

2N VoiceBlue Enterprise

User Manual



Version: 1.11



Dear customer,

*let us congratulate you on having purchased the **2N VoiceBlue Enterprise** system. This new product has been developed and produced in order to provide the maximum utility value, quality and reliability to the user. We hope you will be fully satisfied with the VoIP GSM gateway for a long time.*



- The manufacturer constantly improves the software contained in the product (the so-called firmware). The technology used therein helps you download the latest firmware version to the VoIP GSM gateway using a common PC anytime. For the latest firmware version see www.2n.cz. For necessary instructions refer to Section 13 hereof. We recommend you to apply the latest version to avoid problems that have already been eliminated.
- You also find the latest version of the User Manual at www.2n.cz.
- Check your delivery for completeness according to the packing list and study this manual carefully before installing this product. The manufacturer shall not be responsible for damage caused by any use of this product in contradiction with the User Manual. The warranty terms and conditions do not apply to damage incurred as a result of gross handling and/or undue storing of the product or violation of the technical parameters included herein.
- This manual is very much detailed and includes subsections that are irrelevant for the basic installation purposes as well as subsections referring to other VoIP GSM gateway models. Therefore, pay attention to hints informing you which subsections are necessary for you and which are not.

■ Packing List

Please check your **2N VoiceBlue Enterprise** delivery for compliance with the following packing list.

Item	Pieces
VoIP GSM gateway – check the model type according to the order number, see the type label on the gateway back side	1
12V/2A mains adapter	1
Serial cable	1
Antenna	1
Wall mounting holder	1
Dowels	2
Screws	2
Quick user guide	1
Warranty certificate	1
Compliance certificate	1
2N product CD	1

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SECTION 1

Product Presentation

This section presents the 2N VoiceBlue Enterprise product including its application options and advantages.

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

- Purpose of the product;
- Application possibilities;
- Instructions how to save telephone costs;
- Overview of interesting and useful properties.

1.1. Purpose

- The 2N VoiceBlue Enterprise gateways provide direct interconnection of VoIP and GSM networks with the aid of SIP signalling protocols.
- The voice mode, i.e. an outgoing or incoming call, is the basic function of the system. The gateway is equipped with all functions necessary for such use and provides a very high comfort in this mode.
- In addition to voice transmission, 2N VoiceBlue Enterprise can send and receive short text messages. For remote control purposes, diagnostic SMS can be sent in regular intervals from the gateway to a selected mobile number. These additional functions enhance the utility value of the product.
- 2N VoiceBlue Enterprise contains a built-in SIP proxy server and can thus serve as a PBX for SIP telephones. However, the use of the gateway is not limited to this SIP proxy function only, it can be applied with any SIP supporting PBX.
- No extra equipment (an external GSM telephone, etc.) is needed for normal operation. All programmable parameters are default-preset in such a manner that you can commence your telephone traffic the moment you connect the Ethernet and supply cables, antenna and SIM card and set the IP parameters.

1.2. How to Save Telephone Costs

- Once your 2N VoiceBlue Enterprise gateway has been connected to your VoIP PBX, all calls going out to a mobile network are made directly. **This saves your PSTN - mobile network call costs.** The mobile telephone calls of your personnel in the field are cheaper too.
- You are advised to use the **most advantageous tariff of your GSM provider** for your GSM gateway because all gateway user call accounts are added up for billing purposes.
- If you use a **call answering machine** as a GSM network service, you pay for your voice message deliveries as a rule. However, if you connect your answering machine to the GSM gateway, **you pay nothing for your voice message deliveries.**

- You can bar selected numbers or groups of numbers in your gateway. **You shall pay nothing for the calls you have barred.**
- 2N VoiceBlue Enterprise keeps detailed records on all calls. This helps you **find out easily why your bill is higher than it should be.**
- The Least Cost Router is flexible enough to help you set rules for **GSM calling at the lowest possible operation costs.**

1.3. Other Advantages and Applications

- VoIP-GSM gateways integrate the best of their respective communication technologies.
- With the aid of a built-in SIP proxy server you can create a **complete VoIP network without additional** third party products' costs.
- The **intelligent routing of incoming calls** accelerates the connection of incoming calls and makes calling more comfortable.
- An easily recordable DISA voice message function is available.
- You can use the conditioned or unconditioned call forwarding function.
- Unlike mobile telephones, this system does not expose you to the RF electromagnetic field while making calls.

1.4. RF Radiation Safety Precautions



- ! It is prohibited to use any transmitters, including GSM gateways, in areas where explosives are used, such as quarries.
- ! It is forbidden to use mobile phones and thus GSM gateways too at petrol stations.
- ! A GSM gateway may affect sensitive life-saving devices in medical centres. So it is prohibited to use mobile phones and GSM gateways here.
- ! In general, any prohibition regarding mobile phones based on RF energy radiation applies to GSM gateways.
- ! Where necessary, a GSM gateway may be installed at a safe distance (in the neighbouring building, e.g.) and an Ethernet cable may be carried from the GSM gateway to the original building.
- ! Although GSM gateways are not intended for aeroplanes or cars, all relevant prohibitions and regulations regarding mobile phones apply to them here.

2

SECTION 2

Installation

This section shows how to install the 2N VoiceBlue Enterprise gateway including all accessories.

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

- What you should know before you start;
- Overview of installation steps;
- How to choose the right place;
- Connection to PC network;
- Antenna connection;
- Supply mains adapter connection;
- SIM card replacement;
- PC connection with a serial cable;
- How to read LED statuses.

2.1. Before Installation

Before you start installing your VoiceBlue Enterprise gateway, get familiar with its physical structure, arrangement of connectors and status indicators, see fig. 1 and fig. 2.

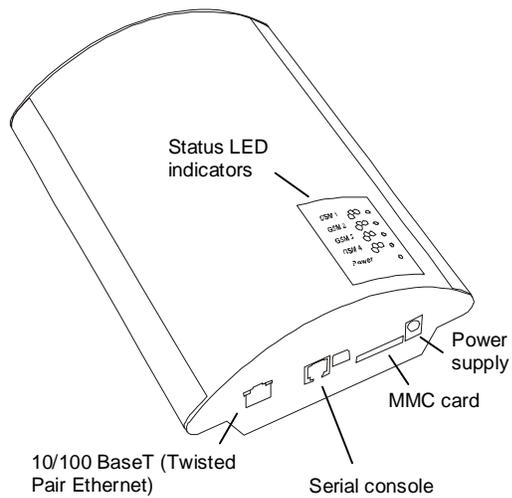


Fig. 1 – Bottom View

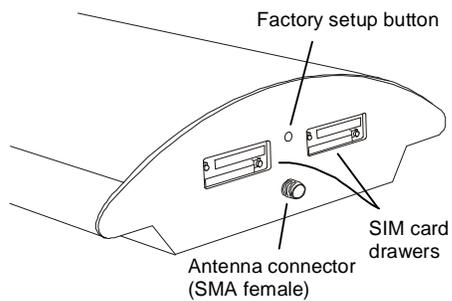


Fig. 2 – Top View

2.2. Brief Installation Guide

- **Proper mounting** – 2N VoiceBlue Enterprise is designed for suspension on a vertical surface. Fit the holder included in the delivery on a wall and hang the gateway on it. For details on the prescribed working position and other recommendations refer to Subsection 2.3.
- **Cable connection** – connect the gateway using a patch-cable to the LAN (Local Area Network). For details on proper wiring refer to Subsection 2.4.
- **Antenna connection** – connect an internal antenna or an external antenna cable to the SMA antenna connector. Place the external antenna on a place with a good GSM signal (refer to Subsection 2.5).
- **Gateway power supply** – the delivery includes a mains adapter. Plug in the adapter connector into the gateway and the power adapter into a mains socket ($\approx 230\text{V}/50\text{Hz}$). The gateway turns on immediately (see Subsection 2.6).
- **SIM card insertion** – SIM cards are inserted in holders on the gateway top. To slide the drawer out of the gateway press the yellow button on the drawer with a suitable tool (pin, pencil, ...), see Subsection 2.7.
- **PC connection and initial configuration** – a web browser is used for gateway configuration in normal operation. If you have not defined the IP address for the gateway, use the serial cable included in the delivery (refer to Subsection 2.8) for initial configuration of the basic parameters (see Section 3).
- **Setting and monitoring** – Having set the basic network parameters (IP address, etc.), you can configure the VoIP gateway using a standard web browser. Configuration options and essential procedures shall be discussed in detail in respective sections.

2.3. Proper Mounting

- The gateway is designed for mounting on a vertical surface. For this purpose a wall mounting holder is available. Just fit the holder with dowels and screws (fig. 3) to the wall and hang the gateway as shown in fig. 4.

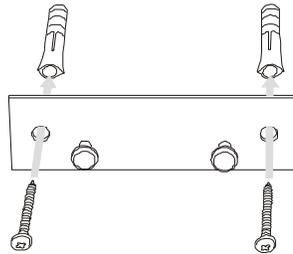


Fig. 3 – Gateway Holder Wall-Mounting

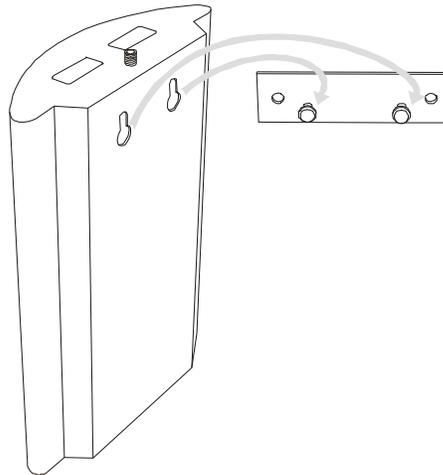


Fig. 4 – Gateway Holder-Hanging

- It is possible to operate the VoIP GSM gateway in another working position (e.g. on a desk) for a short time only, for example in servicing centres for quick testing purposes.

The allowed working temperature and relative humidity ranges are included in Section 14.

- Exceeding the recommended operating temperature values need not affect the gateway function immediately but may result in more rapid ageing and lower reliability.
- The gateway is intended for indoor use. It may not be exposed to rain, flowing water, condensed moisture, fog, or mist.
- The gateway may not be exposed to aggressive gas, acid vapours, solvents, etc. or aggressive liquids, during cover cleaning, for example.
- The GSM gateway is not designed for high-vibration environments, e.g. means of transport, machine rooms, etc.
- Free space has to be left under and over the gateway for cables and agitated air to remove operational heat.
- Install the gateway on a place with a good GSM signal.
- A misplacement of the GSM gateway or its antenna near television, broadcasting or similar RF-sensitive devices may exercise an adverse effect upon their function.
- Being a source of RF energy emission, the gateway antenna should not be located close to human bodies. The hazard is higher than with mobile telephones because the gateway is usually used by many people and thus employed more often.

2.4. LAN Connection

2N VoiceBlue Enterprise is connected to the 10/100BASE-T (Twisted Pair Ethernet) LAN using a standard straight (non-crossed) cable terminated with RJ-45 connectors like other types of equipment (PC, VoIP terminals, etc.).

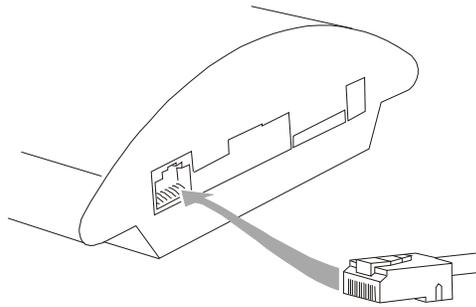


Fig. 5 – LAN Connection

2.5. Antenna Connection

No matter if equipped with two or four GSM modules, the VoIP gateway has just one SMA antenna connector, see fig. 6, to which an external antenna cable is connected. This external antenna should be installed vertically on a place with a good GSM signal.

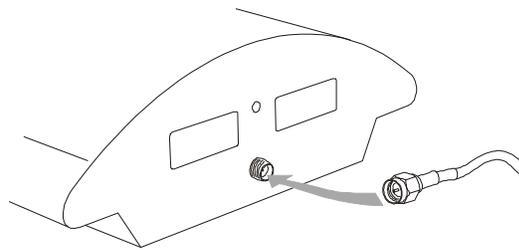


Fig. 6 – Antenna Connector Connection

Tighten the antenna connector **gently** with your hand, never use a wrench!

2.6. Gateway Power Supply

- Be sure to use only the mains supply adapter that is included in the GSM gateway delivery.
- Before plugging in the gateway, make sure that the mains voltage value meets the data given on the mains adapter label.
- Furthermore, make sure that the antenna has been connected. If you connect the gateway to the power supply without an antenna, the GSM module transmitter might get damaged.
- Now plug the supply adapter into a mains socket and connect the adapter connector to the gateway, see fig. 7. The status indicators indicate the proper operation. For their meanings refer to Subsection 2.9.

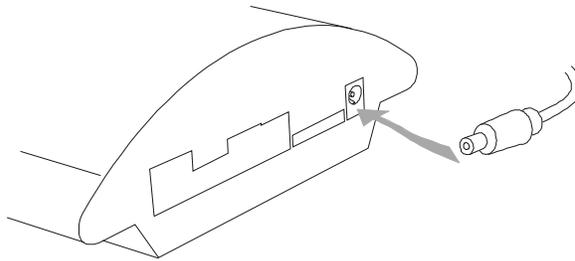


Fig. 7 – Supply Adapter Connection

2.7. SIM Card Insertion

To insert your SIM card in the gateway, press the yellow button on the appropriate card drawer with a suitable tool to make the drawer slide out a little. Remove the drawer, insert the SIM card and replace the drawer. You can replace SIM cards even with your gateway on.

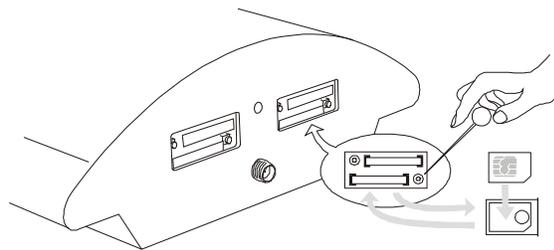


Fig. 8 – SIM Card Inserting Procedure

2.8. PC Connection

You need direct interconnection of your 2N VoiceBlue Enterprise gateway and PC especially when you do not know the gateway IP address and thus cannot configure it using a web browser. In such a case, connect the gateway directly to your PC serial port with the cable enclosed and select the basic network parameters using your console interface.

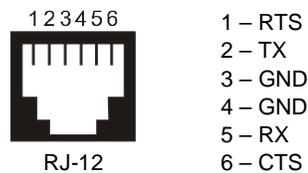


Fig. 9 – Serial Console Pin Connection

You can control the console interface with any terminal-emulating program according to the VT-100 specification at least. The Microsoft Windows OS, e.g., contains a communication tool *HyperTerminal*, which can be used for this purpose.

By default, the gateway serial port communication parameters are set at 115,200 bits per second, 8 data bits, no parity, 1 stop bit, no flow control. Set the communication program on the PC side identically.

2.9. Status Indicators

There is a panel with five LEDs on the gateway upper cover. The *Power* indicator signals that the gateway as a whole is in operation. The other LEDs, marked *GSM 1* to *GSM 4*, indicate the status of each GSM module.

Power Indicator

Basic diagnostic tests are performed automatically whenever the gateway is connected to supply voltage. Each test step is signalled by a specific colour combination of the LEDs. The testing process is usually shorter than 0.5 sec and so it looks like a quick sequence of LED flashes at first sight. If, however, a test step fails, the indicator combination related to the failed test remains lighted. This provides a convenient troubleshooting tool to the technical support personnel.

After all hardware tests have been completed successfully, the operating system kernel is loaded. During the process, all GSM module indicators are off and the blue continuous *Power* light blinks with the period of approximately 1 second.

When the operating system kernel has been loaded, the blue *Power* indicator keeps shining continuously.

In exceptional cases, when a fatal failure of software occurs and the gateway stops working, the *Power* indicator starts flashing quickly. All you can do is disconnect the gateway from the power supply for a few seconds and restart it.

Power Indicator (blue)	
No light	The system is not working.
Blinking	The system kernel is being loaded.
Continuous light	The system is working.
Quick flashing	"Kernel panic" – fatal failure

GSM Indicators

The *GSM 1* to *GSM 4* indicators signal the status of the respective GSM module. After connection to the power supply, the VoIP gateway application starts within two minutes. During that time, the installed GSM modules and SIM cards are detected. If a GSM module or SIM card is not detected, the respective indicator remains off.

If they are detected, the process of module and SIM card initialisation is commenced. The respective indicator shines red during the process. If the process is unsuccessful, the LED goes off. If the indicator turns orange, it means that the module has not logged-in to any GSM network.

If the LED is green, you can make calls and send SMS via the respective module.

The respective module LED blinks green quickly during call establishing. When the connection has been built successfully, the indicator remains green during the whole call.

GSM Indicators	
No light	The module is ready
Red light	SIM card initialisation in progress or no SIM card is in the drawer.
Yellow light	The SIM is not logged-in to any GSM network.
Quick green blinking	A call is being established (dialled).
Green light	A call is being made.

SECTION 3

Initial Setup

This section shows you how to control tools that you will need later for gateway configuring.

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

- Default communication parameters;
- Basic setup via a serial link;
- Description of the web configuration interface arrangement.

3.1. Before You Start Configuring...

The following values are set for the 2N VoiceBlue Enterprise gateways by the manufacturer:

Serial console communication port:

Transmission rate: 115200

Data bits: 8

Parity: none

Flow control: none

Network parameters:

IP address: 10.0.0.1

Network mask: 255.0.0.0

Having unpacked the gateway, you can configure it using any of the following two ways:

- 1) Connect it to your PC via a serial link (refer to Subsection 2.8) and set the basic network parameters using a terminal communication program, e.g. HyperTerminal. Then connect yourself to the gateway through a web browser to complete the setting procedure.
- 2) Interconnect the gateway with your PC using a crossed Ethernet cable, set an arbitrary IP address for the PC from the network 10.x.y.z, other than 10.0.0.1, and set the required IP address for the gateway using a web browser. Having changed the IP address, connect both the PC and gateway to the LAN in a standard way and complete the configuration process using your web browser.

