Proxima PLATAN

IP PBX SERVER

Libra PLATAN

PBX SERVER

PART 3

ProximaWeb, LibraWeb

PROGRAM OPERATION MANUAL

rev. 2.12.01

If a more detailed description (supplement) can be found elsewhere in this document, the following symbols are used:

Note *(a symbol put in the margin)*

Additional information *(a symbol put in the margin)*.

Proxima IP PBX Server, Libra PBX Server
as well as PLATAN ProximaWeb, LibraWeb
are products manufactured by:
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Sopot, June 18th, 2020

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1. Introduction

We are pleased that you have chosen a telecommunications server offered by our company. The Proxima IP PBX Server and Libra PBX Server are PLATAN's products from the latest product line and state of the art IP telecommunications servers. They are characterised by high quality, a wide range of functions and services and easy operation.

Key features of Proxima IP PBX Server and Libra PBX Server:

- Embedded VoIP – IP Gateway, IP EXT, **T.38 support**.
- ProximaWeb, LibraWeb – web-based server management.
- Settings and call history accessible in User Zone via a web browser.
- Advanced VoIP diagnostics.
- Call recording ensuring security and higher quality of provided services.
- Multi Phone – up to 4 devices (including mobile phone) to one extension.
- See Who's Talking – video calls accessible to any number of users.
- Networking of up to 16 Proxima and Libra PBX servers.
- Global phone book with up to 3 000 entries for proprietary and IP phones.
- Conference calls, **conference rooms**.
- Intelligent Call Distribution (ICD) consisting in:
    - Interactive Voice Response (IVR) with multilevel call menu scenarios.
    - Possibility of distributing calls to user groups according to the preset criteria: **queuing**, uniform (UCD), according to the topic selected via IVR, automatically – based on the recognised CLIP number (ACD).
    - 99 voice announcements (up to 30 h) for DISA and IVR.
- Embedded Voicemail (25 channels).
- 4 melodies for waiting connections.
- ARS/LCR – automatic least cost routing.
- CLIP (Calling Line Identity Presentation) on all extensions.
- Call registration and billing – 100 000 billing buffer.
- Automatic fax signal recognition.
- Controlling external devices – automatically or from mobile phone.
- Platan Hotel embedded hotel software and hotel interface – support of external hotel software.
- Open PCTI protocol enabling the server integration with Call center, CRM systems and other applications and devices.
- VEK (VoIP Cost Eliminator).
- Call Through (transit connection with automatic authorisation).

We hope you will be perfectly satisfied with your purchase and we promise to provide professional assistance and information on our products.

In order to operate the PBX Server properly, please read this manual carefully and keep it for future reference.

The Proxima/Libra server manuals are organised as follows:

Part I – PBX Server operation and maintenance manual covering the following issues:

- important features of the server;
- structure and installation procedure of the server.

Part II – PBX Server user manual:

- functions and services provided by the server, divided into incoming and outgoing calls;
- programming of some server functions from the telephone;
- operation of digital proprietary phones, consoles and door phones;
- built-in voicemail;
- User Zone.

The majority of server functions are listed at the end of this manual in a shortened version.

Part III – ProximaWeb/LibraWeb computer program operation manual covering the issues concerning the program operation and the server programming from the PC. The server may be programmed by an authorised person only and this manual is available for Platan's authorised installers.

The server user manual as well as the operation and maintenance manual are available on Libra and Proxima webpages at www.platan.eu website. On this website the information on new products and changes can be also found.

NOTE. Due to safety reasons, it is recommended to connect Proxima IP PBX Server and Libra PBX Server behind a router.

2. Server programming

The server can be configured only by an authorised person using the ProximaWeb /LibraWeb application from the web browser or with Libra PC software..

The server programming process description can be found in this manual.

Some of the server functions may be programmed from the telephone via a dedicated **Programming mode**. This procedure is described in Part II – User manual.

2.1 Equipment requirements

The program is to be installed in a desktop or a mobile computer (laptop) meeting the following minimum requirements:

- Windows 7/8/10 or Linux operating system,
- 100 MB free space on hard disk,
- 128 MB RAM memory,
- network card,
- Internet browser with *Java* enabled or Libra PC software that can be downloaded from www.platan.eu. Java software can be downloaded from www.java.com.

The cable connecting the server to a PC is supplied with the server.

Windows 7 or a later operating system is recommended.

When using the server, particularly for the purpose of analysing and billing calls, the PC must be equipped with a printer.

2.2 Server connection to Ethernet (LAN)

The Proxima IP PBX Server and Libra PBX Server are configured via Ethernet using a PC and an Internet browser or Libra PC software. In order to connect the server to LAN, standard computer network wiring, preferably UTP or FTP, should be used.

The place of connecting the network cable on the servers' processor card is shown below.

Proxima IP PBX Server

After the server has been properly connected to LAN on ETH1 connector, a green LED will light up and a yellow LED will flash on the front panel:

POWER (green LED) – signals that the processor is supplied:
- emits continuous light – the processor is supplied,
- emits no light – the processor is not supplied.

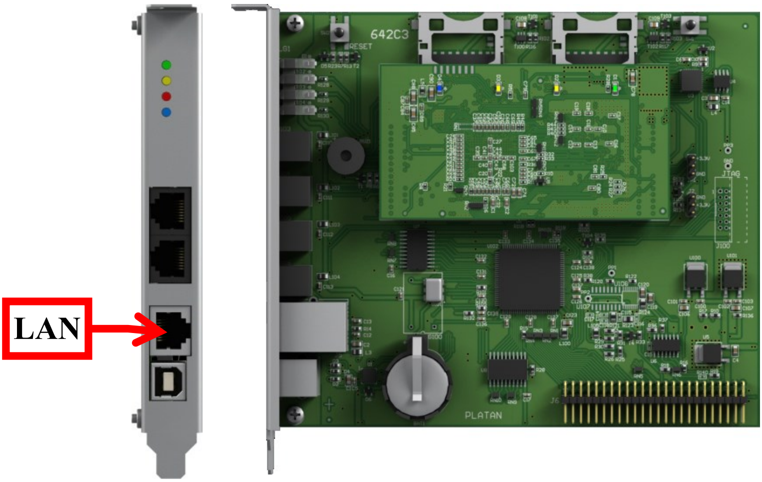
RUN (yellow LED) – signals the PROXIMA-PROC controller status:
- flashes 1.0 s/1.0 s – proper operation of the controller's processor: none server port is active,
- flashes 0.1 s/0.1 s – server operation: at least one of the trunk or extension equipment is active,
- emits continuous light – controller's processor is not working,
- emits no light – controller's processor is not working.

VoIP (blue LED) – signals the VoIP channels and ports status:
- flashes 1.0 s/1.0 s – proper operation; none of the registered IP ports is active (no ongoing VoIP call),
- flashes 0.1 s/0.1 s – at least one IP port (trunk or extension) is active (ongoing VoIP call),
- emits continuous light – controller's processor is not working,
- emits no light – no IP port has been defined or controller's processor is not working.

ALERT (red LED) – signals the incorrect Proxima server status
- emits continuous light – warning: PRA card red alarm, excessive errors number while saving/reading data from microSD card (all warnings and server operational parameters incorrect statuses are saved in the server operation report),
- emits no light – server is working correctly: no warnings, correct status of monitored server operational parameters.

2.2.1 Libra PBX Server

After the server has been properly connected to the LAN, a blue LED will light up on the front panel:



POWER (green LED) – signals that the processor is supplied:
- emits continuous light – the processor is supplied,
- emits no light – the processor is not supplied.

RUN (yellow LED) – signals the LIBRA-PROC controller status:
- flashes 1.0 s/1.0 s – proper operation of the controller's processor,
- flashes 0.1 s/0.1 s – server operation: at least one of the trunk or extension equipment is active,
- emits continuous light – controller's processor is not working,
- emits no light – controller's processor is not working.

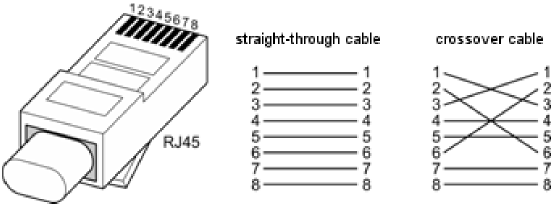
BATTERY (red LED) – signals that the Libra server is supplied in the emergency mode:
- emits continuous light – Libra server emergency supply, no 230 V mains supply,
- emits no light – 230 V mains supply.

ETHERNET (blue LED) – signals the presence of the LAN (ETHERNET) interface physical layer:

– emits continuous light – connection to LAN,
– emits no light – no connection to LAN.

2.2.2 Server connection if Ethernet (LAN) is not available

If LAN is unavailable, it is possible to connect the server to the PC network card. If the PC has the function of automatic cross-linking detection, a straight-through network cable supplied with the server can be used. Otherwise, a crossover network cable must be used:



2.3 IP address settings

The default IP address of Proxima IP PBX Server and Libra PBX Server: **192.168.1.250**.

2.3.1 Setting up the server IP address from the telephone

In order to set up the server IP address from the telephone, the telephone must be first connected. Next, enter the **mode of server programming from the telephone *708 "password"** (*default password: 12345678*), set the IP address and the subnet mask using the following codes:

41 "IP address" # – setting up the IP address, e.g. **41 192*168*1*195#** (the server will restart after programming).

42 "subnet mask" # – setting up the subnet mask, e.g. **42 255*255*255*0#** (the server will restart after programming).
43 "enable DHCP" – enable reading network settings from DHCP server.

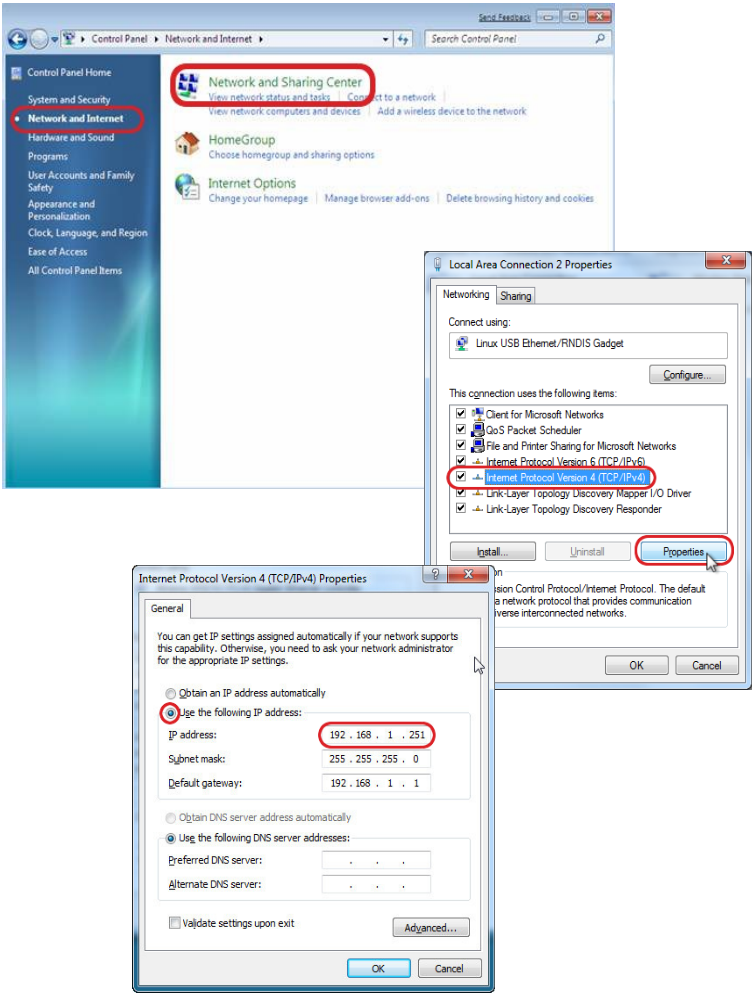
NOTE
708 is a default function code to enter the server programming mode. If the number had been changed (*Common settings → Function codes*), the proper one should be used.

NOTE
After entering the IP address from the telephone, the default port is 80.

2.3.2 Setting up the IP address in the PC directly connected to the PBX server

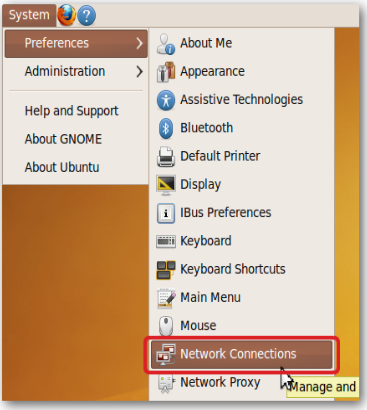
WINDOWS 7

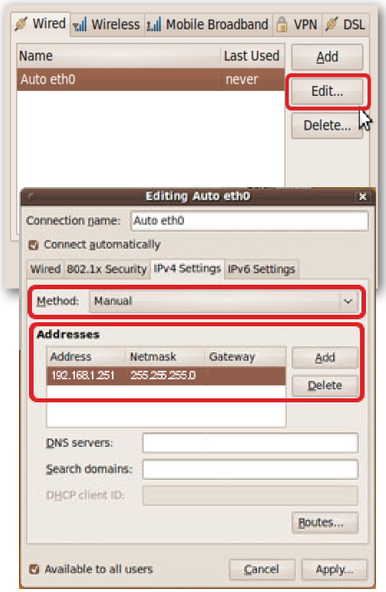
Select *Network and Internet* from the Control Panel and then select the right network from *Network connections*. Select the *Internet Protocol Version 4 (TCP/IP v.4)* from the network properties and enter the network parameters after clicking *Properties*.



LINUX Ubuntu

Go to *System* → *Preferences* → *Network Connections*, select the *IPv4 Settings* tab, change Method to Manual and enter the right network parameters.





The IP address must belong to the same subnet as the Proxima/Libra server and must be different from it, e.g.: IP address – 192.168.1.251
Subnet mask – 255.255.255.0

2.3.3 PBX server address is not compatible with LAN architecture

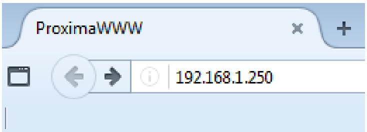
If the default Proxima/Libra server address (192.168.1.250) is incompatible with the architecture of the LAN to which the server has been connected, the server IP address should be changed from the telephone (section 2.3.1) or the server should be connected directly to the PC (PC IP address should also be changed – see section 2.3.2.) and the server IP address should be first of all changed in ProximaWeb/LibraWeb to a compatible one. Should you encounter any problems, contact your computer network administrator.

3. Logging into the PBX Server

In order to manage the Proxima IP PBX Server and Libra PBX Server, the Java environment has to be installed on the computer. The latest Java version you can download from www.java.com website.

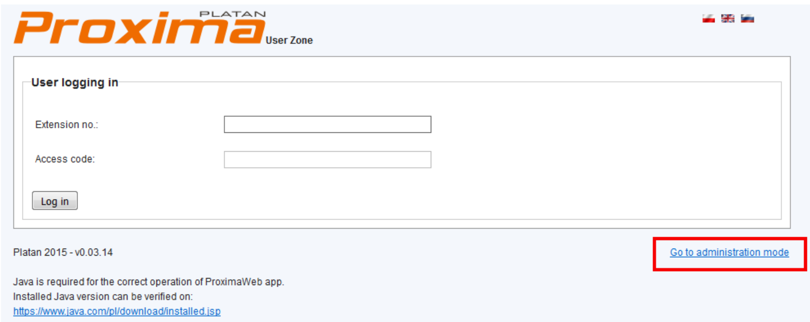
3.1 Connection with the use of web browser

The server can be configured after the 192.168.1.250 address has been entered in the browser address bar.



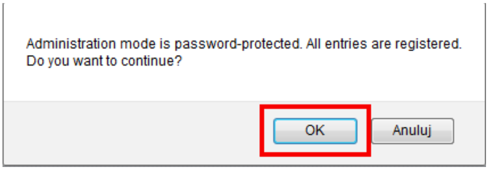
After the IP address has been confirmed, the User Zone (described further in this manual) login page is displayed.

Click the *Go to administration* link in the bottom right corner to access the administration page:



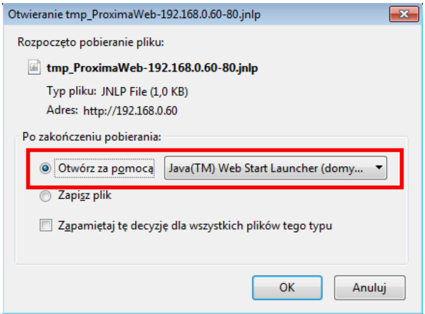
The system will inform you about the registration of all entries to the administration mode and will ask you to confirm your decision.

Note! It may be necessary to add the server IP address to the exceptions in Java security settings (*Control Panel* → *Java* → *Security*).



After having confirmed the entry, you'll be directed to the webpage, where depending on your browser's settings:

- either the automatic download of management program will start
- or the window will display the option of the file opening or saving:



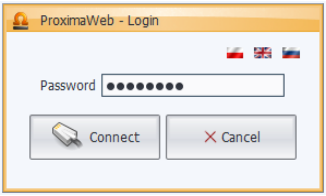
The downloaded file should be opened with Java Web Start Launcher.
If the download does not start automatically, press the *Download ProximaWeb /LibraWeb* button.

The *ProximaWeb/LibraWeb* Java app will require a password on activation (default installer password: 44444444).

3.2 Connection with the use of Libra PC software

Having installed the Libra PC software according to the chapter 15.2 *LibraPC computer program installation*, the LibraPC program should be launched.

When the application from the server site or the Libra PC program are runned, it will ask for the password (installer default password: 44444444).



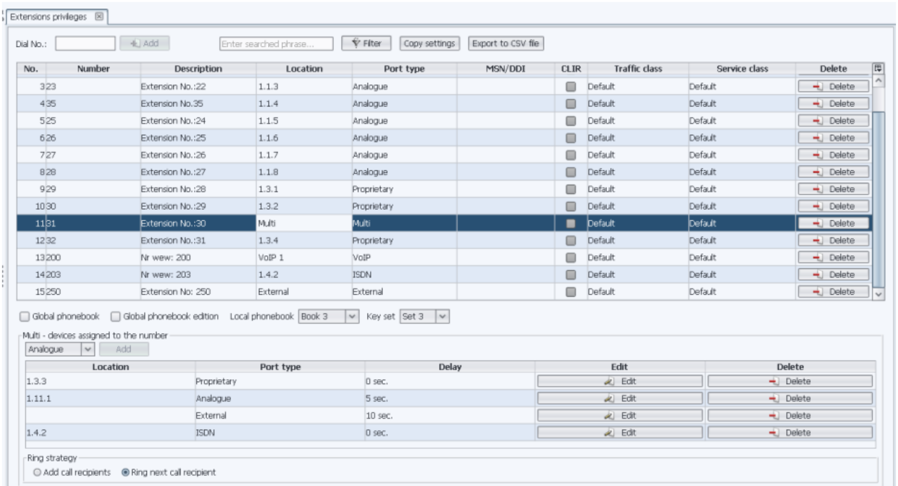
The server configuration settings can be changed in the program tabs. However, the server will apply the introduced changes only after if has received the modified configuration.

Current information about Platan news can be found on the www.platan.eu website.

After the configuration has been downloaded from the server or the *New configuration* option has been selected, the server is ready to be configured.

4. Extensions

4.1 Privileges



The following can be done in the **Privileges** tab:

- Add/Delete an extension.
- Assign/Change dial number – enter the extension number to be assigned to a given terminal in the **Number** field. Remember about the Extensions numeration.
- Edit descriptions – in this field, each extension can be described, e.g. by providing their room number, function, name, etc. These descriptions are used for example in the extension's status or *Call statistics*. The above descriptions are also sent as CLIP on proprietary phones during internal calls.
- Set the CLIR function – to block the presentation of a caller's number.
- Define the traffic class for outgoing calls.
- Define the service class.
- Assign MSN/DDI numbers – each extension can be assigned a 9-digit DDI/MSN number, which will be displayed during outgoing calls via ISDN trunk lines.

  If an extension has a 4-digit or shorter MSN/DDI number assigned, the extension user will be presented in the outgoing ISDN calls with the MSN/DDI leading number of the trunk line with the last two digits replaced by the MSN/DDI digits assigned to this extension.

Attention, a MSN/DDI number assigned to the trunk line is superior to the 5-digit or longer MSN/DDI number assigned to the extension. If a MSN/DDI number sent to the telecom operator is incompatible with the allocated range of numbers, the operator may reject the connection.

- Activate access to the Global phonebook.
- Enable option for the Global phonebook edition.
- Activate access to the Local phonebook for proprietary phones.
- Assign Key sets for speed dialling with BLF (Busy Line Field) for proprietary and IP phones.

- **Programmed buttons** – sets of speed dialling buttons and their corresponding LEDs (for digital proprietary phones) defined in the following tab: *Proprietary Phones* → *Key Sets*. There are 192 sets that can be configured as needed.

- **Global phonebook** – is a common phonebook for proprietary phones with capacity of 3,000 entries. Global phonebook can be edited from a proprietary or IP phone, from *User Zone* or from *ProximaWeb/LibraWeb* configuration program in *Proprietary phones* → *Global phonebook*.

- **Global phonebook edition** – is an option which makes it possible to edit changes in global phonebook from a phone or from *User Zone*.

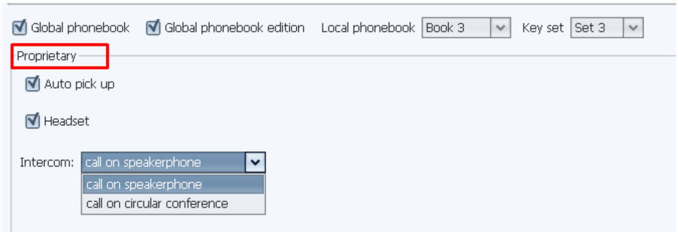
- **Local phonebook** – is a local phonebook of a connected digital proprietary phone. Phonebook can be edited both using the computer program in the *Proprietary phones* → *Local phonebooks* tab and in the phone in which the phonebook is stored.

Note. Only one local or global phonebook can be enabled on a proprietary phone.

- Port type – allows additional settings to be made for the following ports:
  - analogue

Ticking the *Doorphone* option results in different ringtone on analogue corded and proprietary phones while calling from this port.

  - proprietary



Auto pick up is an option which makes it possible to switch on the **speakerphone** system when you press the speed dialling button.

Headset - selecting this option results in switching the phone speaker to the hands-free headset (interchangeably with the *762 and *763 function).

Note. These are **default function codes**. If the numbers had been changed *(Common settings → Function codes),* the proper ones should be used.

Intercom – defines the way the Intercom key on the proprietary phone works. There’re two options:

- **call on speakerphone** – calling another proprietary phone directly on speakerphone to make an announcement;
- **call on circular conference** – calling simultaneously in a circular conference on phones defined as the Hunt Group in the PBX server.

○ <u>IP</u>

Login: to identify and authorise the IP phone in the PBX server.

Password to authorise the IP phone in the PBX server.

Stack – assigning VoIP cards stack for the extension to log in. In order to log in the VoIP phones, Platan Video Softphone or other devices, the address of VoIP card marked as master in a given VoIP stack should be entered in the registration address.

DTMF => FLASH - symbol that executes FLASH function on the server during the call. The following symbols are at disposal: #, *, A, B, C, D or None. Symbols: A, B, C, D can be assigned in some VoIP phones under the programmed keys as Speed Dial function.

T.38 fax relay – function enabling fax transmission with the use of T.38 standard (T.38 requires licence). Option should be ticked for a given VoIP extension if it is used for networking of other Platan systems and the fax transmission between the networked systems is required.

Send re-INVITE for incoming fax – send request for fax in T.38 standard in the case of detecting fax transmission with other codec negociated. It is recommended to enable the function when using fax transmission in T.38 standard.

Videocalls – option that makes it possible to make videocalls through an IP port (requires a licence).

Videocalls – are calls made in Proxima IP PBX Server and Libra PBX Server using VoIP technology. They are available for:

- Proxima/Libra users equipped with Platan Video Softphone app with audio/video accessories or with mobile version of Platan Video Softphone,
- Yealink IP video phones users,
- Helios IP video door phones users.

Videocalls can be made internally between two extensions equipped with the devices mentioned above or between an extension and an external user equipped with video telephone or Platan Video Softphone, when connection is established through an IP trunk line with videocalls service enabled.

Videocall is made bypass media, meaning that the video stream is send outside the Proxima/Libra PBX server. Video call making possibility and its quality depend on the devices used, LAN/WAN network quality and bit rate. Video connection requires for a single call from 64kb/s up to even 4Mb/s in both directions.

Auto provisioning for Platan Video Softphone – function enables reading configuration data required to register Platan Video Softphone app in the Proxima/Libra server.

Signalisation received – signalisation received during the ongoing call (DTMF, RCF 2833 and SIP INFO).

Signalisation sent – signalisation sent during the ongoing call (DTMF, RCF 2833 and SIP INFO).

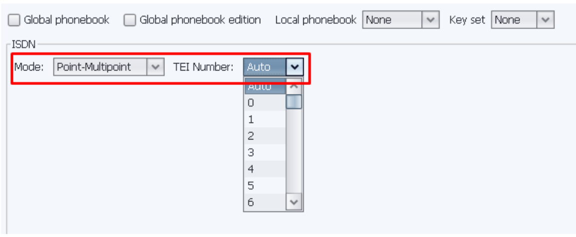
Global phonebook – is a common phonebook for proprietary phones and Platan or Yealink IP phones with capacity of 3,000 entries. Global phonebook can be edited from a proprietary phone, from *User Zone* or from *ProximaWeb/LibraWeb* configuration program in *Proprietary phones → Global phonebook*.

If an IP phone does not use *auto provisioning* function, the access path should be given in *remote phonebook* option of a Platan or Yealink IP phone:

http://Proxima server IP address:port number/get_book.cgi

Waiting messages signalisation – option enables displaying information about the voice message left in the voicemail on the IP phone's display and/or the backlighting of the voicemail button in the phone. The function is available on selected IP phones.

- digital (ISDN)



In order to add an ISDN extension, make sure that the ISDN lines are available or remove unused ISDN BRA lines from menu: *Trunk lines → Privileges*.

To connect the ISDN phone set Mode as *Point-Multipoint* and TEI Number as *Auto*.

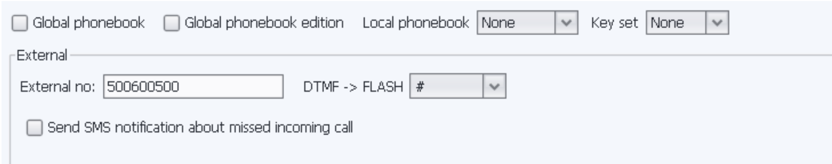
Only the ISDN phones with their own power supply should be connected to ISDN extensions in Proxima and Libra servers.

Only one Gigaset ISDN phone can be connected to each ISDN extension.

In order to connect another PBX system to the Proxima/Libra ISDN extension, set the Mode as *Point-Point* and TEI number as „*0*" (recommended value) or any other.

- external

External port is a specific kind of port that makes it possible to call a defined external number by dialling an extension number. Thanks to that the smartphone user will be available as an extension user, and the defined external number will be able to call into the PBX server and become an extension number with all its privileges.



External no – external number (eg. a mobile phone number) assigned to the given extension number that will receive the connection directed to this extension number.

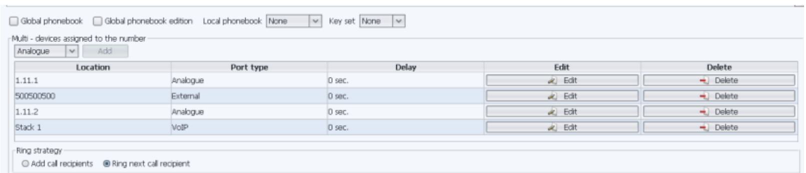
DTMF → FLASH – symbol that executes FLASH function on the PBX server from the phone during the call. There're following symbols at disposal: *, #, A, B, C, D or none. With the selected symbol assigned to the FLASH function a user will be able to put the external calls through to other PBX extension users.

External port is available after buying the licence.

Call made via an external port uses one channel of the trunk line.

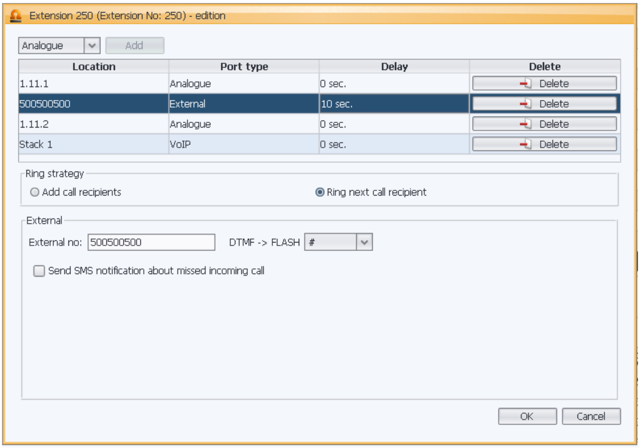
- <u>Multi</u>

Multi port allows to assign to the extension number up to four ports. Thanks to the *Multi* port a user has a possibility of answering calls e.g. on a proprietary phone, when he or she is near the desk, and on analogue DECT phone when moves over the office and on the mobile phone when is outside the company.



Additionally in *Multi* port settings there's an option to set the ring strategy for all ports assigned to *Multi* port.

Click *Edit* to set the ring strategy for all ports assigned to the *Multi* port.



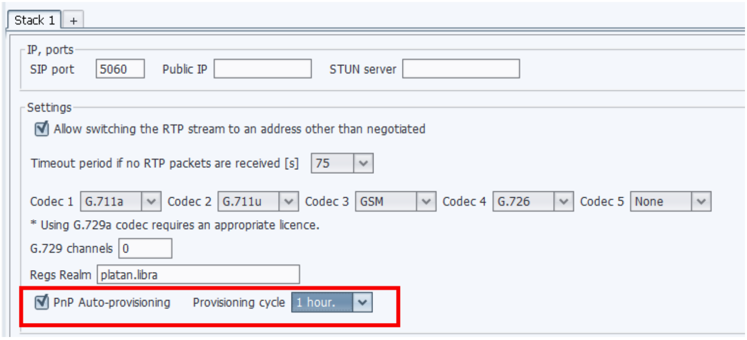
4.1.1 Auto provisioning

Auto provisioning – function which makes it possible to configure automatically via the local network and thanks to the PnP (Plug&Play) function the following IP phones: **Platan IP-T2x**, Yealink T2x, T3x, T4x, in order to work with Proxima IP PBX Server or Libra PBX Server.

Within the auto provisioning function the following data are sent to the IP phones:

- login
- password
- server address
- SIP registration port
- codecs' priorities
- DTMF transmission method
- time settings (time zone, Daylight Saving Time)
- logging password to IP phone
- PnP configuration refreshing
- BLF (Busy Lamp Field) – key sets, available after selecting such option in detailed VoIP extension privileges
- global phonebook (remote phonebook) – available after selecting such option in detailed VoIP extension privileges.

In order to use the auto provision function enable *PnP Auto provisioning* in: *Common settings → VoIP settings*



and set *Provisioning cycle* – frequency of updating the configuration file by the IP phones from the Proxima IP PBX Server and Libra PBX Server.

Then set in the detailed settings of the VoIP phone (menu: *Extensions* → *Privileges*):

- phone MAC address;
- logging password to the phone (recommended);
- refreshing autoconfiguration – updating the configuration changes by the IP phone with the frequency set in the *Provisioning cycle*. If the field is left empty, only the first configuration will be downloaded by the IP phone, updates will not be received;
- key set – setting the BLF (Busy Lamp Field) function in the VoIP phone

! If the phone settings are changed by the Platan or Yealink IP phone user while the option *Refreshing autoconfiguration* is ticked, the phone will restore (after the time set in the *Provisioning cycle*) the settings defined in the Proxima/Libra server.

! If the VoIP card IP address or the SIP registration port have been changed, the IP phones restart is required with *Auto provisioning* service enabled.

4.1.2 Extensions numeration

Extension numbers can be assigned to extensions in the **program mode** of Proxima/Libra server. This applies to program assignment of one specific extension number to each terminal (to each extension line). It provides the possibility of changing the said numeration at any time using the computer program.

Users use the assigned extension numbers during internal and external calls.

Extension numbers can be **0-16 characters** long. Accepted characters are: **0, 1, 2, 3, 4, 5, 6, 7, 8, 9, #, ***. Accepted characters can be used **in any order**.

Assigning extension a 0 characters long number (extension with no number assigned) gives an extension user the possibility to make calls according to their privileges while it will be impossible to call this user.

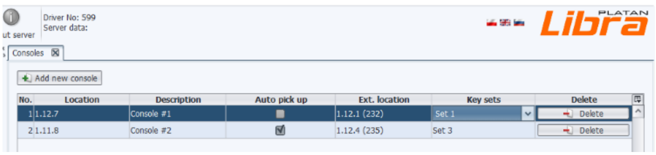
If the emergency numbers (112, 997, 998, 999) are defined in: *Call distribution → Emergency numbers,* extensions cannot include numbers: **1, 9, 11, 99, 112, 997, 998, 999;** in the case of using extension numbers longer than 3 characters, the numbers cannot start with the sequence **112, 997, 998, 999.**

Numeration can be mixed, i.e. 2-, 3- or up to 16-digit numbers can be used at the same time. This numeration method is very flexible. However, it has certain logical restrictions. For example, if we have 3- and 4-digit numbers and one of the extensions has number **100** and subsequent extensions have numbers **101... 105**, etc., there can be no extension no. **10** and **1000-1009, 1010-1019, 1050-1059**, etc.

4.2 Consoles (Libra PBX Server only)

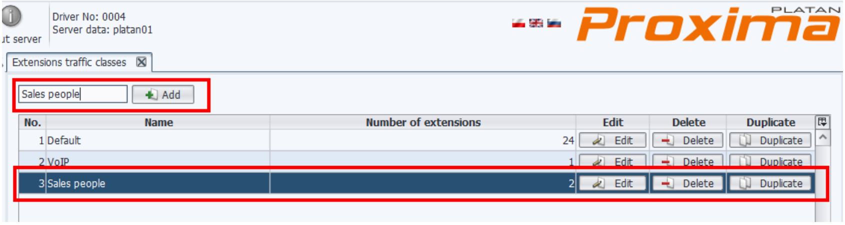
In the *Consoles* tab, the user can assign digital consoles installed in the system to extension users equipped with proprietary phones, assign the key set and the automatic receiver pick up function.

To assign consoles to the system, make sure that at least one proprietary port is available (i.e. no proprietary phone is assigned to it. See: *Extensions → Privileges*, *Port type* column).



4.3 Traffic classes

The *Traffic classes* tab allows you to define 64 traffic classes which can be assigned to server extensions.

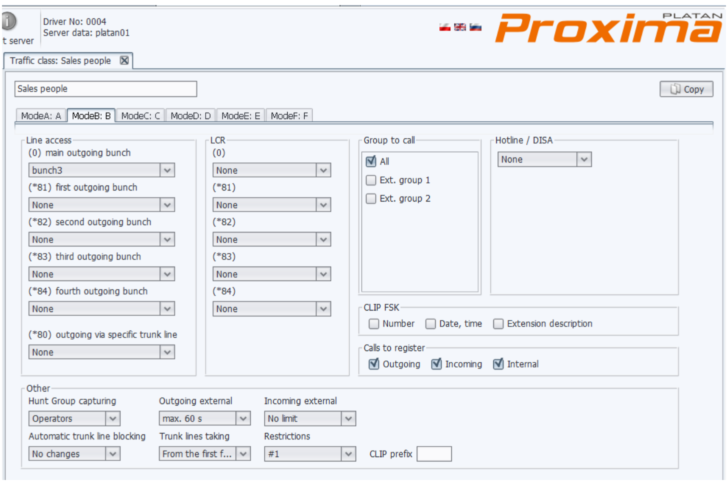


In order to define a new traffic class, enter its name and add it to the list by clicking *Add*. Next, using the *Edit* option, enter the required settings.

Having activated the *Edit* tab but before making any modifications, please check the **Operation Mode** tab as it shows the time in which the displayed settings apply.

4.3.1 Access to trunk lines

Access to trunk lines and outgoing calls are organised based on trunk lines bunches. Trunk lines bunches are defined in the *Trunk lines* → *Bunches* tab. Configuration consists in grouping trunk lines connected to the server into logical groups, e.g. land lines (analogue and ISDN), GSM lines, VoIP lines, etc. Up to 32 trunk lines bunches can be defined.



Next, in the *Traffic classes* tab (while editing a given traffic class), the user can define the trunk lines bunches to which a given extension will have access. The most popular access is the **(0) main outgoing bunch** – after dialling 0, extensions gain access to trunk lines in the assigned trunk lines bunch.

