this action
- **Ti** – Time of triggering action (hh:mm)
- **Cnl** – element which is triggered by action
- **Act** – selected action, depends on controlled element

Control action (**Act**) may mean:

- Switch on or off target relay
- Auto reset PBX service – immediately or after closing all connections
- Trigger temporarily or until canceled of electrolock EZ or/and STA relay of chosen DPH.IP doorphone or all doorphones of chosen ACS area
- Turn off electrolock EZ or/and STA relay of chosen DPH.IP doorphone or all doorphones of choosen ACS area

If current sheet, for DPH.IP doorphone action or ACS area access won't check field **EZ** and/or **STA**, then as default will be taken control elements (**EZ**, **STA**) from field **Act** sheet **Peripherals/Doorphones/DPH.IP/Phone settings**

8 System devices

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

Centrala	<i>CTS-102.CL / CTS-102.HT, CTS-202.CL/CTS- 203.CL/CTS-220/CTS-330 (U terminal)</i>	<i>CTS-102.IP, CTS-203.IP CTS-220.IP CTS-330.IP</i>	<i>Total of CTS and CTS.IP</i>
CXS-0424	12	18	24
CCT-1668 firmware >=5.11	18	60	60
MAC-6400	124	124	124
IPL-256	60	124	124
IPM-032	24 ^{IPM.WM} /32 ^{IPM.EU}	124	124
IPM-032 Low Cost	8	10	10
IPU-14.101 IPU-14.103 IPU-14.105	4	28	28
IPU-14.102 IPU-14.106	4	68	68
IPS-08.100 IPS-08.101		16	
IPS-08.104 IPS-08.105		8	

ATTENTION!

While choosing model type, be aware of PBX minimum required firmware version:

- CTS 330 – fw. 6.20
- CTS 203.CLBT – fw. 6.22
- CTS-220 – fw. 6.50

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

8.1 Private phone book edition

Edition of public phone book can be done only from licensed applications Slican [WebCTI](#)/ [MessengerCTI.Desktop](#)/ [ConsoleCTI](#) or from CTS system phone.

8.2 Public phone book edition

Edition of public phone book can be done only from licensed applications Slican [WebCTI](#)/[MessengerCTI.Desktop](#)/ [ConsoleCTI](#) or from CTS system phone. Without license public phone book can be edited only on subscriber set up CTI.admin (**Subscribers/CTI settings** field **WebAdm**) or login through web browser to *WebCTI* of this subscriber. Changing and adding of entries is possible from sheet **Administration/ Public book**.

Condition that allows editing phone book is having proper permissions by subscriber extension:

- For application *WebCTI* (admin): sheet **Subscribers/CTI Settings** field **WebAdm** (this functionality is not subject of licensing). Changes and adding new entries is possible using application *WebCTI*, from sheet **Administration/ Public book**.
- For applications *WebCTI/ MessengerCTI.Desktop* (user, manager) and *ConsoleCTI*: sheet **Subscribers/Special Services** field **PubPB** = Read and write.
- for CTS system phone: sheet **Subscribers/Special Services** field **PubPB** = Read and write.
- Numbers can be also dialed by index – with code ****XXX** – this index can be edit from application *WebCTI*.

Instrukcje obsługi aplikacji Slican [WebCTI](#), [MessengerCTI.Desktop](#) i [ConsoleCTI](#) znajdują się w serwisie pubwiki.slican.com / pubwiki.slican.pl

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**.*

*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(not applicable for all services). On this sheet we can select if we programming phone buttons or buttons on console which is connected to phone.

As default for system phones CTS-202 i CTS-203 up to two CTS-232 consoles can be connected, although to four system phones in PBX up to five system consoles can be connected. It allows to connected additional BLF maximum up to – for two consoles $2 \times 30(\text{console}) + 12(\text{CTS}) = 72$ and for 5 consoles $5 \times 30 + 12 = 162$. Additional consoles are configured by checking field **CnsSrv** – *Additional console service* in sheet **Peripherals/CTS/Buttons**.

For phones CTS-220 i CTS-330 one console CTS-338 can be connected. Depending on phone type we have– $38(\text{console}) + 19 + 5(\text{CTS-330}) = 62$ lub $38(\text{console}) + 8(\text{CTS-220}) = 46$ BLF button. Additionally, up to four phones have option of connecting four consoles. We connect them just like CTS-202/203 phone. With phones this allow to get for CTS-330 $4 \times 38 + 19 + 5 = 176$ and for CTS-220 $4 \times 38 + 8 = 160$ additional buttons.

Each button can be programmed as:

- subscriber,
- group,
- conference (conference group or room),
- doorphone,
- open doorphone DPH.IP
- open EZ DPH.IP
- open STA DPH.IP
- control EZ DPH.IP
- control STA DPH.IP
- audio device,
- sensor
- relay
- trunk group
- channel
- line
- phone number
- service (including call recording and supervisor²)
- function
- working mode
- eSSL number
- [function number](#)
- [auditorium subscriber](#)
- [announcement play \(PL\)](#)

² This service can be programmed only from inside **ConfigMAN**.

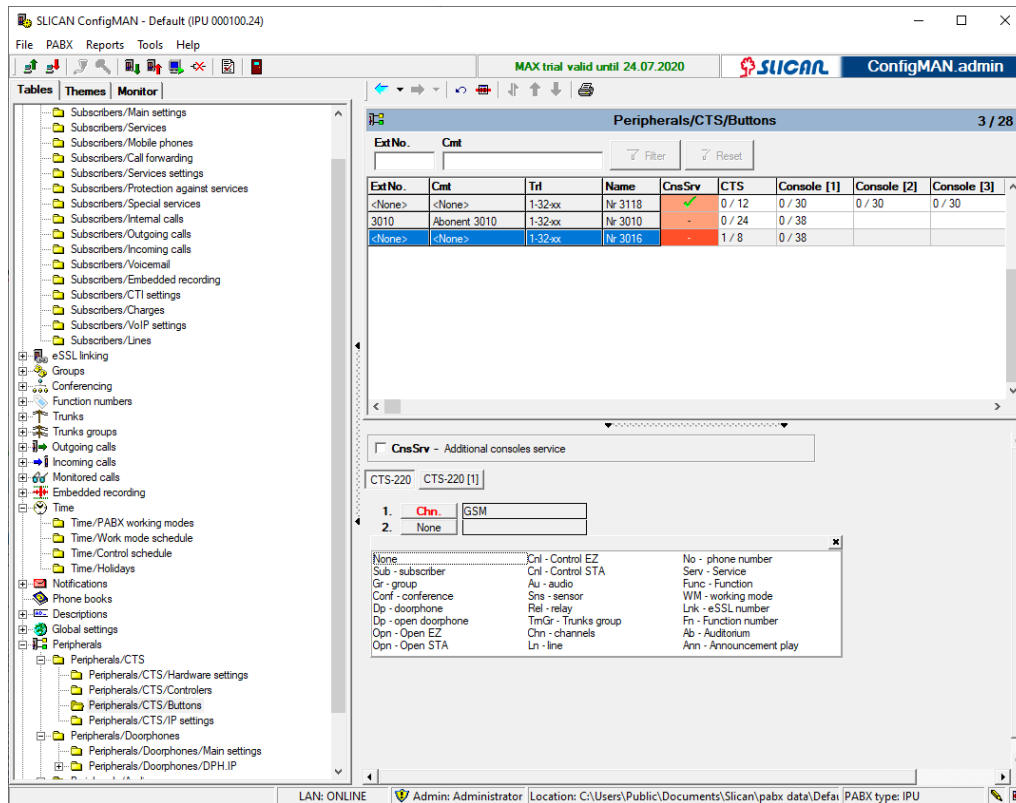


Fig 8.1: Sheet – Devices / CTS / Buttons

From inside *ConfigMAN* application etiquette printout is possible. To do this option *CTS button etiquette* from menu *Reports* should be selected. Subscriber can print etiquette on his own using *WebCTI* sheet *Settings/System phone*.

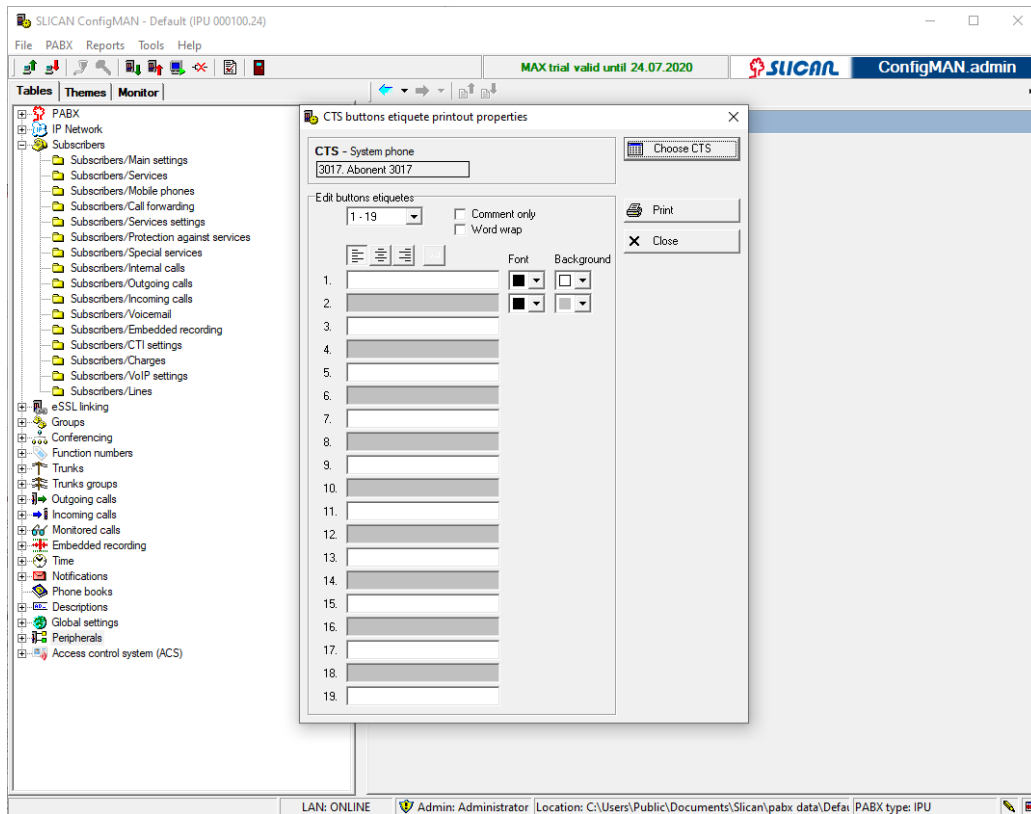


Fig 8.2: :: Button etiquette for CTS

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.

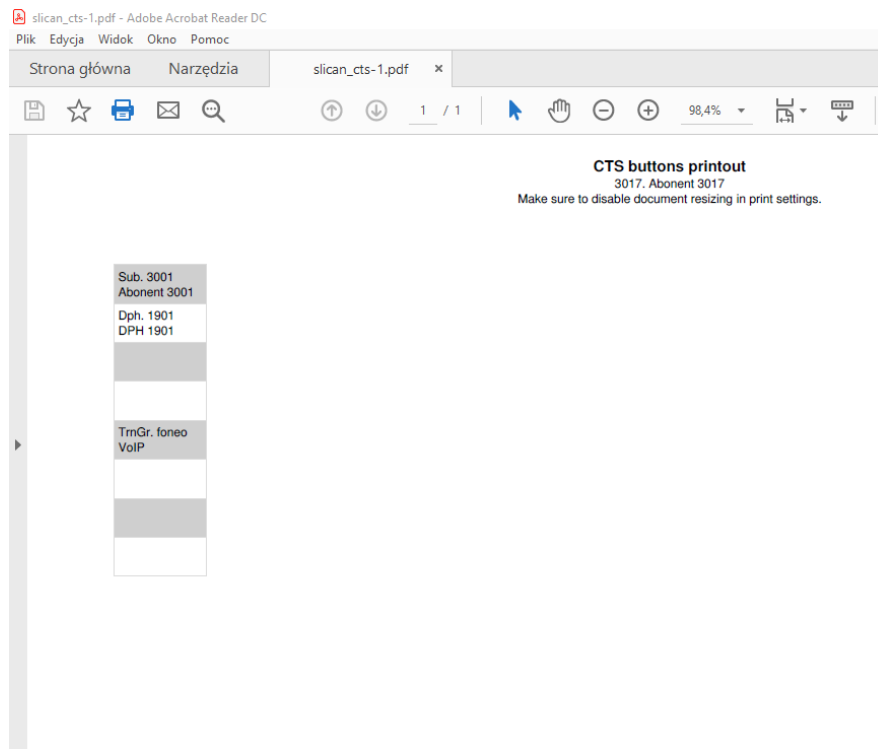


Fig 8.3: CTS etiquette print preview

8.4 System phones CTS IP configuration

Sheet **Peripherlas / CTS / IP settings** contains settings of system phones CTS-102.IP, CTS-203.IP, CTS-220.IP oraz CTS-330.IP that allow them to work with Slican PBXes.

UWAGA!

Since FW: 6.20 phones CTS-202.IP are no longer supported.

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 **Peripherals/ CTS/ IP Settings** will be able to working with PBX.

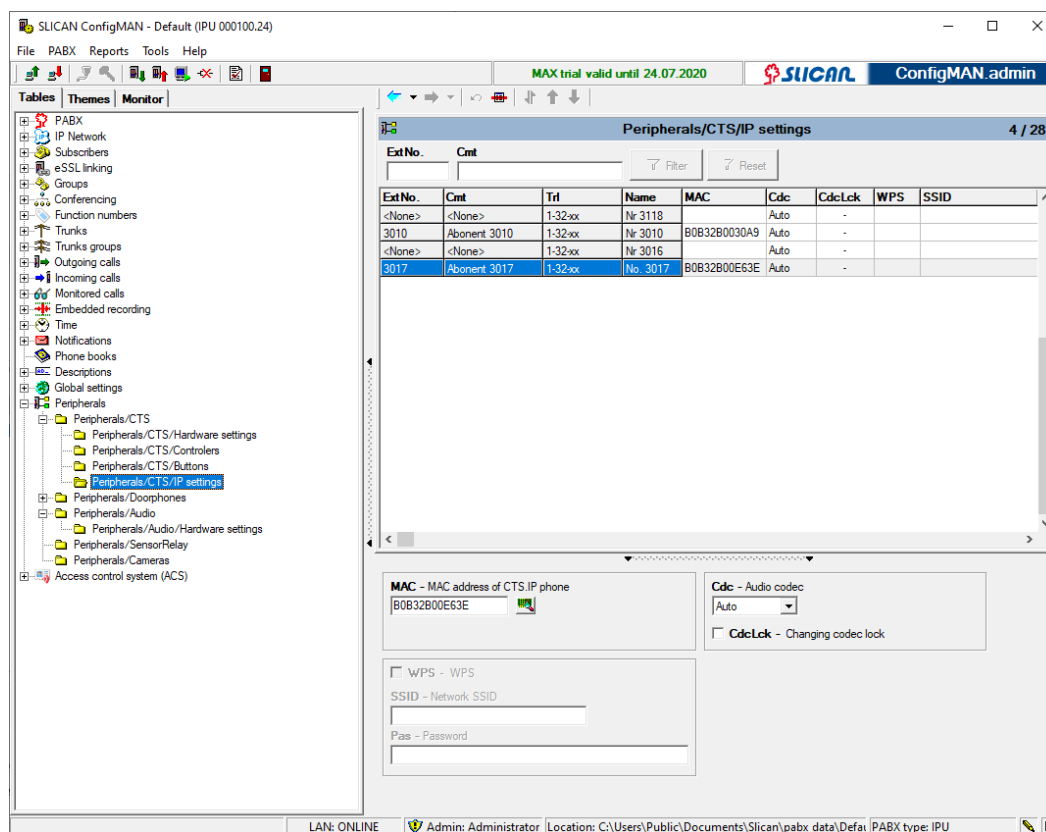


Fig 8.4: Sheet – Peripherals / CTS / IP Settings

To authorize CTS IP system phone in PBX following should be done:

- Configure phone and chose VOIP server (according to system phone manual),
- connect it to the network – properly configured phone will login to PBX,
- w programie **ConfigMAN**, jak widoczne jest na ilustracji, kliknąć w ikonę pola **MAC** - jej naciśnięcie spowoduje odszukanie i wyświetlenie listy aparatów systemowych IP oczekujących na autoryzację w centrali.

ATTENTION!

Configure VoIP server address for CTS.IP for each PBX type:

- IPL, IPM, IPU, IPS – controller address, doesn't matter if VoIP card is installed,
- MAC – VoIP card address,
- CCT – PBX address,
- CXS – VoIP card address



Fig 8.5: Window with detecting IP system 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 address can be also entered manually.

In this sheet, field **Cdc** allows to check which audio codec is used for system phone calls. Currently all CTS.IP phone use only G.711a.

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*. Każde urządzenie audio zajmuje jedno wyposażenie abonenckiego portu analogowego. Every audio device is connected to one analogue port. To configure port as audio device, in sheet **PABX/Cards** port is configured as **Audio**.

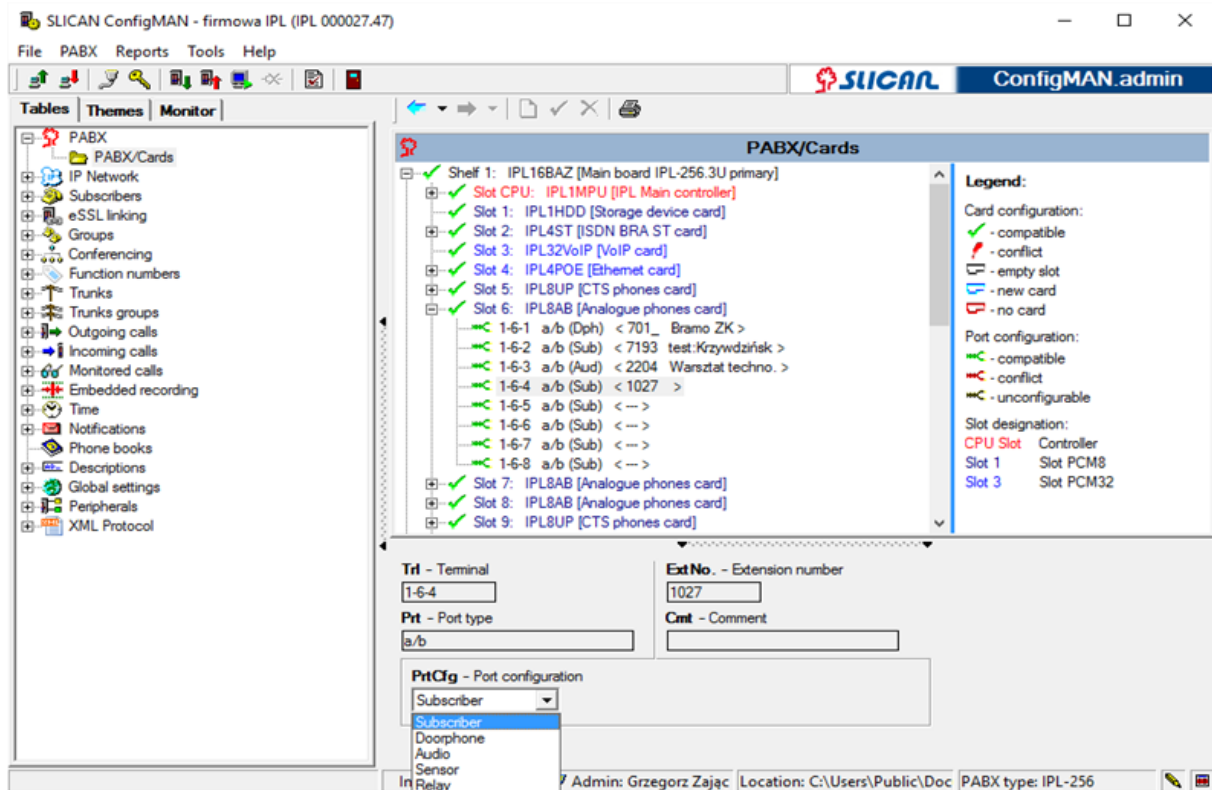


Fig 8.6: Sheet – PABX/ Cards

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

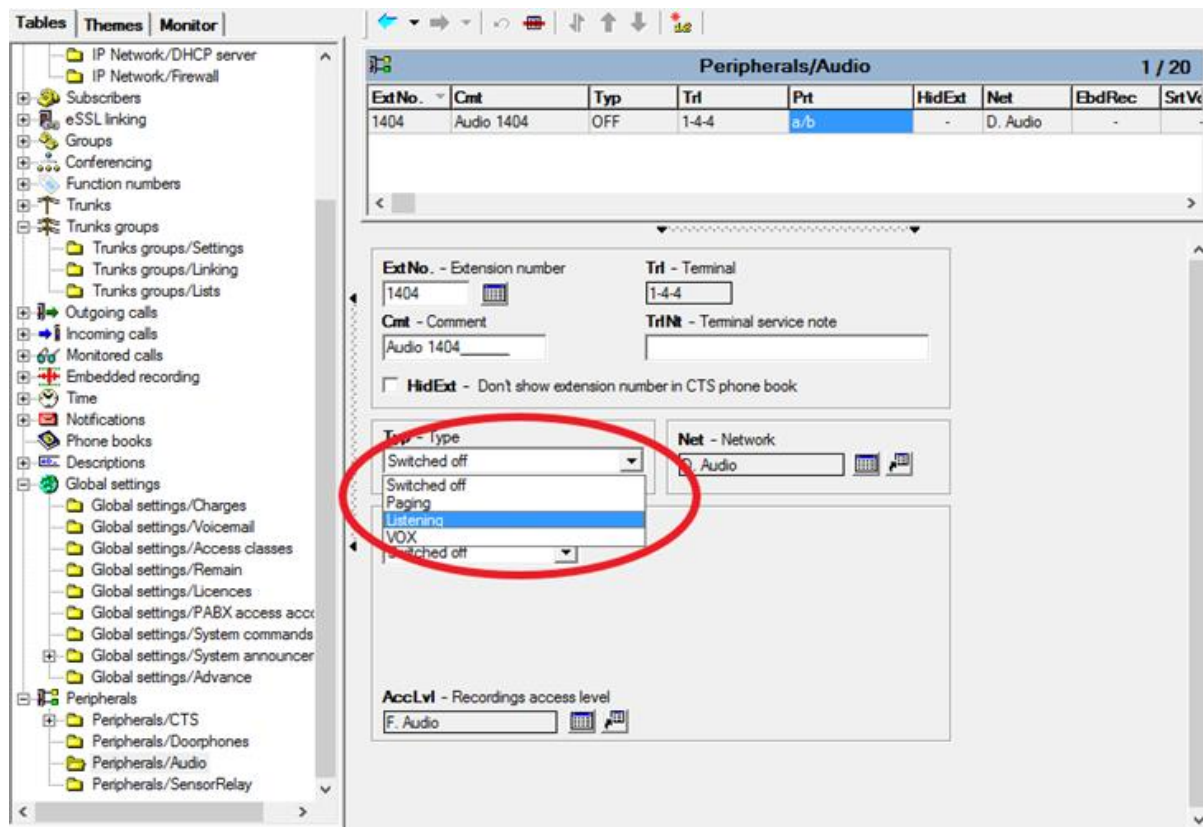


Fig 8.7: Sheet – Peripherals/ Audio

- **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**) – detailed informations about settings are available in [Global settings for PBX](#)
- **Listening**– enables only for one direction communication, ex. listening room, it can be also used for connecting external signal source, ex. CD player,
- **VOX**³ – 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 saves space on recording storage media.

ATTENTION!

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

³ Od ang. *Voice Operated eXchange*

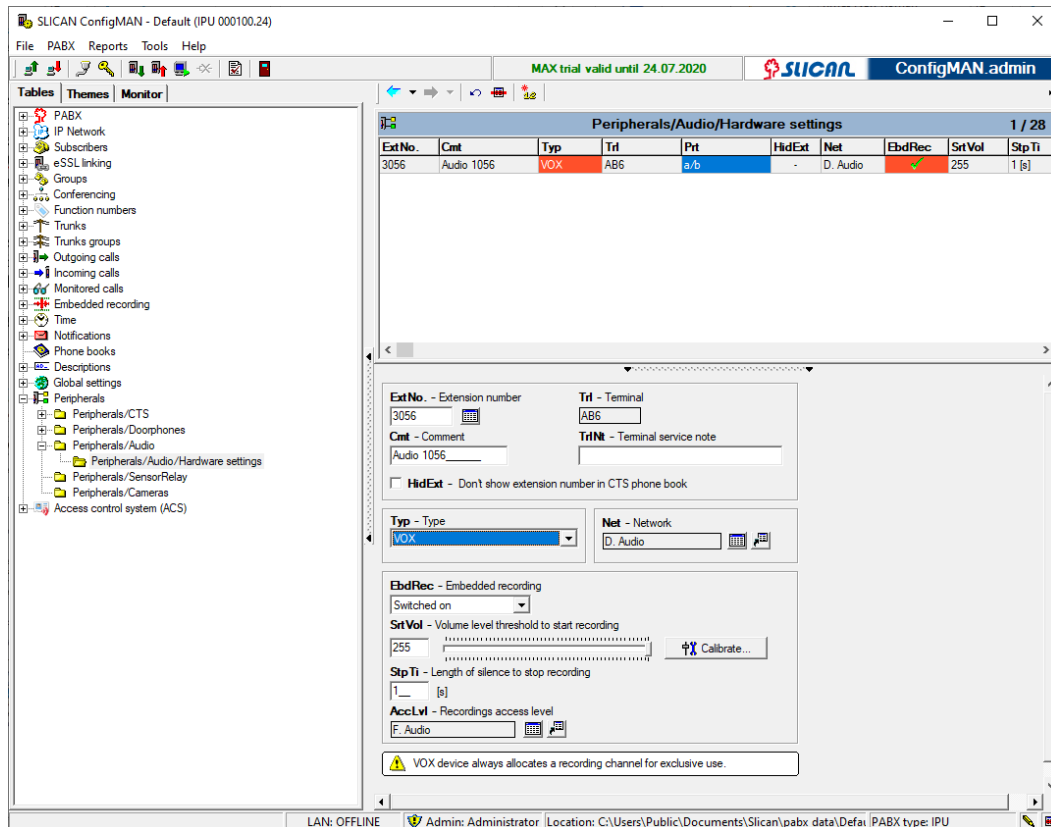


Figure 8.8: Sheet *Peripherals / Audio* for audio device of *VOX* type

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 DPH.AB

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.

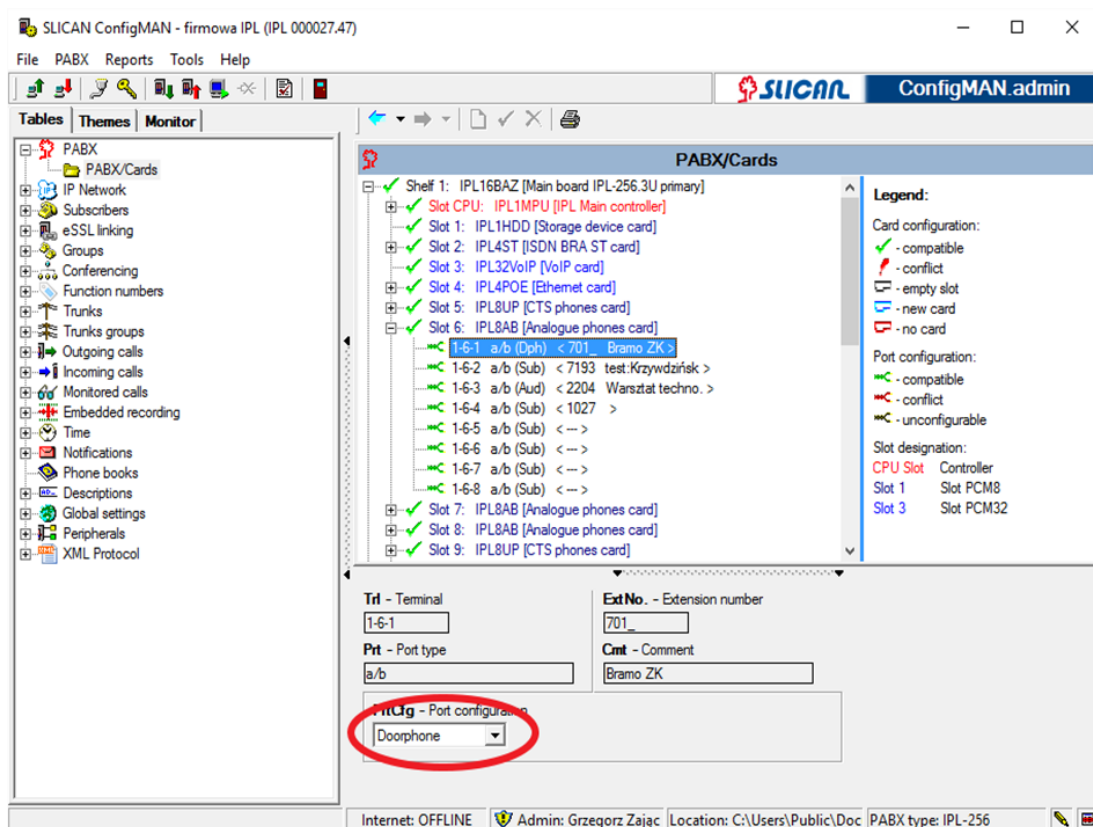


Fig 8.9: Sheet – PABX/ Cards

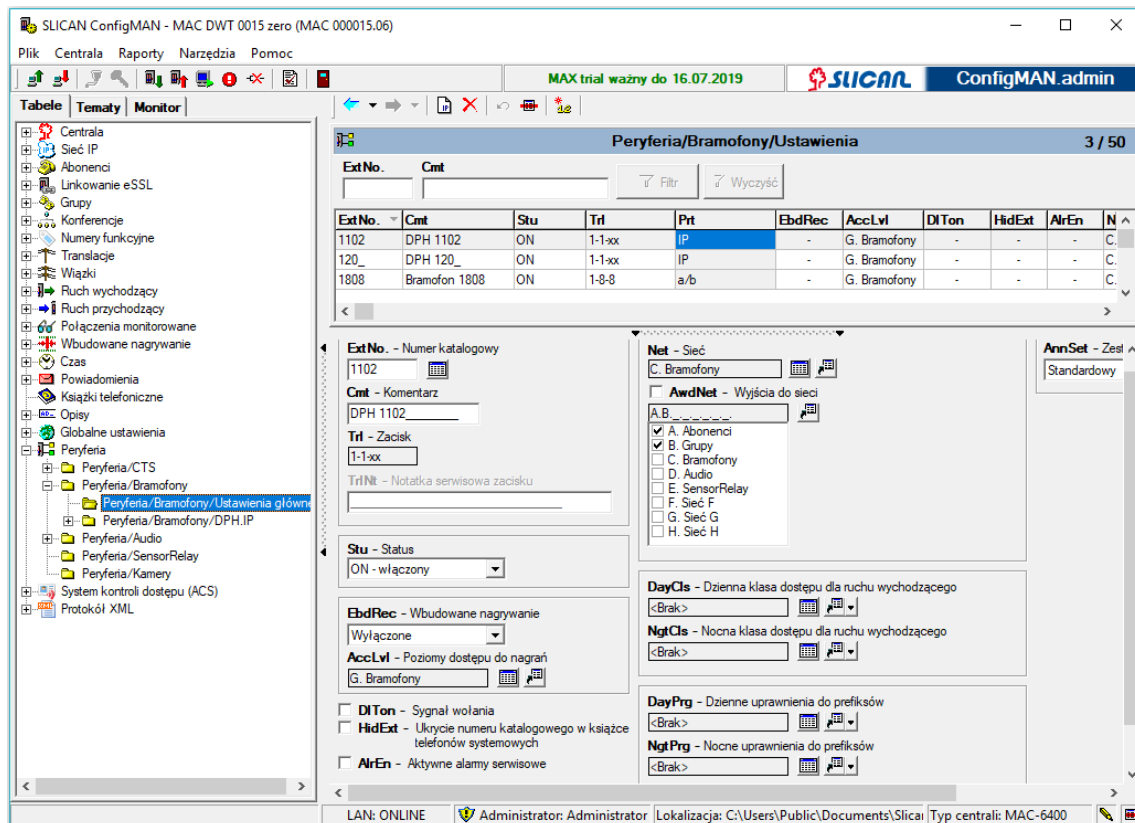


Fig 8.10: Sheet – Peripherals / Doorphones / Main Settings

For doorphone DPH.AB following settings are available:

- **ExtNo.** – enables to assign an extension number,
- **Cmt** – comment entered in this field has only information purpose,
- **Sts** – 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 [Recording 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),
- **DITon** – if this field is not checked, after dialing extension number, an announcement defined in field **MeTyp** 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,
- **HidExt** – allows to hide doorphone extension number and comment for phone book of CTS system phones and CTI applications ([MessengerCTI](#)/ [WebCTI](#)/ [ConsoleCTI](#)),
- **AlrEn** – enables service alert about subscriber port failure (*Notifications/Service Alerts*)
- **DayCls** – defines day access class for outgoing calls
- **NgtCls** – defines night access class for outgoing calls
- **DayPrg** – defines day privileges to prefixes
- **NgtPrg** – defines night privileges to prefixes

8.7 Doorphone DPH.IP

Slican PBX IPx, MAC and CCT since firmware version 6.52 allow to connect doorphones DPH.IP. First step for configuration is connecting doorphone to computer network, after that creating subscriber from PBX using *ConfigMAN* application. Device can be connected using standard patchcord, within same network as PBX. D wpinając go do tej samej podsieci, w której znajduje się centrala. Devices can also be connect from another network eg. through internet. For models DPH.IP-KS1,4,16 optional camera module is available. All events are registered in PBX and doorphones buffers, which after being downloaded with *BillingMAN* software can be analysed and used to generate reports.