Let us discuss the two ways in the subsections below.

3.2. Console Access

The console system is arranged as a set of nested menus. By selecting a menu item you either get into a submenu, or have the required operation executed, or set the selected parameter.

3.2.1 Serial console access

When the gateway is powered on, the main menu should get displayed after terminal connection.

```

VoiceBlue Enterprise V1.01           Main Menu           VoiceBlue

      Option           Value           Description

1 - Configuration      [ menu ] - General configuration
2 - Set Admin password          - Set administration password
3 - Help                - Display help for serial console settings

Enter an option number, <ESC> previous menu
>

```

Every menu consists of the following parts:

- **Header:** Contains (from the left): the product name, firmware version, menu name and network name assigned to the equipment.
- **Options column:** Displays numbers and names of available options.
- **Value column:** If "[menu]" is displayed, the item includes a submenu. Otherwise it shows the actual value of the option.
- **Description column:** Explains briefly the meaning of every option in the menu.
- **„Enter an option number...>“:** Here enter the option number.

To select an option from a menu, enter its first number and press for confirmation.

Submenus

If you select a submenu, this submenu gets displayed. Now you can select items from the selected submenu or return by pressing



Commands and Values

If you are requested to enter some information after selecting an item, you should enter one of the following values:

- **Key word:** A list of one or more fixed strings. To select one, enter only so many initial characters as can differ it unambiguously from other key words.

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

In the above mentioned case, select a key word by entering just an "a", "c", or "t" and press .

- **String:** an arbitrary number of characters. The acceptable value interval may be included in the prompt.

Enter a hostname from 1 to 32 characters: voiceblue

- **Integer:** A decadic integral number. The acceptable value interval may be included in the prompt.

Enter a size between 1 and 100 : 99

A hexadecimal integral number – number to be entered in the hexadecimal format using characters 0÷9 and a÷f or A÷F.

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

- **MAC address:** up to 12 hexadecimal digits of the physical address. Zeros at the beginning can be omitted.

Enter the remote network address : 50C229C4E2

- **IP address:** an Internet address in the format of four digits of 0÷255 separated with dots.

Enter an IP address : 192.168.22.30

As soon as the requested information is entered, the respective operation is executed. If the operation changes the parameter settings, new values are displayed in the redrawn menu.

Some configuration parameters may have just one of two constant values. By selecting such an item you immediately make its value opposite to that preceding the selection. Typical examples are *on* and *off* parameters. If their value is *on*, then it becomes *off* when the item is selected and vice versa.

Some commands execute operations that have a serious impact on the behaviour of the entire system (such as restart). Therefore, the system usually "makes sure" before execution whether it is really your intention.

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

If you give an answer other than "y" or "Y", the command will not be executed.

To cancel the command execution and quit the prompt press the key any time.

About Menus...

We shall discuss the serial console menus in detail in sections devoted to gateway parameter settings. Below is a brief survey of these menus only.

Main menu: Appears after the serial cable is connected to the terminal. From the main menu you can get into the configuration menu, change the administrator password, or display the help.

Configuration menu: Contains two submenus: network configuration and serial console settings.

Network configuration: Used for setting the gateway IP address, network mask, initial router, DNS addresses, network name and domain.

Serial console configuration: Helps set the terminal type and change the initial serial link parameters – transition rate, data and stop bits, parity and flow control type.

DTR Signal Monitoring

The VoiceBlue Enterprise software monitors continuously the Data Terminal Ready (DTR) signal. This signal is used for detecting the presence or absence of terminal connection to the serial console port.

When terminal disconnection is detected, the following operations are performed:

- All initiated operations are cancelled.
- The main menu gets displayed.

If the connecting cable does not interconnect the DTR signal, you cannot work with the serial console!

3.2.2. Telnet console access

This is a short description of new features added to the serial console configuration. For the standard serial console configuration capabilities refer to the 2N VoiceBlue Enterprise

There is also a possibility of using Telnet for 2N VoiceBlue Enterprise configuration in addition to the standard serial console configuration. Connect to your VoiceBlue via Telnet (type telnet <your VoiceBlue IP address> in command line). If successful, VoiceBlue will require an authentication from you. To log in, use valid login and password of one of VoiceBlue's users who is in Administrators group. No user other than user from Administrators group will be logged in. If login and password entered correctly, you will be logged in.

Now you will see the window similar to the standard serial console configuration window and you can work with it in the same way. To log out Telnet configuration, get to the main menu and choose Logout option (this option is not visible in the serial console window).

There are two new menus added to the Configuration menu - menu States and menu Command line. They allow you to control and configure VoiceBlue devices, calls, lines and routes.

3.2.2.1 Menu Command Line

Menu Command line allows you to control and configure VoiceBlue lines, routes and LCR. Because of its complexity, this menu is implemented as a command shell. To return back from this menu type exit (this is the difference to the other menus which you can exit by pressing the Esc key - pressing the Esc key in this menu has no effect). To obtain a list of all the implemented commands type help. Short help for each command can be obtained by typing <command> help.

There are seven commands implemented - calls, devices, disa, h323, lcr, route and sip. They will be discussed in the following (in an alphabetical order).

3.2.2.1.1 Command 'calls'

Command calls prints a list of all active VoiceBlue calls. This command performs the same action as submenu Calls in States menu (see Section 3.2). Printed list has the same form as a list obtained in the Web Browser menu Devices.

3.2.2.1.2 Command 'devices'

Command devices prints a list of all present VoiceBlue devices. This command performs the same action as submenu Devices in States menu (see Section 3.2). Printed list has the same form as a list obtained in the Web Browser menu Devices.

3.2.2.1.3 Command 'disa'

Command disa allows you to set, modify and delete DISA lines. There are following parameters to this command:

help - prints short help

list - prints a list of all DISA lines

ls - same as a previous command

add - adds DISA line to the list, there are other parameters needed, none of them required

type=basic|uploaded - specifies Type of DISA

desc=<description> - specifies Description

replace linenr=<line number> - replaces DISA line with <line number> specified. If

DISA line list is empty, this command is not allowed, same parameters as for previous command are possible, if some parameter is omitted when replacing, original value will be used

del <line number> - deletes DISA line with <line number> specified

Example: There is one Basic DISA line in a list. User decides to add new one with no parameters specified (all parameters will be set to default values). Then he decides to change Type of DISA to Basic DISA with uploaded message. Command line shell, type help for more information or type exit to quit

```
>disa ls
```

```
linenr=26 type=basic desc="first disa"
```

```
>
```

```
>disa add
```

```
No parameters identified, default values will be set
```

```
Added successfully
```

```
>
```

```
>disa ls
```

```
linenr=26 type=basic desc="first disa"
```

```
linenr=27 type=basic desc=""
```

```
>
```

```
>disa replace linenr=27 type=uploaded
```

```
Replaced successfully
```

```
>
```

```
>disa ls
```

```
linenr=26 type=basic desc="first disa"
```

```
linenr=27 type=uploaded desc=""
```

```
>
```

3.2.2.1.4 Command 'h323'

Command h323 allows you to set, modify and delete H.323 lines. There are following parameters to this command:

help - prints short help

list - prints a list of all H.323 lines

ls - same as a previous command

add - adds H.323 line to the list, there are other parameters needed, only numbers parameter is required (parameters can be written in an arbitrary order).

numbers=<numbers> - specifies Numbers, required

c_meth=GK|GW|D - specifies Call method, GK for gatekeeper, GW for gateway, D for direct

d_meth=AUTO|ADDR - specifies Gatekeeper discovery method, AUTO for Discover automatically,

ADDR for Use gatekeeper address

gk_addr=<gatekeeper address> - specifies Gatekeeper address

gk_name=<gatekeeper name> - specifies Gatekeeper name

gw_addr=<gateway address> - specifies Gateway address

prefixes=<prefixes> - specifies Prefixes

endpt=GW|T - specifies Endpoint type, GW for gateway, T for terminal

disp_name=<display name> - specifies Display name

listen_port=<port num> - specifies Listen port number

F_S_EN|F_S_DIS - disables or enables Fast start

E_S_EN|E_S_DIS - disables or enables Early media start

T_EN|T_DIS - disables or enables H.245 tunneling

dtmf=SIG|AN - specifies DTMF type, SIG for Signal, AN for Alphanum

t_meth=CF|CT - specifies Transfer method, CF for Call forwarding, CT for H.450.2 call transfer

codecs= - specifies Codecs, valid values are g711alaw, g711ulaw, g729, separated by semicolon, at most three codecs are allowed

rej_calls=<rejected calls message number> - specifies reference number of Rejected calls code message (from 0 to 26, see a list of all allowed messages and their reference numbers)

no_route=<no route message number> - specifies reference number of No route

code message (from 0 to 26, see a list of all allowed messages and their reference numbers)

desc=<description> - specifies Description

replace linenr=<line number> - replaces H.323 line with a <line number> specified.

If H.323 lines list is empty, this command is not allowed, same parameters as for a previous command are possible, if some parameter is omitted when replacing, original value will be used

del <linenr> - deletes H.323 line with a <linenr> specified

Example: There is one H.323 line in a list. User decides to add a new line with the only required parameter Numbers set to value 1234 (other will be set to the default values). Then he decides to change Display name to value "my name", Description to value "new h323 line" and to disable Fast start. Command line shell, type help for more information or type exit to quit

```
>h323 ls
```

```
linenr=19 c_meth=D d_meth=AUTO gk_addr= gk_name=""  
gw_addr= prefixes= endpt=T di
```

```
sp_name= numbers=1222 listen_port=1720 dtmf=SIG t_meth=CF  
codecs=g711alaw rej_ca
```

```
lls=0 no_route=0 desc=""
```

```
>
```

```
>h323 add numbers=1234
```

```
Added successfully
```

```
>
```

```
>h323 ls
```

```
linenr=19 c_meth=D d_meth=AUTO gk_addr= gk_name=""
gw_addr= prefixes= endpt=T di
sp_name= numbers=1222 listen_port=1720 dtmf=SIG t_meth=CF
codecs=g711alaw rej_ca
lls=0 no_route=0 desc=""

linenr=28 c_meth=GK d_meth=AUTO gk_addr= gk_name=""
gw_addr= prefixes= endpt=T d
isp_name= numbers=1234 listen_port=1720 dtmf=SIG t_meth=CF
codecs= rej_calls=0 n
o_route=0 desc=""
>
>h323 replace linenr=28 disp_name="my name" desc="new h323
line" F_S_DIS
Replaced successfully
>
>h323 ls
linenr=19 c_meth=D d_meth=AUTO gk_addr= gk_name=""
gw_addr= prefixes= endpt=T di
sp_name= numbers=1222 listen_port=1720 dtmf=SIG t_meth=CF
codecs=g711alaw rej_ca
lls=0 no_route=0 desc=""
linenr=28 c_meth=GK d_meth=AUTO gk_addr= gk_name=""
gw_addr= prefixes= endpt=T d
isp_name=my name numbers=1234 listen_port=1720 F_S_DIS
dtmf=SIG t_meth=CF codecs
= rej_calls=0 no_route=0 desc="new h323 line"
>
```

3.2.2.1.5 Command 'lcr'

Command lcr allows you to set, modify, delete and shift LCRs. There are following parameters to this command:

help - prints short help

list [<idx>] - prints a list of all LCRs or lists only LCR with <idx> specified

ls [<idx>] - same as a previous command

add - adds LCR to the list, there are other parameters needed, some of them required (parameters can be written in an arbitrary order).

destination=<destination name> - specifies Destination name, required

prefix=prefix1,prefix2,...,prefix8 - specifies Prefixes, can be regular expressions,

separated by comma, maximal eight prefixes allowed, not required

enabled|disabled - enables or disables added LCR, required

route=<valid route name> - specifies route for LCR, route must be valid, required

priority=pr_type_1,pr_type_2,pr_type_3,pr_type_4 - specifies Priority types,

valid values are first_free, cycles and free_minutes, value default will be used for not specified priority type number (default value is not listed), at most four Priority types are allowed, separated by comma, not required

desc=<description> - specifies Description, not required

replace idx=<idx> - replaces LCR with an <idx> specified. If LCR list is empty, this

command is not allowed, same parameters as for previous command are possible, if some parameter is omitted when replacing, original value will be used

del <idx> - deletes LCR with an <idx> specified

shift from=<from idx> to=<to idx> - shifts LCR from the position <from idx> to the

position <to idx>, this usually takes some time

enable <idx> - enables LCR with an <idx> specified

disable <idx> - disables LCR with an <idx> specified

Example: There are two LCRs in a list. User decides to insert new disabled LCR between them with Destination name value "lcr3",

default prefix and priorities, Route name value "sip" and Description value "third lcr". Then he decides to enable this new one LCR and finally to shift third LCR to the first position.

Command line shell, type help for more information or type exit to quit

```
>lcr ls
```

```
idx=0 destination="lcr1" prefix=.* enabled route="gsm1"
priority=cycle desc="fir
```

```
st lcr"
```

```
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```

```
idx=1 destination="lcr2" prefix=755 enabled route="all gsm"
priority=first_free
```

```
desc="second lcr"
```

```
>
```

```
>lcr insert idx=1 destination=lcr3 disabled route=sip desc="third
lcr"
```

```
Inserted successfully
```

```
>lcr ls
```

```
idx=0 destination="lcr1" prefix=.* enabled route="gsm1"
priority=cycle desc="fir
```

```
st lcr"
```

```
idx=1 destination="lcr3" prefix=.* disabled route="sip" priority
desc="third lcr
```

```
"
```

```
idx=2 destination="lcr2" prefix=755 enabled route="all gsm"
priority=first_free
```

```
desc="second lcr"
```

```
>
```

```
>lcr enable 1
```

```
Enabled successfully
```

```
>
```

```
>lcr ls
idx=0 destination="lcr1" prefix=. * enabled route="gsm1 "
priority=cycle desc="fir
st lcr"
idx=1 destination="lcr3" prefix=. * enabled route="sip" priority
desc="third lcr
"
idx=2 destination="lcr2" prefix=755 enabled route="all gsm "
priority=first_free
desc="second lcr"
>
>lcr shift from=2 to=0
Processing, please wait
Shifted successfully
>
>lcr ls
idx=0 destination="lcr2" prefix=755 enabled route="all gsm "
priority=first_free
desc="second lcr"
idx=1 destination="lcr1" prefix=. * enabled route="gsm1 "
priority=cycle desc="fir
st lcr"
idx=2 destination="lcr3" prefix=. * enabled route="sip" priority
desc="third lcr"
```

3.2.2.1.6 Command 'route'

Command route allows you to set, modify and delete routes. There are following parameters to this command:

help - prints short help

list [<idx>] - prints a list of all routes or lists only route with an <idx> specified

ls [<idx>] - same as a previous command

add - adds route to the list, there are other parameters needed, some of them required (parameters can be written in an arbitrary order).

name=<route name> - specifies Route name, required

lines=<valid_line_1,valid_line_2,...> - specifies Lines for route, all lines must be valid, separated by comma, required

timeint=<valid_timeint_1;valid_timeint_2;...> - specifies time intervals, all time intervals must be valid, separated by semicolon, required
CLIP|CLIR - specifies CLIP or CLIR, if not present, value default will be used (default value is not listed), not required

desc=<description> - specifies Description, not required

replace idx=<idx> - replaces route with an <idx> specified. If route list is empty, this command is not allowed, same parameters as for previous command are possible, if some parameter is omitted when replacing, original value will be used

del <idx> - deletes route with an <idx> specified

Example: There are two routes in a list. User decides to add new one with Name value "gsm1", Lines value GSM1, timeint value weekend;workdays, CLIP and no Description, then to change CLIP parameter to the default value and finally to delete first route in a list. Command line shell, type help for more information or type exit to quit

```
>route ls
```

```
idx=0   name="all   gsm"   lines=GSM1,GSM2,GSM3,GSM4  
timeint=weekdays;weekend;workdays
```

```
CLIP desc="All GSM route"
```

```
idx=1   name="sip"   lines=SIM:8942001103538068335  
timeint=weekdays;weekend;workdays
```

```
desc=""
```

```
>
```

```
>route add name=gsm1 lines=GSM1 timeint=weekend;workdays  
CLIP
```

```
Added successfully
```

```
>
>route ls
idx=0  name="all  gsm"  lines=GSM1,GSM2,GSM3,GSM4
timeint=weekdays;weekend;workdays
CLIP desc="All GSM route"
idx=1  name="gsm1"  lines=GSM1  timeint=weekend;workdays
CLIP desc=""
idx=2  name="sip"  lines=SIM:8942001103538068335
timeint=weekdays;weekend;workdays
desc=""
>
>route replace idx=1 NO_CLIP
Replaced successfully
>
>route ls
idx=0  name="all  gsm"  lines=GSM1,GSM2,GSM3,GSM4
timeint=weekdays;weekend;workdays
CLIP desc="All GSM route"
idx=1  name="gsm1"  lines=GSM1  timeint=weekend;workdays
desc=""
idx=2  name="sip"  lines=SIM:8942001103538068335
timeint=weekdays;weekend;workdays
desc=""
>
>route del 0
Removed successfully
>
>route ls
idx=0  name="gsm1"  lines=GSM1  timeint=weekend;workdays
desc=""
```