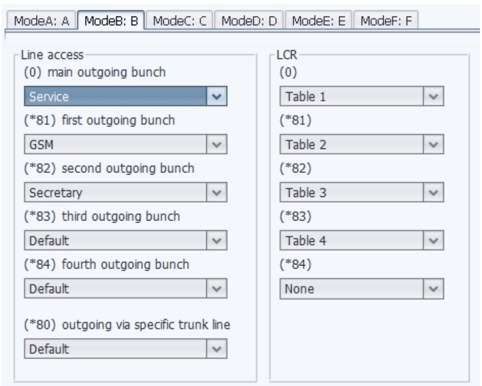
If the extension is assigned a trunk lines bunch only for **(0) main outgoing bunch**, they will hear a ringback tone from the local exchange after dialling "0". After dialling other numbers, i.e. (80xxx) outgoing via specific trunk line – access via the 80a*b*c# function, where: a – unit number, b – trunk line card slot number, c – trunk line card port number, (*81) first outgoing bunch, (*82) second outgoing bunch, (*83) third outgoing bunch and (*84) fourth outgoing bunch, the extension will hear the "unavailability" tone.

Extensions can be assigned various trunk lines bunches for outgoing calls, e.g. in the case of Line access (0) main outgoing bunch, outgoing calls of one extension will be directed through the trunk lines bunch No. 3, while of the other – through the bunch No. 4.

Note. These are **default function codes**. If the numbers had been changed *(Common settings → Function codes),* the proper ones should be used.

4.3.2 LCR

For traffic outgoing through bunches configured in **Line access** box, LCR (*Least Cost Routing*) function can be enabled. Additionally, LCR tables should be configured in *Call distribution → LCR tables.*



There are 4 LCR tables at disposal, which can be freely connected with trunk lines bunch. In the example above:

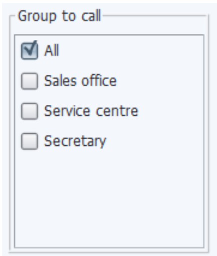
- if an extension user dials **0**, the call will be directed via **main outgoing bunch** (Service) and lines defined in **Table 1** of the LCR box
- if an extension user dials ***81** the call will be directed via **first outgoing bunch** (GSM) and lines defined in **Table 2** of the LCR box

- if an extension user dials ***82** the call will be directed via **second outgoing bunch** (Secretary) and lines defined in **Table 3** of the LCR box, etc.

Note. These are **default function codes**. If the numbers had been changed *(Common settings → Function codes),* the proper ones should be used.

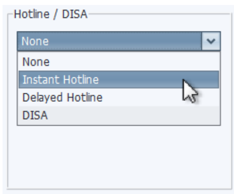
4.3.3 Group to call

In the *Group to call* box, you define the Extension group(s) a given extension can call. Members of Extension groups can be defined in the *Extensions → Extension groups* tab. If all options are selected, the extension will be able to call any other extension assigned to at least one Group. If no option is selected, extensions defined in the given traffic class will not be able to call another extensions.



4.3.4 Hotline/DISA

In order to activate Hotline or DISA function for a given traffic class, select one of the options from the drop-down list in the *Traffic class* tab, *Hotline / DISA* box.



- **Hotline function** – makes the server dial the entered symbols automatically after the extension user has picked up the receiver. An extension number (e.g. a secretary), an external number or any access to trunk lines – e.g. **Line access (0) main outgoing bunch** – can function as a Hotline.

For example, entering "*81" will result in automatic dialling of *81 after picking up the receiver, which means that the extension user will hear the trunk line ringback tone (provided that they have access to this trunk lines bunch). In this example, trunk lines assigned to the **(81) first outgoing bunch** perform the role of a Hotline.

Note. These are **default function codes**. If the numbers had been changed *(Common settings → Function codes),* the proper ones should be used.

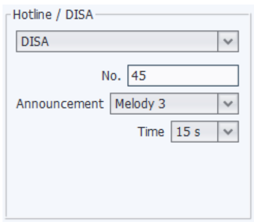
The Hotline function can work in two ways:

⇒ **Instant Hotline** means dialling the programmed digits immediately after picking up the receiver.

⇒ **Delayed Hotline** means that the set number is dialled 3 seconds after picking up the receiver. During this delay, you can dial an extension number and thus interrupt this operation.

The **Hotline** delay time is set in the program: menu *Common Settings → Ringtone and server times → Server times*. This time period is set by default to 3 seconds.
It is possible to exit the **Hotline** and go back to the server services after pressing "#" or "*," depending on which of these symbols is set in the program.
(see menu: *Common settings → Global settings*).

- **DISA** – after picking up the receiver, the server will play the selected announcement or melody. Numbers and server functions can be selected while the announcement is being played. This mode can work as the Delayed Hotline if you provide the server with the number to be dialled after the **DISA time**.



Hotline function is not supported by IP phones.

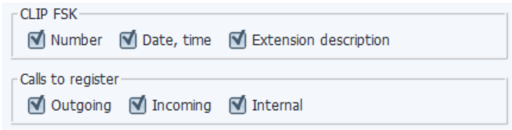
4.3.5 CLIP FSK and calls registration

⇒ **Registration of outgoing calls** – this function activates the registration of billing for outgoing calls. All such calls are registered in the call buffer and can be read and browsed in *Billing – Penny → Call report*. Enabling this function is essential for the use of *Cost limits* function.

⇒ **Registration of incoming calls** – this function activates the registration of incoming calls. All such calls are registered in the call buffer and can be read and browsed in *Billing – Penny → Call report.*

⇒ **Registration of internal calls** – this function activates the registration of internal calls (between extensions). All such calls are registered in the call buffer and can be read and browsed in *Billing – Penny → Call report.*

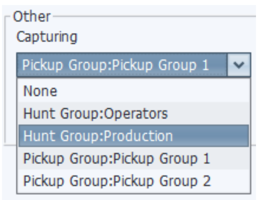
⇒ **CLIP FSK** – means enabling calling line identification presentation for the phones of a given traffic class, both for the incoming external calls (if the trunk line operator provides this type of service) and for the internal calls. Information that can be presented includes the calling line number, date and time of a call and the extension description, if defined in the server. Phones must be equipped with a CLIP receiver (some phones may not support date and description reception).



At the bottom of the tab, in *Other* section, there is a **CLIP Prefix** field in which the trunk line prefix to be attached to the presented calling external line can be entered. E.g. If no prefix has been entered, the presented number is 585558800, while with the “0” prefix – 0585558800. The presented number can also be effectively called back later. <u>If the CLIP Prefix field is empty, the server uses the **0** prefix automatically.</u>

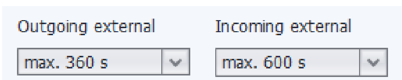
After the **CLIP FSK (Extension description)** option has been selected, part of the description may be cut off. This is due a limited number of displayed characters.

4.3.6 Hunt Group or Pickup Group capturing



Option which makes it possible to capture the call directed to any member of the defined Hunt Group or Pickup Group by using ***710** function (a default function code which can be changed in *Common settings → Function codes)*.

4.3.7 Call time restrictions

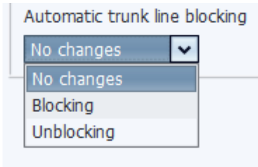


⇒ **Outgoing external** – sets a time limit on outgoing external calls.
⇒ **Incoming external** – sets a time limit on incoming external calls.

If a time limit is set, the extension user will hear an alarm tone 15 seconds before the time limit is reached. After this time, the server will end the call.

4.3.8 Automatic blocking/unblocking of the phone

This function makes it possible to set the automatic blocking/unblocking of the phone for outgoing external calls *(when changing the server operation mode)* – server functions: ***781** and ***780**. Thanks to it, the phone can be for example blocked outside the extension user's working hours and unblocked within these hours.

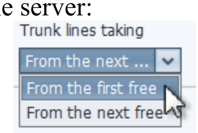


⇒ **Blocking:** – in this mode, the extension will have the ***781** function (automatic outgoing external calls blocked) activated
⇒ **Unblocking:** – in this mode, the extension will have the phone unblocked for outgoing external calls (the *780 function activated).
⇒ **No changes:** – none of the above changes will be introduced.

Note. These are **default function codes**. If the numbers had been changed *(Common settings → Function codes),* the proper ones should be used.

4.3.9 Using trunk lines

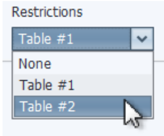
This function defines the method in which a given extension uses trunk lines available for this extension in the server:



⇒ **From the first free** – when outgoing calls are made by the extension belonging to a given traffic class, the server always uses the first free line in a given trunk line bunch.
⇒ **From the next free** – when outgoing calls are made by the extension belonging to a given traffic class, the server uses the external lines one by one, cyclically.

4.3.10 Restrictions

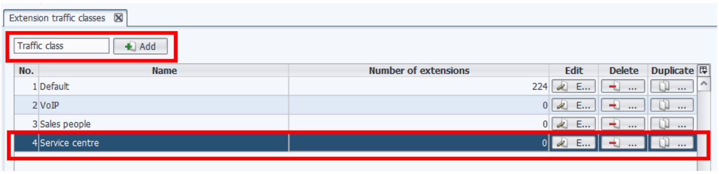
This function makes it possible to use one of 16 available Tables of **Allowed / Denied numbers** containing the set of prefixes or whole external numbers.



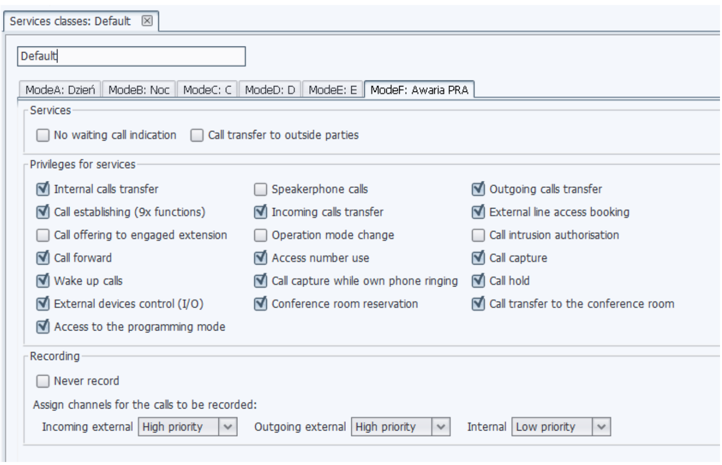
The tables are defined in: *Call distribution → Allowed / Denied Numbers*.

4.4 Privilege classes for services

The *Service classes* tab allows you to define 64 classes which can be assigned to server extensions.



In order to define a new class, enter its name and add it to the list by clicking *Add*. Next, using the *Edit* option, enter the required settings.



In this tab, extensions assigned to a given service class can be privileged to use additional server functions and services.

The following settings are available in the *Services* section:

⇒ **No waiting call indication** – after enabling this service, the extension user will not hear during the call a quiet beeping (the offering tone) signalising the waiting call, and the caller will hear the busy signal. The waiting call indication is used in direct connections only and is not applicable for calls answered by Hunt Groups.

⇒ **Transferring calls to outside parties** – extension user can transfer internal and external calls to outside parties using an idle trunk line.

The following options are available in the *Privileges for services* section:

⇒ **Internal calls transfer** – the extension user can forward internal calls to other server extension users. However, if this function is **not ticked**, no function preceded with the FLASH button can be used.

⇒ **Call establishing (*9x functions)** – the extension user can use the ***9x** function, i.e. redial (*90), booking a call to the last dialled external number (*92), booking a call to any external number (*94) and booking a call to any external number at the specified hour (*95).

⇒ **Call offering to engaged extension** – the extension user can join the ongoing call and offer a call to the engaged user.

⇒ **Call forward** (from own phone to another one) – the extension user can forward their number to another extension number defined in the server dial plan or to another external number – functions ***734, *735, *736, *737**.

⇒ **Wake up calls** – the extension can use the ***731** (single wake-up call) and ***732** function (everyday wake-up calls).

⇒ **External devices control (I/O)** – the extension user can control external devices from the phone (requires the I/O card).

⇒ **Access to the programming mode** – the extension user can change the server settings from the phone (***708**).

⇒ **Speakerphone call** – the extension user can send messages directly onto the proprietary speakerphone – function ***79**.

⇒ **Incoming calls transfer** – the extension user can transfer incoming external calls to other extensions.

⇒ **Operation mode change** – the extension user can manually change the server operation mode – functions ***742**, ***743**,...,***747**.

⇒ **Acces number use** – the extension user can use the server functions (e.g.: return calls, transferring external calls to another external line) requiring the access number defined in the program: *Common settings → Calls*.

⇒ **Call capture while own phone ringing** – the extension user can capture a call directed to another phone while their own phone is ringing – functions ***71** and ***710**. In order for this privilege to function, the *Call capturing* privilege must be ticked as well. <u>For proprietary phones only.</u>

⇒ **Conference room reservation** – the extension user can create many conference rooms for the maximum number of users in a single conference according to the licence – ***85** function.

⇒ **Outgoing calls transfer** – the extension user can transfer outgoing calls to other extension users, e.g. the secretary calls an outside party and then transfers the call to the director.

⇒ **External line access booking** – the extension user can book access to the external line after dialling “0” when the external line is engaged.

⇒ **Call intrusion authorisation** – the extension user allows authorised extensions to intrude their own calls and offer other external calls *(see: call offering)*.

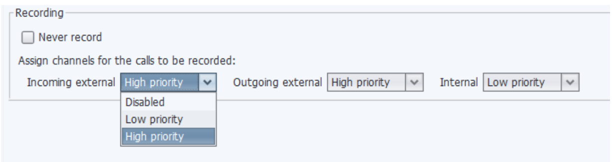
⇒ **Call capture** – the extension user can capture the call directed to another phone – functions ***71** and ***710**;

⇒ **Call hold** (HOLD) – the extension user can put a call on hold for the period of time defined in the program.

⇒ **Call transfer to the conference room** – the extension user can transfer the external call to the conference room.

The following setting is available in the *Recording* section:

Recording – option enabling call recording for extensions assigned to a given service class, separately for incoming external, outgoing external and internal calls. Additionally, the call priorities can be defined: *Low priority* or *High priority*.
With *Never record* option ticked, the recording for a given service class will be disabled, even if other recording criteria are enabled, for example call recording from *Call schemes*.

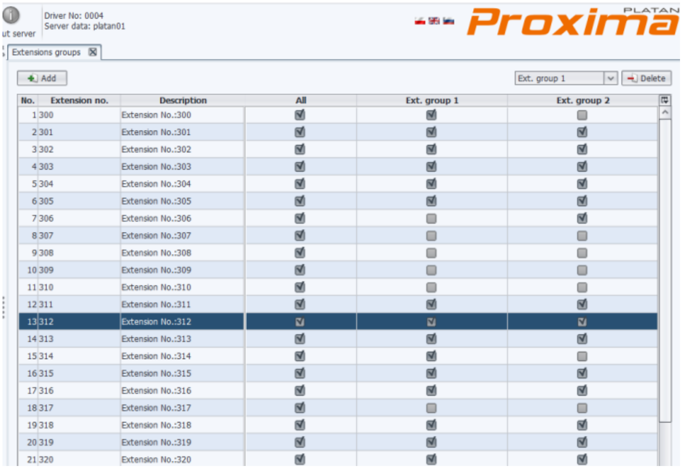


Note. These are **default function codes**. If the numbers had been changed *(Common settings → Function codes),* the proper ones should be used.

4.5 Extension groups

Each server extension should be assigned to at least one **Extension group** (max 32 groups). If an extension is not assigned to any group, no one will be able to call them (also from outside). Each extension can be assigned to several groups and all extensions can be assigned to one group. Dividing extensions into groups makes it possible, for example, to define such server extensions to which direct access through external lines (in the DISA mode) will not be possible, e.g. the company board, etc. Such extensions are assigned to a group not supported in the DISA mode.

In order to assign an extension to a given **Extension group**, the selected *Ext. group* should be ticked for this extension.



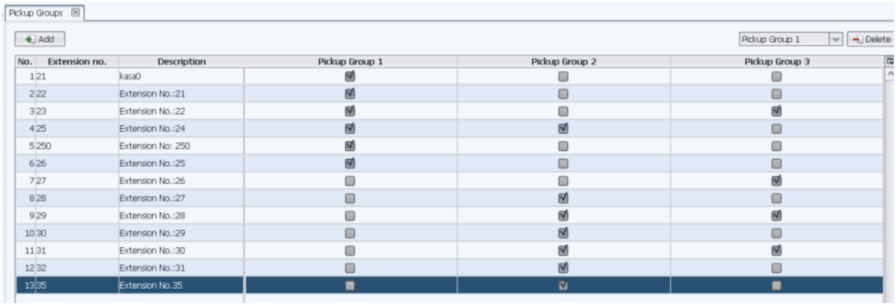
In the *Traffic class* tab the *Group to call* section result from dividing server extensions into Extension Groups. Here, you define the Extension Group a given extension can call. If all sixteen groups are selected, the extension will be able to call any other extension assigned to at least one Group. If no option is selected, a given extension will not be able to call another extension.

In order to facilitate the settings right click on a given Group column permits to choose the option: All, Nobody or Reverse.

4.6 Pickup groups

Pickup groups are the groups of extension users, where a call directed to any user can be picked up (captured) by another user having the privilege to pick up such call. Pickup groups can consist of users belonging to different Hunt Groups, and the call may have different source: direct, from the call scheme etc.

In order to assign an extension to a given **Pickup group**, the selected *Pickup group* should be ticked for this extension.



No.	Extension no.	Description	Pickup Group 1	Pickup Group 2	Pickup Group 3
1	21	kasa0	☑	☐	☐
2	22	Extension No.:21	☑	☐	☐
3	23	Extension No.:22	☑	☐	☑
4	25	Extension No.:24	☑	☑	☐
5	250	Extension No: 250	☑	☐	☐
6	26	Extension No.:25	☑	☐	☐
7	27	Extension No.:26	☐	☐	☑
8	28	Extension No.:27	☐	☑	☐
9	29	Extension No.:28	☐	☑	☑
10	30	Extension No.:29	☐	☑	☐
11	31	Extension No.:30	☐	☑	☑
12	32	Extension No.:31	☐	☑	☐
13	35	Extension No.35	☐	☑	☐

For capturing calls directed to the Pickup groups see: Hunt Group or Pickup Group capturing.

4.7 Call forwarding

The server makes call forwarding possible. Most settings can be made locally by any extension using their phone provided that they are authorised to do so.

The *Call Forward Unconditional (CFU) from “TEL” to me (“I’m here”)* function is set only in the phone to which the call from “TEL” is to be forwarded.

Call forwarding can be configured for each extension in the *Call forwarding* tab (default function codes are presented):

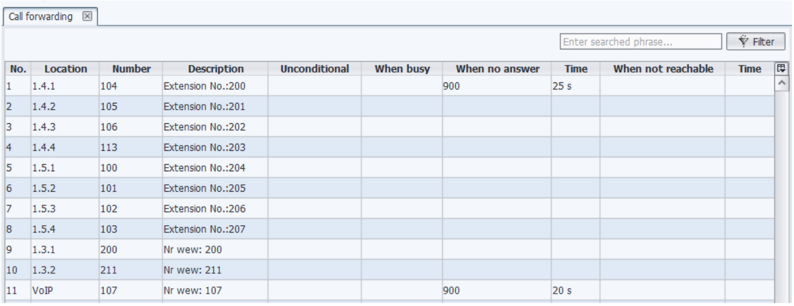
⇒ Call Forward Unconditional (CFU) to “TEL” (function ***734 “TEL”**),

⇒ Call Forward Busy (CFB) to “TEL” when my number is busy (function ***736 “TEL”**),

⇒ Call Forward No Answer (CFNA) to “TEL” when my number does not answer (function ***737 “TEL”**),

⇒ Call Forward Not Reachable (CFNR) to “TEL” when the IP phone is not reachable (function ***738 “TEL”**) for one of the following reasons:

- the user is not registered,
- the 487/Request terminated status,
- there are no IP channels,
- there is no VoIP transmission (e.g. disconnected LAN cable).



Call forwarding

Enter searched phrase... Filter

No.	Location	Number	Description	Unconditional	When busy	When no answer	Time	When not reachable	Time
1	1.4.1	104	Extension No.:200			900	25 s		
2	1.4.2	105	Extension No.:201						
3	1.4.3	106	Extension No.:202						
4	1.4.4	113	Extension No.:203						
5	1.5.1	100	Extension No.:204						
6	1.5.2	101	Extension No.:205						
7	1.5.3	102	Extension No.:206						
8	1.5.4	103	Extension No.:207						
9	1.3.1	200	Nr wew: 200						
10	1.3.2	211	Nr wew: 211						
11	VoIP	107	Nr wew: 107			900	20 s		

In the case of CFNA and CFNR, the time after which the call is to be forwarded should also be specified.

Each extension user can modify these settings in their phone using the ***734**, ***736**, ***737, *738** functions *(see: "Proxima/Libra PBX Server User Manual").*

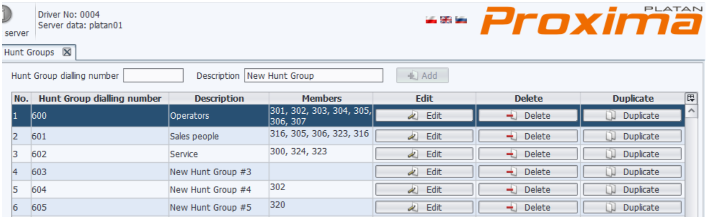
Note. These are **default function codes**. If the numbers had been changed *(Common settings → Function codes),* the proper ones should be used.

When defining Call forwarding in relevant columns (Unconditional, When busy, When no answer) it is important to remember that the external number to which the call is to be forwarded should begin with the escape code defined in *Common settings → Function codes → Accessing trunk line through the main trunk lines bunch set in the configuration (escape code)*. If no number is defined as the escape code, the external number in call forwarding should be entered without it.

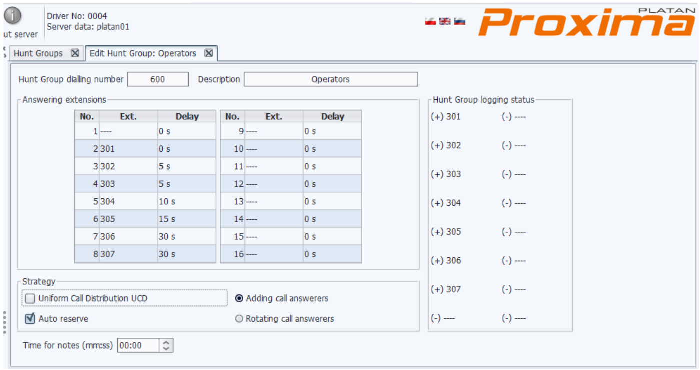
The server allows call forwarding to an extension number, a Hunt Group number, a voicemail number or an external number only through **the main trunk line bunch** (with or without the escape code defined).

4.8 Hunt Groups

A common dialling number can by assigned to selected extensions in the server. The range of numbers for **Hunt Groups** is the same as in the case of the extensions' numeration but they cannot duplicate.



A maximum of 16 extensions can be assigned to one Hunt Group. The *Delay* field is to specify the delay after which a given extension's phone rings. The time of delay is counted from the beginning of calling the Hunt Group.



Up to 64 Hunt Groups can be defined in the server. Each Hunt Group can be assigned an additional description facilitating its identification.

After dialling the Hunt Group dialling number, the internal or external caller calls all extensions assigned to a given group in the declared order or all of them at the same time (depending on the settings defined in the *Delay* field), until the call is answered by any member of the Hunt Group.

With the *Delay* time set on *0 s.* for all the Hunt Group members, the incoming call will ring simultaneously on all the Hunt Group phones, while the next call will receive the busy signal until the first one is answered by one of the Hunt Group members. Thus, it is recommended to set the *Delay* time together with *Auto reserve* option enabled or to define the call distribution to the Hunt Groups in *Call distribution → Call schemes* with *queue* option enabled.

If the *Auto reserve* option is ticked, delay times are not taken into account, e.g. if a Hunt Group contains four members, first two receiving the call with no delay, and other two with defined delay times, and the first two extensions are busy, the call will be directed with no delay to the 'reserve' extensions.

It is also possible to establish a conference call with all extensions within a given Hunt Group. This function is called a **Circular Conference** and is described in the Server User Manual. It applies to internal calls only.

If the trunk line works in the DISA mode, the external caller can dial a Hunt Group number. Then the phones in a given Hunt Group will start ringing according to the declared strategy:

⇒ Adding call recipients
⇒ Rotating call recipients.

The server can be ordered to carry out **Uniform Call Distribution**. It will direct every new incoming call to the listed extensions. However, a different extension will be called first every time. Subsequent extensions will be added with a declared delay *(Common settings → Ringtone and server times → Server times → Joining time in Uniform Call Distribution; default value: 5 s)*. The call distribution procedure described above follows a cyclical pattern. This function is useful for services provided, among others, by **TeleTaxi**, **CallCentres**, etc.

An extension number is superior to the Hunt Group dialling. If it happens that a given group has the same number as the extension, the server will call the extension. Therefore, the Hunt Group dialling number should be excluded from the server numbering plan.

The server makes it possible to log into (***730 xxx 1,** where xxx stands for the Hunt Group dial number) and log out **(*730 xxx 0)** of the Hunt Group. Thanks to this function, you can log out when leaving the workstation, e.g. for a “lunch break,” so that the server does not direct calls to your number. When you come back, you can log into the Hunt Group again.

Time for notes – time for which the Hunt Group member who answered the call will be logged out from the Hunt Group after this call in order to take notes. While being logged out from the Hunt Group, the user will be still able to receive direct calls. When *Time for notes* is enabled, any direct call answered does not prolong the Hunt Group logging out for the time of the call.

For capturing calls directed to the Hunt Groups see: Hunt Group or Pickup Group capturing.

Note. These are **default function codes**. If the numbers had been changed *(Common settings → Function codes),* the proper ones should be used.

The fact of extensions logging into and out of a Hunt Group is recorded in the server operation report.

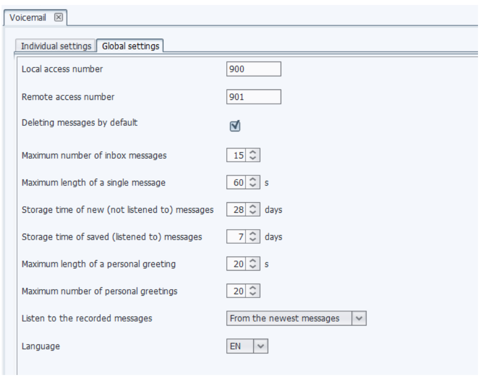
4.9 Voicemail

The server has a built-in voicemail, which can be activated either for all server extensions or only for the selected ones. The voicemail operation is described in the *Proxima IP PBX Server / Libra PBX Server User Manual*. Voicemail is configured in the computer program.

4.9.1 Voicemail settings adjusted in the program

Global settings (applying to all extensions):

⇒ Local and remote access number to the voicemail inbox: the assigned numbers must be free and excluded from the server numeration. If there is no local number, voicemail is switched off.
⇒ Deleting messages by default: *ON by default*
⇒ Maximum number of inbox messages: 1-25, *15 by default*
⇒ Maximum length of a single message: 5-180 s, *60 s by default*
⇒ Storage time of new (not listened to) messages: 1-255 days, *28 days by default*
⇒ Storage time of saved (listened to) messages: 1-255 days, *7 days by default*
⇒ Maximum length of a personal greeting: 1-30 s, *20 s by default*
⇒ Maximum number of personal greetings: 0-255, *20 by default.*
⇒ Listen to the recorded messages: *From the newest messages*, *From the oldest messages*.
⇒ Language – voicemail and voice messages language, e.g. information about IP address: *EN*, *RU*, *PL*.



Additional common settings are made in:
Common settings → Rhythms

⇒ Voicemail call ringtone: *"0.25 pulse/0.75 pause" by default* (for analogue phones)

Common settings → Ringtone and server times → Ringtone times

⇒ Voicemail notification time: 1-255 s, *20 s by default.*

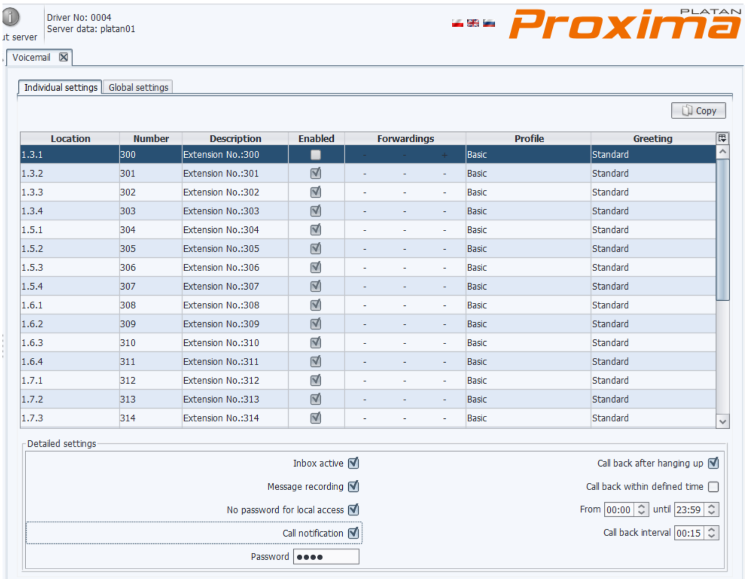
Individual settings (set separately for each extension):

⇒ Voicemail enabled / disabled: *enabled by default*

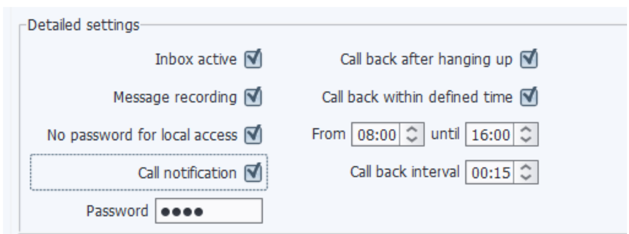
⇒ Voicemail profile: Basic or Advanced (*Basic by default*)

⇒ Greeting: Standard or Personal (*Standard by default*)

⇒ Voicemail inbox calling method defined in *Extensions → Call forwarding* tab are visible in *Forwardings* column: Unconditional, When busy and/or When no answer



The tab above contains individual settings of voicemail users. Detailed parameters should be set in the bottom part after selecting a given server extension.



⇒ Inbox active: *ON by default* (when voicemail enabled)

⇒ Message recording: *ON by default*

⇒ No password required for local access (ON/OFF): *OFF by default*
⇒ Password: Voicemail inbox access password (*1234 by default*)
⇒ Notification method: call notification switching on/off (*OFF by default*), call back after hanging up (*ON by default*) and/or within a defined period of time: *OFF by default*
⇒ Notification time and frequency: 00:00-23:59, *00:00-23.59 by default*, every [hh:mm], *00:15 by default*

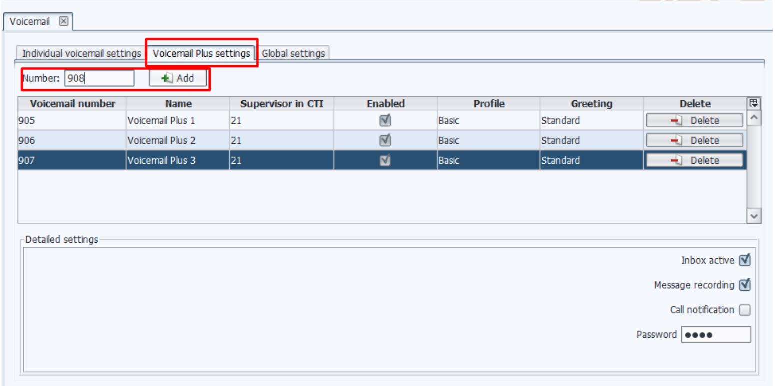
Individual settings introduced in the voice menu (see: *Proxima & Libra PBX Server User Manual*):
⇒ Message recording ON/OFF
⇒ Call attempt notification ON/OFF
⇒ Inbox access password change
⇒ Greeting change: standard or personal
⇒ Inbox operation profile selection: standard or advanced

The "switch on voicemail inbox answering" option is not set in the phone. If it is switched off, the inbox will not handle incoming calls but users will still be able to get through to it.

Before the configuration is changed in the program, its current version should be read from the server. The above settings introduced in the phone will not be deleted.

Voicemail Plus settings

⇒ Voicemail Plus number. The assigned number must be free and excluded from the numbering plan, i.e. not used in *Privileges* window among others.
⇒ Voicemail Plus name.
⇒ Supervisor in CTI – the extension user with Platan CTI software (ver. 2.10.xx or later). Wave files with messages left in the voicemail will be downloaded on CTI client of this extension number.
⇒ Voicemail enabled/disabled: *disabled by default*
⇒ Voicemail profile selection: *Basic* or *Advanced*
⇒ Greeting selection: *Standard* or *Personal*



In the *Detailed settings* there're:

⇒ Inbox active: *enabled by default*

⇒ Enabling / disabling message recording: *enabled by default*

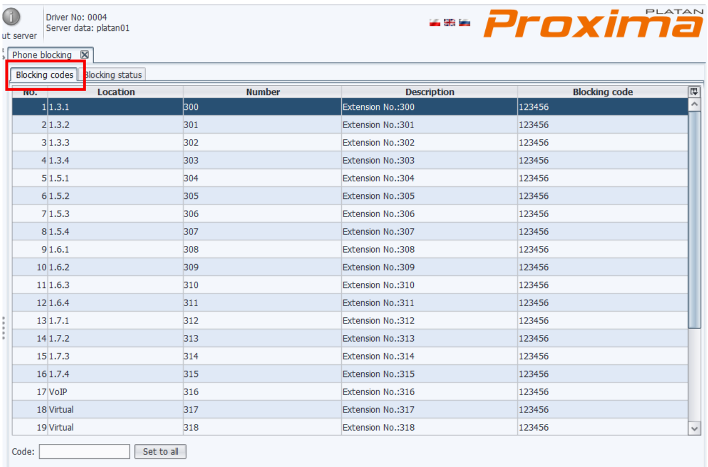
⇒ Enabling / disabling notification about call attempts: *disabled by default*

⇒ Password (access code to the voicemail) change.

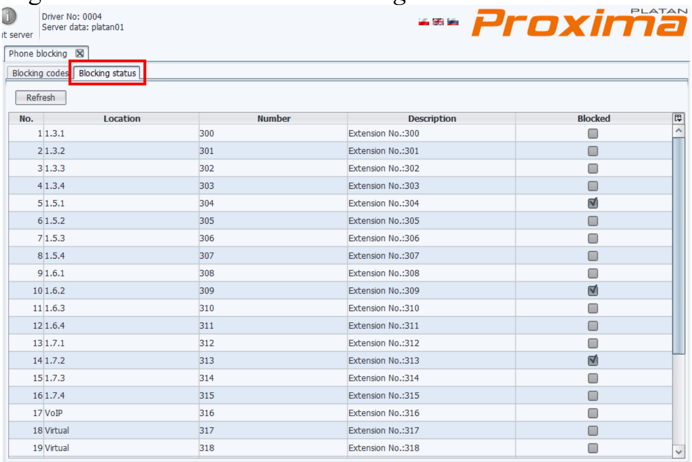
Note! Voicemail Plus inboxes should be used in the scenarios of incoming traffic, i.e. *MSN/DDI tables*, *ACD tables* and in *Call schemes*.

4.10 Phone blocking

In the *Phone blocking* tab, the phone can be blocked/unblocked on external calls. It also allows extension user to change codes blocking their phones on external outgoing calls, e.g. when a given user leaves their workplace for a moment.



The blocking status can be checked in *Blocking status* tab.



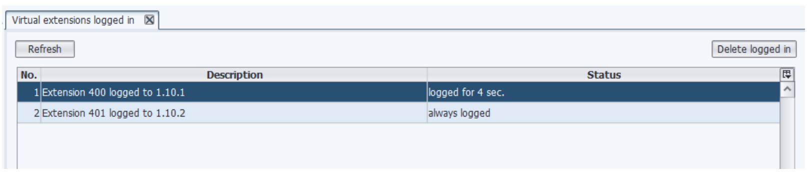
No.	Location	Number	Description	Blocked
1	1.3.1	300	Extension No.:300	
2	1.3.2	301	Extension No.:301	
3	1.3.3	302	Extension No.:302	
4	1.3.4	303	Extension No.:303	
5	1.5.1	304	Extension No.:304	✓
6	1.5.2	305	Extension No.:305	
7	1.5.3	306	Extension No.:306	
8	1.5.4	307	Extension No.:307	
9	1.6.1	308	Extension No.:308	
10	1.6.2	309	Extension No.:309	✓
11	1.6.3	310	Extension No.:310	
12	1.6.4	311	Extension No.:311	
13	1.7.1	312	Extension No.:312	
14	1.7.2	313	Extension No.:313	✓
15	1.7.3	314	Extension No.:314	
16	1.7.4	315	Extension No.:315	
17	VoIP	316	Extension No.:316	
18	Virtual	317	Extension No.:317	
19	Virtual	318	Extension No.:318	

The phone blocking code is also the password to the User Zone described in the *Proxima & Libra PBX Server User Manual*.

