2N® - OMEGA Lite VoIP module

USER'S MANUAL
VoIP

version 1.0

Dear customer,

Our compliments on buying the ***2N*****Ò - OMEGA Lite***. This new product was developed with an emphasis on the maximum possible use value, quality and reliability. We hope that you will be utterly satisfied with the 2N*Ò - OMEGA Lite *for many years to come.*

CONTENTS

1. Packing list

Item	Quantity
VoIP module	1
USB cable	1
PC LAN cable	1
Connecting cable	2
M3x30 spacer	2
Warranty Certificate	1
Conformity Certificate	1
CD	1

2. ATEUS â - Omega VoIP module

The VoIP Module is designed exclusively for **2N® OMEGA Lite** PBXs as a VoIP module for LAN and Internet voice services.

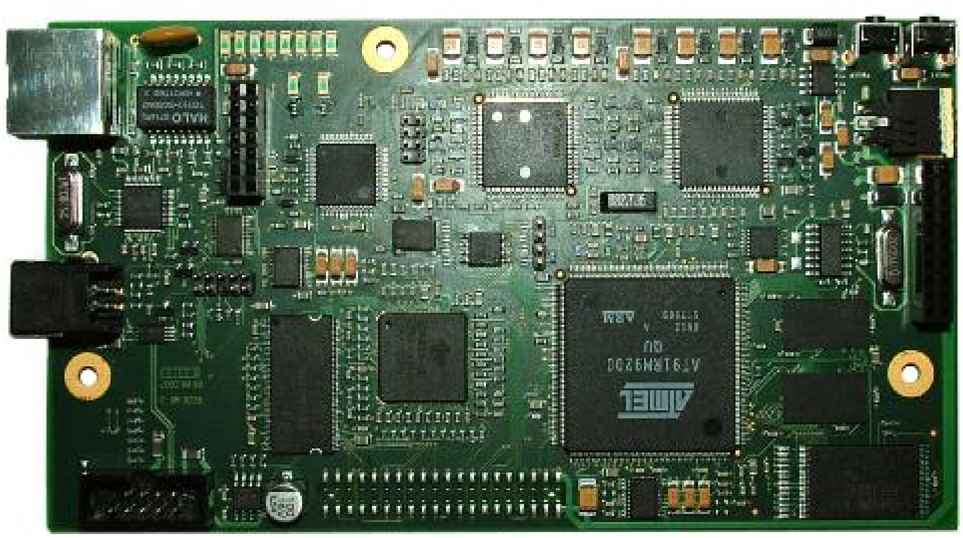
The VoIP Module is to be installed to the left from the power supply unit, above the main board. To connect the module, use the two connecting cables packed in the separate plastic bag (insert them to the J15/J26 connectors on the main board) and the two spacers (also packed in the separate plastic bag). Use the remaining two M3 screws to screw the VoIP Module to the spacers.

Remove the two J8/J9 links on the PBX base plate for appropriate operating efficiency.

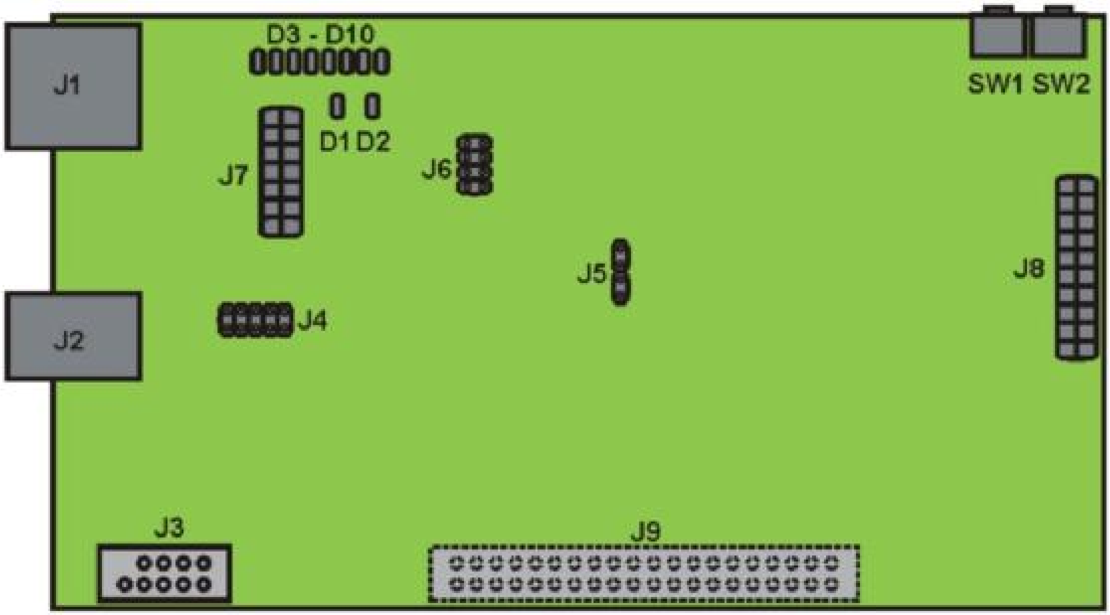
Use the USB cable and the hyper-terminal for the initial setting of the IP address and the network mask; use the (available) web configuration for further settings.

The system connects to the 10/100BASE-T (Twisted Pair Ethernet) computer network through a standard direct cable with RJ 45 connectors at the end.

See the VoIP Module User Manual on the enclosed CD for detailed installation information.



2.1. VoIP module



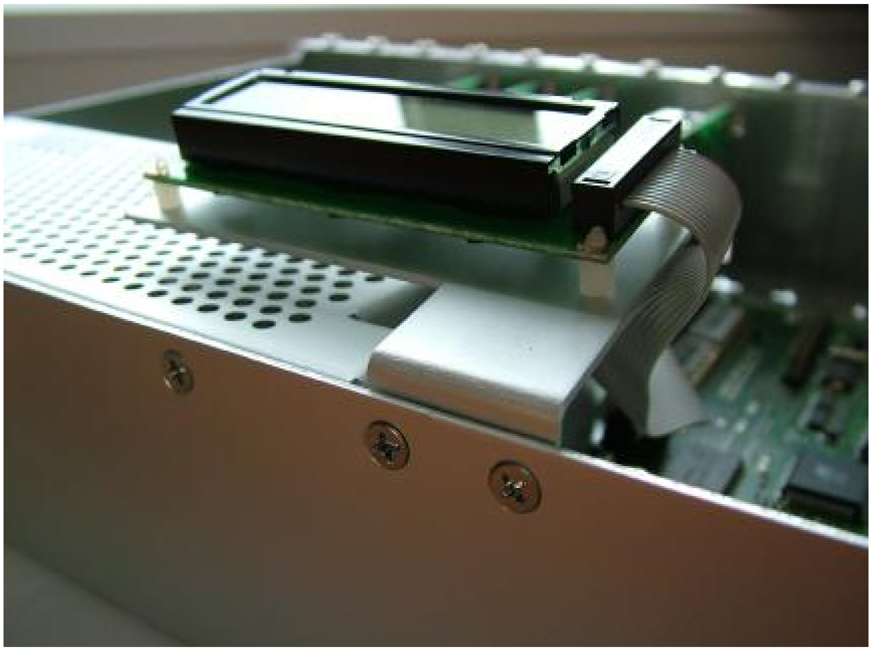
The following is located on the VoIP module:

J1 - RJ45 connector - 8/8 for LAN connection.
J2 - USB connector for USB connection, designed for IP configuration.
J4,J8 - serial connector designed for connection with the main board of the PBX.
J3 - connector for the connection of adapter for an external USB device (USB FLASH disk).*
J5-J7 - connector for maintenance use
*J9 - connector for connection of a functional floor**
SW1 - RESET button. Pressing this button will RESET the CPU board.
SW2 - RESET button. Pressing this button will reset default values (after switching on the PBX hold the button pressed for approximately 20s until LINUX starts).
D1 - green LED – operation status indication (VoIP module ready for operation, approximately 20s after light-up).
D2 - red LED – operation status indication (VoIP module inactive).
D3-D10 - yellow LED – call channels busy signal.

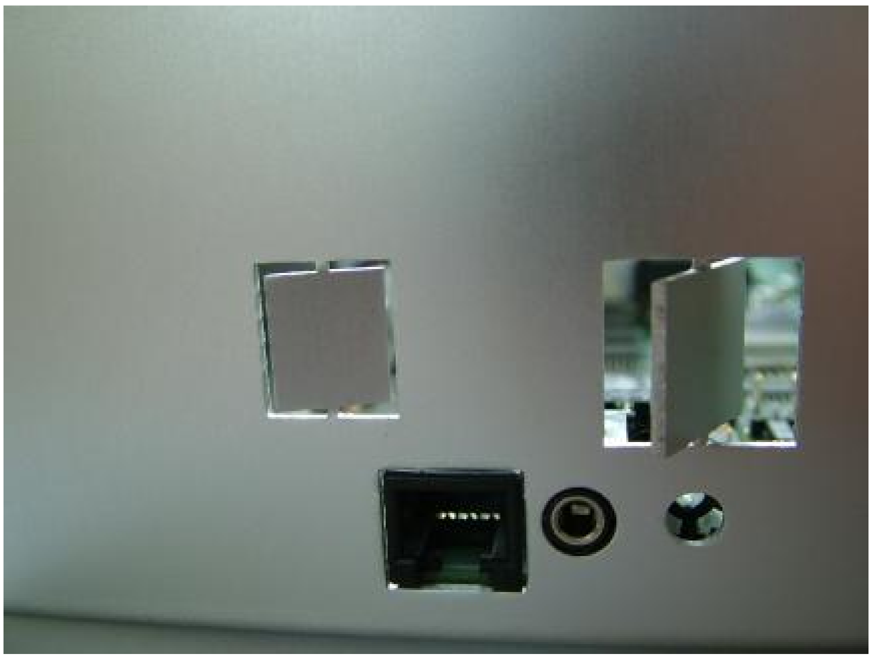
* In preparation

3. Installation of the module

Remove the display holder by unscrewing 2 screws on the chassis.

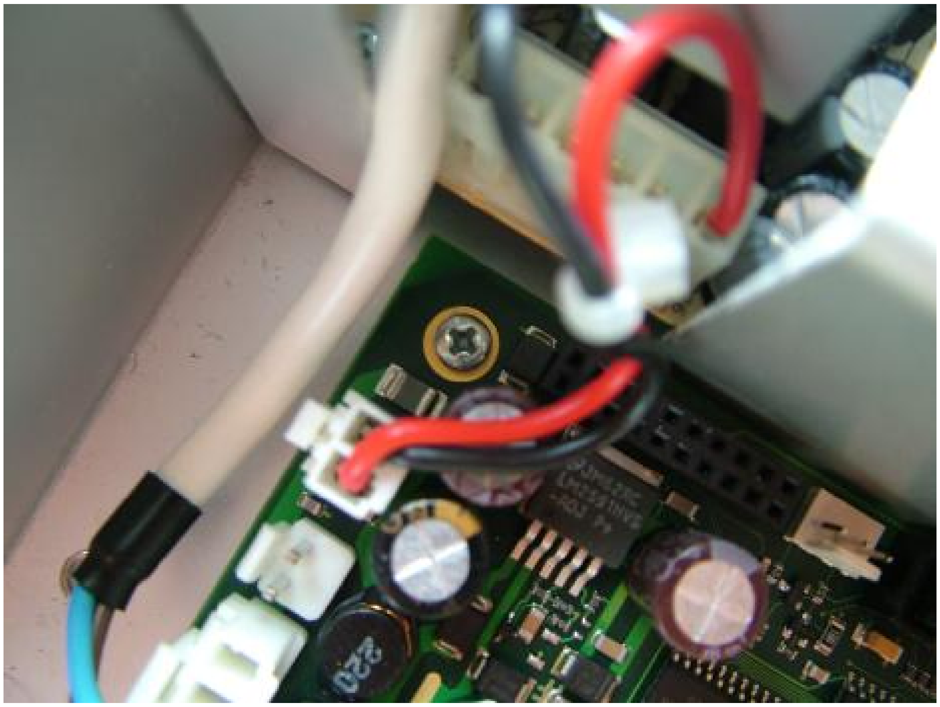


Remove by turning the end cap on the left-hand side of the chassis for J1 and J2 connectors.