```
idx=1      name="sip"      lines=SIM:8942001103538068335
timeint=weekdays;weekend;workdays
desc=""
```

3.2.2.1.7 Command 'sip'

Command sip allows you to set, modify and delete SIP lines. There are following parameters to this command:

help - prints short help

list - prints a list of all SIP lines, REGISTER, CLIP and NAT are listed only if enabled, password is never listed

ls - same as a previous command

add - adds SIP line to the list, there are other parameters needed, none of them required (parameters can be written in an arbitrary order).

server=<SIP server address> - specifies SIP Server address

name=<SIP name> - specifies SIP Name

disp_name=<display name> - specifies Display name

listen_port=<port\> - specifies Listen port

user=<valid user name> - specifies User name, must be valid VoiceBlue user name

passw=<valid password name> - specifies Password for user name

codecs= - specifies Codecs, valid values are g711alaw, g711ulaw, g729, separated by semicolon, at most three codecs are allowed

add_to_req=<request> - specifies Add Phone context to REGISTER request

reg_exp=<seconds> - specifies Register expiration time in seconds

REGISTER|NO_REGISTER - enables or disables SIP registration

CLIP|NO_CLIP - enables or disables CLIP

NAT|NO_NAT - enables or disables NAT

nat_b=<nat port begin> - specifies NAT port begin

nat_r=<nat port range> - specifies NAT port range

nat=<nat address> - specifies NAT IP address

no_route_code=<no route code> - specifies No route code

desc=<description> - specifies Description

replace linenr=<line number> - replaces SIP line with <line number> specified. If

SIP line list is empty, this command is not allowed, same parameters as for previous

command are possible, if some parameter is omitted when replacing, original value will be used

del <linenr> - deletes SIP line with a <linenr> specified

Example: There is one SIP line in a list. User decides to add new one with no parameters

specified (all parameters will be set to default values). Then he decides to change Codecs to value g711alaw, Register expiration time to 5000 and to enable CLIP.

Command line shell, type help for more information or type exit to quit

```
>sip ls
```

```
linenr=25          server=172.20.172.196      name="1111"  
disp_name="1111" listen_port=5061 us  
er="vbegw"        codecs=g711alaw;g711ulaw;g729    add_to_req=  
reg_exp=0 REGISTER nat_b=0  
nat_r=0 nat= no_route_code=0 desc="SIP line"
```

```
>
```

```
>sip add
```

No parameters identified, default values will be set

Added successfully

```
>
```

```
>sip ls
```

```
linenr=25          server=172.20.172.196      name="1111"  
disp_name="1111" listen_port=5061 us
```