ATTENTION!

DPH.IP doorphones usage requires VoIP users and VoIP channels licenses. Basic doorphone functionalities eg.: making calls, opening gate with phone, door button, using sensor to open doors – aren't licensed. Korzystanie z bramofonów DPH.IP wymaga licencji na abonentów VoIP oraz kanały VoIP.

Detailed info and configuration is available on website: [DPH.IP Doorphone](#)

Additional functionalities: using rf identification (cards and key pendants), opening with user code and PIN **requires running Access Control System (ACS).**

Doorphone working within ACS store on its memory complete configuration related to protected area. Even if contact with PBX is lost doorphone is still realising its functions: opening electrolock, letting in users of specific areas, registering events. Doorphone buffer can contain up to 2000 entries. If buffer overflow occurs, next events won't overwrite older events, that are already saved (new ones will be discarded). When doorphones is connected to PBX again, event buffer will be downloaded automatically.

If during startup doorphone won't be able to communicate with pbx (eg.: loss of power, network issues) will work without proper configuration. In that case it will realise opening gate but without controlling users permissions.

ATTENTION!

Usage of Control Access System is licensed.

8.7.1 Basic connectivity with PBX

Basic step for configuration of DPH.IP doorphone is creating extension within PBX. It is done from sheet **Peripheral/Doorphones/Main Settings** and choose **Insert DPH.IP**.

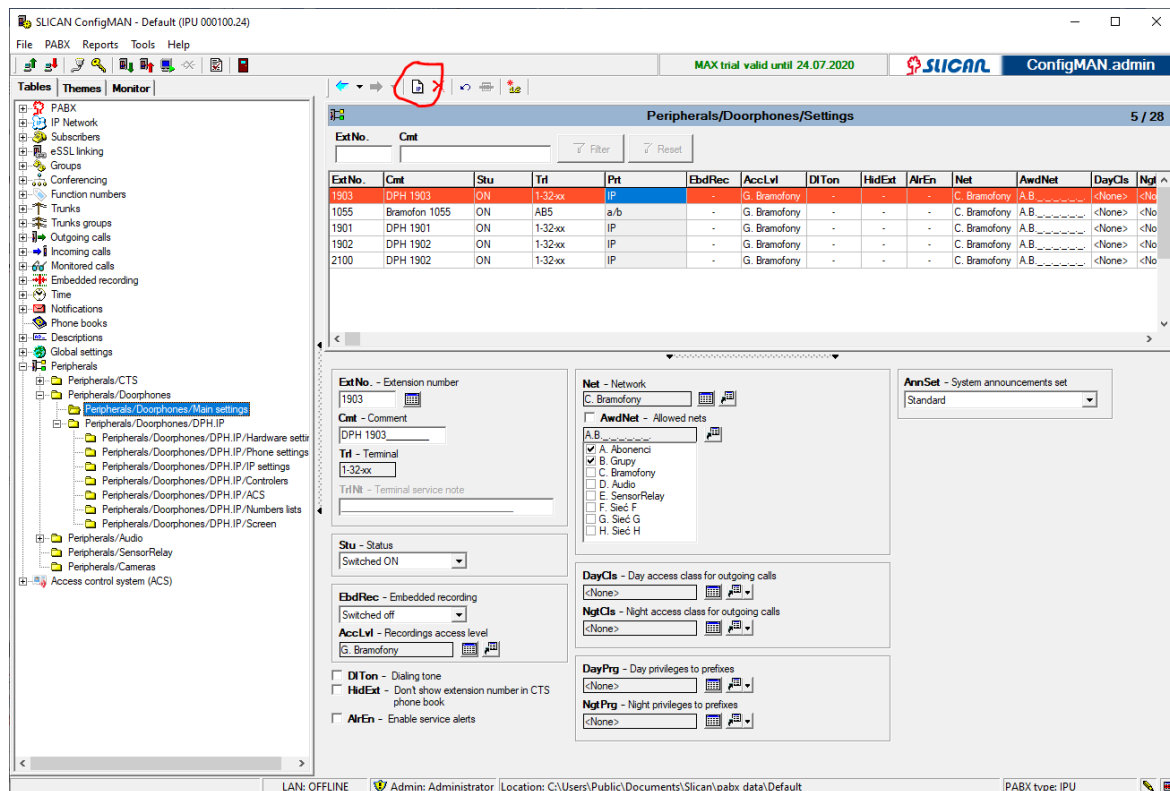


Fig 8.11: Creating DPH.IP doorphone

For DPH.IP doorphones available settings are identical to DPH.AB doorphones (described in chapter above).

DPH.IP doorphones after connecting to local network, by default will try to set up IP address from DHCP server. After that doorphone will broadcast with USRP protocol, so it can be identified by PBX. Doorphone login to PBX is familiar to CTS.IP system phone. From **ConfigMAN** software in sheet **Peripheral/Doorphones/DPH.IP/IP settings** we search for proper MAC address of the device or type it by hand. After sending configuration to PBX and correct doorphone login, in appropriate sheets configuration options will be activated.

IP address, network mask, gateway and DNS server address of all logged doorphones can be configured from **ConfigMAN** application.. Searching for doorphone MAC address and login to PBX will save configuration. It should be noted that configuration errors (incorrect doorphone IP address within network) can cause lack of connectivity and require to repeat device configuration.

KS16 doorphone can have configured it's IP address and VoIP server (PBX) address to connect by hand. Configuration details are in [device manual](#).

All IP doorphones models can log in to PBX from other networks (and from internet). In this case configuration must be done with application [Slican Device Config](#)

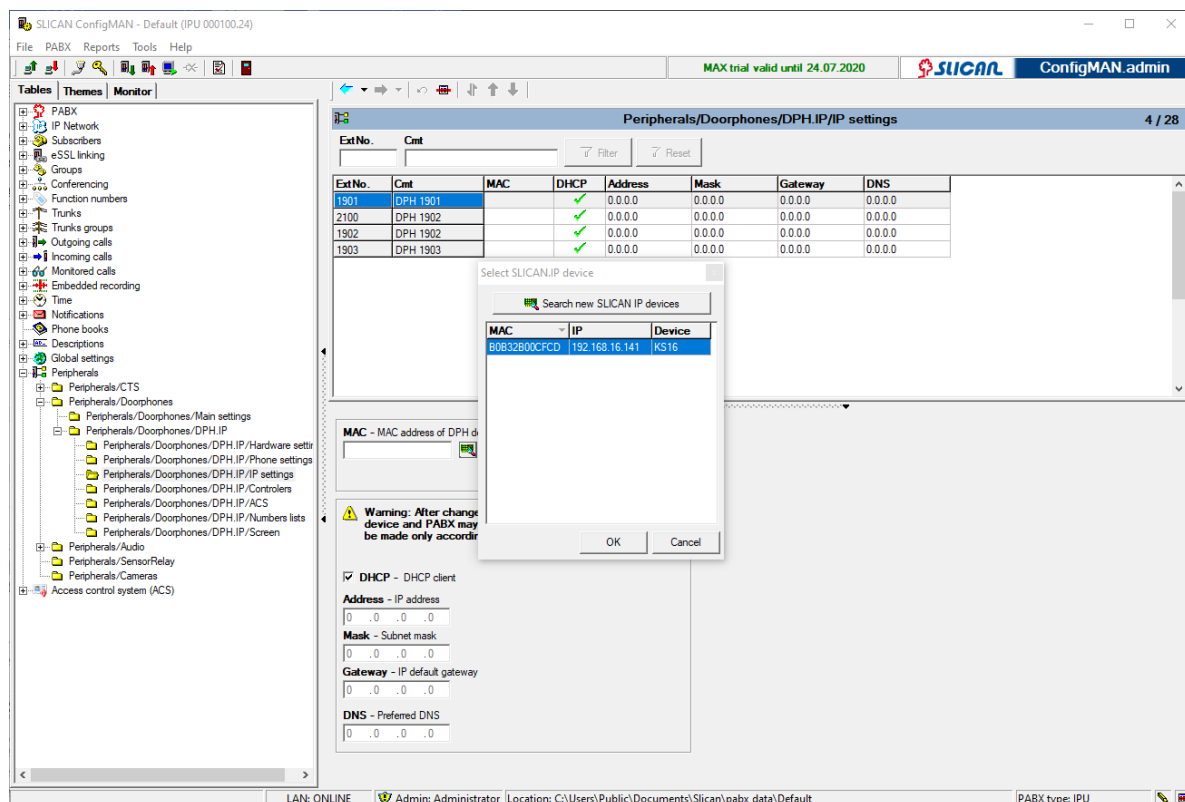


Fig 8.12: Searching for DPH.IP doorphone

8.7.1.1 Doorphone settings configuration

Doorphone configuration is possible with use of SETUP jumper on it's circuit board pins. To avoid need of using jumper each time, within **ConfigMAN**, within sheet **Peripherals/Doorphones/DPH.IP** set up in field **PIN** four digit code. This allows for each use of doorphone configuration menu to press twice „key” button, after that enter PIN code.

After doorphone connection, configuration fields are available.

ATTENTION!

Sending configuration to doorphone takes up to 3 seconds. This should be noted during sending configuration to big amount of doorphones. In case of maximum amount of devices within system, sending configuration takes longer time.

Firmware bramofonów DPH.IP doorphones firmware is stored within PBX. Firmware update is automatic, without technician interference..

8.7.2 Hardware settings

In sheet **Peripherals/Doorphones /DPH.IP/Hardware settings** options can be configured, related to displaying information, opening electrolock, STA relay and sensor or button that open doors.

If doorphones work without Access Control System (ACS), then opening doors, gate(EZ electrolock and/or STA relay) is possible by person that enters or exit with open door buton

POD, by any PBX subscriber using internal phone, MobilePhone or Slican MessengerCTI application (opening doors isn't restricted and can be done by any subscriber).

Available configuration settings:

- **DevTyp** – allows to choose type of connected device, activating proper configuration options:
 - **KS1** – one button doorphone without display,
 - **KS4** – four button doorphone with LCD display,
 - **KS16** – sixteen button doorphone with LCD display,
 - **CR0** – device without display, with integrated RF ID reader,
 - **UB2** – doorphone circuit board that can be mounted inside housing or casing of other manufacturers.
- **Lng** – language of doorphone interface - polish, english or italian,
- **Scr** – allows to set up custom text on display „CHOOSE NUMBER”, custom text are configured in sheet opisy własne są konfigurowane w zakładce **Peripherals / Doorphones /DPH.IP/Screen**, option is available for devices with LCD screen – KS4 i KS16,
- **Hid** – hides/shows text „search” on screen (under screen description), option available for devices with LCD screen,
- **NoLst** – allows to use predefined number list, localised in sheet **Peripherals/Doorphones /DPH.IP/Numbers lists**. W ON doorphones using arrows, numbers can be scrolled only up and down. Available only for doorphones KS4 and KS16,
- **AnyNo.** – allows to choose any number from keyboard. Turning off this fields allows optionally allows to choose pre-configured number list from field **NoLst**. Available only for doorphone KS16,
- **DINo.** (only for KS1 and UB2) – number that will be dialed after pressing device button.
- **SBtMd** (only for UB2) – allows to choose action, that will be initiated by device after pressing second button:
 - **Generate sabotage** – cause to generate alarm within PBX, that can be sent on target e-mail address or SMS to target number. Event is visible in CDR protocol. Configuring notifications about events is done from sheet **Notifications/Events**.
 - **Dial number** – allows to assign number, which will be dialed after pressing second button.
- **SDINo.** (only for UB2) – number dialed after pressing second button. This can be any internal number, external if proper permissions are set in sheet **Peripherals/Doorphones/Main settings**.
- **ELck** – allows to choose how EZ relay works:
 - **Switched off** – relay won't be triggered,
 - **Normally inactive** – doorphone while providing voltage will trigger electrolock, allowing to open doors,
 - **Normally active** – doorphone will constantly provide voltage, disrupting it will trigger electrolock, allowing to open doors.
- **OTi** – depending on EZ output mode, set up time of voltage, how long it will be provided (normalny inactive) or stopped (normalny active)

- **BTi** – sets up for how long doorphone will signal with audio about triggering EZ relay,
- **Rel** – allows to set up work mode of STA relay
 - **Switched off** – relay won't be triggered
 - **Normally open** – in passive state relay is open, while triggered – closed
 - **Normally closed** – in passive state relay is closed, while triggered – open
 - **Light switch** – triggering relay happen after pressing any button, dialing doorphone or during call,
- **PlsDrn** – time of triggering STA relay (depending on work mode – open or close)
- **COD** – activate/deactive (COD) Doors Open Sensor
- **NrmCOD** – set up COD passive state– normally open or normally closed
- **CODTi** – alarm of doors that are open to long. Set up for how long COD must be active to generate notification about event. Setting as 0 will turn off registering events. Events are saved in CDR protocol, register in events biffer and can be sent as SMS or e-mail (sheet Notifications/Events). If CTS phone have doorphone assigned under programmable button then information about opening doors to long will be signaled with blinking button.
- **CODRgs** – set up reaction in case of breaching COD by opening doors without doorphone:
 - As event – in case of mechanical door opening eg. with doorhandle from other side of doors.
 - As alarm – generates alarm within PBX in case of nonautorised opening/closing doors detected by COD – opening by force eg. breaking through doors(sheet *Notifications/Service alerts ACS* field)
- **POD** – field allowing to enable or disable Doors Closing Button (POD). It can work in two modes – as normal button or override button. In second case – using button will dial number and wait for subscriber to pick up - relasing will end dialing or calling.
- **NrmPOD** – set up passive POD state – normally open or normally closed.
- **PODRgs** – checking this field will cause to register action of using POD within CDR protocol and register in events bufffer..

8.7.3 Phone settings

Doorphone configuration is related to phone calls, made by DPH.IP doorphone or action during incoming calls triggered by phone.

- **TiNo.** – time after which doorphone will automatically dial phone number chosen with keyboard,
- **WTi** – set set up maximal waiting time during outgoing calls,
- **CTi** – set up maximal time of doorphone call.
- **Act** – default action done by DPH.IP:
 1. During call from or to doorphone (when voice can be heard)) for chosen target action:
 - „Open electric door lock EZ”, „Turn on relay STA” – choosing "*" asterisk during DTMF in internal phone (CTS, SIP, analog) or cellphone (MobilePhone,

- application **MessengerCTI.Mobile**) will cause: opening EZ or triggering STA, or simultaneous control of EZ and STA,
- „Allow to choose” – choosing digit 1, 2 or 3 during DTMF in internal phone (CTS, SIP, analog) or (MobilePhone, application **MessengerCTI.Mobile**) will cause: 1 - opening EZ, 2 - triggering STA, 3 - controlling EZ and STA,
- 2. During call to or from doorphone or without cost (call wasn't picked up) – with programmed BLF button of CTS.IP or with „door” icon of application **MessengerCTI.Desktop**. For chosen options:
 - „Open electric door lock EZ”, „Turn on relay STA” opening EZ or triggering STA, or simultaneous control of EZ and STA
 - „Allow to choose” – functionality isn't supported

Options „Open electric door lock EZ”, „Turn on relay STA” and „Allow to choose” can be enabled individually or few together as set.

It should be noted that there is higher priority over telephone setting by Control Schedule (sheet **Time/Control schedule**):

- If control schedule contains doorphone which doesn't have assigned actions targets (EZ, STA), then as default will be used targets (EZ and/or STA) from current sheet.
- If control schedule contain doorphones which have assigned targets of actions (EZ, STA), they will be used first as default. After that those from current sheet will be used.
- **AutAct** - from this field we can choose action for DPH.IP, done during incoming call tym polu wybieramy akcję, jaką ma wykonać bramofon DPH.IP w momencie połączenia przychodzącego:
 - Acoustic on – incoming calls will be heard instantly through speaker, without ringing up,
 - Calling – will ring up device, it is possible to reject or pick up call with doorphone keyboard,
 - Listening – call will be picked up automatically with turned on speaker. It is possible to listen to room, without signaling on display.
 - Reject – incoming calls will be automatically rejected.

8.7.4 Contrast and acoustic settings

Settings regarding acoustic and signal for doorphones can be edited from sheet **Peripherals/Doorphones/DPH.IP/Controlers**.

- **Cnt** – allows to change contrast of displayed information
- **Mic** – changes doorphone microphone sensitivity
- **LspVol** – changes loudspeaker volume
- **LspLmt** – set up level limit, switching limit after which speaker will stop working
- **SysVol** – set up sounds level of system sounds and sounds made after pressing any keyboard button
- **RngTon** – set up one of five available ringing tones for incoming calls

- **RngVol** – set up ringing volume for incoming calls

8.7.5 Doorphone actions for Access Control System (ACS)

Sheet **Peripherals/Doorphones/DPH.IP/ACS** allows to enable Access Control System (ACS) on doorphones. Allows to identify users and enable or disable access to specific zones, within set time frames. Turning it on also allows extra functionalities.

If doorphone is working with Access Control System (ACS) enabled, it must have assigned zones. Opening gate, Gateway, doors (with EZ electro lock and/or STA relay) is possible by person (that enters or exits) with door opening button (POD), rf card, entering extension number and PIN on keypad, by PBX subscriber with permission using internal phone, MobilePhone, Slican application [MessengerCTI.Desktop](#) or **MessengerCTI.Mobile** of PBX subscriber assigned to same zone (opening is restricted by permissions).

System is enabled with field **EnACS**.

ATTENTION!

Usage of Access Control System is licensed. License is generated for amount of ACS users. For linked PBXes license of that PBX counts for it's local users

Configuration options:

- **AZn** – allows to choose doorphone access zone. Zone is defined as restricted area, which can be entered through one or more doorphones. Eg.: corridor with two exits, secured with doorphones is zone, both doorphones are in same zone.,
- **CrdAct** – set up action after sensing registered rf card:
 - Open electric door lock EZ and/or Turn on relay STA – depending on configuration it will cause EZ electrolock and STA relay to trigger together or separately EZ or STA,
 - Allow to choose – triggering with digit: 1 – electro lock EZ, 2 – relay STA, 3 - EZ and STA. Sequence for triggering: on DPH.IP keypad choose digit 1, 2 or 3 and after that use rf card.

Options Open electro lock EZ, Turn on relay STA or Allow to choose can be turned on separately or joined together as set.

- **PINAct** – set up action after entering subscriber extension number and assigned Pin code (from sheet **Subscribers/Services** field **PIN**):
 - Open electric door lock EZ and/or Turn on relay STA – depending on configuration it will cause EZ electrolock and STA relay to trigger together or separately EZ or STA,
 - Allow to choose – triggering with digit: 1 - electro lock EZ, 2 – relay STA, 3 - EZ and STA. Sequence for triggering: on DPH.IP keypad – choose subscriber extension number, next press 2x key button, digit 1, 2 or 3, finish with subscriber PIN.

Options Open electro lock EZ, Turn on relay STA or Allow to can be turned on separately or joined together as set.

- **PODAct** – set up action after triggering POD(open door button):

- Open electric door lock EZ and/or Turn on relay STA – depending on configuration it will cause EZ electrolock and STA relay to trigger together or separately EZ or STA,

Options Open electro lock EZ, Turn on relay STA or can be turned on separately or joined together as set.

- **PODNo.** – ustawienie numeru, który zostanie wybrany po zadziałaniu POD

8.7.6 Short numbers from numbers lists

Sheet **Peripherals/Doorphones/DPH.IP/Numbers lists** allows to create and edit separate entries which can be used as number lists available to choose from display of DPH.IP KS4 i KS16 doorphones (settings from field **NoLst** in sheet **.../DPH.IP/Hardware settings**). Those number can be :internal, eSSL subscribers, groups, function or external. They will be analysed just like other numbers during phone calls.

8.8 Cameras

Since fw. 6.52 option of integrating specific extensions with IP cameras was introduced. DPH.IP are special device with built in camera. PBX can also work with some cameras of another vendors.

To display video from cameras, application Slican [MessengerCTI.Desktop](#) was developed. Calls realised to or from extension assigned to camera will cause on user computer that is part of call, to show window on PC with video preview. Subscribers doorphones, groups and other devices eg. controllers – sensor, relays can be those extensions.

Sheet **Peripherals/Cameras** allows to assign few cameras to single extensions, or few extensions to single camera.

Configuration settings:

- **Cmt** – comment
- **ExtNo.** – extension number assigned to camera– subscribers, eSSL subscribers, doorphone, sensor, relay, group, audio device.
- **Lnk** – link to doorphone or doorphone in format, eg: rtsp://192.168.155.20:554/user=admin&password=&channel=1&stream=0.sdp?

Searching for camera within network, access to configuration and video links for DPH.IP doorphone camera is available with application [Slican Device Config](#) (available from ServNET server). Cameras by default use DHCP IP address but for proper operation we recommend to set up static IP address or DHCP reservation on server for camera MAC address. This secures system against errors due to camera links being no longer correct.

Cameras of other vendors must be compatible with libraries (video formats, codecs) of vlc applications. We don't guarantee proper operation of application [Slican Device Config](#) with those devices. In case of issues, use software of that vendor.

9 Access Control System (ACS)

With release of software 6.52, Slican PBX with DPH.IP doorphones can use advanced Access Control System. With proper configuration of devices and users permissions to selected zones, we can control enter and exit actions. Full possibilities are available while using cameras recording video from important areas and application for video preview - [MessengerCTI.Desktop](#).

To run Access Control System few steps must be taken first:

- Create and configure at least one doorphone which will control access to restricted zones
- Configure Access Control System (ACS) – create users, assign them to chosen groups, create zones with access for specific user groups, create rules allowing chosen user to access during specific day time, to specific areas with different methods
- Install and configure [BillingMAN](#) application to prepare reports of enter, exit events
- Optional – configure and assign camera to specific extensions

User of Access Control system can open barriers (doors, gates) with one of few methods:

- Rf cards,
- Using subscriber own number and PIN code (extension number – subscriber, PIN - field **PIN** sheet **Subscribers/Services**),
- Based on MobilePhone number during call,
- By person that picks up call from doorphone:
 - Using phone, during phone call with DTMF code chosen on phone keypad when voice can be heard (*,1,2) z telefonu, podczas rozmowy, za pomocą kodu DTMF wybranego z klawiatury na akustyce (*, 1, 2)
 - Using CTS phone with programmed button (option of opening outside of call eg.: when entering person is visible on camera)
 - With door icon while using application [MessengerCTI.Desktop](#) and **MessengerCTI.Mobile** (option of opening outside of call eg.: when entering person is visible on camera)

9.1 Assigning RF cards to users

ATTENTION!

Maximum number of accounts (also subscribers) using RF cards is 1000.

Assigning RF identifier to user can be done in two ways. It is done in sheet **Access Control System (ACS)/Subscribers**:

- Knowing ID of RF card, number can be typed in field **CrdID** and/or **AddCrdID**. Maximum amount of cards must be set up in field **MaxCrd**.
- If we don't know ID of RF card, doorphones DPH.IP-KS16 can use mechanism of self registering card. Depending on chosen method of adding RF cards in field **RgsPrm**

user can add cards to his extension number (sequence: key button + extension number + subscriber PIN + use RF card 1 + optional: use RF card 2).

To quickly get RF cards numbers, unregistered cards can be used with doorphone and their numbers can be read from **System console** and **cdr** command. For this method of reading RF cards numbers to work, PBX must have embedded recording active.

9.2 Access control system configuration

Access control system based on Slican doorphones and PBXes allows to create advanced access rules for user, used devices and zones.

ATTENTION!

Amount of subscribers/user which have permissions to specific zones and can use doorphone with Pin code/RF card/Mobile Phone is depending on license..

Access Control System (ACS)/Subscribers – in this sheet system users can be added, which are subscribers and virtual accounts of PBX (subscribers logging on eg.: shared phone), which can use Access Control System. For single PBX maximum amount of users depends on its capacity and can't be higher than sum of subscribers and users created in sheet **Subscribers/Main Settings**. For linked PBXes this is sum of users of all PBXes.

Current number of users is visible in right, upper part of window.

Access Control System (ACS)/Groups – to simplify managing ACS rules, access groups can be created. They are used to group chosen subscribers and accounts (eg.: employers of specific departments), which have access to same zones, at same time, with same method (RF card, PIN code or their MobilePhone number).

Each subscriber/user, within ACS must be assigned to at least one group. System can have up to 64 groups.

Within linked system – ACS groups are created only on PBX that is acting as eSSL server. Groups are broadcast automatically and are sent to all client-PBXes (scheme of access group must be the same for whole system). For created and broadcasted groups within all PBX with turned on ACS, can add their local users.

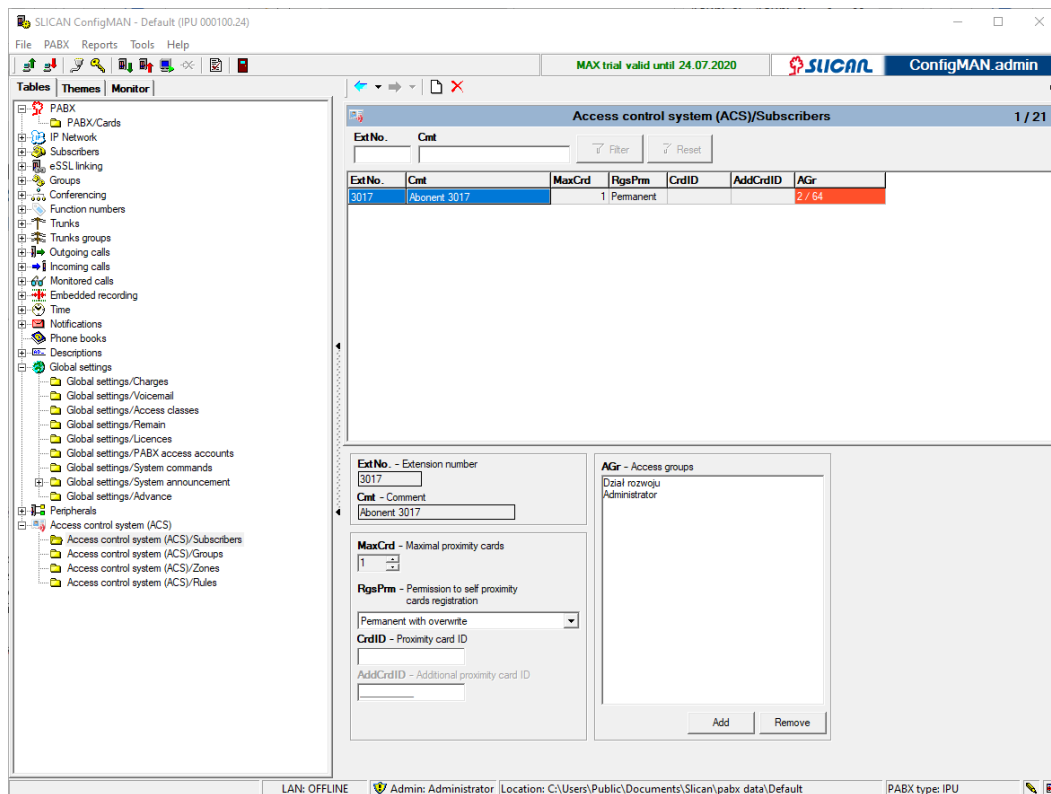


Fig 9.1: Assigning subscriber to access group

Access Control System (ACS)/Zones – within PBX create access zones, which are restricted areas, which can be entered by using one or more doorphones. In this sheet we set up zone names. They are assigned to doorphone in field **Azn** sheet **Peripherals/Doorphones/DPH.IP/ACS**.