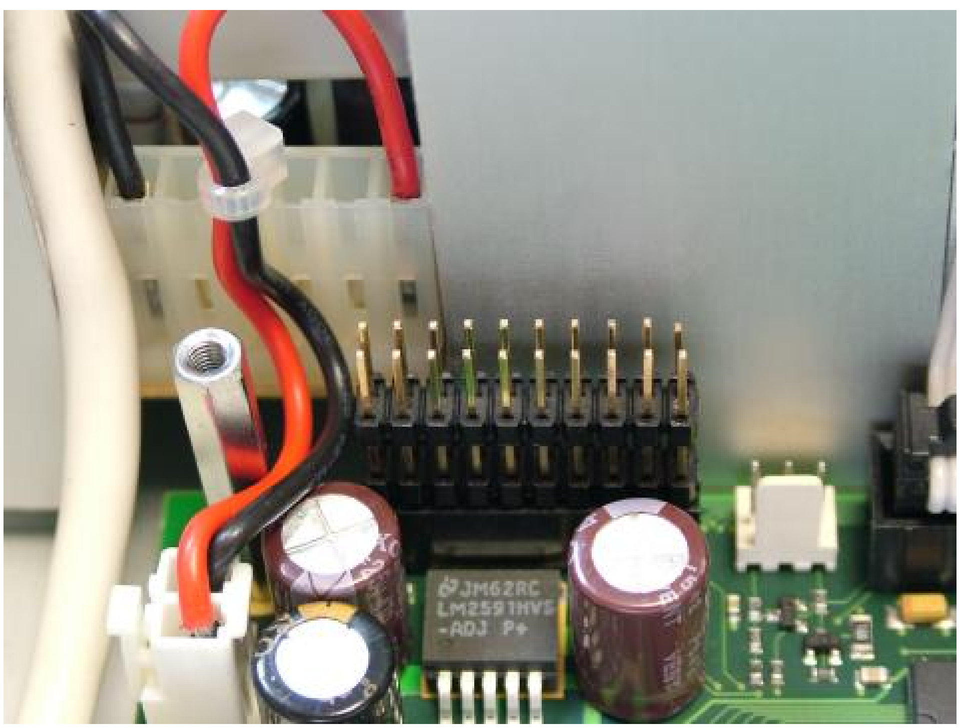
Remove 2 JUMPERs, J8/J9, from the main board.

Unscrew 2 M3 screws from the left and right corner of the main board (these will be used for fixing of the VoIP module) and replace them with the spacers attached.



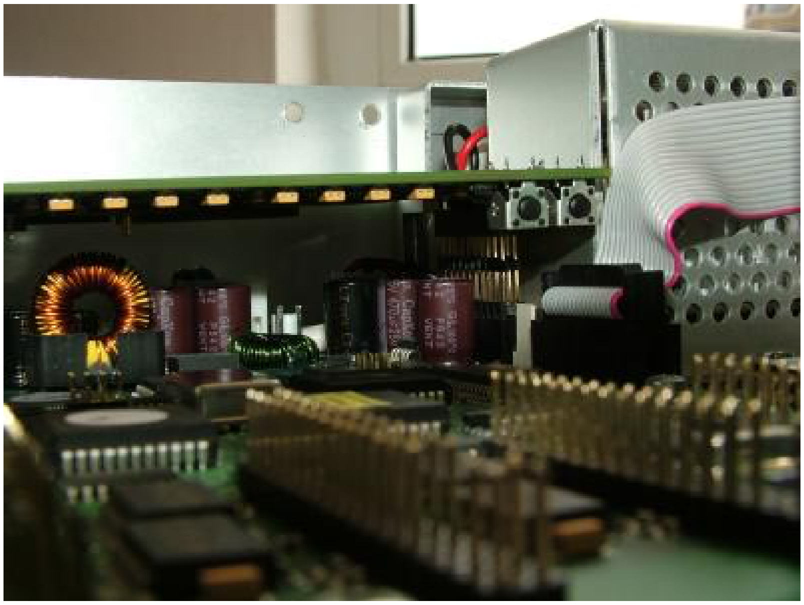

POZOR

Insert the two connectors attached into the J26 and J15 connectors on the main board.

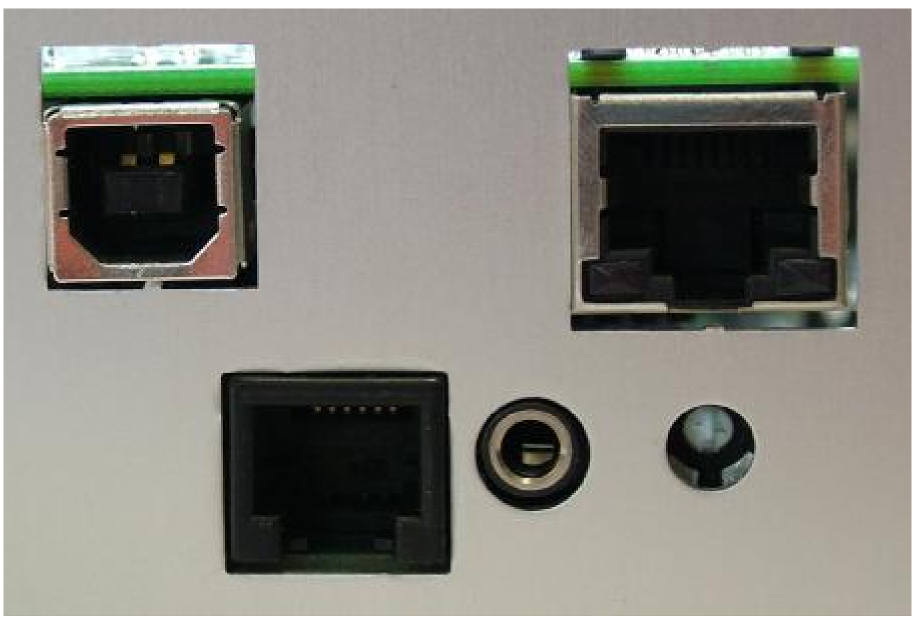


Place the VoIP module onto the connectors inserted and fix it with two screws.

Place the module onto the connectors carefully to avoid shifting of the individual PINS.

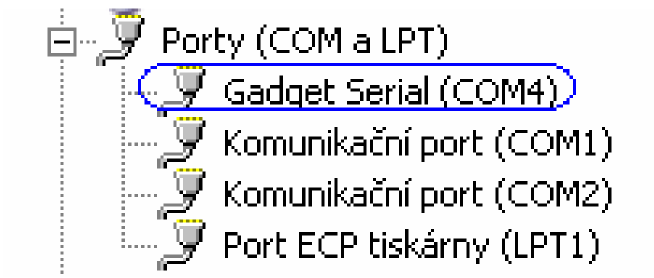


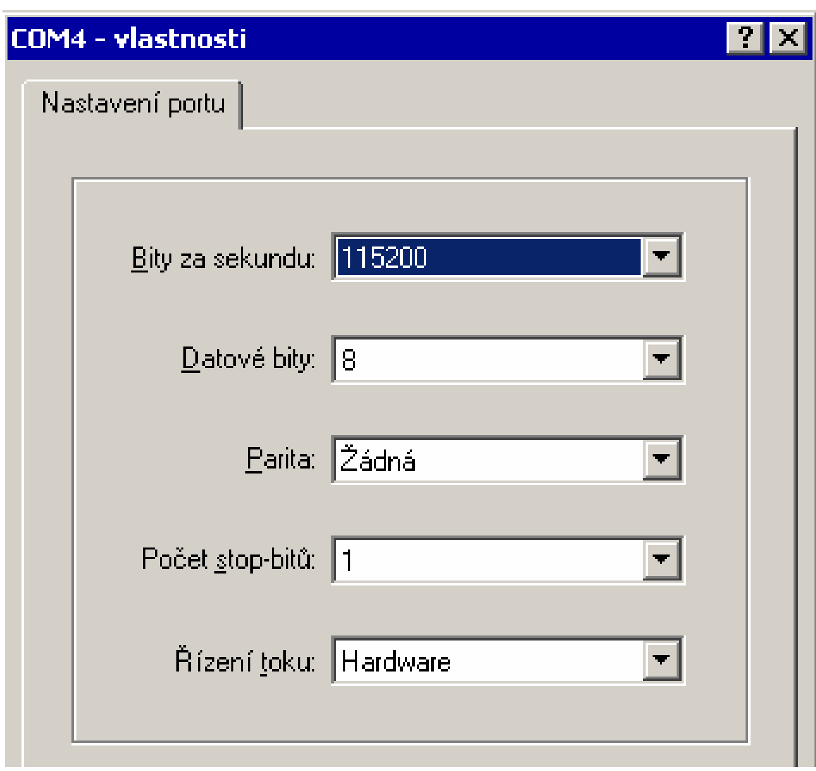

POZOR!
07-1179-0004

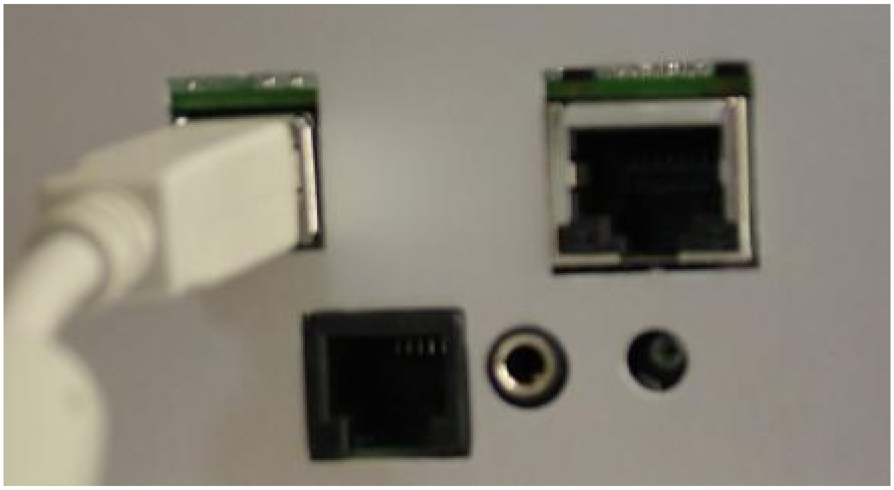


Connect the USB cable inserted (into J2 connector) and connect it with your PC.

When connecting for the first time it is necessary to install USB-COM driver (on the "data/OMEGA Lite/Cz/Software/VoIP/DriverUSB/" CD attached). A new COM PORT is added after installation. After this you may connect by hyper-terminal and perform the necessary settings.







Serves for network parameter setup (IP address/network mask,.).
After the network parameters have been set up you may disconnect.

Always RESET the module using the SW1 button after disconnecting the USB cable.

Connect a standard direct Ethernet cable into the J1 connector.

This cable connects the VoIP module with your Ethernet network.

After turning the PBX power on you may start configuring the VoIP module.

4. Configuration of VoIP module

The VoIP module for the *ATEUS*® - OMEGA central has pre-set from the producer:

Communicating port of serial console:
Speed: 115200
Data bits: 8
Parity: none
Flow control: none

Network parameters:
IP address: 10.0.0.1
network mask: 255.0.0.0

The module may be configured by two ways:

1) Connect the module by means of the USB line to the PC and set the basic network parameters by means of communicating software, e.g. HyperTerminal. Then, connect it by means of the web browser and complete the setting.

2) Connect the module with the PC by means of crossed Ethernet cable, set on PC any IP address from the network 10.x.y.z except the 10.0.0.1, and set the required IP address of the module by means of the web browser. After change of the IP address, connect the PC and the module to the network as usually and complete the configuration via the web browser.

4.1. Connection of USB line

To configure the module connect the inserted USB cable with the VoIP module (J2 connector) and the PC.
Install the driver from the CD attached to create a virtual" COM port (COM4).
Launch e.g., hyper-terminal, set it up for receiving on the COM port created for the new USB connection and set up the network parameters.

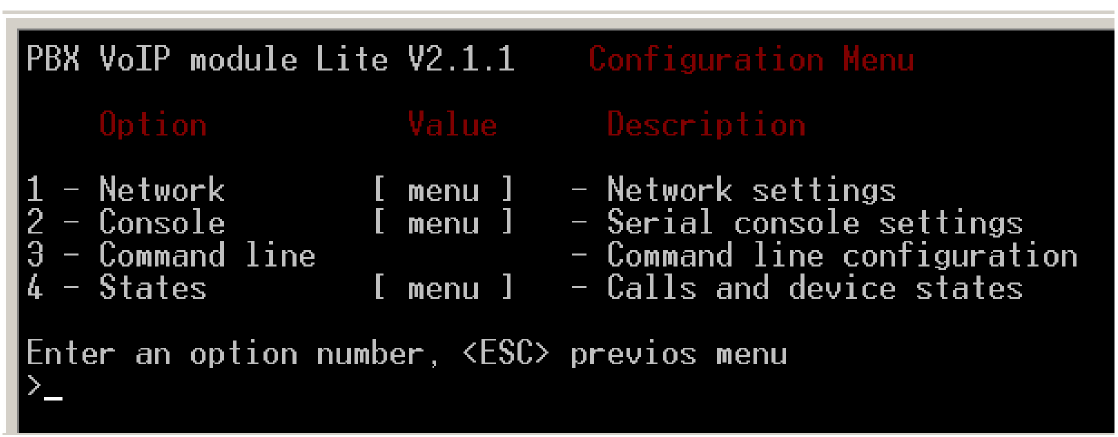
The console system is organized as a set of nested menus. Selecting an item results either in change into sub-menu, performance of the given operation, or set-up of the parameter selected.
The main menu should appear after connecting the terminal if the module is connected. In some cases it will be necessary to press Enter.

Each menu consists from following sections:

- **Heading:** contains (from left) product name, firmware version, menu name and network name assigned to the equipment.
- **Selection column:** displays numbers and names of available options.
- **Value column:** if the „[menu]“ is displayed, then the menu covers another sub-menu. Otherwise, it is actual value of the option.
- **Label column:** briefly describes the meaning of each option in the menu.
- **„Enter an option number… >“:** call for the entering of the option number.