4.11 Virtual extensions logged in

The server features the virtual extension system and makes it possible for an extension user to log into any extension port. This applies both to the existing physical extensions (having server terminals assigned) and to the virtual ones. The function codes are described in the *Proxima & Libra PBX Server User Manual*.

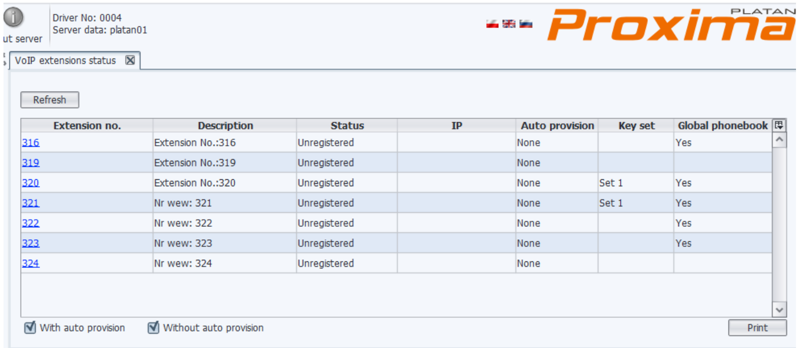
The program allows you to make the list of all virtual extensions logged into the server at a given moment. The list includes the server terminal number, the number of the extension logged into the server and specifies whether the extension is logged into the server for a fixed period of time or permanently.



No.	Description	Status
1	Extension 400 logged to 1.10.1	logged for 4 sec.
2	Extension 401 logged to 1.10.2	always logged

4.12 VoIP extensions status

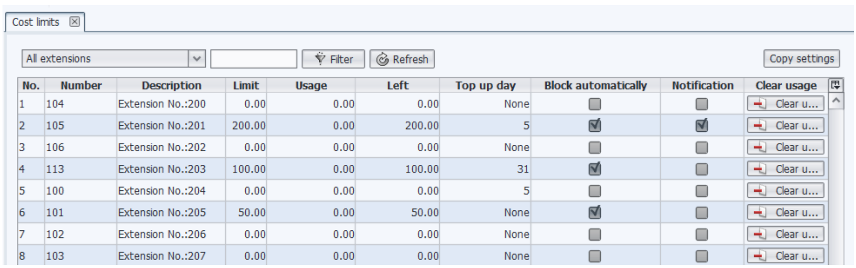
In *VoIP extensions status* tab the VoIP extensions logging status is visible online.



Extension no.	Description	Status	IP	Auto provision	Key set	Global phonebook
316	Extension No.:316	Unregistered		None		Yes
319	Extension No.:319	Unregistered		None		
320	Extension No.:320	Unregistered		None	Set 1	Yes
321	Nr wew: 321	Unregistered		None	Set 1	Yes
322	Nr wew: 322	Unregistered		None		Yes
323	Nr wew: 323	Unregistered		None		Yes
324	Nr wew: 324	Unregistered		None		

4.13 Cost limits

The *Cost limits* function makes it possible to set the quota limit of monthly expenditures on external calls. A separate limit and top up day can be set for every extension.



No.	Number	Description	Limit	Usage	Left	Top up day	Block automatically	Notification	Clear usage
1	104	Extension No.:200	0.00	0.00	0.00	None	☐	☐	Clear u...
2	105	Extension No.:201	200.00	0.00	200.00	5	☑	☑	Clear u...
3	106	Extension No.:202	0.00	0.00	0.00	None	☐	☐	Clear u...
4	113	Extension No.:203	100.00	0.00	100.00	31	☑	☐	Clear u...
5	100	Extension No.:204	0.00	0.00	0.00	5	☐	☐	Clear u...
6	101	Extension No.:205	50.00	0.00	50.00	None	☑	☐	Clear u...
7	102	Extension No.:206	0.00	0.00	0.00	None	☐	☐	Clear u...
8	103	Extension No.:207	0.00	0.00	0.00	None	☐	☐	Clear u...

⇒ **Limit** – quota limit set to an extension for external outgoing calls within a month;

⇒ **Usage** – the quota limit used within the current settlement month;

⇒ **Left** – the quota limit remaining to be used in the current settlement month;

⇒ **Top up day** – day of the month, when the limit will be renewed. If the 31st day is selected, the limit will be always renewed on the last day of the month;

⇒ **Block automatically** – with this option enabled and after the assigned quota limit has been used, the extension user will have no possibility of making outgoing external calls, except the emergency calls;

⇒ **Notification** – option enabling the voice message being played for the extension user who has exceed the assigned quota limit and tries to make an outgoing external call. The voice message should be set in *ProximaWeb /LibraWeb* program in: *Common settings* → *Calls* menu. If the *Notification* option is disabled, the caller will hear an unavailability tone;

⇒ **Clear usage** – clearing the up to date quota limit usage for a given extension;

⇒ **Clear usage to all** (button at the bottom of the page) – the quota limit usage will be cleared for all extensions.

NOTE. In order to have the *Cost limits* working properly, the billing parameters should be configured corectly in the *Billing – Penny* menu.

The *Copy settings* option (button – see the screen below) can be used to copy all or selected parameters of cost limits from one extension to another. If you right-click on the *To* field, the *All*, *Nobody* and *Reverse* options appear. They make it possible to select accordingly all extensions, none or to reverse the selection in order to facilitate the settings copy.

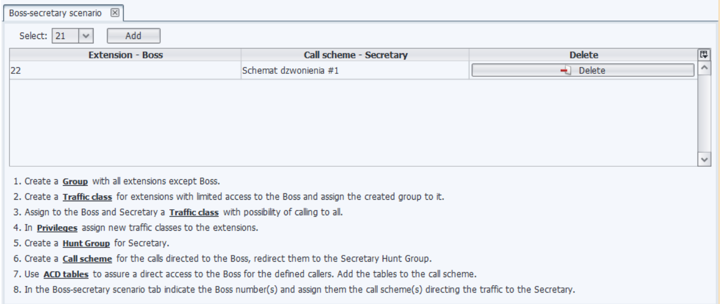
Additionaly, the *Export to CSV file* and *Print* options are available at the bottom of the page to export or print the current state of cost limits.

4.14 Boss-secretary scenario

Option that allows the professional support of calls directed to the managers through secretariat. In order to configure the boss-secretary scenario correctly, the following actions should be made (instructions are also available in the Boss-secretary scenario tab of the software):

1. Create an *Extension group* with all extensions except Boss(es).
2. Create a *Traffic class* for extensions with limited access to the Boss. Assign the group created previously to this traffic class.
3. Assign to the Boss and Secretary the traffic class with possibility of calling all extensions.
4. In *Privileges* assign new traffic classes to the extensions.
5. Create a *Hunt Group* for Secretary.
6. Create a *Call scheme* for calls dirrected to the Boss, direct it to the Secretary Hunt Group.
7. Use *ACD tables* to assure direct access to the Boss for defined callers. Add the tables to the call scheme.

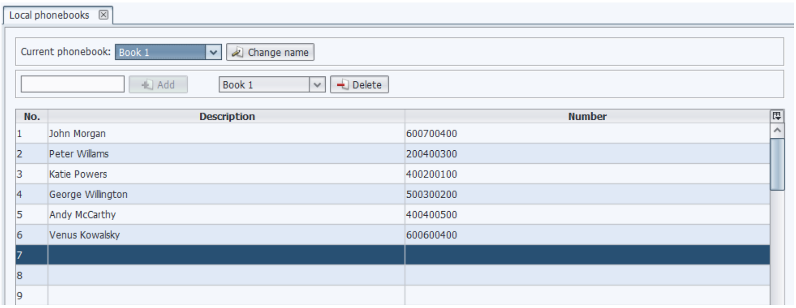
8. In Boss-secretary scenario window indicate the Boss(es) numbers and assign them the call schemes directing traffic to the Secretary.



5. Proprietary phones

5.1 Local phonebooks

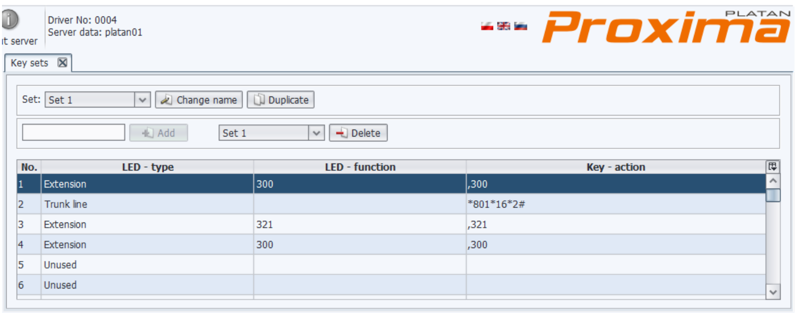
Up to 128 phonebooks can be defined in the *Proprietary phones → Local phonebooks* tab. Each phonebook can contain up to 32 external phone numbers.



5.2 Key sets

One of 192 key sets and LEDs should be assigned to the connected proprietary phones, consoles, Platan or Yealink IP phones with BLF (Busy Lamp Field) function and Platan Video Softphone. This is done in: *Extensions → Privileges,* and for proprietary consoles in *Extensions → Consoles* menu.

The sets define LED signalling and key functioning. A key may activate a server function, an extension or an external number.



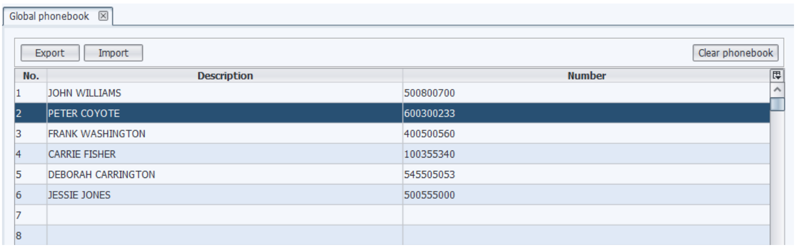
Factory settings can also be restored by pressing **Default**.
Entries can contain 30-digit numbers.

It is possible for the user to configure buttons and LEDs using the proprietary phone keyboard. Therefore, in order to save the user's settings, the server configuration should be downloaded every time before editing these sets because if any configuration is sent to the server, the user's settings will be replaced with the program settings.

If several proprietary phones have the same set assigned, every change to the keys or LEDs settings made on one of these phones' keypad will change the settings of all the other phones with the same set assigned.

5.3 Global phonebook

You can define a global phonebook for proprietary phones, Platan and Yealink IP phones containing extension and external numbers. This is done in *Proprietary phones → Global phonebook* tab.



Global phonebook can be edited by the users with the proper privilege with the use of:

⇒ proprietary phone

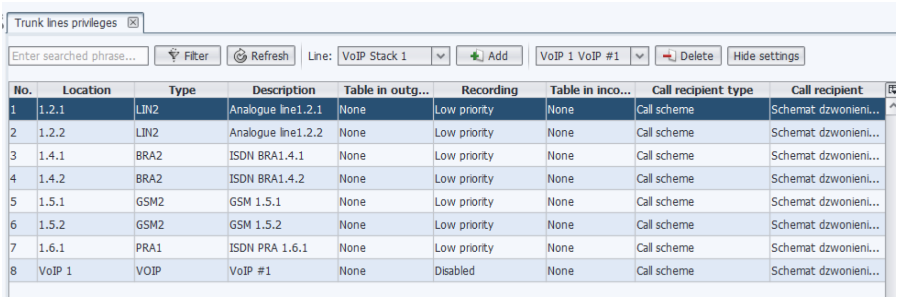
⇒ User Zone.

In the *Global phonebook* tab you can export the settings and import them from a .csv file to facilitate the configuration.

Example of .csv file:
JOHN SMITH;585558800
MARTIN BROWN;585558801.

6. Trunk lines

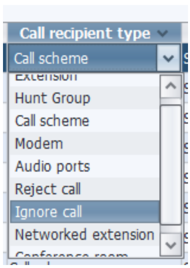
6.1 Privileges



No.	Location	Type	Description	Table in outg...	Recording	Table in inco...	Call recipient type	Call recipient
1	1.2.1	LIN2	Analogue line1.2.1	None	Low priority	None	Call scheme	Schemat dzwonieni...
2	1.2.2	LIN2	Analogue line1.2.2	None	Low priority	None	Call scheme	Schemat dzwonieni...
3	1.4.1	BRA2	ISDN BRA1.4.1	None	Low priority	None	Call scheme	Schemat dzwonieni...
4	1.4.2	BRA2	ISDN BRA1.4.2	None	Low priority	None	Call scheme	Schemat dzwonieni...
5	1.5.1	GSM2	GSM 1.5.1	None	Low priority	None	Call scheme	Schemat dzwonieni...
6	1.5.2	GSM2	GSM 1.5.2	None	Low priority	None	Call scheme	Schemat dzwonieni...
7	1.6.1	PRA1	ISDN PRA 1.6.1	None	Low priority	None	Call scheme	Schemat dzwonieni...
8	VoIP 1	VOIP	VoIP #1	None	Disabled	None	Call scheme	Schemat dzwonieni...

The following can be done in the ***Privileges*** tab:

- Check the trunk line type in a given slot (analogue, BRA, PRA, GSM, VoIP)
- Assign/Change the description of a trunk line.
- Enable recording – all incoming and outgoing call made by a given line will be recorded, excluding the calls directed to extensions with *Service class* with *Never record* option enabled. Two priorities of recording are available: *Low priority* and *High priority*.
- Assign one of the MSN/DDI tables defined previously.
- Define the *Call recipient type*:



  - Extension
  - Hunt Group
  - Call scheme
  - Modem
  - Audio ports (for Libra PBX Server only, if previously defined in *Common settings → External devices control (I/O)*)
  - Digital ports (if previously defined in *Common settings → External devices control (I/O)*)
  - Reject call

  - Ignore call
  - Networked extension
  - Conference room
- Assign the *Call recipient* to the selected *Call recipient type.*

With the *Call recipient type* set as *Extension* or *Hunt Group*, the incoming external calls from GSM/ISDN/VoIP lines to the busy extension or Hunt Group will receive the busy signal. The same will happen if the *Call recipient type* is set as *Reject call*.

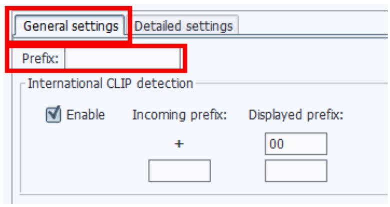
In the same cases for the external calls incoming from the analogue trunk lines, the Proxima/Libra server will ignore the connection and the caller will hear a ringback tone (as in *Ignore call*).

6.1.1 Analogue trunk lines settings

After having selected the given analogue trunk line in the *Trunk lines → Privileges*, the *General settings* and the *Detailed settings* tabs for this line will appear in the bottom part of the window (remember to click *Show settings* button if they are hidden).

General settings

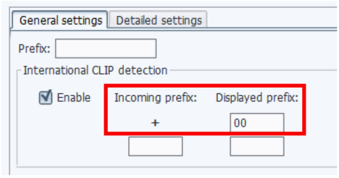
In the *General settings* tab the prefix that will be added to all outgoing calls may be defined.



The *International CLIP detection* (enabled by default for all types of lines) makes it possible to modify the CLIP number in incoming calls and thus serve properly the international numbers, which may differ in their international call prefixes.

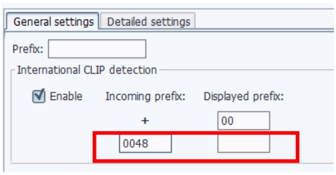
If the incoming number will be preceded by "+", the server will present the number defined in the *Displayed prefix* field.

Example 1



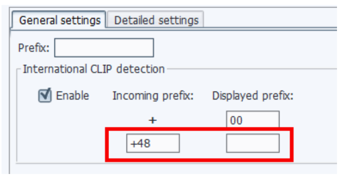
If the operator sends the number: *+48585558800* in the incoming call, with the above settings, the number presented on the phones will be *00048585558800*, where: the first *0* is the default escape code (access to the external line), and the following *00* were added after the server had received the information about the international call.

Example 2



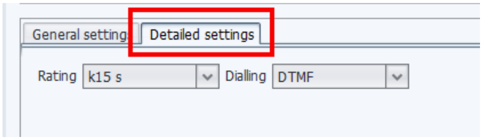
If the operator sends the number: *0048585558800* in the incoming call, with the above settings, the number presented on the phones will be *0585558800*, where: the first *0* is the default escape code (access to the external line). The server will cut off the *0048* prefix from the presented number. The call from the *0049245245245* number will be presented as follows: *00049245245245*.

Example 3



If the operator sends the number: *+48585558800* in the incoming call, with the above settings, the number presented on the phones will be *0585558800*, where: the first *0* is the default escape code (access to the external line). The server will cut off the *+48* prefix from the presented number. The call from *+49245245245* number will be presented as follows: *00049245245245*.

Detailed settings



Rating – parameter defining the call start in order to rate and charge the call correctly:

- *National* – connection's start and end are signalised by polarisation reversal made by the telecom operator on the analogue trunk lines;
- *Time* – connection starts after the preset period of time, counted from the first dialled digit (*25-75 s*), or from the last dialled digit (time proceeded by 'k' letter: *k5-k40 s*).

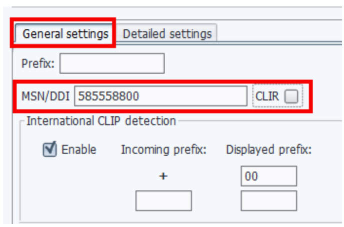
Dialling – defines the mode in which dialled digits are transmitted to the telecom operator:

- DTMF – tone dialling;
- *pulse (66/34, 50/50, 60/40)* – pulse (decadic) dialling.

6.1.2 ISDN BRA (2B+D) trunk line settings

General settings

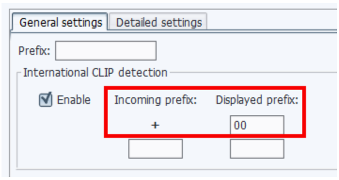
For the ISDN BRA (2B+D) lines in the *General settings* tab the prefix that will be added for all outgoing calls can be defined, the MSN/DDI number that will be used for the users' presentation in the outgoing calls can be assigned to the ISDN BRA (2B+D) line, and the CLIR function can be enabled for this line.



The MSN/DDI number set in *Trunk lines → Privileges* is superior to the MSN/DDI number set in *Extensions → Privileges*. It is recommended to assign full MSN/DDI number in *Trunk lines privileges* and enter up to 4 last digits of the number in *Extensions privilege*s. Thus the last digits of external number will be exchanged on the number's ending entered for the extension and the extension will be presented by its MSN/DDI number.

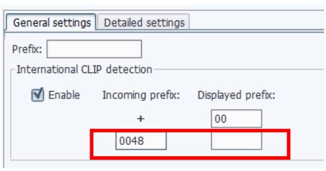
International CLIP detection (enabled by default for all types of lines) makes it possible to serve properly the international incoming calls. If the connection was marked by the operator as an international call (in digital lines only) or the number is preceded by a universal *"+"* symbol, the server will present the number defined in the *Displayed prefix* field.

Example 1



If the incoming call has the information about the international number and the operator sends the number *48585558800 (*without "+*"* or *00)*, with the above settings, the number displayed on the phones will be *00048585558800*, where: the first *0* is the default escape code (access to the external line), and the following *00* were added after the server had received the information about the international call.

Example 2

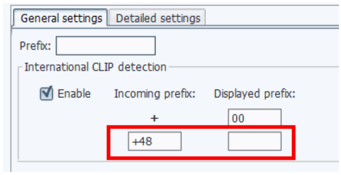


If the incoming call has the information about the international number and the operator sends the number *48585558800 (*without "+*"* or *00)*, with the above settings, the number displayed on the phones will be *0585558800*, where: the first *0* is the default escape code (access to the external line). The server will cut off the *0048* prefix from the presented number. The call from the *49245245245* number (no matter if the operator sends it with "+" or *00*) will be presented as follows: *00049245245245*.

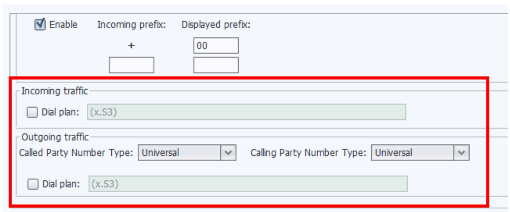
Example 3

The incoming call **does not have** the information about the international number and the operator sends the *0048585558800* number. Then, according to the settings shown in **Example 2**, the number displayed on the phone will be 0585558800, where: the first *0* is the default escape code (access to the external line). The server will cut off the *0048* prefix from the presented number. The call from the *0049245245245* number will be presented as follows: *00049245245245*.

Example 4



The incoming call **does not have** the information about the international number and the operator sends the number *+48585558800*, with the above settings, the number displayed on the phones will be *0585558800*, where: the first *0* is the default escape code (access to the external line). The server will cut off the *+48* prefix from the presented number. The call from *+49245245245* number will be presented as follows: *00049245245245*.



Dial plan for incoming traffic – disabling dial plan for incoming traffic causes receiving dialled numbers as whole numbers, with option enabled dialled numbers will be received digit by digit.

Called Party Number Type – setting the type of called party number sent to the operator.

Calling Party Number Type – setting the type of caller number that will be presented.

The available options of number type:

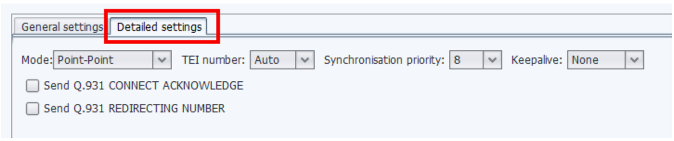
- *Universal* – universal type of dialled number,

- *National* – 9-digit number,
- *International* – 00 + country code + area code + subscriber number.

In the case of selecting *National* or *International* option make sure that the telecom operator handles the selected numbering type.

Dial plan for outgoing traffic – enabling dial plan for outgoing traffic causes sending dialled numbers as whole numbers, with option disabled dialled numbers will be sent digit by digit.

Detailed settings



Mode – defines the operation modes: point – point or point – multipoint.

TEI – assigns the TEI number (*Terminal Endpoint Identifier*) to devices connected to the port. For ISDN trunk line working in point-point operation mode TEI value should be set to "0".

Synchronisation priority – synchronisation priority (1-16). The higher the value, the more important the line, meaning that the synchronisation of other lines will be based on it. Should the same values be set for more than one BRA/PRA equipment, synchronisation will be based on the line that synchronises with the local exchange trunk line as the first one.

Keepalive – options: *None*, *Layer 1*, *Layer 2*.
When enabled, the server checks ISDN Layer 1 (physical) and Layer 2 (data) statuses and activates the layers in the case of hibernation.
It is recommended to set *Keepalive Layer 2* for ISDN lines with the highest synchronisation priority and *Keepalive Layer 1* for the remaining ISDN lines.

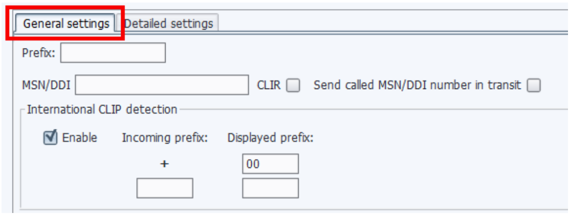
Sending message Q.931 CONNECT ACKNOWLEDGE – sending ACKNOWLEDGE message after every CONNECT frame.

Sending message Q,931 REDIRECTING NUMBER - sending information to NET about redirecting call by the Proxima IP PBX Server.

6.1.3 ISDN PRA (30B+D) settings

General settings

For the ISDN BRA (30B+D) lines in the *General settings* tab the prefix that will be added for all outgoing calls can be defined, the MSN/DDI number that will be used for the users' presentation in the outgoing calls can be assigned to the ISDN BRA (30B+D) line, and the CLIR function can be enabled for this line.

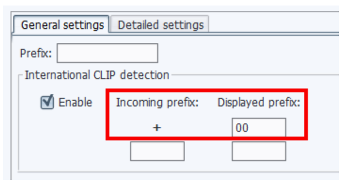


The MSN/DDI number set in *Trunk lines* → *Privileges* is superior to the MSN/DDI number set in *Extensions* → *Privileges*. It is recommended to assign full MSN/DDI number in *Trunk lines privileges* and enter up to 4 last digits of the number in *Extensions privileges*. Thus the last digits of external number will be exchanged on the number's ending entered for the extension and the extension will be presented by its MSN/DDI number.

Send MSN/DDI number in transit – option is visible only if the given PRA line is defined as a transit line. Option makes it possible to transmit the information about the given MSN/DDI number to another PBX system.

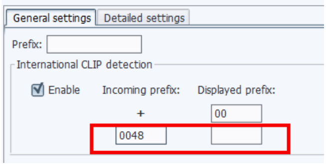
International CLIP detection (enabled by default for all types of lines) makes it possible to serve properly the international incoming calls. If the connection was marked by the operator as an international call (in digital lines only) or the number is preceded by a universal *"+"* symbol, the server will present the number defined in the *Displayed prefix* field.

Example 1



If the incoming call has the information about the international number and the operator sends the number *48585558800* (without "+" or *00*), with the above settings, the number displayed on the phones will be *00048585558800*, where: the first *0* is the default escape code (access to the external line), and the following *00* were added after the server had received the information about the international call.

Example 2

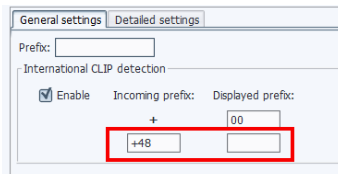


If the incoming call has the information about the international number and the operator sends the number *48585558800* (without "+" or *00*), with the above settings, the number displayed on the phones will be *0585558800*, where: the first *0* is the default escape code (access to the external line). The server will cut off the *0048* prefix from the displayed number. The call from the *49245245245* number (no matter if the operator sends it with "+" or *00*) will be presented as follows: *00049245245245*.

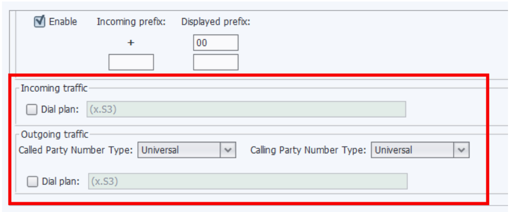
Example 3

The incoming call **does not have** the information about the international number and the operator sends the *0048585558800* number. Then, according to the settings shown in **Example 2**, the number displayed on the phone will be 0585558800, where: the first *0* is the default escape code (access to the external line). The server will cut off the *0048* prefix from the presented number. The call from the *0049245245245* number will be presented as follows: *00049245245245*.

Example 4



The incoming call **does not have** the information about the international number and the operator sends the number *+48585558800*, with the above settings, the number displayed on the phones will be *0585558800*, where: the first *0* is the default escape code (access to the external line). The server will cut off the *+48* prefix from the presented number. The call from *+49245245245* number will be presented as follows: *00049245245245*.



Dial plan for incoming traffic – disabling dial plan for incoming traffic causes receiving dialled numbers as whole numbers, with option enabled dialled numbers will be received digit by digit.

Called Party Number Type – setting the type of called party number sent to the operator.

Calling Party Number Type – setting the type of caller number that will be presented.

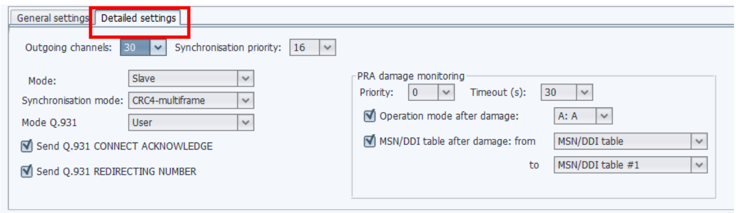
The available options of number type:

- *Universal* – universal type of dialled number,
- *National* – 9-digit number,
- *International* – 00 + country code + area code + subscriber number.

In the case of selecting *National* or *International* option make sure that the telecom operator handles the selected numbering type.

Dial plan for outgoing traffic – enabling dial plan for outgoing traffic causes sending dialled numbers as whole numbers, with option disabled dialled numbers will be sent digit by digit.

Detailed settings



Outgoing channels – number of outgoing channels.

Synchronisation priority – synchronisation priority (1-16). The higher the value, the more important the line, meaning that the synchronisation of other lines will be based on it. Should the same values be set for more than one BRA/PRA equipment, synchronisation will be based on the line that synchronises with the local exchange trunk line as the first one.

Mode: synchronisation operation mode *Slave* or *Master*.

Synchronisation mode: setting synchronisation mode as *CRC4-multiframe* or *Doubleframe*.

Mode Q.931: PRA card operation mode as USER (client) or NET (network side). When connecting two PBX systems by the PRA interface, the synchronisation operatiom mode in the master unit should be set as *Master* and the Q.931 mode as NET. The master system must synchronise with the trunk line. In the slave system the synchronisation operation mode should be set as *Slave* and Q.931 mode as *User*.

When connecting the PBX system to the operator, the synchronisation operation mode should be set as *Slave* and Q.931 mode as *User*.

Send Q.931 CONNECT ACKNOWLEDGE – sending ACKNOWLEDGE message after every CONNECT frame.

Send Q.931 REDIRECTING NUMBER – sending information to NET about redirecting call by the Proxima/Libra PBX Server.

PRA damage monitoring – PRA card activity monitoring. This function will switch the server to a different operation mode, if the PRA card remains inactive for a specific period of time.

Operation mode after damage – the mode to which the server will switch if the PRA card is inactive. This emergency mode should be configured in such a way as to prevent the outgoing calls from using PRA channels of an inactive card.

Priority – with several PRA trunk lines, the priority of changing to *Operation mode after damage* should be defined in the case of inactivity of more than one PRA trunk line. The higher the value, the higher priority for PRA trunk line.

Timeout [s] – time of PRA card inactivity (in sec.), after which the server will consider the PRA card damaged and will switch to *Operation mode after damage*.

Operation mode after damage – the operation mode the PBX server will switch to when detecting the PRA card inactivity. The PBX server emergency operation mode should be configured so that the outgoing traffic does not use the PRA channels on inacvite PRA card.

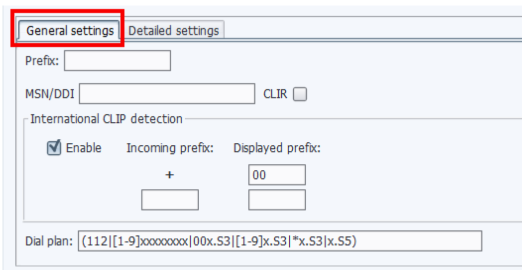
MSN/DDI table after damage – should there be the damage on PRA trunk line detected, the MSN/DDI tables will be switched for the whole system according to these settings, i.e. the table selected on the *from* position will be replaced by the MSN/DDI table selected on *to* position for all trunk lines where the table defined in the *from* field is used.

Should there be problems with activating the ISDN PRA trunk line, the information about the ISDN line parameters, such as for example the synchronisation mode and audio, should be obtained from the telecommunications operator. Next, the above information should be entered in the PRA card configuration tab.

6.1.4 GSM settings

General settings

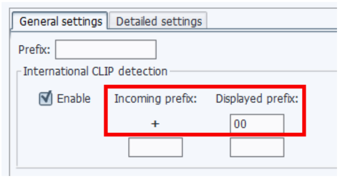
In the *General settings* tab the following settings can be made for the GSM lines: the prefix that will be added for all outgoing calls can be defined, the MSN/DDI number that will be used for users' presentation in the outgoing calls can be assigned to the GSM line, and the CLIR function can be enabled for this line.



MSN/DDI – the entered number will be presented in FLASH SMS message sent. If the field remains empty, the FLASH SMS messages will present the MSN/DDI numbers set to extensions in *Extensions → Privileges*.

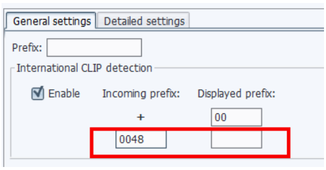
International CLIP detection (enabled by default for all types of lines) makes it possible to serve properly the international incoming calls. If the connection was marked by the operator as an international call (in digital lines only) or the number is preceded by a universal *"+"* symbol, the server will present the number defined in the *Displayed prefix* field.

Example 1



If the incoming call has the information about the international number and the operator sends the number *48585558800 (*without "*+*" or *00)*, with the above settings, the number displayed on the phones will be *000485855558800*, where: the first *0* is the default escape code (access to the external line), and the following *00* were added after the server had received the information about the international call.

Example 2

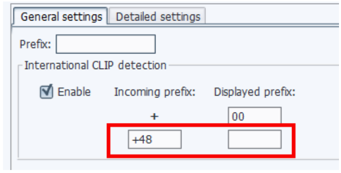


If the incoming call has the information about the international number and the operator sends the number *48585558800 (*without "*+*" or *00)*, with the above settings, the number displayed on the phones will be *0585558800*, where: the first *0* is the default escape code (access to the external line). The server will cut off the *0048* prefix from the displayed number. The call from the *49245245245* number (no matter if the operator sends it with "+" or *00*) will be presented as follows: *00049245245245*.

Example 3

The incoming call **does not have** the information about the international number and the operator sends the *0048585558800* number. Then, according to the settings shown in **Example 2**, the number displayed on the phone will be 0585558800, where: the first *0* is the default escape code (access to the external line). The server will cut off the *0048* prefix from the presented number. The call from the *0049245245245* number will be presented as follows: *00049245245245*.

Example 4

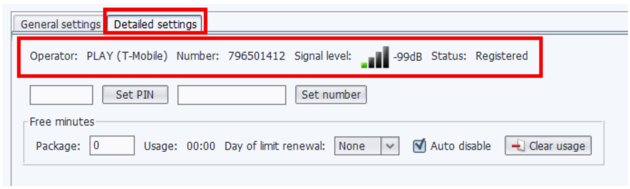


The incoming call **does not have** the information about the international number and the operator sends the number *+48585558800*, with the above settings, the number displayed on the phones will be *0585558800*, where: the first *0* is the default escape code (access to the external line). The server will cut off the *+48* prefix from the presented number. The call from *+49245245245* number will be presented as follows: *00049245245245*.

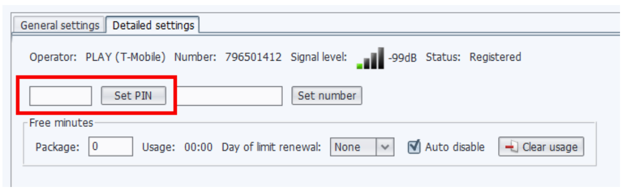
The *Dial plan* should be defined for the GSM lines (the dial plan for Poland is entered by default).

Detailed settings

The GSM line status preview is available in the GSM line detailed settings.

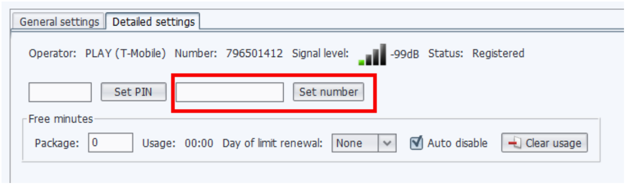


In order to use a given GSM line, a proper SIM card should be activated by sending a correct PIN code to it (*Set PIN* button):

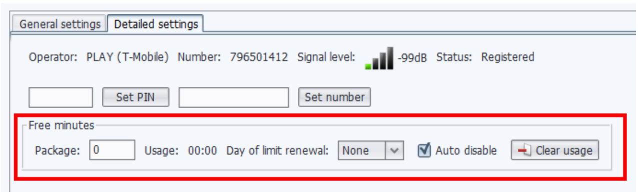


If you send an incorrect PIN code three times in a row, the SIM card will be locked. Therefore, you should make sure that the entered code is correct before you send it to the SIM card.