Doorphone can secure up to maximum of 64 zones. For linked system (eSSL) zones are created in each PBX separately, because they are assigned for DPH.IP doorphones that were created and logged locally.

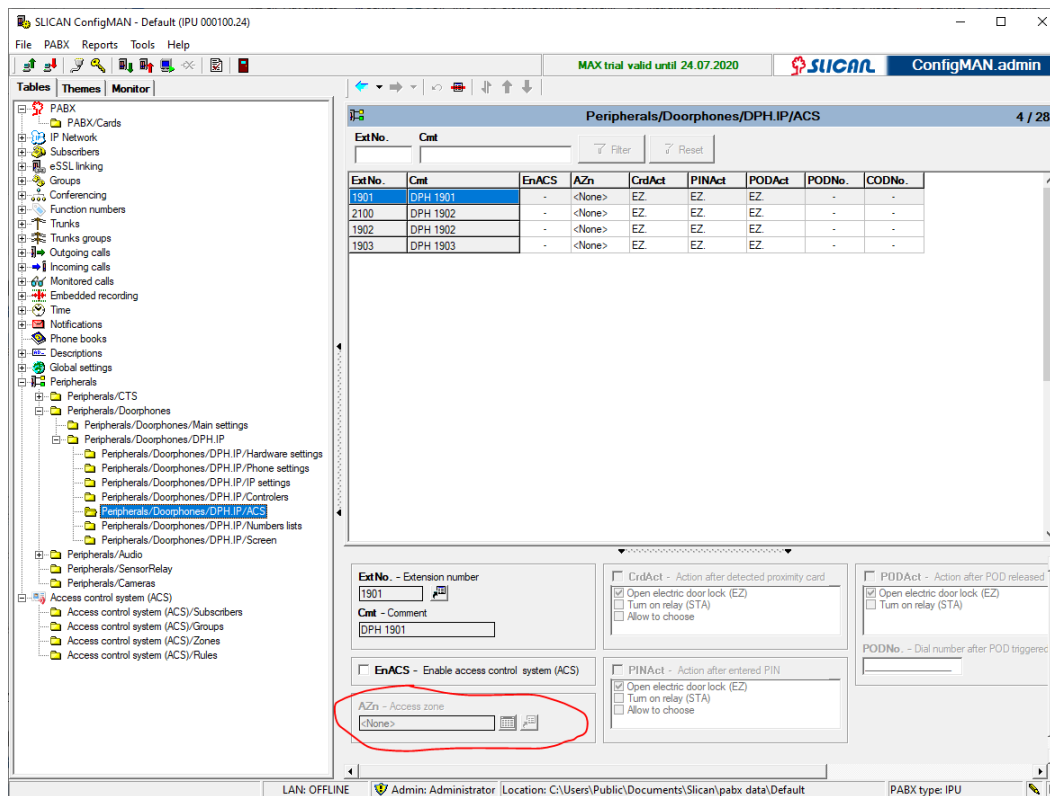


Fig9.2: Assigning zone to doorphone

W zakładce **System kontroli dostępu (ACS)/Reguły** tworzymy wpisy określające, do jakich stref, w określone dni oraz godziny, posiadają dostęp wybrane grupy użytkowników ACS.

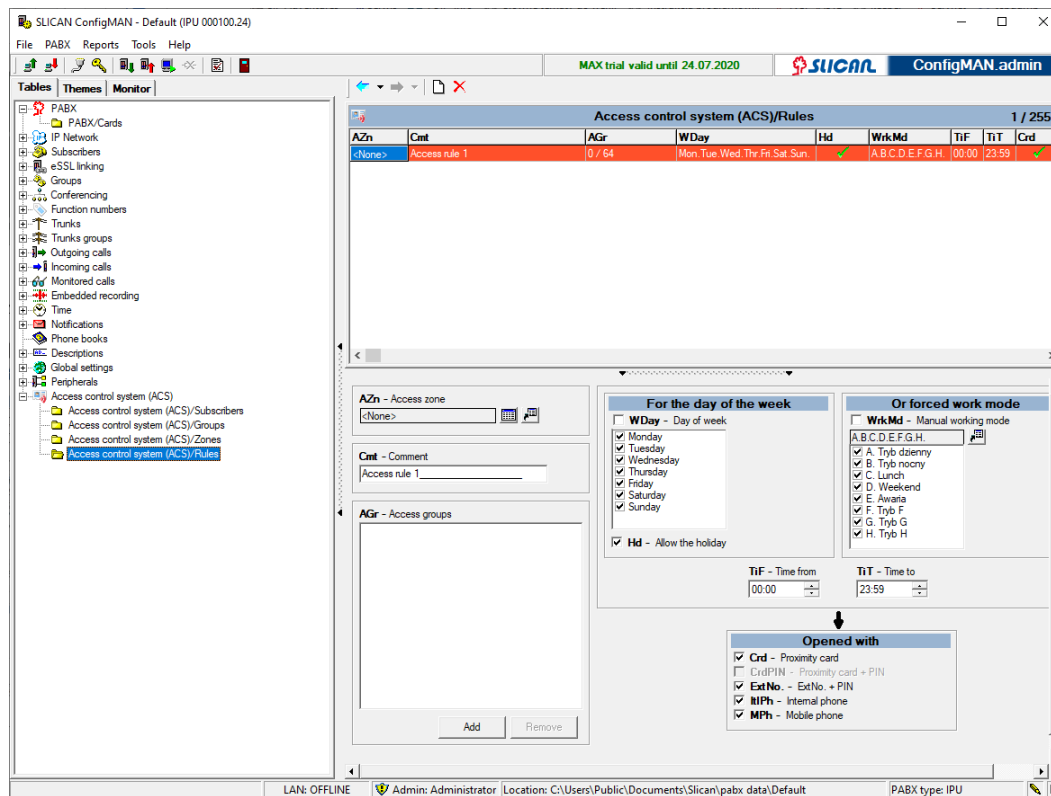


Fig 9.3: Creating access rules

To create access rule, those fields must be set:

- **WDay** – allows to set up rules according to day of the week (allows to divide rules into specific days of the week)
- **TiF** and **TiT** – hours for which rules is active
- **WrkMd** – PBX work modes during which rule is active

After setting up which ACS group have access to which zones during specific time or PBX work mode, we set up method of trigger doorphone:

- **Crd** – with RF card
- **ExtNo.** – with doorphone keypad – enter your own extension number, after that press key button and confirm with PIN code
- **IdPh** – during phone call with PBX subscriber (with permissions) (phone, CTS, VoIP, analog phone, *MessengerCTI.Desktop*), that is part of proper access group, subscriber can use 1,2 or * buttons, open with *MessengerCTI.Desktop*, *MessengerCTI.Mobile* icon or CTS.IP programmable button
- **MPh** – charge fee doorphone opening by gsm – call from MobilePhone or application *MessengerCTI.Mobile* to properly routed PBX external number,

ATTENTION!

Each subscriber/account, which can controll with actions, must be configured according to above steps

PBX subscribers that picks up doorphone call transferet to another subscriber, mast have proper permissions during opening barrier, else he won't be able to trigger any action.

10 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 [Keyboard shortcuts](#)).

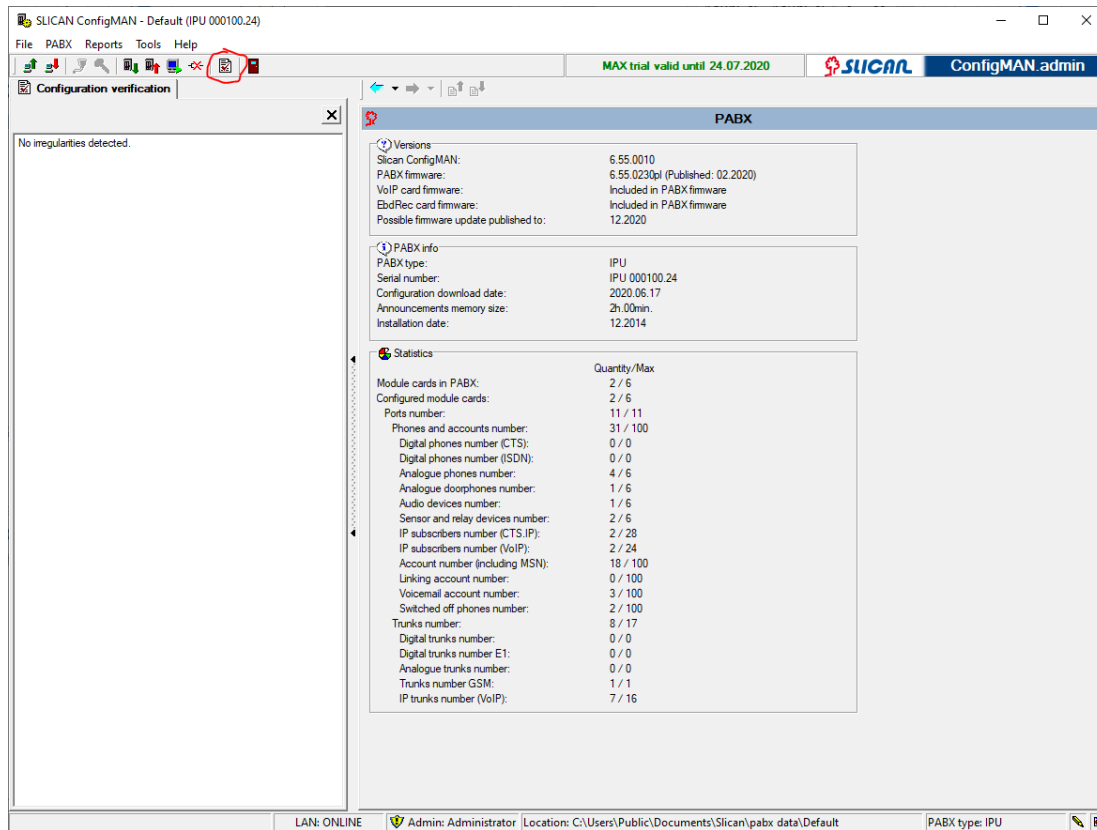


Fig 10.1: Configuration verification

If configurations errors are detected then **ConfigMAN** will display issues in this sheet *Configuration Verification*. Below examples shows that number wasn't used within trunk..

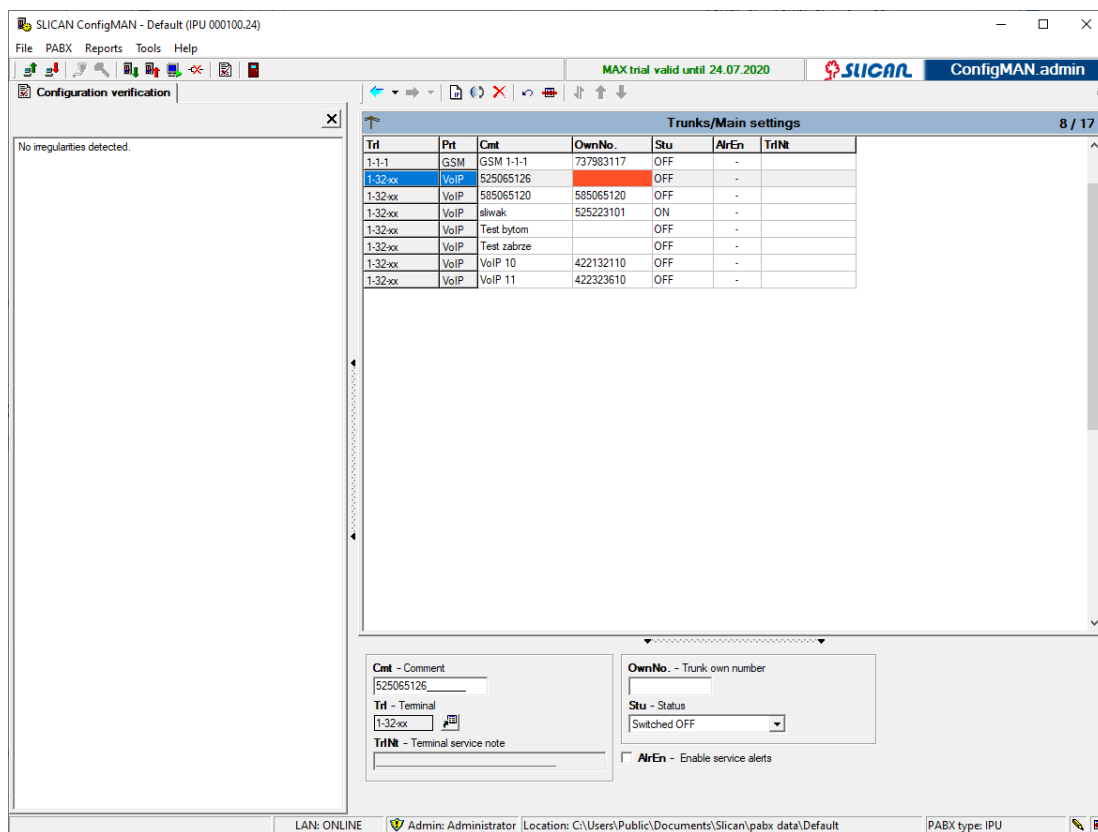


Fig 10.2: Showing lack of trunk own number

11 PBX reset

11.1 PBX reset

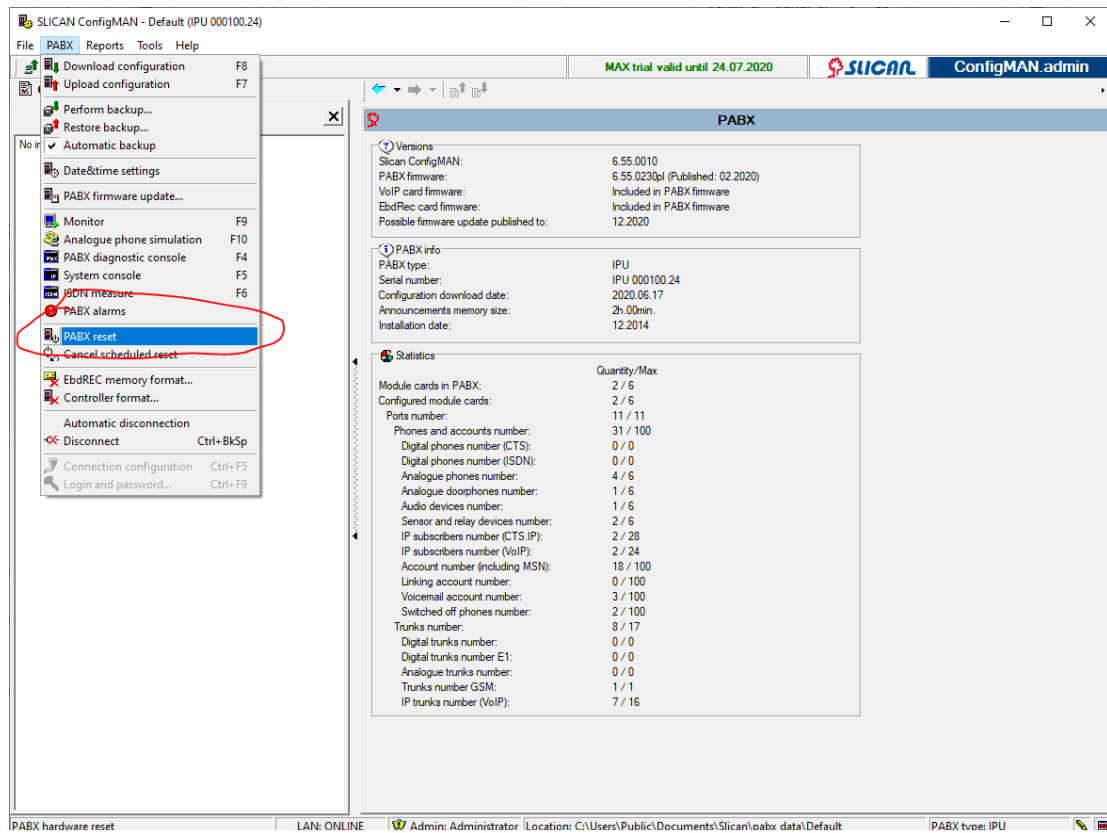


Fig 11.1: PABX reset

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 immediatly* – stops work, cancel all current and parked calls.

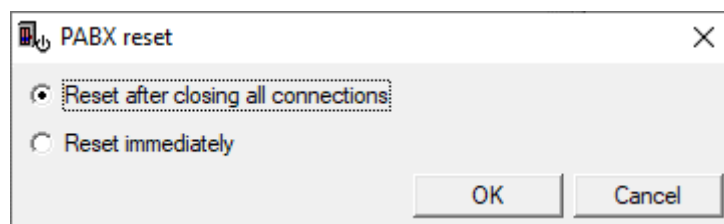


Fig 11.2: Reset mode selecting window

ATTENTION!

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

12 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.

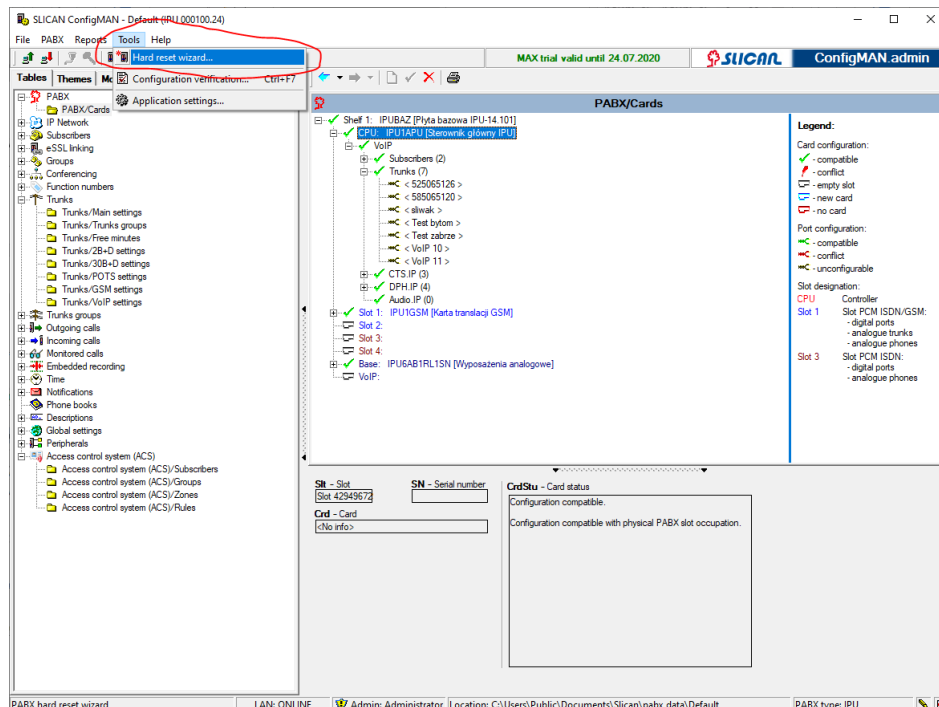


Fig 12.1: Basic PBX configuration wizard

After starting the **Hard reset wizard** the following message will be displayed:

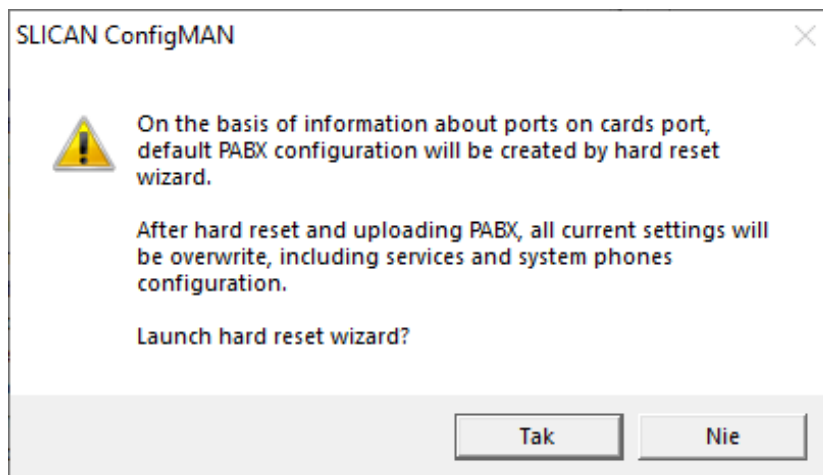


Fig 12.2: PABX hard reset wizard message

Next, the system will request to choose how it will be named, confirm international prefix number and specific configuration depending on detected external trunk ports cards. First are ISDN trunks (if installed) – set up their work mode: PP – point – point or PMP – point – multipoint, after that trunk main number. (in case of PP – amount of DDI numbers).

If the system detects analogue POTS or GSM trunks, it will request their own numbers. Additionally for GSM trunks SIM card PIN number. Also it is important to provide the correct area code, for the PBX's location. Described above steps are presented on following figures. Described above steps are presented on following figures.

PABX hard reset

Hard reset wizard will prepare default configuration for installed module cards and ports.

Select PP or PMP configuration for ISDN trunks and enter trunks own numbers.

PP trunks

Trl	Prt
1-10-1	E1

DDI own number

Quantity of DDI numbers:

PMP trunks

Trl	Prt	MSN own number
1-9-1	S/T	
1-9-2	S/T	
1-9-3	S/T	
1-9-4	S/T	

Warning:
Own number should be in national format, ex.: 523251111.

Fig 12.3: Hard reset wizard – ISDN trunks

PABX hard reset

Hard reset wizard will prepare default configuration for installed module cards and ports.

Enter POTS trunks own numbers.

Trl	POTS own number
1-6-1	
1-6-2	
1-6-3	
1-6-4	
1-6-5	
1-6-6	
1-6-7	
1-6-8	

Warning:
Own number should be in national format, ex.: 523251111.

Fig 12.4: Hard reset wizard – POTS trunks

PABX hard reset

Hard reset wizard will prepare default configuration for installed module cards and ports.

Enter numbers assigned by GSM provider or PIN number required for SIM card authorization.

Trl	GSM own number	PIN
1-7-1		
1-7-2		
1-7-3		
1-7-4		

Warning: Own number should be in national format, ex.: 523251111.

Cancel <<Back Next>>

Ilustracja 12.5: Hard reset wizard – GSM trunks

ATTENTION!

Hard reset wizards allows to add only VoIP Foneo Sip trunk. Its own number and password should be assigned. Z poziomu kreatora twardego resetu dodajemy tylko translację VoIP Foneo. Other trunks, depending on needs should be added in sheet **Trunks/Main settings** and **Trunks/VoIP settings**. See also chapter concerning VoIP functionality.

PABX hard reset

Hard reset wizard will prepare default configuration for installed module cards and ports.

☐ DHCP - DHCP client of LAN interface

Enter IP configuration of LAN interface.

LAN Interface

LANAdr - IP address of LAN interface
192 . 168 . 0 . 248

LANMsk - IP mask of LAN interface
255 . 255 . 255 . 0

LANGt - IP default gateway of a LAN interface
192 . 168 . 0 . 1

Cancel <<Back Next>>

Ilustracja 12.6: Hard reset wizard - LAN interface configuration

*** PABX hard reset** [X]

Hard reset wizard will prepare default configuration for installed module cards and ports.

Enter foneo VoIP trunk own number and password.

VoIP trunk				
Trl	Ptl	VoIP own number	Quantity of VoIP DDI	Pas
1-1-xx	SIP		1	

Cancel <<Back Next>>

Ilustracja 12.7: Hard reset wizard –VoIP trunks

Po zdefiniowaniu translacji należy w kolejnym kroku określić sposób wychodzenia „na miasto” (rodzaj analizy numeru w ruchu automatycznym) oraz ustalić sposób automatycznego nadawania numerów katalogowych abonentom w centrali.

Ilustracja 12.8: Hard reset wizard – extension numbering and outgoing method

Next step is setting parameters of PBX admin account:

- login: **admin**,
- password,
- code (key)⁴.

Due to security concerns, system won't allow to set up password or code/key last four digit of serial number. Ze względów bezpieczeństwa system nie pozwoli na umieszczenie w haśle i kodzie czterech ostatnich cyfr numeru seryjnego centrali.

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.

^{4 4} Used by services *87 – Manual working mode change and *92 – Announcement recording.

PABX hard reset

Wizard will create default administrator access account. Please specify login, password and four digits key for this account.

Login:

Password:

☒ Hide password on screen using * characters

Four digits key:

☒ Hide key on screen using * characters

Warning: Login and password are used by applications like ConfigMAN and BillingMAN, to connect with PABX.

Warning: Four digits key is needed to access phone services like announcement recording or working mode switching.

Cancel <<Back Create

Ilustracja 12.9: Hard reset wizard – admin account

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

PABX hard reset

Hard reset:

- ✓ Database preparation
- ✓ Module card approval
- ✓ Global settings
- ✓ Subscribers and doorphones configuration
- ✓ Group configuration
- ✓ Trunks groups configuration
- ✓ Trunks configuration
- ✓ Trunks list configuration
- ✓ Incoming calls configuration
- ✓ Outgoing calls and LCR table configuration

Hard reset wizard has successfully created PABX configuration.

Prepared configuration should be uploaded!

Required changes can be introduced before uploading configuration.

Cancel <<Back Close

Fig 12.10: Hard reset wizard – result

ATTENTION:

After using [Hard Reset Wizard](#) , the configuration needs to be uploaded to the PBX.

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

REMEMBERREMEMBER: *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 lose these data there will be no possibility to connect with PBX using **ConfigMAN** and **BillingMAN** applications).*

13 Controller format

Controller formatting is exclusively a service function. **Going through the complete procedure, causes all the information in the PBX to be lost – configuration, saved calls, announcements, billing data!** Controller memory will contain last saved firmware necessary to running PBX.

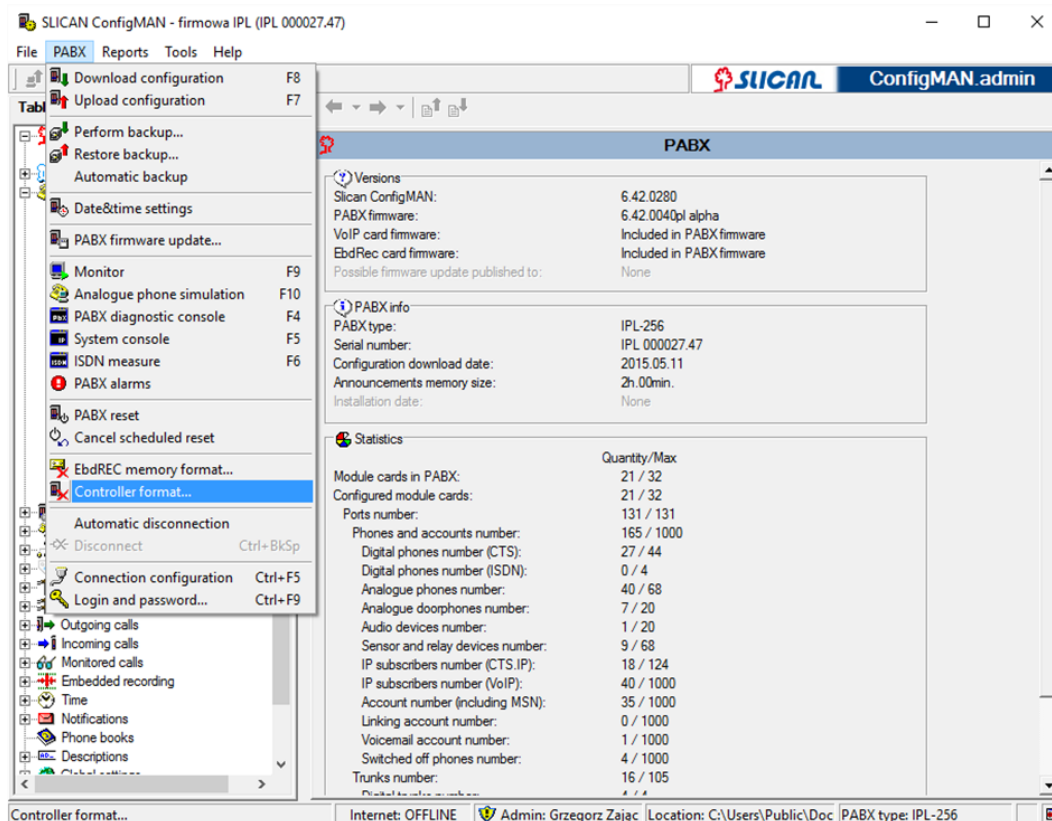


Fig 13.1: Controller format

After selecting option *Controller format* following window appear:

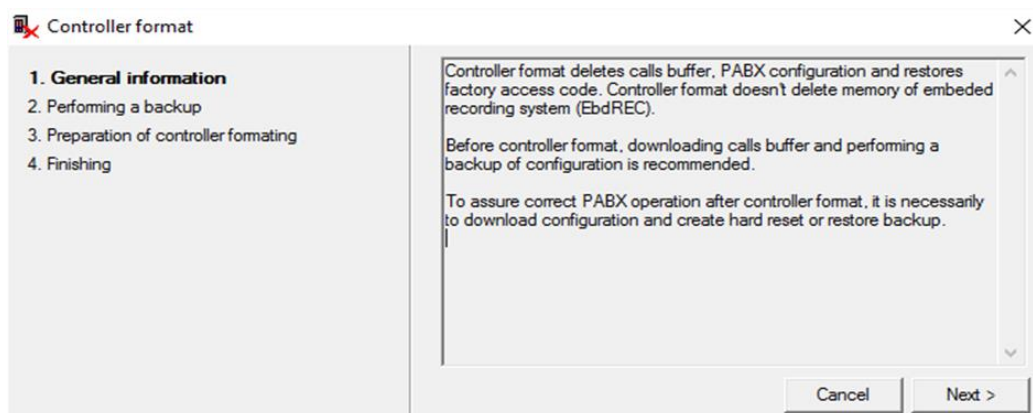


Fig 13.2: Controller format - general information

ATTENTION!

Before using command, it is recommended to download the call buffer and make a configuration backup copy.