The option from the menu is realised by entering of its number and it is confirmed by pressing of the Enter.

4.1.1. Sub-menus



If we select the item corresponding to the sub-menu, this sub-menu will be displayed. Now, we may select the items from the selected menu or get back by pressing of the  **key.**

4.1.2. Commands and values

If the entering of some data is required, it may be the value of one of the following types:

- **Key word:** list of one of more fixed strings. For selection of one of them the entering of such number of characters, which distinguishes it from other key words, is sufficient.

  `Enter one of [ansi,color,teletype] : a`

  In above stated example, we may select the option by entering the character: „a“, „c“ or „t“ followed by pressing of the 

key.

- **Chain:** any number of characters. The range of the available value length may be displayed.

  `Enter a hostname from 1 to 32 characters: voiceblue`

- **Integer number:** Decimal integer number. The range of the available value length may be displayed.

  `Enter a size between 1 and 100 : 99`

  Hexadecimal integer number – number entered in hexadecimal number system using the characters 0÷9 and a÷f of A÷F.

  `Enter a hex number between 1h and ffh : 1a`

- **Network address:** 12 or less hexagonal numbers of the physical address. Zeros at the beginning may be omitted.

  `Enter the remote network address : 50C229C4E2`

- **IP address:** internet address in form of four numbers 0÷255 separated by dots.

  `Enter an IP address : 192.168.22.30`

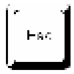
As soon as the required information is entered, the relevant operation will be realised. If the given operation changes the setting of the parameters, the new values will be displayed in the updated menu.

Some of the configuration parameters may be of one or more fixed values. If you select such item of the menu, its value is changed immediately to the opposite state compared to the state before the selection. Typical examples are the parameters of the values *on* or *off*. If the value of such item is *on*, then after its selection the value is switched to *off* and vice versa.

Some commands realise the operations, which significantly influence the behaviour of whole system (e.g. restart of the unit). Before their realisation, the system usually confirms, that you really wish their realisation.

```
Are you sure [y/n] :
```

If you answer to such question anything else than „y“ or „Y“, the command will be aborted.

The call for entering of any information may be aborted at any time by pressing the  **button, and the command realisation will be aborted.**

4.1.3. About menus …

Main menu: is displayed after connection of the serial cable to the terminal. From the main menu, we may get to the configuration menu, change the administrator passwords or to display the help.

Configuration menu: contains two sub-menus: network configuration and serial console configuration.

Network configuration: serves for setting of the IP address, network mask, initial router, DNS server addresses, network name and domain.

Serial console configuration: enables setting of the terminal type and change the initial parameters of the serial line – speed, number of data bits and stop bits, parity and type of the flow control.

4.1.4. Monitoring of the DTR signal

The firmware continuously monitors the Data Terminal Ready (DTR) signal. This signal is used for detection of presence or absence of the connection of the terminal for the serial port of the console.

Following activities will be realised after detection of the terminal disconnection:

- **Initialised operations will be aborted**
- **Main menu will be displayed**

4.2. Access from the web browser

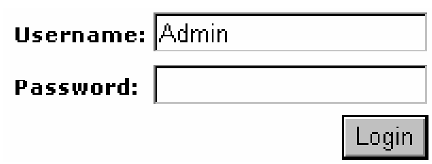
While the serial console interface enables changing of only basic parameters, the setting of all parameters of the VoIP module and all its services is available via the web browser.

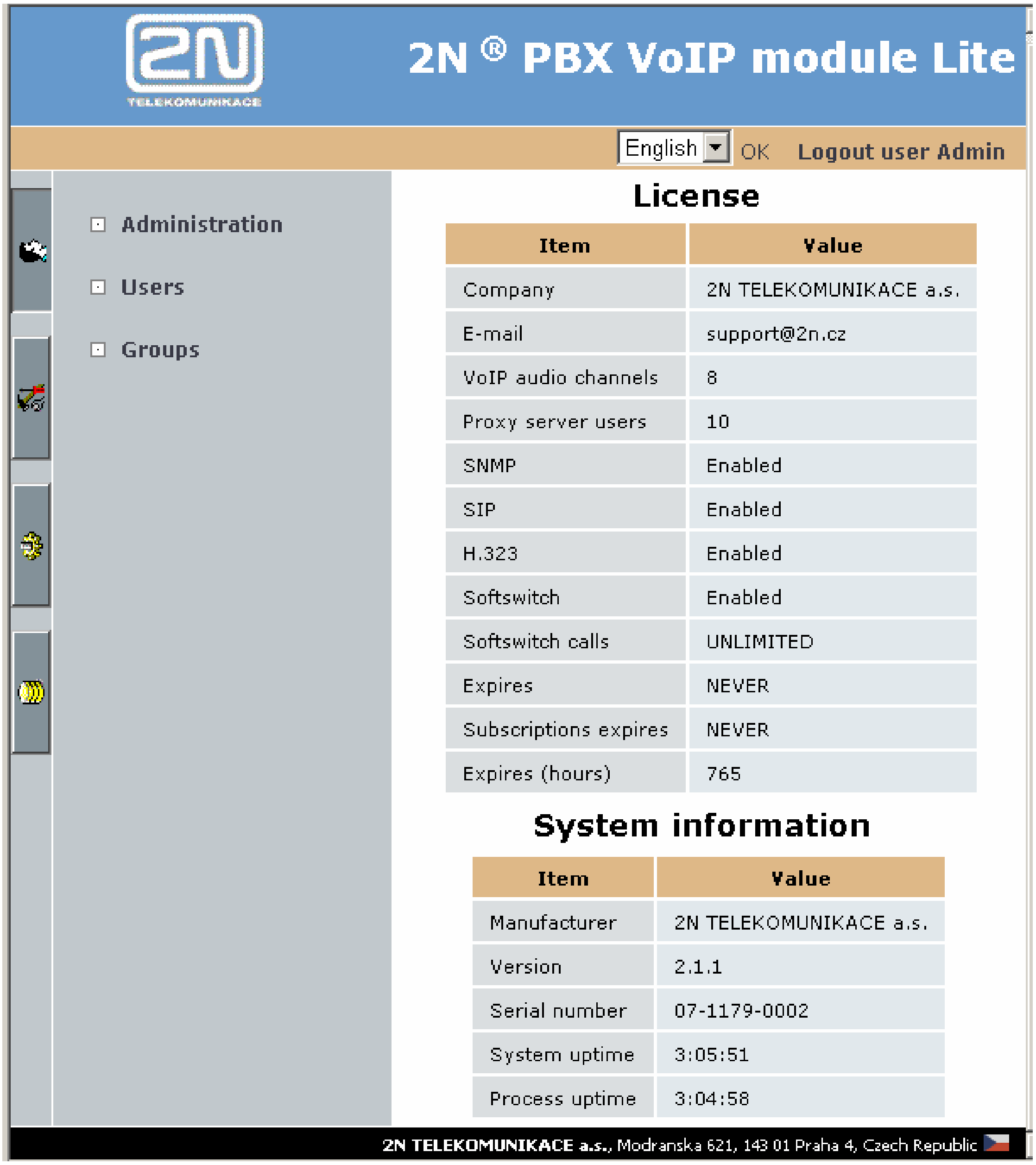
Connection with the module is started by entering the IP address of the module into the line for entering of the Internet address, e.g.:

http://10.0.0.1

If everything is connected properly and IP addresses of the module and PC with the web server are set correctly, prompt for entering of the user name and password should be displayed.

> The producer delivers the VoIP modules with pre-set user name *Admin* without the password. The user name and password are case specified. One of first operation should be change of the administrator password due to the safety.





License

Item	Value
Company	2N TELEKOMUNIKACE a.s.
E-mail	support@2n.cz
VoIP audio channels	8
Proxy server users	10
SNMP	Enabled
SIP	Enabled
H.323	Enabled
Softswitch	Enabled
Softswitch calls	UNLIMITED
Expires	NEVER
Subscriptions expires	NEVER
Expires (hours)	765

System information

Item	Value
Manufacturer	2N TELEKOMUNIKACE a.s.
Version	2.1.1
Serial number	07-1179-0002
System uptime	3:05:51
Process uptime	3:04:58

After successful log-in into the system, we will get into the basic display of the web application. The window is divided into four basic parts.

Heading line: the window heading contains the pull-up menu for selection of language, in which the user relation will take place; and further the name of actually logged user is displayed with the button for log out.

Ears of the menu tabs: the gateway settings are divided into theme groups, which are arranged into the tabs. The clicking to the tab ear, you will be switched to the given setting group and its options will be displayed in the group menu.

Group menu: the settings in tabs are arranged into double-level menus. Clicking to the first-level menu item will display the sub-menus, which are on the second level. If the item on the first level does not have any sub-menus, or clicking to some sub-menu, the appropriate application form will be displayed on the residual area of the window.

Application form: it is main part of the module user interface. It contains the control elements specific for given selected item from the group menu.

4.2.1. Overview of group tabs and menus

Administration of users: all necessary for the administration of the user accounts is here. Here are following menus:

- **Users** – administration of the accounts themselves
- **Groups** – administration of the account groups

Configuration: the *Configuration* tab covers most of basic settings related to the main functions of the module **–** VoIP lines and connection of the calls.

- **Administration** – common settings of the gateway
- **Equipment** – configuration of VoIP lines
- **Network configuration** – module network parameters
- **Firmware updating** – updating of the gateway firmware

Services: besides its main function, the VoIP module *2N*® - OMEGA Lite offers also other network services. The forms for their setting and control elements for the service switching on/off are in this tab.

- **SIP proxy** – setting of the SIP Express Router service (SIP proxy)

States and records: records on the module operations are written during its activities. These records, their settings and actual state information are available from the tag menu *States and records.*

- **States** – information of actual calls
- **Call tariffication** – records and settings of the calls tariffication
- **Records** – operation records of the firmware
- **Records loading** – packing and loading of operation records

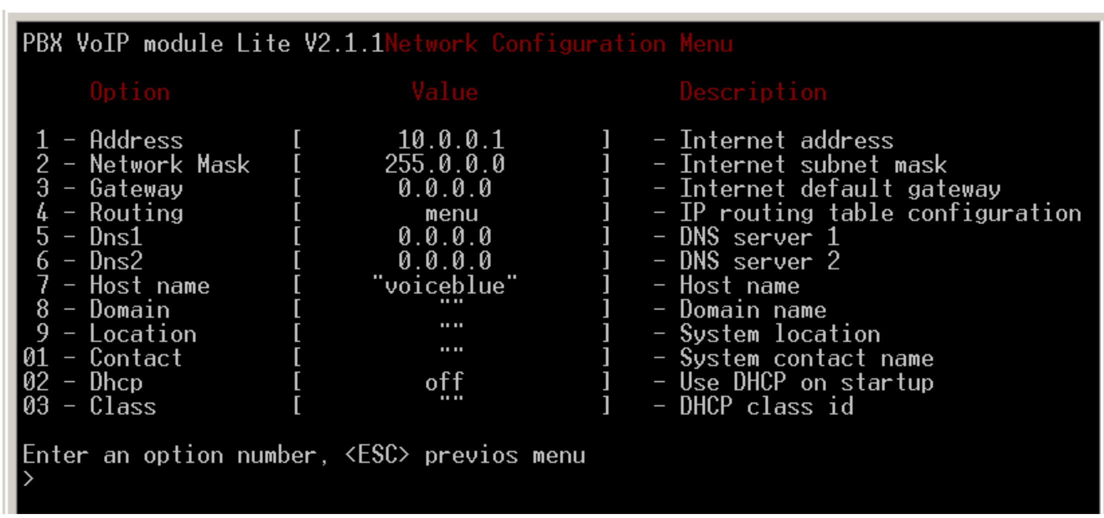
4.3. Setting of the network parameters

Before the module is used for telephoning, it is necessary to set its network parameters. It may be done by two different manners. If the gateway is pre-set from the production or if we know its IP address, we may connect to it the computer by the crossed Ethernet cable and to set the appropriate parameters via the web browser.

In other cases, we appreciate the possible access via the serial port by means of character terminal, e.g. HyperTerminal.

4.3.1. Setting by means of the serial console

The module must be connected to the PC according to the instructions in chap. „*Configuration of VoIP module by connecting of serial line*“. Main menu will be displayed on the terminal. In main menu, we select the *Configuration* item and then the *Network* item. Following menu will be displayed:



```
PBX VoIP module Lite V2.1.1Network Configuration Menu

     Option                  Value               Description

 1 - Address          [     10.0.0.1       ]   - Internet address
 2 - Network Mask     [     255.0.0.0      ]   - Internet subnet mask
 3 - Gateway          [     0.0.0.0        ]   - Internet default gateway
 4 - Routing          [       menu         ]   - IP routing table configuration
 5 - Dns1             [     0.0.0.0        ]   - DNS server 1
 6 - Dns2             [     0.0.0.0        ]   - DNS server 2
 7 - Host name        [    "voiceblue"     ]   - Host name
 8 - Domain           [        ""          ]   - Domain name
 9 - Location         [        ""          ]   - System location
01 - Contact          [        ""          ]   - System contact name
02 - Dhcp             [       off          ]   - Use DHCP on startup
03 - Class            [        ""          ]   - DHCP class id

Enter an option number, <ESC> previos menu
>
```

The menu contains following items:

- **Address** – IP address of the module. If the dynamic assignment of the addresses is activated by means of the DHCP protocol, the value may not be manually changed.
- **Network Mask** – network mask. If the dynamic assignment of the addresses is activated by means of the DHCP protocol, the value may not be manually changed.
- **Gateway** – IP address of initial router. It is used for routing of the data traffic outside the network limit. If the dynamic assignment of the addresses is activated by means of the DHCP protocol, the value may not be manually changed.
- **Dns1** – IP address of the first DNS server.