The SIM card number can also be changed (*Set number* button):



The free minutes package obtained from the mobile operator can be assigned to the given GSM line in the bottom part of this tab. If all free minutes have been used up for the GSM line, and the *Auto disable* option is selected, the server will turn off this line for outgoing calls and notify the user about this fact by a short text message (SMS):



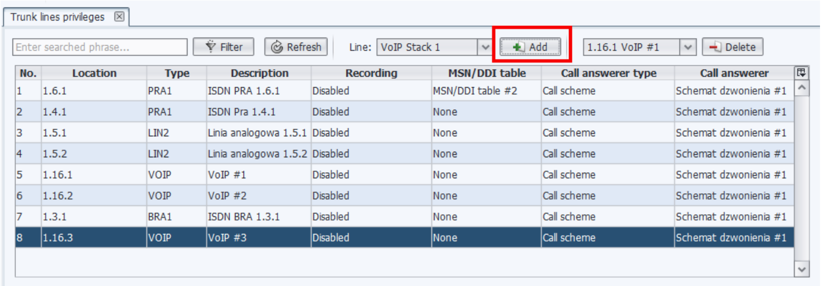
In the same tab you can obtain information about the number of free minutes left and assign the day of the month on which the free minute package will be renewed (*Day of limit renewal*). The free minutes usage may be cleared by clicking the *Clear usage* button.

If the package is to be renewed on the last day of the month, 31st day should be selected.

6.1.5 VoIP settings

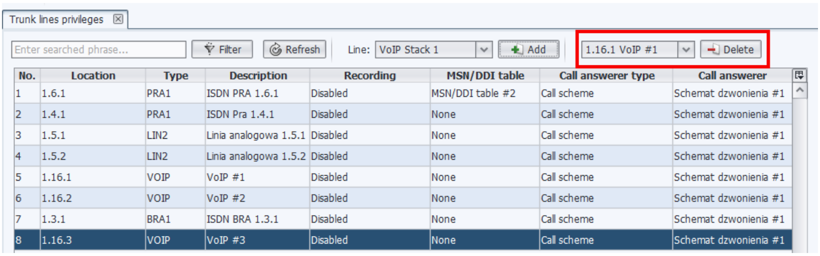
In *Trunk lines* → *Privileges* tab the VoIP accounts can be added and configured to ensure the proper communication with the VoIP operator and the logging status of the ports defined beforehand can be checked.

In order to add a new VoIP account, click *Add* button in *Lines* box, selecting the given *VoIP Stack*:



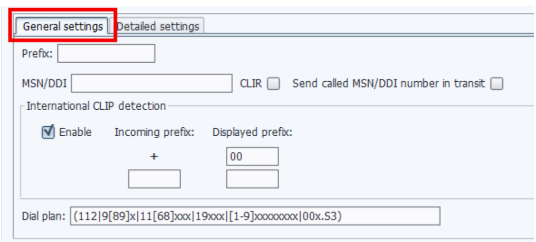
A new VoIP account will appear on the list. Once it has been selected on the list, it can be configured in the *Detailed settings* tab with the data received from the VoIP operator. Up to 64 external VoIP ports/accounts (IP GW) can be configured.

In order to delete the given VoIP account, select it from the drop-down list and click the *Delete* button:



General settings

In the *General settings* tab the following settings can be made for the VoIP lines: the prefix that will be added for all outgoing calls can be defined, the MSN/DDI number that will be used for users' presentation in the outgoing calls can be assigned to this VoIP line, and the CLIR function can be enabled for this line.

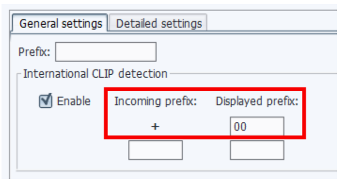


The MSN/DDI number set in *Trunk lines → Privileges* is superior to the MSN/DDI number set in *Extensions → Privileges*. It is recommended to assign full MSN/DDI number in *Trunk lines privileges* and enter up to 4 last digits of the number in *Extensions privilege*s. Thus the last digits of external number will be exchanged on the number's ending entered for the extension and the extension will be presented by its MSN/DDI number.

Send called MSN/DDI number in transit – option is visible only if the given VoIP line is defined as a transit line. Option makes it possible to transmit the information about the dialled MSN/DDI number to another PBX system.

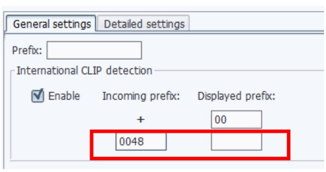
International CLIP detection (enabled by default for all types of lines) makes it possible to serve properly the international incoming calls. If the connection was marked by the operator as an international call (in digital lines only) or the number is preceded by a universal *"+"* symbol, the server will present the number defined in the *Displayed prefix* field.

Example 1



If the incoming call has the information about the international number and the operator sends the number *48585558800 (*without "+*"* or *00)*, with the above settings, the number displayed on the phones will be *00048585558800*, where: the first *0* is the default escape code (access to the external line), and the following *00* were added after the server had received the information about the international call.

Example 2

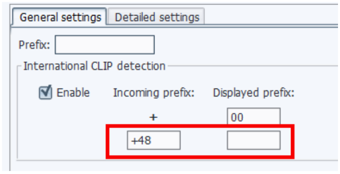


If the incoming call has the information about the international number and the operator sends the number *48585558800 (*without "+*"* or *00)*, with the above settings, the number displayed on the phones will be *0585558800*, where: the first *0* is the default escape code (access to the external line). The server will cut off the

0048 prefix from the displayed number. The call from the *49245245245* number (no matter if the operator sends it with "+" or *00*) will be presented as follows: *00049245245245*.

Example 3
The incoming call **does not have** the information about the international number and the operator sends the *0048585558800* number. Then, according to the settings shown in **Example 2**, the number displayed on the phone will be 0585558800, where: the first *0* is the default escape code (access to the external line). The server will cut off the *0048* prefix from the presented number. The call from the *0049245245245* number will be presented as follows: *00049245245245*.

Example 4



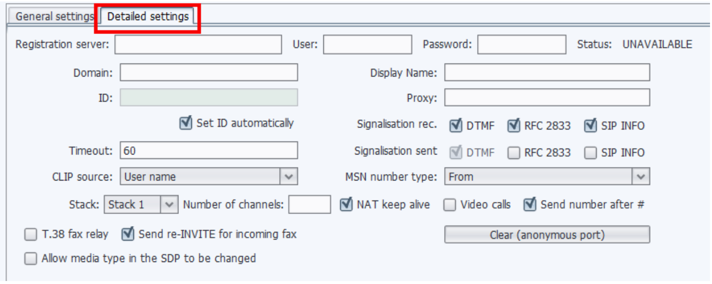
The incoming call **does not have** the information about the international number and the operator sends the number *+48585558800*, with the above settings, the number displayed on the phones will be *0585558800*, where: the first *0* is the default escape code (access to the external line). The server will cut off the *+48* prefix from the presented number. The call from *+49245245245* number will be presented as follows: *00049245245245*.

Dial plan – a dial plan appropriate for the VoIP operator. Dial plan – a collection of contexts defining the sequences of symbols that can be dialled, and the method of executing them.

Dial plan structure:
( ) – dial plan opening and closure,
| – context separator,
x – any single digit,
x. – infinite number of any digits,
[136] – required number 1 or 3 or 6
[5-9] – a required number from 5 to 9 inclusive,
Sy – processing the sequence after y seconds without the need of confirming it with #
Example of a dial plan for Poland:

(112|9[89]x|11[68]xxx|19xxx|[1-9]xxxxxxxx|00x.S3)

Detailed settings



Registration server – a VoIP provider address server.

User – the username/login received from the VoIP provider.

Password – the password received from the VoIP provider.

Status – information about the VoIP account logging status.

Domain – a VoIP provider domain (optional).

Display Name – SIP display name, value sent in the SIP protocol.

ID – an additional authorisation field (automatic by default: *"user"@"registration server"*).

Proxy – proxy server has the task to pass a connection request to the right domain. Proxy directs the connection to another network node basing on the address contained in the INVITE command of SIP protocol. This way of carrying out the connections permits to avoid problems connected with addresses translation (NAT).

Timeout – the time for re-registration.

Signalisation rec. – signalisation received during the ongoing call

Signalisation sent – signalisation sent during the ongoing call

CLIP source – settings concerning the frame in which the information about CLIP number will be sent. *User name* (default) and *Display name* are available.

MSN number type – this setting indicates the way the MSN number is to be sent during the outgoing SIP connection. MSN number can be included in the SIP headings: *From* (default) or *Remote-Party-ID*.

Stack – assigning VoIP stack (set of VoIP cards) to a VoIP trunk line.

Number of channels – defining the maximum number of channels for incoming and outgoing calls for a given VoIP account. The entered value will reserve a given number of channels from a given VoIP stack resources for this VoIP account.

NAT keepalive – client's registration keepalive behind NAT. Function should be enabled if the VoIP operator requests the new registration after the *Timeout* higher than 80 sec.

Video calls – option that makes it possible to make video calls using VoIP port (requires licence).

Send number after # - dialled number confirmation and sending to the operator with the use of the '#' key.

T.38 fax relay – option that enables fax transmission in T.38 standard on VoIP trunk lines.

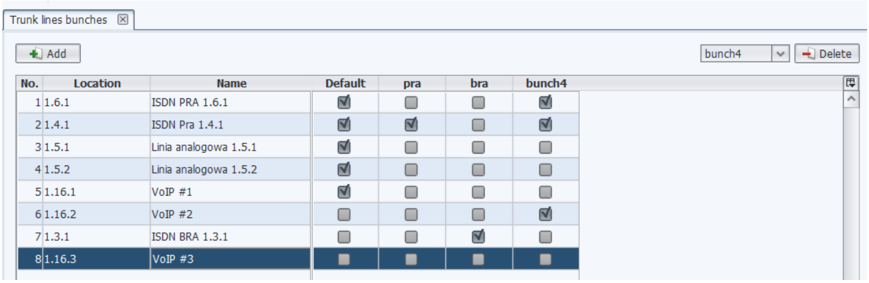
Send re-INVITE for incoming fax – option of sending a request for fax transmission in T.38 standard in the case of detecting a fax transmission with other codec negociated for the established connection. It is recommended to enable this option when using T.38 fax transmission standard.

Note: if the VoIP operator sends re-INVITE command for incoming and outgoing fax transmissions, the option *Send re-INVITE for incoming fax* should **not** be ticked.

Allow media type in the SDP to be changed – option that makes it possible to resend the message initiating connection with the changed type of media (i.e. audio, image, etc.).

6.2 Bunches

In *Trunk lines* → *Bunches* tab create a new bunch of trunk lines that will be used to assign available lines to extensions and to direct outgoing traffic effectively thanks to LCR (*Least Cost Routing*) function.



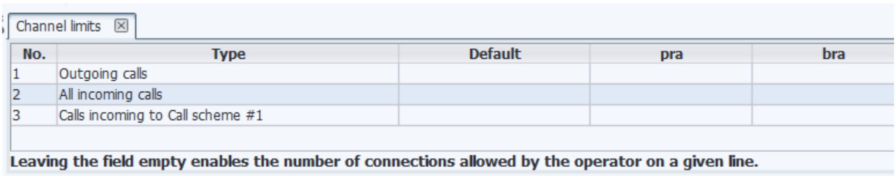
No.	Location	Name	Default	pra	bra	bunch4
1	1.6.1	ISDN PRA 1.6.1	☑	☐	☐	☑
2	1.4.1	ISDN Pra 1.4.1	☑	☑	☐	☑
3	1.5.1	Linia analogowa 1.5.1	☑	☐	☐	☐
4	1.5.2	Linia analogowa 1.5.2	☑	☐	☐	☐
5	1.16.1	VoIP #1	☑	☐	☐	☐
6	1.16.2	VoIP #2	☐	☐	☐	☑
7	1.3.1	ISDN BRA 1.3.1	☐	☐	☑	☐
8	1.16.3	VoIP #3	☐	☐	☐	☐

In order to facilitate the settings, right click on a given Bunch permits to choose the option: All, None and Reverse.

6.3 Channel limits

In *Trunk lines → Channel limits* you can set the restrictions on particular trunk lines bunches in incoming and outgoing traffic, and on call schemes in incoming traffic.

The restrictions make it possible, in the case of the intense incoming traffic, to reserve the channels in order to make also the outgoing calls.



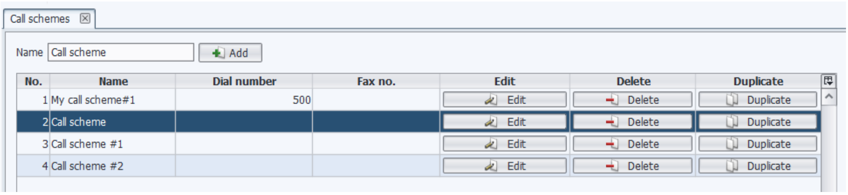
No.	Type	Default	pra	bra
1	Outgoing calls			
2	All incoming calls			
3	Calls incoming to Call scheme #1			

Leaving the field empty enables the number of connections allowed by the operator on a given line.

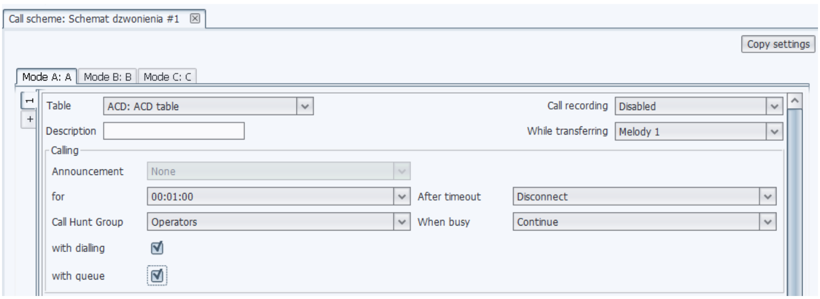
7. Call distribution

7.1 Call schemes

In *Call distribution → Call schemes* tab up to 64 call schemes can be defined, which can be assigned to the servers' trunk lines.



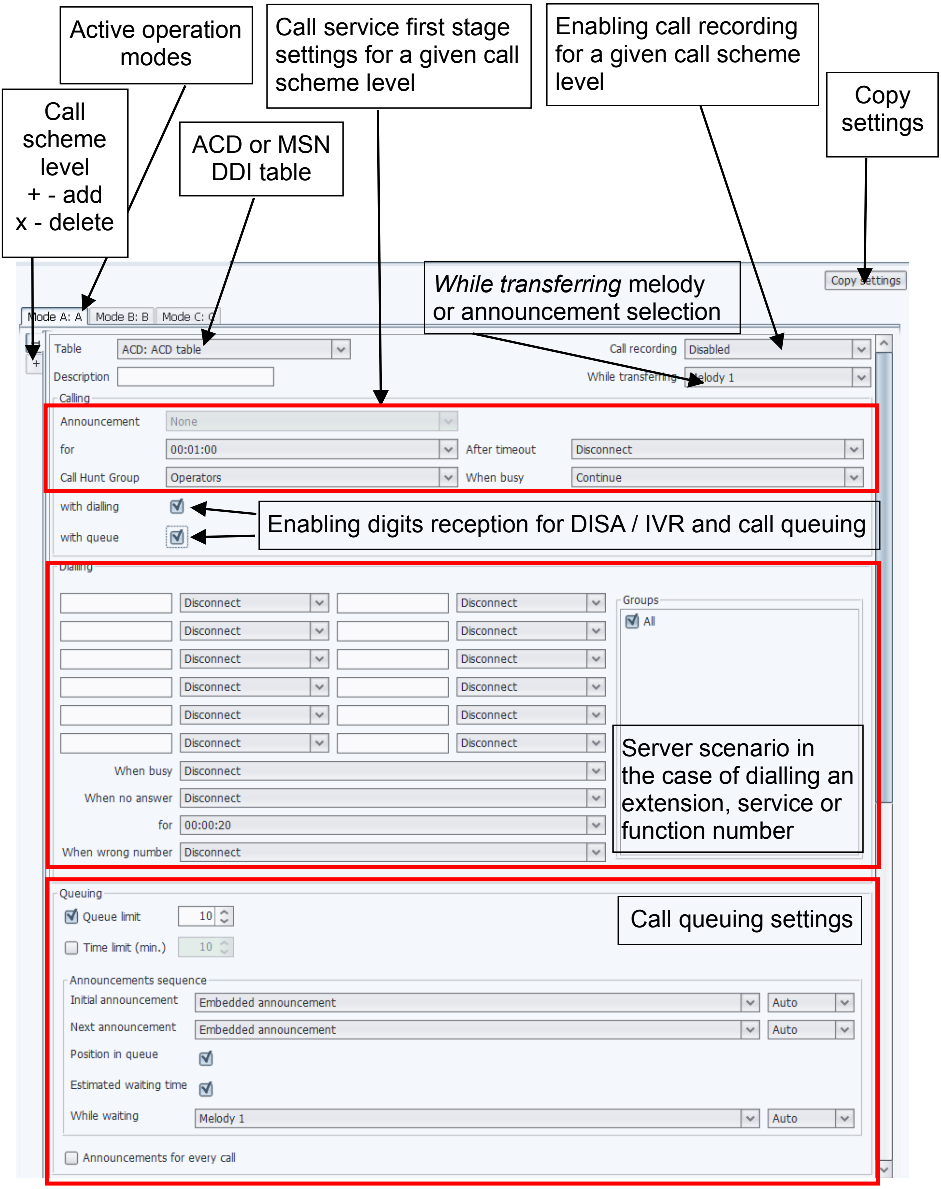
In order to define a new class, enter its name and add it to the list by clicking *Add*. A *Dial number* can be assigned to the call scheme in order to enable the *Call schemes* call transfer, and the *Fax no.* in order to enable automatic fax transfer. Next, using the *Edit* option, enter the required settings.



Before making any changes, the server operation mode, i.e. the time of day in which the settings will apply, must be selected.

In call schemes for every *Operating mode* and every *Call scheme* level the separate *ACD tables* and *MSN/DDI tables* can be defined, the *Call recording* can be enabled and anouncement or melody used *While transferring* can be set.

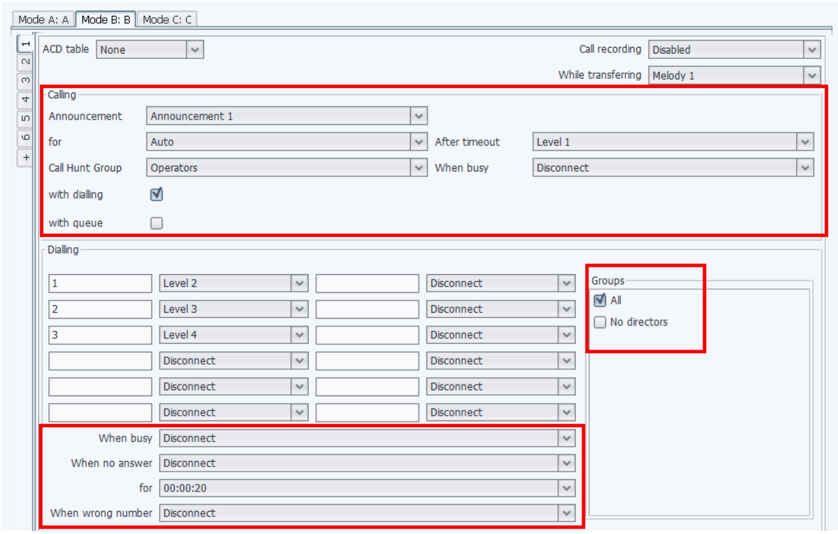
The following modes can be set in *Calling*, *Dialling* and *Queuing* boxes: DISA, IVR, calling on Hunt Groups, Queuing, and mixed mode, using the above modes.



Active operation modes
Call service first stage settings for a given call scheme level
Enabling call recording for a given call scheme level
Copy settings
Call scheme level + - add x - delete
ACD or MSN DDI table
While transferring melody or announcement selection
Copy settings
Mode A: A
Mode B: B
Mode C: C
Table
ACD: ACD table
Call recording
Disabled
Description
While transferring
Melody 1
Calling
Announcement
None
for
00:01:00
After timeout
Disconnect
Call Hunt Group
Operators
When busy
Continue
with dialling
with queue
Enabling digits reception for DISA / IVR and call queuing
Dialling
Disconnect
Groups
All
When busy
When no answer
for
00:00:20
When wrong number
Server scenario in the case of dialling an extension, service or function number
Queuing
Queue limit
10
Time limit (min.)
Call queuing settings
Announcements sequence
Initial announcement
Embedded announcement
Auto
Next announcement
Position in queue
Estimated waiting time
While waiting
Announcements for every call

7.1.1 DISA mode

In the *DISA* (Direct Inward System Access) mode, the external caller will hear an *Announcement (Announcement 1-99)* selected by the user. During that time, a server extension's number can be dialled, but it has to be the number of an extension belonging to designated Extension Groups. In this way, access from DISA to selected extensions can be limited.



In the DISA mode the played announcement together with the DISA mode duration can be set. In the DISA mode time (*for*) one of the following options should be selected:

- *Auto* – the DISA mode duration is equal to the duration of the announcement set in the *Announcement* field.
- time [00:00:01 – 01:00:00] – DISA mode duration. If the set time is longer than the duration of the announcement set in the *Announcement* field, the announcement will be looped. With this option selected, it is recommended to set the multiplication of the announcement duration, so that the selected announcement will be looped and played in total.

The digits reception when playing the announcement should be enabled (*with dialling* option ticked) and the *Groups* to which one can call from the DISA mode should be defined.

Additional scenarios can be defined with the following options:

- *After timeout* – if a caller during DISA mode duration did not dial any digits
- *When busy* – the called number is busy
- *When no answer* – (*for*) a defined time
- *When wrong number* – when the dialled number is not assigned in the server numbering plan.

The DISA time is usually set a few seconds longer than the recorded announcement. Thanks to that, the external caller can listen to the whole announcement and has time to dial the extension number. It is also recommended to record an announcement starting and ending with a moment of silence.

7.1.2 IVR mode

The Proxima/Libra server makes it possible to handle the incoming calls in the IVR (*Intelligent Voice Response*) mode. It is a very useful service, especially for companies with complex structures.

In order to use the IVR mode, an external caller must be equipped with a phone set with the tone dialling function.

99 announcements with the total duration of up to 30 hours are available in the server by default.

Recording of own announcements and melodies is described in *Proxima IP PBX Server/Libra PBX Server User manual*.

IF THE DIALLED EXTENSION NUMBER IS ENGAGED OR IS NOT ANSWERING, YOU CAN PRESS "*" AND DIAL A DIFFERENT EXTENSION NUMBER.

A sample IVR design is presented below. It is a simplified project because it does not aim at showing numerous configuration possibilities offered by the IVR but at explaining the methodology of operation and the general principle of its functioning.

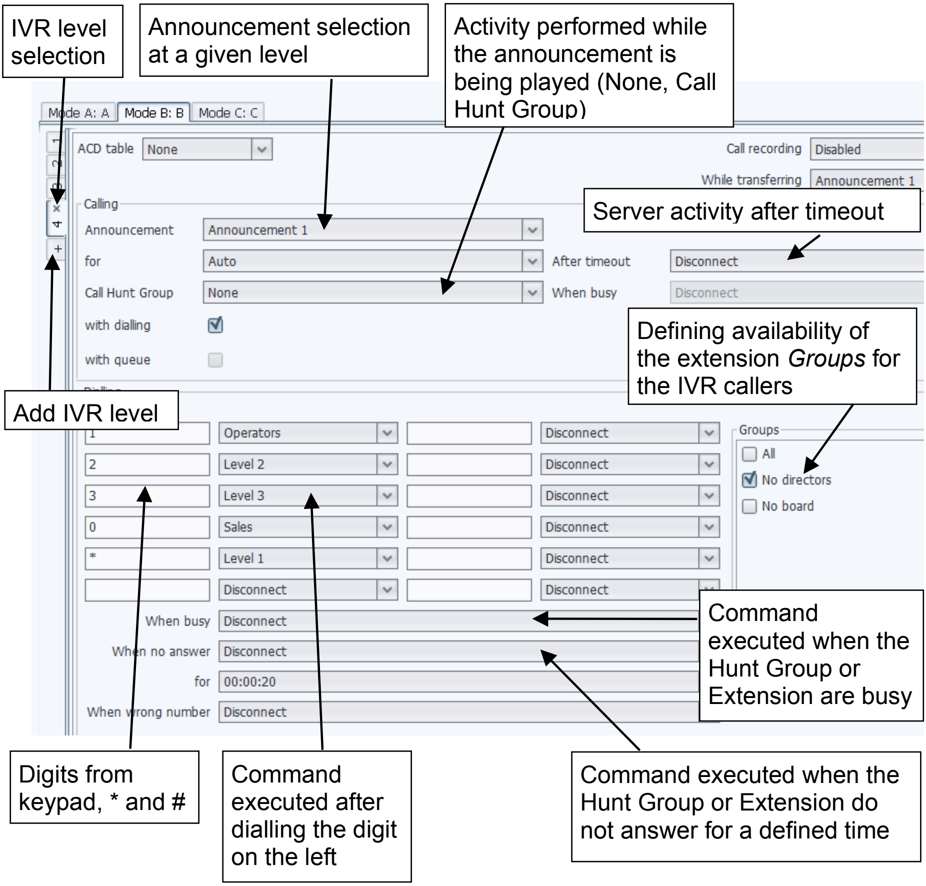
EXAMPLE:

The following abbreviations have been used in the example:

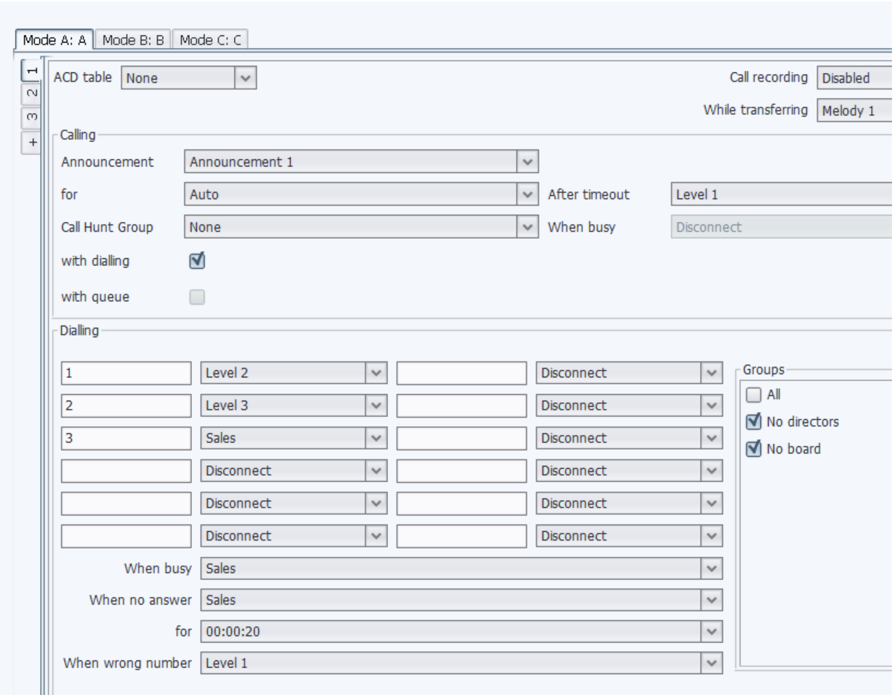
ExC – external caller
HG – Hunt Group

Principles of a sample IVR:

There are three Hunt Groups. The first one informs about product A, the second about product B and the third group is the Sales Department. From the basic level, an external caller can decide whether they want to obtain information about one of the products or to connect directly with the Sales Department. Connection with the Sales Department will be established with a server ring-back tone in the background, while calling information groups – with a melody. If a given Hunt Group is busy or does not answer for the declared time, the server should answer with an announcement about the consultants being busy, the caller's queue position and the expected waiting time. The system announcements will be played for a declared max. waiting time (*Queuing* box → *Time limit*).

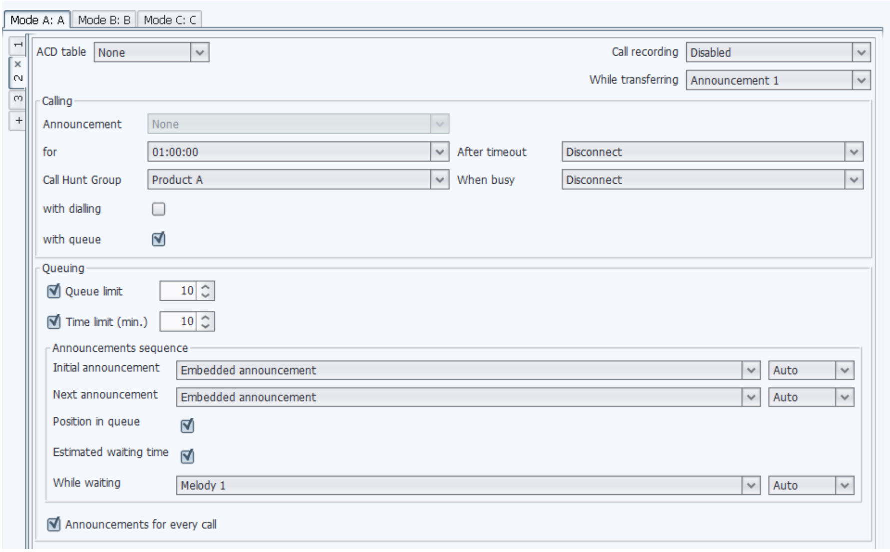


Level 1:



The principle is that an External Caller (ExtC) can obtain information about product A (by dialling 1 and calling the Hunt Group 1) or product B (by dialling 2 and calling the Hunt Group 2), or contact the Sales Department by dialling 3 and calling the Hunt Group 3. If the Hunt Group 3 (Sales) does not answer for a specific period of time or is busy, the server will play the ringback tone and will call the Hunt Group 3 (Sales) again.

Level 2:

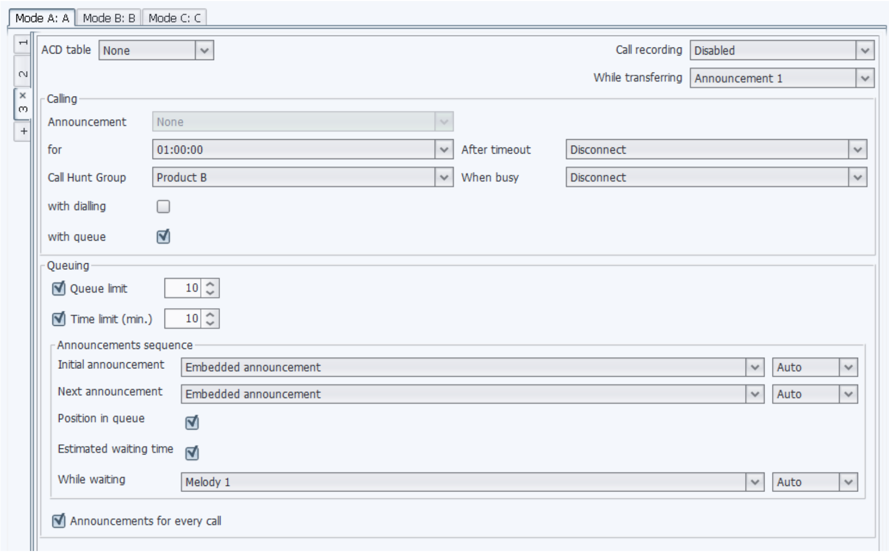


On this level, the server calls the HG 1 (Product A), while the ExtC hears the ringback tone if the Hunt Group members are idle or in the case they are busy, but the option *Continue* in *When busy* field is set.

If the *with queue* option is ticked and the Hunt Group is busy, the server responds (for the time set in the *for* field) with announcements about the consultants being busy, the caller's queue position and the expected waiting time.

Level 3:
Level 3 is analogue to Level 2. The server calls the HG 2 (Product B), while the ExtC hears the ringback tone if the HG members are idle or in the case they are busy, but the option *Continue* in *When busy* field is set.

If the *with queue* option is ticked and the Hunt Group is busy, the server responds (for the time set in the *for* field) with announcements about the consultants being busy, the caller's queue position and the expected waiting time.



7.1.3 Hunt Group calling mode

In this mode, the server directs external incoming calls to extensions selected in the *Calling* box → *Call Hunt Group* field for the time defined in the *for* field. A caller hears the ring back tone, and in the case of the Hunt Group being busy, the action defined in the *When busy* field will be executed. The following actions are available: different call scheme level, different call scheme, calling different Hunt Group, continue calling the busy Hunt Group (the caller hears ring back tone) or disconnect. One of the above actions can be also assigned in the *After timeout* field.

In the Hunt Group calling mode the call queuing mode can be also enabled (licence required).

7.1.4 Call queuing mode

Call queuing mode requires a queuing licence.

Call queuing mode makes it possible to handle more calls than the Hunt Group members can answer at the same time. The callers who do not get through will hear the message about their position in queue and estimated waiting time.

The following waiting call limits can be set in the queue option:

- *Queue limit* – the surplus calls number handled by a Hunt Group. Up to 40 people can be served by a queue.
- *Time limit (min.)* – max queue waiting time. If the actual waiting time is longer than this limit, the server will not admit the next call to the queue, even if the *Queue limit* has not been achieved.
- *Announcements for every call* – with this option ticked, the calls directed to the free Hunt Group members, instead of a ring back tone, will be admitted to the queue and the announcements about a position in queue and estimated waiting time will be played, according to the set options.

Limits can be used individually or simultaneously. In the latter case, the exceeding of one limit will suspend the calls admission to the queue.

Note! In the case of enabling a call queuing mode, it is recommended to set the max calling time (1h) in the *for* field. This parameter is a global waiting time for a connection with a Hunt Group. If it is set for instance on 60 sec, the server after 60 sec. of queue waiting time will execute the scenario set in *After timeout* field.

Note! The estimated waiting times are reset after the Proxima IP PBX Server/Libra PBX Server restart and the system learns again the traffic distribution in a given queue.

In the call queuing mode the announcements sequence can be set together with their duration. The *Auto* parameter (recommended) sets the duration equal the recorded announcement length. They are two options at disposal: *Initial announcement* and *Next announcement*. Default factory announcements are:

Initial announcement: *"All consultants are busy"*.
Next announcement: *"All consultants are still busy"*.

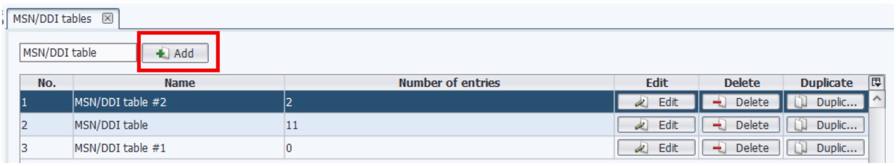
Additionally, it is possible to enable the information about the caller's position in a queue, estimated average waiting time and additional melody between all the announcements.

Exemples of voice announcements, their recording and listening to methods are described in the *Proxima IP PBX Server / Libra PBX Server User manual*.

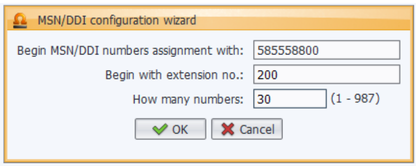
7.2 MSN/DDI tables

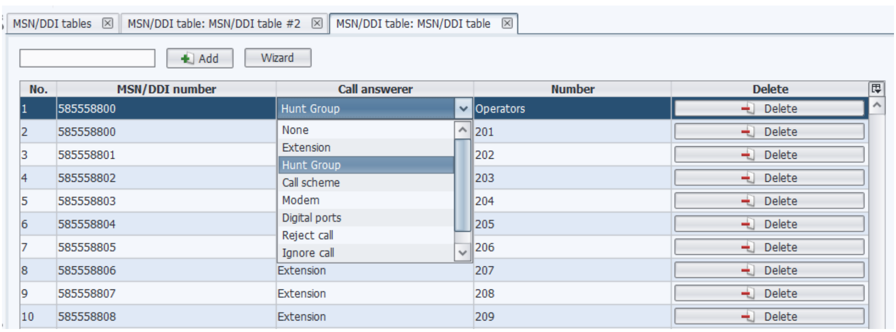
All numbers (the leading number, the MSN / DDI digital numbers) have equal status for the server. It is the installer that decides while programming the server how it will react to external incoming calls. The tab below illustrates the process of handling calls incoming through the ISDN line. Up to 1000 MSN/DDI numbers can be used.

In order to create a new *MSN/DDI table*, click **Add**.



Select a given MSN/DDI table from the list and click *Edit* button to set the MSN/DDI parameters. You can use a MSN/DDI configuration wizard to facilitate the process:





⇒ **MSN/DDI number** – here you enter MSN or DDI numbers, depending on which of them have been assigned by your telecommunications operator. You do not need to enter the whole number; final leading numbers are enough. In this field, you enter as many digits of the MSN/DDI number as required for the dial plan to be clear, i.e. if there are 10 DDI numbers ranging from 0 to 9, it is enough to enter the last digit; if there are 100 DDI numbers ranging from 00 to 99, at least the last 2 digits must be entered.

The case is similar with the telecommunications operator – it must provide as many digits as to make it clear what the right number is.

