

V O I P G S M G A T E

BlueGate SIP



Quick Installation guide V1.1

Technical parametres

Dimensions	133 x 233 x 60 mm
Weight	850 g
Operating position	various
Operating condition	temperature: +5°C ÷ +40° C humidity: 10% ÷ 80% při 30° C
Power supply	9-15V ss or 8-12Vst, 1.5 A

VoIP

- Ethernet – 10/100Mb with standard BaseT and 100BaseTx, connector RJ45
- SIP connection P2P or IP PBX net unit – tested with Cisco Call Manager, Alcatel OMNI PCX, Asterisk, Nexspan, Panasonic...
- 2 VoIP channels (2 IP adresses)
- Codecs: G711u, G711a, G726, GSM
- VAD (Echo cancellation)
- Protocols: IP, TCP, UDP, http, TELNET, SIP, RTP
- Web server for remote management – BOA
- Software for internet phone (VoIP) – Linphone
- WEB – firmware upgrade
- WEB – interface for parametres programming IPDP
- Connection with DTMF dial in
- connection to 2 operators after time out (adjustable) for direct dialing in
- transmit of calling part number (CLIP)
- incoming call restriction from GSM network
- outgoing call restriction to GSM network
- Priority connection through either the 1st or the 2nd GSM module (LCR)
- Smart Call back – automatic incoming calls routing up CLIP
- Direct acces – assign of IP adress to GSM channels
- Echo canceller – switching ON/OFF
- volba poskytovatele GSM služeb samostatně pro každý GSM modul
- OGM module for recording voice message*
- PIN protection of SIM card

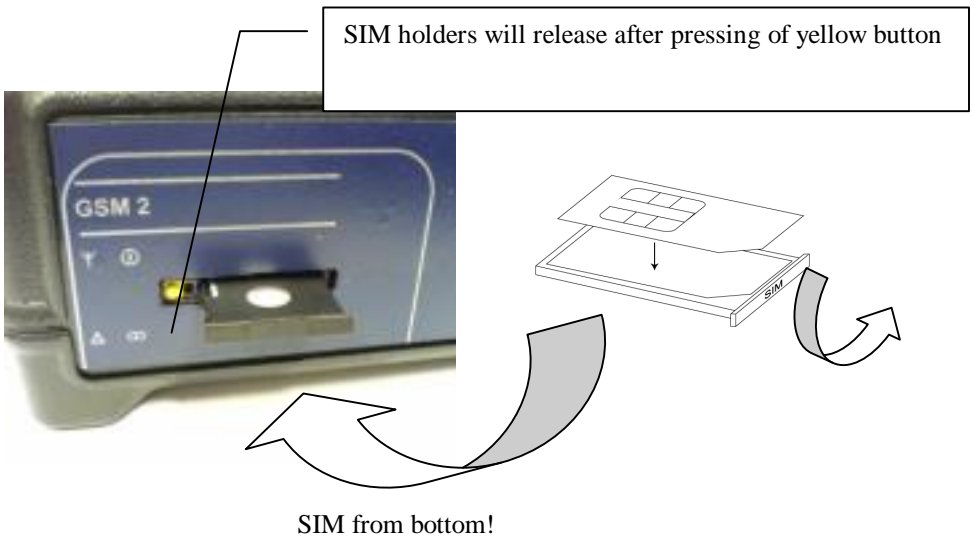
GSM

- GSM 900 (class 4 – 2 W)
- GSM 1800 (class 1 – 1 W)
- Antenna connector SME/SMA, 50Ω
- SIM card: 3/1,8 V
- 2 GSM channels