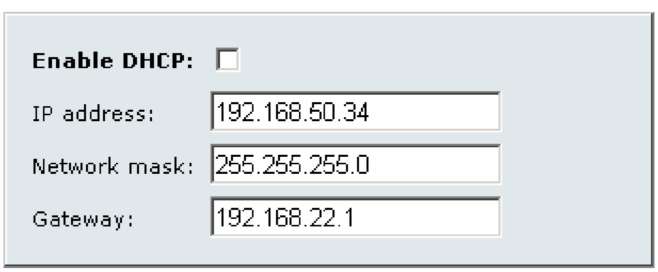
- **Dns2** – IP address of the second DNS server.
- **Host Name** – network name of VoIP module.
- **Domain** – domain name.

- **Location** – any text documenting the module location. This value is published by means of SNMP.
- **Contact** – text containing the contact information to the module administrator. This value is published by means of SNMP.
- **Dhcp** – flag of the dynamic assignment of the IP addresses. If it is activated, the module gets its network setting from the DHCP server. Otherwise the data must be entered manually.
- **Class** – name of the equipment class, which is sent in the request to the sending of the network setting. According to the setting, the DHCP server may distinguish various types of the equipment and to assign them relevant configuration parameters.

Minimally the *Dhcp* or *Address/Network Mask* parameters must be set correctly to enable the communication of the VoIP module in the TCP/IP network. Further, it is recommended to set the address of at least one DNS server.

4.3.2. Setting from the web browser

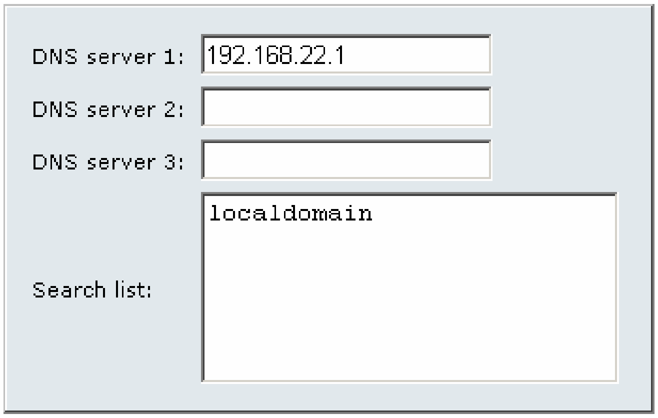
If we are able to connect to the module by the web browser and we know the administrator password, then we can set the network parameters in *Setting* tab in the *Network configuration* menu. After opening of the menu, the actual setting of the network is displayed. The change form may be opened by clicking to the *Change* item located at the right side under the table with the network parameters. Following form will be displayed in the browser:



The form contains following fields:

- **Enable DHCP** – flag, whether the network parameters may be obtained from the DHCP server, or if they have to be entered manually.
- **IP address** – IP address of the module. If the dynamic assignment of the addresses is activated by means of the DHCP protocol, the value may not be manually changed.

- **Network Mask** – network mask. If the dynamic assignment of the addresses is activated by means of the DHCP protocol, the value may not be manually changed.
- **Gateway** – IP address of initial router. It is used for routing of the data traffic outside the network limit. If the dynamic assignment of the addresses is activated by means of the DHCP protocol, the value may not be manually changed.



- **Dns server 1** – IP address of the first DNS server.
- **Dns server 2** – IP address of the second DNS server.
- **Dns server 3** – IP address of the third DNS server.

Changes must be confirmed by clicking to the *Change* item. Setting of the new values takes certain time. Please wait for the web browser response.

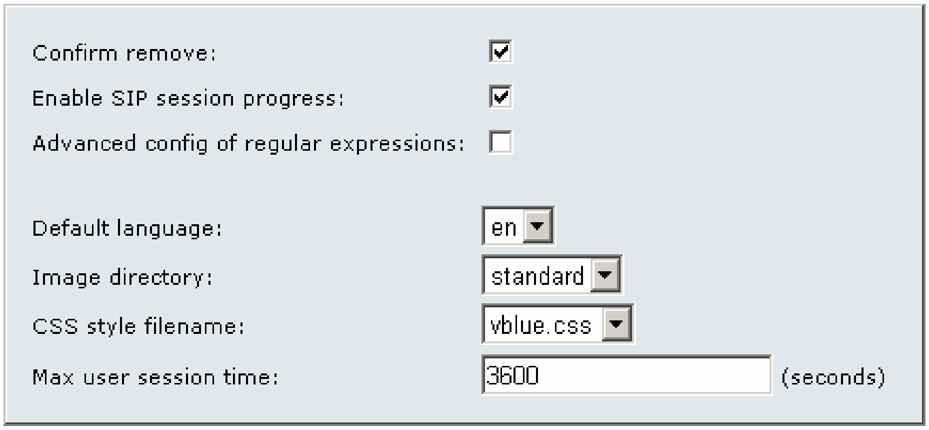
The values of the fields may be returned to the state, in which they were before opening of the form by means of the *Initial* item.

4.4. Administration

In this chapter, we will describe the setting of the general parameters of the VoIP module *2N*® - OMEGA Lite, with creating of the user accounts and the authorisation groups.

4.4.1. General gateway parameters

In the *Configuration* tag, there is the item *Administration*. Clicking to it will open the page with the general parameters of the module. In this form, it is not possible to change the parameter values. We may move to the change mode by clicking to the *Change* item.



The form contains following fields:

- **Confirm remove** – If this mark is selected, then any deleting command is accompanied by a confirmation window.
- **Enable SIP session progress** – This mark enables the SIP stack to send so-called *session progress* messages during call establishing.
- **Extended configuration of regular expressions –** enables configuration of regular expressions in the LCR/Normalizing section.
- **Default language** – The language in which the web interface login page will be displayed.
- **Image directory** – This setting influences the initial appearance of the login page. You can select any set of graphic elements in the pop-up menu.
- **CSS style file name** – This setting influences the initial appearance of the login page. You can select any set of graphic elements in the pop-up menu.
- **Max user session time** – The web interface relation shall be disconnected automatically after a predefined inactivity timeout.

4.4.2. User accounts

The user accounts for log-in into the web interface of for authentication of the SIP telephones to the integrated SIP proxy are defined in the *User administration* tag in *Users* menu. After its selection, the list of user accounts will be displayed.

Name	Group	Language	Line number	Description			
Admin	Administrators	English		Administrator			
siphone	Administrators	English	1111				

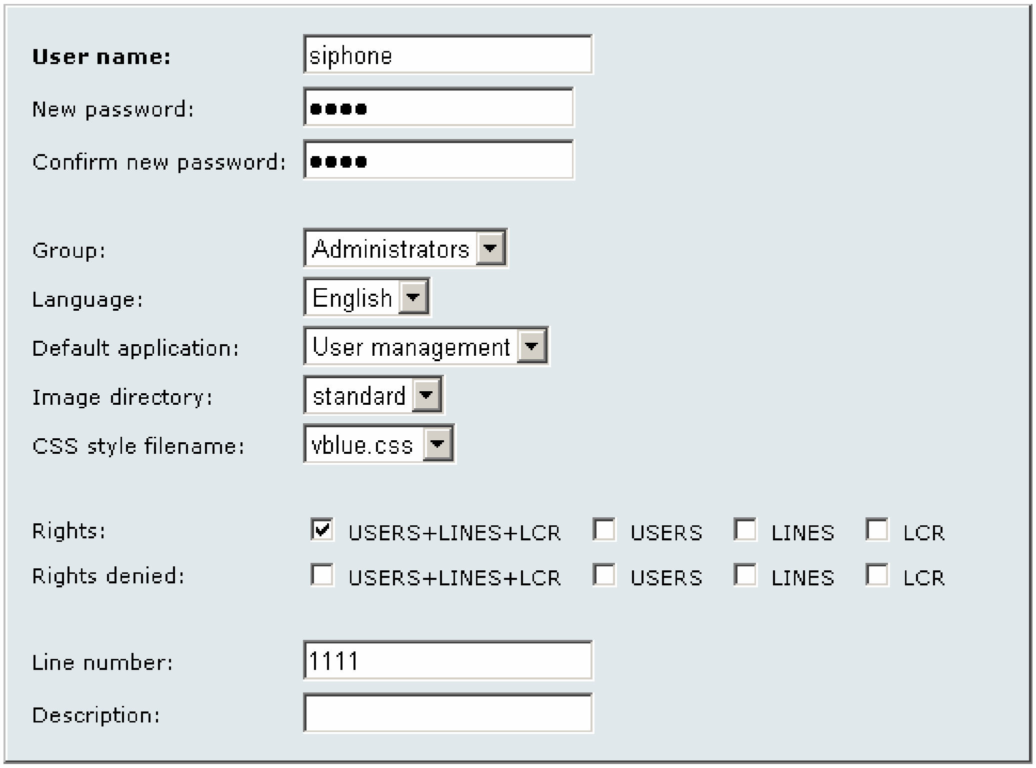
Following items are displayed in individual table columns:

- **Name** – name of the user account.
- **Group** – authorisation group.
- **Language** – language, to which the web interface will be switched to after the user log-in.
- **Description** – any text describing the meaning of the account.

Clicking to the symbol moves us to the user account detail, in which the changes may be done.

Symbol serves for the clearing of the account. The accounts may be cleared also by ticking of the field in the last column of the table and clicking to the *Remove selected* item.

Under the table, there is also the *Add user* item. Clicking to it will display the form for definition of the new user.


User name: siphone
New password: ●●●●
Confirm new password: ●●●●
Group: Administrators
Language: English
Default application: User management
Image directory: standard
CSS style filename: vblue.css
Rights: ☑ USERS+LINES+LCR ☐ USERS ☐ LINES ☐ LCR
Rights denied: ☐ USERS+LINES+LCR ☐ USERS ☐ LINES ☐ LCR
Line number: 1111
Description:

The form contains following fields:

- **User name** – name of the user account. It must be unique, contain alphanumeric characters only and the Upper/Lower Case must be respected.
- **New Password** – the password to be entered for user login. Dots are displayed instead of characters for safety reasons.
- **Confirm new password** – Since dots are displayed instead of characters, re-enter the password to avoid typing errors.
- **Group** – the rights group. You can create different rights groups for different user groups. To define the range of user rights effectively simply choose a certain rights group. In addition to this, you can "tailor" the safety rules for a specific user, refer to the descriptions of the *Rights* and *Barred rights* fields.
- **Language** – language to which the web interface switches upon user login.
- **Default application** – the initial group bookmark that is active at the instant of login.
- **Image directory** – This setting influences the web interface appearance. You can select various sets of graphic elements from the pop-up menu. The selected set is activated upon user login.

- **CSS style filename** – This setting influences the web interface appearance. You can select various sets of graphic elements from the pop-up menu. The selected set is activated upon user login.
- **Rights** – access rights beyond the selected rights group.
- **Rights denied** – rights that shall be barred to a user although the user should have them because of the selected rights group.
- **Line Number** – states the number of line, under which the internal SIP Proxy server IP telephone may be registered.
- **Description** – any text that describes the meaning of an account.

4.4.3. Authorisation groups

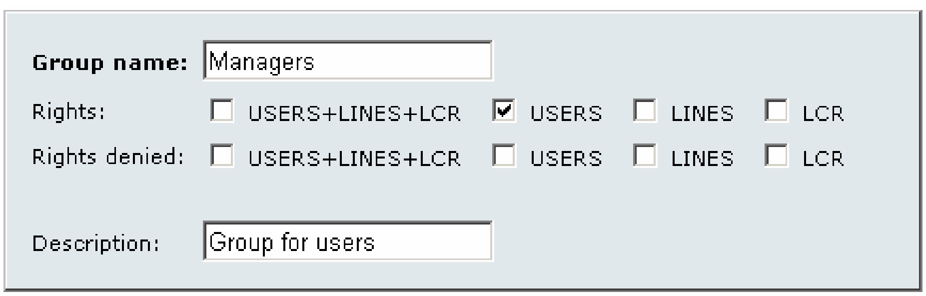
The authorisation groups are created for the reason, that we do not want to specify at each creation of the user account, to which parts of the system the user may have access. The resulted subset of assigned and refused rights is composed during the log-in from the settings of assigned authorisation group and from eventual corrections, which are adjusted during establishment of the user account.
The authorisation groups are defined in the *User administration* item in the *Groups* menu. After clicking to it, the list of created groups is displayed.

Name	Description			
Administrators	Group for administrators			
Managers	Group for users			

Following items are displayed in individual columns of the table:

- **Name** – authorisation group name.
- **Description** – any text documenting the group meaning.