⇒ **Call recipient** – a called party type for an incoming call is defined; it can be an extension or a Hunt Group, the call can be handled in accordance with a Call scheme, sent to a modem, a PIN number (if defined), a digital or audio (for Libra PBX Server only) output from the I/O card (after having defined at least one code: activation, deactivation, switch over, temporary activation), it can be also rejected or ignored.

If no code is defined for a given digital port in *External devices control (I/O) → Digital ports*, it will be not possible to assign an MSN/DDI number to it.

⇒ **Number** – the precise call recipient is defined, depending on the call recipient type set in the previous column – it can be an extension number, a Hunt Group name or a defined call scheme for incoming calls, according to which a given call will be handled, a given PIN number (if defined) or a code defined in the I/O card outputs.

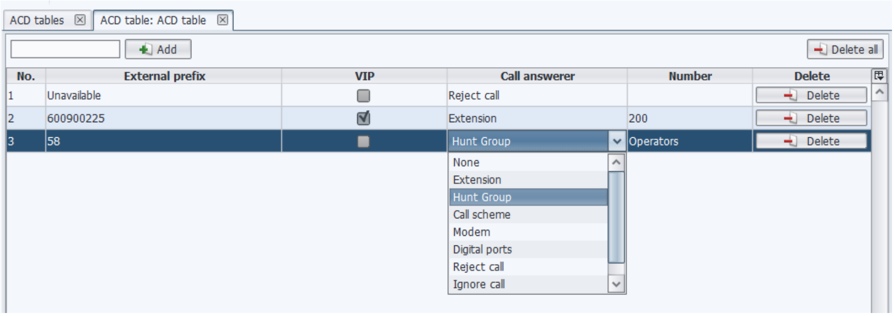
If a call is directed to a *Call scheme*, the scheme should be configured in *Call distribution → Call schemes*.

It should be remembered that when MSN numbers are called from the external lines, an engagement tone will not be heard if the called extension number is engaged. Nevertheless, an engagement tone will be heard when calling DDI numbers, but only if the *No waiting call indication* option is ticked for the given *Service class*.

7.3 ACD table – automatic call distribution

The ACD function directs incoming calls to selected extensions based on the ACD table.

The ACD tables are defined in *Call distribution → ACD tables*. Up to 100 ACD tables can be defined with totally up to 1000 entries. A new table with its name and number of entries will be added after clicking **Add**. An **External prefix**, a **Call recipient** and an recipient 's **Number** should be entered for every table.



External prefix

An incoming external number prefix or the whole number should be entered in the field. Therefore, if 5 is entered, all numbers identified by CLIP starting with 5 (e.g. 58..., 503...) will be directed according to the further configuration.

An *Unavailable* option represents the ex-directory numbers.

VIP

Ticking *VIP* field gives the defined callers a priority in the queue (if call scheme with call queuing mode is enabled).

Call recipient

In this field, the call recipient type for a given call is selected from a drop-down list. It can be:

⇒ *Extension* (the call will be directed to the selected server extension).

⇒ *Hunt Group* (the server will direct the call to the selected group).

⇒ *Call scheme* (the call will be handled according to the call scheme defined in *Call distribution → Call schemes*).

⇒ *Modem.*

⇒ *Digital ports* on I/O card.

⇒ *Audio ports* on I/O card (in Libra PBX Server only).

⇒ *Reject call* – for analogue line the call from that line will be ignored.

⇒ *Ignore call.*

⇒ *Trunk line transit* can be made on ISDN PRA or VoIP trunk line.

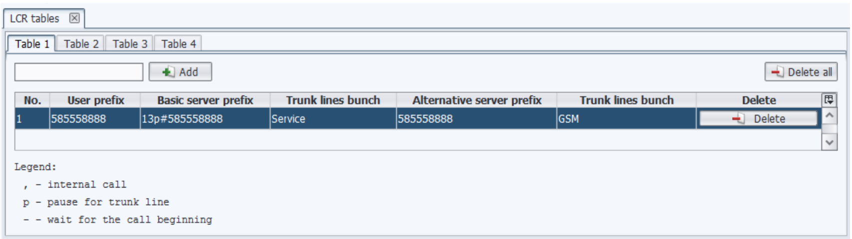
Number

In this column, the number for the call recipient defined beforehand should be entered. If a *Call scheme* or a *Hunt Group* has been selected as the *Call recipient* , its name should be entered here, etc.

Only numbers defined in the server numbering plan can be entered in the ACD table.

7.4 LCR tables

Up to 2048 User prefixes can be defined in Proxima IP PBX Server / Libra PBX Server. They can be allocated freely in four LCR tables in *Call distribution → LCR tables*.



LCR table is related to the trunk line access through a given trunk lines bunch configured in *Extensions → Traffic classes [Edit]*

7.4.1 Functionality

LCR *(Least Cost Routing)* is designed to reduce the costs of phone calls. Effectiveness of this service depends on the configuration of extensions and of the *LCR tables***.** The LCR function consists in the server automatically analysing the external number dialled by an extension. After the analysis, the server (in accordance with the *LCR tables*) directs the call through the indicated trunk lines. It can also modify the dialled number. The call will be established if the extension is privileged to use a given trunk line.

The LCR function is active only in the case of outgoing external calls made after having dialled (in default settings):

0 – line access through the main trunk lines bunch set in the configuration
*81 – line access through the first trunk lines bunch set in the configuration
*82 – line access through the second trunk lines bunch set in the configuration
*83 – line access through the third trunk lines bunch set in the configuration
*84 – line access through the fourth trunk lines bunch set in the configuration.

Note. These are **default function codes**. If the numbers had been changed *(Common settings → Function codes),* the proper ones should be used.

The **LCR table** is analysed by the server after each digit dialled by the extension. The process is concluded (the trunk line bunch is assigned) when dialling any other digit will not change the decision made by the LCR system. A longer prefix entered in the LCR table is "more important" than the shorter one. The time for dialling a number by an extension and for making a decision by the LCR system is limited (the so-called timeout), i.e. after a specific period of time, the caller will hear an unavailability tone.

It is possible to define in the LCR tables an alternative route through which the call is to be made if the first route is engaged or damaged. In such a case, alternative trunk line bunches are entered and prefixes are modified.

The extension's privileges to trunk lines included in the trunk line bunch indicated by the LCR table should be checked. Access to trunk lines for extensions is defined in *Extensions privileges* (and selected *Traffic classes*). Before establishing a call, the server checks which trunk lines (from among the ones indicated by the LCR table) a given extension is authorised to use. Only via such lines can the call take place.

Such a solution makes it possible to use the LCR service as a specific call discriminator.

7.4.2 LCR prefix table

Effectiveness of this service depends on the LCR tables configuration. When configuring the LCR tables, you will encounter such notions as a *Trunk lines bunch*, a *User prefix*, a *Basic server prefix* and an *Alternative server prefix*.

User prefix is the initial part of the external number dialled by a server extension, informing for example about the telecommunications operator.

Basic server prefix is a prefix entered in the LCR table, which will be in fact dialled by the server. If an **Alternative server prefix** is defined, should a connection with a Basic server prefix not be established (e.g. if the prefix is unavailable or a given trunk lines bunch is either engaged or damaged), the server will try to use an alternative configuration.

When configuring the LCR table, it should be remembered that trunk lines can be assigned to different trunk lines bunches. One trunk line can be used in more than one trunk lines bunch.

7.4.3 Defining the LCR tables

Defining the LCR table consists in configuring user prefixes, basic and alternative server prefixes and trunk lines bunches properly and logically. Before starting the configuration, preparation of its initial project is recommended.

A new server configuration contains four empty LCR tables. One of them should be selected. In order to add a rule, click **Add** and introduce necessary settings. The LCR table allows up to **2048** lines with LCR rules to be added. These 2048 lines with LCR rules can be freely divided into four tables. The same user prefix can be used in different tables, but it cannot be repeated in one table.

7.4.4 Sample LCR table project

It is an example and should be used only for training purposes.

! To be in line with the idea behind the LCR service, it is recommended that no more than one operator be assigned to one trunk lines bunch within the LCR table.

Example of defining Trunk lines bunches for LCR

	Defined Trunk lines bunches (TLB)							
	"for LCR":				**available trunk lines "for extension"**			
	All lines	Operator A	Operator A (II)	GSM Gateway	Board	Sales	Production	Common Room
1	X	X			X	X	X	X
2	X	X			X	X	X	
3	X		X		X			
4	X			X	X	X		
	TLB=1	TLB=2	TLB=3	TLB=4	TLB=5	TLB=6	TLB=7	TLB=8

TLB – Trunk lines bunches defined in *Trunk lines → Bunches.*
The server is connected to a public network via trunk lines bunches (TLB):

TLB = 1	all lines
TLB = 2	operator A landline network (Operator A)
TLB = 3	operator A landline network (Operator A [II])
TLB = 4	GSM gateway to mobile network (GSM gateway)

Sample LCR table

LCR Table

Add | Delete all

No.	User prefix	Server prefix	Trunk lines bunch	Alternative prefix	Trunk lines bunch	Delete
1	#57	585558800	Operator A	585558800	Operator A [II]	Delete
2	*57	600900500	Operator A	600900500	Operator A [II]	Delete
3		1090	Operator A	1090	Operator A [II]	Delete
4	00	708108107	Operator A	708108107	Operator A [II]	Delete
5	50	50	GSM	50	Operator A [II]	Delete
6	51	51	GSM	51	GSM	Delete
7	60	60	GSM	60	Operator A [II]	Delete
8	66	66	GSM	66	Operator A [II]	Delete
9	69	69	GSM	69	Operator A [II]	Delete
10	88	88	GSM	88	Operator A [II]	Delete
11	1	1	Operator A	1	Operator A [II]	Delete
12	9	9	Operator A	9	Operator A [II]	Delete

Caption:
, - internal call
p - pause for trunk line
- - wait for the call begining

When defining external numbers in the LCR table, the following symbols can be used:

"–" (dash) – waiting for the call to be initiated
"," (comma) – extension number
"p" – pause

Note to configuration data:

Line 1:
Calls to an external (e.g. home) number of a server extension number 57 (dialling '0' and then the pre-arranged '#' symbol before 57) can be carried out via the Operator A landline network (Operator A) – physical trunk lines No. 1 and 2. Thanks to the **number swap,** an extension user does not have to dial full number (and does not even have to know it). If those lines are unavailable, an alternative trunk lines bunch can be used – on the Operator A landline network (Operator A [II]) – physical trunk line No. 3. Only the extensions that have been assigned TLB Board have access to this trunk lines bunch and only for them can the alternative call be established (*discrimination* is practised by means of restricting access to trunk lines).

Line 2:
Calls to a mobile number of a server extension number 57 (dialling '0' and then the pre-arranged '*' symbol before 57) can be carried out via the GSM Gateway (trunk line No. 4, not available to Production and Common Room). Thanks to the number swap by LCR, an extension user does not have to dial full number (and does not even have to know it).

Line 3:
Calls are directed through the first two trunk lines of the server to the Operator A landline network, with the Operator B prefix added (10xx) – **pre-selection** is made (also for local calls). It should be noted that the extensions that have been assigned TLB Common Room will not be able to make a call when the server establishes a connection with the second trunk line. In the case of an alternative connection, the call will be made via Operator A [II] landline network.

Line 4:
International calls are directed through trunk lines No. 1 and 2 (Operator A landline network) using the access number to the foreign operator gate.
In the case of an alternative connection, the call will be made via Operator A [II] landline network, **omitting the prefix** of the access number. Only the extensions that have been assigned TLB Board will be able to make such a call.

Line 5, 6, 7, 8, 9, 10:
The server establishes calls to GSM operator subscribers via the GSM Gateway installed on the trunk line No. 4 (not available to Production and Common Room). The specificity of GSM operators requires the ommitting of the leading '0' in the target prefix, which is restored for the alternative connection.

Line 11, 12:
Emergency calls (like 112, 9xx) and calls to other special numbers are carried out without dialling area numbers.

In addition:
An empty server prefix (empty field) and a selected trunk lines bunch can also be assigned to a user prefix. Such settings may be used to delete the user prefix from the whole dialled number.

In the LCR table, an empty field can also be set as the user and the server prefix, and a trunk line bunch can be selected. In such a case, every number dialled by extensions that is not in accordance with the prefixes defined in the LCR table is dialled with no changes and the call is established via a specific trunk line bunch.

If there is no alternative route available, the **NONE** option should be selected in the LCR table, in the field for selecting an alternative trunk lines bunch.

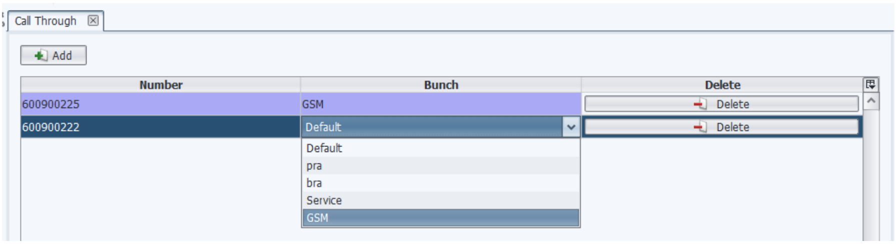
The LCR table functions only on the trunk lines bunches – i.e. Line access after having dialled 0, *81, *82, *83, *84.

In order not to make it possible for the LCR table to be omitted by the users, the defined LCR tables should be assigned to all outgoing trunk lines bunches in *Extensions → Traffic classes [Edit] → LCR* box.

Regardless of the LCR table settings, restrictions concerning outgoing calls according to the *Allowed/Denied numbers* tables apply.

7.5 Call Through

Call Through – a function enabling the users whose external phones, e.g. mobile phones, are defined in the ProximaWeb/LibraWeb, to make incoming calls from external lines at the expense of the server.



In order to use the Call Through function within the server, incoming calls must be directed to the call scheme working in DISA mode, i.e. with the announcements and the *with dialling* option enabled.

The defined telephone numbers should be saved in ProximaWeb/ LibraWeb as 9-digit numbers, e.g. 600200200 or 585558800.

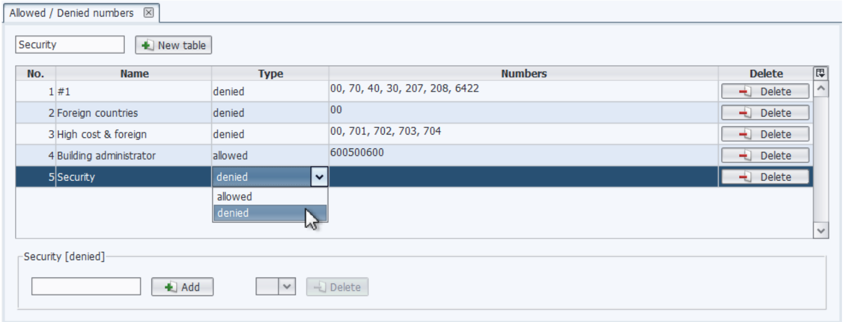
7.6 Allowed / Denied numbers

Specific external numbers, e.g. long-distance calls, can be either blocked or made available for a given user. You can define sets of external numbers and prefixes which, if dialled, will result in calls either being or not being established. **16 tables** to be defined in any way are available in the system.

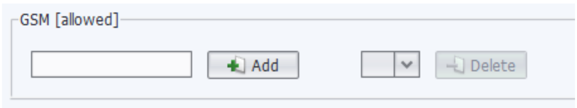
The created Tables can be used in two ways:

⇒ **to deny** users calling certain numbers,

⇒ **to allow** users calling certain numbers.



A new table should be added in the *Allowed / Denied numbers* tab. Next, numbers should be added / deleted at the bottom. The numbers entered cannot be longer than 16 characters.



Applying the Table to a given traffic class is set in the *Traffic class* tab, in the *Restrictions* drop-down list. The name of the table with restrictions is specified.

Table setting: **Denied**

⇒ If the digit entered is "5" the server will not establish a call to any external number starting with "5".

⇒ If the number entered is "701" this function will work as follows:
  - The server will allow the user to dial the first digit with no interference ("7" can also be dialled).
  - The case will be similar with the second digit (even if "70" is dialled).
  - If the third digit dialled is "1" and:
    - it was preceded by "70" – the server will interrupt the call and an unavailability tone will be heard.
    - it was not preceded by "70" – the server will allow the call to be established.

Table setting: **Allowed**

⇒ If the number entered is "5", the server will establish only the connections to numbers starting with "5".

⇒ If the number entered is "58", the server will establish only the calls to numbers starting with "58".

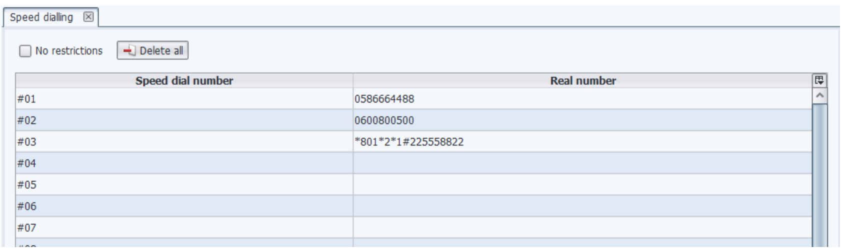
Denied and allowed numbers are defined for external numbers. This means that, when defining e.g. an external number 585558888, “0” (or any other escape code – access prefix to trunk lines) is not added at the beginning.

7.7 Speed dialling

The server users can use the common Table of speed dial numbers containing up to **1000 real numbers** not longer than **24 characters** (0-9 digits, #, * and p-pause, approx. 1 second long). **Speed dial** number can be up to **16 characters** long (0-9 digits, #, *). When using speed dial numbers, individual extensions' restrictions concerning outgoing calls apply (i.e. access to trunk lines, restriction tables, etc.). <u>External numbers should be preceded by an outgoing trunk line access prefix</u>. Prefixes define the trunk line bunch through which the server is to establish a call with a local exchange (0, *81-*84). A prefix for a specific trunk line can also be used, i.e. *80a*b*c#, where a – unit number, b – slot number, c – card port number.

Note. These are **default function codes**. If the numbers had been changed *(Common settings → Function codes)* the proper ones should be used.

The real number should begin with the access code to the main or any trunk lines bunch defined in *Common settings → Function codes*. If no number is defined as the escape code (access through the main trunk lines bunch) the real number should be entered without it.



If the *No restrictions* option is ticked, no restrictions will be checked in the tables of *allowed/denied numbers* after dialling any of the defined speed dial numbers.

7.8 Emergency numbers

In *Call distribution → Emergency numbers* you can define the numbers that will be dialled by extension users and target emergency numbers that will be sent by Proxima IP PBX Server/ Libra PBX Server via a defined trunk lines bunch. You can define up to 16 emergency numbers in the server.

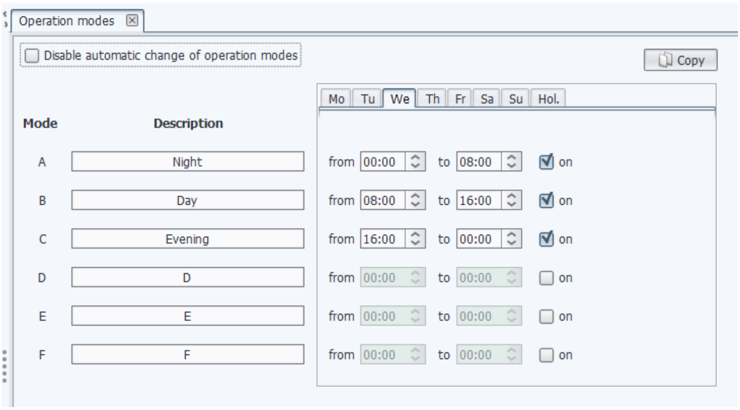
Emergency numbers

Dialled number	Emergency number	Bunch
112	112	Default
997	997	Default
998	998	Default
999	999	Default
		None

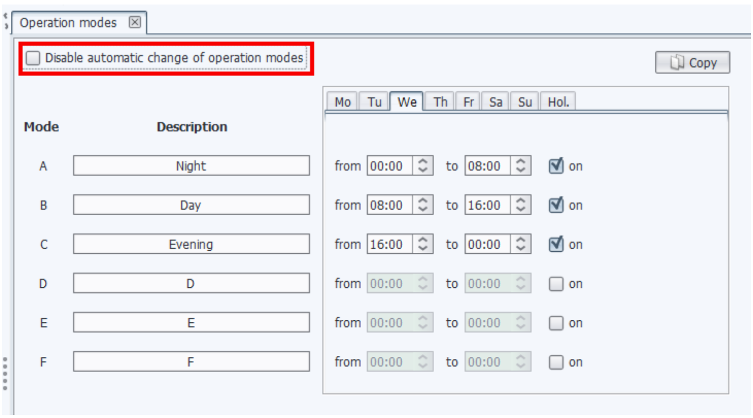
8. Common settings

8.1 Operation modes

The server can operate in six modes, i.e. every day of the week can be divided into the maximum of six periods. Operation modes are configured in the program *(menu: Common settings → Operation modes)* In fillable fields, you should enter the time at which a given mode will begin and end. In order to activate an operation mode, the **"ON"** field should be ticked.



The automatic change of operation modes can be enabled/disabled in Proxima/Libra servers. It can be done in the ProximaWeb/LibraWeb:



The automatic change of operation mode can be enabled/disabled also with the use of the following codes entered from the phone:

⇒ *764 – disable automatic change of operation modes

⇒ *765 – enable automatic change of operation modes.

8.1.1 Any day of the week

Operation modes can be defined as needed. Operation modes can be nested *(Example 2)*. It should be then remembered that the server "reads" the modes from F to A. Therefore, when nesting them, you should start with the longest (mode A) and end with the shortest mode (e.g. mode C according to the example below).

EXAMPLE 1
Mode A from 08 to 16
Mode B from 16 to 08

EXAMPLE 2
Mode A from 00 to 00
Mode B from 08 to 18
Mode C from 12 to 13

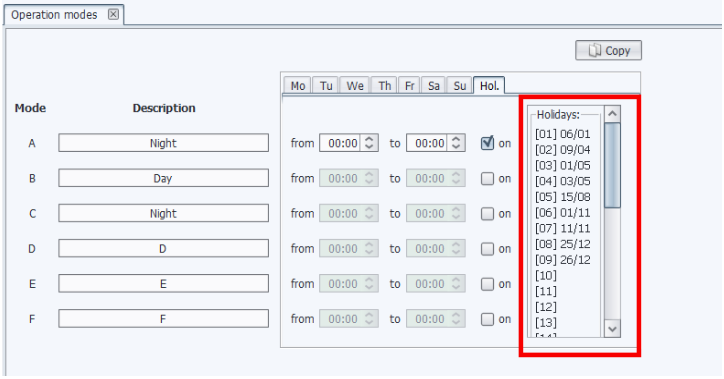
Should the server operation modes overlap, the server will not inform us about it.

In *Example 1*, the server operation within the company's working hours has been set to mode A and the server operation outside the company's working hours – in mode B.

One operation mode can be configured in the case of an ISDN PRA trunk line damage – see menu: *Trunk lines → Privileges [ISDN PRA] → Detailed settings [Operation mode after damage]*.

8.1.2 Holidays

Apart from standard days of the week, the server operation modes can also be defined on holidays if the company's working hours are different than for example on Sunday. In such a case, you should configure the **Holidays** tab and fill in the **Holidays** field.



After double-clicking any item in the Holidays list, the **Select day** window is displayed in which you select the required month using arrows, then click the selected day and confirm the holiday with **OK**.

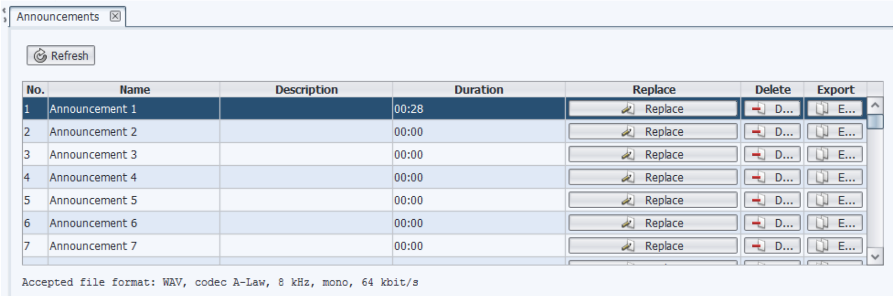
Information about the current operation mode can be found in: *Monitor → Server*.

8.2 Announcements

The server records 99 voice announcements (up to 30 hours of voice announcements in standard – for Proxima plus and for Libra with controller 2.7 and highers; up to 1 hour for Proxima and for Libra with controllers lower than 2.7) that can be used when configuring the **DISA, IVR and Announcement** modes in the *Call schemes*. Announcements are recorded by the user or by the server administrator. A **WAV file** can be used as an announcement (A-Law codec, 8 kHz sampling, 64 kbit/s). It only needs to be uploaded to the server *(Common settings → Announcements)*. Announcements can be recorded and played using a phone connected to the server. This procedure is described in the *Proxima IP PBX Server/ Libra PBX Server User Manual*.

With the *Export* option a copy of the announcement can be made to a *wave* file.

Apart from the announcements, the server makes it available 4 Melodies, a Ringback tone, a Route and a Three-tone signal.

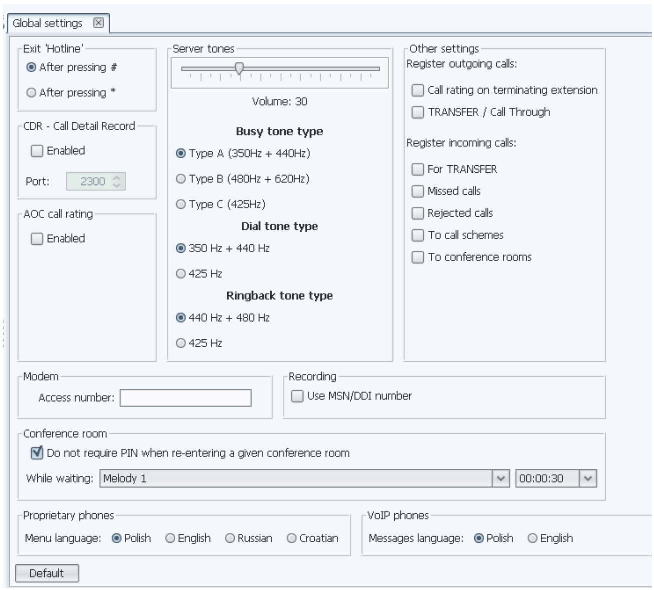


8.3 Global settings

⇒ **Exit 'Hotline'** – defines the key that allows the extension user to exit the Hotline, return to the server functions and e.g. dial the extension number.

⇒ **CDR – Call Detail Record –** you can enable sending call details after a completed call.

⇒ **AOC call rating** – enabling call rating for outgoing calls based on AOC (Advice of Charge) sent by the operator. With this option enabled the billing based on options set in *Billing – Penny* will be disabled.



⇒ **Server tones:**

– **Volume** adjustment for all server tones heard in the receiver. Adjustment range: **0** – mute to **100** – the loudest.

– **Busy tone type** – busy tone selection.

- **Dial tone type –** dial tone selection.
- **Ringback tone type** – ringback tone selection.

⇒ **Other settings:**

- **Register outgoing calls:**
  - **Call rating on terminating extension –** when an outgoing call is forwarded by extension A to extension B, the call will be charged to extension B.
  - **TRANSFER/Call Through** – every external outgoing return call will be registered. It will be available for viewing in the *Billing – Penny*.
- **Register incoming calls**:
  - **For TRANSFER** – every external incoming return call (for TRANSFER function) will be registered. It will be available for viewing in *Billing – Penny*.
  - **Missed calls** – every external incoming missed call will be registered. It will be available for viewing in *Billing – Penny*.
  - **Rejected calls** – every external incoming rejected call will be registered. It will be available for viewing in *Billing – Penny*.
  - **To call schemes** – every incoming call to call schemes will be registered. It will be available for viewing in *Billing – Penny*.
  - **To conference rooms** – every incoming call to conference rooms will be registered. It will be available for viewing in *Billing – Penny*.

⇒ **Modem:**

- **Access number** – extension number to access the PBX server through the analogue modem.

⇒ **Recording:**

- **Use MSN/DDI number** – setting that changes the extension number presentation to its MSN/DDI number for recorded calls in Agent 003 app.

⇒ **Conference room**:

- **Do not require PIN when re-entering a given conference room** – option that disables the PIN request for users re-entering a given conference room.
- **While waiting** – selection of a melody played for a user being the only participant of a conference and its duration (*00:00:30* by default)

⇒ **Proprietary phones:**

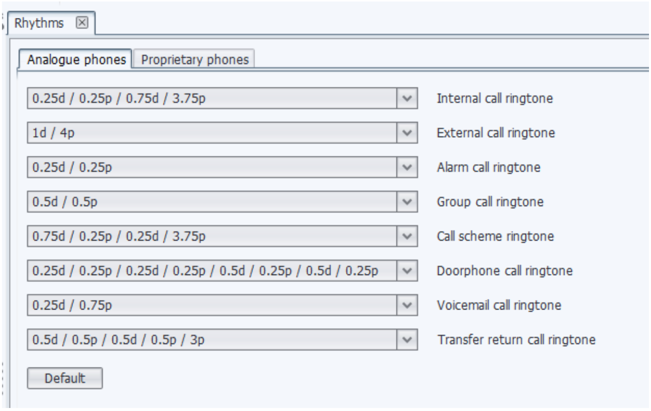
- **Menu language –** language selection for messages displayed on the proprietary phones (Polish, English, Russian and Croatian available).

⇒ **VoIP phones:**

- **Messages language –** language selection for messages send by the PBX server to the display of IP phones (Polish and English available).

8.4 Rhythms

8.4.1 Analogue phones



⇒ **Internal call ringtone** – ringing rhythm in analogue extension phones if another extension is calling.

⇒ **External call ringtone** – ringing rhythm in analogue extension phones if the external incoming call is made directly by a given trunk line.

⇒ **Alarm call ringtone** – ringing rhythm in analogue extension phones if a “wake-up call” has been booked.

⇒ **Group call ringtone** – ringing rhythm in analogue extension phones belonging to the same Hunt Group if the whole Hunt Group is called.

⇒ **Call scheme ringtone** – ringing rhythm in analogue extension phones if an external call comes through a call scheme.

⇒ **Doorphone call ringtone** – ringing rhythm in analogue extension phones called by doorphones.

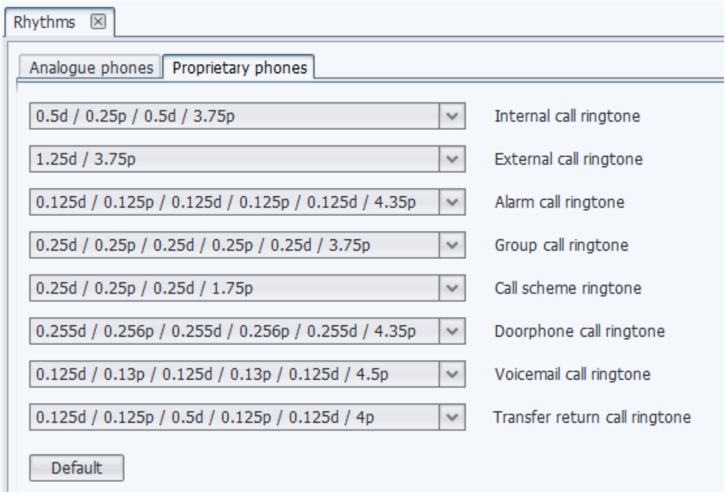
⇒ **Voicemail call ringtone** – ringing rhythm in analogue extension phones called by voicemail.

⇒ **Transfer return call ringtone** – ringing rhythm in analogue extension phones when a transferred call returns to a given extension after not being answered.

d – ringing **p – pause**

8.4.2 Proprietary phones

Like in the case of standard analogue phones, ringing rhythms in proprietary phones can be set in the program. The picture below illustrates default server settings.



⇒ **Internal call ringtone** – ringing rhythm in a proprietary phone if another extension is the calling party.

⇒ **External call ringtone** – ringing rhythm in a proprietary phone if the external incoming call is made directly by a given trunk line.

⇒ **Alarm call ringtone** – ringing rhythm in a proprietary phone if a “wake-up call” has been booked.

⇒ **Group call ringtone** – ringing rhythm of proprietary phones belonging to the same Hunt Group if the whole Hunt Group is called.

⇒ **Call scheme ringtone** – ringing rhythm in a proprietary phone if an external call comes through a call scheme.

⇒ **Doorphone call ringtone** – ringing rhythm in proprietary phones called by doorphones,

⇒ **Voicemail call ringtone** – ringing rhythm in proprietary phones called by voicemail.

⇒ **Transfer return call ringtone** – ringing rhythm in proprietary phones when a transferred call returns to a given extension after not being answered.

Where: **d – ringing** **p – pause**

8.5 Ringtone and server times

Ringtone times of phones connected to the server are defined in: *Global settings → Ringtone and server times*, where: **Ringtone times** and **Server times** tabs are located – to modify factory settings as needed.

8.5.1 Ringtone times

⇒ **Internal call ringtone time** – ringing time of extension phones,

⇒ **Ringtone adjustment time** – ringing time of extension phones when the installer is checking the ringtone system in the extension phone,

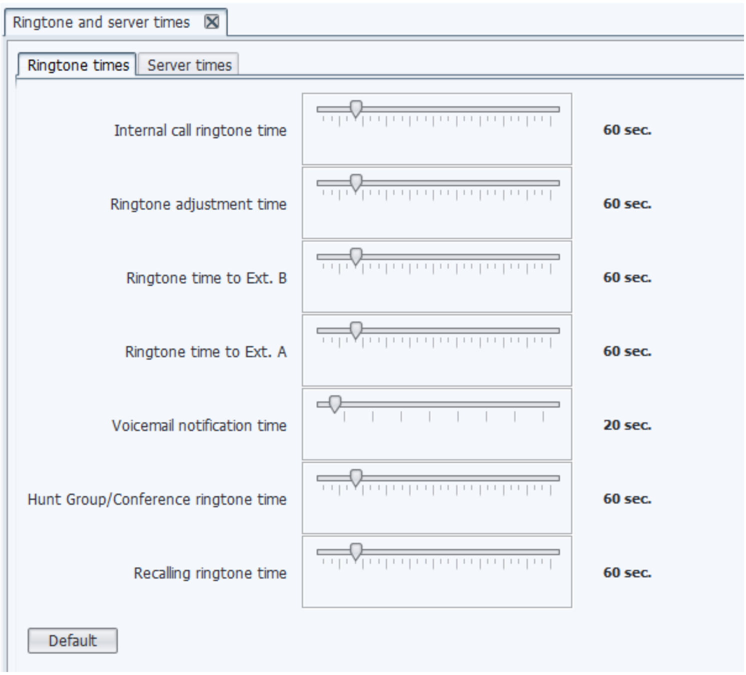
⇒ **Ringtone time to Ext. B** – ringtone time to extension B after the call has been transferred from extension A,

⇒ **Ringtone time to Ext. A** – ringtone time signalling the return of the call transferred to extension B,

⇒ **Voicemail notification time –** ringing time of extension phones when notifying about a message left in the voicemail inbox,

⇒ **Hunt Group / Conference ringtone time –** ringing time of phones belonging to a given Hunt Group if the Group is called and/or a conference takes place,

⇒ **Recalling ringtone time –** ringing time of extension phones when services such as external line booking, calls to busy parties, etc. are booked.



8.5.2 Server times

⇒ **Minimum FLASH time –** a minimum time after which the server will react to pressing the phone cradle (temporary disconnection) in the same way as to pressing FLASH;

⇒ **Maximum FLASH time** – a time with the phone cradle pressed after which the phone will end the call;

⇒ **Hotline delay time** – a period of time for user(s) with *Delayed Hotline* option defined (*Traffic classes → Hotline/DISA*) to dial an external number; after that time the server will establish a call with the number defined as *Delayed Hotline.*

⇒ **Do Not Disturb time** – a period of time in which the phone with the Do Not Disturb service activated will be unavailable (busy signal) for incoming calls;

⇒ **HOLD time** – a period of time for which the hold function is active; after that time, the server resumes the call that has been put on hold;

⇒ **Joining time in Uniform Call Distribution** – a period of time after which the server will add other extensions during a call initiated via a trunk line if the Uniform Call Distribution option is selected;

⇒ **Time for trunk line polarisation to stabilise** – waiting time until the external local exchange answers;

⇒ **Log out virtual extension after** – time after which the virtual extension logged on using the 784 “tel” code function will be automatically logged out. The time is always counted after the phone has been hung up. If the phone is picked up again, the time is counted anew.