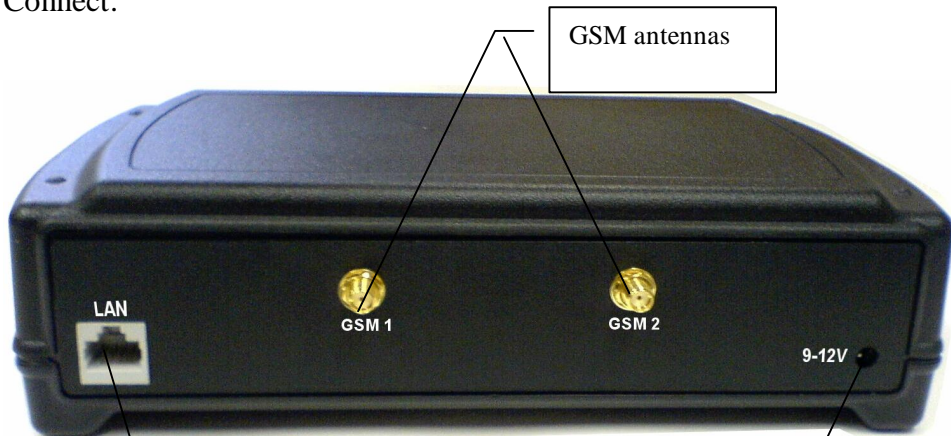
Installation

Insert SIM cards.

Before SIM card inserting check their status in mobile phone (starting with PIN or without PIN). *When you want use start with PIN you have to firstly programm this PIN to Blue Gate SIP. Without this setting the Blue Gate SIP will not registrate to GSM network!*



Connect:



GSM antennas

LAN

GSM 1

GSM 2

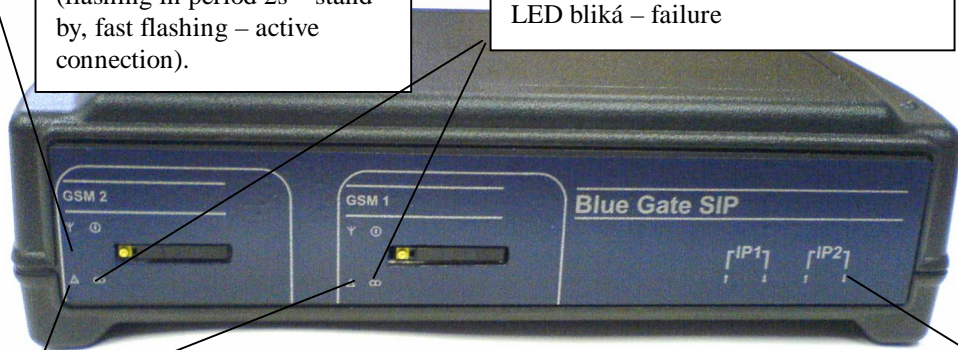
9-12V

LAN cable

Power supply –
connect at end

Green LED during operation indicates GSM modules powering. The Red LED indicates GSM net operation (flashing in period 2s – stand by, fast flashing – active connection).

Error status indication on channels (RED):
LED is off – stand by
LED bliká – failure



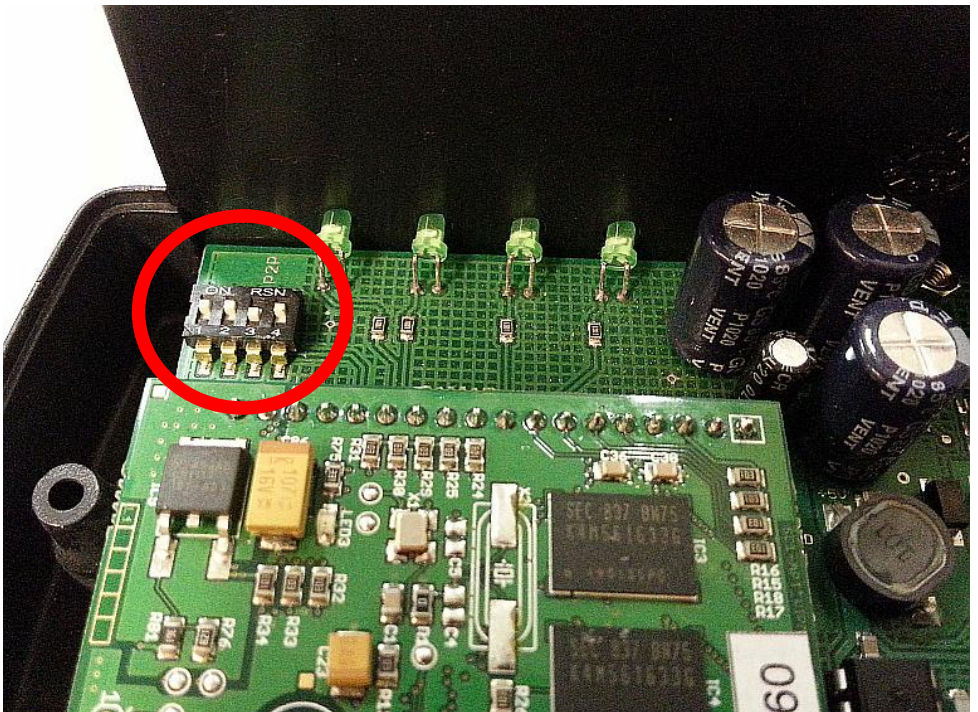
GSM1,2 operation (green):
LED flashing – modul registrate to
GSM net, stand by
LED lighting – connection (call)