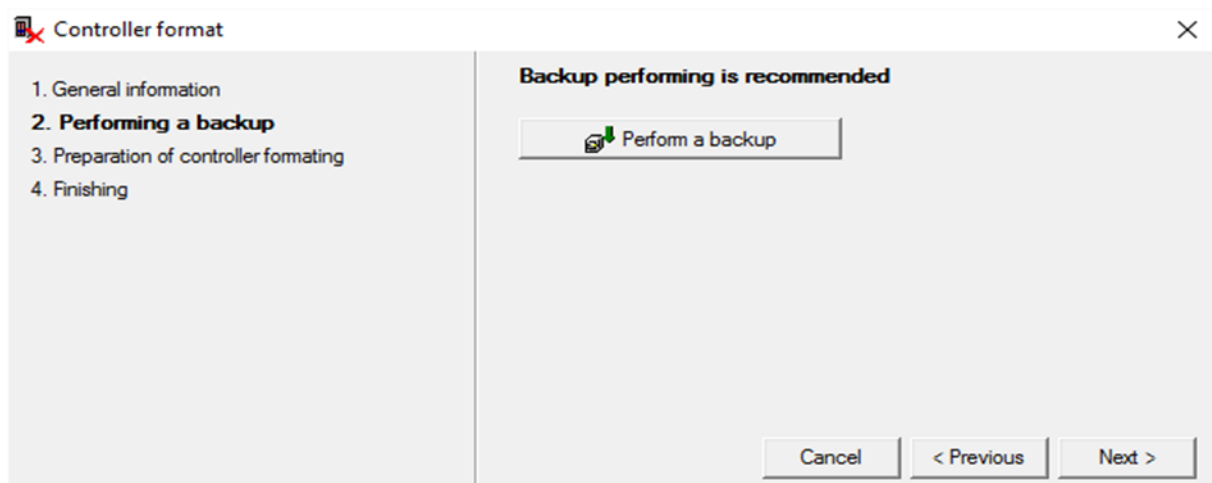


Fig 13.3: Controller format – performing backup

Window which appears in the next step depends on firmware version in PBX:

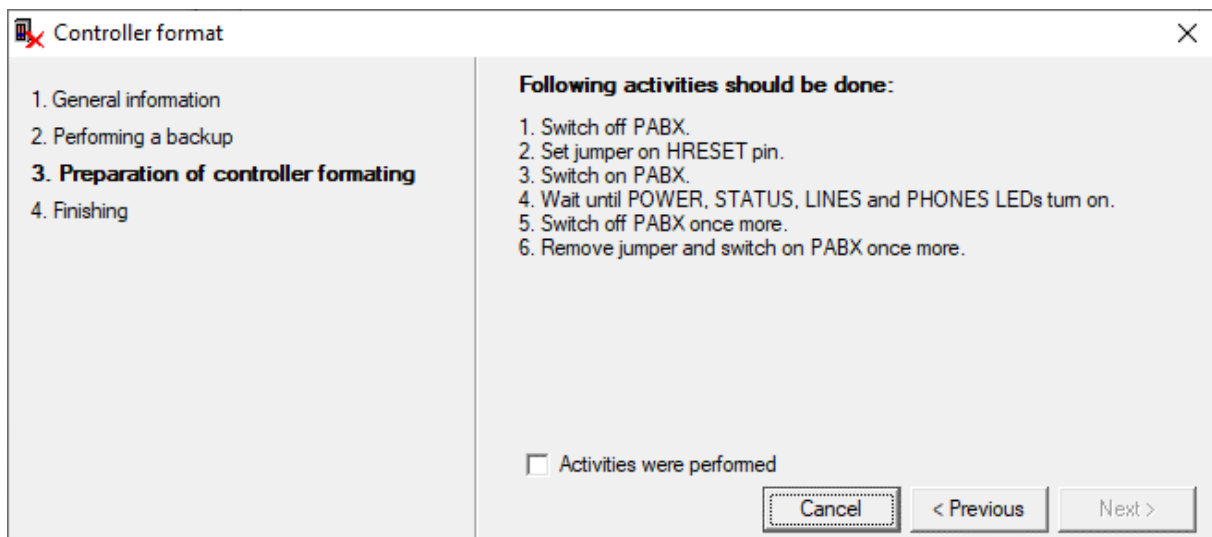


Fig 13.4: Controller format of IPx and CCT PBX

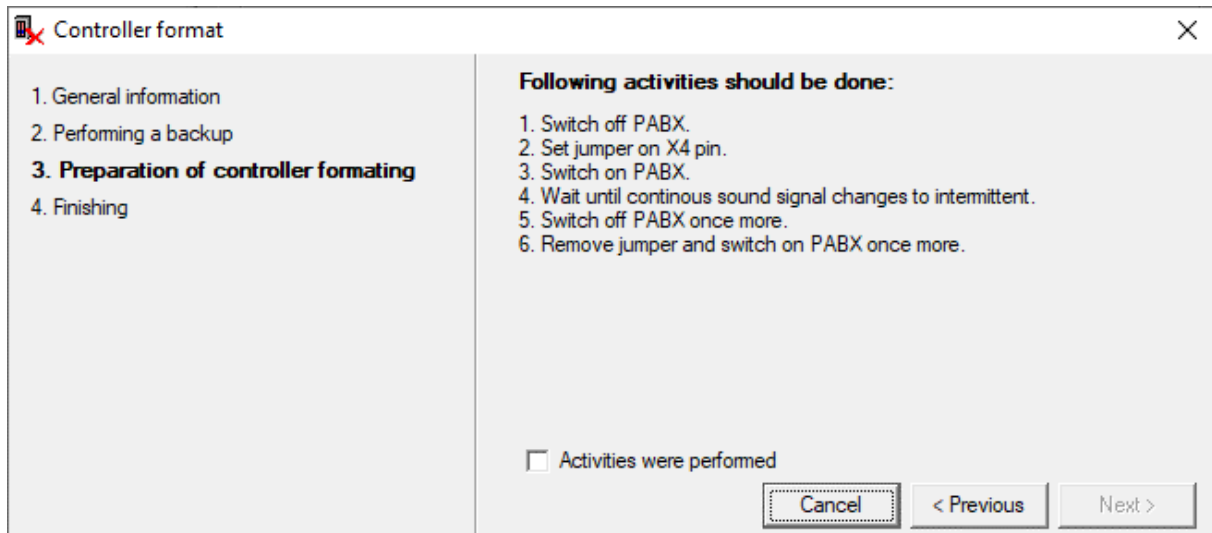


Fig 13.5: Controller format of MAC-6400 PBX

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:

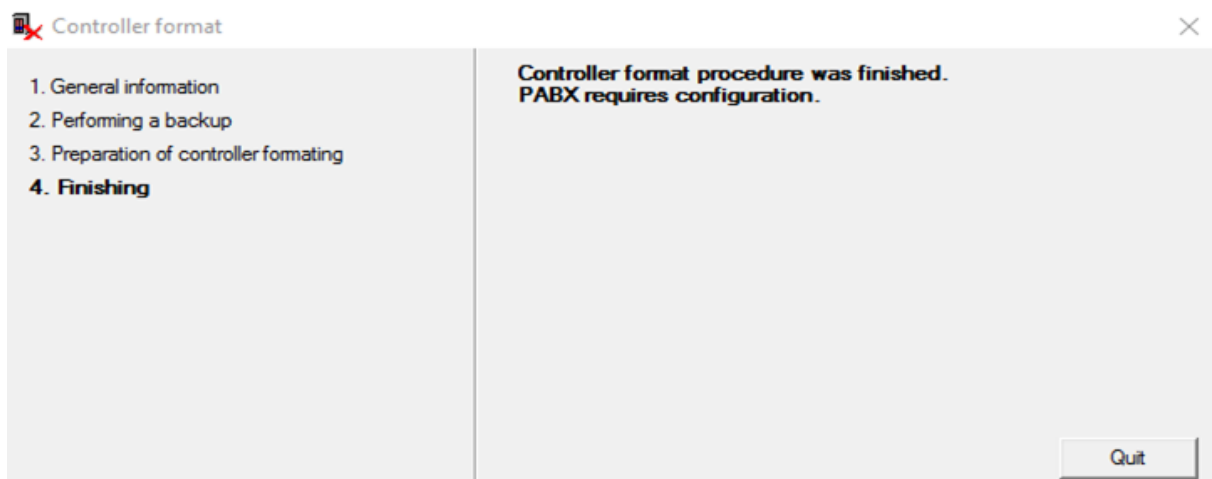


Fig 13.6: Controller format - finish

14 Uploading PBX firmware

14.1 Uploading controller firmware

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

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 continue longer then usual.

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

Firmware uploading can be performed via any interface but USB is recommended. To replace the firmware, use the **Firmware update** option in the **PABX** menu.

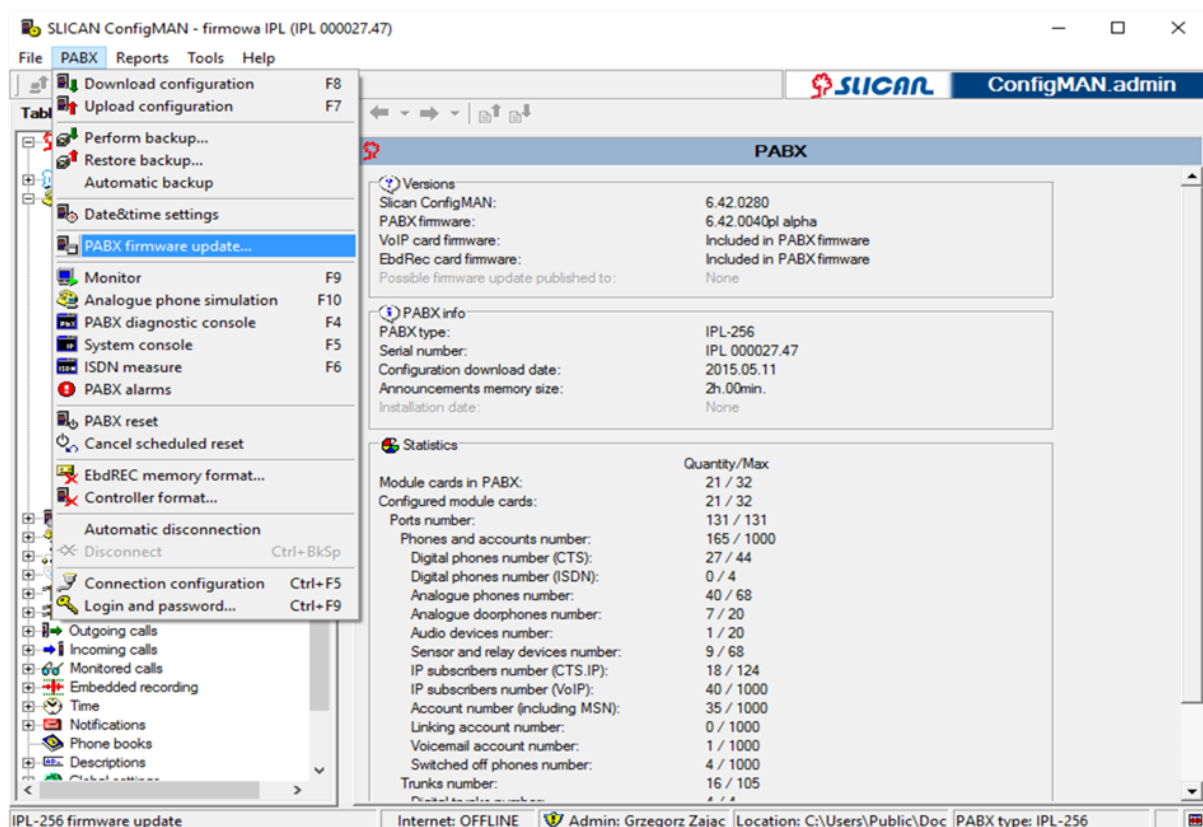


Fig 14.1: Update PABX firmware

Selecting this option will start the automatic firmware replacement wizard.

First message about requirement of downloading configuration before firmware uploading appears on screen:

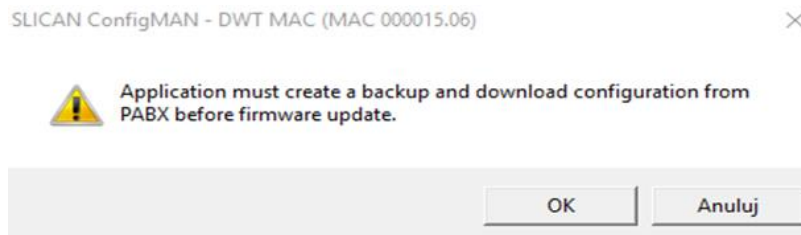


Fig 14.2: Firmware updating – download configuration request

After clicking **OK** button system will request the location of the current PBX firmware and make backup copy.

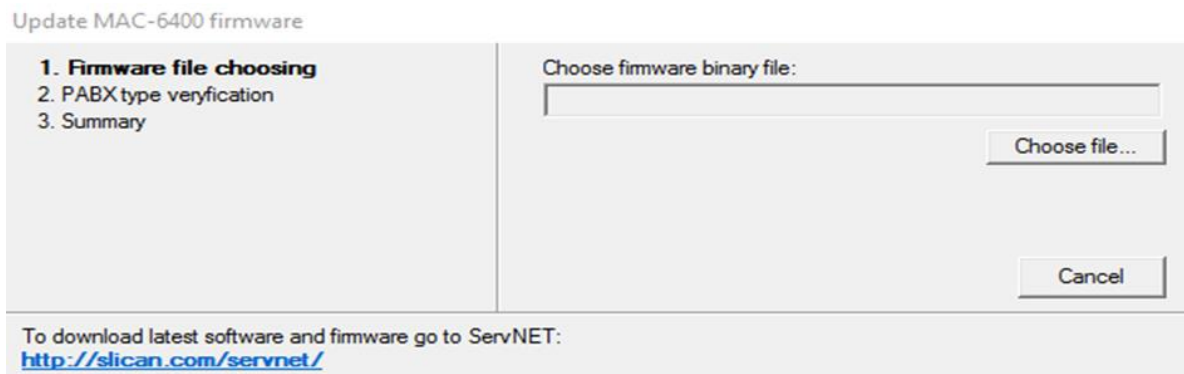


Fig 14.3: Firmware update – displaying first window

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 [Licenses for PBX firmware.](#)

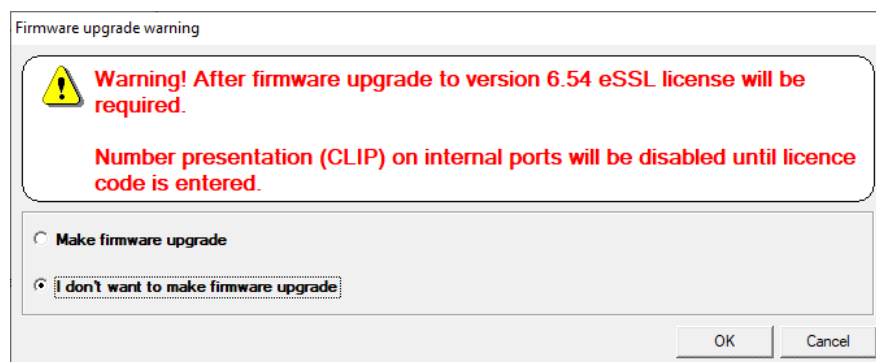


Fig 14.4: Firmware updating – license warning

After acquiring the file to be transmitted, the dialog box will display the firmware version, and the PBX families, this firmware is designed for.

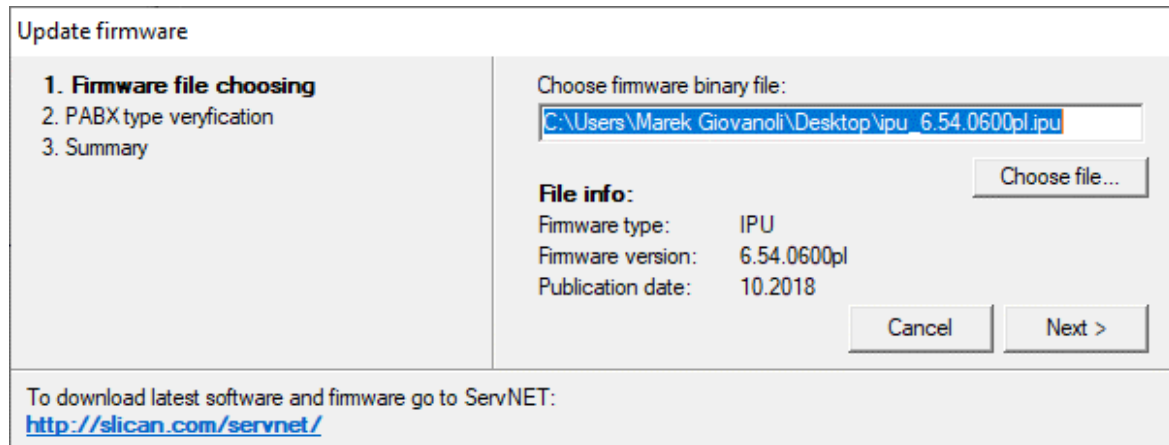


Fig 14.5: Firmware update – selecting file including appropriate firmware version

After confirming the firmware replacement process, the system will warn, that a reset of the PBX will be needed.

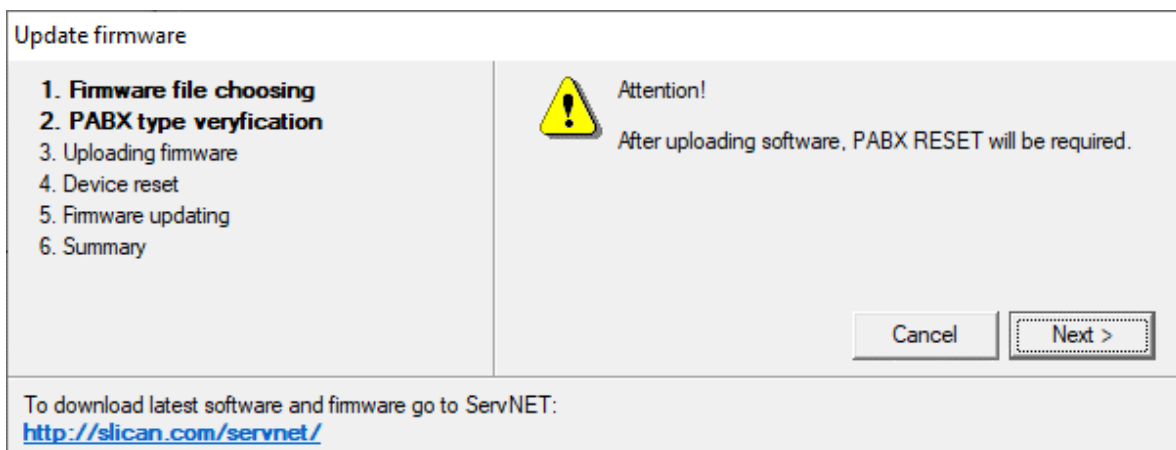


Fig 14.6: Firmware update – starting updating process

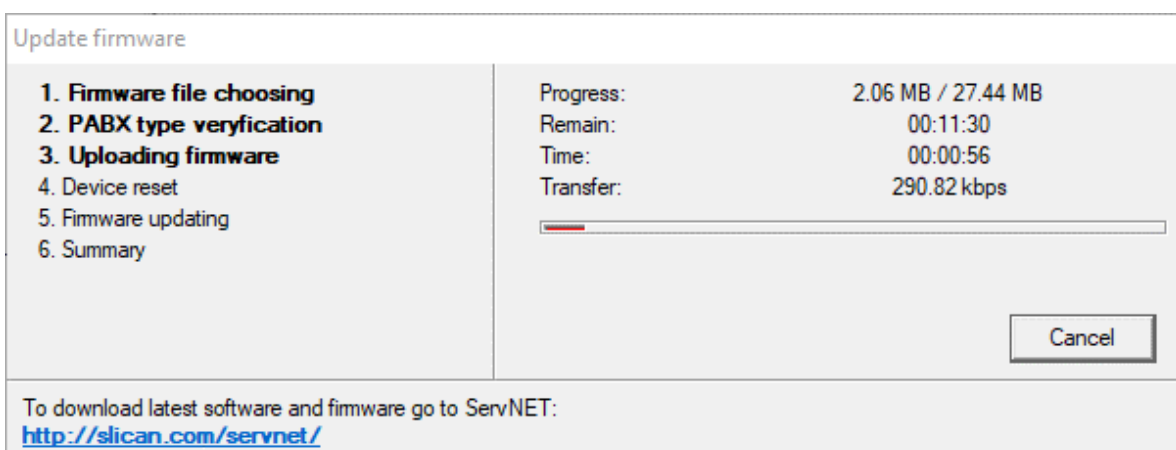


Fig 14.7: Firmware update – uploading firmware to PABX

After finishing uploading, the system will ask PBX reset. PBX reset is necessarily to proper functioning new firmware.

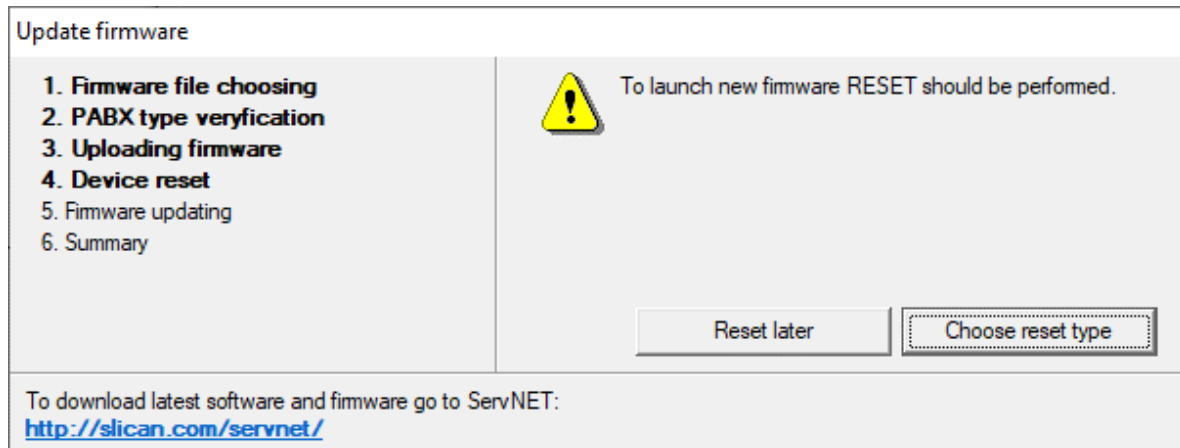


Fig 14.8: Firmware update – performing PABX reset

After finishing the process, the system will display a summary, or any possible causes of a failure.

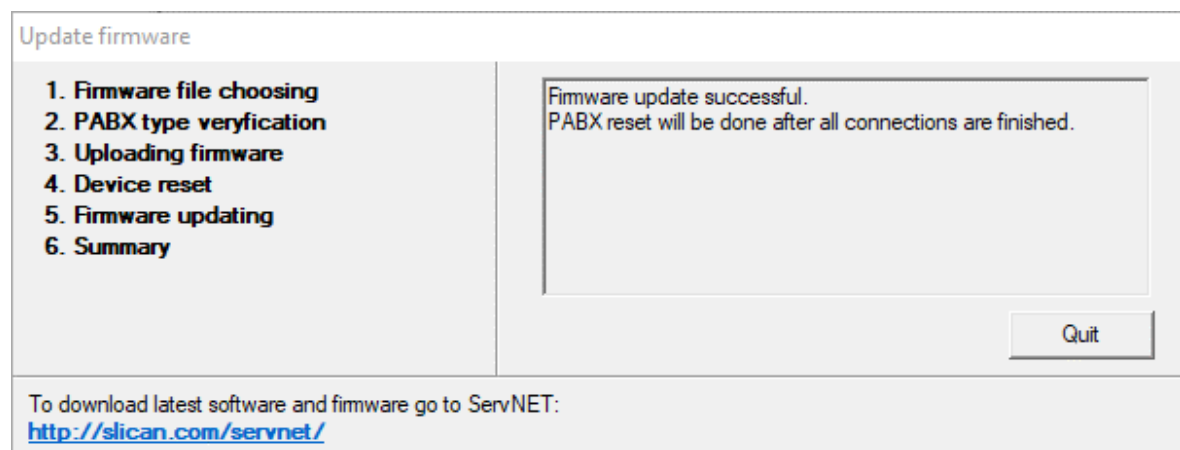


Fig 14.9: Firmware update – summary

14.2 VoIP card firmware changing

ATTENTION!

During VoIP card firmware changing power supply cannot be switched off and connection to PBX can't be lost. Since firmware version 6.xx VoIP firmware update concern only MAC and CXS PBXes..

MAC-6400 PBX can change VOIP card firmware using any connection media – USB, Ethernet network, Internet network – keeper. It is recommended to use USB port.

After connecting to PBX choose menu **PABX** option **VoIP card firmware update**.

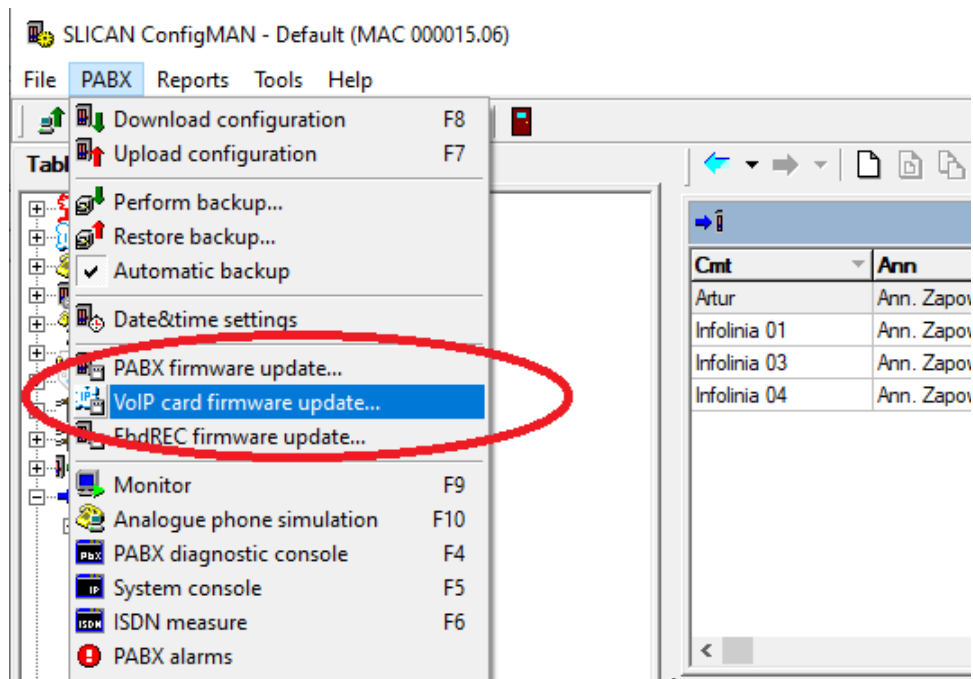


Fig 14.10: VoIP card firmware update – select option

During next step this window will be visible:

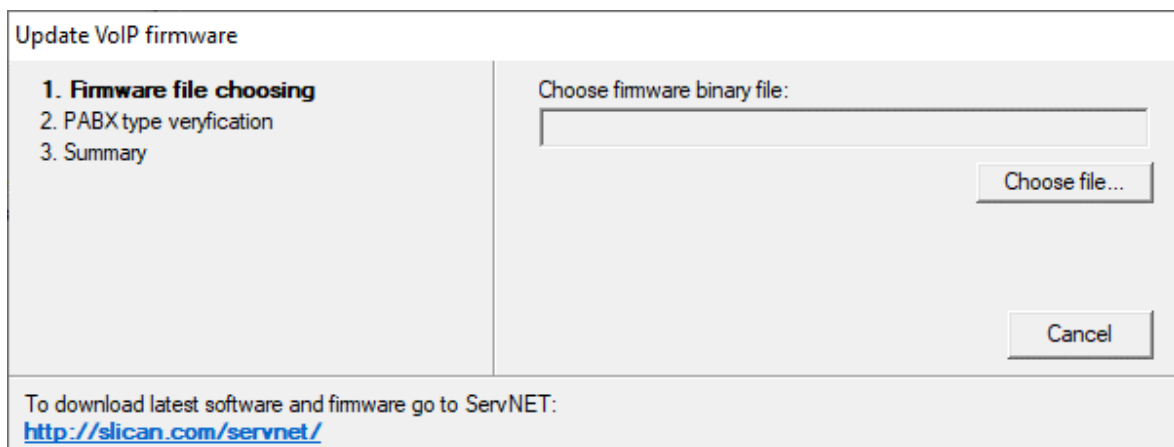


Fig 14.11: VoIP card firmware update – selecting firmware file

Next we select location of new firmware using button *Choose file...* and click *Next*.

Update VoIP firmware

1. Firmware file choosing 2. PABX type verification 3. Summary	Choose firmware binary file: C:\Users\Marek Giovanoli\Desktop\macvoip_6.55.0300.voip File info: Firmware type: VoIP for MAC-6400 Firmware version: 6.55.0300 Choose file... Cancel Next >
---	--

To download latest software and firmware go to ServNET:
<http://slican.com/servnet/>

Fig 14.12: VoIP card firmware update – firmware file was selected

Remember After clicking *Next*, PBX **cannot be reseted** during firmware updating.

Update VoIP firmware

1. Firmware file choosing 2. PABX type verification 3. Uploading firmware 4. Device reset 5. Firmware updating 6. Summary	 Attention! After uploading software, VoIP card will self RESET.
--	---

To download latest software and firmware go to ServNET:
<http://slican.com/servnet/>

Fig 14.13: VoIP card firmware update – warning about reset

After pressing *Next* key, new firmware is uploaded to the card:

Update VoIP firmware

1. Firmware file choosing 2. PABX type verification 3. Uploading firmware 4. Device reset 5. Firmware updating 6. Summary	Progress: 1.48 MB / 13.12 MB Remain: 00:08:33 Time: 00:01:05 Transfer: 188.42 kbps 
---	---

To download latest software and firmware go to ServNET:
<http://slican.com/servnet/>

Fig 14.14: VoIP card firmware update – firmware uploading

After successful finishing updating application informs as follows.