⇒ **Time for BRA lines to activate** – the BRA ISDN lines are usually dormant and need to be activated in order to make a call. You should define the time period given to the server to activate the ISDN lines. A period that is too long will be troublesome for users, while if it is too short, you may hear an busy signal when trying to use the ISDN trunk line;

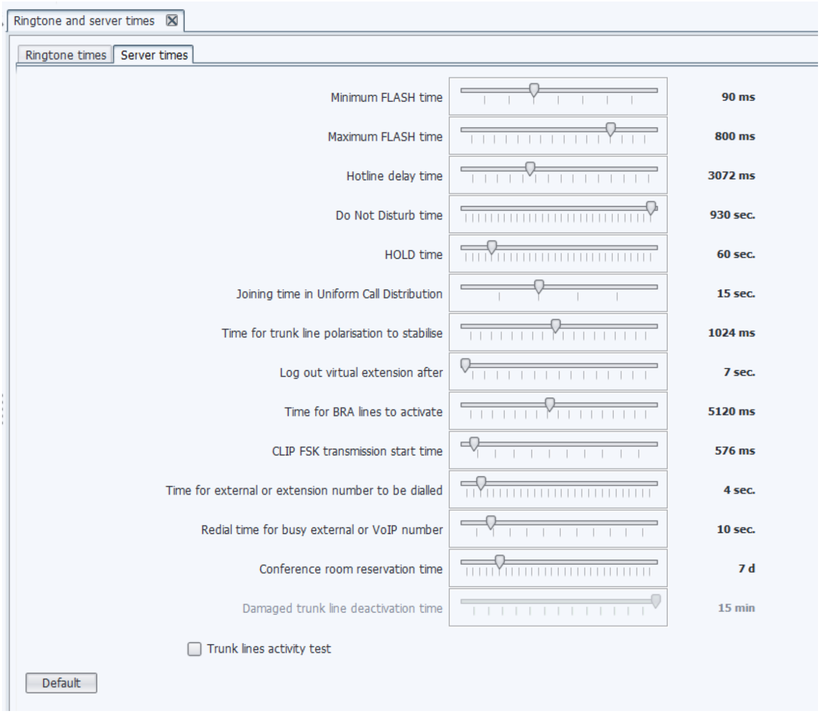
⇒ **CLIP FSK transmission start time** – time after which the server will start transmitting CLIP FSK to the phone;

⇒ **Time for external or extension number to be dialled** – waiting time for next digit to be dialled. It is an important parameter for handling dial numbers in the case of no escape code (accessing trunk line through the main trunk lines bunch set in the configuration without any number);

⇒ **Conference room reservation time** – time for which the conference room will be reserved (*7 d*(ays) by default)

⇒ **Damaged trunk line deactivation time** – the server checks if trunk lines are working. Should a damaged line be detected, the server will exclude it from the trunk lines bunch until it is fixed, which means that this line will not be used for calls. In the program, time intervals in which the server is to check the status of trunk lines are defined.

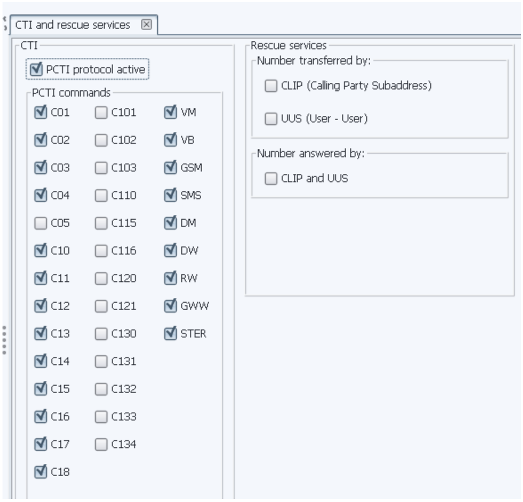
⇒ **Trunk lines activity test** – when ticked, the trunk lines test is activated, and the option *Damaged trunk line deactivation time* becomes available for setttings.



8.6 CTI and rescue services

PCTI commands – server interaction with the PCTI protocol. The PCTI commands box and all the commands must be ticked in order for the server to interact with the **PLATAN CTI** system. In addition, in order to assure the proper operation of CTI Clients installed on users' computers, **TCP** and **UDP 1000** ports must be opened on firewall. PCTI is an auxiliary protocol for CRM systems. Its description can be obtained from the Sales Department.

Rescue services – enabling functions in *Rescue services* box makes it possible to exchange the information about the transferred calls between the Platan PBX systems. The function is dedicated to handle the emergency calls by rescue services.



8.7 Calls

Settings concerning calls can be defined in the *Common settings* → *Calls* tab.

The following server services are defined in **Return calls** box:

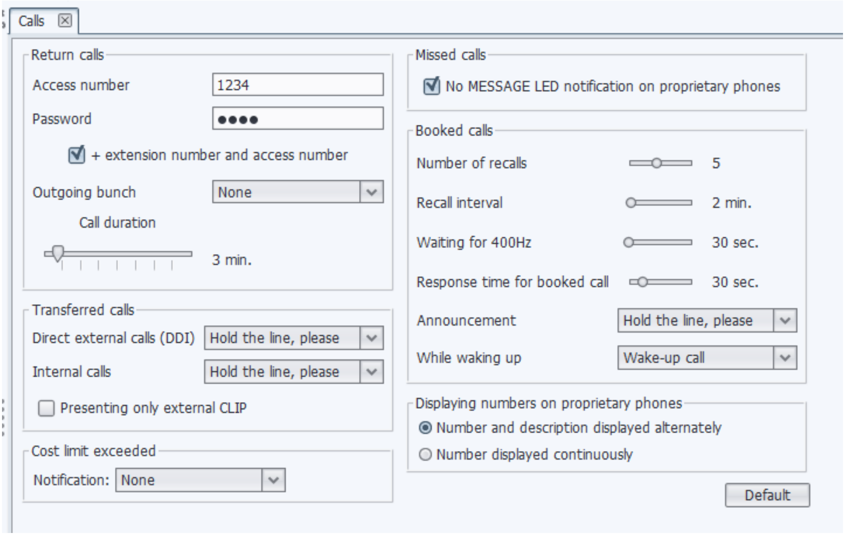
⇒ **Access number** – here you enter a service access number which the caller calling the server from the outside must dial during the DISA announcement. This number must be different from all other numbers in the server.

⇒ **Password** – a password with the maximum length of eight digits, which additionally protects the service from unauthorised people.

⇒ **+extension number and access code** – the safest method of using the external Return calls. This option activates the mode of individual external call settlement – you should enter your extension number and your own access password set in: *Extensions* → *Phone blocking* → *Blocking codes*. In addition, an extension user must have a *Service class* assigned with the option *Access number use* ticked.

⇒ **Outgoing bunch** – defines a trunk lines bunch through which external return calls will be made.

⇒ **Call duration** – time limit on return calls.



The following server services are defined in **Transferred calls** box:

⇒ **Direct external calls (DDI)** – select the announcement played during the transfer of calls made directly on extension's DDI number.

⇒ **Internal calls** – select the announcement played during the transfer of internal calls.

⇒ **Presenting only external CLIP** – defining the way of presentation of the transferred calls. With the option ticked, the transferred calls will be presented with the external caller's number only (external CLIP). Otherwise, the transferring user extension number will be presented at the beginning of a transfer and when they disconnect – it will be replaced by the external caller's number. Any option selected, the external caller's number will be presented in the call history.

Cost limit exceeded:
Selecting the announcement notifying about exceeding the cost limit (if it is set for a given user).

Missed calls:

– **No MESSAGE LED notification on proprietary phone**: option disabling the notification about the missed calls by the LED lamp on the proprietary phones.

Booked Calls – in these fields, you can adjust the *Call booking* function (functions *92, *94 and *95 described in *the Proxima IP PBX / Libra PBX Server Server User Manual*).

⇒ **Number of recalls** – defines the number of times the server should attempt to establish a call to an external number if it is busy or does not answer.

⇒ **Recall interval** – defines how many minutes should pass before another attempt at establishing an external call is made.

⇒ **Waiting for 400Hz** – a parameter useful if the trunk line does not function properly. Should it happen that after booking a call to a given number neither a busy tone nor a ring-back tone is received (e.g. a trunk line is disconnected), the server will end such a call after a time period set here in order to release an extension.

⇒ **Response time for booked call** – extension response time for a call with outside party, booked previously by this extension and already established.

Either an announcement or a melody can be selected in for booked calls:

⇒ **Announcement** – an announcement heard by a party with whom the call was booked before it is established,

⇒ **While waking up** – an announcement heard by an extension user called by the server with the “wake-up call” function.

In **Displaying numbers on proprietary phones** box, you can define whether the calling line number alternates with the description or is displayed on the proprietary phone continuously.

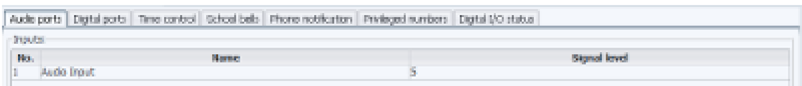
8.8 External devices control (I/O)

In *Common settings → External devices control (I/O)* you can define the operation settings of PROXIMA-IO/LIBRA-IO card, which enables the support of audio systems (Libra PBX Server only) and the signal reception from various sensors and external devices control with the use of adapter with relays.

8.8.1 Audio ports in Libra PBX Server

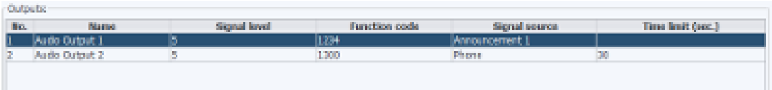
In *Audio ports* tab you can configure:

- **Audio Input** – input of external sound source by setting the *Signal level* value;



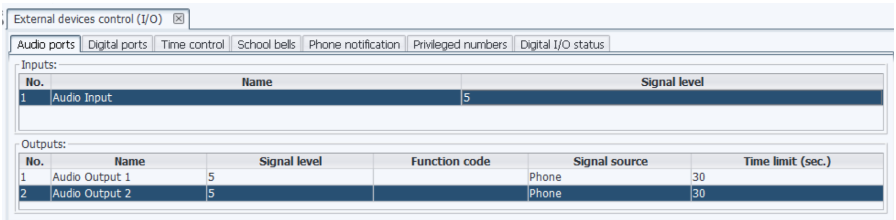
- **Audio Output** – e.g. as output for two independent radio broadcasting systems.
  In order to configure Audio Output you have to set:
  - **Signal level**,
  - **Function code** – number dialled by callers on the telephone in order to make an announcement through the radio broadcasting system,

- **Signal source** – selection of an announcement that will be broadcasted through the radio broadcasting system;
- **Time limit (sec.)** – parameter limiting the maximum duration of an announcement made by the telephone.



Outputs:

No.	Name	Signal level	Function code	Signal source	Time limit (sec.)
1	Audio Output 1	5	1234	Announcement 1	
2	Audio Output 2	5	1300	Phone	30



External devices control (I/O)

Audio ports | Digital ports | Time control | School bells | Phone notification | Privileged numbers | Digital I/O status

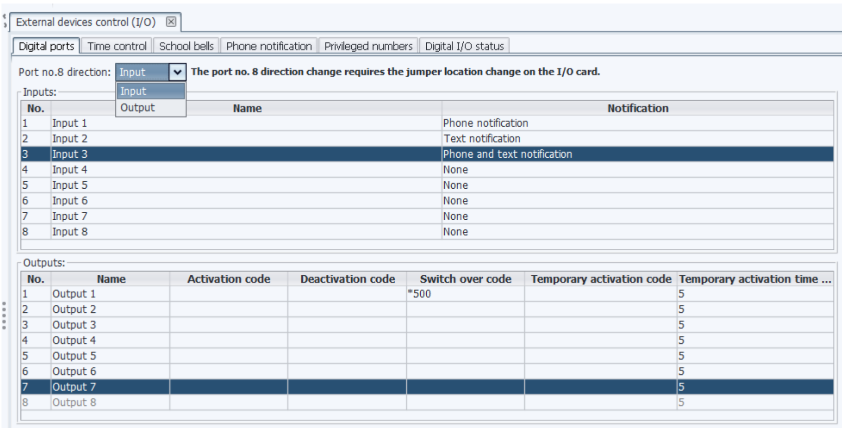
Inputs:

No.	Name	Signal level
1	Audio Input	5

Outputs:

No.	Name	Signal level	Function code	Signal source	Time limit (sec.)
1	Audio Output 1	5		Phone	30
2	Audio Output 2	5		Phone	30

8.8.2 Digital ports



External devices control (I/O)

Digital ports | Time control | School bells | Phone notification | Privileged numbers | Digital I/O status

Port no.8 direction: Input (Input / Output) The port no. 8 direction change requires the jumper location change on the I/O card.

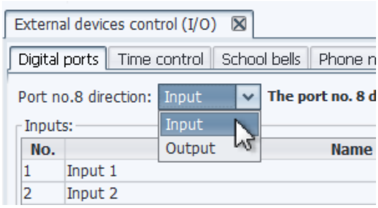
Inputs:

No.	Name	Notification
1	Input 1	Phone notification
2	Input 2	Text notification
3	Input 3	Phone and text notification
4	Input 4	None
5	Input 5	None
6	Input 6	None
7	Input 7	None
8	Input 8	None

Outputs:

No.	Name	Activation code	Deactivation code	Switch over code	Temporary activation code	Temporary activation time ...
1	Output 1			*500		5
2	Output 2					5
3	Output 3					5
4	Output 4					5
5	Output 5					5
6	Output 6					5
7	Output 7					5
8	Output 8					5

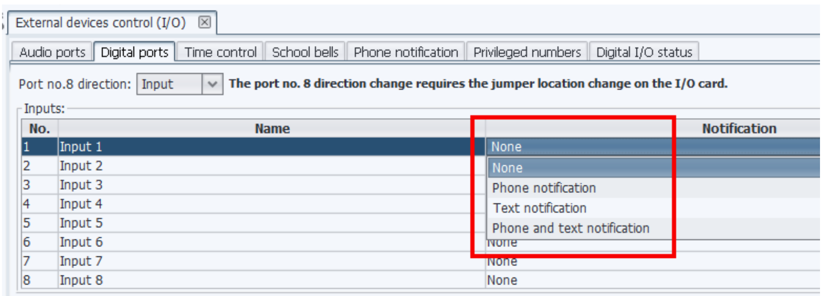
In the *Digital ports* tab sensors and relays' parameters can be set. You have at disposal: 7 inputs, 7 outputs and one port that can be defined either as an input or as an output.



Inputs - sensors

In the **Inputs** table the Proxima IP PBX Server/ Libra PBX Server's reaction on short circuit or voltage signal received by PROXIMA-IO/LIBRA-IO card is defined: The following notifications are available:

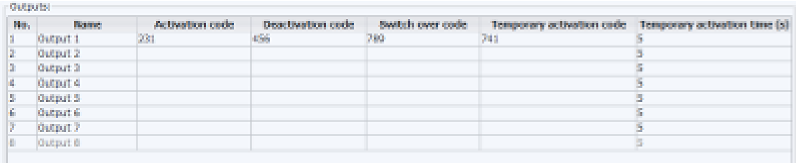
- **Phone notification –** on numbers which are set in *Phone notification* tab in *Common settings → External devices control (I/O)*
- **Text notification** – information about the change of sensor's status is sent on mobile phone numbers defined in *GSM → Settings*
- **Phone and text notification –** combining the two first options.



Outputs

In the **Outputs** table the following codes controlling the relays in the adapter are defined:

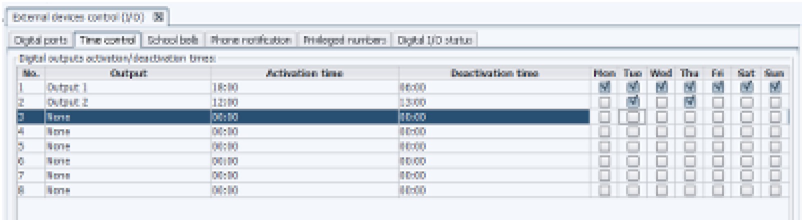
- Activation code,
- Deactivation code,
- Switch over code,
- Temporary activation code,
- Temporary activation time (s)



Codes can consist of 1-16 digits and cannot coincide with server's numbering plan.

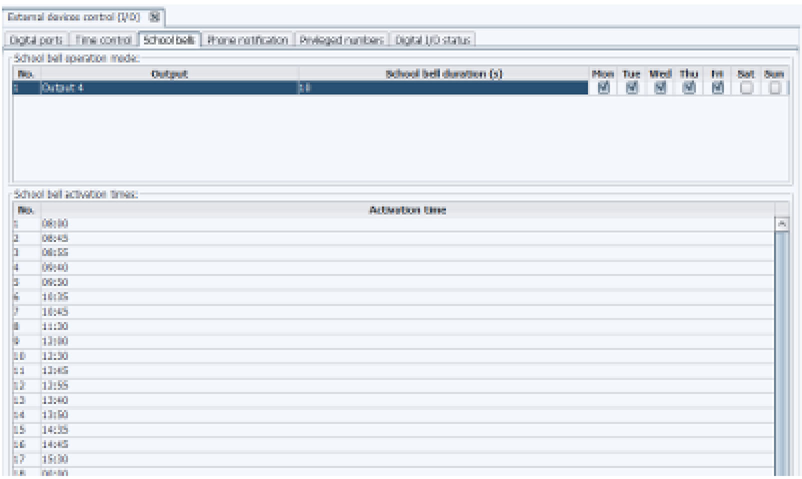
8.8.3 Time control

In *Time control* tab the Activation and Deactivation time of outputs (relays) can be defined for specific days of the week.



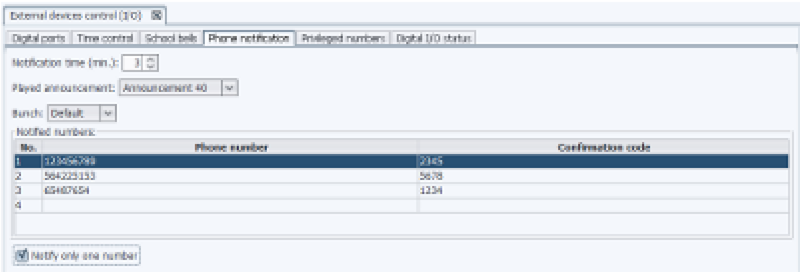
8.8.4 School bells

A school bell can be connected to one of the PROXIMA-IO/LIBRA-IO card's outputs; hours and ringing duration can be defined in *School bells* tab.



8.8.5 Phone notification

In *Phone notification* tab, phone numbers, confirmation codes and played announcement can be defined for notification receivers. Notifications are sent through the selected trunk lines bunch in the case of the short circuit or voltage signal on PROXIMA-IO/LIBRA-IO card's input if a phone notification option has been selected (and jumpers on PROXIMA-IO/LIBRA-IO card have been configured).

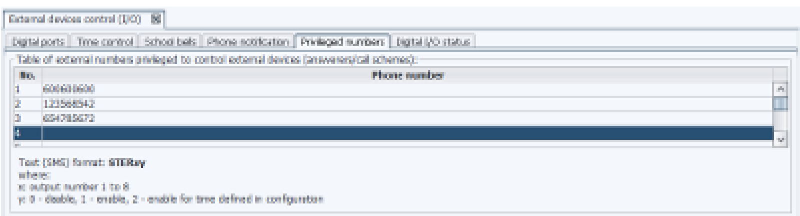


If the option *Notify only one number* is ticked, the server, after having received the confirmation code from one of the notified numbers, will not continue notifying next defined numbers.

8.8.6 Privileged numbers

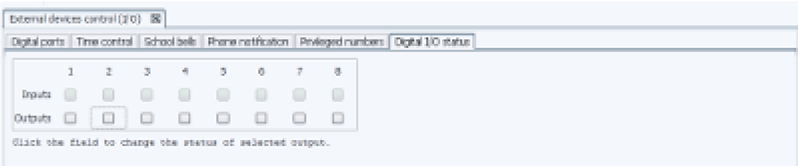
In *Privileged numbers* tab you can define external numbers (e.g. mobile phone numbers), which will be privileged to call on PROXIMA-IO/LIBRA-IO outputs and change remotely status of these outputs.

The card control can be executed by text messages (SMS) using the command STERxy, where x – output number (1-8), y – 0 (disable), 1 (enable), 2 (enable temporarily for the time defined in the *Digital ports* tab).



8.8.7 Digital I/O status

In the *Digital I/O status* tab the status of inputs and outputs can be observed, and the output status can be changed.



8.9 Function codes

Server function codes can be assigned with great flexibility. Function code may consist of **0 to 16 characters**. Accepted characters are: 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, #, *. Characters can be used in any order.

If no dial number is assigned to a function, this function will be not accessible for the extension users. The only exception is made for the **escape code** (accessing trunk line through the main trunk lines bunch set in the configuration) which can be left without any number.

In *Function format* column the server functions are presented as codes used by the extension users. They can be printed by clicking on *Print* button. The current settings are also available for users in the *User Zone*.

The initial default function codes can be restored by clicking on the *Restore default settings* button.

Functions dial numbers

Restore default settings | Print

Functions dial numbers:

Function name	Dial number	Function format
Accessing trunk line through the main trunk lines bunch set in the configuration (escape code)	0	0
Accessing trunk line through the first trunk lines bunch set in the configuration	*81	*81
Accessing trunk line through the second trunk lines bunch set in the configuration	*82	*82
Accessing trunk line through the third trunk lines bunch set in the configuration	*83	*83
Accessing trunk line through the fourth trunk lines bunch set in the configuration	*84	*84
Defining individual speed dial number	*89	*89 x "TEL"
Call Forward Unconditional CFU - (I'M THERE)	*734	*734 "TEL"
Cancelling Call Forward Unconditional (CFU)	*734	*734
Call Forward Unconditional CFU - (I'M HERE)	*735	*735 "TEL"
Call Forward Busy - CFB	*736	*736 "TEL"
Cancelling Call Forward Busy - CFB	*736	*736
Call Forward No Answer (CFNA) - if no answer within 20 sec.	*737	*737 "TEL"
Call Forward No Answer (CFNA) - if no answer within given time	*737	*737 "TEL" # "tt"
Cancelling Call Forward No Answer	*737	*737
Call Forward Not Reachable - CFNR	*738	*738 "TEL"
Call Forward Not Reachable (CNFR) - if not reachable within given time	*738	*738 "TEL" # "tt"
Cancelling Call Forward Not Reachable (CFNR)	*738	*738
Redialling the last dialled external number (REDIAL)	*90	*90
Dialling individual speed dial number	*91	*91 x
Booking a call to the last dialled external number	*92	*92
Booking a call to any external number	*94	*94 xxxxx
Cancelling services connected with booking calls to external numbers	*94	*94

If the **escape code** (accessing trunk line through the main trunk lines bunch) has **no number** assigned, the dialling process proceeds as follows:

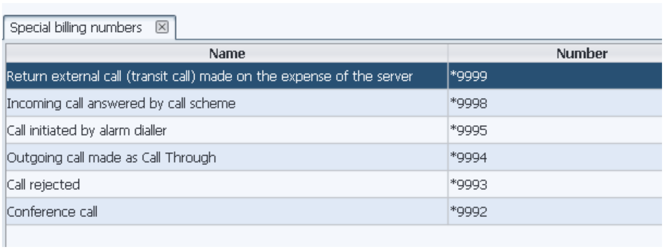
After the first dialled digit server waits for the successive digit for a preset time, checking in the meantime whether the dialled number is among the defined functions dialled numbers. After the next dialled digit the process repeats. If a server recognises the dialled sequence as a dial number it executes the adequate function or connects to the right extension number. If the dialled number is not recognised by the server, it is treated as an external number and the connection is being established with it.

Waiting time for the next digit is set in *Common settings → Ringtone and server times → Server times → Time for external or extension number to be dialled*. The default waiting time is 4 sec. For many Proxima server users this time may seem too long, thus it should be set experimentally.

If the escape code is empty it should be remembered that when defining **speed dialling** numbers (*Call distribution → Speed dialling*), the numbers in the *Real number* column should be entered without the access code to the main trunk lines bunch.
The same applies when setting the **call forwarding** (*Extensions → Call forwarding*) – the numbers in the *Unconditional*, *When busy* or *When no answer* columns should be entered also without the escape code.

8.10 Special billing numbers

For the call rating (billing) purposes the special virtual extension numbers are used so that certain types of calls can be registered (default settings):
*9999 – return external call made after previous logging into the server, on the expense of the server (transit call),
*9998 – incoming call answered by call scheme,
*9995 – call initiated by alarm dialer,
*9994 – outgoing call made as Call Through,
*9993 – call rejected,
*9992 – conference call.
Numbers can be changed in line with the numbering plan rules.

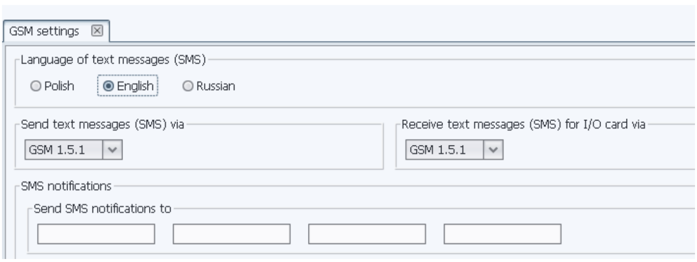

Special billing numbers

Name	Number
Return external call (transit call) made on the expense of the server	*9999
Incoming call answered by call scheme	*9998
Call initiated by alarm dialler	*9995
Outgoing call made as Call Through	*9994
Call rejected	*9993
Conference call	*9992

8.11 GSM settings

The GSM card settings are available in *Common settings → GSM settings*. In the configuration tab, you can:

⇒ select the language of text messages (SMS) – English, Russian or Polish,
⇒ identify the external GSM line via which the SMS text messages are to be sent,
⇒ identify the GSM line via which the SMS text messages for optional PROXIMA-IO/LIBRA-IO card (control of external devices) will be received,
⇒ define the phone numbers to which the notifications are to be sent in the defined situations. The program allows you to identify up to 4 numbers in an international format,



The messages are generated by the server and sent to the GSM network immediately after the occurrence of a given situation.

8.11.1 Types of SMS notifications

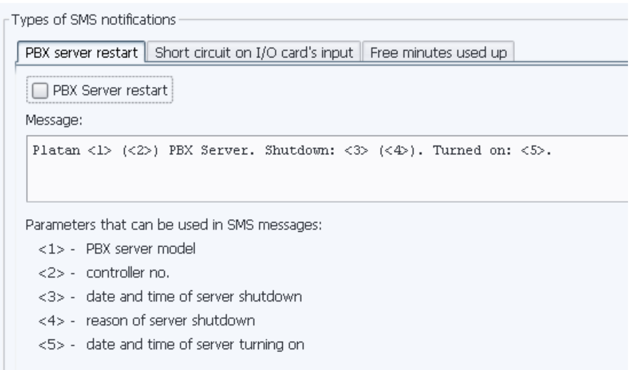
Service SMS

Define which important events in the server should result in a notification in the form of a short text message (SMS) being sent. The program allows you to activate SMS notifications in the following cases:

- PBX server restart,
- short circuit on I/O card’s input (notification about alarm),
- free minutes used up.

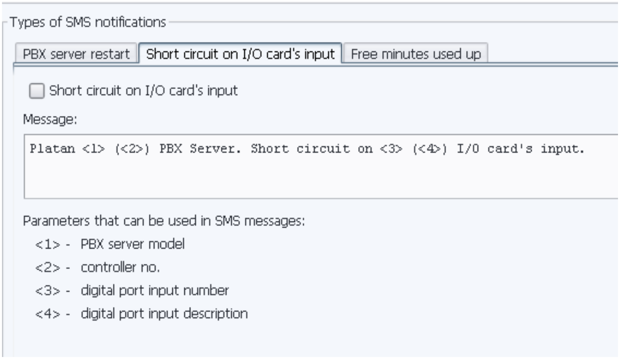
The SMS text message sent by the server in the cases defined above contains detailed information about the date and time at which a given event occurred and a controller number of the server involved. The default message can be modified according to the user’s preferences.

- <u>PBX server restart</u>



The default message is as follows: “Platan <PBX server model> (<controller no.>) PBX Server. Shutdown: <date and time od server shutdown>. Turned on: <date and time of server turning on.”
Both text and variables used in the SMS can be modified.

- <u>short circuit on I/O card's input</u>



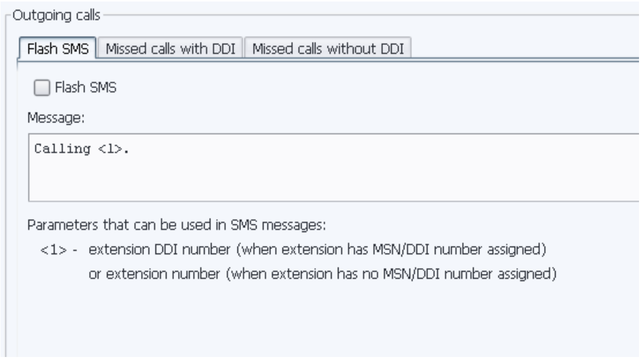
The default message is as follows: "Platan <PBX server model> (<controller no.>) PBX Server. Short circuit on: <digital port input number> (<digital port input description>) I/O card's input."
Both text and variables used in the SMS can be modified.

Outgoing calls
Three types of text messages (SMS) can be sent for outgoing calls:

- Flash SMS
- Missed calls with DDI
- Missed calls without DDI.

These are the text messages that are sent to users from outside the PBX server for the calls made by PBX server users.

- Flash SMS

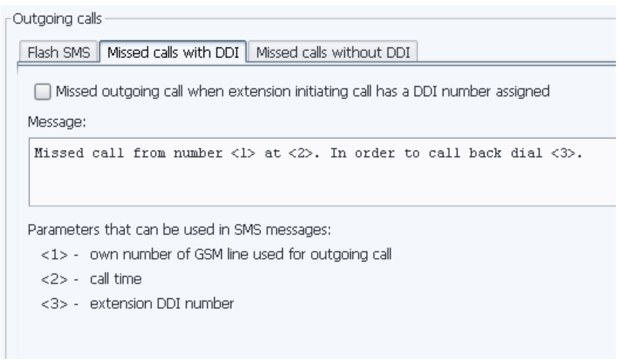


Switch on/off the option of sending Flash SMS text messages to the mobile phones to which calls via the GSM card are made. Such a message results in a called party receiving information about the number of the calling party.

The flash SMS is displayed directly (simultaneously with an incoming call) on the called party's mobile phone with information about the caller's PBX server extension number (otherwise the main trunk line number is displayed).

The default text message is: "Calling <extension DDI number (when extension has MSN/DDI number assigned) or extension number (when extension has no MSN/DDI number assigned)>."
Both text and variables used in the SMS can be modified.

- Missed calls with DDI

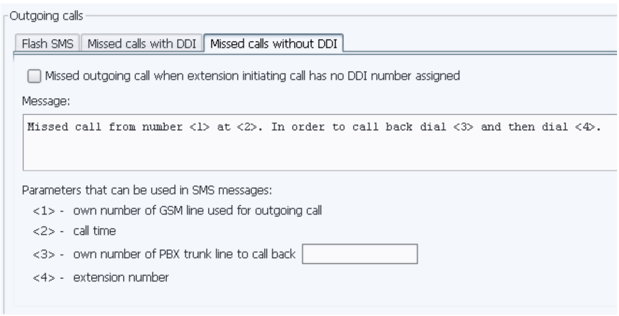


Switch on/off the option of sending SMS text messages for missed outgoing calls made via the GSM card when extension initiating call has a DDI number assigned.

This text message is sent to the called party's mobile phone when the recipient fails to answer a call incoming from the PBX server's extension having a DDI number assigned.

The default text message is as follows: "Missed call from number <own number of GSM line used for outgoing call> at <call time>. In order to call back dial <extension DDI number>."
Both text and variables used in the SMS can be modified.

- <u>Missed calls without DDI</u>



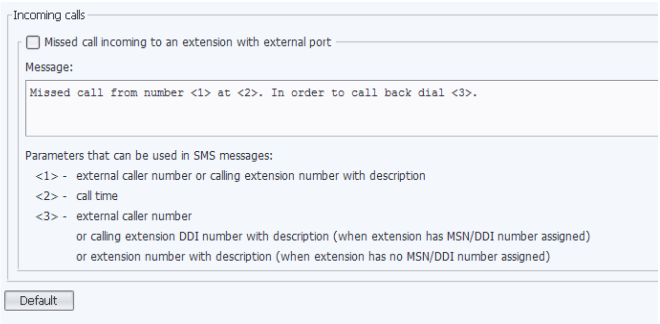
Switch on/off the option of sending SMS text messages for missed outgoing calls made via the GSM card when extension initiating call has not a DDI number assigned.

This text message is sent to the called party's mobile phone when the recipient fails to answer a call incoming from the PBX server's extension without a DDI number assigned.

The default text message is as follows: "Missed call from number <own number of GSM line used for outgoing call> at <call time>. In order to call back, dial <own number of PBX trunk line to call back> and then dial <extension number>."
Both text and variables used in the SMS can be modified.

Incoming calls

Switch on/off the option of sending text messages to the mobile phones being assigned to the external ports of a PBX server.



This type of text message is sent to the mobile phones used as PBX server's external ports when the user of such port fails to answer a call incoming from an external number via the PBX system. Without this SMS the recipient of such call will only have the information about an attempt of a call made by the PBX server's GSM line.

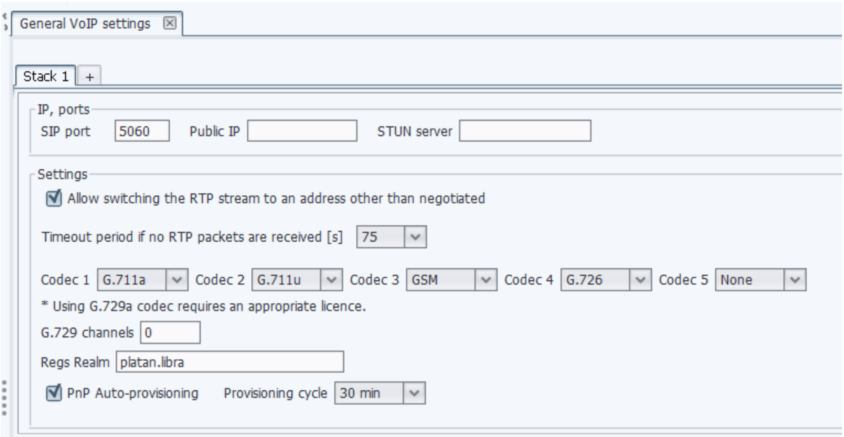
The default text message is as follows: "Missed call from number <external caller number or calling extension number with description > at <call time>. In order to call back, dial <external caller number / calling extension DDI number with description (when extension has MSN/DDI number assigned) / extension number with description (when extension has no MSN/DDI number assigned)>."
Both text and variables used in the SMS can be modified.

The *Default* key at the bottom of the page serves to restore the default messages used for all types of SMS.

8.13 VoIP settings

8.13.1 VoIP transmission settings

The *VoIP settings* tab is used to configure VoIP transmission parameters.



SIP port – VoIP server registration port (*port 5060 by default*). If Proxima IP PBX Server/ Libra PBX Server is used as the registration server for clients (VoIP extensions) from outside the local network, this port (UDP) needs to be forwarded in the router.

RTP port – defines the start of the range of ports used for voice transmission. The number of ports in use equals 2 x the maximum number of VoIP connections, which means another 128 ports (i.e. UDP 4000...4127 ports by default). If Proxima/ Libra is used as the registration server for clients from outside the local network, this port range (UDP) needs to be forwarded in the router.

Public IP – (optional) can be required when logging to the server of VoIP extensions from outside the local network.

STUN server – (optional) a network protocol allowing a client behind a NAT (or multiple NATs) to find out its public IP addresses, the type of NAT they are behind and the Internet port associated by the NAT with a particular local port.

***In the case of logging IP phones from outside the local network into the Proxima IP PBX Server/ Libra PBX Server, enter the Public IP address or STUN server in* VoIP → General settings menu.**

Allow switching the RTP stream to an address other than negotiated – this option allows changing the negotiated IP addresses during the call. If this option is off and the address is changed, the server will end the call. It has to be ticked if Proxima/ Libra functions as a registration server for clients from outside the local network (who do not have the public address or the STUN server configured).

Timeout period if no RTP packages are received [s] – a time period after which the server will end the established call if no RTP packages are received.

Codec 1...5 – a drop-down list from which the preferred VoIP transmission codecs should be selected. Available codecs:

- G.711a-law
- G.711µ-law
- GSM
- G.726
- G.729a*

* Using G.729a codec requires an appropriate licence.