```
er="vbegw"   codecs=g711alaw;g711ulaw;g729   add_to_req=
reg_exp=0 REGISTER nat_b=0
nat_r=0 nat= no_route_code=0 desc="SIP line"
linenr=29 server= name="" disp_name="" listen_port=5061
user="" codecs= add_to_r
eq= reg_exp=300 REGISTER nat_b=0 nat_r=0 nat=
no_route_code=0 desc=""
>
>sip replace linenr=29 reg_exp=5000 CLIP codecs=g711alaw
g711alaw
Replaced successfuly
>
>sip ls
linenr=25      server=172.20.172.196      name="1111"
disp_name="1111" listen_port=5061 us
er="vbegw"   codecs=g711alaw;g711ulaw;g729   add_to_req=
reg_exp=0 REGISTER nat_b=0
nat_r=0 nat= no_route_code=0 desc="SIP line"
linenr=29 server= name="" disp_name="" listen_port=5061
user="" codecs=g711alaw
add_to_req= reg_exp=5000 REGISTER CLIP nat_b=0 nat_r=0
nat= no_route_code=0 desc
=""
```

3.2.2.2 Menu States

Menu States allows you to control all the VoiceBlue devices and present calls. It consists of two submenus: Devices and Calls. Submenu Devices prints a list of all present VoiceBlue devices. Submenu Calls prints a list of all active calls. These submenus are just a text form of the same graphical menus available in VoiceBlue web browser.

H.323 messages

Reference number Message

- 0 Local endpoint application cleared call
- 1 Local endpoint did not accept call
- 2 Local endpoint declined to answer call
- 3 Remote endpoint application cleared call
- 4 Remote endpoint refused call
- 5 Remote endpoint did not answer in required time
- 6 Remote endpoint stopped calling
- 7 Transport error cleared call
- 8 Transport connection failed to establish call
- 9 Gatekeeper has cleared call
- 10 Call failed as could not find user (in GK)
- 11 Call failed as could not get enough bandwidth
- 12 Could not find common capabilities
- 13 Call was forwarded using FACILITY message
- 14 Call failed a security check and was ended
- 15 Local endpoint busy
- 16 Local endpoint congested
- 17 Remote endpoint busy
- 18 Remote endpoint congested
- 19 Could not reach the remote party
- 20 The remote party is not running an endpoint
- 21 The remote party host off line
- 22 The remote party failed temporarily, app may retry
- 23 The remote party ended the call with unmapped Q.931 cause code
- 24 Call cleared due to an enforced duration limit
- 25 Call cleared due to invalid conference ID

26 Call cleared due to no circuit channel available

3.3 Web Browser Access

While the serial console interface enables you to change the basic gateway parameters only, the web browser gives you access to all parameter settings and services available in the VoiceBlue Enterprise gateway. To establish connection with the gateway, enter the gateway IP address into the Internet address setting line in the browser, e.g.:

<http://192.168.22.30>

If you have connected all parts properly and set the correct gateway and PC IP addresses, the request to enter the access user name and password should get displayed, see fig. 10.



The manufacturer supplies the gateways with an integrated user account *Admin* without having set its password. Mind the Upper/Lower Case while entering the user name and password! The first thing you should do for safety reasons after putting your gateway in operation is to change the administrator password.

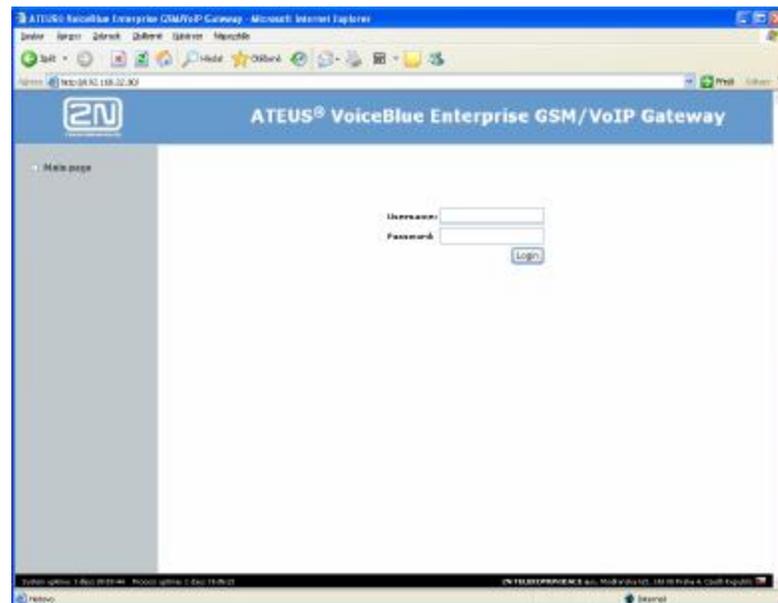


Fig. 10 – Web Administration Login Page

After a successful login you get onto the main page of the web application. The window is divided into four parts, see fig. 11.

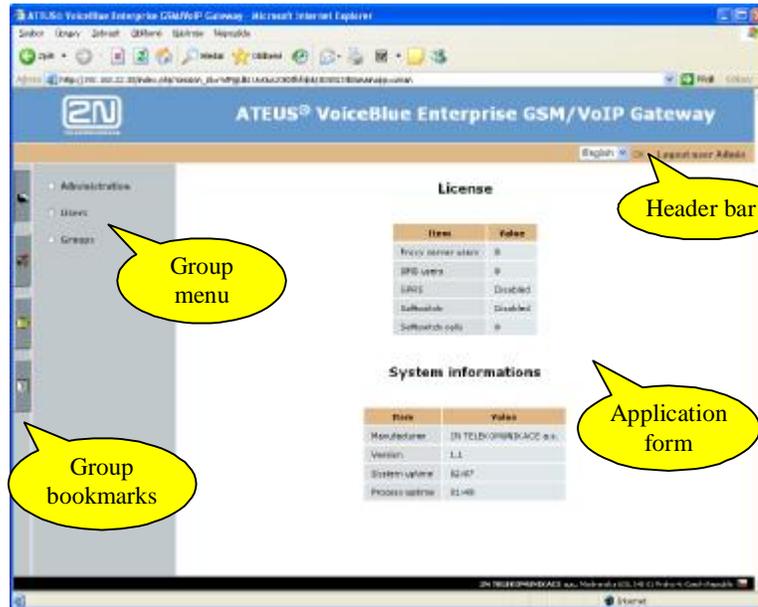


Fig. 11 – Web Interface Desktop

Header bar: The window header contains a pull-down menu, which helps you select the user relation language, displays the currently logged-in user and a log-out button.

Group bookmarks: The gateway settings are grouped in "cards" accessible under bookmarks. By clicking on a bookmark "ear" you get into the required setting group and see the group menu.

Group menu: The bookmark settings are arranged in two menu levels. By selecting a Level 1 item you display the respective Level 2 subitems. If the Level 1 item contains no more subitems or if you select a subitem, the appropriate application form gets displayed on the remaining window space.

Application form: Is the main part of the gateway user interface. It contains specific controls to the selected group menu item.

Survey of Group Bookmarks and Menus

Individual types of forms, their meanings and options will be discussed later. Now a survey of setting groups and menus in bookmarks is only given.

User administration: You will find all you need for user account administration in this bookmark. The following menus are available:

- Users – own account administration;
- Groups – account group administration.

Configuration: The *Configuration* bookmark integrates the majority of basic settings connected with the main gateway functions – especially GSM and VoIP line settings and call interconnection.

- Administration – common gateway settings;
- Devices – GSM and VoIP line configuration;
- Network configuration – gateway network parameters;
- GSM routing – routing of incoming GSM calls;
- Time intervals – routing and tariff metering time interval settings;
- Least Cost Router (LCR) – routing of incoming calls;
- Firmware update – gateway software updating.

Services: In addition to its basic function, i.e. a VoIP-to-GSM gateway, 2N VoiceBlue Enterprise provides a number of other network services. This bookmark integrates setting forms and on/off controls for these services.

- SIP proxy – SIP Express Router (SIP proxy) settings;

Statuses and records: Records of operation statuses are made in the gateway. To get access to these records and their settings and know the current gateway status information, select the *Statuses and records* bookmark menu.

- Statuses – information on currently made calls, tariff drawing statuses.
- Call accounting – call accounting records and settings.
- Records – software operation records.
- Record downloading – operation record packing and downloading.

SMS: This bookmark is reserved for SMS sending and receiving.

- SMS folders – folders for SMS and SMS sending form;
- SMS delivery – setting of the gateway response to SMS arrival.

3.3. Network Parameter Setting

Before starting to use the 2N VoiceBlue Enterprise gateway for making calls, remember to set its network parameters. You can do this in two ways. If the gateway has a factory setup or you know its IP address, you can connect a PC to it using a crossed Ethernet cable and select the respective parameters using the web browser.

Or, in other cases, you will definitely appreciate access via the serial port using a character terminal, e.g. the *HyperTerminal*.

Serial Console Access

Connect the gateway to your PC as described in Subsection 2.8. The main menu gets displayed on the terminal. Select the *Configuration* item and then *Network*. The menu shown in fig. 12 will get displayed.

VoiceBlue Enterprise		Main Menu	
Option	Value	Description	
1 - Address address	[10.0.0.1]	- Internet address	
2 - Network Mask subnet mask	[255.0.0.0]	- Internet subnet mask	
3 - Gateway default gateway	[0.0.0.0]	- Internet default gateway	
4 - Routing table configuration	[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	[""]	- Location	
01 - Contact	[""]	- Contact	
02 - Dhcp	[off]	- Use DHCP on	
03 - Class	[""]	- DHCP class	
Enter an option number, <ESC> previous menu			
>			

Fig. 12 – Network Parameter Configuration by Serial Console

The menu in fig. 12 contains the following items:

- **Address** – the IP address of the gateway. If the dynamic address assignment function using the DHCP is enabled, the value cannot be changed manually.
- **Network mask** – the network mask. If the dynamic address assignment function using the DHCP is enabled, the value cannot be changed manually.
- **Gateway** – the initial router IP address. It is used for routing of data traffic beyond the network limits. If the dynamic address assignment function using the DHCP is enabled, the value cannot be changed manually.
- **Dns1** – IP address of the 1st DNS server.
- **Dns2** – IP address of the 2nd DNS server.
- **Host name** – the network name of the GSM gateway.
- **Domain** – domain name.
- **Location** – any text that describes the gateway location. This value is published through the SNMP (in special versions of firmware).
- **Contact** – a text containing the gateway administration contact information. This value is published through the SNMP (in special versions of firmware).
- **Dhcp** – a mark of the dynamic address assignment function. If it is selected, the gateway gets its network settings from the DHCP server. If not, the values must be entered manually.
- **Class** – the equipment class name that is sent in the network setting sending request. This string can help the DHCP server distinguish different types of equipment and assign configuration parameters to them accordingly.

The *Dhcp* or *Address/Network mask* parameters at least must be set correctly for the gateway communication to be established within the TCP/IP network. The address of one DNS at least should be set too.

Web Browser Access

If you can get connected to the gateway with a web browser and know the administrator password, you can set the network parameters in the *Network Configuration* menu in the *Setting* bookmark. When the menu is opened, the current network setup is displayed. To get into the form for changes click on the *Modify* link located on the right under the network parameter table. A form gets displayed in the browser as shown in fig. 13.

The screenshot shows a web interface for network configuration. On the left is a sidebar menu with the following items:

- [-] Network config
 - [-] Filtering
 - [-] Port mapping
- [+] GSM routing
- [-] Time intervals
- [+] LCR
- [-] Set timezone
- [-] Synchronize time
- [+] Update
- [+] Upload / Download
- [-] Reboot

On the right, there are two configuration panels:

The top panel is titled "Enable DHCP:" and has a checked checkbox. Below it are the following fields:

- IP address: 192.168.50.66
- Network mask: 255.255.255.0
- Gateway:

The bottom panel contains the following fields:

- DNS server 1: 62.84.128.6
- DNS server 2: 192.168.50.1
- DNS server 3:
- Search list:

Fig. 13 – Network Parameter Setting by Web Browser

The form contains the following fields:

- **Enable DHCP** – a mark indicating that the network parameters should be gained from the DHCP server or set manually.
- **IP address** – the gateway IP address. If the dynamic address assignment function using the DHCP is enabled, the value cannot be changed manually.
- **Network mask** – the network mask. If the dynamic address assignment function using the DHCP is enabled, the value cannot be changed manually.

- **Gateway** – the initial router IP address. It is used for routing of data traffic beyond the network limits. If the dynamic address assignment function using the DHCP is enabled, the value cannot be changed manually.
- **Dns1** – IP address of the 1st DNS server.
- **Dns2** – IP address of the 2nd DNS server.

To confirm the changes, click on the *Modify* link. The process of value setting takes some time. Please wait for the browser's response.

Firewall

You can set packet filtering in the section Network config > Filtering.

The screenshot shows a configuration form for a firewall rule. At the top, there is a checkbox labeled "Enabled:" which is checked. Below this are several fields:

- Protocol:** A dropdown menu with "all" selected.
- Action:** A dropdown menu with "DROP" selected.
- Source IP address:** A text input field with a placeholder "[!] address[/mask]".
- Destination IP address:** A text input field with a placeholder "[!] address[/mask]".
- Source port:** A text input field with a placeholder "[!] port[:port]".
- Destination port:** A text input field with a placeholder "[!] port[:port]".
- Description:** A text input field.

- **Protocol** – filtered protocol; you can select UDP, TCP or both.
- **Action** – packets from/to defined IP address may be dropped or accepted
- **Source IP address** – source IP address of packet
- **Destination IP address** – destination IP address of packet

GPRS router

Some of GSM providers allow to assign static/public IP address to certain SIM card. Then it is possible to access devices in the network behind VoiceBlue Enterprise via IP (when GPRS connection is established) thanks to port mapping.



The screenshot shows a configuration panel for a port mapping rule. It includes the following fields:

- Enabled:** A checkbox that is checked.
- Protocol:** A dropdown menu currently set to 'tcp'.
- Interface:** A text field containing 'ppp0'.
- Port:** An empty text input field.
- New IP:** An empty text input field, followed by the label 'address[:port]'.
- Description:** An empty text input field.

- **Protocol** – selection of UDP or TCP traffic
- **Port** – external port mapped to the internal network
- **New IP** – internal network IP address to which is traffic mapped from predefined port
- **Description** – description of this port mapping rule

4

SECTION 4

Administration

This section describes how to set general parameters of the 2N VoiceBlue Enterprise gateway, create user accounts and rights groups.

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

- General gateway parameters;
- User accounts;
- Rights groups;
- Emergency change of administrator password.

4.1. General Gateway Parameters

There is an *Administration* item in the *Configuration* bookmark. By clicking on it you get on a page containing general parameters of the gateway. It is impossible to change the parameter values in this form. To change the values, click on the *Modify* link.

Confirm remove:	<input checked="" type="checkbox"/>
SIM card routing:	<input type="checkbox"/>
Enable SIP session progress:	<input checked="" type="checkbox"/>
Advanced config of regular expressions:	<input type="checkbox"/>
PIN for SIM cards:	
Disable GSM alerting detection:	<input type="checkbox"/>
Default language:	en
Image directory:	standard
CSS style filename:	vblue.css
Max user session time:	3600 (seconds)
Simple login page:	<input type="checkbox"/>
Mobility Extension:	
Outgoing ME enabled:	<input checked="" type="checkbox"/>
Incoming ME enabled:	<input checked="" type="checkbox"/>
Mobility extension SIP route:	192.168.50.111:5061

Fig. 14 – General Gateway Parameter Setting

The form in fig. 14 contains the following fields:

- **Confirm remove** – If this mark is selected, then any deleting command is accompanied by a confirmation window.
- **SIM card routing** – This selection binds the routing rules to specific SIM cards regardless of their physical position in a drawer.

- **Enable SIP session progress** – This mark enables the SIP stack to send so-called *session progress* messages during call establishing.
- **Advanced config of regular expressions** – enables usage of regular expressions in the LCR table
- **PIN for SIM cards** – all SIM cards should have the same PIN code, recommended is to disable PIN code
- **Disable GSM alerting detection** – enables to send SIP session progress immediately - gateway does not wait for alerting message from GSM network.
- **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.
- **Simple login page** – blank login page with Username and password fields only.
- **Outgoing ME enabled** – enables outgoing mobility extension feature.
- **Incoming ME enabled** - enables incoming mobility extension feature.
- **Mobility Extension SIP route** – IP address and port of ME device – typically there will be written IP address of VBE and port of internal SIP line.

4.2. User Accounts

The user accounts for web interface login or SIP telephone authentication to an integrated SIP proxy are defined in the *User administration* bookmark in the *Users* menu. If you select it, a list of user accounts gets displayed (fig. 15).

Name	Group	Language	Description			
admin	Administrators	English	Administrator			
admin	Administrators	English				<input type="checkbox"/>

Fig. 15 – Survey of User Accounts

The table columns in fig. 15 include:

- **Name** – user account name.
- **Group** – user rights group.
- **Language** – language to which the web interface switches after user login.
- **Description** – any text specifying the account meaning.

By clicking on the pencil symbol  you transit into the user account detail and can change the settings.

The cross  is used for deleting an account. Another way to delete an account is to tick off the field in the last table column and click on the *Delete selected items*.

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

User name: 100

New password: xoxox

Confirm new password: xoxox

Group: SMS

Language: English

Default application: SMS

Image directory: standard

CSS style filename: vblue.css

Rights: USERS+LINES+LCR USERS LINES LCR SMS

Rights denied: USERS+LINES+LCR USERS LINES LCR SMS

Line number: 100

Description: Line 100

Mobility Extension

Outgoing ME enabled:

Outgoing ME number: +44123123123

Incoming ME enabled:

Incoming ME number: +44123123123

Fig. 16 – User Account Setting

The form in fig. 16 contains the 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.
- **Password** – the password to be entered for user login. Dots are displayed instead of characters for safety reasons.
- **Confirm 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 (in special versions of firmware). 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 (in special versions of firmware). 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. This number also serves as a Mobility Extension trigger.
- **Description** – any text that describes the meaning of an account.
- **Outgoing ME enabled** – enables outgoing mobility extension for this line.
- **Outgoing ME number** – specifies outgoing mobility extension number for this user.
- **Incoming ME enabled** - enables outgoing mobility extension for this line.
- **Incoming ME number** – specifies incoming mobility extension number for this user. In most cases will be the same as a Outgoing ME number.

4.3. Rights Groups

Rights groups are created to eliminate the need to specify access rights for each user individually. Upon login, the resultant set of assigned and barred rights consists of the selected rights group settings and, if necessary, corrections made during the user account creation.

To define the rights groups use the *Groups* menu in the *User administration* bookmark. Having done this, you get a list of groups created.

Name	Description			
Administration	Group for administrators			
SMS	only for sms			<input type="checkbox"/>

Fig. 17 – Survey of Rights Groups

The table columns shown in fig. 17 include:

- **Name** – name of the rights group.
- **Description** – any text that describes the meaning of a group.

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

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

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

Group name:

Rights: USL*2-L.NL3-LCK JSLK*3 L.NL2 LCK

Rights denied: USL*2-L.NL3-LCK JSLK*3 L.NL2 LCK

Description:

Fig. 18 – Rights Group Setting

The form in fig. 18 contains the following fields:

- **Group name** – a unique identifier.
- **Rights** – parts of the system to be made available.
- **Rights denied** – parts of the system to be barred.
- **Description** – any text that describes the meaning of a group.

The web interface is divided into parts to/from which access rights can be assigned/removed. Tick-off fields in the above-mentioned forms, one pair per group, are used for rights group and user account setting. They are as follows:

- **ALL** – the whole system.
- **USERS** – definition of user accounts and groups.
- **LINES** – communication line settings.
- **LCR** – the Least Cost Router configuration.

4.4. Emergency Change of Administrator Password

Do not despair in emergency when you forget the *Admin* account password. Connect yourself to the gateway using a serial cable and select item 2 in the terminal console main menu – *Set Admin password*. Enter a new password and press .

4.5. Switching of protocols SIP and H.323

If you have not purchased the licence for usage of both SIP and H.323 protocols, it is necessary to choose between the protocols. In the gateway, the pre-set protocol is the H.323 protocol.

Switching between the protocols is realised by *Switch to SIP*, respectively *Switch to H.323* in menu – see fig. 19. Restart of the whole gateway will be done by switching of the protocols.

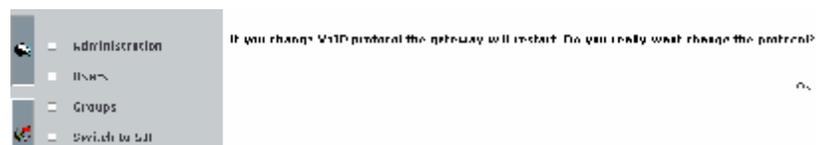


Fig. 19 – Switching to SIP protocols

5

SECTION 5

Insight into VoIP Gateway

This section deals with typical arrangements of VoIP networks, focusing on various applications of the 2N VoiceBlue Enterprise gateway. The meaning of the frequently used term "line" are explained and routing options available in the gateway are presented.

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

- "Point-to-Point" arrangement versus VoIP PBX networks;
- Gateway between VoIP and GSM networks;
- Gateway as a SIP proxy;
- Gateway between SIP and H.323 networks;
- Lines;
- Routing, Least Cost Router, DISA & co.

5.1. VoIP Network Arrangement

VoIP telephony communication has two components – signalling and voice. Signalling is primarily used for establishing and terminating calls, telephone login to a PBX, negotiating parameters and control of the speech/voice channel. The voice channel is only used for transmission of encoded digitised voice information.

Typically, VoIP telephones are operated together with a PBX, which coordinates their traffic. For a schematic arrangement see fig. 20. The VoIP telephony PBX is a software application that looks like a traditional PBX. Its functions include mainly numbering plan consistence maintenance, routing, user or telephone rights, call billing, call forwarding, DISA, etc. But it can integrate more functions, e.g. voice mail. SIP telephone PBXs are usually called SIP proxies while PBXs for H.323 are called gatekeepers.