Incoming and outgoing call via
individual IP channels

DIP switch

The program reads the DIP Switch state after restart. The DIP switch has 4 switches.

1. switch is for default IP address. For normal Gateway operation set this switch to ON position.
2. switch is for default VoIP parameters. For normal Gateway operation set this switch to ON position
3. switch for P2P or SIP Server mode. For SIP Server mode set the switch to off position, for P2P set the switch to on position
4. without function

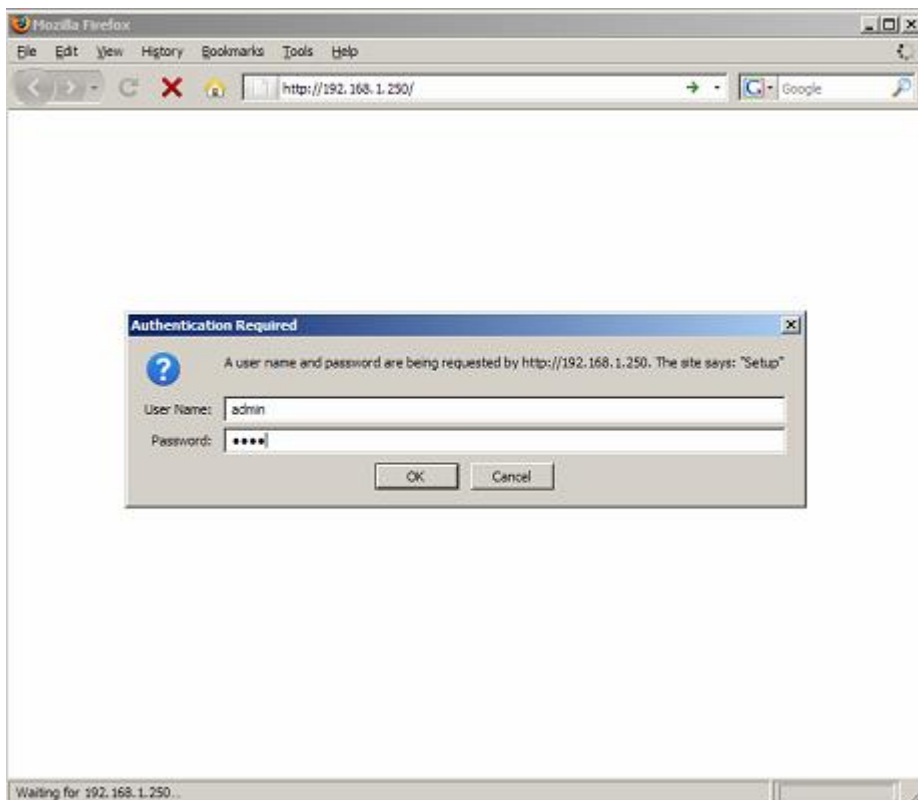


On PC connected to LAN from which you can control connected Gateway run various web browser. Insert IP address of gate (default 192.168.1.250) and you can continue in parametres setting.