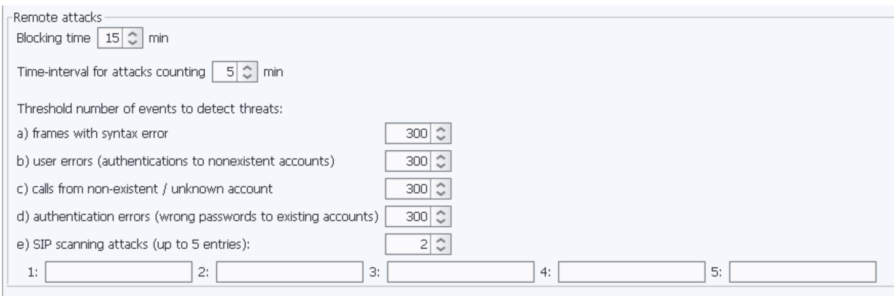
Regs Realm – an authorisation domain in the server, increasing the authorisation security and making it more likely that IP phones will be connected to the right VoIP server (server). Default *Regs Realm* domain name: platan.proxima.

PnP Auto-provision – option that enables the provisioning for Platan and Yealink IP phones and Platan Video Softphone.

Provisioning cycle – time interval for the changes to be sent to the IP phones.

8.13.2 Remote attacks

Option protecting Proxima IP PBX Server/ Libra PBX Server from attacks on VoIP which consists in blocking IP addresses from which the attacks originate within a defined period of time.



Blocking time – time for which an IP address that originates inquiries will be blocked.

Time-interval for attacks counting – time for attacks counting (events of a given type), after which the IP address will be black-listed.

Threshold number of events to detect threats – setting a limit for a given type of events. The following events are being monitored:

- frames with syntax error
- user errors (authentications to nonexistent accounts)
- calls from non-existent / unknown account
- authentication errors (wrong passwords to existing accounts)
- SIP scanning attacks (up to 5 entries).

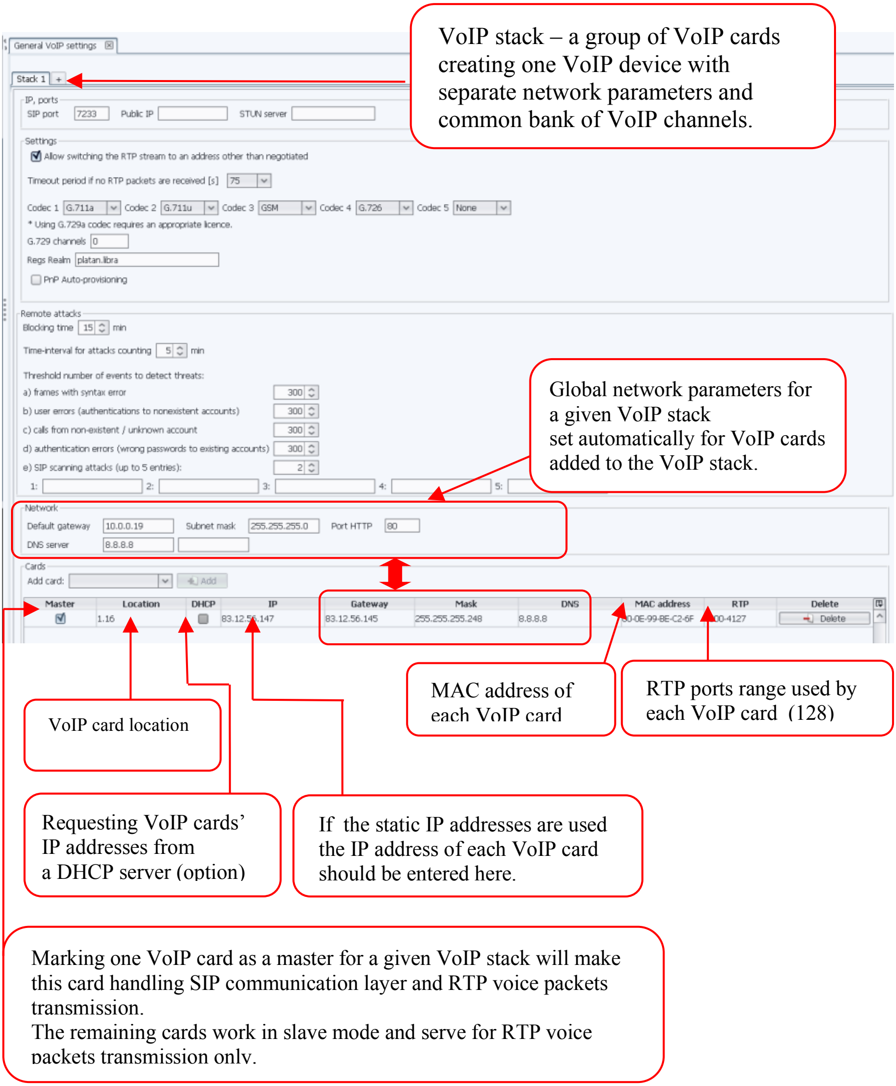
8.13.3 VoIP card settings

When using a PROXIMA-VOIP/ LIBRA-VOIP card, which makes it possible to increase the number of VoIP channels (simultaneous VoIP calls), set the method of obtaining the IP address: automatic (DHCP) or static.

If the option of a static IP address is selected, you have to enter:

- IP address of VoIP card,
- subnet mask,
- default gateway,
- DNS server – **enter the DNS server address received from your Internet provider**.

Additionally card's MAC address and port can be changed.



In the case of using the VoIP cards, we recommend removing the processor card from the VoIP stack.

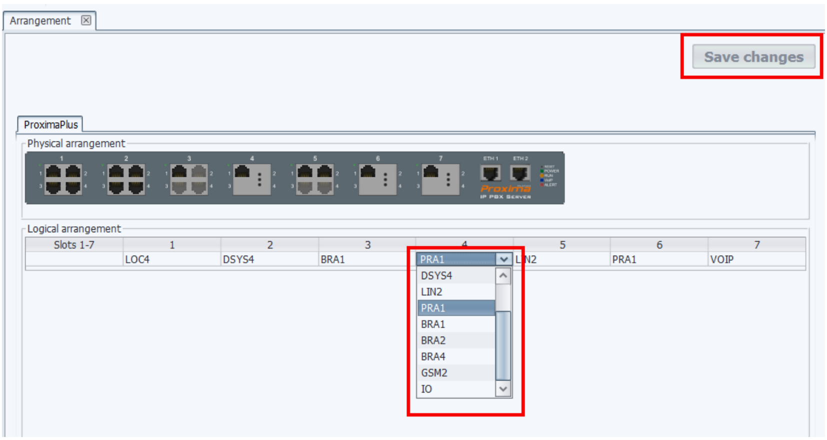
In Proxima IP PBX Server and Libra PBX Server with controllers no. 3, 4, 5 and 6 the support of VoIP is possible only with the use of VoIP card.

9. Administration

9.1 Proxima IP PBX Server arrangement

The *Arrangement* tab contains visualisation of the physical package arrangement in the Proxima / Proxima plus server. Apart from that, you can manually define the arrangement of packages in server units using the *Logical arrangement* field. Available cards:

- LOC4 – 4 analogue extensions card
- DSYS4 – 4 digital proprietary phones card
- DSYS2 – 2 digital proprietary phones card
- LIN2 – 2 analogue (CO) trunk lines card
- PRA1 – ISDN PRA (30B+D) interface card
- BRA1 – 1 ISDN BRA (2B+D) interface card
- BRA2 – 2 ISDN BRA (2B+D) interfaces card
- BRA4 – 4 ISDN BRA (2B+D) interfaces card
- GSM1– 1 GSM interface card
- GSM2 – 2 GSM interfaces card
- IO – card for digital inputs and outputs
- VOIP – VoIP card for up to 64 VoIP channels



After having assigned the cards to the proper slots, the settings should be saved by clicking the *Save changes* button. After saving, the ports will be visible in the *ProximaWeb/ProximaPlusWeb* program.

9.2 Libra PBX Server arrangement

The *Arrangement* tab contains visualisation of the physical package arrangement in the Libra PBX Server and the number of Libra units. There can be up to 6 units in one Libra PBX Server. Apart from that, you can manually define the arrangement of packages in server units using the *Logical arrangement* field. Available cards:

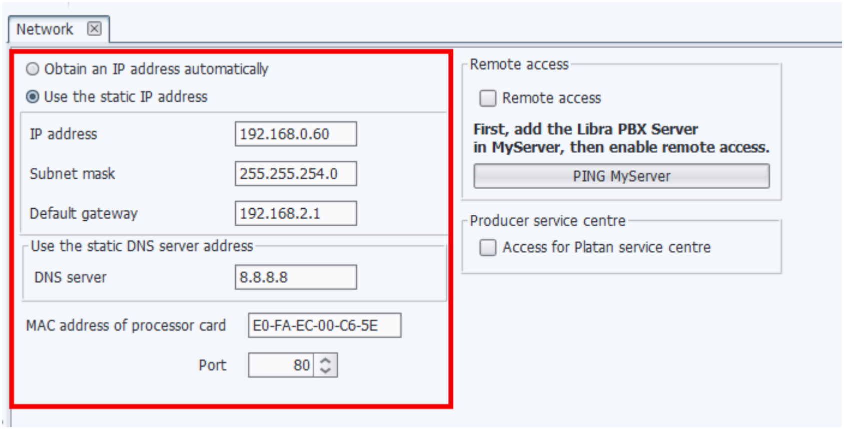
- LOC4 – 4 analogue extensions card
- LOC8 – 8 analogue extensions card
- DSYS4 – 4 digital proprietary phones card
- DSYS8 – 8 digital proprietary phones card
- LIN2 – 2 analogue (CO) trunk lines card
- LIN6 – 6 analogue (CO) trunk lines card
- PRA1 – ISDN PRA (30B+D) interface card
- BRA2 – 2 ISDN BRA (2B+D) interfaces card
- BRA4 – 4 ISDN BRA (2B+D) interfaces card
- BRA8 – 8 ISDN BRA (2B+D) interfaces card
- GSMH2– 2 GSM interface card
- GSMH4 – 4 GSM interfaces card
- IO – card for audio and digital inputs and outputs
- VOIP – VoIP card for up to 64 VoIP channels

After having assigned the cards to the proper slots, the settings should be saved by clicking the *Accept changes in arrangement* button. After saving, the ports will be visible in the *LibraWeb* program.

9.3 Network

The Proxima/Libra server Ethernet (LAN) interface settings are available in *Administration → Network*. Default settings are as follows:

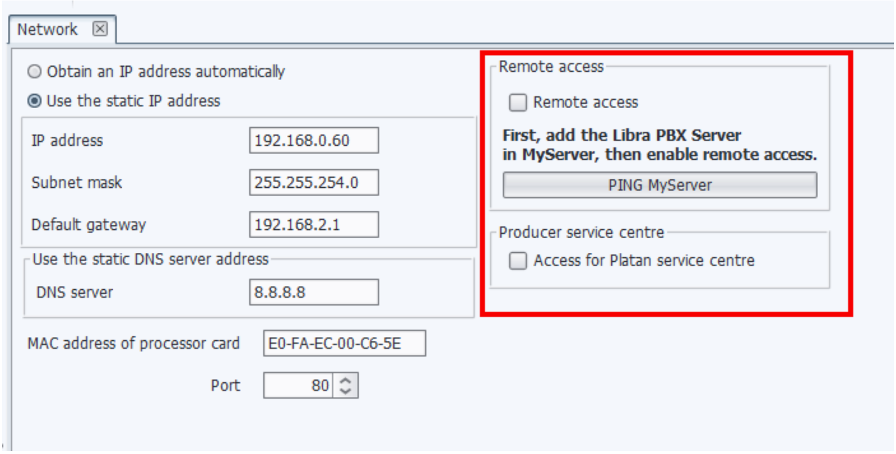
- IP address: **192.168.1.250**
- subnet mask: **255.255.255.0**
- default gateway: **192.168.1.1**
- port: **80**
- DNS server: 194.204.159.1 – **DNS server address received from the Internet provider should be entered**.



You will obtain necessary configuration data from the computer network administrator.

9.4 Remote access

Function enables remote management and configuration of Proxima IP PBX Server/ Libra PBX Server through *MyServer.platan.eu* platform.



MyServer.platan.eu platform is available for Platan Partners, Authorised Installers and other Platan installers.

For security reasons, function *Remote access* requires obtaining a written consent of Proxima/Libra server's owner.

9.5 Licences

In the *Licences* tab you can enter the licence key unblocking the additional PBX Server functions, such as VoIP ports and channels, video ports, channels for call recording and call queuing function, etc.

Licence keys are available as auxiliary option, in order to receive the licence key from the Sales Department, the server controller number is required (it can be found in *Administration → Server info*).

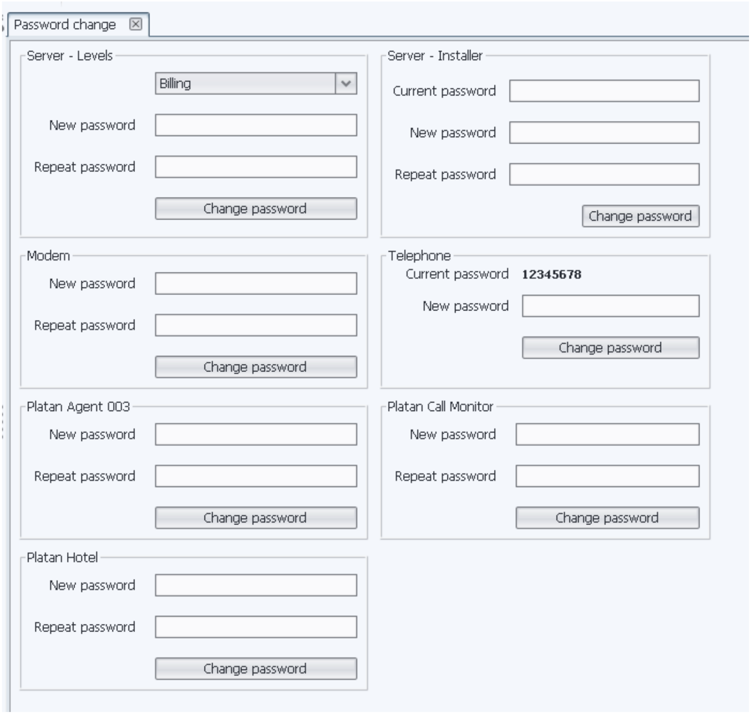
Note. Maximum quantity of VoIP licences and on call recording depends on the type of controller used in Proxima IP PBX Server/ Libra PBX Server.

For more than 10 VoIP channels we recommend to install the LIBRA-VOIP card in the 15th slot of any Libra server's unit.

Temporary licence codes are generated for 14 to 30 days for testing and service purposes. Once the temporary licence expires, the default licences assigned to the server are enabled.

9.6 Password change

In *Administration → Password change* tab, access password to Proxima IP PBX Server/ Libra PBX Server can be changed.



There are following access passwords in Proxima/Libra server:

Server – Levels

Passwords for PBX server users, giving them access to different PBX settings and data:

- *Billing* – password to access the billing data only.
- *Reading* – password to read the configuration from the server, without the possibility of making any changes.
- *Manual 1* and *Manual 2* – passwords to access and make changes in settings defined by the installer in *Administration → Manual level*.

Installer – password for installer giving access to all changes made in the server.

Modem – password to access the PBX server through ProximaPC/LibraPC offline program with the use of the embedded modem.

Telephone – password enabling programming several functions from the telephone connected to the server.

Installer password and passwords to different user levels have to consist of at least 8 characters, including 2 digits or 2 letters.

Agent 003 – password enabling access to Agent 003 application (call recording management).

Agent 003 password can be changed only when Proxima IP PBX Server/ Libra PBX Server is connected to the Agent 003 server.

Note. After Agent 003 password change the encryption algorithm for call recording in Proxima IP PBX Server/ Libra PBX Server will be changed as well. Before changing the password make sure that the past recordings have been downloaded to Agent 003 application, as they will be not longer available after the password change.

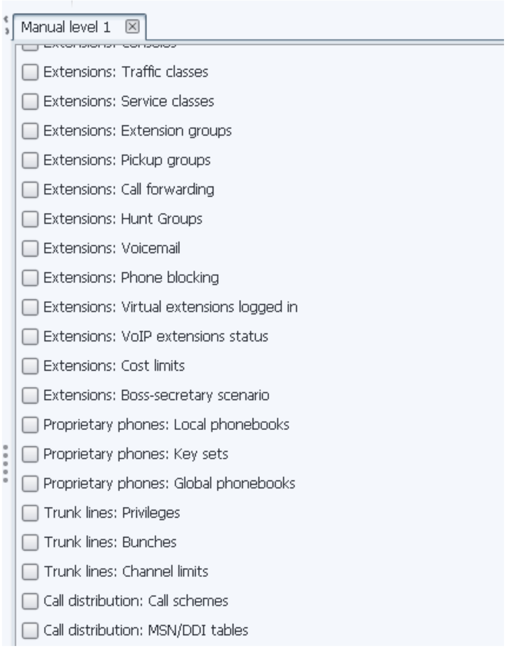
After having saved the new Agent 003 password, enter the same password in *Agent 003* application → *Configuration* tab in order to update the recordings decryption key and to re-connect.

Platan Hotel – password to access Platan Hotel application – software embedded in the Proxima/Libra PBX servers that supports hotel functions in the PBX.

Platan Call Monitor – password to access Platan Call Monitor application (to support and monitor the call reception by groups of users).

9.7 Manual level

In *Administration* → *Manual level 1* and *Manual Level 2* tabs you can tick the options available for edition after entering the password for *Levels: Manual 1* or *Manual 2*. It gives installer the possibility to grant users two levels of access to the selected PBX settings.



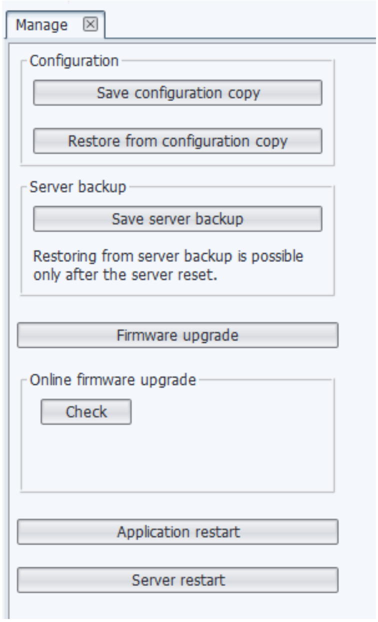
9.8 Manage

In *Administration → Manage*, you can save a backup copy, restore the previously saved configuration and upgrade the firmware (either online or from a file).

Save configuration copy – configuration backup. The following data are saved in the configuration copy:

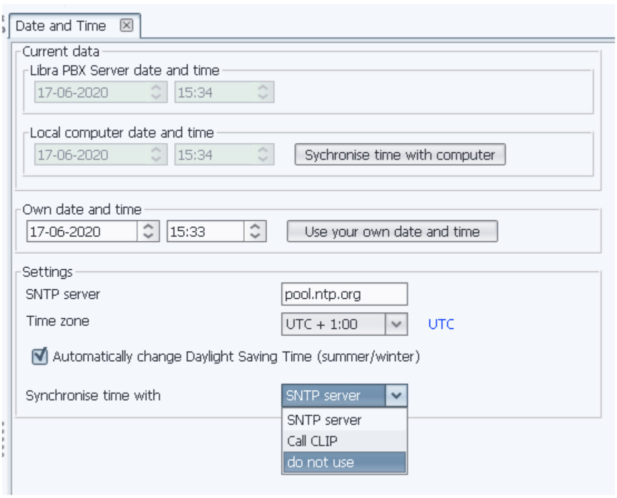
- server configuration
- voice announcements
- billing
- personal greetings and messages left in voicemail.

Application restart – ProximaWeb/LibraWeb application restart.
Server restart – Proxima IP PBX Server/ Libra PBX Server restart



9.9 Date and Time

Set the server time by synchronising it with the SNTP time server, the incoming call CLIP or the computer connected to the PBX server.

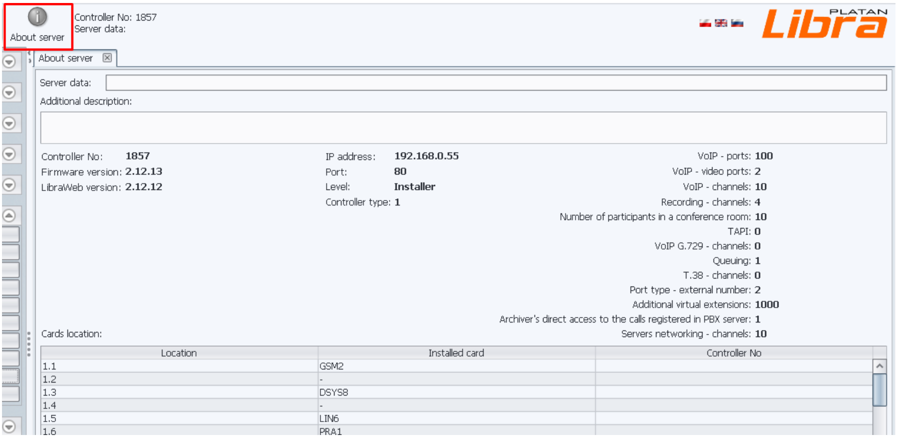


9.10 About server

All the basic information about the server are available in this tab, including:

- firmware and ProximaWeb/LibraWeb program versions,
- controller no and type,
- PBX Server's IP address and port
- level of access (installer, manual etc.)
- licences
- cards arrangement.

The same information can be accessed by the *About server* button on the main top panel.



10. Networking

Networking makes it possible to connect up to sixteen Proxima IP PBX Servers and/or Libra PBX Servers via the IP network into one uniform telecommunication system with coherent extension numbering.

In order to connect PBX servers in one network, each of them must be equipped with VoIP64 card and have licences for server networking channels. The number of required licences depend on the number of simultaneous calls made among the servers. The channel licences are required for all servers creating a network.

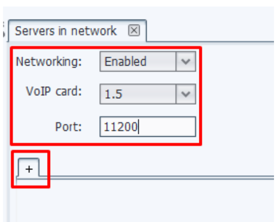
10.1 Servers

In *Networking* → *Servers* menu:

- enable networking function for a given PBX server,
- indicate the VoIP card that will take part in servers networking,
- enter the UDP network port that will be used in this server for servers networking.

The total numer of VoIP channels and channels used for servers networking cannot exceed 63 for one VoIP64 card indicated for servers networking.

When the PBX systems are linked via the Internet network, the UDP port indicated for servers networking should be forwarded in the router to the IP address of a PBX server's VoIP card located in the LAN network and used for servers networking.

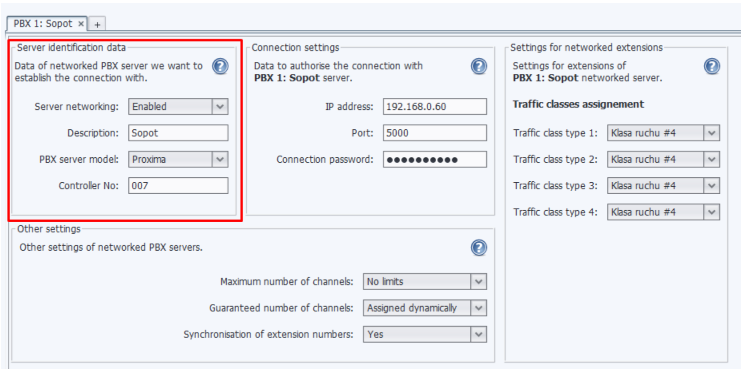


Then click the "+" symbol below to add the authorisation data of the PBX server that will be linked. A new window will appear.

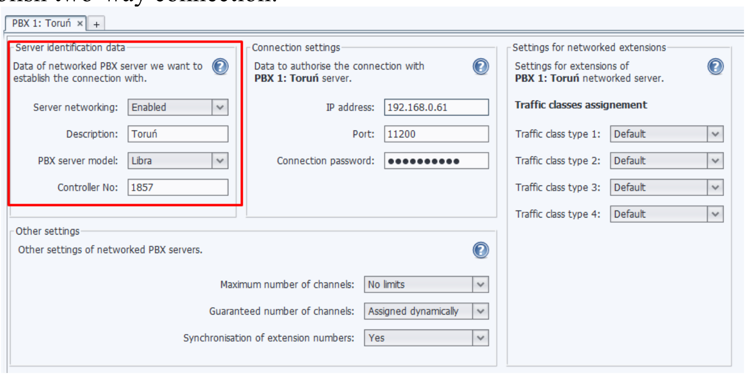
10.1.1 Server identification data

Enter data of a PBX server you want to link:

- Server networking – enable the networking with a given PBX server.
- Description – enter a description that will identify the PBX server. This description will appear automatically in additional comments and help.
- PBX server model – select Proxima or Libra.
- Controller No – controller no of the remote PBX server.



In the remote PBX server the data of the first PBX server should be entered as well to establish two-way connection.

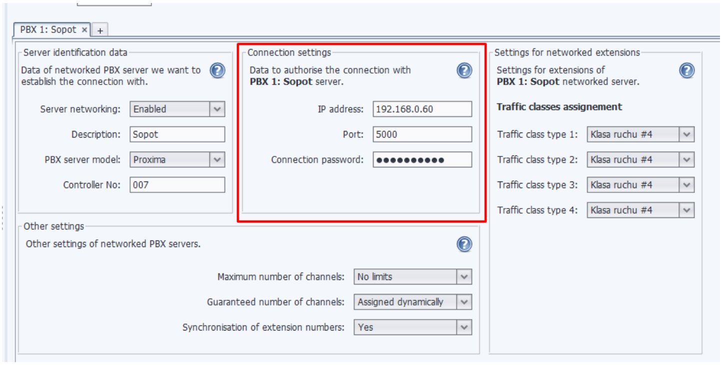


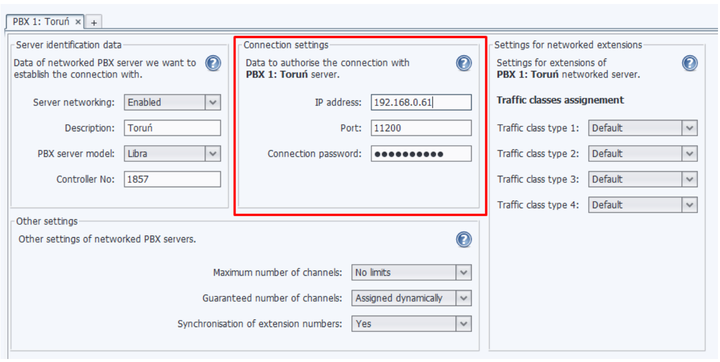
10.1.2 Connection settings

Data authorising the connection with a networked server. PBX server's IP address and port should be entered, indicating the location of the PBX system.

If a networked PBX server is behind NAT, the UDP port should be forwarded properly to this server.

The connection password is required and serves to establish an encrypted connection between the pair of PBX servers. Password must be identical in the settings of both servers in every pair of networked devices.

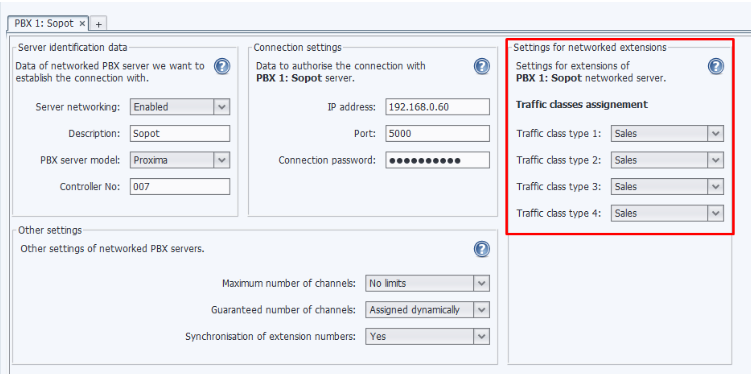




10.1.3 Settings for networked extensions

Settings for extensions of a networked PBX server. Extensions of a networked PBX server have a given type of traffic class, e.g. "Traffic class type 1" defined for networking in the *Traffic classes* menu.

A traffic class defined in the local PBX server should be assigned to these types. It determines the privileges of networked extensions to make calls within the local PBX server.

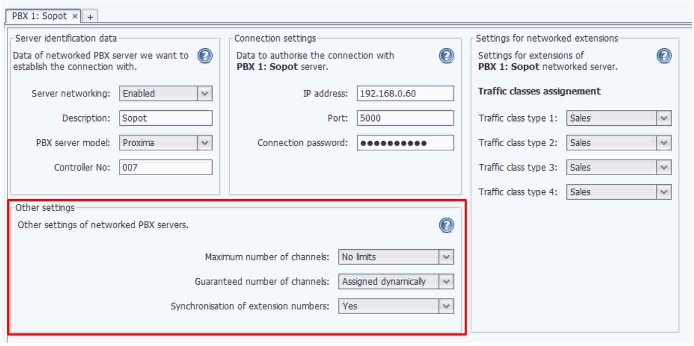


10.1.4 Other settings

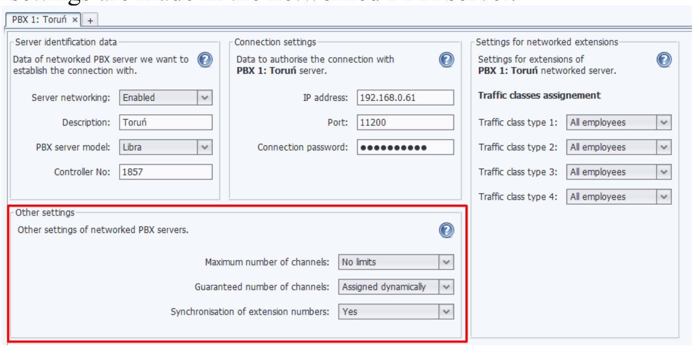
Maximum number of channels – the limit of simultaneous calls between a pair of PBX servers. The “No limits” option makes it possible to establish the maximum number of simultaneous calls equivalent to the number of licences for networking channels bought reduced by the number of channels used by other pairs of PBX servers. The value “0” blocks the possibility of connections between the pair of servers.

Guaranteed number of channels – guaranteed number of simultaneous calls between a pair of PBX servers. System allows making the higher number of calls than guaranteed only if there are licences for network channels available (not used by other network connections between the servers). The “Assigned dynamically” option enables the dynamic assignement of channels for networking among the pairs of servers within the number of licences bought.

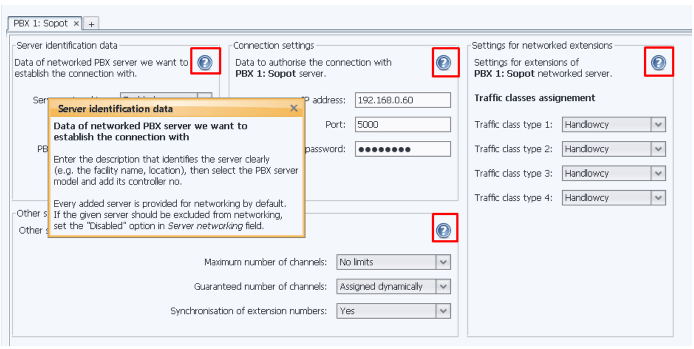
Synchronisation of extension numbers – access to the extension numbers defined in the networked PBX server. Lack of synchronisation makes it impossible to read the extension numbers of the networked server by the local server and does not allow to establish connections with them by the extensions of the local PBX server.



The same settings are made in the networked PBX server.

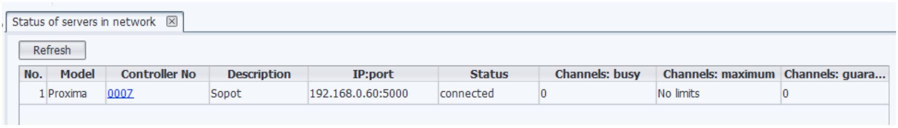


Note: Click the blue question mark to access the additional window with help.



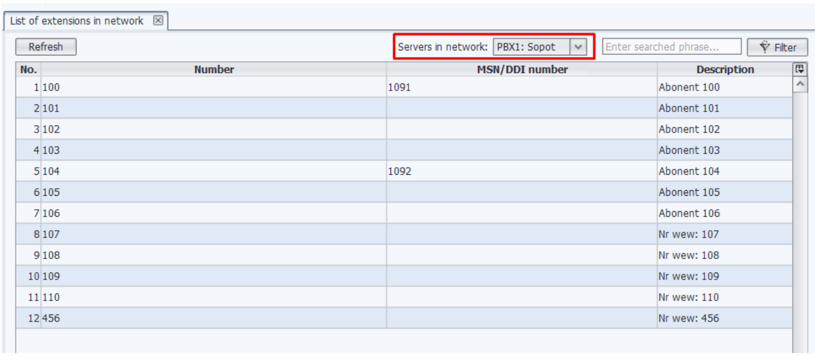
10.2 Statuses

In *Networking* → *Statuses* there's information about a current status of networked servers' connection and the number of channels currently used for linking the PBX servers.

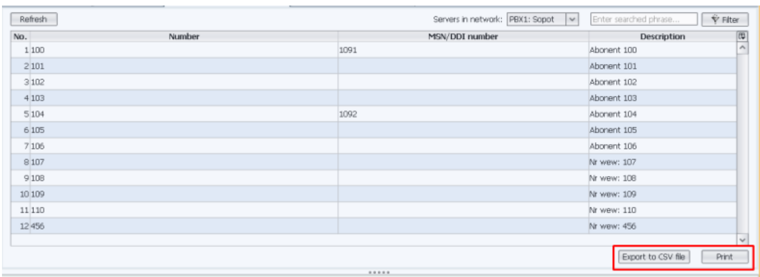


10.3 Numbers

In *Networking* → *Numbers* the extension numbers of the networked servers are listed. The required PBX server can be selected from the drop-down list.



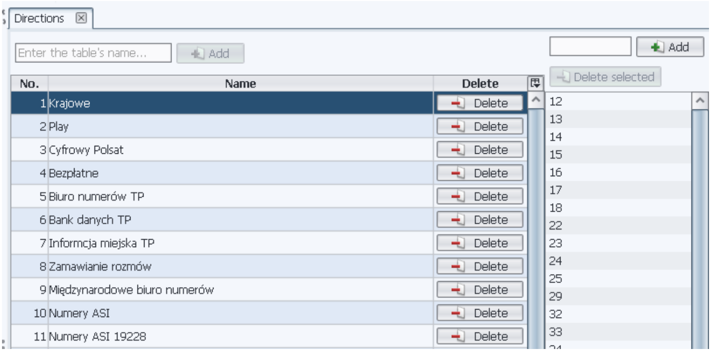
Extension numbers of the networked server can be printed or exported to a .csv file.



11. Call rating and billing – Penny

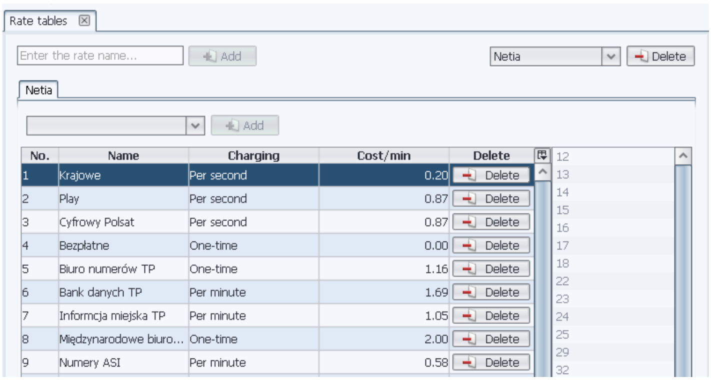
11.1 Directions

In *Directions*, you should define prefixes based on which call costs will be calculated.

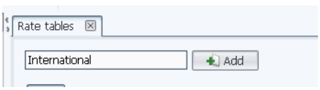


11.2 Rate tables

The *Rate tables* tab is for determining the rates that will be used to define the rate schedule.



In order to create a new rate, you should name it and accept it by clicking *Add*.



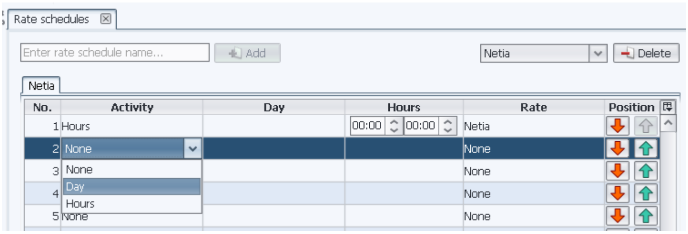
Next, *Directions* (prefix tables), e.g. GSM, International, etc. and call rates should be assigned to such a new rate.

11.3 Rate schedule

You can add/delete a rate schedule in the *Rate schedule* tab. In order to define a rate schedule, you should decide if it is to apply all day long – for this purpose you should set the rate schedule activity option to *Day* – or at specific hours – for this purpose you should set the rate schedule activity option to *Hours*.