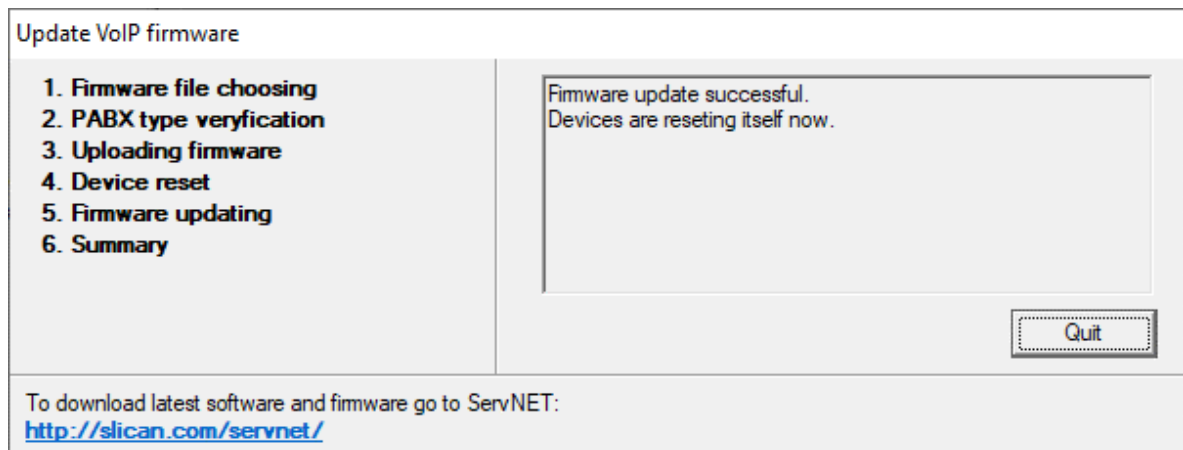


Fig 14.15: VoIP card firmware update- summary

Information about firmware updating can be found in *PABX monitor*, in sheet *PABX events*.

ATTENTION:

If interrupting connection system informs about it. Perform all described above activities once more. If it is unsuccessful, contact with service.

15 PBX alarms

A useful function of the software that have possibility of tracing all alarms, notifying us of any possible irregularities.

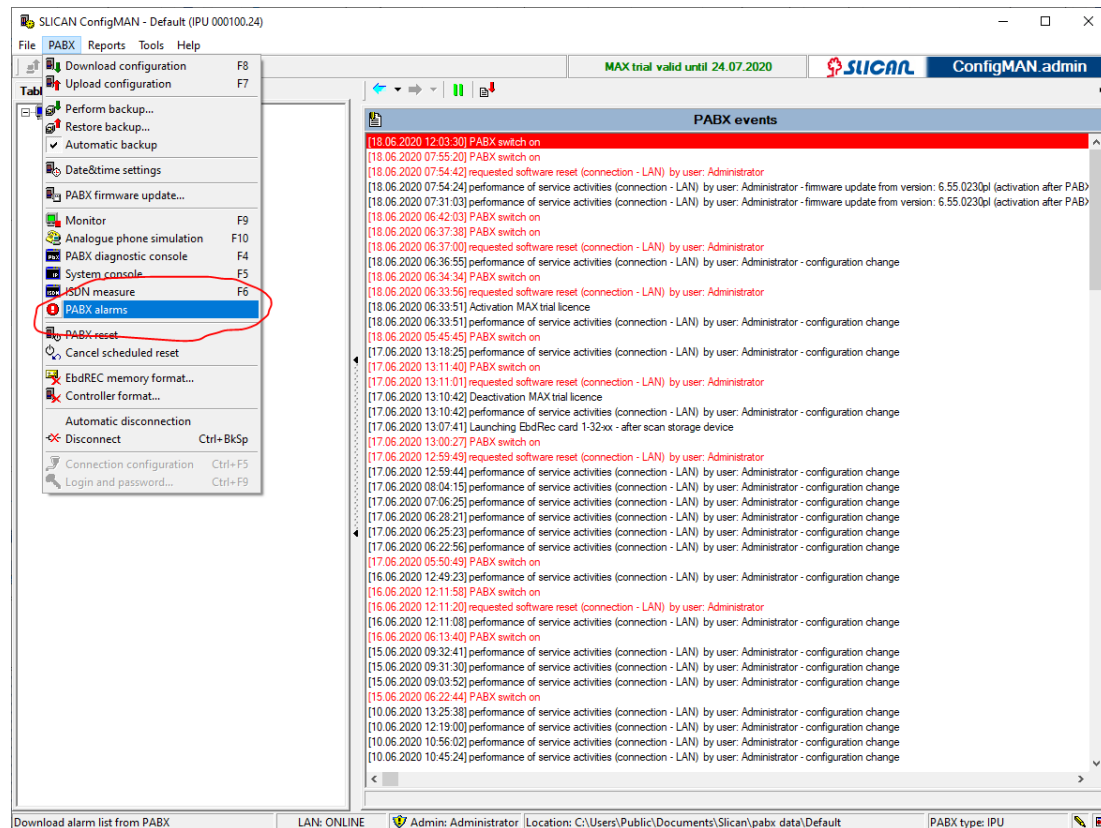
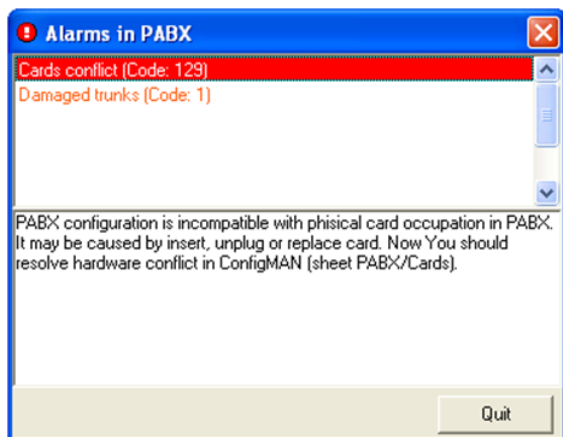


Fig 15.1: PABX alarms

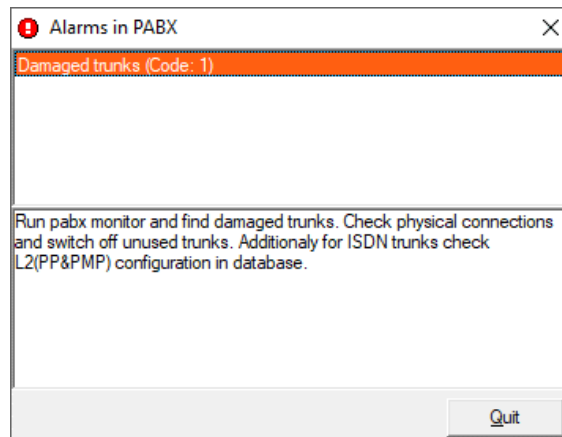
Examples of critical errors:

- card conflict,
- konflikt wyposażenia,
- port conflict,
- improper firmware,
- invalid value of battery voltage,
- no PBX serial number,
- no power supplied to device,
- call recording card is 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.



Ilustracja 15.2: PABX alarm – example 1



Ilustracja 15.3: PABX alarm – example 2

16 Notifications

Current PBX monitoring can be easier due to notification functionality. Due to this functionality, SMS or e-mail messages are sent to previously defined GSM numbers or e-mail addresses. These messages inform about events of some types. Basic requirement of this functionality is installing GSM card in PBX and/or configuring SMTP server for sending mail messages.

SMS configuration is done in sheet **Notifications**:

- **NtfSMS** – method of sending SMS notifications:
 - using *SMS center* notifications
 - define GSM trunk which will be sending notifications
- **ChrAcc** – in this field notification SMS cost payer should be defined.

ATTENTION!

To enable sending notifications set up field **SmsStu** in sheet **Trunks/GSM settings** option **SMS center**.

SMTP server configuration is done from sheet **Notifications/SMTP client**:

- ~~Enable field **ErnSMTP** – mail messages will be send using dedicated Slican server – removed since firmware version 6.52~~
- Enable field **ErnSMTP** – mail messages will be send by server configured in this field..

Additionally in sheet **Notifications/Events** in field **LogCmt** can have comment info, which will display failed login to PBX **ConfigMAN** i **BillingMAN** software.

16.1 Service alerts

In this sheet up to 8 numbers (field **No.**) can be defined, to these numbers SMS (or e-mail) notification, informing about specified alerts are send. Depending on notification type in field **NtfTyp**, field **Rcp** defines number or e-mail address that will receive notifications about alerts. Notification about following alerts can be switched on::

- **SysFlr** – system failure e.g.:
 - uszkodzenie prądu dzwonka
 - 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** or **AlrEn** is checked in ports sheet) eg.:
 - trunk failure/fixing (*Trunks/Main settings*)
 - subscriber port failure/fixing (*Subscribers/Special services*)
 - DPH.AB and DPH.IP doorphones failure/fixing (*Peripherals/Doorphones/ Main settings*)
- **Cfg** – uploading new configuration to the PBX.:
 - configuration change,
 - firmware updat
 - announcement upload
 - LAN/VoIP/EREC card firmware update
 - memory storage format
- **EvtLog** – fulfilling event buffer in 85% or its overload
- **RecFlr** – recording system failureand monitoring RM.server connection status. Status checking frequency (None, 10 minutes, 24 hours) is configured in **Embedded recording** tab in **ChkCnn** fiel,
- **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,
- **ACS** – events related to Access Control System

Additionaly number set up in field **Rcp** will be informed with **SMS** about configuration change for his phone settings.

UWAGA!

*To enable sending notifications set up field **SmsStu** in sheet **Trunks/GSM settings** option **SMS center**.*

16.2 Events

In this sheet up to 50 events can be defined, which will trigger sending SMS or e-mail notification. 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 triggered (specific sensor, any sensor – sending notification from any sensor)
- door opening time exceeded
- sensor return to rest state (specific sensor, any sensor – sending notification from any sensor)

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.

17 Monitor

Monitor is a **ConfigMAN** function, designed to supervise PBX operations. It can be used to observe:

- PBX port status,
- external line status,
- internal line status,
- VoIP channels status,
- recordings channel status,
- PBX linking state - eSSL
- status of calls made by PBX,
- PBX events.
- Trunks groups busy statistics

The PBX can be monitored using any available connection method.

UWAGA!

*For the **Monitor** to 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, using icon .

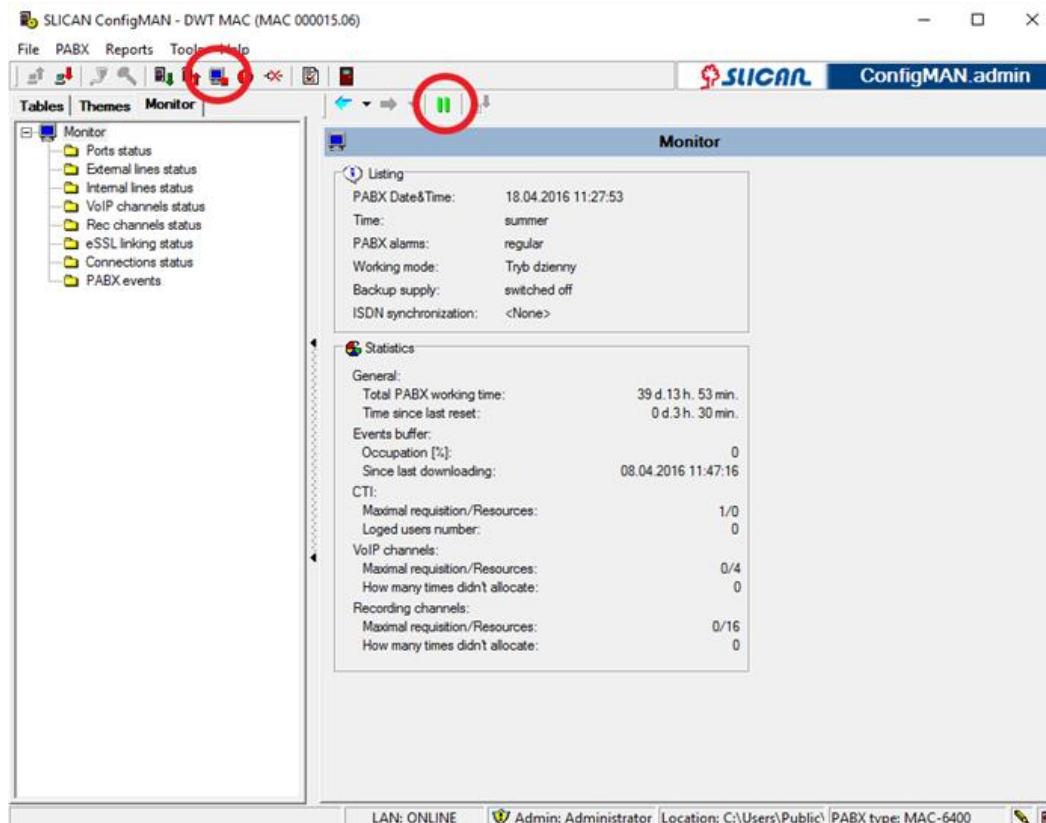


Fig 17.1: PBX monitor

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 screen, making them easier to analyze in detail. Clicking again the button  or F11 refreshes the status of monitored lines and ports.

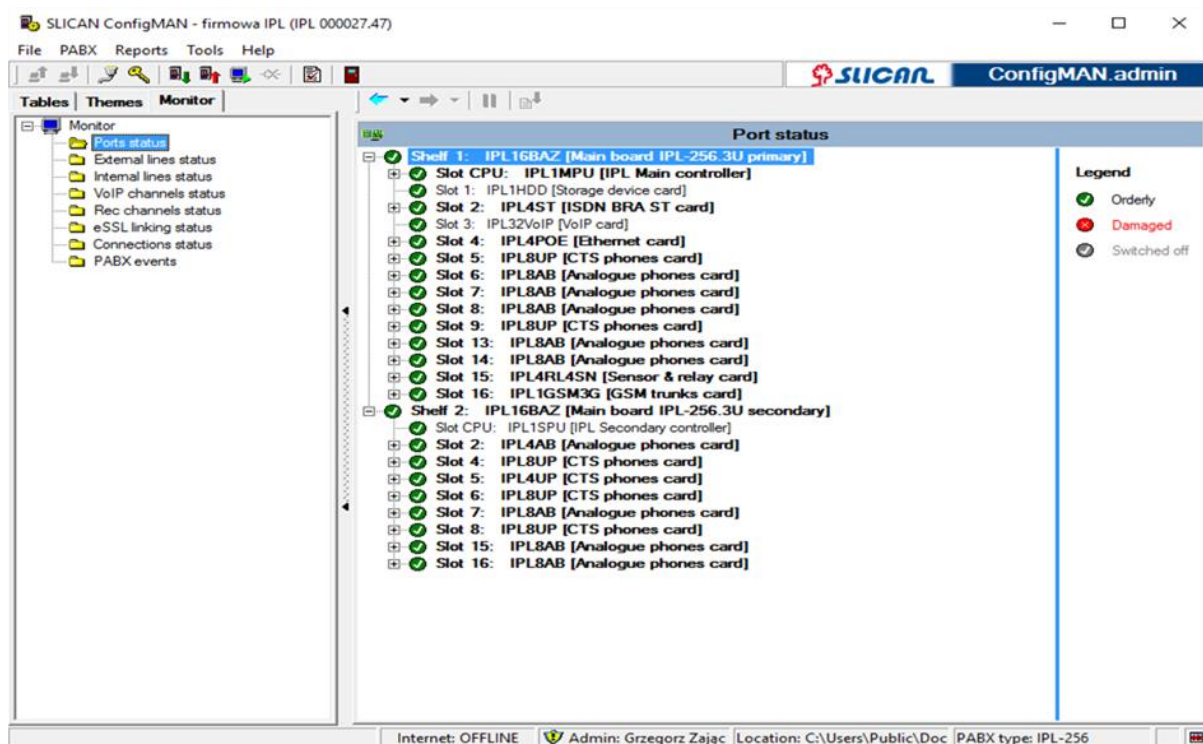
17.1 Monitoring port status

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,
- VoIP trunks,
- CTS.IP subscribers.

Additionally information about status of LAN/WAN ports and connection to keeper.slican.pl server are presented.



IlustFigracja 17.2: PBX Monitor – Port status

✓ < 2071 Marek > <<Subscriber is not logged>> <<0.0.0.0>>

Fig 17.3: Port status for VoIP subscriber after switching on PBX

✓ < 2118 Lechu SIP 2118 > <<192.168.16.10:5260>>

Fig 17.4: Port status for VoIP subscriber after log in – used by subscriber IP address is visible and used port

✓ < 2067 Artur test voip1 > <<Subscriber is not logged>> <<192.168.16.50:64787>>

Ilustracja 17.5: Prezentacja stanu portu abonenta VoIP po wylogowaniu – widoczny jest adres IP z którego ostatnio logował się dany abonent

For GSM trunks it is possible to monitor current status selected GSM port (ex. information about log in to network or SIM card installation, network mode 2G/3G) as well as signal level information which can be used while aerial installation. GSM should be set up during phone call due to additiona signal loss.

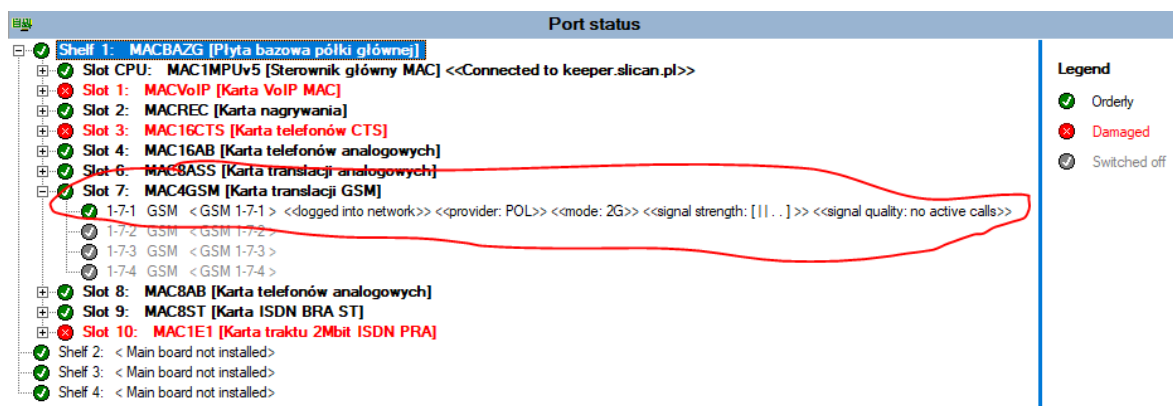


Fig 17.6 PABX Monitor – Port status – GSM ports

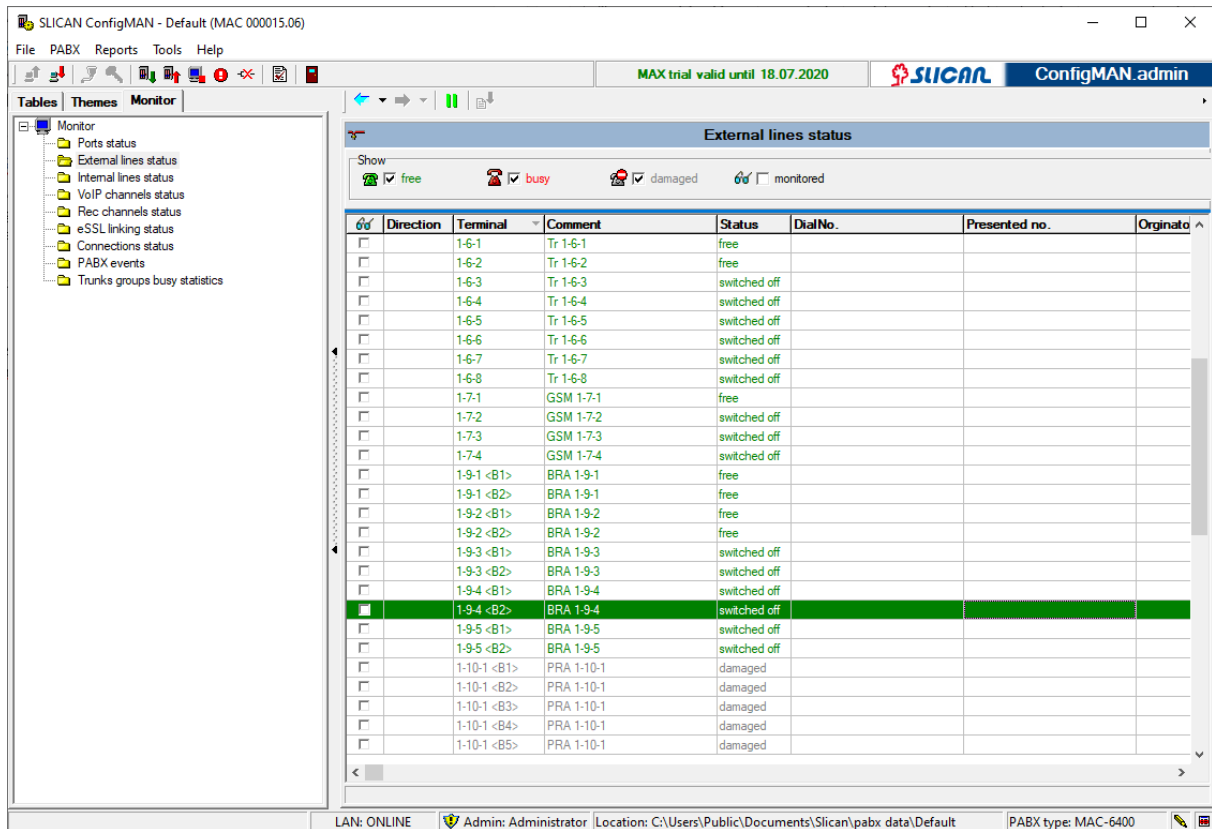
17.2 Monitoring of external lines

select individual trunks to be watched (toggling them with the glasses icon).

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 below.

In the **Show** field, you can set one of four available filters:

- *free* – unused trunks,
- *busy* – trunks currently in use,
- *damaged* – trunks marked by the PBX as damaged,
- *monitored* – only lines set up as monitored ones.



Ilustracja 17.7: Monitor – External lines status

17.3 Monitoring of internal line status

Monitoring internal lines shows, in real time (online), all subscriber status (internal ports) in the PBX. If a subscriber is making, or receiving a call, the number dialed is displayed.

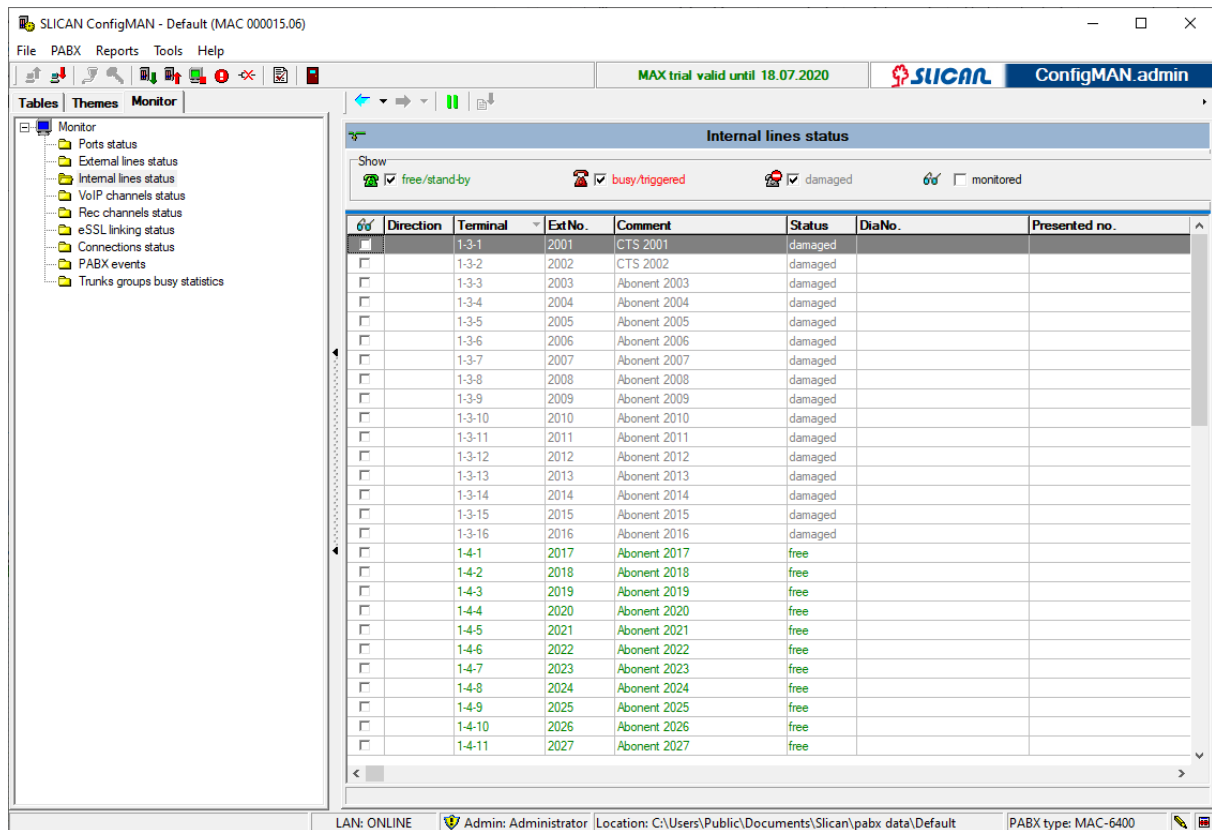


Fig 17.8: Monitor – Internal lines status

Displayed bellow figure presents, in Monitor window, one connection between extension numbers. In this situation from extension number 2017 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 "DINo" and "Presented No" include the same values. It is because in presented situation connection is internal.

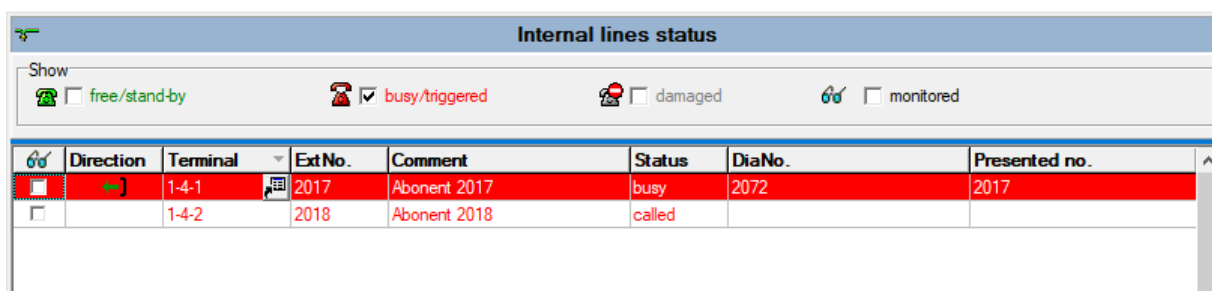


Fig 17.9: Monitor – Internal lines status – Internal connections

17.4 VoIP channels status

If in PBX VoIP card is installed, monitor enables to monitor channel occupation. Table presents outgoing call (column ***Direction***) with VoIP caller info (column ***Comment***), dialed number (column ***DialNo.***). Caller presented number (column ***Presented no.***). Channel status is defined in column ***State***. Established can connection occupy only one VoIP channel, it means that is established between VoIP and regular internal extension number (analogue phone, ISDN, CTS Phone) or external number in PSTN network.

The screenshot shows the SLICAN ConfigMAN - Default (MAC 000015.06) interface. The 'Monitor' tab is selected, and the 'VoIP channels state' table is displayed. The table has columns: Direction, Channel, Comment, State, DialNo., Presented no., and Signal. The table shows four rows of data, all with a 'free' state.

Direction	Channel	Comment	State	DialNo.	Presented no.	Signal
	<B1>		free			
	<B2>		free			
	<B3>		free			
	<B4>		free			

The interface also includes a sidebar with a tree view of monitoring options: Ports status, External lines status, Internal lines status, VoIP channels status, Rec channels status, eSSL linking status, Connections status, PABX events, and Trunks groups busy statistics. The status bar at the bottom shows LAN: ONLINE, Admin: Administrator, Location: C:\Users\Public\Documents\Slcan\pabx data\Default, and PABX type: MAC-6400.

Fig 17.10: Monitor – VoIP channel status

17.5 Recording channel status

SLICAN ConfigMAN - Default (MAC 000015.06)

File PABX Reports Tools Help

MAX trial valid until 18.07.2020

ConfigMAN.admin

Tables Themes Monitor

Monitor

- Ports status
- External lines status
- Internal lines status
- VoIP channels status
- Rec channels status
- eSSL linking status
- Connections status
- PABX events
- Trunks groups busy statistics

Rec channels state

Show ☒ free ☒ busy

Terminal	Originator	State	ID	Forward ID	Terminator	Dialed/CLIP
1-2-1 <B9>		free	0000000	00		
1-2-1 <B16>		free	0000000	00		
1-2-1 <B15>		free	0000000	00		
1-2-1 <B14>		free	0000000	00		
1-2-1 <B13>		free	0000000	00		
1-2-1 <B12>		free	0000000	00		
1-2-1 <B11>		free	0000000	00		
1-2-1 <B10>		free	0000000	00		
1-2-1 <B1>		free	0000000	00		
1-2-1 <B8>		free	0000000	00		
1-2-1 <B7>		free	0000000	00		
1-2-1 <B6>		free	0000000	00		
1-2-1 <B5>		free	0000000	00		
1-2-1 <B4>		free	0000000	00		
1-2-1 <B3>		free	0000000	00		
1-2-1 <B2>		free	0000000	00		

LAN: ONLINE Admin: Administrator Location: C:\Users\Public\Documents\Slisan\pabx_data\Default PABX type: MAC-6400

Ilustracja 17.11: Monitor – Recording channels status

If in PBX cpu controller or REC 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 **Originator** 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.client* to find specified call recordings.

17.6 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.