STEP 1 - choosing a mode and login

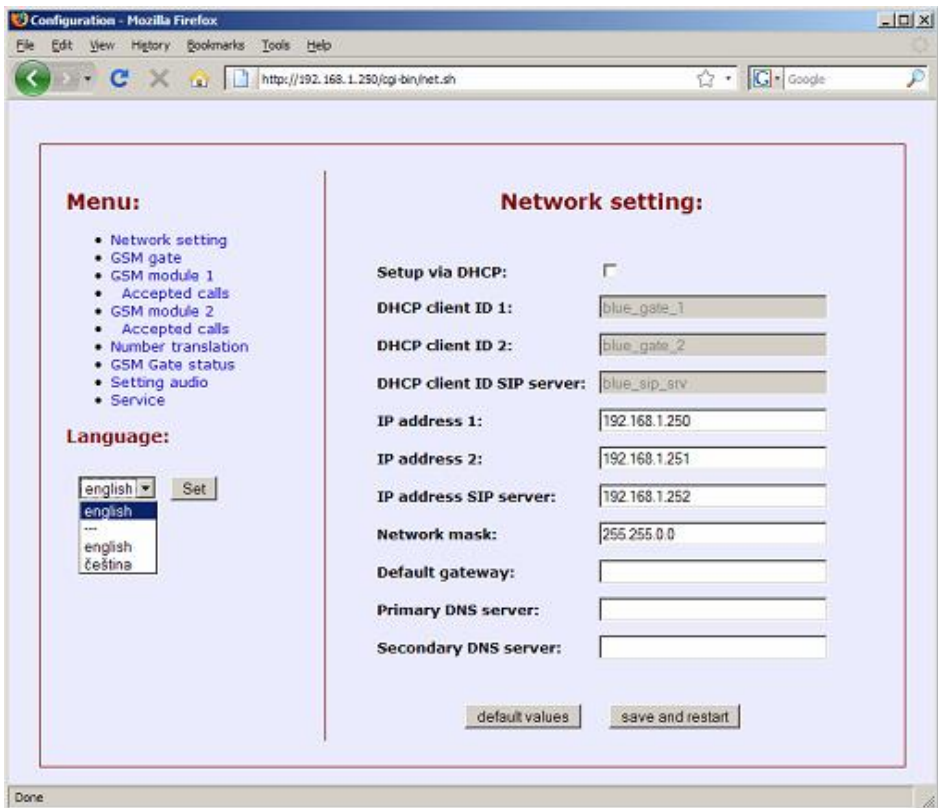
It is important to choose a gate mode first. The gate can work in the PeerToPeer mode or SIP server mode. The mode setting can be made by a relevant switch. In the SIP server mode is possible to choose SIP server internal or external. It can be set in a configuration interface of the gate.

In your web browser enter IP address of the gate, default is **192.168.1.250**. Enter user name and password. User name is **admin**, default password is **1234**.



STEP 2 - language option:

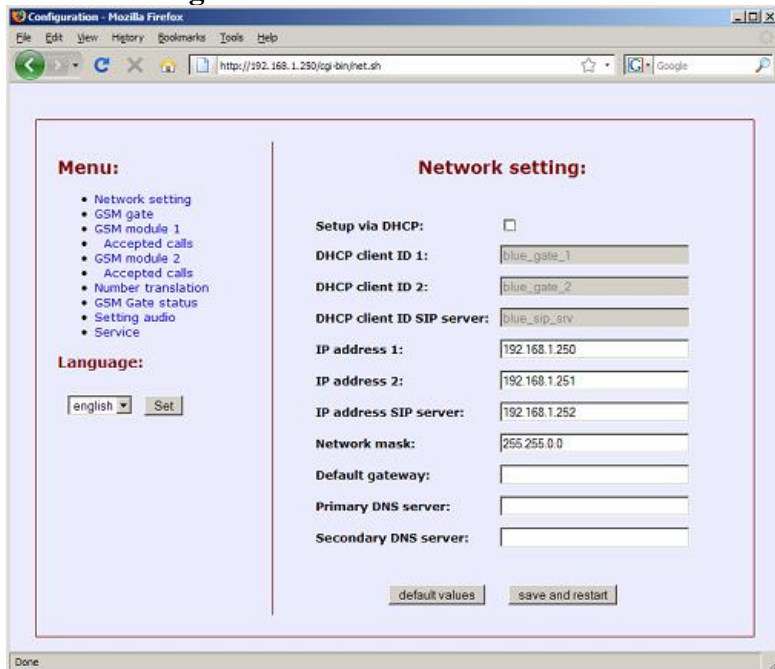
Language setting can be made in a menu on the left panel.



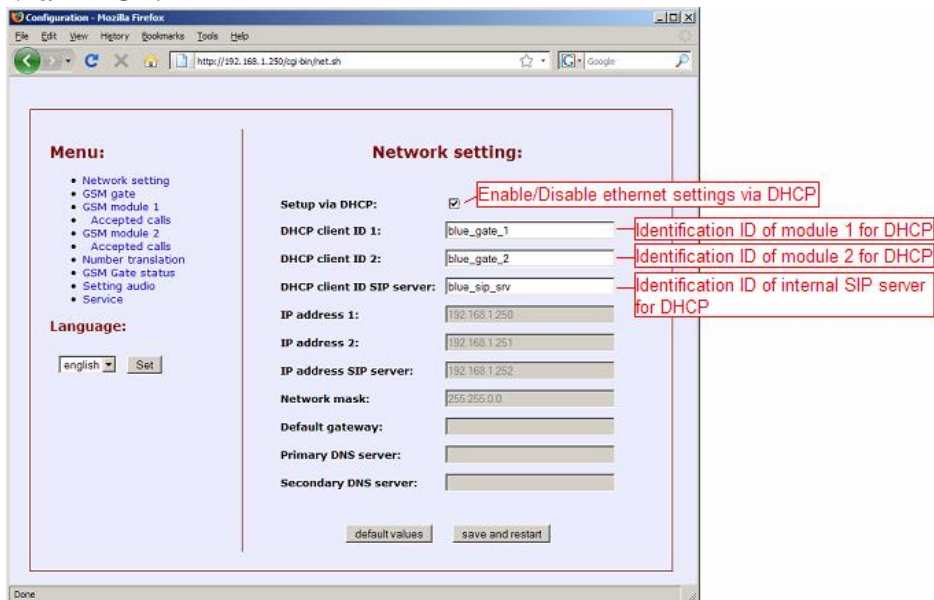
STEP 3 - Network settings:

Network settings are located in the **Network setting** menu item. It is possible to use DHCP service or you can enter IP addresses manually. After making changes click on a **save and restart** button.

Manual configuration:



Via DHCP:



STEP 4 - GSM gate settings

GSM gate settings can be made in a section **GSM module 1** and **GSM module 2**. It is important to set a customer number, which will be dialed by the modul at an incoming calling. Next, it is important to set SIM card PIN, if it is required. If not, write **0000**. After making changes click on a **save changes** button.