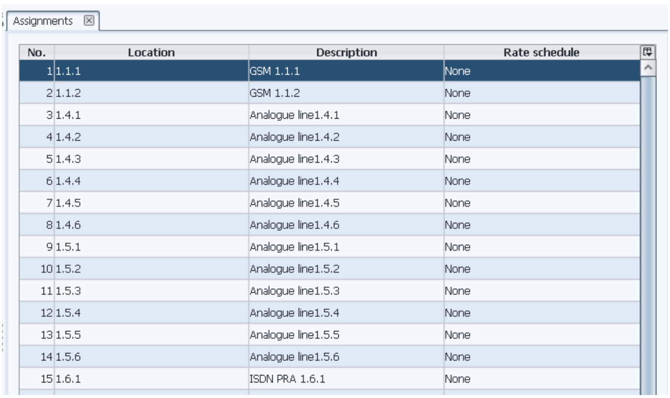
For *Day* activity, one of three modes can be selected in the *Day* column from a drop-down list: *Saturday*, *Sunday* or *Holiday*.
For *Hours* activity, the start and the end of the period in which the rate schedule applies should be defined in the *Hours* column.

In the *Rate* column, you assign one of the previously prepared rates from a drop-down list.



11.4 Assignments

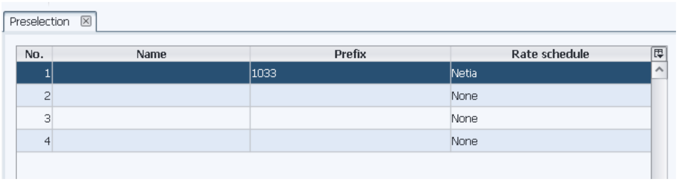
In *Assignments* tab you should set a predefined *Rate schedule* selected from a drop-down list.



No.	Location	Description	Rate schedule
1	1.1.1	GSM 1.1.1	None
2	1.1.2	GSM 1.1.2	None
3	1.4.1	Analogue line1.4.1	None
4	1.4.2	Analogue line1.4.2	None
5	1.4.3	Analogue line1.4.3	None
6	1.4.4	Analogue line1.4.4	None
7	1.4.5	Analogue line1.4.5	None
8	1.4.6	Analogue line1.4.6	None
9	1.5.1	Analogue line1.5.1	None
10	1.5.2	Analogue line1.5.2	None
11	1.5.3	Analogue line1.5.3	None
12	1.5.4	Analogue line1.5.4	None
13	1.5.5	Analogue line1.5.5	None
14	1.5.6	Analogue line1.5.6	None
15	1.6.1	ISDN PRA 1.6.1	None

11.5 Preselection

The *Preselection* tab allows you to set the call rating for calls made using the prefix of any operator, e.g. 1033, 1044, etc.

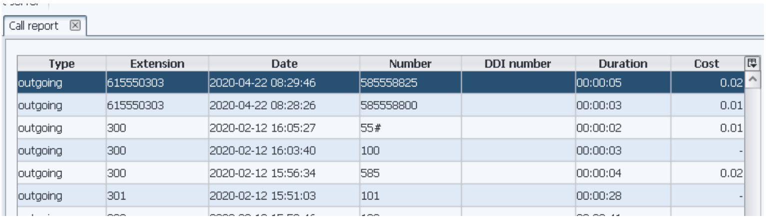


Preselection

No.	Name	Prefix	Rate schedule
1		1033	Netia
2			None
3			None
4			None

11.6 Call report

The *Call report* tab contains the list of calls made by extension users along with the call costs calculated after rate schedules have been defined and assigned to trunk lines.



Call report

Type	Extension	Date	Number	DDI number	Duration	Cost
outgoing	615550303	2020-04-22 08:29:46	585558825		00:00:05	0.02
outgoing	615550303	2020-04-22 08:28:26	585558800		00:00:03	0.01
outgoing	300	2020-02-12 16:05:27	55#		00:00:02	0.01
outgoing	300	2020-02-12 16:03:40	100		00:00:03	-
outgoing	300	2020-02-12 15:56:34	585		00:00:04	0.02
outgoing	301	2020-02-12 15:51:03	101		00:00:28	-

Calls registered on *Special billing numbers* stand for (default settings):

*9999 – return external call made after previous logging into the server, on the expense of the server (transit call),
*9998 – incoming call answered by call scheme,
*9995 – call initiated by alarm dialer,
*9994 – outgoing call made as Call Through,
*9992 – rejected call,
*9991 – conference call.

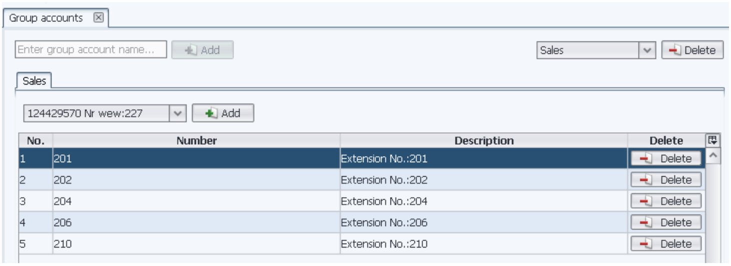
i Special billing numbers can be modified in *Common settings → Special billing numbers*.

11.7 Export / Import

Here you can export/import the rate schedule prepared beforehand.

11.8 Group accounts

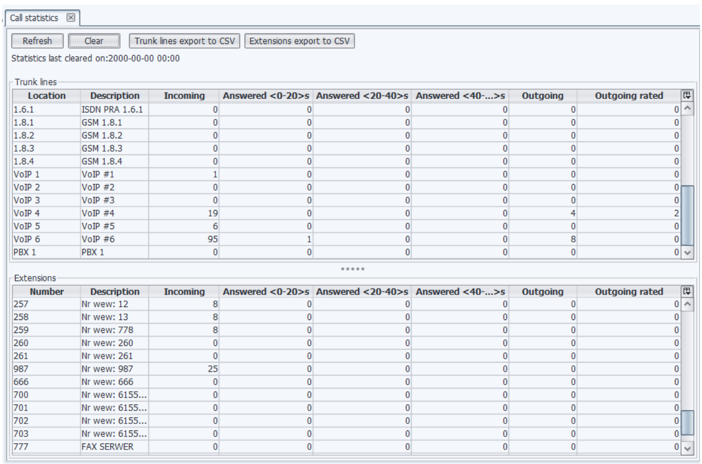
In the *Group accounts* tab you can create group accounts (e.g. for different departments), thus enabling the filtering of billing data in the *Call report* and graphic visualisation of the statistics for these groups.



11.9 Call statistics

Call statistics make it possible to estimate the scale and proportions of incoming and outgoing traffic intensity on particular trunk lines (upper part of the window) and evaluate calling activity and call answering by extension users (bottom part of the window).

In *Call statistics* tab there is a list of all incoming and outgoing phone calls made since the last restoration of the default settings or since the last statistics' clearing.



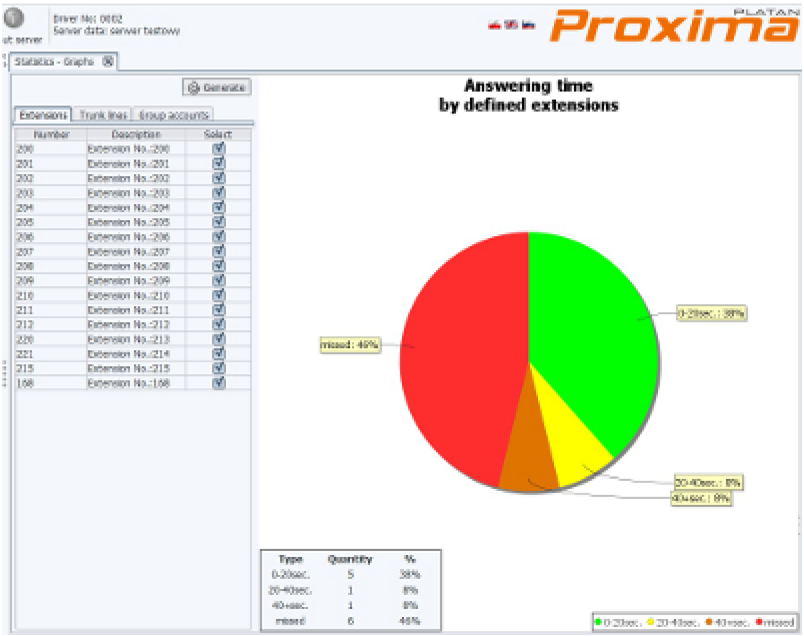
- **incoming** – number of all external connections incoming through respective trunk lines

- **answered** – number of incoming answered calls divided into groups based on the time after which the call was answered
- **outgoing** – number of outgoing calls made through respective trunk lines;
- **outgoing rated** – number of outgoing calls made through respective trunk lines, rated in rate schedule.

Before opening the *Call statistics* window connect to the Proxima IP PBX Server/ Libra PBX Server and read the configuration. Otherwise the displayed billing data may be out-of-date.

11.10 Statistics - Graphs

In *Statistics – Graphs*, the answering time by defined extension users, account groups and coming through the defined trunk lines are visualised. The answering time is given in the following groups: less than 20 s, 20-40 s, over 40 s and missed.



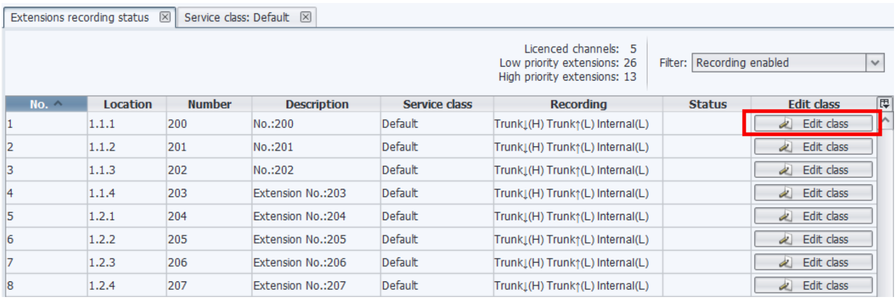
Note: For better support of calls answered by groups and more advanced statistics check the Platan Call Monitor software.

12. Call recording

12.1 Extensions recording status

In *Recording → Extensions recording status* you can find information about current status of call recording service for extension users:

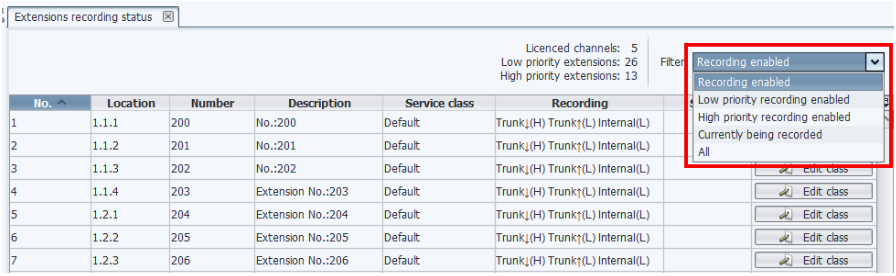
- *Licenced channels* – number of licenced call recording channels.
- *Low priority extensions* – number of extensions with low priority recording assigned.
- *High priority extensions* – number of extensions with high priority recording assigned.
- *Location*, *Number, Description* and *Service class* of extensions with recording assigned.
- *Recording* – recording priority.
- *Status* – recording status, information about recording of a current call made by an extension.
- *Edit class* – possibility of *Service class* edition.



No.	Location	Number	Description	Service class	Recording	Status	Edit class
1	1.1.1	200	No.:200	Default	Trunk↓(H) Trunk↑(L) Internal(L)		Edit class
2	1.1.2	201	No.:201	Default	Trunk↓(H) Trunk↑(L) Internal(L)		Edit class
3	1.1.3	202	No.:202	Default	Trunk↓(H) Trunk↑(L) Internal(L)		Edit class
4	1.1.4	203	Extension No.:203	Default	Trunk↓(H) Trunk↑(L) Internal(L)		Edit class
5	1.2.1	204	Extension No.:204	Default	Trunk↓(H) Trunk↑(L) Internal(L)		Edit class
6	1.2.2	205	Extension No.:205	Default	Trunk↓(H) Trunk↑(L) Internal(L)		Edit class
7	1.2.3	206	Extension No.:206	Default	Trunk↓(H) Trunk↑(L) Internal(L)		Edit class
8	1.2.4	207	Extension No.:207	Default	Trunk↓(H) Trunk↑(L) Internal(L)		Edit class

For easier preview one of the following filters may be selected:

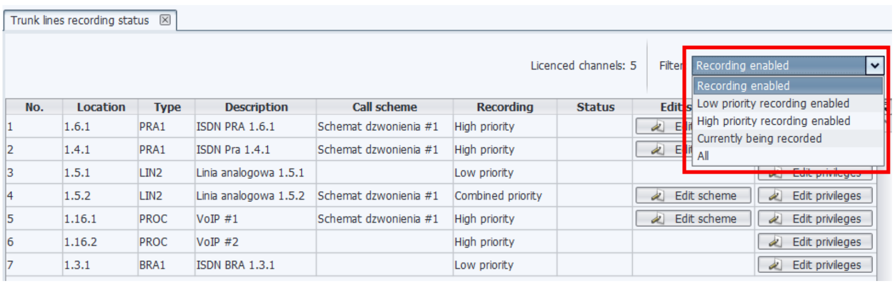
- Recording enabled.
- Low priority recording enabled.
- High priority recording enabled.
- Currently being recorded.
- All.



12.2 Trunk lines recording status

In *Trunk lines recording status* tab you can find information about current status of call recording service for trunk lines:

- *Licenced channels* – number of licenced call recording channels.
- *Low priority lines* – number of lines with low priority recording assigned.
- *High priority lines* – number of lines with high priority recording assigned.
- *Location*, *Type, Description* and *Trunk lines class* of trunk lines with recording assigned.
- *Recording* – recording priority.
- *Status* – recording status, information about recording of a current call made by a trunk line.
- *Edit class* – possibility of *Trunk lines class* edition.

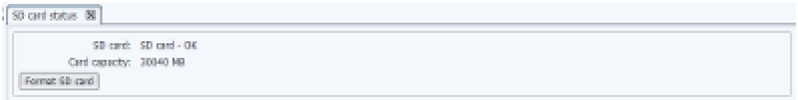


For easier preview one of the following filters may be selected:

- Recording enabled.
- Low priority recording enabled.
- High priority recording enabled.
- Currently being recorded.
- All.

12.3 SD card status

The SD card status can be checked in *Recording* → *SD card status* menu.



13. Monitor

13.1 Extensions

In *Monitor → Extensions* tab the current status of Proxima IP PBX Server/ Libra PBX Server extensions are visible: idle, engaged, calling. The called extension, number and type of a trunk line by which a call is made, and the call direction (incoming/outgoing) can be observed as well.

Extensions

No.	Extension number	Location	Status	Number	Location	Type	Direction
1	201	1.3.1	-----				
2	202	1.3.2	-----				
3	203	1.3.3	-----				
4	204	1.3.4	-----				
5	205	1.3.5	-----				
6	206	1.3.6	-----				
7	207	1.3.7	-----				
8	208	1.3.8	-----				
9	209	1.7.1	-----				
10	210	1.7.2	-----				
11	211	1.7.3	-----				
12	212	1.7.4	-----				
13	213	1.7.5	-----				
14	214	1.7.6	-----				

☑ Idle ☑ Engaged

Legend: <-- Incomming call --> Outgoing call

13.2 Trunk lines

In *Trunk lines* tab the trunk lines statuses are presented: idle, engaged, no SIM card, last received CLIP and possible line damage is signalised.

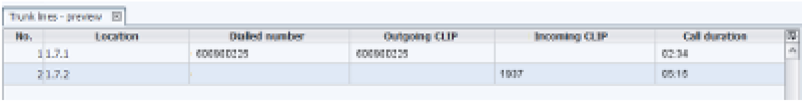
Trunk lines

No.	Location	Line type	Status	Last received CLIP	Damaged
7	1.5.5	Analogue	-----		
8	1.5.6	Analogue	-----		
9	1.6.1	ISDN PRA	-----		
10	1.8.1	GSM	No SIM card		
11	1.8.2	GSM	No SIM card		
12	1.8.3	GSM	No SIM card		
13	1.8.4	GSM	No SIM card		
14	1.14.1	ISDN PRA	-----		
15	VoIP 1 (lm)	VoIP	-----		
16	VoIP 2 (lm)	VoIP	-----		
17	VoIP 3 (lm)	VoIP	-----		
18	VoIP 4 (lm)	VoIP	-----		
19	VoIP 5 (lm)	VoIP	-----		
20	VoIP 6 (lm)	VoIP	-----		

☑ Idle ☑ Engaged ☑ GSM errors ☑ Damaged

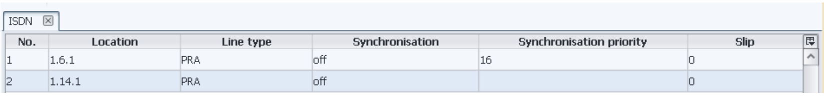
13.3 Trunk lines – preview

The *Trunk lines – preview* tab shows information about currently established connections.



13.4 ISDN

In *Monitor → ISDN* tab, the following information about ISDN BRA and ISDN PRA lines status are visible: *synchronisation*, *synchronisation priority* and *slip*.

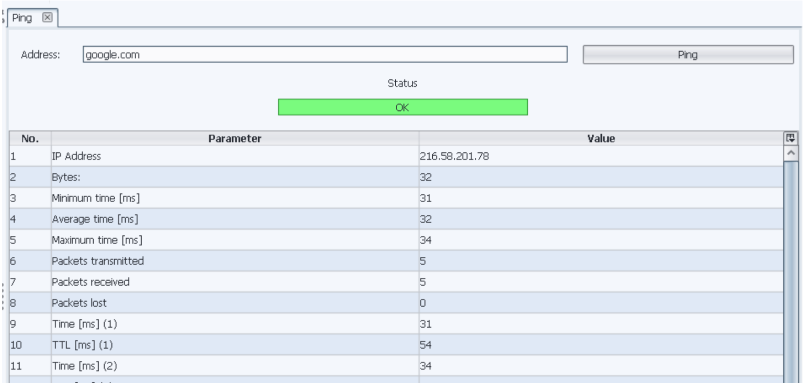


13.5 Server

Basic current information about the Proxima IP PBX Server/ Libra PBX Server: *date and time*, *server operation mode* and *counted calls*.

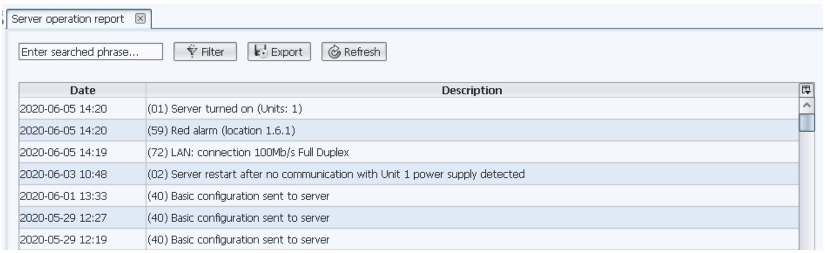
13.6 Ping

In *Monitor → Ping* tab, the basic diagnostics of network connections is possible. In the *Ping* tab you can check whether a connection between testing and tested hosts exists. The number of lost packets and network delays (ping time) can be measured.



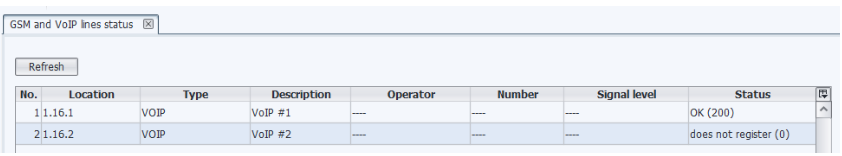
13.7 Server operation report

This tab contains information about the last events occurred in the Proxima IP PBX Server/ Libra PBX Server.



13.8 GSM and VoIP lines status

In *Monitor → GSM and VoIP lines status*, the GSM and VoIP lines status can be previewed.

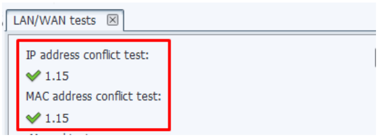


14. VoIP diagnostics

Software tools for running tests and a preliminary diagnosis of VoIP communication.

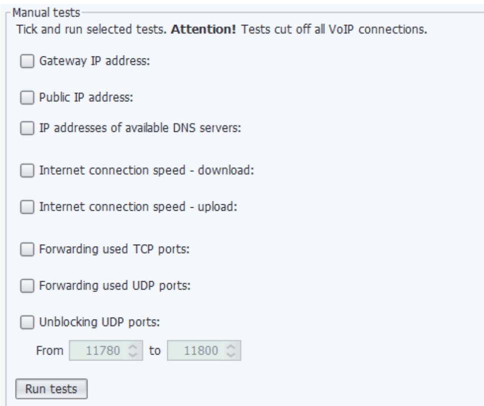
14.1 LAN/WAN tests

At the beginning *IP address conflict test* and *MAC address conflict test* are run automatically. Their results are shown above the *Manual tests* box.



14.1.1 Manual tests

IP settings tests in Proxima IP PBX Server/ Libra PBX Server:

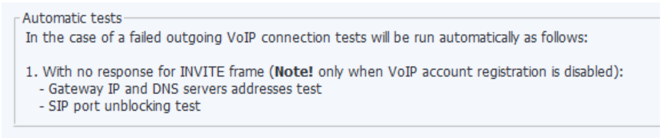


- *Gateway IP address*
- *Public IP address*
- *IP addresses of available DNS servers*
- *Internet connection speed – download*
- *Internet connection speed – upload*
- *Forwarding used TCP ports*
- *Forwarding used UDP ports*
- *Unblocking UDP ports*

Tick the required tests and click *Run tests* button to run the selected tests.
The manual test results are saved in: *VoIP diagnostics → VoIP work report*.

14.1.2 Automatic tests

The tests presented in the *Automatic tests* box are run automatically in the case of a failed outgoing VoIP connection.



The automatic tests results are saved in: *VoIP diagnostics → VoIP work report*.

14.2 VoIP efficiency tests

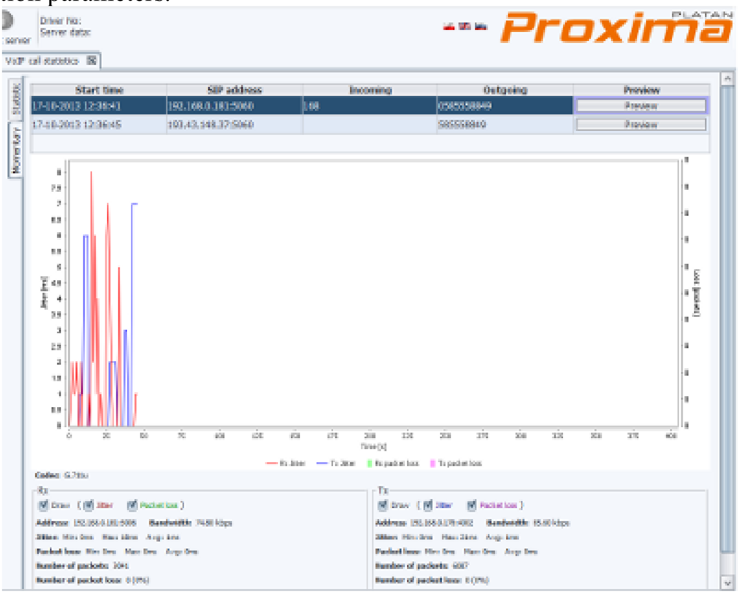
VoIP efficiency tests make it possible to check the number of simultaneous connections along with their transmission and reception parameters. The tests are based on the *test server.*

14.3 VoIP call statistics

VoIP call statistics make it possible to check parameters of VoIP connections. Installer has the access to the call statistics from a given period (*Statistic* tab) or current calls statistics (*Momentary* tab).

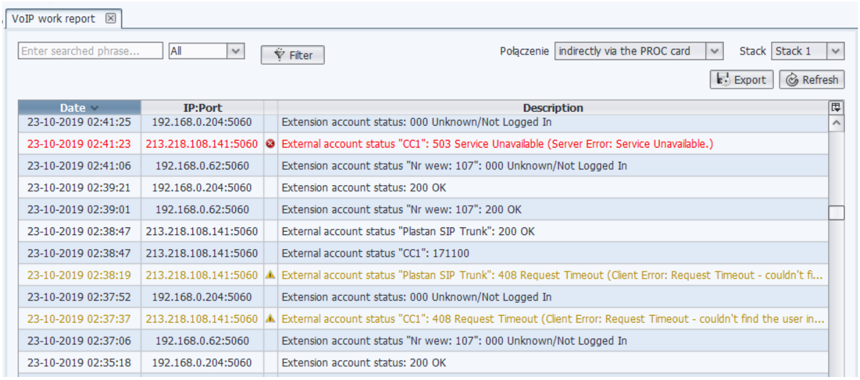
Using Ctrl+ mouse button you can move the graph, and by highlighting the area you would like to see in details – you can enlarge it.

In *Momentary* tab, after clicking the *Preview* button you'll see the graph of a given connection parameters.



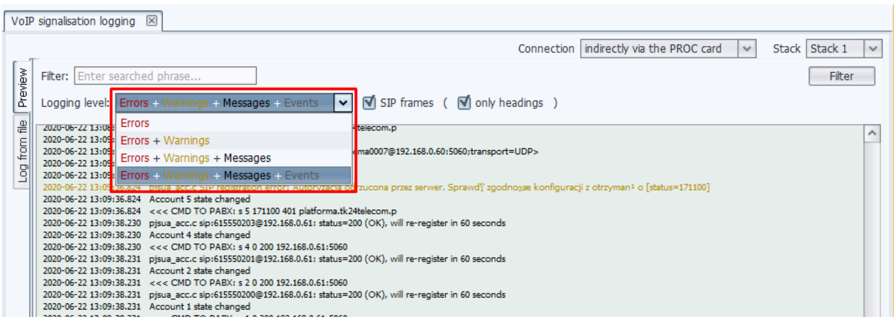
14.4 VoIP work report

In *VoIP work report* tab, VoIP events that occurred in Proxima IP PBX Server/ Libra PBX Server are saved.



14.5 VoIP signalisation logging

In *VoIP signalisation logging* it is possible to follow the VoIP signalisation in Proxima IP PBX Server/ Libra PBX Server.



15. Server programming without connection and by modem

The server offline configuration (without connection to the server) is made by an authorised person with the use of *LibraPC* computer program (the same for Libra and Proxima servers).

15.1 Hardware requirements

The program is to be installed in a desktop or a mobile computer (laptop) meeting the following minimum requirements:

- processor: 1,6 GHz;
- RAM memory: 1 GB;
- graphic card with 128 MB RAM;
- MS Windows XP/Vista/7/10/Linux, MAC operating system;
- Java environment installed.

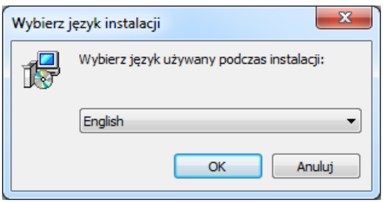
Java environment can be downloaded from www.java.com website.

15.2 LibraPC computer program installation

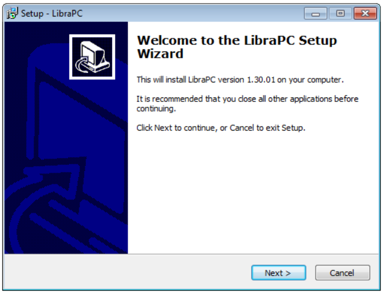
The computer program necessary to operate and programme Proxima IP PBX Server / Libra PBX Server offline is available for Platan Partners.

Installation of the Proxima IP PBX Server / Libra PBX Server offline operation software. Press: *Install LibraPC.*

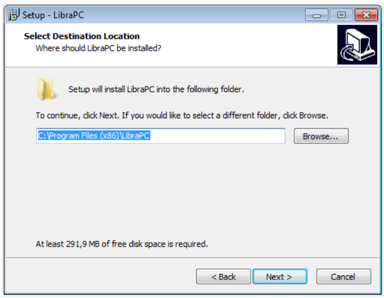
Start the LibraPC_Setup.exe file. At the beginning of the installation process, the installation language can be selected and accepted by pressing *OK*.



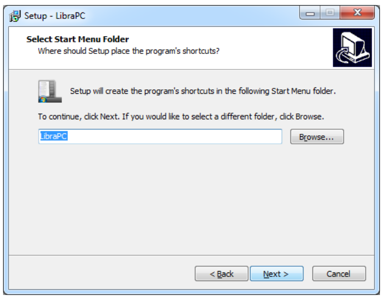
Read the information about installation and click *Next*.



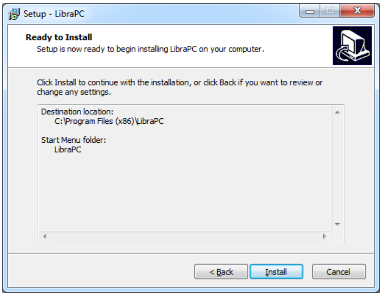
Select the destination location and click *Next*.



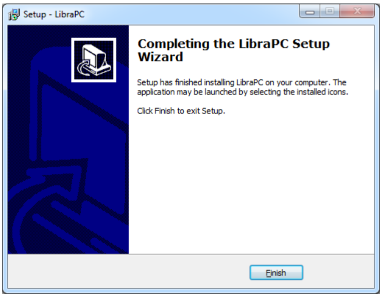
Select the Start Menu folder to create the program's shortcuts and click *Next*.



Confirm installation data by clicking *Install*.



After completing the installation click *Finish*.

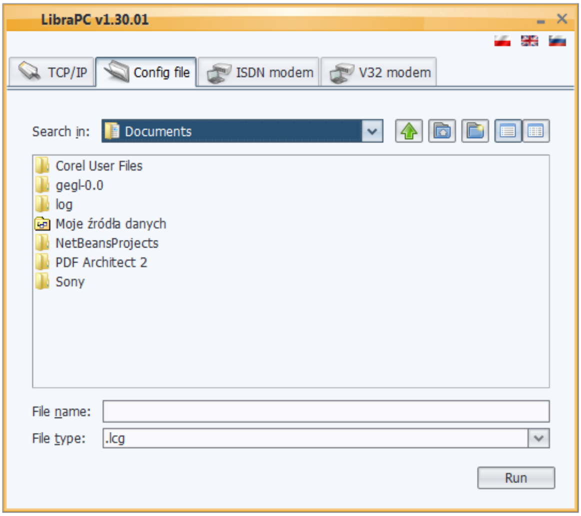


15.3 Libra Launcher

The *LibraPC* icon will appear on your desktop. Double click it to launch the problem or open *LibraLauncher.jar* file with java.

After launching the program, a welcome window opens, where the type of work can be selected:

- *TCP/IP* – work with Proxima IP PBX Server/ Libra PBX Servier via the Ethernet network, alternative to the access via website.
- *Config file* – offline work with a previously saved configuration file.
- *ISDN modem* – work with Proxima IP PBX Server/ Libra PBX Server via ISDN modem connection.
- *V32 modem* – work with Proxima IP PBX Server/ Libra PBX Server via V32 analogue modem connection.

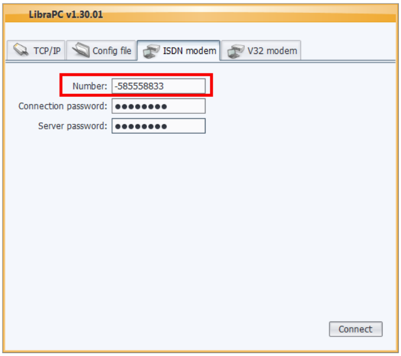


15.3.1 ISDN/V.32 (analogue) modem

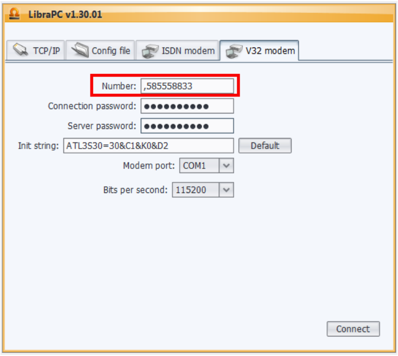
Before connecting to the Proxima/Libra server via modem, a modem password should be defined in *Administration* → *Password change* → *Modem* and a modem *Access number* should be entered in *Common settings* → *Global settings.*

Then the specific MSN/DDI number should be set for modem in *Call distribution* → *MSN/DDI tables* or the incoming connection with a CLIP defined in the ACD table should be directed automatically to a modem (*Call distribution* → *ACD tables*, an incoming number defined in *External prefix* column and ‘modem’ selected in *Call recipient* column).

ISDN modem access number in LibraPC should be preceded with a dash (“-“).



V32 modem access number in LibraPC should be preceded with a comma (“,”).



15.4 Programming

Server programming in LibraPC program is the same as the one described in this manual for ProximaWeb/ LibraWeb.

16. Platan Hotel software

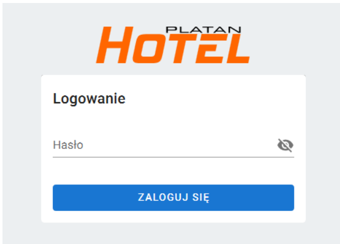
Platan Hotel is a software embedded in Proxima IP PBX Server and Libra PBX Server. A licence is required to use the software.

Platan Hotel software is available from the Internet browser (the latest versions of Chrome and Firefox browsers are recommended).

In order to activate the software window, enter the following adres in the Internet web browser address bar:

http://<PBX server's IP address>/hotel/

The logging window will appear:



The default software language depends on the language of the computer system. It will be Polish, Russian or English (for all other languages). It can be changed in *Opcje (Settings) -> Ogólne (General)* after logging in.

Default logging password: 77777777

The programming videomanual of Platan Hotel software can be found on our YouTube channel in "Wideoinstrukcje" playlist (with English subtitles)

https://www.youtube.com/user/PlatanPl

17. Glossary

DTMF (Dual Tone Multi Frequency) is the name of tone signalling used in telephone devices. It is also known as TouchTone®. DTMF is an example of MFSK (Multiple Frequency Shift Keying).

DISA (Direct Inward System Access) is the name of a functionality offered by PBX telecommunications systems, enabling you to call an extension user without going through an operator.
Having connected with a given PBX system, a caller hears an announcement and while or after listening to it, dials the desired extension number using a telephone with DTMF tone dialling.
A drawback of this system is that the caller pays for the call from the moment the DISA device answers the phone, i.e. they pay for the time that passes until the phone is answered after the extension number has been dialled.

DDI (Direct Dial-In (Europe) or Direct Inward Dialing (USA)) is the name of a service in PBX systems, enabling you to call an extension user directly.
A PBX server user is provided by the operator with at least 10 telephone numbers and each of them is assigned to individual server extension users. For example, in the case of 7-digit numeration, a user receives 100 numbers in the 1234500-1234599 range. 12345 functions then as the server number, while the last two digits are used as the number of an extension user. DDI numeration using the last one or the last three digits is also possible (a user may then receive either 10 or 1000 numbers). A connection between the server and the operator’s local exchange is most often established using an ISDN line.

DDI offers the following advantages:
It makes the PBX server operator's assistance no longer necessary and eliminates the need of using an additional extension number.
It is also cheaper than buying from the operator a separate telephone line for each extension user. For example, you pay the operator for a few lines only and have 100 DDI numbers at your disposal.

CLIP (Calling Line Identification Presentation) is a service that consists in presenting the caller's number on the telephone display (or on a special telephone snap-in’s display). This service is available for both land line and mobile networks.

CLIR (Calling Line Identification Restriction) is a service that blocks the presentation of the caller's number. Thanks to this functionality, the number of the user initiating the call will not be shown even if the other party has the CLIP function on.

MSN (Multiple Subscriber Number) is the name of the service in PBX systems using ISDN phone lines.
Thanks to MSN several (up to 8) appliances (phone, modem, fax) can be connected to an ISDN terminal and assign each of them a different telephone number.
MSN offers the following advantages:

– It makes the PBX server operator's assistance no longer necessary and eliminates the need of using an additional extension number.
– It is also cheaper than buying from the operator a separate telephone line for each extension user. For example, you pay the operator for one ISDN line with a packet of MSN numbers and have 5 MSN numbers at your disposal, which is much cheaper than buying 5 separate lines.

Prefix is a string of digits following the 10XX or 10XXX pattern (where X stands for any digit), identifying a given telecommunications operator.
Telecommunications operators and the calls made by their customers are identified within the telecommunications networks by means of assigned prefixes. Prefix is officially referred to as the network access number. In a sense, prefixes enable communication between customers and telecommunications networks (owned by telecommunications operators). By choosing a given prefix, a customer chooses the operator through the network of which a call is to be carried out.