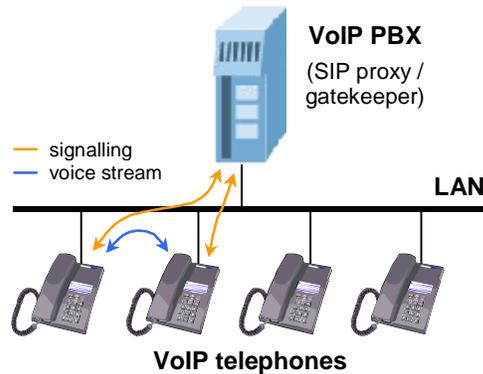


Fig. 20 – VoIP PBX Arrangement

To work properly, a VoIP PBX must process all signalling traffic. Unlike this, voice channel data are transmitted directly between the terminal points. This is, among others, a difference from the model described below.

It is because a PBX is not necessarily required in the IP telephony. Signalling protocols are designed in such a manner that it is possible to call from one terminal to another directly, without any mediator. To do this, you have to know the full ID of the terminal to be called, i.e. also the IP address and destination port, of course. This is the main disadvantage compared with the preceding model for multiple-telephone locations.



Fig. 21 – "Point-to-Point" Arrangement

If you use a PBX, the terminal points must know the PBX address only. The PBX is able to route requests according to information acquired in the process of telephone login to the PBX. With the "Point-to-Point" arrangement, all telephones must be equipped with information on surrounding telephones. This model is mostly applied in direct calling via the Internet where you do not want or cannot use services of any telecom provider.

The 2N VoiceBlue Enterprise gateway is capable of working in both the types of environment, with a PBX and without a PBX.

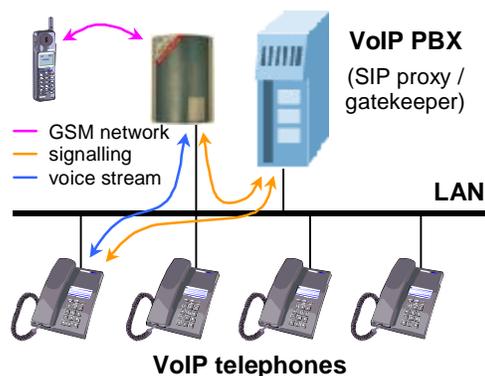


Fig. 22 – VoIP-to-GSM Gateway with PBX

It can even replace the PBX completely for SIP telephones because it is equipped with an integrated SIP proxy. This simplifies the VoIP network structure as shown in fig. 23.

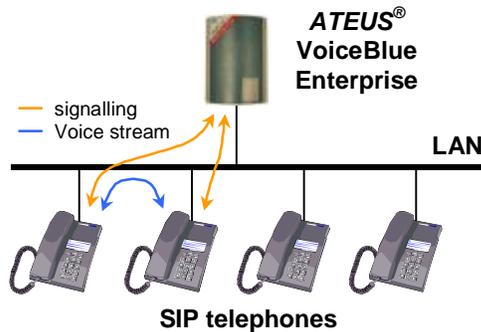


Fig. 23 – 2N VoiceBlue Enterprise as SIP Proxy

Although call interconnection between VoIP and GSM networks is the primary function of 2N VoiceBlue Enterprise, the gateway can also be used for bridging VoIP networks with different signalling standards – SIP and H.323. For such arrangement see fig. 24.

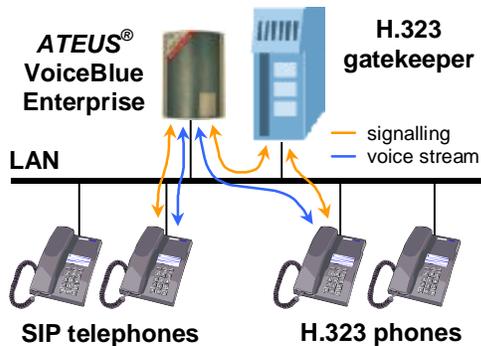


Fig. 24 – 2N VoiceBlue Enterprise as SIP - H.323 Gateway

The situation shown in fig. 24 differs from simple SIP-based call connection in that the gateway transmits both signalling and voice channels here, like in interconnecting calls from VoIP to GSM. This results in a certain limitation of the maximum number of calls due to a limited hardware pass-through capacity of the gateway.

5.2. Lines

Call interconnecting and routing in 2N VoiceBlue Enterprise takes place between lines. This means that where no line is available the gateway cannot connect a call. Nevertheless, the notion "line" is less materialistic in VoIP gateways than in classical telephone systems.

There are always lines corresponding to the GSM modules available in the gateway at least. To make the equipment work as a gateway, another line at least has to be created to the VoIP environment. This line is either used for gateway - VoIP PBX connection or direct gateway - VoIP telephone connection. From the viewpoint of the network environment of the gateway, a VoIP line represents a virtual VoIP telephone set.

It is possible to define more VoIP lines than one in the gateway. Then the gateway appears like a system of multiple virtual VoIP telephones. All types of lines are equal for routing and connecting purposes. Routing rules can be defined for VoIP lines mutually as for VoIP - GSM lines. This can be useful when more VoIP networks need to be interconnected, e.g. SIP and H.323.

GSM and SIM

We mentioned that a line corresponds to each GSM module. These lines are called "GSM module - 1" to "GSM module 4"¹ and correspond to respective SIM card drawers. If we define that all incoming calls should go via the "GSM module - 1" line, then the gateway shall always connect calls through the SIM card located in the first drawer.

If you operate the gateway with different providers' SIM cards, you will definitely want to use the GSM provider's SIM card for GSM call routing. In that case, however, avoid any misplacement of SIM cards after having configured the routing rule.

¹ If you have purchased a two-channel gateway, then only the "GSM module - 1" and "GSM module - 2" lines will be available.

The above-described behaviour, i.e. routing via GSM modules, is the default behaviour of the gateway. 2N VoiceBlue Enterprise offers another option - routing via SIM cards. When a new SIM card is inserted, the gateway creates automatically a "SIM card – xxx" line where xxx is the SIM card's Id (typically printed on the card next to contacts). This behaviour must be enabled explicitly in the common gateway parameter settings. If routing via SIM cards is activated, you can bind the routing rules both to the "GSM module" lines and "SIM card" lines. In the latter case, the physical location of the SIM card does not matter in call connection.

SIP

One or more lines can be defined in 2N VoiceBlue Enterprise via which the gateway communicates with other VoIP equipment using the SIP. Each line has a number of its own and receiving UDP port. You can register the line on the SIP proxy or communicate directly with another VoIP terminal. The registration can be either anonymous or related to gateway authentication.

H.323

Another signalling standard supported by the IP telephony is a group of protocols called H.323. Like for SIP lines, it holds true that one or more H.323 lines can represent the gateway. These lines can log in to the gatekeeper or communicate directly with VoIP telephones or other gateways.

5.3. Routing

For the purpose hereof, routing means a process of retrieving a route from the calling party to the called one. There are various routing methods according to the type of call and place from which the call is coming.

Within VoIP Network

Speaking of routing within a single VoIP network, we think of the activity executed by the PBX whenever a registered user initiates a new call. The result of this process is a route to another registered user based on the called number analysis. The called number can be an IP telephone, GSM gateway, PSTN gateway, another VoIP PBX, etc.

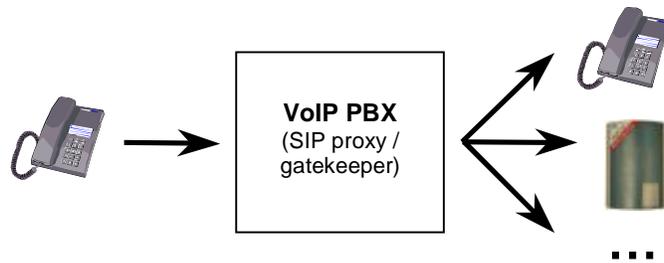


Fig. 25 – Call Routing by VoIP PBX

The routing options are dependent on the capabilities of the particular PBX. For SIP-based telephones, a SIP proxy server built in 2N VoiceBlue Enterprise can be used as a PBX. For details refer to Subsection 0.

From VoIP to GSM

On coming to an 2N VoiceBlue Enterprise VoIP line, a call proceeds to the Least Cost Router (LCR). The LCR decides which line the call shall use for going out. The decision depends on the called number and LCR settings, refer to Section 7.

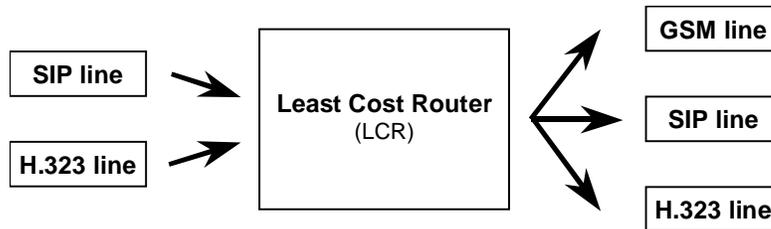


Fig. 26 – Routing of Incoming Calls from VoIP

Although the Least Cost Router is, as its name suggests, designed primarily for searching the most cost-efficient route to a GSM network, it is not limited just to this function. Generally, it can interconnect incoming calls from VoIP lines to any lines that are available in the gateway regardless of their types. This is very useful for interconnecting heterogeneous VoIP networks, such as SIP and H.323 environments.

Incoming Calls from GSM

A special routing mechanism is applied to calls coming to the 2N VoiceBlue Enterprise gateway from GSM networks.

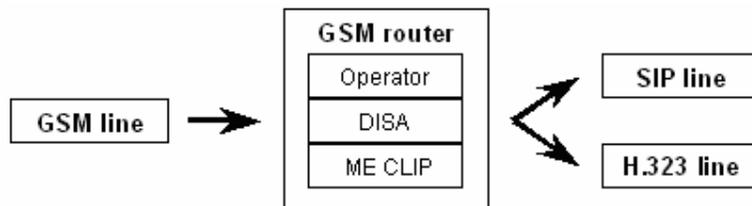


Fig. 27 – Routing of Incoming Calls from GSM

As selected, the GSM router: 1) forwards the call to a fixed operator; 2) plays ME message and allow to dial local or remote number according to the ME CLIP; or 3) forwards the call to the DISA automat where the calling party chooses the route using a voice menu and tone dialling.

Since incoming calls from GSM networks can only be routed to VoIP lines in the gateway it is impossible to connect a call back to the GSM network in this type of routing. This, however, does not mean that this operation cannot be executed on another routing level, e.g. in a PBX.

If the VoIP telephones used are equipped with the Calling Line Identification Presentation (CLIP) function, either the internal number of the gateway output line used for the incoming GSM call, or the CLIP obtained from the GSM network can be displayed. This VoIP line parameter is set at the first option by default. To make the gateway convey a public number to the VoIP telephone it is necessary to ensure that all routing elements be able to process the CLIP correctly in the opposite direction.

6

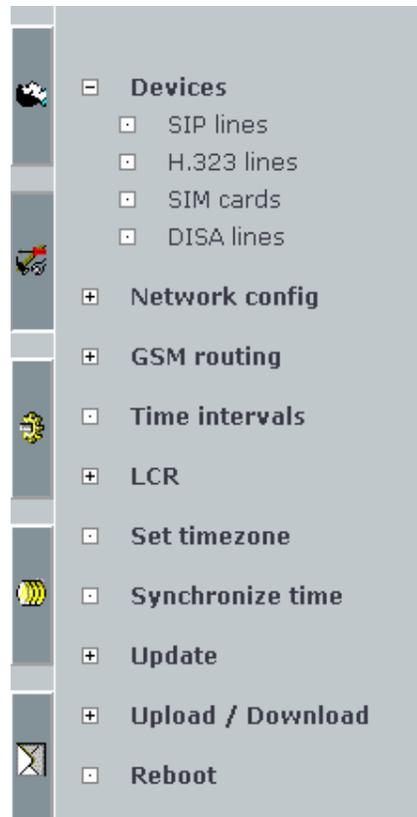
SECTION 6

Communication Line Setting

This section reverts to communication lines. Each line type is discussed separately and you will learn how to create and configure such lines.

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

- SIP lines;
- H.323 lines;
- SIM cards.
- DISA lines



To define line parameters use the *Configuration* bookmark, *Devices* menu. It is possible to add and modify VoIP lines and change SIM card settings. To set a certain type of lines select the respective subitem in the *Devices* menu.

To make 2N VoiceBlue Enterprise work as a real gateway between VoIP and GSM networks, create one line at least. You need not create lines corresponding to the GSM modules.

6.1. SIP lines

All defined SIP lines are tabulated clearly on the initial SIP line setting line, see fig. 28.

Line ID	SIP server	Phone number	Description			
[12]	192.168.22.30	1111	My SIP line			

Fig. 28 – Survey of SIP Lines

The columns include:

- **ID lines** – an internal line identifier used by the gateway software. Click on the number to get line details.

- **SIP server** – the IP address of the SIP proxy on which the line is registered.
- **Phone number** – line number used for calling.
- **Description** – any text that describes the meaning of a line.

By clicking on the pencil symbol  you transit into the line setting detail and can change the settings.

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

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

SIP Line Parameters

fig. 29 shows a SIP line in detail. To add a new line, use the *Add* link under the form, to modify an existing line use *Change*. By clicking on it you confirm the data entered and request execution.

There are also *Reset* and *Back* links under the form. The first one restores the values that were valid at the instant of form opening and the other one puts you back on the preceding page.

SIP server address:	<input type="text" value="192.168.2.240"/>	
SIP name:	<input type="text" value="vbegw"/>	
Display name:	<input type="text" value="vbegw"/>	
Listen port:	<input type="text" value="5061"/>	If exists SIP proxy registration, listen port cannot be 5060
User name:	<input type="text" value="vbegw"/>	
Password:	<input type="password" value="*****"/>	
Codecs:	<input type="list" value="G.729 8000 bps"/> <input type="list" value="G.711 A Law 64000 bps"/> <input type="list" value="G.711 u Law 64000 bps"/>	<input type="button" value="▲ Shift up"/> <input type="button" value="▼ Shift down"/>
Add Phone context to REGISTER request:	<input type="text"/>	
Register expires (seconds):	<input type="text" value="0"/>	
Register with proxy:	<input checked="" type="checkbox"/>	
Enable CLIP:	<input checked="" type="checkbox"/>	
Enable NAT:	<input type="checkbox"/>	
NAT port begin:	<input type="text" value="0"/>	
NAT port range:	<input type="text" value="0"/>	
NAT IP address:	<input type="text"/>	

Fig. 29 – SIP Line Detail

The form contains the following fields:

- **SIP server address** – the IP address or fully qualified network name of the SIP proxy to which the line is to log in.
- **SIP name** – line number for calling.
- **Display name** – a text to be displayed for the called line.
- **Used port** – the UDP port on which the line receives calls.
- **User name** – the name used by the line for SIP proxy registration.
- **Password** – the password used by the line for SIP proxy registration. Dots are displayed instead of characters for security reasons.
- **Codecs** – a list of codecs to be provided by the gateway for voice channel negotiation.
- **Add Phone context to REGISTER request** – advanced description for accounting.
- **Register expires** – allow to customize time of registration.
- **Register with proxy** – enable/disable registering with proxy.
- **Enable CLIP** – enable/disable of presenting by CLIP in case of outgoing call.
- **Enable NAT** – enable/disable to use “fake” (public IP address of router) IP address instead of real (private) IP address and select RTP port range.
- **NAT port begin** – first RTP port.
- **NAT port range** – range of used RTP ports.
- **NAT IP Address** – public IP address of NAT router; it is used as a “fake” IP address.
- **Description** – any text that describes the meaning of a line.

6.2. H.323

We can see a tabulated survey of defined lines (fig. 30) on the entry screen of the H.323 line configuration.

Line ID	Call method	Display name	Numbers	Listen port	Description			
[13]	Direct	VoiceBlue Ent	1234	1720	My H.323 link			<input type="checkbox"/>

Fig. 30 – Survey of SIP Lines

The columns include:

- **Line ID** – an internal line identifier used by the gateway software. By clicking on the number you get the line detail.
- **Call method** – defines how a line should initiate a call.
- **Display name** – line number used for calling.
- **Numbers** – line numbers used for calling.
- **Listen port** – the port on which the line receives calls.
- **Description** – a text entered by the administrator that can describe the meaning of a line if there are more lines than one.

By clicking on the pencil symbol  you transit into the line setting detail and can change the settings.

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

There is an *Add H.323 line* link under the table too. By clicking on it you get a new line defining form.

H.323 Line Parameters

fig. 31 shows a H.323 line detail. To add a new line, use the *Add* link under the form, to modify an existing line use *Change*. By clicking on it you confirm the data entered and request execution.

There are also *Reset* and *Back* links under the form. The first one restores values that were valid at the instant of form opening and the other one puts you back on the preceding page.

Call method: Gateway

Gatekeeper discovery method: Discover Gatekeeper automatically (use multicast)

Gatekeeper address:

Gatekeeper name:

Gateway address:

Gateway prefix(es):

Endpoint type: Gateway

Display name:

Numbers:

Listen port: 1720

Disable Fast start:

Disable H.245 tunneling:

Disable Early media start:

DTMF type: Signal

Transfer method: Call forwarding

Codecs:

- G.729 8000 bps
- G.711 A Law 64000 bps**
- G.711 u Law 64000 bps

Shift up

Shift down

Rejected call code: Local endpoint application cleared call

No route code: Local endpoint application cleared call

Description:

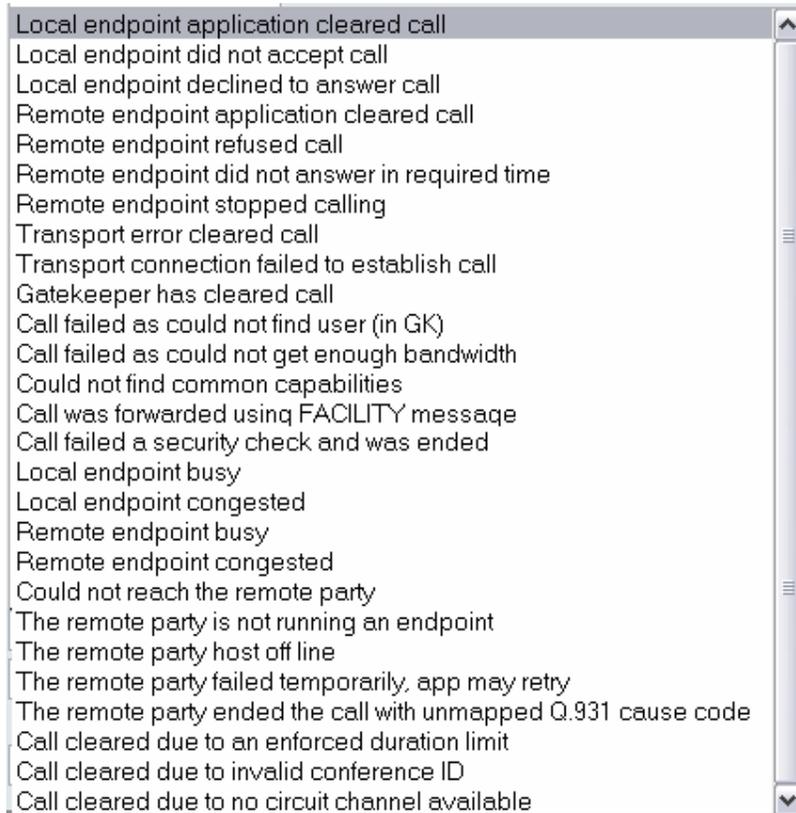
Fig. 31 – H.323 Line Detail

The form contains the following fields:

- **Call method** – defines how to initiate a new call. There are three options: 1) *direct* – calling including identification of the fully qualified network name of the called line; 2) *gatekeeper* – the selected gatekeeper shall find the route and establish connection; or 3) *gateway* – calling via a selected gateway.
- **Gatekeeper discovery method** – defines how to find the available gatekeeper address. The address can be entered statically or determined by network retrieval.

- **Gatekeeper address** – If not determined by network retrieval, the gatekeeper address 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 H.245 tunnelling** – disallows using of H.245 tunnelling.
- **Disable Early media start** – disallows using of *Early media strat* 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.
- **Rejected call code** – code which gateway returns back to gatekeeper/gateway in case of rejected call.

- **No route code** - code which gateway returns back to gatekeeper/gateway in case of no route for call (no GSM module/SIP line/H.323 line/SIP phone) is ready. This is used mainly for overflow/emergency routing.



- **Description** – any text that describes the meaning of a line.

6.3. SIM

Each SIM card inserted in the gateway has a set of parameters of its own. Their values are bound to the card Id and remain stored in the gateway even after the SIM card is removed. Thus, if the SIM card is reinserted, the parameters need not be reset.

The list of cards whose configuration parameters are available are displayed in a table on the introductory page of the SIM card configuration.

Present	SIM card number	Operator ID	Operator name	Description		
<input checked="" type="checkbox"/>	891202339051181538E	23002	EURATEL - CZ			
<input type="checkbox"/>	891202339051181538F	0	0			

Fig. 32 – Survey of SIM Cards

The columns include:

- **Present** – a mark that the SIM card has been inserted in the gateway.
- **SIM card number** – a unique card identifier. It is usually printed on the card next to contacts. By clicking on the number you display the card setting detail.
- **Operator ID** – a mobile network provider's identifier.
- **Operator name** – a mobile network provider's name.
- **Description** – a text entered by the administrator.

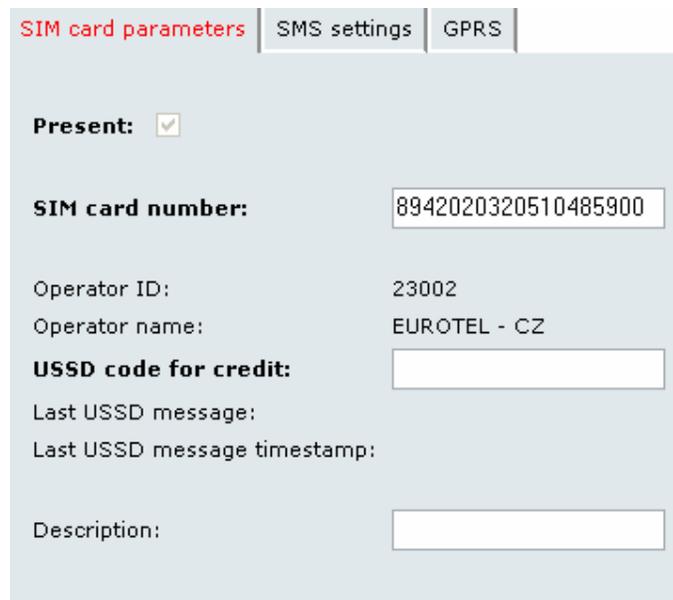
By clicking on the pencil symbol  you transit into the line setting detail and can change the settings. The cross  is used for deleting a line. There is an *Update* link under the table too for table updating.

SIM Card Detail

The SIM card setting form is arranged in three bookmarks:

- **SIM card parameters** – general parameters;
- **SMS settings** – SMS sending settings;
- **GPRS** – GPRS connection configuration.

SIM Card Parameters



The screenshot shows a web interface with three tabs: "SIM card parameters" (highlighted in red), "SMS settings", and "GPRS". The "SIM card parameters" tab is active, displaying a form with the following fields:

Present:	<input checked="" type="checkbox"/>
SIM card number:	<input type="text" value="8942020320510485900"/>
Operator ID:	23002
Operator name:	EUROTEL - CZ
USSD code for credit:	<input type="text"/>
Last USSD message:	
Last USSD message timestamp:	
Description:	<input type="text"/>

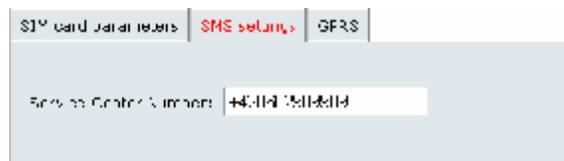
Fig. 33 – SIM Card Detail, *SIM Card Parameters* Bookmark

The bookmark contains the following fields:

- **Present** – a mark (for reading only) that the SIM card has been inserted in the gateway.
- **SIM card number** – a unique card identifier (for reading only). It is usually printed on the card next to contacts too.
- **Operator ID** – a mobile network provider's code (for reading only).
- **Operator name** – a mobile provider's name (for reading only).

- **USSD code for credit** – code used for credit balance checking (for prepaid SIM cards only). Works only if it is supported by GSM operator. For proper code contact your GSM operator/SIM card supplier.
- **Description** – any text that describes the meaning of a line.

SMS Setting



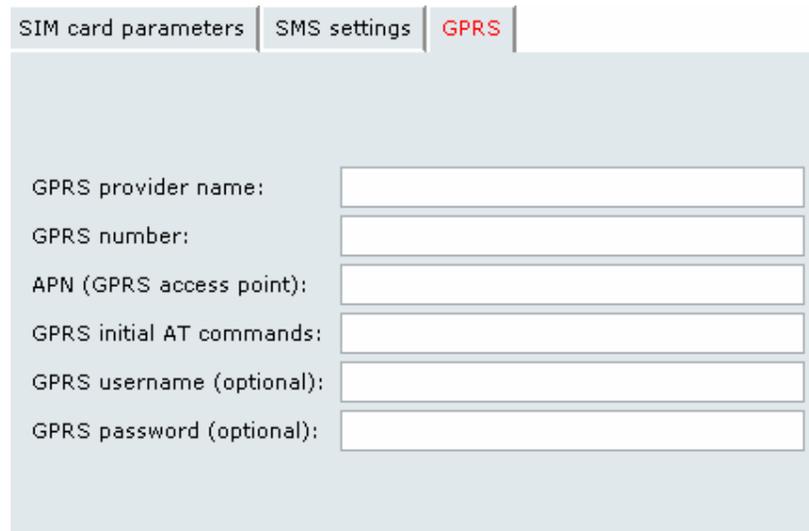
The screenshot shows a mobile application interface with three tabs at the top: "SIM card parameters", "SMS settings", and "GPRS". The "SMS settings" tab is selected and highlighted in red. Below the tabs, there is a single input field labeled "SMS Center Number:" with the value "+4411 1111111" entered.

Fig. 34 – SIM Card Detail, *SMS Setting* Bookmark

The bookmark contains the following fields:

- **SMS center number** – the number of the SMS centre through which short text messages are sent.

GPRS



The screenshot shows a mobile application interface with three tabs at the top: "SIM card parameters", "SMS settings", and "GPRS". The "GPRS" tab is selected and highlighted in red. Below the tabs, there are six input fields, each with a label to its left: "GPRS provider name:", "GPRS number:", "APN (GPRS access point):", "GPRS initial AT commands:", "GPRS username (optional):", and "GPRS password (optional):". All input fields are currently empty.

Fig. 35 – SIM Card Detail, *GPRS* Bookmark

The bookmark contains the following fields:

- **GPRS provider's name** – Name of GPRS service provider. Please fill in only alphanumeric chars.
- **GPRS number** – GPRS number which will be dialled during GPRS connection establishment. You can get this number from your GPRS provider.
- **APN (GPRS access point)** – The name of GPRS access point to internet network (it is usually something like this: *internet.<provider name>.<national domain abbreviation>*). You can get the APN from your GPRS provider.
- **GPRS initial AT commands** – Optional parameter. GPRS provider may use special AT command to establish GPRS connection. You can fill this command to this column.
- **GPRS username** – username for GPRS connection – optional in case when it is required
- **GPRS password** - password for GPRS connection – optional in case when it is required.

6.4. DISA lines

DISA line is used for incoming call. Most typical usage will be for incoming calls from GSM network. It might also serve for incoming calls from VoIP provider (SIP/H.323 line). DISA line plays message/dial tone and then wait for DTMF dialing. After receiving # (hash/pound key) character dials collected digits via predefined SIP/H.323 line.

Type of DISA:	Basic DISA
Description:	Basic DISA Basic DISA with uploaded message

- **Type of DISA** – selection of Basic DISA (continuous dial tone) or Basic DISA with uploaded message (message must be in ISDN a-law format and must not be bigger than 200kB).

- **Description** - text that describes the meaning of a line.

SECTION 7

LCR – Saving Automat

In this chapter, we will describe the significance and operation of basic routing mechanism of the gateway, i.e. saving automat. We will explain its operation and configuration.

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