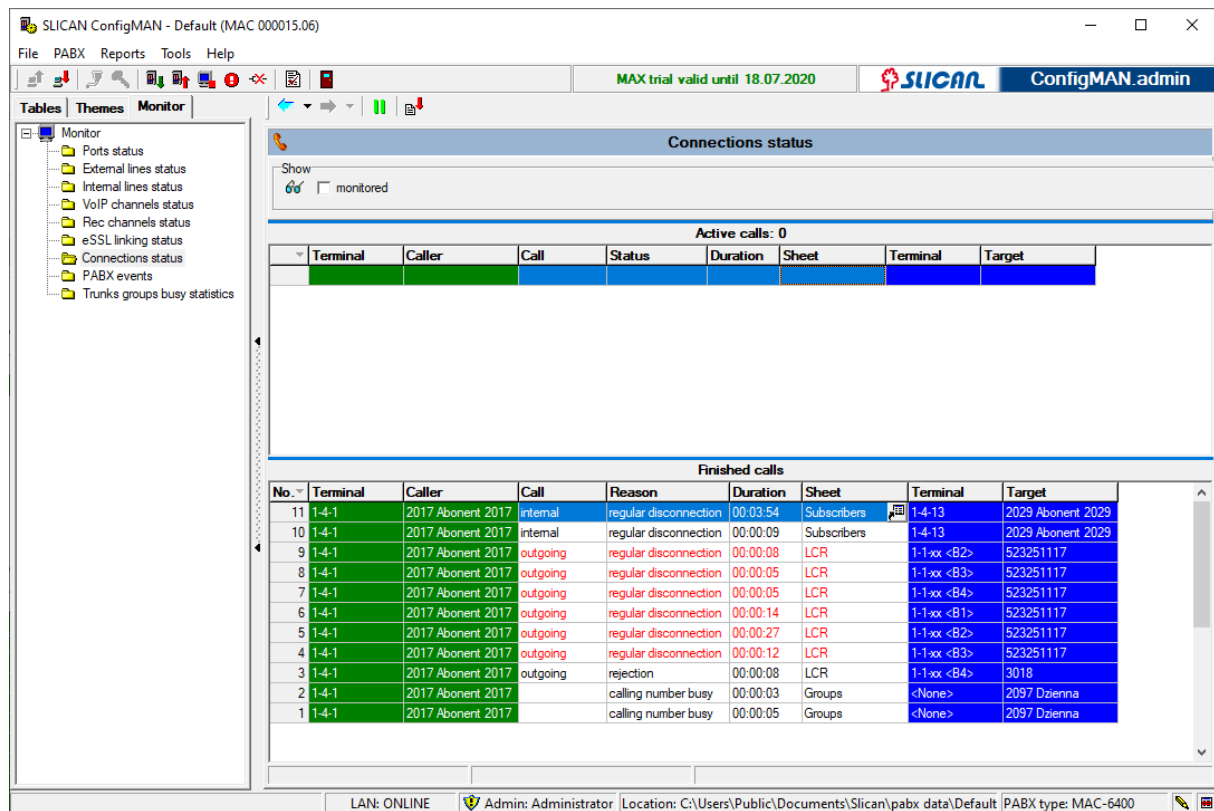


Fig 17.12: Monitor – Connection status

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 - **Call** 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 **Terminal** field is a part of external call.
- Internal line status – if the clicked **Terminal** 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. Additionally, you can apply the **monitored** filter to limit the number of records displayed.

Connections status								
Show ☒ monitored								
Active calls: 1								
	Terminal	Caller	Call	Status	Duration	Sheet	Terminal	Target
	1-4-1	2017 Abonent 2017	internal	calling	00:00:16	Subscribers	1-4-13	2029 Abonent 2029
Finished calls								
No.	Terminal	Caller	Call	Reason	Duration	Sheet	Terminal	Target
10	1-4-1	2017 Abonent 2017	internal	regular disconnection	00:00:09	Subscribers	1-4-13	2029 Abonent 2029

Fig 17.13: Monitor – Connection status window

Window *Active calls* contains few hidden 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 17.14)
- placing mouse cursor over field **Target** causes displaying box with information about presented number (Fig 17.15)
- placing mouse cursor over field **Duration** causes displaying box with information about time of establishing call (Fig 17.16)

Connections status								
Show ☒ monitored								
Active calls: 1								
	1-4-1	2017 Abonent 2017	outgoing	call	00:00:16	LCR	1-1-xx <B1>	523251117
		Dialed no.: 523251117						
Finished calls								
No.	Terminal	Caller	Call	Reason	Duration	Sheet	Terminal	Target
11	1-4-1	2017 Abonent 2017	internal	regular disconnection	00:03:54	Subscribers	1-4-13	2029 Abonent 2029
10	1-4-1	2017 Abonent 2017	internal	regular disconnection	00:00:09	Subscribers	1-4-13	2029 Abonent 2029
9	1-4-1	2017 Abonent 2017	outgoing	regular disconnection	00:00:08	LCR	1-1-xx <B2>	523251117

Fig 17.14: Monitor – Connection status – Information about dialed number

Connections status									
Show  ☐ monitored									
Active calls: 1									
No.	Terminal	Caller	Call	Status	Duration	Sheet	Terminal	Target	
	1-4-1	2017 Abonent 2017	outgoing	call	00:00:12	LCR	1-1-xx <B4>	523251117	Presented no.: 525065118
Finished calls									
No.	Terminal	Caller	Call	Reason	Duration	Sheet	Terminal	Target	
12	1-4-1	2017 Abonent 2017	outgoing	regular disconnection	00:00:24	LCR	1-1-xx <B1>	523251117	
11	1-4-1	2017 Abonent 2017	internal	regular disconnection	00:03:54	Subscribers	1-4-13	2029 Abonent 2029	

Fig 17.15: Monitor – Connection status – Information about presented number

Connections status									
Show  ☐ monitored									
Active calls: 1									
No.	Terminal	Caller	Call	Status	Duration	Sheet	Terminal	Target	
	1-4-1	2017 Abonent 2017	outgoing	call	00:00:06	LCR	1-1-xx <B3>	523251117	Connection hour: 08:43:11
Finished calls									
No.	Terminal	Caller	Call	Reason	Duration	Sheet	Terminal	Target	
13	1-4-1	2017 Abonent 2017	outgoing	regular disconnection	00:01:53	LCR	1-1-xx <B4>	523251117	

Fig 17.16: Monitor – Connection status – Information about call starting hour

17.7 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,
- LAN network issues,
- call recording card damaging,
- wykonanie czynności serwisowych (zmiana konfiguracji, aktualizacja firmware'u centrali, karty VoIP/EbdRec),
- reset free minutes assigned to the trunk.

Event list can be write in text (*.txt) or *.csv file.

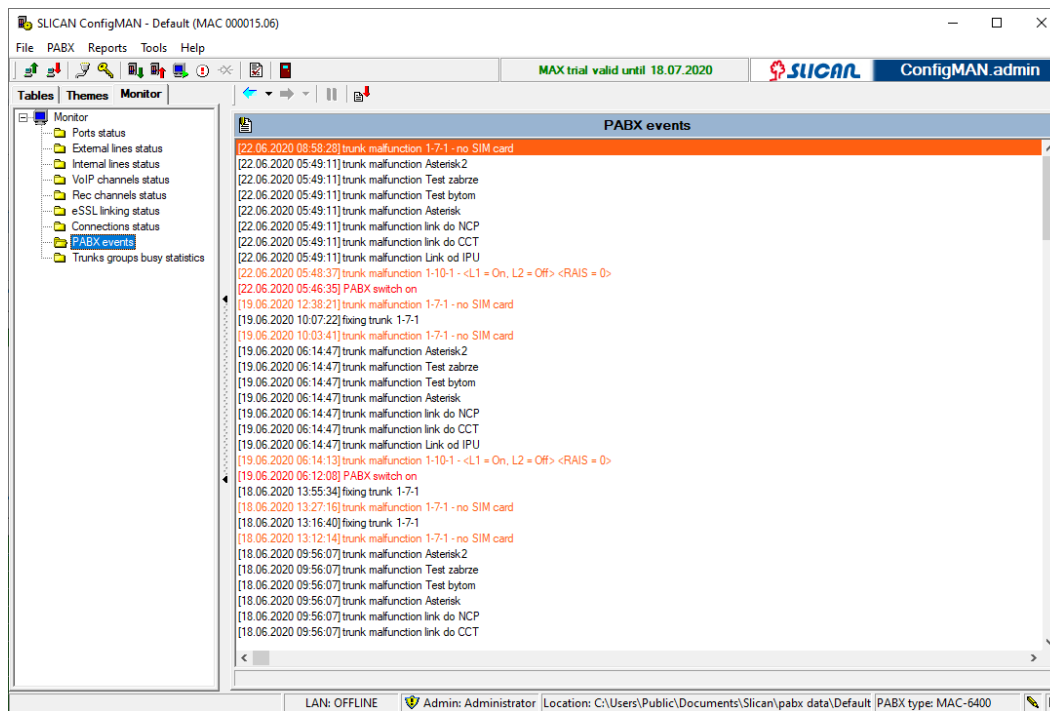


Fig 17.17: Events in PBX

17.8 Trunks groups busy statistics

This sheet allows to analyse PBX traffic. It allows to find GNR/HMT (hour of most traffic), details of this traffic, amount of calls made during same time or for specific trunk groups. Statistics are shown as bar graph. They are usefull then considering about increasing (higher bandwith) or decreasing (savings) trunks in each direction.

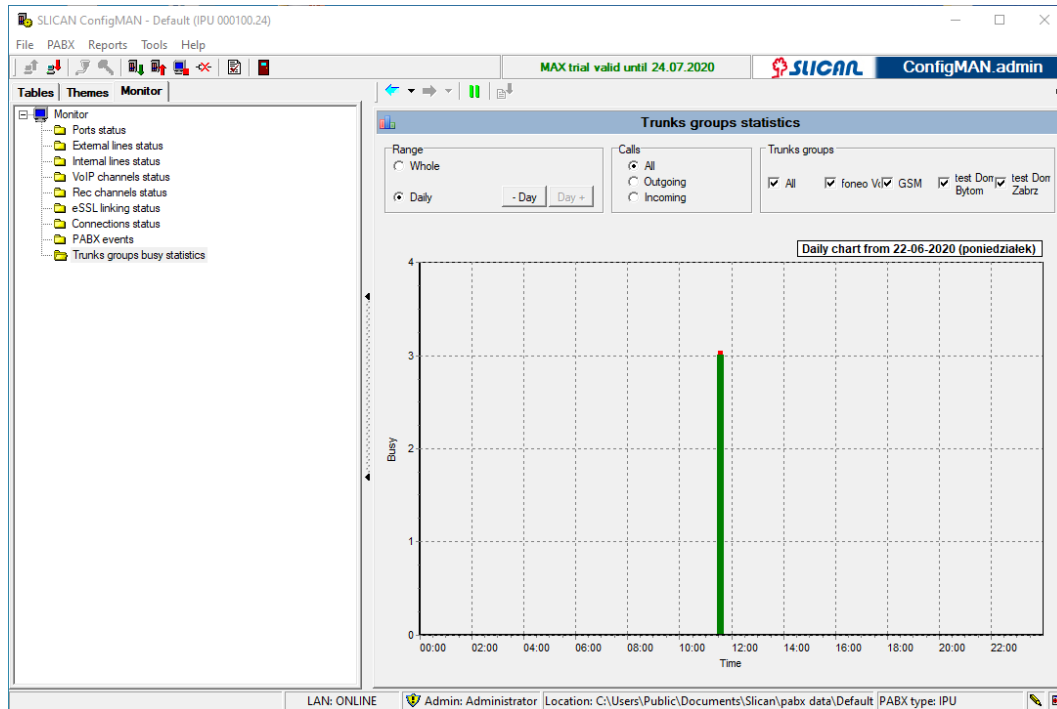


Fig 17.18: Trunks groups busy statistics

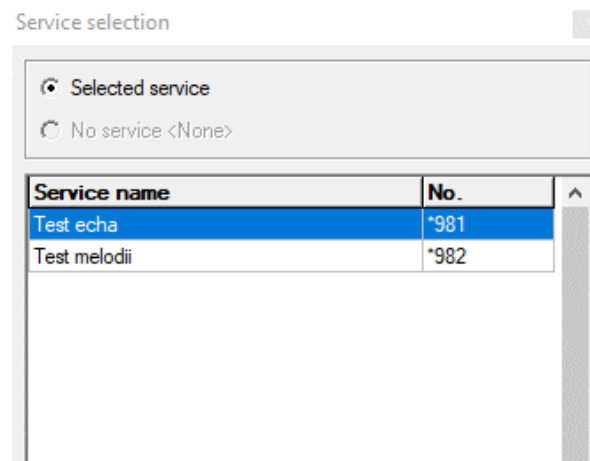
18 Acoustic test services

Slican PBX have two service for testing trunk voice quality. Internally those are services numbers:

- Echo test - *981,
- Melody test - *982.

Codes can be used with any phone connected to PBX and during incoming call to DISA. During „echo test” there is a loop so testing person will hear themselves with slight delay. During „Melody test” testing person will hear melody played by PBX during hold. .

Incoming calls can be routed to service in sheet **Incoming calls/Settings/Main** field **ExtNo=Test**.



Service name	No.
Test echa	*981
Test melodii	*982

Fig 18.1: Choosing test type

Service is especially usefull during remote management eg. testing VoIP calls quality. Testing allows to decide about changing audio codec to improve call quality.

19 Licenses

In PBX licenses are required for:

- PBX firmware,
- software: *TelefonCTI* i *TelefonCTI.plus*,
- software: [*MessengerCTI.Desktop*](#), [*MessengerCTI.Mobile*](#), [*WebCTI*](#),
- software: [*ConsoleCTI*](#),
- CTIP protocol
- sending CTIP protocol to RS-232 interface (included in license for *TelephoneCTI*),
- TAPI workstation,
- VoIP subscribers ports,
- VoIP channels,
- embedded recording system (license includes number of recording channel, *RecordMAN.server* and *RecordMAN.client* accounts, *ftp*),
- Infolines (number of infolines and maximal time of all recorded announcements),
- number of conference participants,
- MobilePhone and CallBack service
- Number of simultaneous calls via XML.IVR,
- Amount of GSM trunks, which can be used to sending and receiving SMS (*SenderSMS*, *TelefonCTI*, external applications)
- Amount of subscribers within eSSL system, in this linked PBX
- Amounts of users within Access Control System

19.1 Błąd! Nie można odnaleźć źródła odwołania.

Entering license code for firmware is required if you updating firmware from version 3.08 (or earlier) to current version.

Full table of relations between firmware version and serial number is available on site: [Serial numbers for firmware version](#).

Firmware license requirements can have two variants:

- PBX fw. version higher or including 6.22 – updating firmware to specific fw. version is verified by checking PBX serial number (from specific table of CCT, MAC. IPL) to fw version (fw version column). Proper version can be uploaded to PBX without extra license if that version is same or lower eg. for CCT nr 151000 uploaded firmware can be up to 6.21.
- Since fw. 6.30 PBX verifies how long device was working. For 14 days since first start works as startup. After that it goes into normal mode. In this moment specific month and year of firmware publish is set, which can be uploaded into PBX without any additional licenses. Example, PBX started working from 24.08.2017, after 14 days, at 07.09.2017 PBX goes into working mode (SDN key saves installation date). Firmware release date that can be uploaded to PBX is set to 09.2018. Checking fw. version can be done in colum Firmware publish date on table from this site - [Serial numbers for firmware version](#)

When updating firmware with required purchase of license, during update (look at chapter: [Uploading PBX firmware](#)) warning will be displayed:

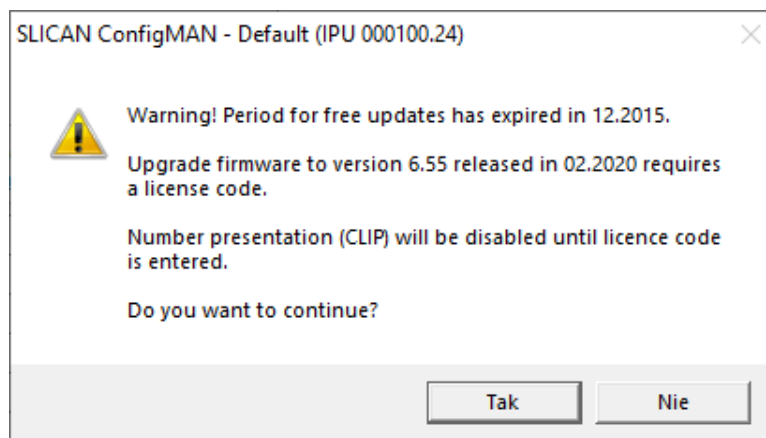


Fig 19.1: Warning about requirement of license purchase during update of PBX firmware

License is issued for specific serial number and model of PBX. Lack of license doesn't block device but introduced restrictions that disables number presentation (CLIP) for internal and external calls. Instead of proper presentation, message „lack of license” and number „0000” will be displayed on CTS phone, analog phones will display „0000” number.

License can be added in sheet Odpowiednią licencję należy wprowadzić w arkuszu **Global settings / Licenses/ PABx firmware** field **LicFrm**.

19.2 CTI software (MessengerCTI/ WebCTI)

CTI software license also allows to service subscriber with CTIP protocol and XML protocol subscribers modules.

Without CTI license, each PBX subscriber have access to basic services with *WebCTI*. By login to their subscriber account using web browser they have access to: basic info about their extensions number, enforced limits and calls fees, blocking phone with services, calling account, alarm clock, forwarding calls, MobilePhone forwarding, DND, I am here download *MessengerCTI.Desktop* application.

License code is entered in field **LicCTI** in sheet **Global settings/Licenses**. Entering license code causes unlocking PBX to using application according acquired license code. Information about maximal number of CTI user, according to acquired license code is displayed after entering code.

ATTENTION!

Every subscriber with CTI license enabled can use all features of WebCTI. They also have Windows application available: [MessengerCTI.Desktop](#) and Android [MessengerCTI.Mobile](#) – depending on license type (CTI.user, CTI.user.Plus)

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**.

CTI.user license allows to use of applications:

- *MessengerCTI.Desktop* – VoIP and video camera preview disabled
- Modules of Calls/Service/Term XML protocol and CTIP protocol,

CTI.user.Plus license allows to use:

- *MessengerCTI.Desktop* full functionality – VoIP and video camera preview enabled
- *MessengerCTI.Mobile* full functionality – VoIP and GSM calls
- Modules of Calls/Service/Term XML protocol and CTIP protocol

ConfigMAN can't assign CTI permissions to more user then license allows. To login applications *MessengerCTI.Desktop/ MessengerCTI.Mobile/ WebCTI* to PBX, they require subscribers number and password defined in sheet **Subscribers / CTI settings**.

19.3 ConsoleCTI

Functionality added in fw. 6.22

After entering license code possibilities of using application according this license is unlocked. Information about maximal number of [ConsoleCTI](#) **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**.

License must be added in sheet **Global Settings/Licenses/ConsoleCTI** field **LicConsoleCTI**.

19.4 TouchCTI i TouchCTI.VoIP sfotware

This functionality is not supported from version 6.20.

19.5 PayMAN software

This functionality is not supported from version 6.20.

19.6 TAPI workstations

Starting from firmware version 4.02 interface **TAPI** is available. Starting from version 4.07 in PBX there is accessible also TAPI for server.

Interface tapi is used to communicate with all *Microsoft* applications using this protocol. 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.

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 / CTI settings**.

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**.

Password for TAPI server should be defined in field **TAPIPas**, sheet **Global settings/Remain**.

19.7 VoIP card channels and subscriber ports

Number of possible VoIP subscribers and used VoIP channels is defined using license code.

Since fw. 6.50 DEMO isn't available. .

Each PBX may have up to maximal amount of VoIP subscribers:

- 100 subscribers for PBXes IPL-256, IPM-032 (except low cost - LC), MAC-6400, CCT-1668, CXS-0424
- 64 subscribers for PBXes IPU-14.102 i IPU-14.106
- 24 subscribers for PBXes IPU-14.101, IPU-14.103 i IPU-14.105
- 16 subscribers for PBXes IPS-100 i IPS-101
- 8 subscribers for PBXes IPM-032.LC, IPS-08.104 i IPS-08.105,

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/ VoIP**. 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 of licensing. Depending on license code and type of VoIP card following VoIP channels can be used:

- IPL-256, IPM-032 – up to 64
- MAC-6400 - up to 32
- IPU-14.102 and IPU-14.106 – up to 16
- IPU-14.101, IPU-14.103, IPU-14.105, IPS-08.100, IPS-08.101, CCT-1668, CXS-0424 - up to 8
- IPM-032.LC, IPS-08.105 – up to 4
- IPS-08.104 – up to 2

ATTENTION!

Due hardware revisions amount of available VoIP subscribers and channels for each PBX should be verified with current Technical Documentation..

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 more channels.

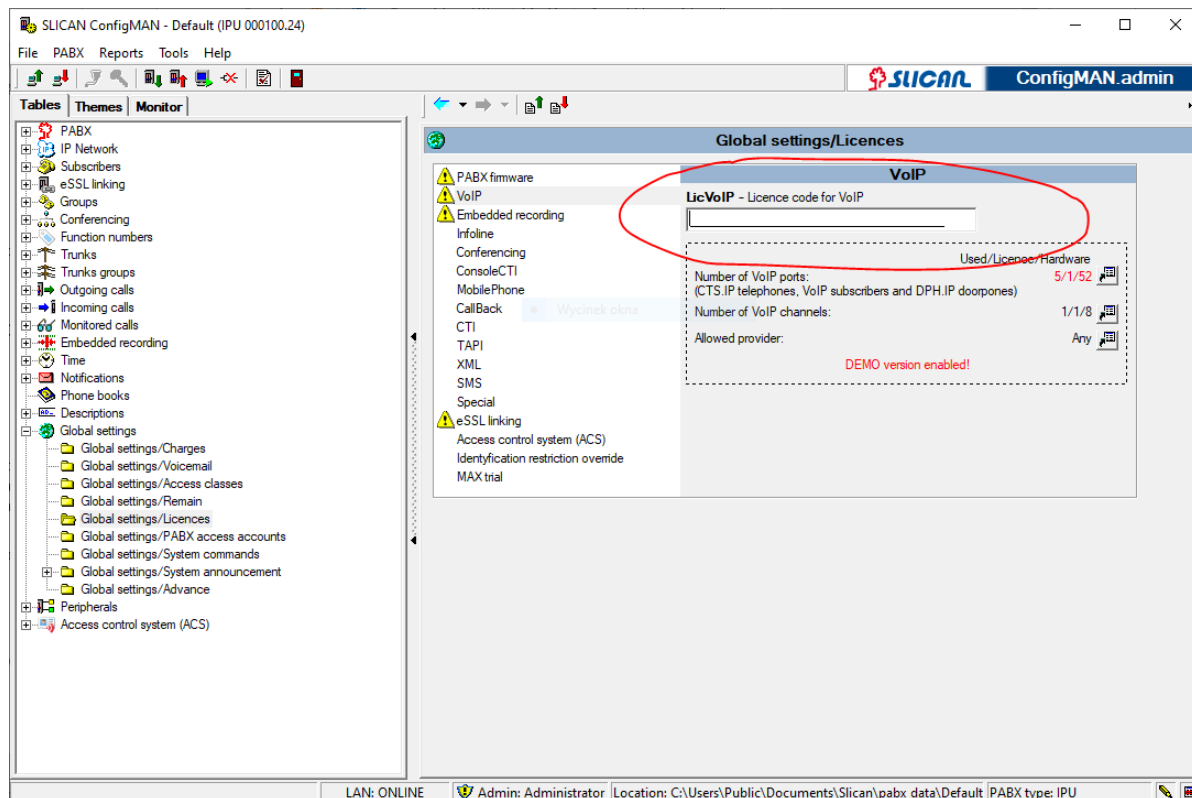


Fig 19.2: Adding VoIP channels and subscribers licenses

UWAGA!

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 due to lack of licenses or resources).

19.8 License for embedded recording

Since fw 6.50 version enabling recording requires proper licenses (DEMO isn't available).

Subjects of licensing are:

- recording channels on call recording cards,
- account of **RecordMAN.server**,
- account of **RecordMAN.client** (single account is included with PBX, cost free)
- access to recordings from external applications FTP/SFTP,

Hardware – limited maximum amount of recording channels (number of simultaneously recorded calls) for each PBX:

- IPL-256, CCT-1668: **12** with SD card/ **24** with HDD
- IPM-032⁵: **12** with SD
- IPU-14, IPS-08⁶: **8** with SD card

⁵ - LowCost versions support 4 recording channels

⁶ - LowCost versions support 4 recording channels

- *MAC-6400* – up to **128** (8 *MAC.M-16REC* modules installed).
- *CXS-0424* - **8** (after installing any LAN or VoIP card)

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.

Licenses can be added in sheet ***Global settings / Licenses/ Embedded recording*** field **LicEbdRec**.

ATTENTION!

In monitor statistics there is possibility to check how many calls were no recorded, because lack of available recording channels – see chapter [5.5 Optimization of system resources needed for call recording](#)

19.9 Infolines and announcement license

Since fw. 6.50 infoline usage requires proper license.

It is possible to define maximal 99 infolines. Number of possible info-lines depends on acquired licenses.. Maximal time of all voice announcements is also licensed. *Without 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. It means possibility of acquiring license expanding announcements time from 20 up to 120 minutes. License number is entered in sheet ***Global settings / Licenses*** field **LicInf**.

PBX enable to record up to 99 announcements (both DISA/Infolines and private DND messages).

19.10 Number of conference participants,

Since fw. 6.50 conferences usage requires proper licenses (free license with 4 conference participants was removed).

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* 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:

<i>PBX type</i>	<i>Maximal number of participants (currently talking) in one conference,</i>	<i>Maximal number of simultaneous conferences,</i>
	<i>License</i>	
IPS-08	8	4
IPL-256, IPM-032, IPU-14, CCT-1668	12	4
MAC-6400, CXS-0424	30	10

Table 4: Hardware limitation concerning conferences

Enter proper license in sheet *Global settings / Licenses/ Conferencing* field **LicCnf**.

19.11 MobilePhone license

Service available since fw. 4.07.

Since 6.50 PBX firmware version using any MobilePhone any subscriber requires a proper licence key (cost free license for 1 user was removed).

Since above firmware MobilePhone license is permanently linked to Callback functionality. Detailed information can be found in chapters: [MobilePhone](#) oraz [CallBack](#)

Enter proper license in sheet *Global settings / Licenses / MobilePhone* field **LicMbIPh**.

19.12 CallBack

Available since fw. version 4.07.

Since 6.50 PBX firmware version CallBack licence has been removed and merged with MobilePhone licence.

More information concerning this functionality can be found in chapter [CallBack](#)

19.13 XML.IVR license

Available since FW version 4.11 (5.11 for CCT).

Od wersji fw. 6.50 dostęp w wersji DEMO na okres 30 dni został wyłączony.

License 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>.

XML.CDR license (added with fw. 6.42 version) allows to connect and observe XML stream for extracting information about PBX events eg. ringing up, putting on hold, parking, disconnecting and halting recording. Details are available after registering on <http://sdk.slican.pl>.

Enter proper license in sheet *Global settings / Licenses / XML* field **LicXml**.

19.14 SMS license

Available since fw. version 6.20.

Defines amount of GSM trunks within PBX that can be used to send and receive SMS messages with *TelefonCTI*, *SenderSMS* and third party applications. Concerns settings form sheet *Trunks / GSM settings*.

Enter proper license in sheet *Global settings / Licenses / SMS* field **LicSMS**.

19.15 eSSL linking license

Available since fw. version 6.40.

Field **LicESSL** must have proper license key – only for PBX with fw. ≥ 6.40 , that will be linked to other PBX (eSSLv1 or eSSLv2 protocol). 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 and system phones will display information about lack of license.

Enter proper license in sheet *Global settings / Licenses / eSSL linking* field **LicESSL**.

19.16 Access Control System (ACS)

Available since 6.52 version of firmware.

Amount of users that can be assigned to system is licensed. Available in sheet *Global settings/Licenses/Access Control System (ACS)* field **LicACS**.

If ACS system is working on linked PBXes, license must be purchased separately for each PBX with ACS. Each PBX z turned on ACS must have set amount of users (sheet *Access Control System(ACS)/Subscribers*).

Example: 4 PBXes are linked, 2 of them are using ACS system. Licenses must be purchased only for those 2 PBXes. For each of them, numbers of system users must be estimated. Licenses must be purchased according to specific PBX (don't add up users of complete system).

19.17 Time licence – Auto MaxTrial

Newly purchased PBX, since firmware 6.50 have their DEMO licenses and functionalities (infoline, conferences, mobilephone, 1 VoIP subscriber and channel, 1 recording channel, 30 day usage of CTI application). Instead time license AutoMAXtrial was added. After enabling this license user will be able to try, test all licensed functionalities before purchasing specific license.

License can be enabled only one-time, by technician, person providing technical support for PBX in sheet *Global Settings / Licenses/ MaxTrial* field **MaxTrial**. License is valid for 30 days after activation..

ATTENTION!

For PBXes which had firmware version lower then 6.50 when they were purchased and updated to 6.5x at later date – they still conform to previous rules eg. free functionalities and DEMO license for CTI are active. Access to AutoMaxTrial is blocked.

License expiration time is visible on toolbar, within main window of **ConfigMAN** application. Expiration date is notified twice to people from sheet **Notifications/Service alerts** ([Wiki article](#)).

19.18 Time license – TRIAL

License provided by manufacturer in case of PBX issue or for purpose of testing different licensed functionalities by client before final purchase of normal license.

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 remaining licenses.

License expiration time is visible on toolbar, within main window of **ConfigMAN** application. Expiration date is notified twice to people from sheet **Notifications/Service alerts** ([Wiki article](#)).

19.19 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 to 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), infoline, recording calls, as well there is no possible to establish VoIP calls.

ATTENTION!

*PBX **locks** above functions after removing sub module during operation. After installing hardware device number during PBX operation alarms are not deleted, until reset. PBX reset verifies hardware device number.*

20 Replacing cards in PBX – activating (deactivating) cards,

20.1 Accepting a new card

- turn the PBX off and disconnect 230v power supply,
- 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.

20.2 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.

21 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:

- 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**, **BillingMAN**, **RecordMAN** and using CTI application.
- 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.
- 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	Kodeki VoIP			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 analog standard and assures the same quality. In practice for Slican PBX it should be assumed that total bandwidth for establishing single call (to occupy one channel on card) equals **110kb/s** 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:

- CTS.IP system phones,
- SIP phones – any vendor VoIP phones using SIP protocol,
- computer software – so called softphone supporting SIP.

ATTENTION!

All PBX Slican services are available from system phones CTS-202.IP. While using IP phones of other manufacturer or soft phones some services may be realized improper, ex. call transfer (FLASH) or three way call.

For PBX CCT-1668 network card is inserted into port marked as LAN slot.

2. CCT-1668 - PBX firmware => 5.11slot

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. To increase capabilities, it is possible to install 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 with 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**, **BillingMAN** 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. Allow to download recordings using **RecordMAN** software,

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. PBX (despite of its type) can be installed only one VoIP card.