Clicking to the symbol moves us to the authorisation group detail, in which the changes may be done.
Symbol serves for the clearing of the authorisation group. The authorisation groups may be cleared also by ticking of the field in the last column of the table and clicking to the *Remove selected* item.
Under the table, there is also the *Add group*. Clicking to it will display the form for definition of the new authorisation group.



The form contains following fields:

- **Group name** – unique identification.
- **Rights** – parts of system, which are to be made available.
- **Rights denied** – parts of the system, the access to which is to be refused.
- **Description** – any text documenting the group meaning.

The web interface is divided into parts, to which the rights may be assigned or refused. One pair of ticking fields corresponds to each authorisation group in above-described forms for setting of the authorisation groups and user accounts. They are as follows:

- **ALL** – whole system.
- **USERS** – defining of user accounts and groups.
- **LINES** – setting of communication lines.
- **LCR** – configuration of the saving machine.

4.4.4. Emergency change of the administrator password

In emergency cases, when we forget the password of the *Admin* account, some things may be yet saved. We must connect to the gateway by means of the serial cable and in the main menu of the terminal console we must select the item no. 2 – *Set Admin password*. Then, we only set the new password and press Enter.

4.4.5. Switching over of the SIP and H.323 protocols

If you have not purchased the licence for simultaneous usage of the SIP and H.323 protocols, you must then select between the protocols. In the gateway, the H.323 protocol is pre-set.

Switching the protocols is realised by option *Switch to SIP,* or *Switch to H.323* in the following menu. At the switching between the protocols, the whole module will be restarted.

4.5. Setting of communication lines

The parameters of lines are defined in *Configuration* item in the *Equipment* menu. The VoIP lines (SIP and H.323). may be added and modified. The setting of particular type of the lines may be done via appropriate sub-item in the *Equipment* menu.

We must establish at least one SIP or H.323 line to enable proper functioning of the VoIP module *2N*® - OMEGA Lite.

SIP lines

H.323 lines

4.5.1. SIP

On the SIP line setting initial screen, there are displayed all defined SIP lines in the table:

Line ID	SIP server	Phone number	Description			
[12]	10.0.0.1	1111	Moje SIP linka			

Following items are displayed in particular columns of the table:

- **Line ID** – internal identification of the line used by module firmware. The line detail will be displayed after clicking to the number.
- **SIP server** – IP address of SIP proxy, to which the line is registered.
- **Phone number** – line calling number.
- **Description** – any text documenting the line meaning.

Clicking to the symbol moves us to the line setting detail, in which the changes may be done.

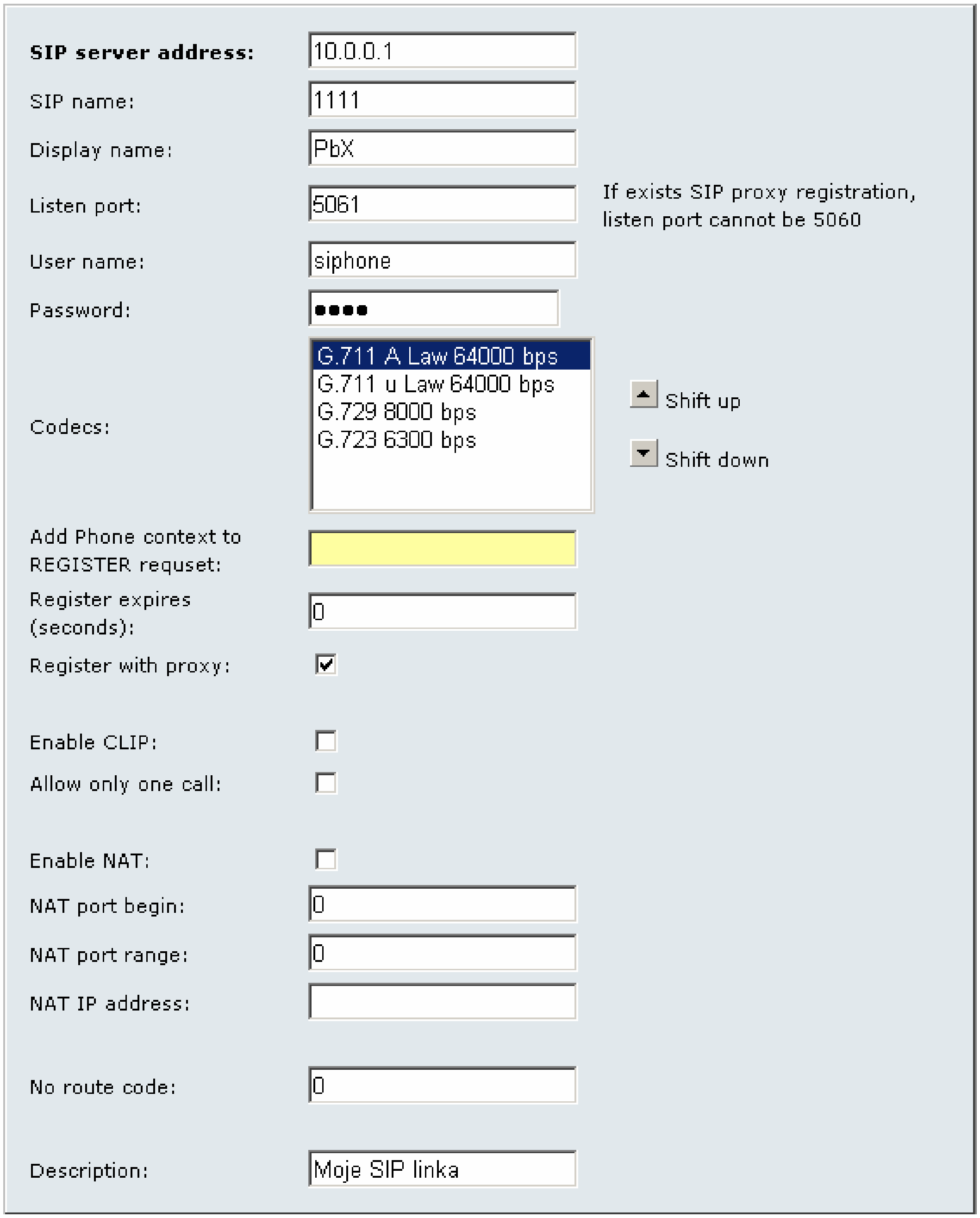
Symbol serves for the clearing of the line. The lines may be cleared also by ticking of the field in the last column of the table and clicking to the *Remove selected* item.

Under the table, there is also the *Add SIP line*. Clicking to it will display the form for definition of the new line.

4.5.2. Parameters of SIP line

If we add the new line, there is item *Add* below the form. During the modification of existing line, there is the item *Change*. Clicking to it, the entered data are confirmed and we request the realisation of relevant operation.

Under the form, there are further the items *Initial* and *Back*. The first one returns the value of all fields to the values before the form opening, and the second one moves to the previous screen.



The form contains following fields:

- **SIP server address** – IP address or fully qualified network name of SIP proxy, to which the line is to be registered.
- **SIP name** – calling line number.
- **Display name** – text, which is to be displayed to the called person.
- **Listen port** – UDP port, on which the line accepts.

- **User name** – name, by which the line is specified during registration to SIP proxy.
- **Password** – password, by which the line is authorised during registration to SIP proxy. Due to safety, the dots are displayed instead of characters.

- **Codecs** – list of codecs, which will be provided by the gateway during negotiation of the voice channel.
- **Description** – any text documenting the line meaning.

4.5.3. H.323

On the initial screen of the H.323 line configuration, there is overview table of defined lines.

Line ID	Call method	Display name	Numbers	Listen port	Description			
[13]	Direct	Omega	1234	1720	H.323 line			

Following items are displayed in particular columns of the table:

- **Line ID** – internal identification of the line used by module firmware. The line detail will be displayed after clicking to the number.
- **Call method** – method, by which the line activate a new calling.
- **Display name** – text, which is to be displayed to the called person.
- **Numbers** – calling line numbers.
- **Listen port** –port, on which the line accepts.
- **Description** – text entered by the administrator, by which we may document the line meaning, if there are more lines.

Clicking to the symbol moves us to the line setting detail, in which the changes may be done.

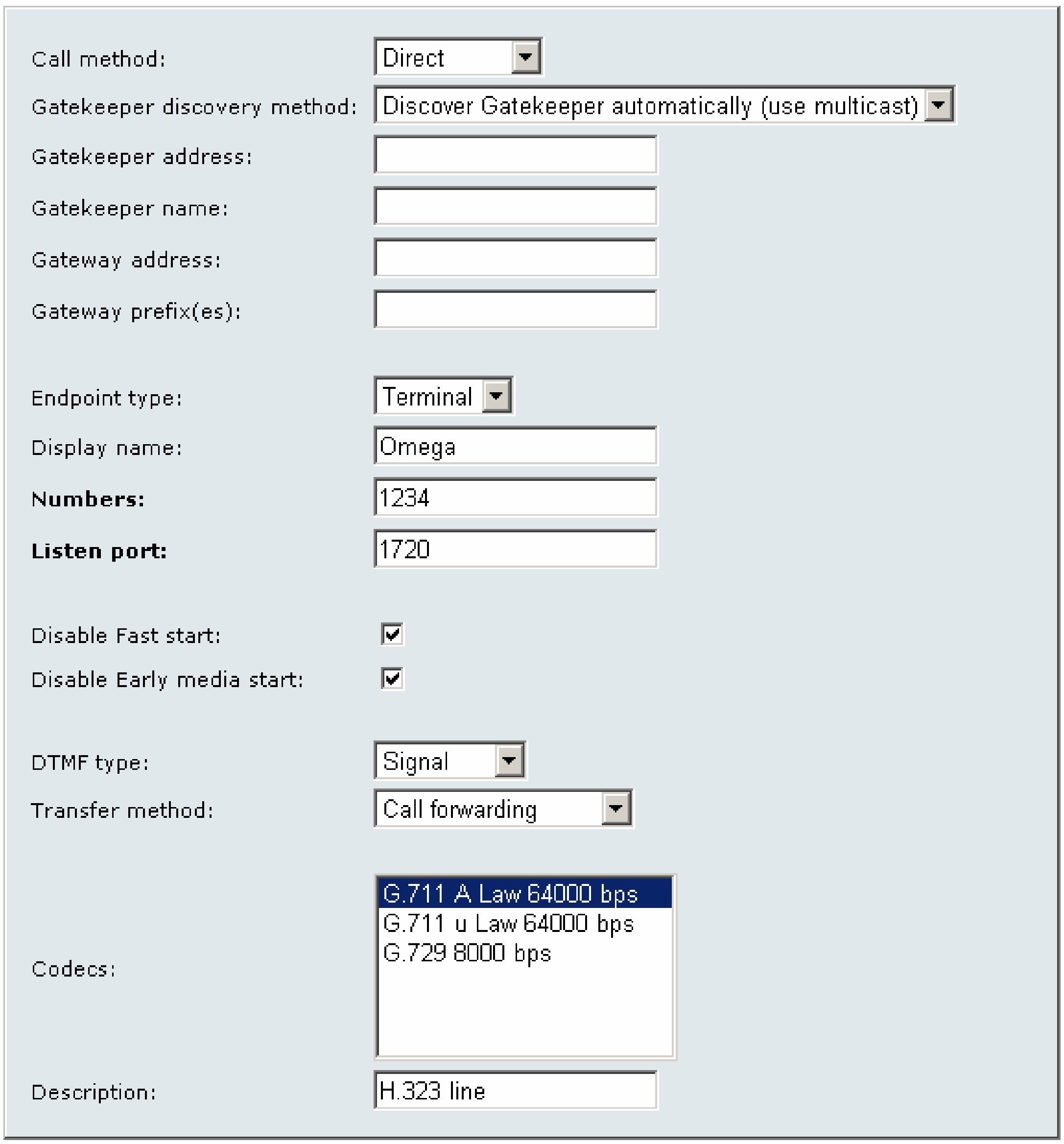
Symbol serves for the clearing of the line. The lines may be cleared also by ticking of the field in the last column of the table and clicking to the *Remove selected* item.

Under the table, there is also the *Add H.323 line*. Clicking to it will display the form for definition of the new line.

4.5.4. Parameters of H.323 line

If we add the new line, there is item *Add* below the form. During the modification of existing line, there is the item *Change*. Clicking to it, the entered data are confirmed and we request the realisation of relevant operation.

Under the form, there are further the items *Initial* and *Back*. The first one returns the value of all fields to the values before the form opening, and the second one moves to the previous screen.



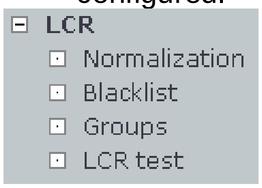
The form contains following fields:

- **Call method** – method of the new calling activation. There are three options: 1) *direct* – calling with specified fully qualified network name of the called person, 2) *gatekeeper* – the search of the relevant path and connection is realised by selected Gatekeeper; 3) *gateway* – calling through selected module.
- **Gatekeeper discovery method** – determines, how the gateway finds the address of available Gatekeeper. The address may be entered as static, or it may be determined by searching in the network.