- Routing process in the saving automat
- Routing rules – destinations, paths, lines and time intervals
- Normalisation of numbers
- Tariffication of calls
- List of prohibited numbers
- How to test the saving automat setting

7.1. What is saving automat?

The goal of the saving automat is find the optimal output line for called number. The LCR process has several phases:

Input normalisation: transformation of the numbers of calling and called participant into so-called normal form before the input into the saving automat.

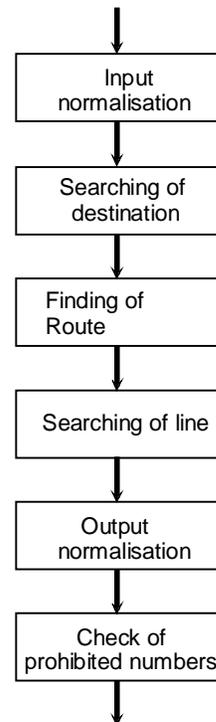
Searching of normalisation: the destination means the target side of the call. The part of the destination is Route and manner of passing of individual lines in the Route. The Route is searched according to prefix of the called normalised number.

Searching of Route: by the Route means the logical assembly of lines. The part of the Route is also time limit of the path validity within particular days of week.

Searching of line: the path may consist from one or more lines. The line, to which the call will be at last switched, is determined according to selected algorithm of line choice. There are three methods of the search: 1) *first free* – finds first free line, 2) *cycle* – search for free line with oldest time of the last call, and 3) *free minutes* – uses free line with highest number of remaining free minutes. Proper operation of the method requires setting of tariffs for relevant lines.

Output normalisation: transformation of calling and called numbers into normal form before switching into output line.

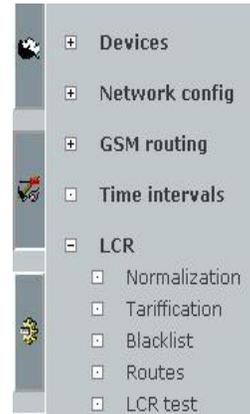
Check of prohibited numbers: after output normalisation and before switching into found output line, the table of *prohibited numbers* is searched for and in case of compliance the call is refused.



7.2. Saving automat setting

In the group bookmark *Configuration*, there is the *Saving automat (LCR)* item. After its opening, there are other five sub-items as described in the fig. on the right. Setting of individual phases of the saving automat process may be done through them. The sub-items will be discussed in following paragraphs.

The saving is pre-set from the manufacturer in such manner, that it connects all calls from VoIP into any GSM module without the called number modification.



7.3. Routing rules

The routing rules are basis of the saving automat. They consist from relatively high number of parameters, which provides considerable flexibility for the routing process. There is a guide to enable the most possible comfortable setting for the user. It will guide us through the configuration of rules.

We start our way by click on the *Saving automat (LCR)* item in menu of group bookmark *Configuration*. In window, there will be displayed the table with the pre-defined routing rules and with set of buttons, which enable to change their sequence, add new ones, modify them and cancel the actual rules, see fig. 36.



Fig. 36 – Overview of routing rules of the saving automat

All buttons except for the *Add* button are in initial state non-functional (greyed). They are activated just after selection of at least one routing group by ticking the tip field in second column of the relevant line. After pressing of such activated button, the operation above the selected routing rules is realised.

Adding of rule

The guide through the adding of the routing rule will be started by pressing the button *Add*, resp. *Add after* or *Insert before*. The guide will be launched in the new window, see fig. 37.



- Fig. 37 – Guide
- choices: 1) *first free* – first unoccupied line in natural sequence is selected, 2) *cycle* – preferably the line, which is not in operation for longest time, is used (uniform distribution of called time through the particular lines) – necessary to set Tariffication for each SIM is group, 3) *free minutes* – the line with highest number of remaining free minutes is preferably used (uniform drawing of free minutes).
- **Description** – any text documenting the path importance.

Change of rule

When we decide to change the setting of any routing rule, then we click to the name of the relevant destination in the overview of rules, see fig. 38. This will launch the guide, which is very similar to the guide, with which we have acquainted in the previous paragraphs. The only difference is the presence of the *Change* button instead of the *Add* button.

Destination		Enabled	GSM groups	Description
GSM	<input type="checkbox"/>	<input checked="" type="checkbox"/>	GSMGROUP	

Fig. 38 – Change of routing rule

Changes of routing rules effect the gateway operation immediately after their realisation. Therefore it is not needed to restart the whole equipment.

Removal of rule

The routing rule, which is no longer required, might be cancelled by ticking the tip field in the second column of the relevant line and pressing the *Remove* button, see example on fig. 39. Removing of several rules at the same time may be realised by simultaneous ticking of more lines.

Destination	Enabled	GSM groups	Description
GSM	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	GSMGROUP

Fig. 39 – Removing of routing rule

Change of sequence

The rule sequence may be changed by similar procedure, as the removal of the rules. We select the relevant line by ticking and press the *Move up* resp. *Move down* button.

Forbidding/permitting of rule

As we already know, the routing rule affects the routing process only if it is permitted. During establishing and/or change of the rule setting, we may change the state of the *Permitted* button - see

fig. 37. Operatively the activity may be realised also in the overview of the routing rules by clicking to the tip field in the third column of the relevant line.

Destination		Enabled	GSM groups	Description
GSM	<input type="checkbox"/>	<input checked="" type="checkbox"/>	GSMGROUP	

Fig. 40 – Forbidding/permitting of the routing rule

7.4. Routes

Routes are part of LCR rules, as described in previous chapter.

List of Routes

The Routes may be listed by clicking to the *Routes* folder in the *LCR* sub-menu, as illustrated on fig. 41.

Route name	Route lines	Time intervals	Description		
GSM	GSM module - 2 GSM module - 3	weekdays weekend workdays			

Fig. 41 – Removal of routing rule

Adding of Route

Route may be added by clicking on the *Add Route* item. In the window, following form (see fig. 42) is displayed, in which following data must be entered:

The screenshot shows a web-based configuration interface for adding a route. It features several input fields and buttons:

- Route name:** A text input field containing "GSM".
- Lines of route:** A list box containing "GSM module - 1", "GSM module - 2", "GSM module - 3", and "GSM module - 4". The second and third items are highlighted in blue, indicating they are selected. To the right of the list box are three buttons: "Add SIP line", "Add H.323 line", and "Add SIM card".
- Time intervals:** A list box containing "weekdays", "weekend", and "workdays". To the right are two buttons: "Add day group" and "Add time range".
- CLIP/CLIR:** A dropdown menu currently set to "Default".
- Description:** An empty text input field.

Fig. 42 – Adding of Route

- **Route name** – the name of the Route may contain any alphanumerical characters. The LCR rules then refer to the name, as was described in previous chapter.
- **Lines of Route** – list of the available lines, which create the Route. The selection is made / cancelled by clicking of the left mouse button to the relevant item. In the list, several items may be selected at the same time by pressing the  button, while clicking by the left mouse button. The selected items are in blue. The new lines may be established by pressing the *Add SIP line* resp. *Add H.323 line* button.
- **Time intervals** – determine the time validity of the routing rule. Three time groups are standardly pre-defined: *weekdays* – Monday to Friday for whole day, *workdays* – Monday to Friday from 7:00 AM to 17:00 PM, *weekend* – Saturday and Sunday. New time intervals may be created after pressing of the *Add day group* resp. *Add time range* button.
- **Description** – any text documenting the path significance.

The *Limits* bookmark and other parameters of the R are not functional in this version.

Change of a Route setting

If you decide to change the setting of any Route, then click to the symbol of pencil  of appropriate Route. This starts the form, which is very similar to the form described in previous paragraphs. The only difference is the presence of the *Change* button instead of the *Add* button.

Removing of a Route

If you decide to remove the Route, then click to the cross symbol  and appropriate Route will be erased. The Route may be also erased by ticking the field in the last column in the table and clicking to the *Remove selected* bookmark, by which all ticked Routes are erased at the same time.

7.5. Normalisation of numbers

During description of the saving automat operation on the Chapter 7.1 we have mentioned also the modifications of the numbers before the input and after the output of the routing process – so-called normalisation. They are conditional transformations of the called and/or calling number. This ensures the transformation into the unified format, which then simplifies the definition of own routing rules.

The normalisation prescription is determined by three parameters:

- 1) **Normalisation type** – determines, in what stage of the process (input / output) and to what numbers it should be used (called/calling).
- 2) **Conditions** – transformation will be applied only for numbers, which fulfil the condition. The condition is specified by prefix – chain, by which the number must begin. The prefix is automatically detached and the residual part of the number after the prefix goes into the transformation phase.
- 3) **Transformation prescription** – enables modification of the number by detaching of certain number of the characters from its starting, respectively adding of the new prefix.

To illustrate the practical possibilities, which are provided to us by the transformation manner, we state the two typical, but entirely different usages of the normalisation.

Example 1: It flows from the numbering plan of VoIP network, that we must dial „0“ to call outside this network. Therefore, if we want to call the mobile user with the number „602 123 456“, we must dial the „0 602 123 456“ on the VoIP telephone. Before entry into the saving automat, usually transformation into the international format is realised. For the Czech Republic, it would be „+420 602 123 456“.

In this case, the prefix of normalisation will be „0“, number of subsequent detached characters 0 and the new added prefix will be „+420“.

Example 2: the second example will be the relatively unusual requirement for transformation of any number into the single particular number. What is the purpose? Let us say, that we do not want to enable the international calls through our gateway for the

employees of our company. While also another procedures are possible, we decide to switch all calls starting with „00“ (our international call flag) to the voice information service, which will nicely announce to the user, that such calls are prohibited. The voice service is located in our VoIP network and let us say that it has number 9876.

This behaviour may be achieved by setting the normalisation, which has the prefix „00“, number of subsequently detached characters, e.g. 50, and new added prefix „9876“. Regarding the fact that we do not suppose the occurrence of the phone number, which is longer than 50 characters, then we may expect, that nothing remains from the original number, and the result of the transformation will be only „prefix“ 9876.

Adjustment of the normalisation prescriptions is available via the group *Configuration* in menu *Saving automat (LCR)* via the *Normalisation* item. The initial display is the list of established normalisation rules.

Prefix	Remove count	Add number	Type	Description			
0	0	+420	Called outgoing				<input type="checkbox"/>

Fig. 43 – Overview of normalisations

By clicking to the symbol of pencil  we move to the form, in which we may made the changes.

Symbol  serves for cancelling of the normalisation prescription. These may be cancelled also together by ticking in the last column of the table and clicking to the *Remove selected* located under the table.

Under the table, there is also the *Add to LCR normalisation* reference. By clicking to it, the form for definition of new line is displayed.

Meaning of the field in the form is the same as the meaning of columns in the normalisation table in fig. 43:

- **Prefix** – prefix, by which the number must begin to enable the applying of the transformation prescription to it.
- **Remove number** – number of characters, which are detached from the beginning of the number after the prefix.

- **Add the number** – prefix, which is added before the number residue after detaching of the prefix and given number of the initial numerals.
- **Type** – normalisation type determines, in which phase the normalisation rule will be applied. We may chose from following four possibilities: *incoming call* – on input for calling numbers; *incoming called* – on input for called numbers, *Outgoing calling* – on output for calling numbers, *Outgoing called* – on output for called numbers.
- **Description** – any text describing the meaning of the normalisation rules.

7.6. Tariffication of calls

Tariffication of calls in the 2N VoiceBlue Enterprise gateways means the sophisticated recording of the called time on individual SIM cards. The actual version of the software does not evidence the price for calls in any way. Monitoring of the free minutes is realised with view to optimal usage of the bought call tariffs and is used in the routing process of the saving automat.

The tariffication scheme divides the call into two time ranges. In each rang, the tariffication may be realised per various time units. Example of the typical tariff is illustrated in fig. 44. First 10 minutes is tarifficated per the whole minutes, and after it there is the tariffication per secondes.

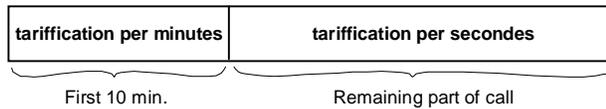


Fig. 44 – Example of calling tariff definition

However, to one SIM card various tariffication schemes for various days in week and various parts of day may be defined and composed together. This mechanism may cover most part of the tariffs usually offered by the mobile operators.

The tariffs are established and assigned to the SIM cards in the group bookmark *Configuration* in menu *Saving automat (LCR)* after opening of the item *Tariffication*. Input point is the general table of the defined tariffs - see fig. 45.

Tariff name	Line ID	Day of restriction call limit	Tariffication [sec.]	Free min/seconds	Description			
PROFITING P	2-4 000-142012-900 / 4012-11		00:00	00:00				

Fig. 45 – General tariff table

In individual phases, there is displayed:

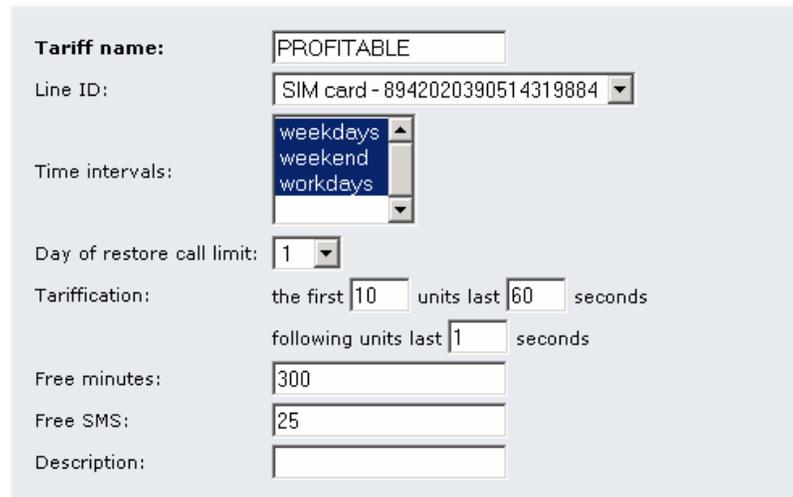
- **Tariff name** – each tariff must be named.
- **Line ID** – identification of the SIM card.

- **Day of the calling limit recovery** – counters of called minutes are reset each month in given day.
- **Tarifficate [sec]** – tariffication scheme. It consists from two data for each time range one. For the first time range: from how many time units it is composed and how many seconds has one unit. Second part of the call: information on length of the tariffication unit.
- **Free minutes/SMS** – number of free minutes, resp. SMS.
- **Description** – any text documenting the meaning of tariff.

Clicking on the symbol of pencil  moves into the tariff details, in which we may realise the changes.

Symbol  serves for removing of the tariff. The tariffs may be removed also by ticking the field in the last column in the table and clicking on the *Remove selected* item.

Under the table, there is also the *Add LCR tariff* reference. By clicking to it, the form for the definition of new tariff is displayed.



Tariff name: PROFITABLE

Line ID: SIM card - 8942020390514319884

Time intervals: weekdays

Day of restore call limit: 1

Tariffication: the first 10 units last 60 seconds
following units last 1 seconds

Free minutes: 300

Free SMS: 25

Description:

Fig. 46 – Tariff detail

The form on fig. 46 contains following fields:

- **Tariff name** – each tariff must be named.
- **Line ID** – list if SIM cards, by which the connection may be realised.

- **Time intervals** – specify the time validity of the tariff.
- **Day of the calling limit recovery** – counters of called minutes are reset each month in given day.
- **Tarifficate** – number of time units in first range, first range unit length, second range unit length.
- **Free minutes** – number of free minutes.
- **Free SMS** – number of free SMS.
- **Description** – any text documenting the meaning of tariff.

7.7. Forbidden numbers

After the output normalisation before the switching of the call into the found output line, the check of the called number against the forbidden number list is realised. If the number begins with the string stated in the list the call is refused. It means, that we may input into the list both the particular whole telephone numbers, and only their starting parts (prefixes) and collectively exclude e.g. all international calls of calls to special services tarifficated by higher rates (erotic lines etc.).

Adjustment of the forbidden numbers is realised in the group bookmark *configuration* in menu *Saving automat (LCR)*. After selection of the *Forbidden numbers* item, the list of forbidden numbers is displayed, see fig. 46.

Regular expression	Description			
+420906				<input type="checkbox"/>

Fig. 47 – List of forbidden numbers

Following items are displayed in individual columns:

- **Prefix** – initial part of the forbidden number (also whole number may be entered).
- **Description** – any text, documenting the meaning of the item.

Clicking the symbol of pencil  moves into the detail, in which the changes may be realised. Only above-mentioned two fields are filled in the change form.

The symbol  serves for removing of the forbidden number. The numbers may be removed also by ticking the field in the last column of the table and clicking to the *Remove selected* item.

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

7.8. LCR Test

The *LCR Test* item in the menu *Saving automat (LCR)* in the group bookmark *Configuration* serves for testing of the changes in the saving automat adjustment. After entering of the calling and called number and clicking to the *LCR Test* resp. *Repeat LCR Test* reference located under the form, the simulation of the LCR process is simulated.



The image shows a web form for the LCR test. It contains two input fields: 'Caller:' and 'Called number:'. Below the form, there is a link labeled 'LCR test | Reset'.

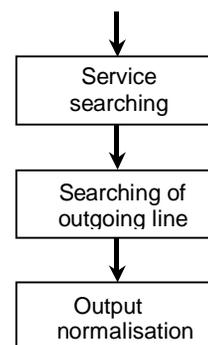
Fig. 48 – Parameters of the LCR test

After the simulation, following items are displayed: normalised calling and called number, name of the found output line or reason, why the search of the line was not successful.

Routing of calls from GSM

Processing of the incoming calls from the GSM network is realised simply by the gateway by routing of the incoming call into the VoIP network to one particular number. The routing may be set for each SIM card of GSM module separately.

During processing of the incoming calls from the GSM side, the automat firstly tries the service *Operator* according to the GSM module, then according to the number of SIM card. After the service is found, the line, which is assigned to the *operator* service, is selected for the outgoing call. Finally, the calling number is normalised according to the normalisation table.



8.1. Setting of the operator service

Routing setting is realised by clicking to the *GSM routing* reference. Firstly must be set the *operator* service. Clicking to the *Operator*



item, the table of configured operators is listed (see fig. 49)

GSM routing - Operator

Service name		Operator number	VoIP line	Description
Cisco7940	<input type="checkbox"/>	2265	SIP - My line	

Operation:

Fig. 49 – List of the operator services

Adding of the operator service

New service *operator* may be added by clicking to the *Add* button. Then, the form for adding of the *operator* service is displayed se (see fig. 50).

Add GSM routing Operator service

Service name:

Operator number:

VoIP line:

Description:

Add | Reset | Back

Fig. 50 – Adding of the operator service

- **Service name** – The field may contain any alphanumerical characters.
- **Operator number** – number in VoIP network, to which the incoming call from the GSM network is routed.
- **VoIP Line** – line used for dialling of the number into the VoIP network (might be also DISA line).
- **Description** – any text documenting the meaning of the *operator* service.

The adding of the *operator* service itself is done by clicking to the *Add* reference.

Change of the operator service

Change of the *operator* service is realised by selecting of relevant line (choice is done by ticking in second column of table displayed in fig. 49) and clicking on *Change* button. Further procedure is the same as in the previous paragraph with the exception that the saving of the changes is done by clicking to the *Change* item.

Removing of the operator service

The *operator* service may be removed by selecting of relevant line (similarly as in previous paragraph) and clicking to the *Remove*.

Change of the operator service sequence

Change of the sequence of the *operator* services is made similarly as in chapter 7.3.

8.2. Adjusting of SIM routing rule

Adding of SIM routing rule

Clicking to the *SIM* item in left menu will display the list of the SIM routing rules (see fig. 51).