4. IPL-256 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 EbdRec 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 modules mentioned above:

- **IP Network / LAN Interfaces** – for configuration of all mentioned above modules,
- **IP Network / WAN Interfaces** – for configuration cards with WAN port,
- **IP Network / DNS Configuration** – for translating domain names by PBX,
- **IP Network / POE Configuration** – for PoE card setting – powering, supported interface, IP addressing
- **IP Network / VoIP Configuration** – for VoIP parameters configuration – signaling port, voice ports, network location of VoIP phones,
- **IP Network / DHCP server** – enable DHCP server on LAN interface with configured address pool

- **IP Network / Firewall** – for configuration of PBX internal firewall
- **IP Network / Router** – for configuration of PBX router while working in LAN+WAN mode.

Additionally sheet **IP Network / LAN Interfaces** is located in sheet **Themes**, group **Basic settings**.

PoE card configuration

In sheet **IP Network/PoE Configuration** appropriate settings should be entered in following fields:

- **Cmt** – port comment
- **Pwr** – port power supply can 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.

22 VoIP functionality in Slican systems

22.1 Basic information

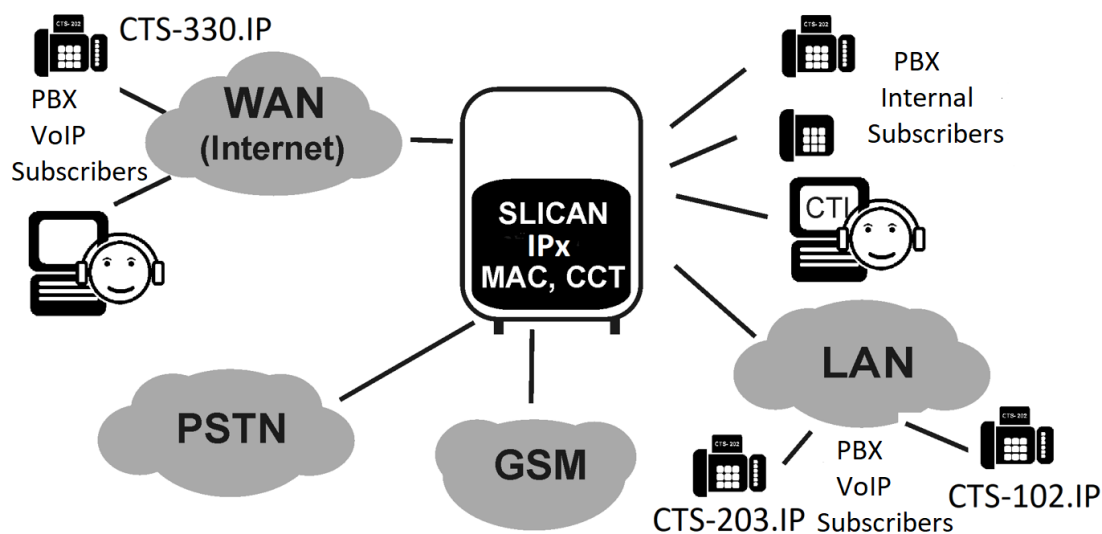


Fig 22.1: Slican PBX with VoIP functionality – general view

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 CTS.IP system phones.

Number of VoIP extension number – in Slican PBXes it is possible to create two typed of VoIP extension numbers:

- **CTS.IP extension numbers** - ports destined **only** for system VoIP phones manufactured by Slican (system phone uses dedicated communication protocol),
- **VoIP extension numbers** – ports destined for IP phones of other manufacturers and softphones, which uses SIP protocol.

VoIP trunks – PBX can create logical VoIP trunks.

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 – Slican PBXes equipped in VoIP card can work in one of following mode when connecting to company network (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 versa, 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 those are available:

- **Keeper** – skierowanie wywołań do serwera Keeper na gniazdo LAN lub WAN,
- **VoIP** – skierowanie wywołań do operatora VoIP do gniazda LAN lub WAN.

There are are access rules for services that requires setting up **CTI** server on PBX – 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 and 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.

22.2 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).

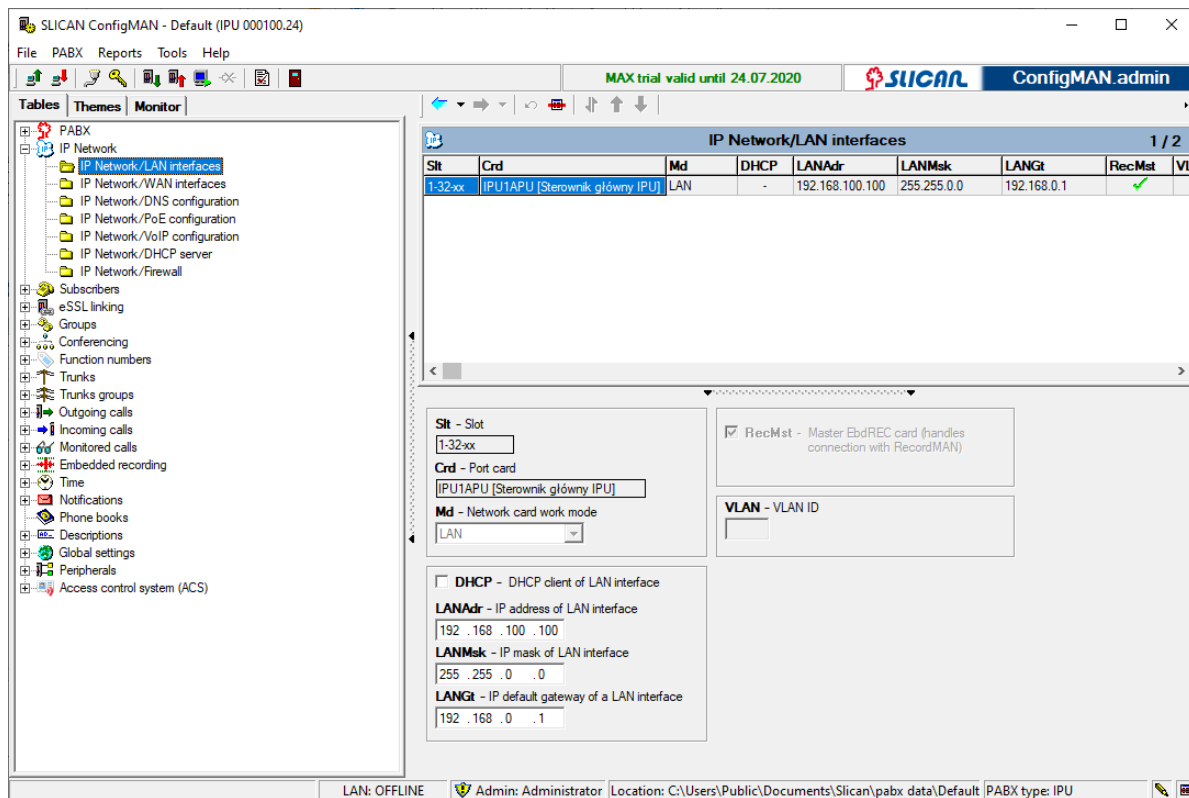


Fig 22.2: Sheet Devices/LAN card/LAN configuration

For IPx i CCT PBX settings are available within cpu controller, for MAC PBX on cpu controller and MACVoIP card.

Basic settings in this mode are:

- **LANAdr** – PBX (VoIP card) 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 and define subnet mask as 255.255.255.0 to further PBX configuration via LAN network.

Performing above activities enables to login to PBX through LAN network and use of CTI application.

In the next step from sheet **IP Network / DNS configuration** fields **DNS1** and **DNS2** should be entered. These fields define addresses of DNS servers, preferred and alternative, which enable connecting 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**. DNS addresses can be acquired from DHCP server after enabling field **AutDNS**. After LAN card installation DNS by default are set to addresses DNS servers belonging to Polish National provider **TPSA**.

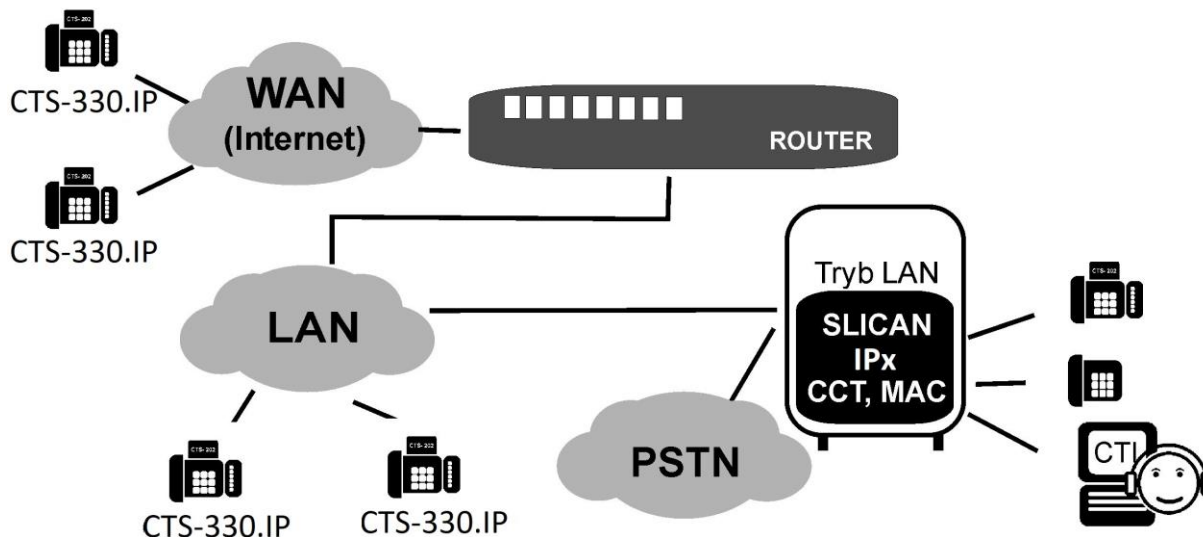


Fig 22.3: Working chart of PBX in LAN network - with VoIP calls establishing possibility

When using VoIP or CTS.IP phones outside of PBX local network or when VoIP trunk are configured as server (to connect another PBX, basic linking, eSSL linking)W przypadku, gdy korzystamy z telefonów VoIP lub CTS.IP znajdującymi się poza siecią lokalną centrali, lub jeżeli łączy VoIP centrali skonfigurowane są jako serwer (w przypadku łączenia innych central, linkowania prostego, linkowania eSSL) PBX must be configured so it will be sending it's public (external) IP address. This allows phone (PBX connecting as client) sending packets to eg. public address of router connected to Internet network. Configuration can be done from sheet **IP Network/ VoIP configuration**. Field **FwdPrt** have three settings:

- VoIP subscribers in my network (default) – use only when VoIP phone are on local network
- VoIP subscribers outside my network (auto) – if phones are located outside PBX local network, then PBX is sending public ather from sheet **Monitor** or system console after using command „show pubip”. This is public adres which PBX is using to connect to keeper server.
- VoIP subscribers outside my network (manual) – is used for network configurations which force using address from field **PubIP** sheet **IP Network/VoIP configuration**. Used when we want to use IP address different then used as default public address.

PBX working in LAN mode is located behind NAT (routerd), according this – to enable VoIP transmission, login to PBX to phones (PBXes) outside of network, **it is necessary that proper UDP or TCP ports to be forwarded.**

For particular protocols following ports are used as default:

- 5060, 8100-8300 UDP – for VoIP phones and trunks when linking eSSLv1 and eSSLv2,
- 5529 TCP – when linking with eSSLv2,

- 5522, 8100-8300 UDP - for CTS.IP system phones.

For ports 5060 and range 8100 – 8300 there is option for changing them in sheet **IP Network/VoIP configuration** (changing ports range 8100-8300 isn't possible for IPx PBXes with VoIP card).

Additionally it is important to assure QoS on PBX VoIP routers, especially transmission bandwidth, 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 in sheet **IP Network/VoIP configuration** into VLAN field, according information from admin.

22.3 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 two separate networks devices eg. to service provider device (ex. DSL modem,router) as well as local company network. Connection to LAN network enables to using CTI application, managing PBX via LAN network and establish calls with VoIP users in LAN.

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 network. To eliminate reasons of poor calls quality, separate dedicated Internet connection should be used to establish VoIP calls in Slican PBX

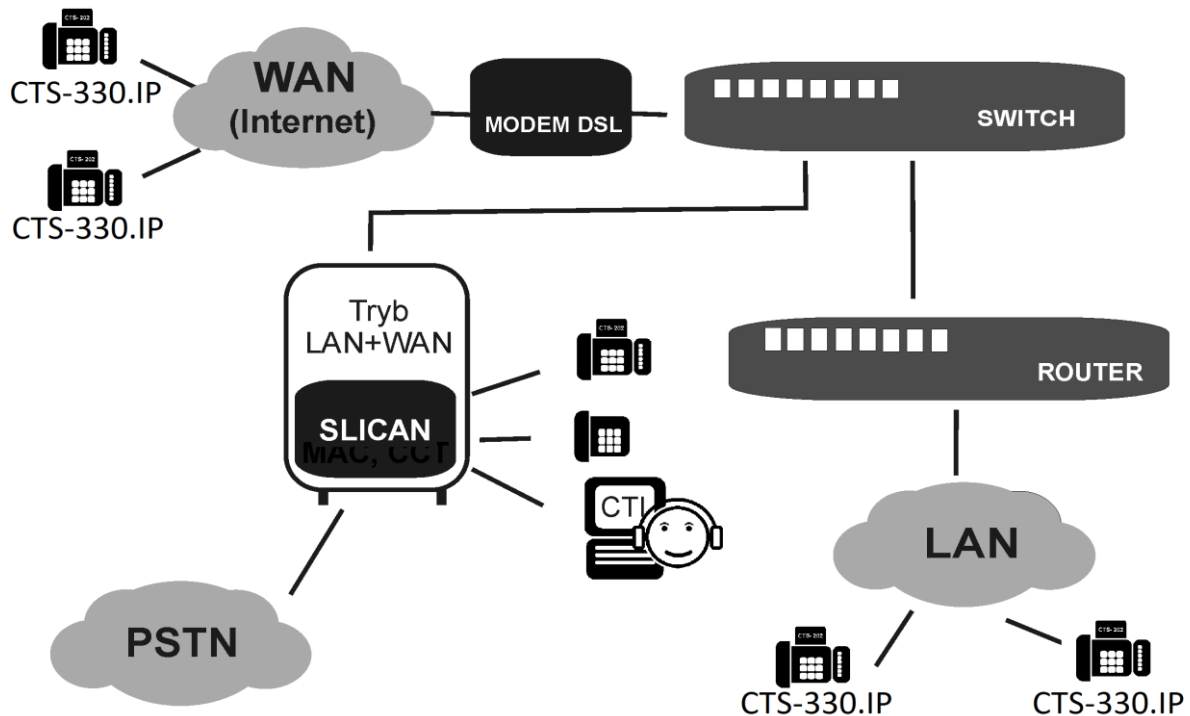


Fig 22.4: Sharing Internet interface in LAN+WAN mode

2. PBX using separated Internet interface for VoIP calls

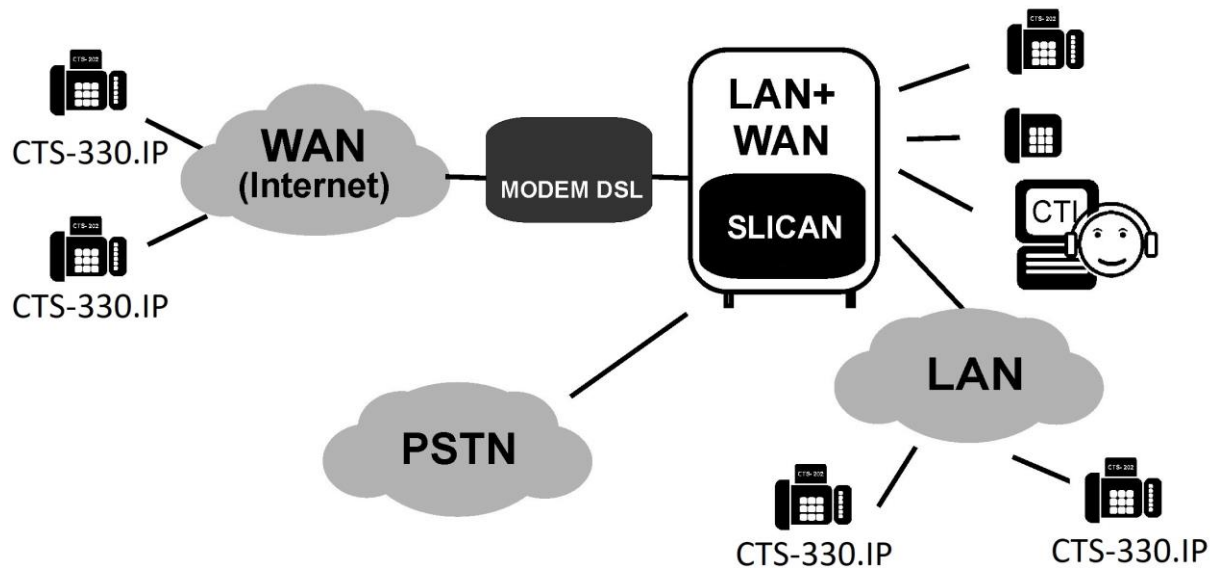


Fig 22.5: Using separate Internet connection to establish VoIP calls

To provide the best quality of service it is recommended to use dedicated Internet connection to VoIP calls establishing. With this configuration company's local computer network should use separate Internet connection.

3. PBX as 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:

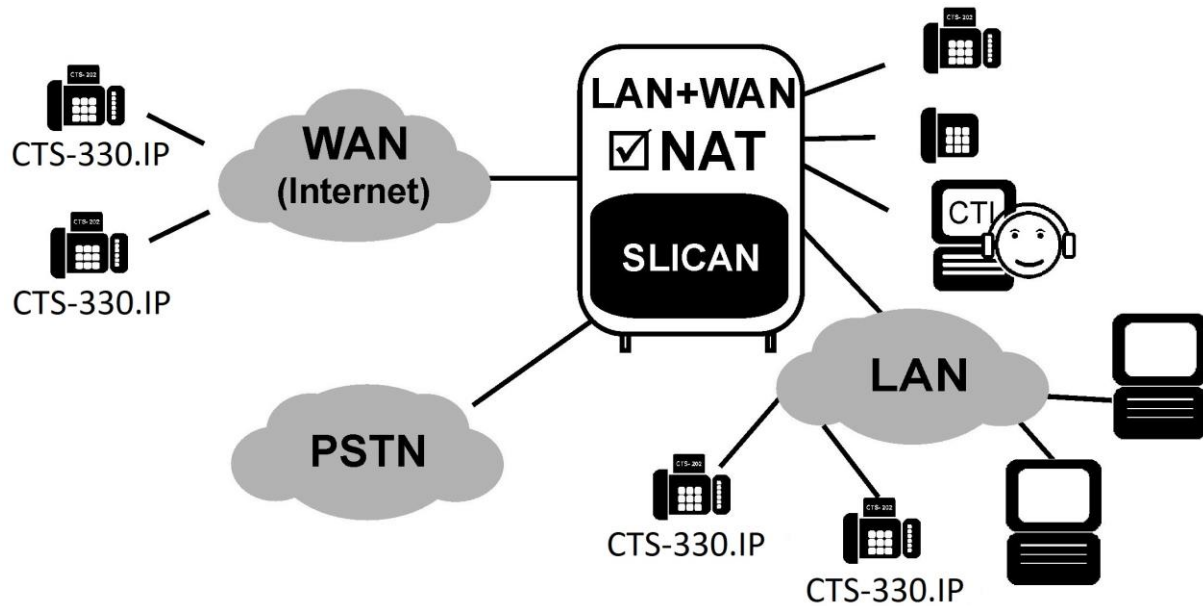


Fig 22.6: Working scheme for Slican PBX used as router

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** – PBX IP address in WAN network,
- **WANMsk** – subnet mask,
- **WANGt** – default gateway address.

After checking **DHCP** field above mentioned settings are downloaded from DHCP server.

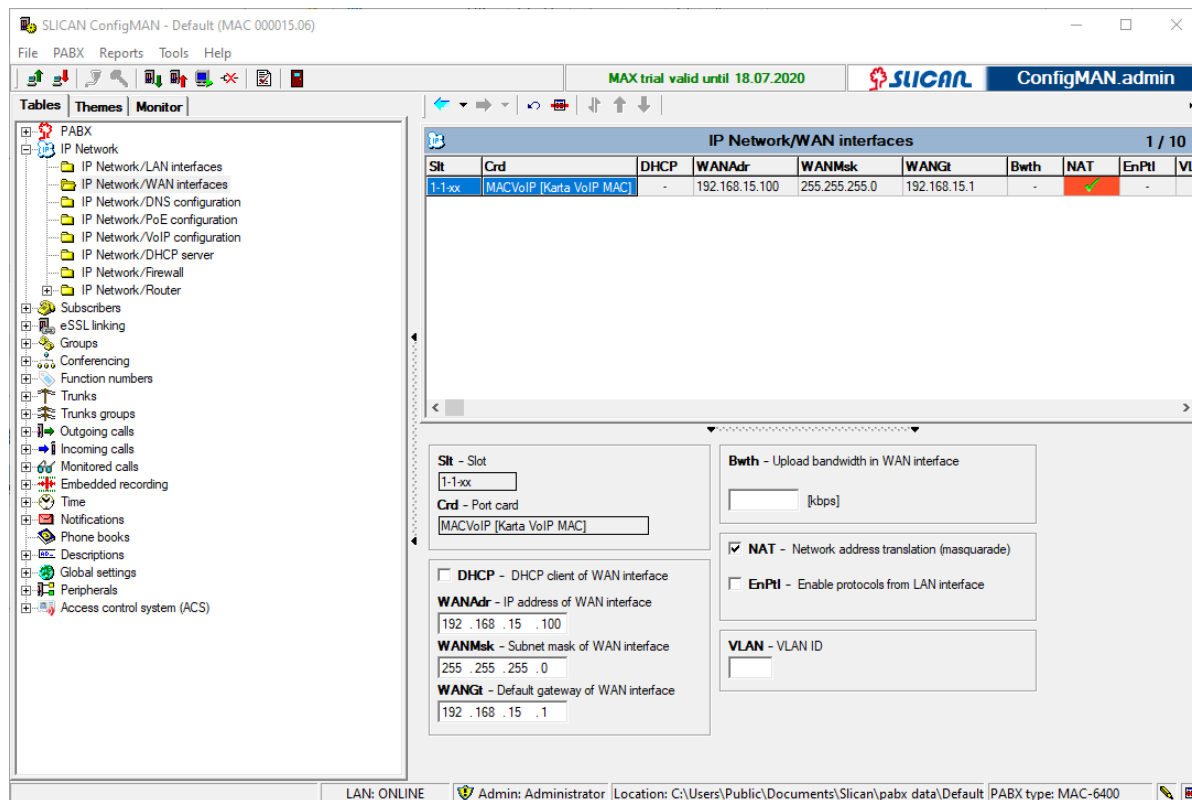


Fig 22.7: Sheet IP Network / WAN Interfaces – LAN+WAN (Router) mode

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 work within network which uses assymetrical Internet access (download higher then upload), within field **Bwth** we can enter Upload parameter.

If there for Wide Area Network are used VLANs for WAN port of PBX working, PBX can be assigned to one of them by entering appropriate number in field **VLAN**, according LAN admin suggestions.

Field **EnPtl** can be enabled, if we want to reverse use of LAN or WAN ports to connect local network through WAN and dedicated VoIP circuit through LAN port. Enabling field forwards ports (to WAN interface) that are used for applications **ConfigMAN**, **BillingMAN**, **RecordMAN**.

22.4 Basic methods of using VoIP functionality in Slican PBXes

22.4.1 Operation with VoIP extension number

There is possible to establish following calls for VoIP extension numbers in Slican PBXes:

- wewnętrzných – internal – between themselves and with regular extension numbers (analogue, ISDN, CTS system phone),
- outgoing – to PSTN, GSM, VoIP networks,
- incoming – from PSTN, GSM, VoIP networks.

To make calls, follow these steps:

In Slican PBX:

1. Allow VoIP functionality – on controller (IPU, IPM, IPU, IPS, CCT) or install and accept new VoIP card (IPL, IPM, MAC). Add VoIP licenses in sheet **Global Settings / Licenses / VoIP**. In sheet **Centrala / Cards** choose controller (IPx, CCT) or VoIP card (MAC) within menu **VoIP channels** set up amount of channels that be used by:
 - VoIP subscribers (SIP phone, softphone like *X-Lite*, Zoiper),
 - Subscribers that use CTS.IP system phones,
 - VoIP trunks (or this example VoIP trunks there are not used).
2. Select VoIP card/controller working mode ([LAN](#) lub [LAN + WAN](#))
3. For VoIP subscribers - sheet **Subscribers / Main settings** allows to create new subscriber (amount depends on used license). Detailed configuration can be set from sheet **Subscribers / VoIP settings**.
4. For CTS.IP phones – from sheet **Peripherals / CTS / IP settings** search and add proper phones.
5. Plan and configure settings and service privileges for each subscriber.
6. Configure incoming (**Incoming calls / Settings**) 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 system phones you should:

1. Configure communication method with VoIP server (PBX).
2. Configure phone settings for local network where it will work.

ATTENTION!

Pełne informacje dotyczące konfiguracji znaleźć można w instrukcji instalacji i obsługi aparatów systemowych.

To use softphone as VoIP subscriber device following activities should be done:

1. Download from <https://www.zoiper.com/> freeware version of software *Zoiper* (available for both Windows and Linux) - this is only one example of available softphones.
2. Install *Zoiper* software on PC.
3. Configure softphone to work with PBX.

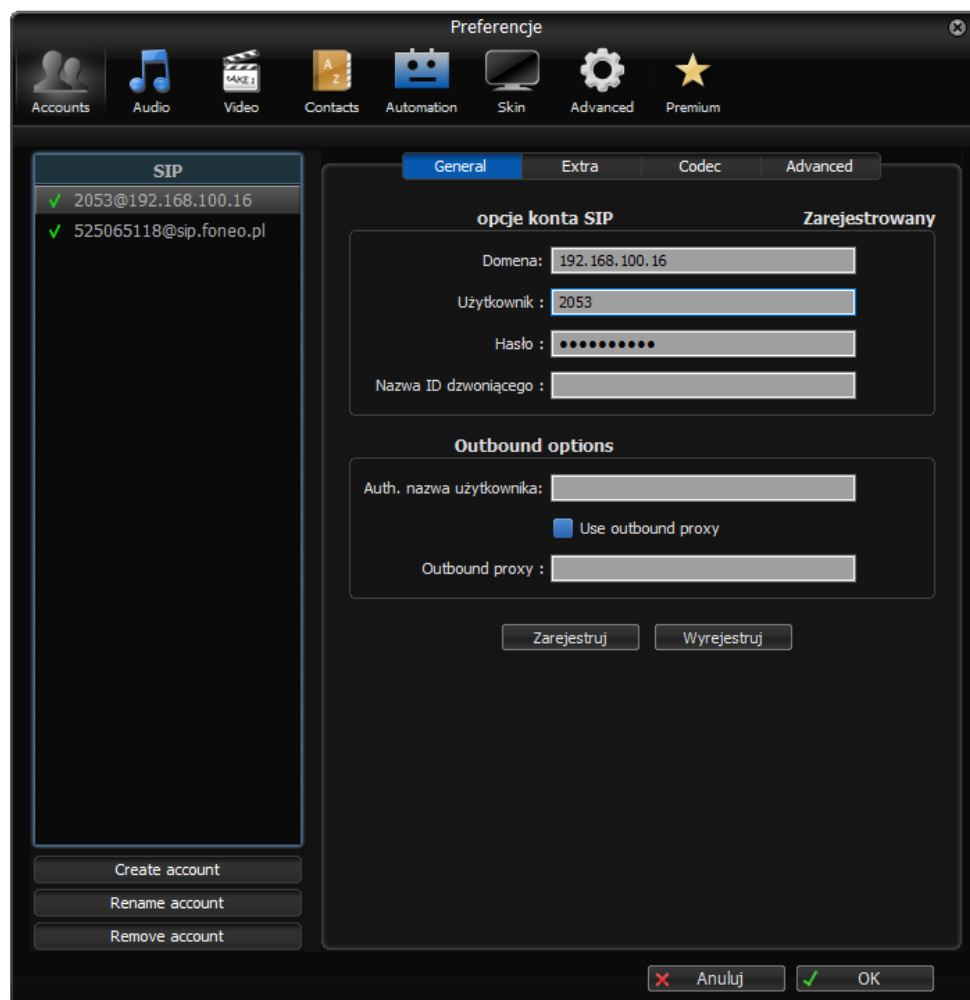


Fig 22.8: Zoiper software configuration

To configure Zoiper software provide following information:

- *Domain (domena)* – IP address of controller (IPx, CCT) or VoIP card (MAC) of PBX,
- *Username (użytkownik)* – extension number of VoIP subscriber,
- *Password (hasło)* – VoIP subscriber password, defined in ConfigMAN in sheet **Subscribers** / **VoIP settings** field **Pas**.

Within Extra tab set up: **Register during startup**.

ATTENTION!

Zoiper software is one of the free software that can work with SIP protocol. Below table present possible software which be used with Slican PBXes.

<i>Software</i>	<i>Used protocol</i>	<i>Operating system</i>	<i>Web page address</i>
X-Lite	SIP	Windows/Linux	http://www.counterpath.com/
Twinkle	SIP	Linux	http://www.twinklephone.com/
Zoiper	SIP, IAX	Windows/IOS /Android	http://www.zoiper.com/

22.4.2 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.

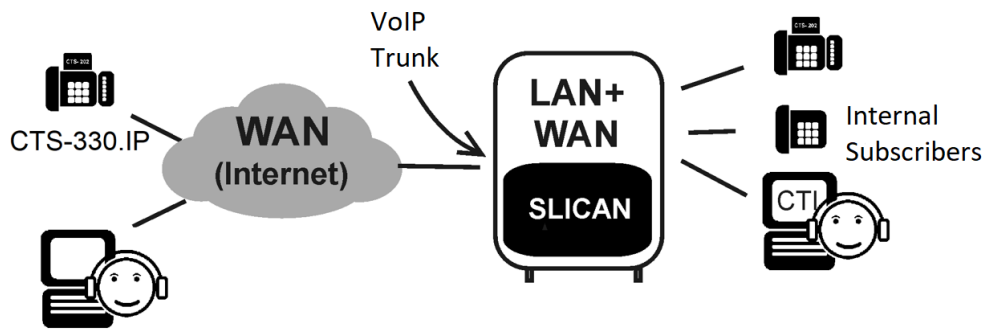


Fig 22.9: Calls from anonymous callers, routed via VoIP trunk

To enable communication between anonymous subscribers:

1. In PBX create VoIP trunk for incoming calls and 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:

- In sheet **PABX / Cards** choose controller (IPx, CCT) or VoIP card (MAC) form **Channels configuration** set up amount of channels that can be used by trunk,
 - In sheet **Trunks/ VoIP settings** choose protocole used for connecting calls (SIP),
 - Select audio codes
 - select DTMF transmission mode
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.