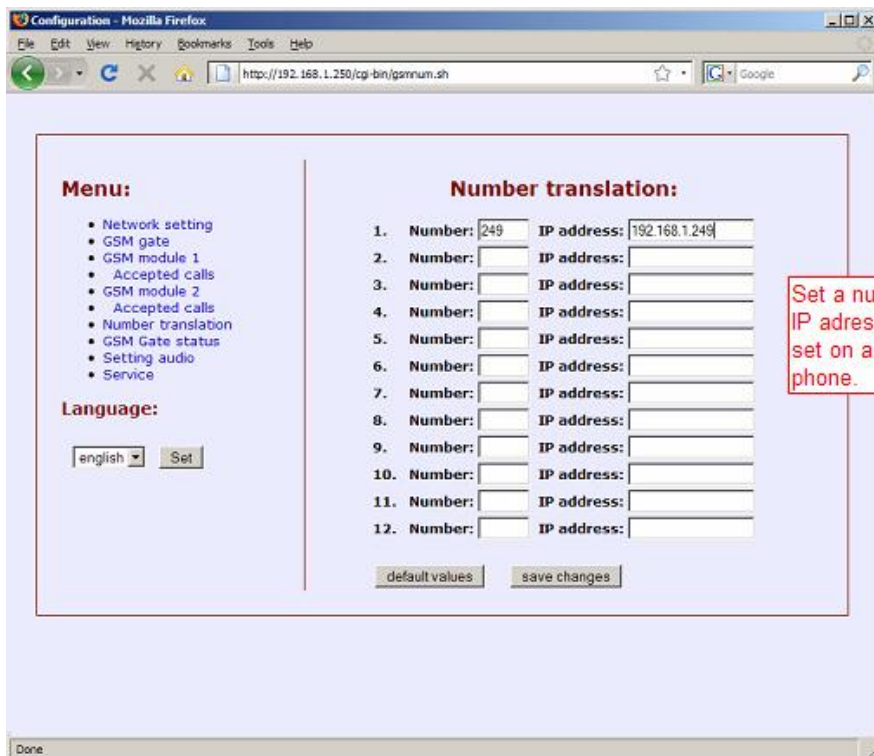
The screenshot shows a web browser window titled "Configuration - Mozilla Firefox" with the address bar displaying "http://192.168.1.250/cgi-bin/gsm1.sh". The main content area is divided into two sections:

- Menu:** A list of navigation links including Network setting, GSM gate, GSM module 1, GSM module 2, Accepted calls, Number translation, GSM Gate status, Setting audio, and Service.
- Language:** A dropdown menu set to "english" with a "Set" button.
- GSM Gate - module 1:** A configuration panel with the following fields:
 - PIN:** Input field containing "0000". A red callout box labeled "SIM card PIN" points to this field.
 - Subscriber number:** Input field containing "249". A red callout box labeled "Set subscriber number" points to this field.
 - Volume GSM:** Input field containing "1".
 - Volume ISDN:** Input field containing "1".
 - Incoming calls:** Checkmark
 - Outgoing calls:** Checkmark
 - Call progress tones:**
 - Redirection to GSM 2:**
 - Smart callback:**
 - O:**
 - CLIR:**

At the bottom of the configuration panel, there are two buttons: "default values" and "save changes".

STEP 5 - GSM gate settings in the P2P mode

The gate should be set to the P2P mode, viz. **STEP 1**. In this mode it is important to set a translation of numbers into IP addresses. This setting can be made in the **Number translation** section. Item „Number“ is a phone number, IP address is an address of the selected phone. After making changes use the **save changes** button.



The screenshot shows a web browser window titled "Configuration - Mozilla Firefox" with the address bar displaying "http://192.168.1.250/cgi-bin/gsmnum.sh". The page content is divided into two main sections: "Menu:" and "Number translation:".

Menu:

- Network setting
- GSM gate
- GSM module 1
- Accepted calls
- GSM module 2
- Accepted calls
- Number translation
- GSM Gate status
- Setting audio
- Service

Language:

english

Number translation:

1.	Number:	<input type="text" value="249"/>	IP address:	<input type="text" value="192.168.1.249"/>
2.	Number:	<input type="text"/>	IP address:	<input type="text"/>
3.	Number:	<input type="text"/>	IP address:	<input type="text"/>
4.	Number:	<input type="text"/>	IP address:	<input type="text"/>
5.	Number:	<input type="text"/>	IP address:	<input type="text"/>
6.	Number:	<input type="text"/>	IP address:	<input type="text"/>
7.	Number:	<input type="text"/>	IP address:	<input type="text"/>
8.	Number:	<input type="text"/>	IP address:	<input type="text"/>
9.	Number:	<input type="text"/>	IP address:	<input type="text"/>
10.	Number:	<input type="text"/>	IP address:	<input type="text"/>
11.	Number:	<input type="text"/>	IP address:	<input type="text"/>
12.	Number:	<input type="text"/>	IP address:	<input type="text"/>

A red callout box on the right side of the page contains the text: "Set a number and IP address, which is set on a desired phone."