GSM routing - SIM

SIM card	Service name	Description		
8942001032430068798	Cisco7940			

Add SIM card routing

Fig. 51 – List of SIM routing rules

Clicking to the *Add routing of SIM card* item will display the form for adding of the routing rule (see fig. 52).

Add SIM card routing

SIM card:	<input type="text" value="8942001032430068798"/>
Service name:	<input type="text" value="Cisco7940"/>
Description:	<input type="text"/>

Add SIM card routing

Fig. 52 – Adding of SIM routing rule

- **SIM card** – the number of the SIM card is selected to this field. The incoming calls through the SIM number will be assigned to the service selected in the *Service name* field.
- **Service name** – service, through which the calls have to route into the VoIP network, is entered to this field.
- **Description** – any text documenting the SIM route rule meaning.

Change of the SIM routing rule

Change of the SIM routing rule is realised by clicking to the symbol of pencil . After clicking, the form similar to the form from the previous chapter will be displayed. The changes are saved by clicking to the *Change* button.

Removing of the SIM routing rule

The SIM routing rule is removed by clicking to the symbol .

8.3. Setting of the GSM routing rule

Adding of SIM routing rule

Clicking to the *GSM* item in left menu will display the list of the GSM routing rules (see fig. 53).

GSM routing - GSM

GSM device	Service name	Description		
GSM module - 3	Cisco7940			

Add GSM device routing

Fig. 53 – List of GSM routing rules

Clicking to the *Add routing of GSM equipment* item will display the form for adding of the routing rule fig. 54).

Add GSM device routing

GSM device:

Service name:

Description:

Add | Reset | Back

Fig. 54 – Adding of GSM routing rule

- **GSM equipment** – the GSM module number is entered into this field. The incoming calls through this module will be assigned to the service specified in the *Service name* field.
- **Service name** – service, through which the calls have to route into the VoIP network, is entered to this field.
- **Description** – any text documenting the GSM route rule meaning.

Change of the SIM routing rule

Change of the GSM routing rule is realised by clicking to the symbol of pencil . After clicking, the form similar to the form from the previous chapter will be displayed. The changes are saved by clicking to the *Change* button.

Removing of the SIM routing rule

The GSM routing rule is removed by clicking to the symbol .

8.4. Normalisation

Work with normalisation table is described in chapter 7.5. The table normalises the calling (CLIP) numbers of the incoming calls from GSM network.

The normalisation should be made to display the calling number on the IP telephone in such manner, which enables proper routing of the back call into the 2N VoiceBlue Enterprise gateway.

If e.g. the gateway is configured to dial the calls into the GSM network by means of prefix 0 and the incoming CLIP from GSM network starts with +420, the prefix +420 must be detached and replaced by 0 (see fig. 55).

GSM routing - Normalization

Prefix	Remove count	Add number	Description			
+420	0	0				<input type="checkbox"/>

Add GSM routing normalization | Remove selected

Fig. 55 – Normalisation of CLIP

9

SECTION 9

SIP Proxy

This section discusses in detail the capabilities of an integrated SIP proxy server, its VoIP PBX function, routing configuration, authentication and monitoring.

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

- Routing rules;
- Authentication for SIP terminal registration;
- Survey of registered terminals.

The 2N VoiceBlue Enterprise gateway is equipped with an integrated SIP proxy, which can serve as a PBX for SIP telephones. It is easy to configure, the only function to be set is routing.

If prefix	Strip	Add	Do action	With parameter			
sip 0	-	+120	connect to VoiceBlue	SIP - My SIP line			<input type="checkbox"/>
else	0		no routing operation				

Fig. 56 – Example of Integrated SIP Proxy Server Routing Table

Fig. 56 shows an example of a completed routing table that gets displayed when you open the *SIP proxy* item in the *Services* group bookmark. Each line represents one rule. Having received a call establishing request, the SIP proxy searches this table from the top to bottom for the appropriate routing rule. Doing this, it compares the called destination with the value in the first table column. If the called subscriber's identifier starts with the prefix included in the *If prefix* field of the routing rule, the SIP proxy terminates searching and uses the rule found.

The last table line represents the so-called initial routing rule. It cannot be deleted. It is applied to all calls for which no explicit routing rule has been found.

The routing rule defines what the SIP proxy should do with a call. It can reject it, forward it to another host and/or port, transfer it to a gateway VoIP line (i.e. to the LCR), or search for the called subscriber in the database of logged-in SIP telephones. Before one of the above mentioned operations is executed, it is possible to modify the called subscriber's identifier by removing a certain number of characters from the left or adding a new string to its beginning (refer to the *Remove* and *Add* columns).

By clicking on the pencil symbol  you transit into the routing rule detail and can change the settings.

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

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

Fig. 57 shows a form for setting the SIP proxy routing rules. The form fields correspond to the header of the table in fig. 56.

If prefix:	sip:
Strip:	1
Add:	+420
Do action:	Connect to VoiceBlue
With parameter:	SIP - My SIP line

Fig. 57 – SIP Proxy Routing Rule Detail

The form and the table include the 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 VoiceBlue* – 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 VoiceBlue* action.

The call forwarding parameter can be: a host address (2) in the format of the IP address or fully qualified network name, port number (3), or both at the same time (4), with the host specification being separated from the port number with the ":" character.

9.1. Survey of Registrations

It is possible to monitor via the web interface which equipment units are registered on the SIP proxy. For the survey click on the *Registration* item in the *SIP proxy* menu in the *Services* bookmark.

9.2. Switching on/off

If you use external SIP proxy server and you do not wish to use the integrated proxy server, you have the possibility to switch it off.

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

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



10

SECTION 10

SMS Server

This section shows how to set the delivering rules for an integrated SMS server.

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

- Message delivery according to contents;
- Reply to message;
- VIP numbers.

10.1. Message Delivery

Thanks to an integrated SMS server, the 2N VoiceBlue Enterprise gateway is capable of not only sending but also receiving and delivering SMS. The delivery to the user's SMS box is controlled by three rules that can be either enabled or disabled. These rules are applied to an incoming message in the sequence as described below. The delivery configuration settings are available in the *SMS delivery* menu in the *SMS* group bookmark, see fig. 58.

Distribute incoming messages:

Parse contents:

Message response:

VIP numbers:

Save outgoing messages to Sent folder:

Module 0 - Deliver SMS to user: 100

Module 1 - Deliver SMS to user: 101

Module 2 - Deliver SMS to user: 200

Module 3 - Deliver SMS to user: Admin

For outgoing SMS use: LCR

Automatic answers delivering:

Answer arrives until: 10 Minutes

SMTP/POP3 server settings:

Email domain name: 192.168.50.111

Delete SMS with POP3: Delete permanently

POP3 email subject: SMS from %n to %u, date %d

Maximum number of SMS per email: 1

SMTP IP access control:

SMTP authentication required:

Allow 16-Bit SMS:

Allow SMS concatenating:

Insert email sender to SMS:

Insert email subject to SMS:

Fig. 58 – SMS Delivery Settings

The form contains the following fields:

- **Parse contents** – enables SMS delivery according to contents.
- **Message response** – enables response deliveries.
- **VIP numbers** – enables deliveries according to the list of VIP numbers.
- **Reply comes within** – response timeout, refer to the *Reply to message* delivery mechanism.

Delivery According to Contents

With this rule activated, every incoming SMS is analysed for the #User# string where the *User* is the login name of a valid gateway user. If the string is found, the message is delivered into the *User* box.

Reply to Message

This delivering method decides whether the message received is a reply to a message sent earlier. For this purpose, the gateway keeps a list of sent messages. If, within a certain time limit, a reply comes from the inquired mobile telephone, it is delivered into the original message sender's box.

VIP Numbers

The last way of delivering is to examine the table of so-called VIP² numbers. The table contains a part of or the whole VIP number that is compared with the message sender number. Each VIP number is assigned a list of boxes to which VIP senders' messages are to be saved.

10.2. List of VIP Numbers for SMS Delivery

There is a *VIP numbers* item in the *SMS delivery* menu in the *SMS* group bookmark. When you open it, you get a list of predefined VIP numbers that is searched whenever a new SMS is received if the *VIP number* delivery rule is active.

² They are called VIP numbers because they are typically used for defining the list of very important clients, e.g. a helpdesk system. If such a client sends SMS to your gateway you want to be sure that it will be delivered into the right hands in any case.

VIP number	Description			
+120602223155				<input type="checkbox"/>

Fig. 59 – List of VIP Numbers for SMS Delivery

The columns include:

- **VIP number** – beginning (or the whole) of the SMS CLIP to which a list of addressees is assigned.
- **Description** - any text that describes the meaning of an item.

By clicking on the pencil symbol  you transit into the VIP number detail and can change the settings.

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

There is an *Add VIP number* link under the table too. By clicking on it you get a VIP number defining form, see fig. 60.

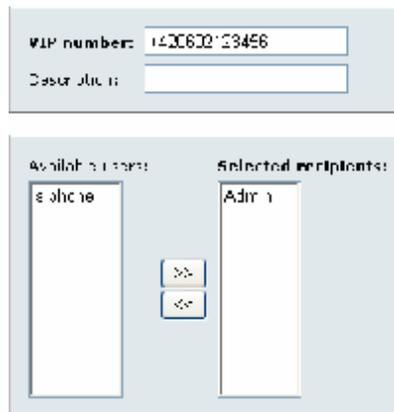


Fig. 60 – VIP Number Detail for SMS Delivery

The form contains the following fields:

- **VIP number** – the beginning or whole of the SMS CLIP.
- **Description** - any text that describes the meaning of an item.
- **Available users** – a list of users to be delivered SMS.
- **Selected recipients** – a list of selected gateway users to be delivered SMS meeting the specified criterion.

You can move user names from one list to another using handling arrows located between the lists. To do that, select the user with

the left-hand mouse button and then click on the arrow pointing at the other list.

10.3. SMS sending by SMTP/POP3

POP3 server: provides SMS downloading from VBE via e-mail.

Authorization goes through USERNAME and PASSWORD, which are used for registering to web-based interface of VBE.

Each SMS will be changed to e-mail in the Inbox directory, after successful authorization of VBE. There will be SMS sender tel. number@domain name and SMS receiver tel. number@ domain name (domain name could be set through web-based interface, according to fig 58 in the header of e-mail, subject of the message will be filled according to VBE settings (fig 58). You can use formatting strings in the text as well (%n – sender number, %u – receiver number, %d – date).

Message, which is already read, could be deleted from POP3 server. You can delete the message from VBE, or mark it as already read (i.e. it is still available in the Inbox directory through web-based interface, but POP3 server can not see it, fig. 58). It depends on the settings.

SMTP server: provides sending e-mail messages as SMS from VBE. USERNAME and PASSWORD for authorization are the same as the USERNAME and PASSWORD for web-based interface of VBE. Authentication could be set in web-based interface of VBE. Domain name must be equal to POP3 domain name (fig 58). Server awaits e-mail in format Text/Plain, Text/HTML, Multipart/Mixed or Multipart/Alternative. From the message in format Text/HTML, only the body of HTML message, without formatting, is put onto the SMS message. From messages Multipart/Mixed and Multipart/Alternative only the first part of message is used, enclosed between strings, mentioned in the header under name “boundary“.

Server supports the only messages in 7 bit or 16 bit format in encoding according to ISO-8859-1 (messages with the different encoding could be illegible after receiving message in form of SMS). SMTP server enables sender IP address control as well as authentication. With help of web-based interface, you can enable

or disable IP address control as well as set the list of allowed IP addresses.

SMTP server makes one or more SMS messages from incoming e-mail (maximum number of created SMS messages is set in web-based interface. It is possible to set from 0 to 100 messages, Fig. 58 in form F:<sender><CRLF>S:<e-mail subject (max. 10 characters)><CRLF>e-mail body. Messages with length of 160 characters are sent subsequently to receiver via Admin account (already sent SMS messages are available in SENT directory of user Admin).

Set IP address

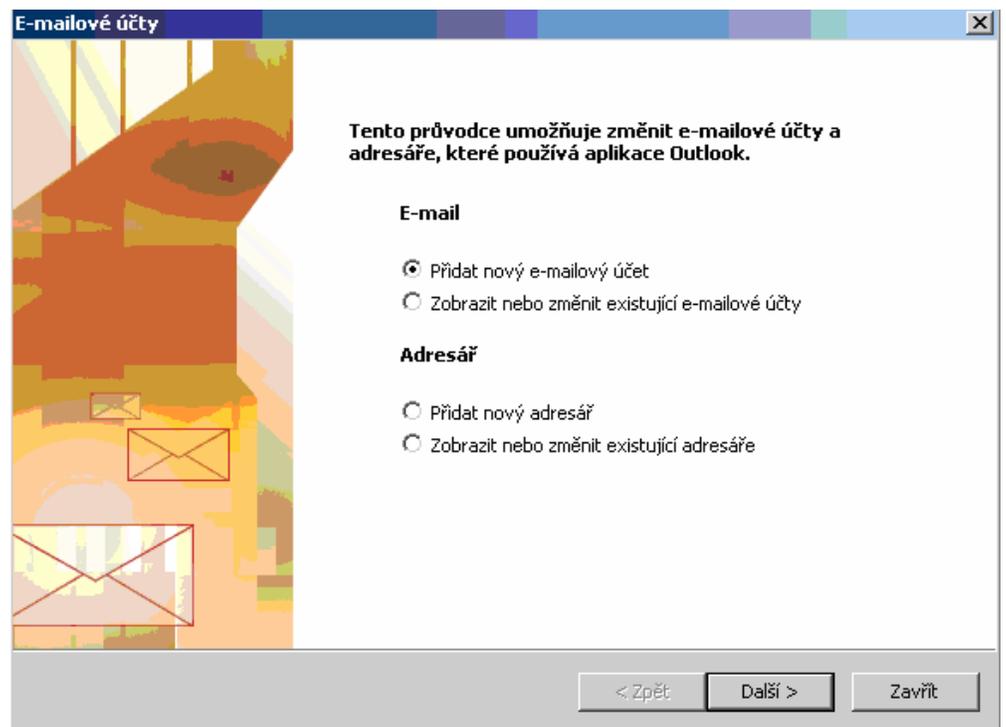
IP address/Mask	Description			
192.168.2.50/16	My office			<input type="checkbox"/>
50.30.48.1/32	My home			<input type="checkbox"/>
192.168.10.45/13	Public			<input type="checkbox"/>

Add IP address | Remove selected

The list of allowed IP addresses for VBE SMTP server

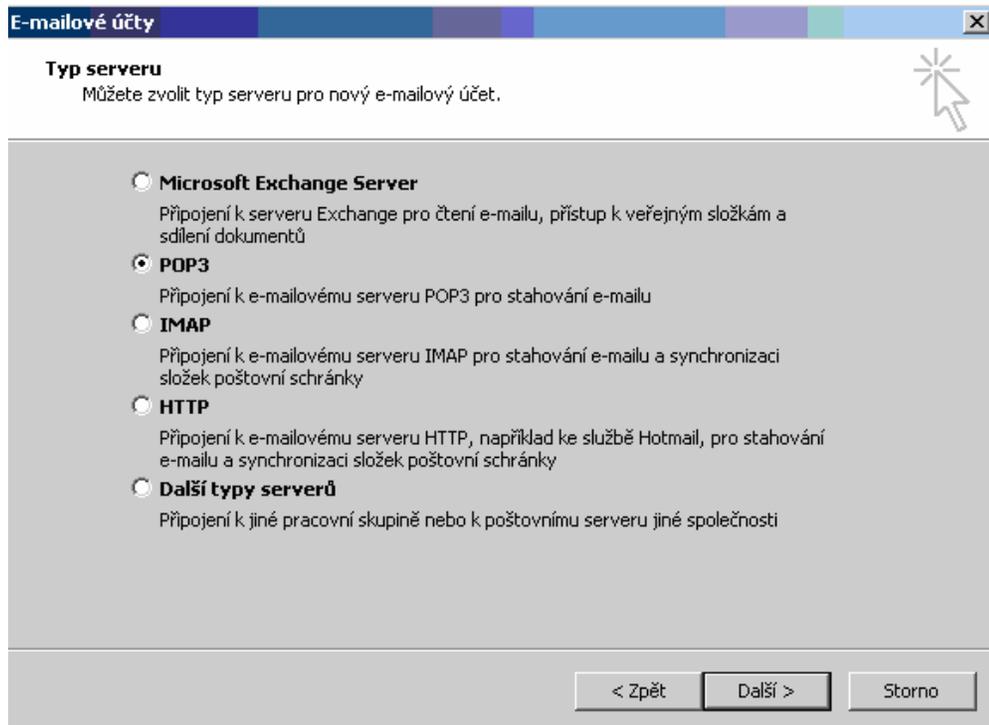
Example: Configuration of Microsoft Office Outlook 2003 for use with POP3 server

Choose the item Tools from upper menu, then choose E-mail accounts inside this menu. E-mail accounts window opens immediately (Pict. 3). Point out the item Add new e-mail account, and then click on Next.



Adding the new e-mail account

You have to choose type of server in the next opened window. Point out the item POP3 and click on Next button.

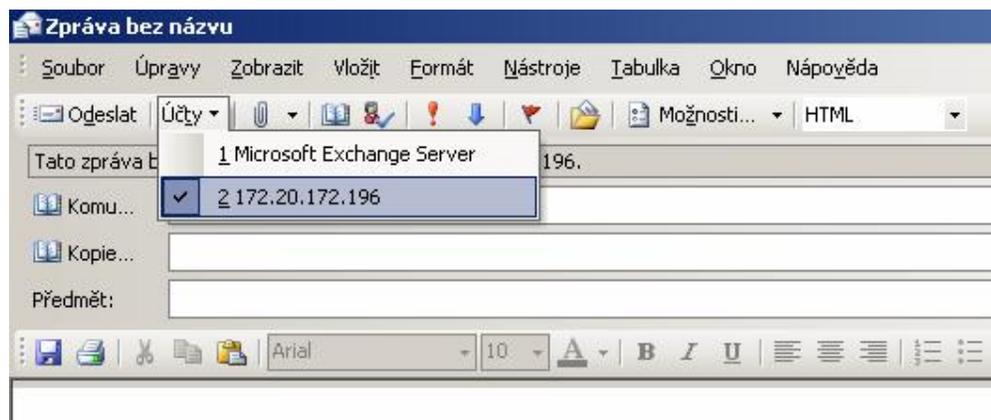


Selection of the server type

New window opens. Enter your name and e-mail address into fields Name and E-mail address. Enter IP address of your VBE into fields Incoming mail server (POP3) and Outgoing mail server (SMTP). Enter the valid user name and password of VBE into fields User name and Password – it is necessary to add the IP address of your computer into the list of allowed IP addresses via the web-based interface to VBE.

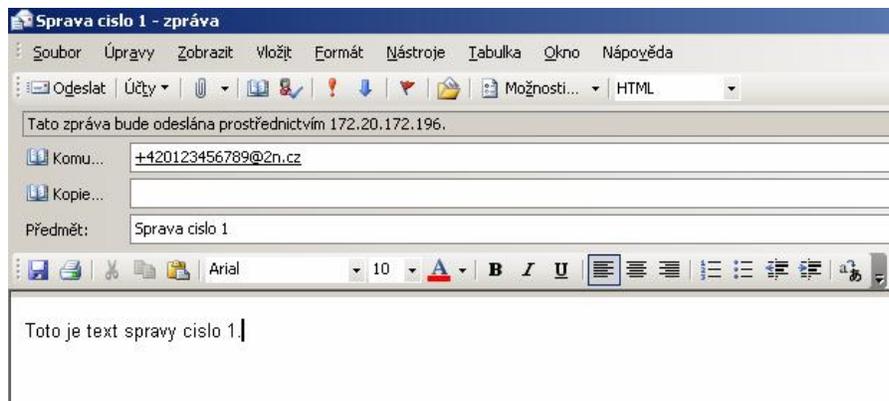
Settings of entries for SMTP/POP3 server

Sending SMS messages via VBE SMTP: in upper menu click on item New, the empty page with new e-mail opens. Click on item Accounts, choose SMTP/POP3 account of your VBE (Pict. 6)



Setting SMTP/POP3 account for new message

Fill to the field To: for proper sending of the e-mail you have to know the domain name, previously set in your VBE, in case, that you enter an incorrect domain name, VBE reject this message. Example: domain name is 2n.cz, receiver telephone number is +420 123 456 789, in field To make entries as follows: +420123456789@2n.cz. Fill in the field Subject and then write the message. You can use national characters in text but supported encoding is UTF-8 only. Format of message must be Plain text.



Example of SMS message sent through VBE SMTP server

Click on Send button. If your message is sent correctly, it will appear in Outgoing messages. You can see already sent message in web-based interface of VBE, SMTP server send the messages through Admin account, i.e. you can see the message in the Sent directory after login to Admin account in VBE (tuck SMS, item SMS).

Receiving e-mails through VBE POP3: SMS message sent to the number of the sim card in VBE could be downloaded as the e-mail in two ways. Either you click on Send and receive button, or you open the menu bellow this button and chose item Incoming messages in the menu Only <IP_address of_VBE>. All the SMS messages, which are saved in VBE in logged in account, will be downloaded.

SECTION 11

GPRS

In this chapter we will describe, how to connect into the Internet by means of the GPRS technology via the 2N VoiceBlue Enterprise gateway.

Following figure illustrates the connection of local LAN network into the Internet by means of the 2N VoiceBlue Enterprise gateway. After successful dialling of the GPRS connection, the line will automatically route the local LAN network into the Internet.

button is changed to *Switch GPRS OFF* button. The dialling of GPRS connection is time demanding – therefore it is necessary to wait several seconds after pressing the *Switch GPRS ON* button.

11.2. Setting of computer

To enable connection to the Internet by means of the 2N VoiceBlue Enterprise gateway, it is necessary to correctly set the connection to the local network on the computer (resp. TCP/IP protocol) in such manner, that the *Output gateway* will be the IP address 2N VoiceBlue Enterprise. Further, it is necessary to set the *preferred and alternative DNS server* to IP address, which will be displayed after successful dialling of the GPRS connection or is communicated to you by your GPRS provider.

On following figure, there is displayed the setting of local network, in which the computer has the IP address 10.0.0.100, gateway 2N VoiceBlue Enterprise has IP address 10.0.0.1 and the GPRS provider has primary DNS server 62.77.73.11 and secondary GPRS server is 81.95.96.9.

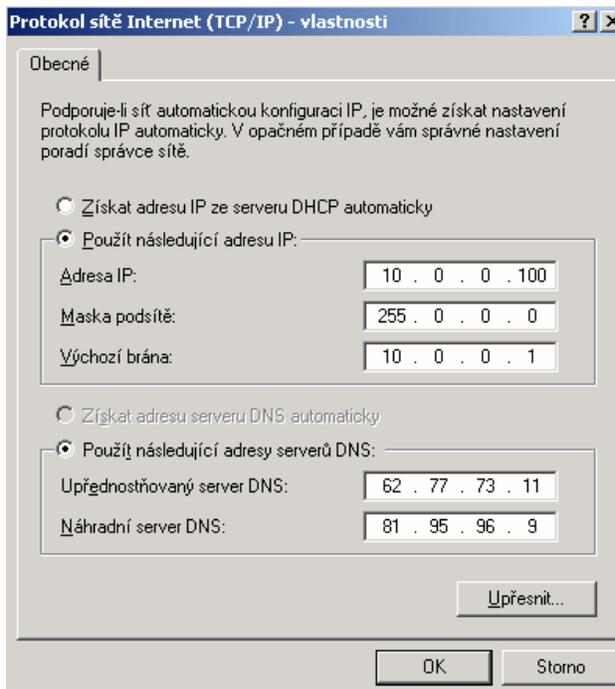


Fig. 63 – Setting of PC for connection into Internet by means of GPRS

SECTION 12

Monitoring and Diagnostics

This section deals with how the 2N VoiceBlue Enterprise gateway captures and records important operational data, accounting records and exceptional events. It also shows how to work with relevant data acquiring tools.

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

- Tariff drawing status;
- Currently made calls;
- Call accounting data;
- Operation journals;
- Downloading of journal files.

12.1. Tariff Drawing Status

If you use routing according to free minutes (refer to Subsection 7.3), you must have defined tariff rates pursuant to 7.6 and assign them to SIM cards. Now you definitely appreciate the possibility to assess the number of called/free minutes. To do this, open the *Tariffication* item in the *Statuses* menu in the *Statuses and records* bookmark. A tabulated survey gets displayed as shown in fig. 64.

Tariff name	Line ID	Day of restore call limit	Tariff rate [sec]	Free min./SMS	Called min./SMS	Last time used	Last day of restore call limit
SIM#-1	SIM - 13412039014014014	1	10:50 no 1	JUL20	JUL20	None	JUL20

Fig. 64 – Tariff Drawing Statuses

The columns include:

- **Tariff name** – the number under which you created the tariff.
- **Line ID** – the SIM card number to which the tariff is assigned. This number is usually printed on the card next to contacts.
- **Day of restore call limit** – The counters of called minutes and sent SMS are reset on this day every month.
- **Tariff rate [sec]** – a tariffication scheme. It consists of two values – one for each time zone. The number of time units and number of seconds per time unit are displayed for the first time zone. Information on the tariffication unit length is displayed only for the second call part.
- **Free minutes/SMS** – number of free minutes or SMS per month.
- **Called minutes/SMS** – number of called minutes or sent SMS after the last reset.
- **Last time used** – time of the last call established or SMS sent via the particular SIM card.
- **Last day of restore call limit** – The counters of called minutes and sent SMS will be reset on this day next time.

By clicking on the tariff name you get the tariff detail showing basically the same tariff information, yet arranged more clearly.

12.2. Currently Made Calls

To display the survey of currently made calls select the *Calls* item in the *Statuses* menu in the *Statuses and records* bookmark.

Od	Jméno	ID linky	Do	Jméno	ID linky	Trvání	Stav
4444	Jakub's testing phone	SIP - 12	+420602123456		11 - 11	0:00:10	CONNECTED

Fig. 65 – Survey of Calls Currently Made Via Gateway

The columns include:

- **From** – the calling number.
- **Name** – the calling line name. It is completed only if this function is supported by the call-initiating IP telephone.
- **Line ID** – the line via which the call came to the gateway.
- **To** – the called number.
- **Name** – the called line name. It is completed only if this function is supported by the call-initiating IP telephone.
- **Line ID** – the line via which the call left the gateway.
- **Duration** – call duration time.
- **Status** – connection status.

12.3. Call Accounting Records

Records on accomplished calls are saved into files in the text format. The file that the gateway currently saves call data into is called "current.acc". At the end of a predefined period of time, the file is renamed automatically and the gateway starts saving new call records into an empty file. The names of the closed files correspond with the respective period of time.

To display the list of saved call data files open the *Call accounting* menu in the *Statuses and records* bookmark.

Report name	Size	Last changed
current.acc	0 K B	2005/03/21 08:36:15

Fig. 66 – List of Call Record Files

The columns include:

- **Report name** – record file name.
- **Size** – file size in bytes.
- **Last changed** – date and time of the last file update.

Each file line relates to one call. For an example of a record see the frame below.