22.4.3 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).

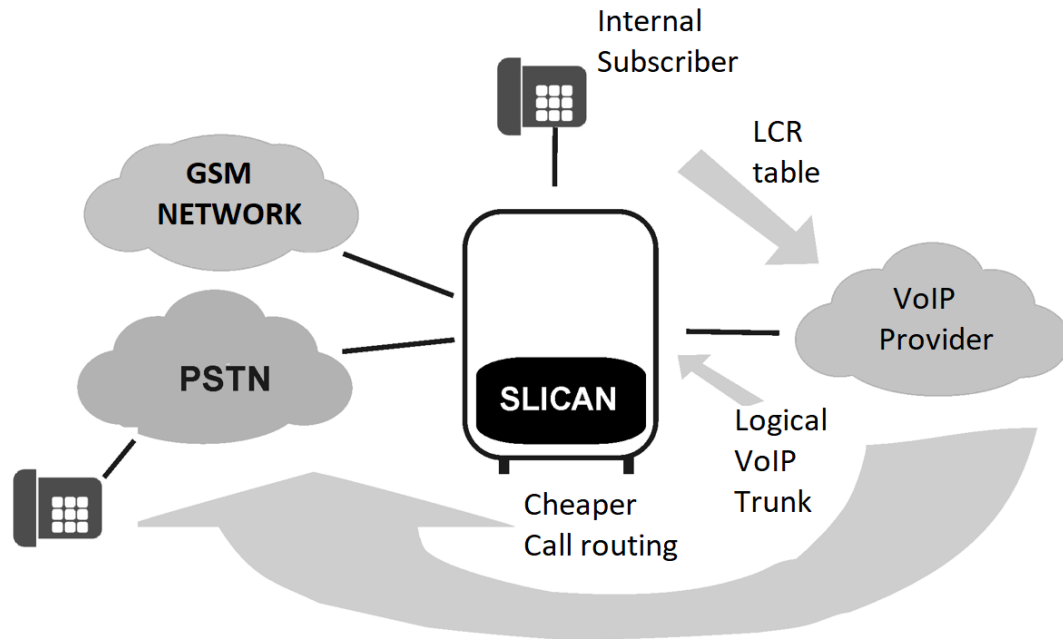


Fig 22.10 Cooperation of Slican PBX with VoIP providers

To establishing such calls VoIP trunk should be created and set up appropriate fields according information received from VoIP provider. Next step is table of outgoing calls which should be programmed to according needs. If from any extension number international number is dialed and appropriate entries in LCR table enabled to dial this number, call will be routed to VoIP provider which will establish connection to PSTN network.

To enable PBX connection 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).

23 PBX linking with eSSL v1

23.1 General information,

Linking of Slican PBXes can be executed using VoIP or ISDN trunk (PRA or BRA). For E1 trunk it is 300 meters and for 2B+D trunk maximal linking range is 1000 meters.

Linking protocols:

- eSSL (ISDN or VoIP) – full integration, necessary condition: all PBXes must use firmware version 6.20 or later.
- Simple linking (ISDN or VoIP) – older Slican PBXes or PBXes from third party manufacturers.

23.1.1 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 *MessengerCTI.Desktop/ MessengerCTI.Mobile* 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.

23.1.2 Assumptions

- eSSL (extended Slican Smart Link) 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 overlap.
- 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 have** different PBXes models 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)

23.2 eSSL network topology

23.2.1 Serial/Cascade

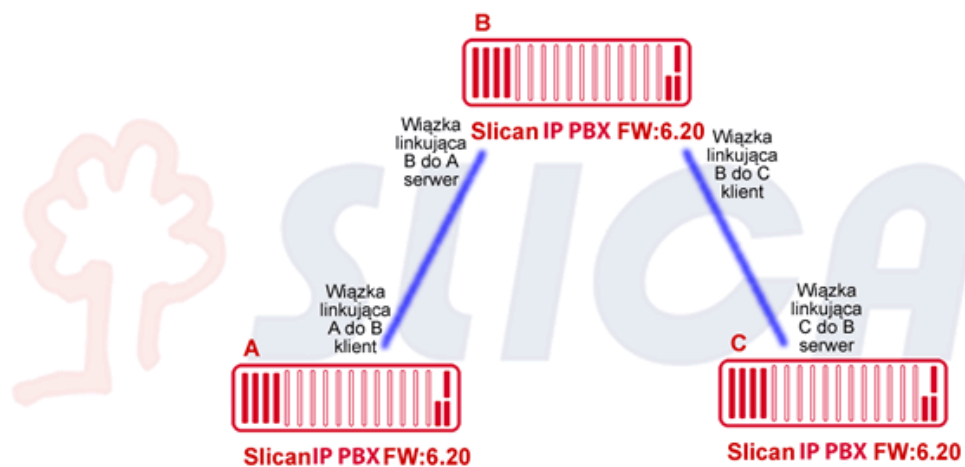


Fig 23.1: Serial topology (up to 1 transit PBX))

23.2.2 Each other topology

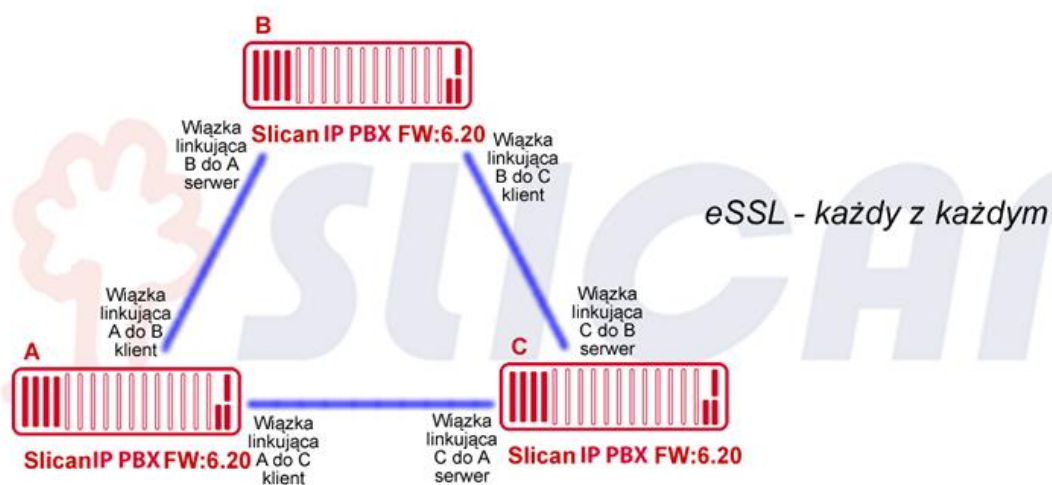
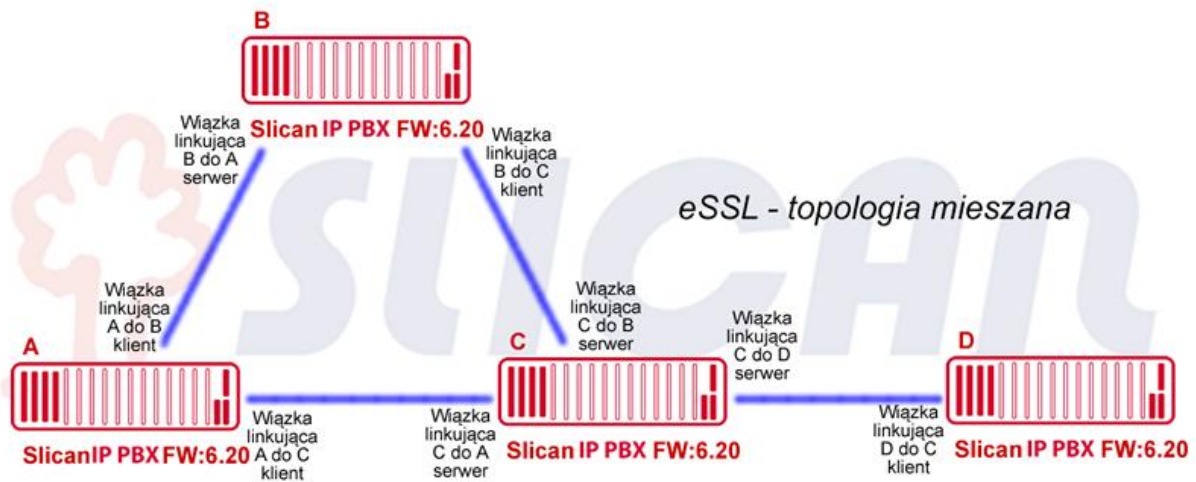


Fig 23.2: Each other topology (optimized channel usage)

23.2.3 Mixed topology



Ilustracja 23.3: Mixed topology

23.3 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 unambiguous comment.

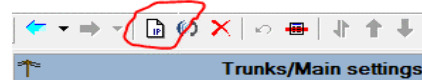
23.3.1 Connection via VoIP



Fig 23.4: For each eSSL connection (VoIP) – one PBX must act as server, second as client

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 unambiguous comment. Field **OwnNo.** remains 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.

23.3.2 Connection via ISDN

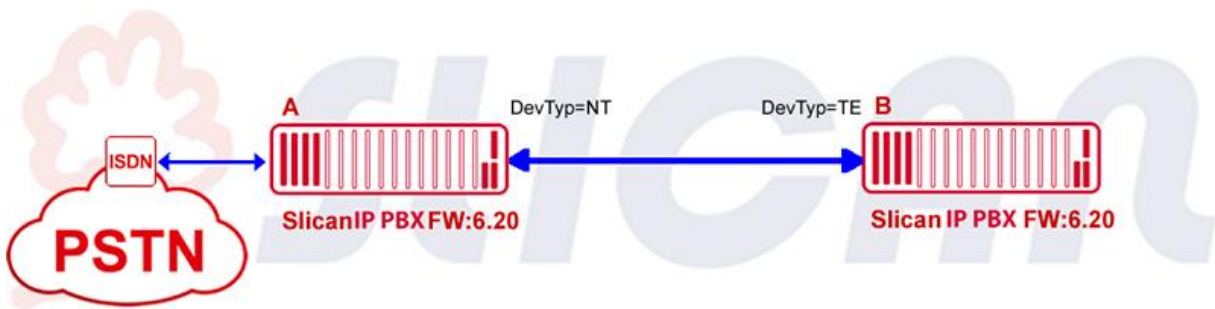


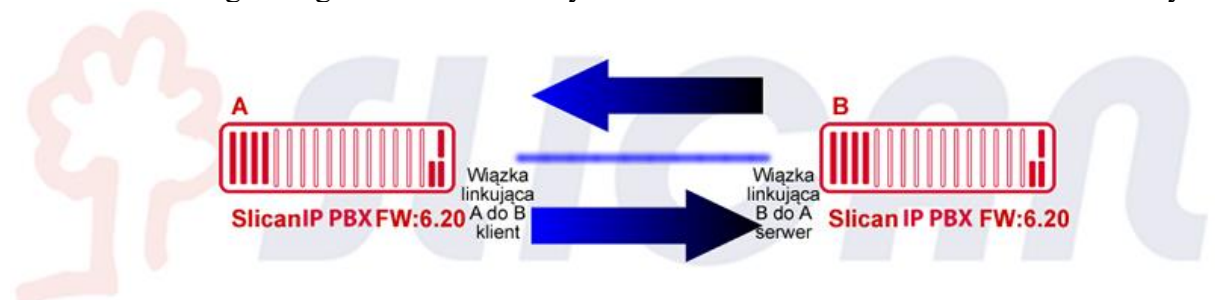
Fig 23.5: In every relation for eSSL (ISDN) – one PBX is NT device and second TE

1. Select ISDN port/ports, which will be operate as linking trunks (**Trunks/Main settings**). According good practice, it is recommended to assign it unambiguous 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.

23.3.3 Information synchronization in PBX network

After sending configuration to PBX a synchronization 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. Numbers not shown on phone book won't be synchronized and used with eSSL system.

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 is presented:

- PBX comments in field **PBXCmt**, in sheet *Global settings/Remain*
- **PBX** serial number
- Correctness status of linking trunk group **State**
- **Prg** – progress of eSSL number database synchronization.

eSSL linking state			
Linked PABX name	Linked PABX	State	Synchronized

ATTENTION!

Depending on PBX capacity eSSL PBX network, database synchronization progress may take several minutes.

23.3.4 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.

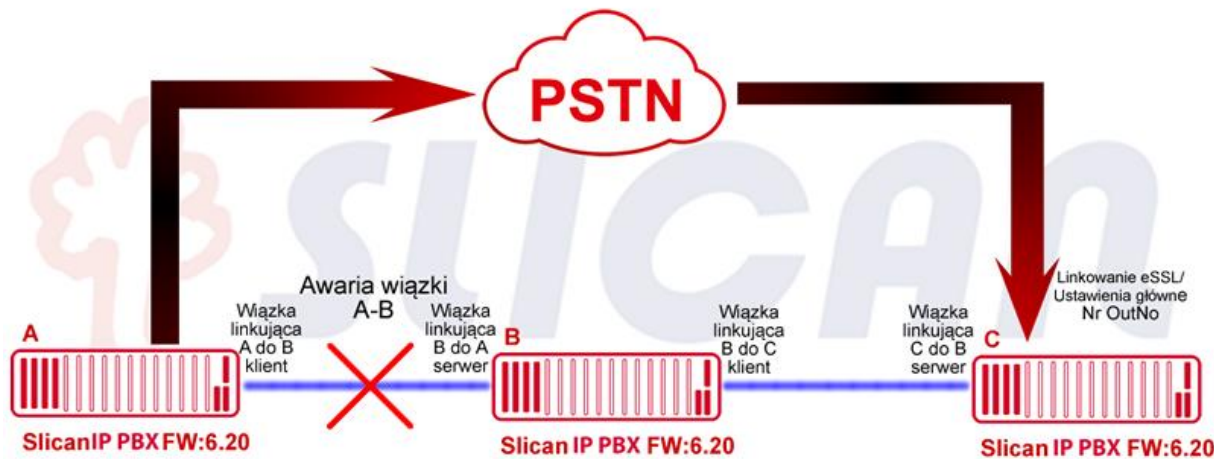


Fig 23.6: Alternative call maintenance, if connection with linking PBX is not possible

If it is foreseen to use alternative call maintenance in eSSL PBX system, field **OutNo** in sheet *eSSL Linking/Main settings* should be fulfilled with number used to establish call if there are some problems with eSSL link. Two types of alternative call maintenance are possible.

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**.

23.3.5 Automatic bypass 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 and C PBXes.
- If direct link A-C is damaged, A PBX will find bypass route via B PBX.

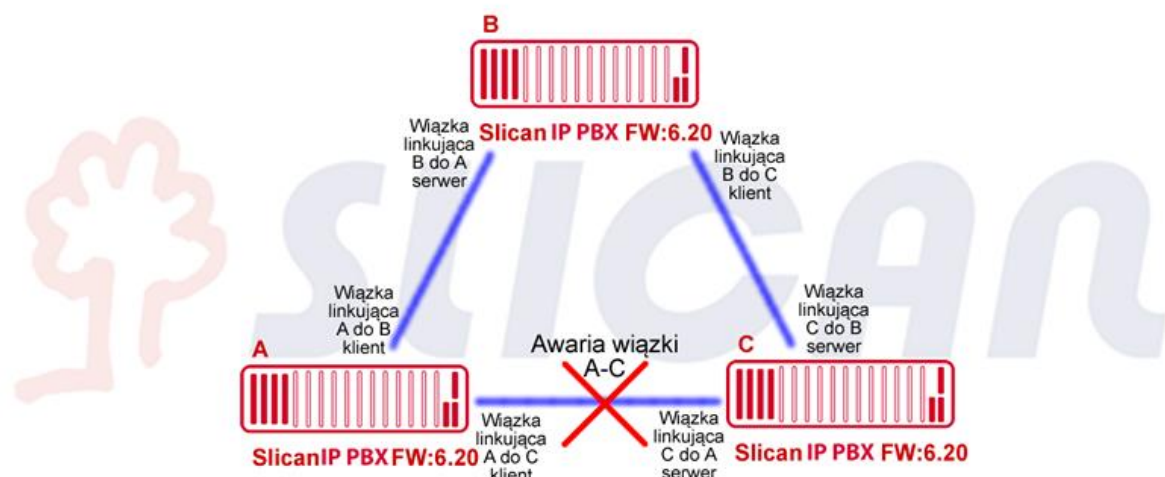


Fig 23.7: Automatic bypass call maintenance

This function shouldn't be configured – it works automatically.

24 PBX linking with eSSL v2 protocol,

24.1 Basic information

eSSLv2 system is an 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.

It should be noted that for eSSL v2 system:

1. Only single PBX acts as eSSL server, other PBXes are eSSL clients (configured in sheet **eSSL Linking/Main settings**), exchanging comments between PBXes is done with TCP protocol on port 5529 – network or internet connection between linked PBXes is required – sending phone book, chat, SMS
2. Create trunk between PBXes that will be used for phone calls – VoIP or ISDN (configured in sheet – general settings: **Trunks/Main Settings**, VoIP: **Trunks/VoIP Settings**, ISDN: **Trunks/2B+D Settings** i or **Trunks/30B+D Settings**) Use standard practices of creating normal trunks:
 - VoIP – trunk configured as client server (one side is PBX with trunk configured as client, another side of trunk is PBX with trunk configured as server). VoIP trunk with client-server configuration (single PBX may have many client or server VoIP trunks) with eSSL system client – server (only one PBX acts as server).
 - ISDN - trunk NT – TE with point-point configuration (on one side PBX with trunk configured as NT, on other side PBX with TE trunk)
3. eSSL system integration (point .1) with voice trunk (pkt. 2) is done using trunks groups (chapter : 24.4 eSSL with trunks groups) or directly within configuration VoIP (chapter: 24.5 eSSL trunks without trunk groups)

24.1.1 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.

- **Integration**

Subscriber statuses from PBXes included in eSSLv2 system can be presented on other system phone buttons assigned to subscriber and in [MessengerCTI.Desktop](#), [MessengerCTI.Mobile](#). 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 *MessengerCTI.Desktop/MessengerCTI.Mobile* within linked PBXes. Static and dynamic SMS messages routing is also available in eSSLv2 linking.

24.1.2 Assumptions,

- eSSLv2 (extended Slican Smart Link 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.
- eSSLv1 and eSSLv2 systems can't be used at same time.
- 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 edge 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 same serial number **cannot** be used.
- Time required for transfer statuses of all subscribers after establishing link takes up to 60 seconds
- Time required for exchanging comments between PBXes – up to 10 minutes (1 comment per second).
- Time required for exchanging public phonebook's records depends on its size and can take (during first synchronization) up to one hour.

24.2 eSSL v2 network topology

24.2.1 Serial connection (required)

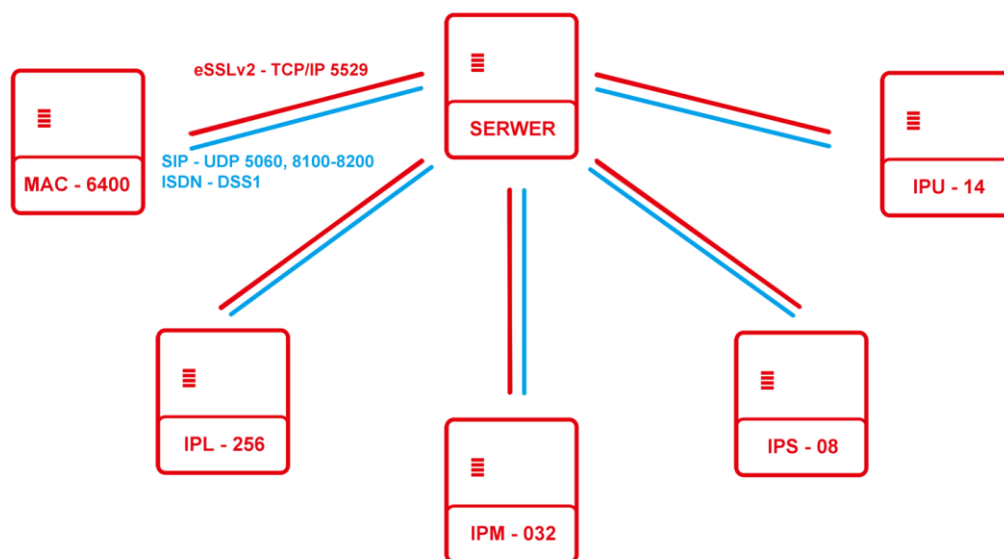


Fig 24.1: Serial connection (required) to PBX-server

24.2.2 Cross connections

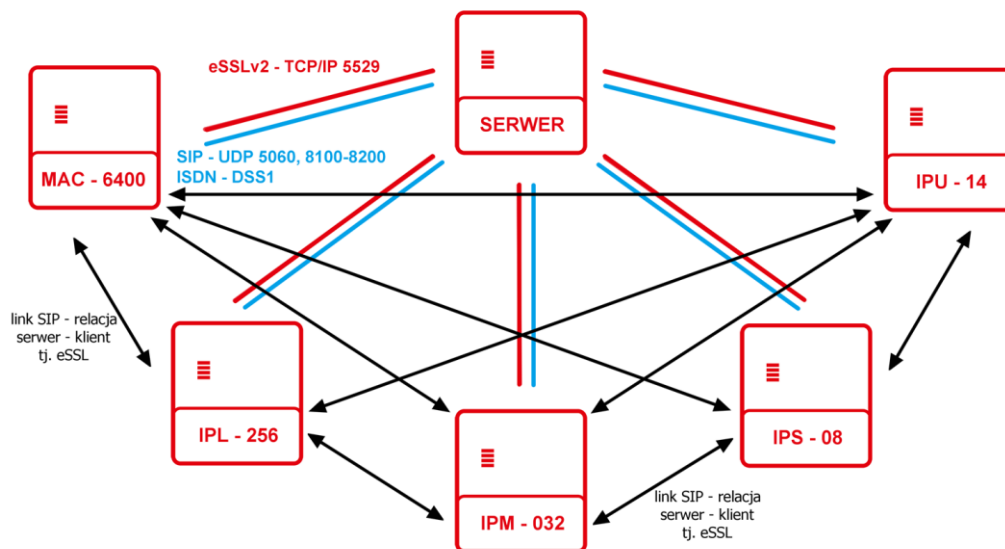


Fig 24.2: Voice cross connections between PBX-clients

Above example connections aren't required. Although, it secures PBXes against connection problems with PBX-server if one linking trunks group is damaged.

In this example, each linking trunk should be assigned to separate trunks group. Each trunk group will have different **PBXNo.** Used, sheet *Trunks groups/Linking*.

24.3 eSSL v2 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 server(only one server at same time) for whole system.
4. For server – we choose which elements (phone book, chat, SMS) will be synchronized in linking system).
5. For client – enter eSSL server controller card IP address.

24.4 eSSL with trunks groups

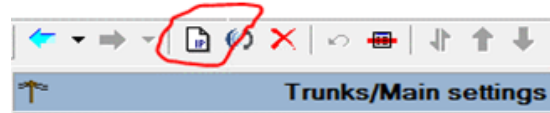
1. Create linking trunk group (for each PBX that will be linked/will have direct SIP or ISDN trunk) in sheet *Trunk groups/Linking* and define its type as **LnkTyp**→**eSSL**. According good practice it is recommended to assign unambiguous comment.
2. For PBX – eSSL server. In **PBXNo.** Field on selected trunk group, type serial number of the linked PBX (e.g. IPL000232.29).
3. For linked eSSL PBX, for trunk group connecting to eSSL server choose **Connection with server** for field **CnnTyp** .

4. In all cross connections between client PBXes (linking between clients), correct serial number “of the other side” is also required (**PBXNo.** field in **Trunks groups/Linking**), also choose **Connection with other client** on field **CnnTyp**.

24.4.1 Connection via VoIP

Create pair of trunks (client-server):

1. Create linking trunk (in sheet **Trunks/Main settings**). It is good practice to assign it unambiguous comment. Field **OwnNo.** remains empty.
2. Configure trunk **Trunks/VoIP settings** complete fields **OprMd/Adr/Usr/Pas** by client side and fields **OprMd /Usr/Pas** on server side. Remember 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** trunk should be assigned to earlier created linking trunk group.



From this moment eSSLv2 linking trunk group is ready to work.

24.4.2 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 unambiguous 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 or 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 v2 is ready to operation.

24.5 eSSL trunks without trunk groups

ATTENTION!

Described feature does not apply to ISDN linking. Trunk groups is required.

If features related to trunk groups does not interest us (call quantity limitations, trunks group recording, dynamic call routing, call charging), we can create VoIP trunk without assigned trunk group.

Create pair of trunks (client - server):

1. Create linking trunk (in sheet **Trunks/Main settings**). It is good practice to assign it unambiguous comment. Field **OwnNo.** remains empty.

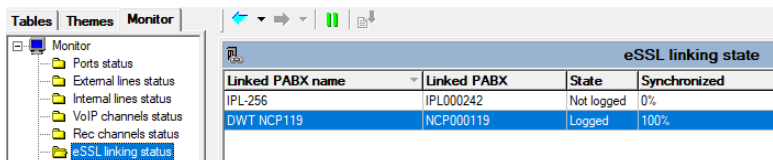
2. **Port status** – it is done automatically and refresh together with any change of linked port status.

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,
- progress of eSSL number database synchronization.



Linked PABX name	Linked PABX	State	Synchronized
IPL-256	IPL000242	Not logged	0%
DWT NCP119	NCP000119	Logged	100%

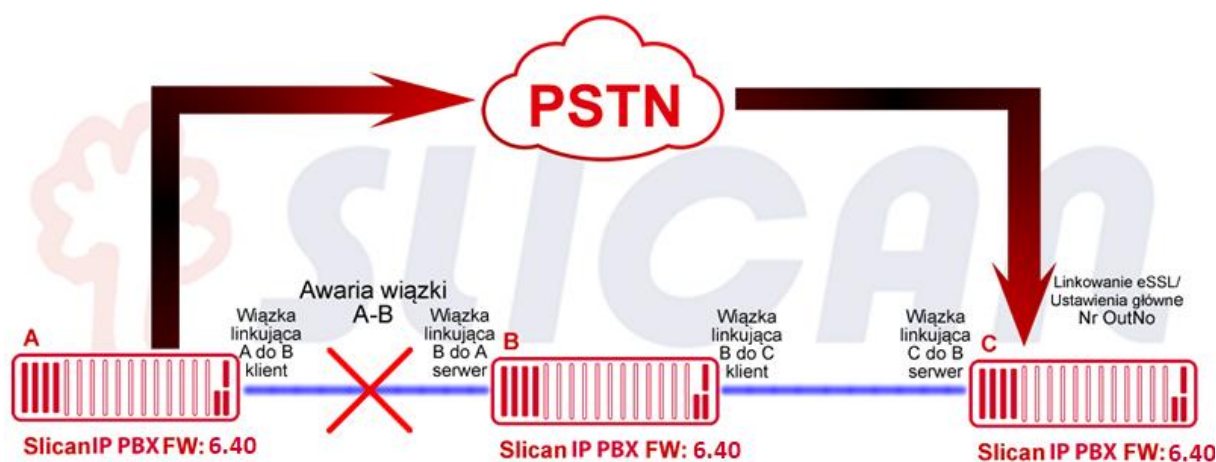
ATTENTION!

Depending on PBX capacity eSSL PBX network, database synchronization progress may take several minutes.

24.7 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.
- v.
- tego typu połączenia możliwe są **wyłącznie** w ramach sieci eSSL.



Goh 24.4: Alternative call maintenance if no connection to linked PBX is available

If it is foreseen to use alternative call maintenance in eSSL PBX system, field **OutNo** in sheet **eSSL Linking/Main settings** should be fulfilled 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**

24.8 Automatic bypass 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 and 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.

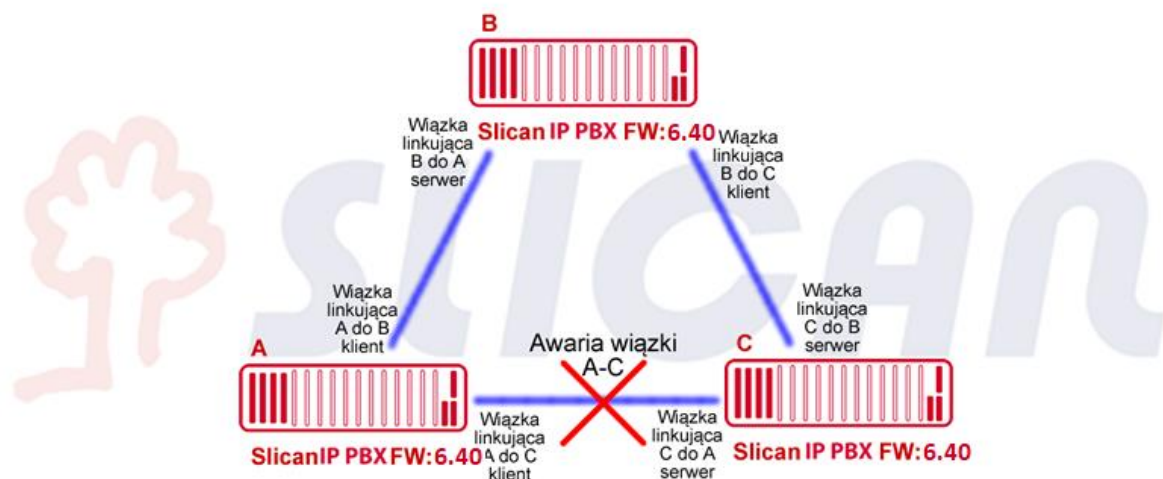


Fig 24.5: Automatic bypass call maintenance

25 Basic linking

While linking older 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.

25.1 Configuration of basic linking method

1. In sheet **Global settings/Advanced** switch on the field **LnkEn** – to show options concerning with basic linking.
2. (Linking account number should be outside of PBX numbering plan). This account represents privileges and registers calls of all subscribers from linked PBX.
3. **LnkTyp**→**Basic**. According good practice it is recommended to assign it unambiguous 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
 - **DgtDI** – number of digits, which should be send to linked PBX (starting from the right).

ATTENTION!

No matter regarding type 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.