STEP 6 - GSM gate settings in the SIP server mode (internal)

A module should be set to a mode with SIP server, viz **STEP 1**. Activation, deactivation and a real settings of the internal SIP server can be made in the **SIP server** section. After making changes use button **save and restart**.

Menu:

- Network setting
- GSM gate
- GSM module 1
- Accepted calls
- GSM module 2
- Accepted calls
- GSM Gate status
- SIP parameters
- SIP server
- Setting audio
- Service

Language: english

SIP Server:

Enable SIP server: Activate/Deactivate Internal SIP server

Server name (realm): blue_sip_srv SIP server name

Prefix 1: 0 Prefix for calling through module 1

Prefix 2: 9 Prefix for calling through module 2

	Number:	Password:
1:	249	249249
2:		
3:		
4:		
5:		
6:		
7:		
8:		
9:		
10:		

Table of enabled phones. Set a user name and password, which is set on a desired phone.

STEP 7 - GSM gate settings in the SIP server mode (external)

The module should be set to a mode with SIP server, viz **STEP 1**. Next, internal SIP server must be deactivated, viz **STEP 5**. Settings can be made in a **SIP parameters** section. It is important to set IP address or the SIP server name, possibly username and password for every module. It depends on the SIP server settings. After making changes use the **save changes** button.

The screenshot shows a web browser window titled "Configuration - Mozilla Firefox" with the address bar displaying "http://192.168.1.250/cgi-bin/sip.sh". The main content area is divided into two sections: "Menu:" and "SIP parameters:".

Menu:

- Network setting
- GSM gate
- GSM module 1
- Accepted calls
- GSM module 2
- Accepted calls
- GSM Gate status
- SIP parameters
- SIP server
- Setting audio
- Service

Language:

english

SIP parameters:

SIP proxy server 1 Address: Port: 5060

Account module 1 Name: Password:

SIP proxy server 2 Address: Port: 5060

Account module 2 Name: Password:

Two red callout boxes point to the "Address" input fields for "SIP proxy server 1" and "SIP proxy server 2". The first callout contains the text "Name or IP address of external SIP server for module 1". The second callout contains the text "Name or IP address of external SIP server for module 2".

2. Description of GSM Parameters

Menu: <ul style="list-style-type: none">• Network setting• GSM gate• GSM module 1<ul style="list-style-type: none">• Accepted calls• GSM module 2<ul style="list-style-type: none">• Accepted calls• GSM Gate status• SIP server• Setting audio• Service Language: <input type="text" value="english"/> <input type="button" value="Set"/>	GSM Gate: DISA: <input type="text" value="0"/> OGM: <input type="checkbox"/> rec: <input type="checkbox"/> Wait: <input type="text" value="0"/> Erase CLIP: <input type="text" value="0"/> Preanswer: <input checked="" type="checkbox"/> Modules directly: <input type="checkbox"/> Echo canceller: <input checked="" type="checkbox"/> Add zero before CLIP: <input type="checkbox"/> Start with module 1: <input type="checkbox"/> <input type="button" value="default values"/> <input type="button" value="save changes"/>
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DISA

The incoming calls can be handled in two ways:

All calls can be automatically connected to a selected extension in the connected exchange. Connection takes place in accordance with the selected Subscriber number or direct dialling numbers of the operator separately for each channel. In this way, it is even possible to distinguish the calls incoming to individual telephone numbers of the GSM channels of the gate - the calls incoming to the telephone number of the first channel will be connected to the direct dialling code 11, incoming calls to the number of the other channel e.g. to the direct dialling code 12.