```
** 30.03.05/08:35:57 O-OK CAU-000 gsm/1 0:08 000:10 123456789 +420602123456 4444
8942020390514319884
```

The record consists of fields separated with spaces:

- ****** – initial record sequence.
- **30.03.05/08:35:57** – date and time of call arrival.
- **O-OK** – call type and status. The leading character indicates the call direction. *I* means calls from a GSM network and *O* calls to a GSM network. The two characters following the hyphen inform how the call was terminated. *OK* means correctly, *FD* indicates call disconnection due to an error.
- **CAU-000** – call termination code (errors). The values are as follows:

Code	Meaning
CAU-000	Normal end
CAU-001	Internal protocol stack error
CAU-002	Protocol error
CAU-003	Connection transferred to another application
CAU-004	No free line
CAU-005	Busy line
CAU-006	Rejected call
CAU-007	Called line does not answer

- **gsm/1** – module number via which the call was made.
- **0:08** – call establishing time. The period of time between receipt of the call by the gateway and the moment of line seizure by the called party.
- **000:10** – call duration (from line seizure by the called party).
- **123456789** – gateway identification – serial number (optional).
- **+420602123456** – called number.
- **4444** – calling number.
- **8942020390514319884** – SIM card identification.

Call Record Saving Setting

There is a *Change of accounting setting* link to the right of the list of call record files. Click on it to open a call accounting setting form.

The screenshot shows a web-based form titled "Call Accounting Setting". It is organized into four main sections:

- Accounting periods:** Contains five radio button options: "Day", "Week", "Month", "Quarters", and "Year". The "Month" option is currently selected.
- Report options:** Includes a dropdown menu for "SIM id" with "SCID" selected, and a checked checkbox for "Gateway S/No".
- To send reports by e-mail:** Features two text input fields: "Recipient(s)" and "Message subject".
- To delete reports:** Contains two radio button options: "Never" and "On demand". The "On demand" option is selected. To the right of the "On demand" radio button is a text input field and a dropdown menu set to "Months".

Fig. 67 – Call Accounting Setting

The form in fig. 67 contains the following fields:

- **Accounting period** – the period of time after which the call record file is closed and renamed. If you choose a weekly interval, enter the starting day.
- **SIM Id** – defines whether the call record should contain the Id of the SIM card used for the call. There are two ways of identification: SCID (SIM Card Identification Number) – the card number printed on the card next to contacts, or IMSI (International Mobile Subscriber Identity).
- **Gateway S/No** – an indicator saying whether the accounting record should include the gateway Id (serial number).
- **Recipient(s)** – It is possible to send a closed file by e-mail at the end of the accounting period. There may be more addressees than one. Their addresses are separated with semi-colons. The call record file is attached to the message.

- **Message subject** – the subject of messages sent at the end of the accounting period, refer to the parameter above.
- **Delete reports** – The gateway capacity is limited and so it is advisable to delete old accounting records. Define for how long the files should be kept.

To save the changes click on the *Modify* link located to the right of the form. There is an *Reset* link next to the *Modify* link used for restoring values.

12.4. Operation Journals

Events that come into being during the gateway operation are recorded in operation journals for subsequent error analysis purposes. The events are divided into five groups and saved into separate files:

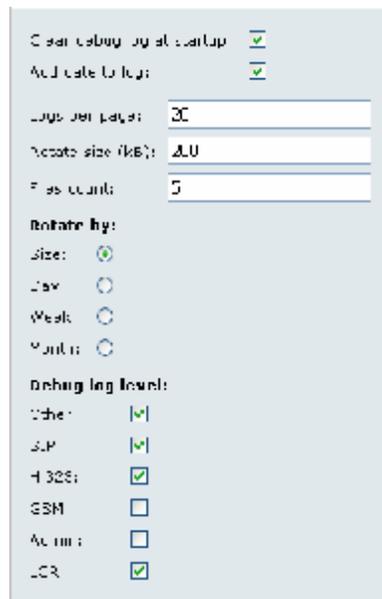
- **Warning** – events that should not occur normally but their presence has no effect on the gateway behaviour.
- **Errors** – errors that limit or jeopardise the gateway function. Their occurrence usually signals an approaching fatal failure.
- **GSM AT** – a gateway - GSM module communication record. This journal includes all AT commands sent to the GSM modules by the gateway and responses sent back.
- **Debug** – extended information used for development/diagnostic purposes. The debug information categories to be recorded are selected in the journal setting menu.

To view the contents of the journal files use the *Records* menu in the *Statuses and records* bookmark. Each journal type has an item of its own in this menu. If you click on it, the last file lines get displayed and you can change the position of the displayed slice using navigation buttons *First*, *Preceding*, *Following* and *Last*. To control the size of the slice to be displayed use the journal setting menu.

The *Delete* key is used for manual deleting of the journal file contents. Otherwise the files are cleared automatically according to specified criteria, see below.

Operation Journal Setting

You can set the behaviour of the operation journals in the *Statuses and records* bookmark. Click on the *Records* menu to get the necessary form, see the following figure.



The screenshot shows a configuration form for operation journals. It contains the following elements:

- Checkboxes for "Clear debug log at startup" and "Add date to log", both of which are checked.
- Text input fields for "Logs per page" (value: 30), "Rotate size (kB)" (value: 100), and "Files count" (value: 5).
- A "Rotate by:" section with radio buttons for "Size" (selected), "Day", "Week", and "Month".
- A "Debug log levels:" section with checkboxes for "Info" (checked), "SUP" (checked), "INFO" (checked), "WARN" (unchecked), "ERROR" (unchecked), and "FATAL" (checked).

Fig. 68 – Operation Journal Settings

The form in fig. 68 contains the following fields:

- **Clear debug log at startup** – defines whether the tuning journal file should be deleted automatically upon the gateway power up.
- **Add date to log** – defines whether the gateway should add a time stamp to the events recorded.
- **Logs per page** – the number of lines to be displayed at the same time on one journal page.
- **Rotate size (kB)** – If the "rotating" criterion is set at the value *Size*, see below, the journal files shall "rotated off" when the defined size is exceeded (in kilobytes).
- **Files count** – the maximum number of "rotated-off" files of each type that are stored in the gateway memory. The earlier ones are deleted automatically.

- **Rotate by** – the "rotation" criterion. The journal files are closed and renamed when a certain condition is satisfied, a new file is used for event recording. The files can be closed at the end of a defined period of time (day, week, month) or when the specified size is achieved.
- **Debug log level** – defines which tuning record categories shall be recorded.

12.5. Journal File Loading

The viewing of the journal files using a web browser as described in the preceding subsection is ineffective for complicated retrieval of events that happened in the distant past. In this case, it is more advantageous to download the journal files to your PC and view them using more sophisticated tools.

To download packed journal files click on the *Download...* key on the page that appears when you click on the *Download records* item in the *Statuses and records* bookmark.

SECTION 13

Firmware Update

In this chapter we will describe recording of the licence file into the 2N VoiceBlue Enterprise gateway and procedure of the firmware updating.

13.1. Licence file

The licence file serves for switching on the purchased services providing by the 2N VoiceBlue Enterprise gateway. The new purchased gateway 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 gateway is realised by clicking to the *Update* item in the *Setting* menu, and then to the *Upload licence* item. Then, the dialog for adding of the licence file will display on the



screen, as illustrated on fig. 69.



Fig. 69 – Adding of licence file

Select the path to the licence file by pressing the *Browse* button and record the licence file into the 2N VoiceBlue Enterprise gateway by pressing the *Upload* button.

13.2. Firmware updating

The firmware updating serves for exchange of the firmware of the 2N VoiceBlue Enterprise gateway. 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 gateway may contain, or for recording of new versions of software, which contain new functions.

The file with the firmware is named *root.tgz*, and should have up to 10 MB. The firmware updating lasts cca 5-10 minutes, when the firmware is recorded into the gateway 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 *Update firmware* item. Then, the dialogue for adding of the licence file is displayed, as illustrated on fig. 70.



Fig. 70 – Updating of firmware

Select the path to the file containing the firmware by pressing the *Browse* button and record the new firmware into the 2N VoiceBlue Enterprise gateway by pressing the *Update* button.

Before updating process is recommended to reboot 2N VoiceBlue Enterprise.

SECTION 14

Technical Parameters

VoIP

Signalling standards	SIP, H.323
Number of channels	4
Codecs	G.711 PCM (64 kbps) G.729A CS-ACELP (8 kbps) G.723.1 MP-MLQ/ACELP 6.3 kbps (optional)
PCM quantizing	A-law / μ -law

GSM

Mobile network type	GSM 900 phase II EGSM 1800 MHz
SIM cards	Plug-in 3V ("small")
Transmission power	2W (1W)
Receiver sensitivity	-104 dBm

Antenna

Frequency	900 / 1800 MHz
Impedance	50 Ω
Maximum output power	2W
Antenna connector type	SMA

Power Supply

Adapter	230V \pm 10%, 50/60Hz / 12V DC
Supply voltage	12V / 2A
Supply connector	DC jack, 2.1 mm
Power input	max. 20W

Interface

Serial	RJ- connector, see 115200 bps, 8 bits, 1 stop bit, no parity, no flow control
MMC	
Ethernet	RJ-45 10/100 Base T
Protocols	IP, ICMP, TCP, UDP, DHCP, RTP, TELNET, HTTP, HTTPS

Operational Conditions

Temperature	0°C ÷ 40°C
Relative humidity	5 ÷ 95% non-condensing

Others

Dimensions (without connectors)	250×150×50 mm
CE certification	EN 60 950:2000, EN 55 024, EN 55 022 Class B