- **Gatekeeper address** – in case the Gatekeeper address is not searched in the network, it must be entered in this field.
- **Gatekeeper name** – the name of Gatekeeper to which the line is to log in.
- **Gateway address** – the address of the gateway to be used for calling if the *gateway* calling method is selected.
- **Gateway prefix(es)** – the prefixes used by gateway for registration.
- **Endpoint type** – the gateway can register itself to Gatekeeper either as a *gateway* for a prefix (destinations starting by the prefix are routed through the gateway) or as a terminal (virtual telephone).
- **Display name** – the text to be displayed to the called line.
- **Numbers** – numbers of the line working in the *terminal* mode.
- **Listen port** – the TCP port on which the line receives H.323 connections.
- **Disable Fast start** – The *Fast start* method provides faster start of the voice stream while a new H.323 connection is setting up.
- **Disable Early media start** – disallows using of *Early media start* method. This method establishes the voice channel before the call is wholly set up. It's useful for transmission of so called "progress tones" from the GSM network.
- **DTMF type** – choose the set of supported DTMF tones; either numbers only (the *Signal* option) or all of alphanumeric characters (the *Alphanum* option).
- **Transfer method** – calls can be transferred either by the *Call forwarding* method or the *H.452.2 call transfer* method.
- **Codecs** – a list of codecs to be provided by the gateway for voice channel negotiations.
- **Description** – any text that describes the meaning of a line.

4.6. Least Cost Router - LCR

This section deals with the purpose and operation of the gateway cost saving machine - the Least Cost Router. Let us explain how it works and can be configured.



Here is a survey of what you will find in this section:

- Least cost routing process;
- Routing rules – destinations, routes, lines and time intervals;
- Number normalizing;
- List of barred numbers (Blacklist);
- How to test the LCR configuration.

The purpose of the Least Cost Router is to find the optimum output line for the called number. The LCR process has several stages:

Input normalizing: The calling and called line numbers are transformed into a normalized format before entering the LCR.
Destination searching: A destination means the target party to the call. The destination includes a group and type of line searching in the group. The group is searched according to the prefix of the normalized called number.
Group searching: A group means a logical set of lines. The group definition includes a route time limit within a week.
Line searching: A route can consist of one or more lines. The final line is determined according to the selected line selection algorithm. Three ways of searching are available: 1) *the first free line* – finds the first free line, 2) *cycle* – selects the free line with the earliest time of the last call, and 3) *free minutes* – uses the free line with the highest number of remaining free minutes. Correct tariff rates must be selected for the lines for the method to work effectively.
Output normalizing: The calling and called line numbers are transformed into a normalized format before the call is forwarded to the output line.
Check for barred numbers: The *Barred numbers* table is searched for match after output normalizing and before forwarding to the successfully found output line, and, if a match is found, the call is rejected.

By default, the LCR connects all VoIP calls to the VoIP module without modifying the called number.

4.6.1. Routing rules

Routing rules are the core of the LCR system. They consist of a relatively high number of parameters, which make the routing process highly flexible. A guide is available to make your configuration steps as convenient as possible.

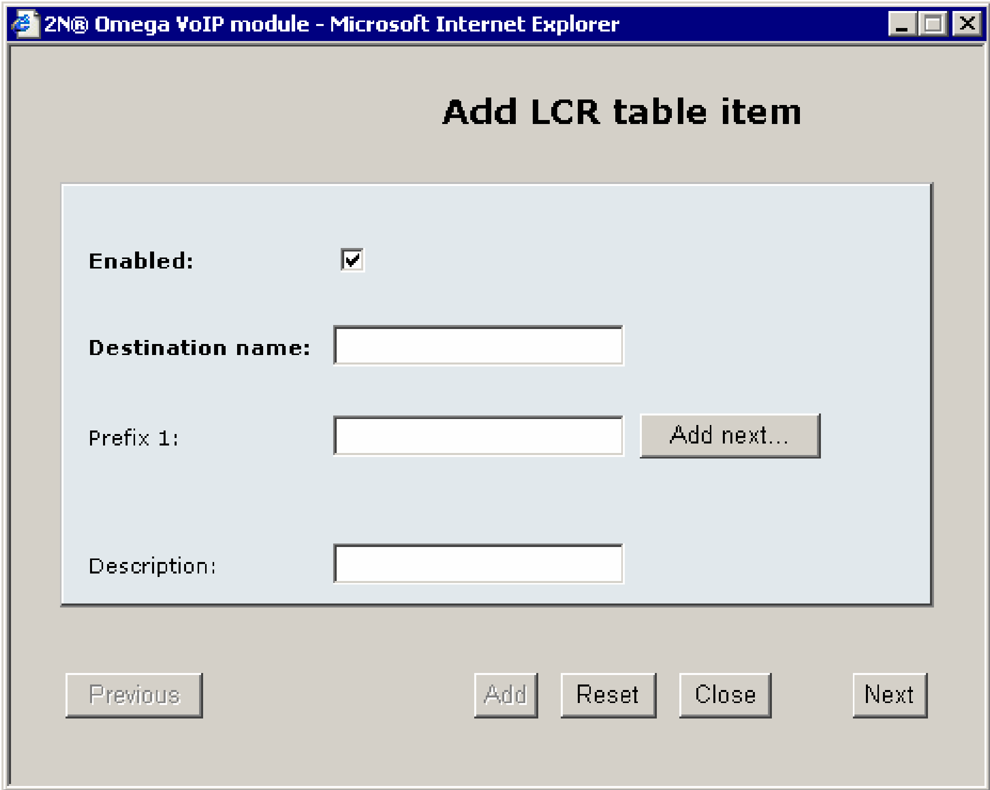
Start with clicking on the *Least Cost Router (LCR)* item in the *Configuration* group tag menu. A table gets displayed with predefined routing rules and buttons that help you change the sequence, add new items, modify and delete existing rules.

Destination		Prefix	Enabled	Groups	Description
TOOMEGA	☐	1	☑	OMEGA	
TOVOIP	☐	2	☑	VOIP	

All the above-mentioned buttons except for *Add* are inactive in the initial status (greyish). They will not become active until one routing rule at least is selected by putting the check-mark in the second column of the respective line. By pressing the button activated in this way you get executed the operation above the selected routing rules.

4.6.2. Add rule

To run the routing rule adding guide click on the *Add* or *Add after* or *Insert before* key. The guide will get displayed in a new window.



The first step is to define the destination. Basically, the destination is a set of prefixes. Looking for the appropriate routing rule, the LCR searches the allowed routing rules from the top to bottom for a match between the normalized called number prefix and the destination number prefix. The first matching prefix stops the searching process and the call is routed according to the respective destination rule.

The form contains the following fields:

- **Enabled** – A routing rule can be defined yet need not be used in the routing process. Enabled rules are used only.
- **Destination name** – The destination must be named briefly and clearly, e.g. according to the mobile provider or any other characteristic feature that differs the destination from the others. The name may contain alphanumerical characters only.
- **Prefix *n*** – the destination prefix where *n* is a serial number. The prefix is a beginning of the target telephone number. If you leave the first prefix blank and do not enter any other, then the particular destination includes all called numbers. To enter more prefixes, add new items by pressing the *Add others* key.
- **Description** – any text describing the meaning of a destination.

To restore the values available at the instant of form opening press the *Reset* button. To close the guide window click on *Close*. To proceed to the second part of creation, press the *Next* key. The *Previous* and *Add* keys are not active in the first step because they cannot initiate a meaningful operation.

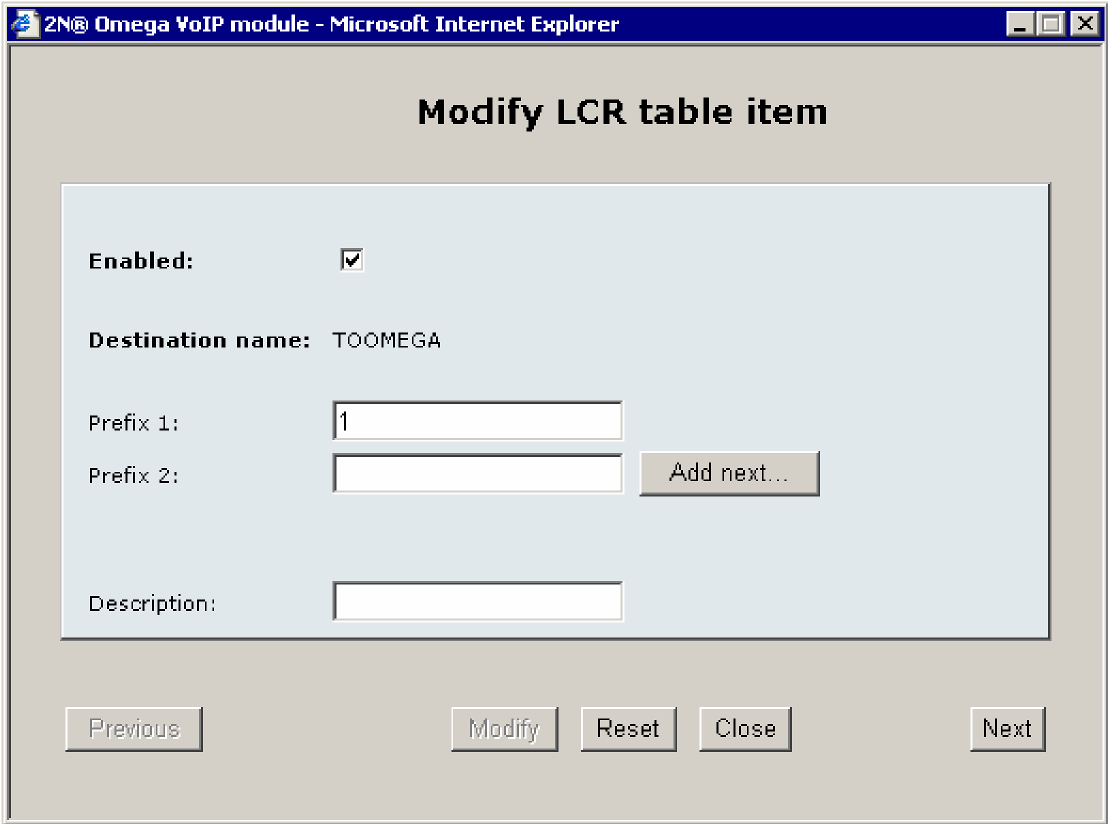
In the second step select the group whose lines are to be used for this rule. To do this, you can either use a list of earlier-defined groups from the pop-up menu or create a new group. Moreover, define the way of selection of lines included in the particular group.

4.6.3. Modify rule

To change a routing rule setting, click on the destination name in the survey of routing rules. This displays a guide similar to that introduced in the text above. The only difference is the presence of the *Modify* key instead of the *Add* key.

Destination		Prefix	Enabled	Groups	Description
TOOMEGA	☑	1	☑	OMEGA	
TOVOIP	☐	2	☑	VOIP	

All routing rule changes influence the gateway behaviour immediately upon execution. Therefore, it is unnecessary to restart the whole equipment.


2N® Omega VoIP module - Microsoft Internet Explorer
Modify LCR table item
Enabled:
Destination name: TOOMEGA
Prefix 1:
1
Prefix 2:
Add next...
Description:
Previous
Modify
Reset
Close
Next

4.6.4. Remove rule

To delete a useless routing rule put the check-mark in the second column of the respective line and press the *Remove* button.

4.6.5. Change sequence

To change the sequence of rules, take steps similar to those for rule removal. Tick off the respective line and click on the *Shift up* or *Shift down* key.

4.6.6. Disable/Enable rule

We already know that only an enabled routing rule can affect the routing process. You can change the status of the *Enabled* field during rule setting and/or setting change. This operation can be performed flexibly in the survey of routing rules too by clicking on the check-mark in the third column of the respective line.

4.7. LCR - Groups

LCR groups are part of the LCR rules as mentioned in the preceding section.

4.7.1. List of groups

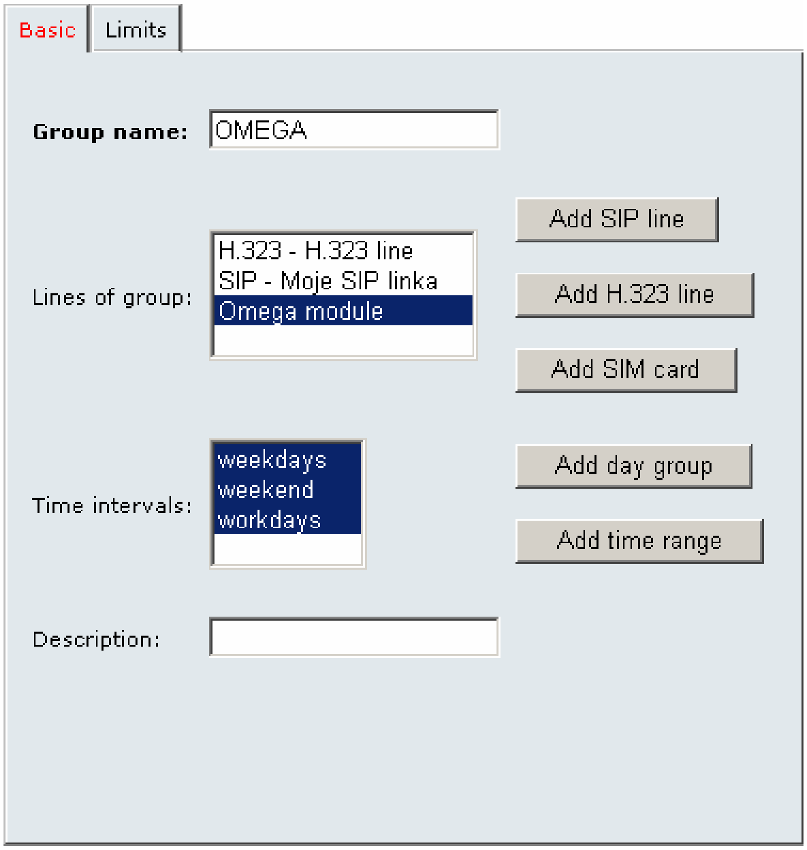
To display a list of groups click on the *Groups* link in the *Least Cost Router* submenu.

Group name	Path lines	Time intervals	Description			
OMEGA		weekdays weekend workdays				
VOIP	SIP - Moje SIP linka	weekdays weekend workdays				

4.7.2. Add group

Click on the *Add group* link to add a group. A form gets displayed in the browser for you to complete the following data:

- **Group name** – The group name may contain any alphanumerical characters. Refer to the LCR rules in the preceding section.
- **Lines of group** – a list of available lines that form a group. To select/remove a line click on the respective item with the left-hand mouse button. You can select more list items at the same time by pressing the Ctrl key along with the mouse click. The selected items are illuminated blue. To create new lines press the *Add SIP line* or *Add H.323 line* key.
- **Time intervals** – define the routing rule time validity. There are three time groups by default: *weekdays* – the whole of Mondays till Fridays, *workdays* – from 7 a.m. to 5 p.m., Mondays till Fridays, *weekend* – Saturdays and Sundays. To create new time intervals click on the *Add day group* or *Add time range* key.
- **Description** – any text that describes the meaning of a route.


Basic
Limits
Group name:
OMEGA
Add SIP line
Lines of group:
H.323 - H.323 line
SIP - Moje SIP linka
Omega module
Add H.323 line
Add SIM card
Time intervals:
weekdays
weekend
workdays
Add day group
Add time range
Description:

4.7.3. Modify group settings

To modify a group setting, click on the pencil symbol of the respective group. This displays a guide similar to that introduced in the text above. The only difference is the presence of the *Modify* key instead of the *Add* key.

4.7.4. Remove group

To delete a group click on the symbol. Or, you can also select the field in the last table column and click on the *Remove selected items* to delete all selected groups at the same time.

4.8. LCR - Number normalizing

When we introduced the LCR operations, we mentioned modifications of numbers before input into and after output from the routing process - so-called normalizing. Normalizing means conditioned transformations of the called and/or calling numbers into a unified format, which facilitates definition of the routing rules. The normalizing regulation is determined by the following three parameters:

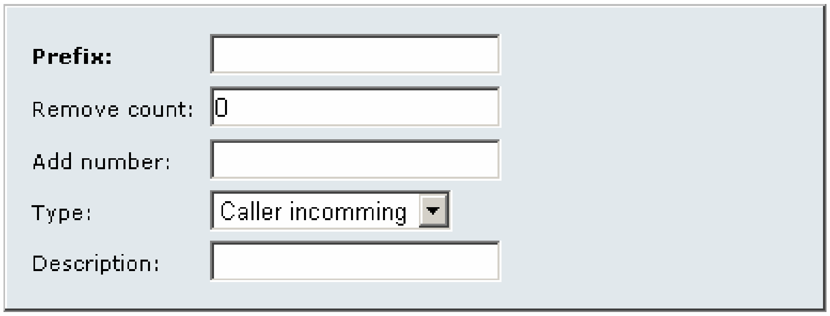
1) **Normalizing type** – defines at which process stage (input/output) and for which numbers (called/calling) normalizing should be used.
2) **Condition** – Transformation is only applied to numbers that meet a condition. The condition is specified by a prefix – a string that the number starts with. The prefix is separated automatically and the remaining part of the number is transformed only.
3) **Transformation regulation** – enables to modify a number by removing a certain number of characters from the number beginning, or adding a new prefix.

To set the normalizing rules use the *Normalizing* item in the *Least Cost Router (LCR)* menu in *Configuration* group tag. The initial display is a list of defined normalizing rules.

By clicking on the pencil symbol you transit into the form and can change the settings.

The cross is used for deleting a normalizing rule. To delete the rules collectively tick off the field in the last table column and click on the *Remove selected items* under the table.

There is an *Add LCR normalizing* link under the table too. By clicking on it you get a new rule defining form.



The meanings of the fields in the form are identical with the following columns of the normalizing table:

- **Prefix** – a prefix that a number must start with to meet the transformation regulation.
- **Remove count** – a count of characters to be removed from the number beginning behind the prefix.
- **Add number** – a prefix to be added before the rest of the number after removal of the prefix and a defined count of digits.

- **Type** – defines at which stage the normalizing rules shall be applied. There are four options: *Incoming calling* – on the calling number input, *Incoming called* – on the called number input, *Outgoing calling* – on the calling number output, *Outgoing called* – on the called number output.
- **Description** – any text that describes the meaning of a normalizing rule.

4.9. LCR - Barred numbers

The *Barred numbers* table is searched for match after output normalizing and before call forwarding to the successfully found output line. If the normalized number starts with a string included in the table, the call is rejected. This means that we can enter both whole telephone numbers and prefixes into the list and thus eliminate, e.g., all international calls or special high-rate calls (erotic lines, etc.).

To select barred numbers use the *Blacklist* item in the *Least Cost Router (LCR)* menu in the *Configuration* group tag. A list of barred numbers gets displayed.

The columns include:

- **Prefix** – the barred number beginning (or the whole number can be entered).
- **Description -** any text that describes the meaning of an item.

By clicking on the pencil symbol you transit into the detail and can change the settings. The above-mentioned two fields are to be completed in the modifying form only.

The cross is used for deleting a barred number. Another way to delete a barred number is to tick off the field in the last table column and click on the *Remove selected items*.

There is an *Add barred number* link under the table too. By clicking on it you get a new number defining form.

4.10. LCR - Test

The *LCR test* item in the *Least Cost Router (LCR)* menu in the *Configuration* group tag is used for testing changes in the LCR settings. Enter the caller's and called numbers and click on the *LCR test* or *Repeat LCR test* link located under the form to initiate the LCR process simulation.

After simulation, the normalized calling and called numbers plus the name of the successfully found output line or the reason why the line search was unsuccessful get displayed.

4.11. SIP proxy

The VoIP module *2N*® - OMEGA Lite is delivered with integrated SIP proxy, which is able to fulfil the function of branch central for the SIP telephones. Its configuration is very easy and it consists only of easy setting of the routing.

If prefix	Strip	Add	Do action	With parameter			
sip:1	0		connect to OMEGA VoIP module	SIP - omega 1			
else	0		lookup registration				

Here is the example of filled routing table, which is displayed after opening of the item *SIP proxy* in the group tag *Services*.

Each line represents one rule. After obtaining of request for assembly of the call, the SIP proxy goes through the table downward and search the rule, according to which it will do further routing. The relevant rule is determined on basis of comparing of the called destination with the value in the first column of the table. If the identification of the called participant starts with the prefix stated in the column *When prefix* of the routing rule, the search terminates and the call is routed according to found rule.

The last column in the table represents so-called initial routing rule. It cannot be cancelled. According to the rule, all calling, for which the explicit routing rule could not be found, are routed.

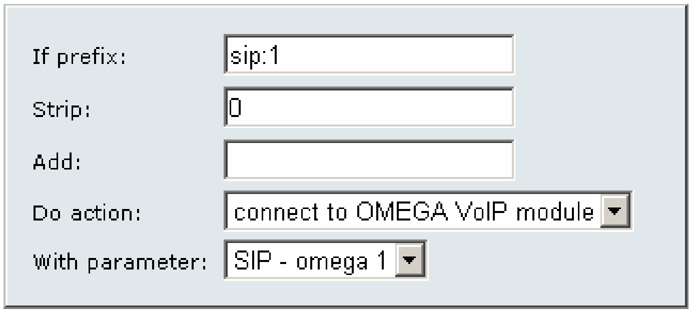
The routing rule determines, what the SIP proxy are to do with the call. The call may be refused, routed to another host and/or port, routed to the VoIP line of the module or search for the called participant in the databases of the registered SIP telephones. Before realisation of one of the mentioned operations, the identification may be modified by subtracting of certain number of characters from the left, eventually by adding of the new string to the beginning of the identification (see columns *Subtract* and *Add*).

Clicking on the symbol moves us to the routing rule detail, in which the changes may be done.

Symbol serves for the clearing of the routing rule. The routing rules may be cleared also by ticking of the field in the last column of the table and clicking to the *Remove selected* item.

Under the table, there is also the *Add rule*. Clicking to it will display the form for definition of the new rule.

On the picture, there is a form in which the setting of parameters of the SIP proxy routing rules may be done. The form fields correspond to the heading of above-displayed table.



The form and the table contain following fields:

- **If prefix** – If the called subscriber's URI (Uniform Resource Identifier) starts with this string, this rule is used for routing. In the SIP environment, the URI is introduced with the "sip:" prefix, which must be included in the value in this field.
- **Strip** – the number of characters following the "sip:" prefix to be removed from the URI before processing.
- **Add** – the string to be inserted in the URI behind the "sip:" prefix.

- **Do action** – what to make with the call. There are six potential actions in the pop-up menu but, in principle, there are only three of them – rejection, forwarding and connecting within the SIP proxy registrations. However, let us mention all options briefly to have the full picture: 1) *Reject* – the called line gets the busy tone; 2) *Overwrite host* – forwards the call to the same port of the selected host; 3) *Overwrite port* – forwards the call to the selected port of the same host (this can have the same effect like option 5); 4) *Overwrite host and port* – forwards the call to any port of the selected host, 5) *Connect to OMEGA VoIP module* – connects the call to the selected gateway SIP line and thus to the LCR, and finally 6) *Search registration* – tries to search the SIP proxy registered users for the required URI and forward the call to the appropriate host.
- **With parameter** – The above actions, except for the first and last ones, require a parameter to be set. This parameter is a new routing destination for call forwarding and the SIP line name for the *Connection to OMEGA VoIP module* action.

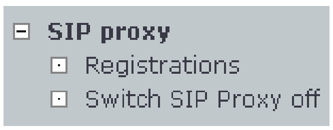
The parameter in the re-routing may be: host address (2) in form of IP address or fully qualified network name, port number (3), or both of them (4), and the host specification is separated from the port number by the character „:“.

4.11.1. Overview of registrations

Via the web interface of the module, it is possible to see, which units are registered to the SIP proxy. This list is available via *Registration* item in the menu *SIP proxy* in tag *Services*.

4.11.2. Switching on/off

If you are using external SIP proxy server and you do not wish to use the services of the integrated SIP proxy server, the integrated proxy server may be switched off.



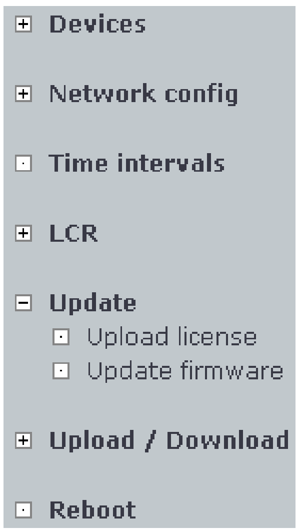
If the integrated SIP proxy server is switched on, it may be switched off by clicking to the item in menu – *Switch SIP Proxy off*.

If the integrated SIP proxy server is switched off, it may be switched on by clicking to the item in menu – *Switch SIP Proxy on.*

4.12. Licence file

The licence file serves for switching on the purchased services providing by the 2N® OMEGA Lite VoIP module. The new purchased module is not functional without the licence file. Before recording, save the licence file on your PC hard-disc and remember the path of its saving.

Recording of the licence file into the module is realised by clicking to the *Update* item in the *Setting* menu, and then to the Add licence item. Then, the dialog for adding of the licence file will display on the screen, as illustrated.



Select the path to the licence file by pressing the *Browse* button and record the licence file into module by pressing the *Add* button.

4.13. Firmware updating

The firmware updating serves for exchange of the firmware of the 2N® OMEGA Lite VoIP module. The updating must be realised exclusively by the firmware delivered by the gateway producer, or by firmware loaded from the web pages of the producer.

The firmware updating serves mainly for recording of repairs of potential failures, which the module may contain, or for recording of new versions of software, which contain new functions.
The file with the firmware is named root.tgz.gpg and should have up to 10 MB. The firmware updating lasts approx. 5-10 minutes, when the firmware is recorded into the module from the local network. If the firmware is updated from the Internet, the delay caused by the speed of your internet connection must be added.

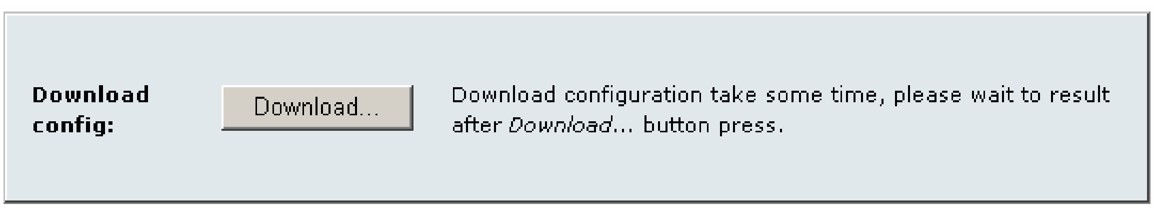
The updating is realised by clicking to the *Update* item and then to the *Firmware update* item. Then, the dialogue for adding of the licence file is displayed, as illustrated

Select the path to the file containing the firmware by pressing the *Browse* button and record the new firmware into the *2N*® OMEGA Lite VoIP module.

4.14. Configuration downloading

Any configuration setting can be downloaded and saved into a file.

To save the configuration click on the *Upload/Download* item in the *Configuration* menu and then on the *Transfer* item to get the configuration saving dialogue.

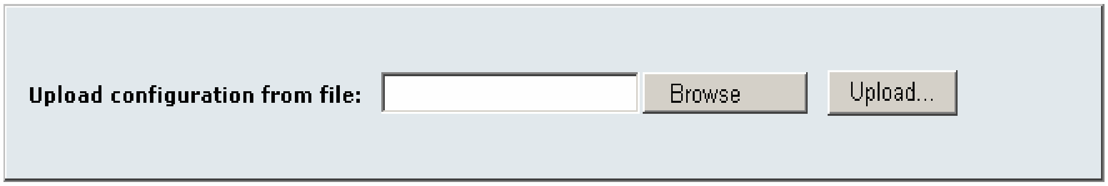


Save the successfully transferred configuration *backup.tgz* to an unambiguous destination.

4.15. Configuration uploading

You can upload the saved configuration setting into the VoIP module.

To do this, click on the *Upload/Download* item in the *Configuration* menu and then on the *Upload* item to get the configuration uploading dialogue.



Find the saved configuration *backup.tgz* and press *Upload.*

5. Configuration of VoIP module parameters in PbX config tool

For correct function of the VoIP module it is necessary, that the *2N®* - OMEGA Lite central is loaded with the firmware of minimal version 4.02REV11. Otherwise the upgrade is necessary.

5.1. Upgrade of firmware to the ATEUS® - OMEGA central unit

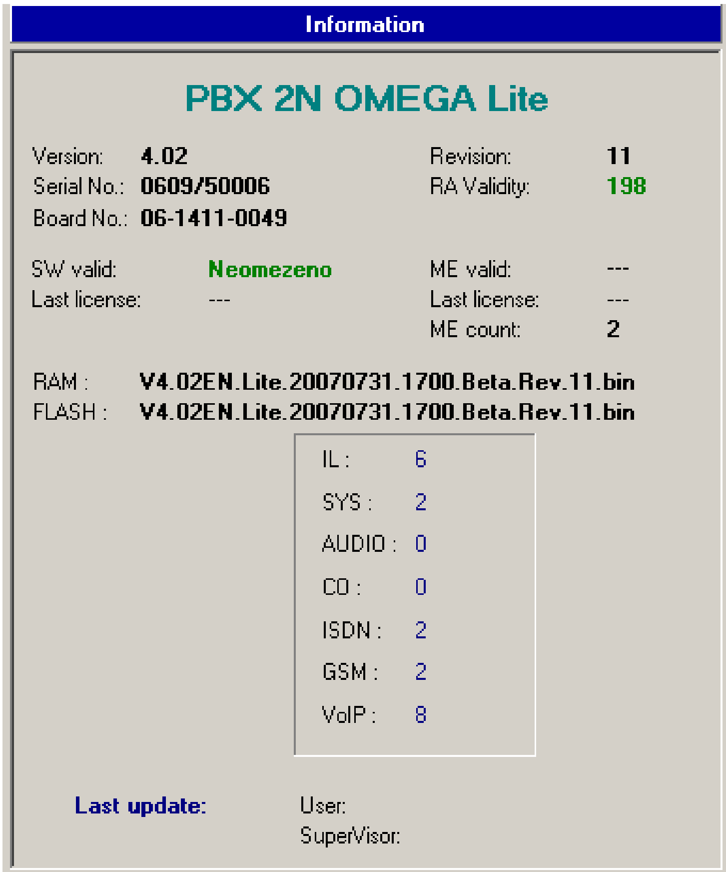
Here is described proper procedure for upgrade of the firmware:

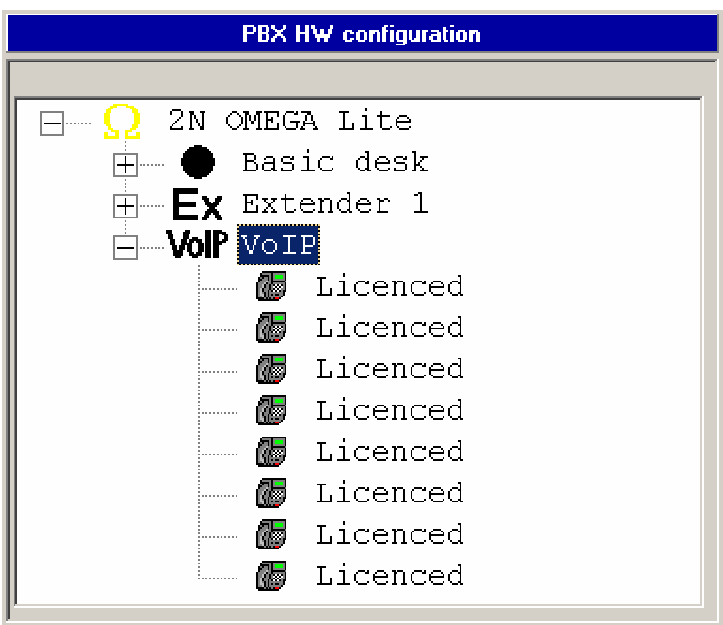
1. Read and save the initial settings of the central unit by the PbX config tool.
2. Read all accounting data (if they are further processed).
3. Terminate all open connections.
4. Use the original configuration tool to import firmware in the **Data – Saving software in the PBX** menu and reset and boot from flash (you will be prompted automatically by the configuration tool).
5. Press both reset buttons on the central motherboard and reset to the manufacturing values (eventually after starting press the SW1 button on the motherboard (left one), until the **Program Data Cleared** is displayed on the central display). **No longer!** (The central would have pass to the service mode - then it is necessary to repeat the procedure).
6. Launch the current configuration tool and create a new configuration (upon upgrade from versions before 4.02, in other cases load the original stored configuration).
7. Check the setting of <u>all</u> parameters of the central.
8. Back-up the configuration to the hard disc.
9. Export data to the central unit.

5.2. Hardware configuration

After the firmware upgrade of the central unit insert and connect the VoIP module (the central unit must be switched off). After the central unit is switched on, load the data and adjust all settings of the module and the central.

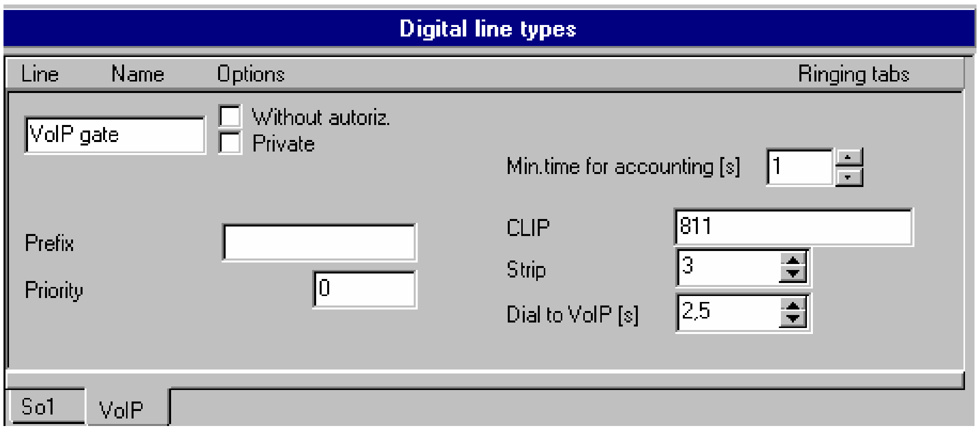
After the data loading, the presence of the VoIP module is signalled in the PbX config tool module *Global data / information* and in module *Global data/Hardware.*


Information
PBX 2N OMEGA Lite
Version: 4.02
Revision: 11
Serial No.: 0609/50006
RA Validity: 198
Board No.: 06-1411-0049
SW valid: Neomezeno
ME valid: ---
Last license: ---
Last license: ---
ME count: 2
RAM : V4.02EN.Lite.20070731.1700.Beta.Rev.11.bin
FLASH : V4.02EN.Lite.20070731.1700.Beta.Rev.11.bin
IL : 6
SYS : 2
AUDIO : 0
CO : 0
ISDN : 2
GSM : 2
VoIP : 8
Last update:
User:
SuperVisor:


PBX HW configuration
2N OMEGA Lite
Basic desk
Ex
Extender 1
VoIP
VoIP
Licenced
Licenced
Licenced
Licenced
Licenced
Licenced
Licenced
Licenced

5.3. Setting of VoIP line

Setting of the VoIP line must be adjusted in module OmegaProgram *External lines/Digital line types/VoIP*



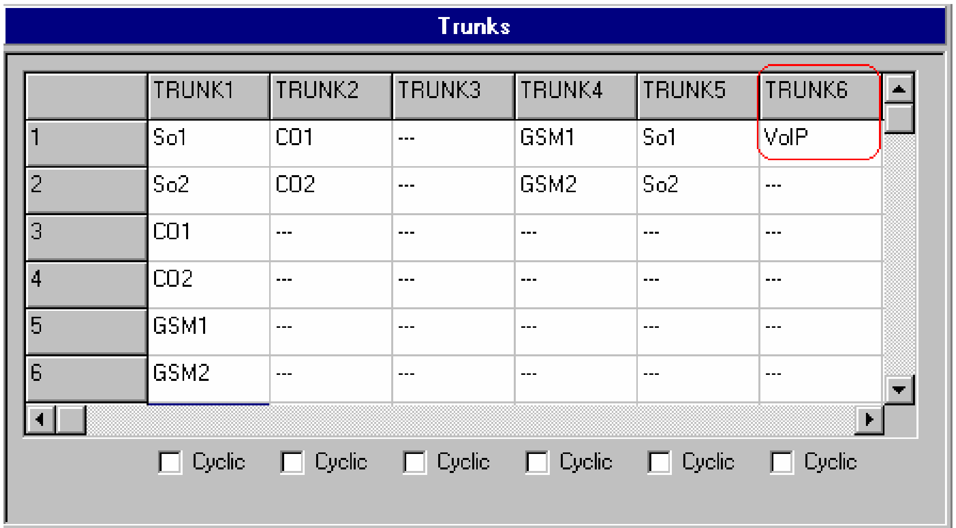
Set:

- **Name –** line name of max. 14 characters
- **Without autoris. –** the authorisation check will not be performed on this line
- **Private –** line connected to the private network, no accounting is performed
- **Prefix –** prefix of up to four digits, which will be automatically called after outgoing occupation of the line as the first (DIN via the superior central). Prefix is not recorded into the accounting line.
- **Minimum time for accounting –** time subtracted from the call time for pseudo-accounting.
- **Priority –** internal line must be of higher priority to use the external line.
- **CLIP** – enter, how should the line be identified in the outgoing direction
- **Strip** – enter number of digits from the incoming identification, which should be subtracted in the input direction in order that the remaining digits are the DIN to the participant of DIN to the ringing table.
- **Dial to VoIP** – maximal period for delay of the next selection into the VoIP line. The time survey is reset after each obtained digit from the internal participant and after its expiring, the dialling mode is changed to the call mode. If it is activated, the expiring of the period and end of the dialling is announced by a short beep. Any other dialling is considered a service into the call.

5.4. Outgoing calling

For outgoing calling, it is necessary to include the VoIP lines into some of the outgoing bundles. During creation of the manufacturing settings, the VoIP are automatically included into the bundle no. 6 and the approach of all lines to the bundle is enabled by the service *Approach to bundle of all external lines* by dialling 86.

Include the VoIP lines into the LCR for comfort calling.


Trunks

	TRUNK1	TRUNK2	TRUNK3	TRUNK4	TRUNK5	TRUNK6
1	So1	CO1	---	GSM1	So1	VoIP
2	So2	CO2	---	GSM2	So2	---
3	CO1	---	---	---	---	---
4	CO2	---	---	---	---	---
5	GSM1	---	---	---	---	---
6	GSM2	---	---	---	---	---
	Cyclic	Cyclic	Cyclic	Cyclic	Cyclic	Cyclic

5.5. Incoming ringing

In the incoming direction, the behaviour of the VoIP module is the same as the behaviour of the ISDN BRI module with DDI pro-selection. It must be specified in the setting, how many digits must be subtracted from the incoming identification, and the residual digits are considered as the pro-selection. The pro-selection then may be the direct one for the particular participant of the central unit or to the ringing tables we already used at the ISDN pro-selection.

The manufacturer reserves the right, in contrast to the submitted documentation, to make modifications to the product that will improve the product's properties.

Please use the product in accord with the instructions and for the purpose for which it was designed and manufactured.
After the product or its components have come to the end of their lifespan please dispose of them in accordance with the valid legal provisions for environmental protection.