If a number of digits in the direct dialling code are being set, the number will be higher than zero (max. 4 - **Caution: the number should correspond with the number of digits of the direct dialling code of the exchange and selected Subscriber number of individual GSM modules!**), the callers may themselves dial directly using tone selection by preset number of digits to the required extension.

OGM

The gate can be provided additionally with a board with a digital recorder of “messages” for individual GSM channels. When collecting an incoming call through direct dialling, these messages are always replayed.

rec

The new OGM modul doesn't use this parameter.

Wait

The automatic operator waits for a pre-selected number of seconds for direct dialling code selection. If it is not selected, the pre-selected number of the operator (according to a channel the call is incoming from) is dialled again.

Erase Clip

The gate supports transfer of CLIP from the GSM network , i.e. to a telephone connected to an extension (if the device allows it). Here it is advantageous to use the parameter for deleting the initial digits from CLIP (e.g. deleting +42 in order not to “confuse” LCR exchanges when calling back to the displayed number).

Modules directly

By activation this feature you disabled LCR (permitted direction). The IP adress 1 is automatically assigned to 1 SIM card and IP adress 2 is assigned to 2 SIM card. The LCR you can do directly in the SIP server.

Echo canceller

This feature restrict ECHO during transffer between digital and analog telephone systems. (2wire - 4 wire). When on SIP server are connected only

digital (4 wire) lines (digital phones) you can by activation this parametr (switch OFF ECHO canceler) reach better voice quality.

Add zero before CLIP

It adds automatically 0 before each incoming number.

Start with module 1

The Gate start testing LCR in Gate with module 1.

Menu: <ul style="list-style-type: none">• Network setting• GSM gate• GSM module 1• Accepted calls• GSM module 2• Accepted calls• GSM Gate status• SIP server• Setting audio• Service Language: <input type="text" value="english"/> <input type="button" value="Set"/>	GSM Gate - module 1: PIN: <input type="text" value="0000"/> Subscriber number: <input type="text" value="102"/> Volume GSM: <input type="text" value="1"/> Volume ISDN: <input type="text" value="1"/> Incoming calls: <input checked="" type="checkbox"/> Outgoing calls: <input checked="" type="checkbox"/> Call progress tones: <input type="checkbox"/> Redirection to GSM 2: <input type="checkbox"/> Smart callback: <input type="checkbox"/> O: <input type="checkbox"/> CLIR: <input type="checkbox"/> <input type="button" value="default values"/> <input type="button" value="save changes"/>
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PIN

Set PIN up SIM card in modul. When you use SIM card without PIN then this window is not used.

Subscriber number

Is designed for storing automatic dial in - operator - particularly for each GSM modul (more via. DISA Setting).

Volume GSM

The Volume GSM allows adjusting call volume in the outgoing direction. Volume can be set in degrees 1 to 7.

Volume ISDN

The Volume ISDN allows adjusting call volume in the incoming direction. Volume can be set in degrees 1 to 4.

Incoming calls

Permission of incoming calls.

Outgoing calls

Permission of outgoing calls.

Call progress tones

Switching ON/OFF tones signalling searching of called part in GSM net.

Redirection to GSM 2

In case of marking the "Re-route in GSM2" (GSM1) field, "overflow" of the enabled directions (selection see below) in the adjacent channel is allowed. If the channel with the enabled direction is already busy calling, the gate will not refuse the call, but it will utilize the other channel (even if the direction is not on the list of enabled directions for this channel).

Smart callback

When you use this parametr then all outgoing calls which has been missed or refused are stored. It stores PBX extension which made the call + called number. When called part calling back then the call is automatically routed to extension which made the call. When connection was succesfull data are erased from memory.

0

It adds automatically 0 before each outgoing dialled number.

clir

Switch OFF outgoing CLIP.

<p>Menu:</p> <ul style="list-style-type: none">• Network setting• GSM gate• GSM module 1• Accepted calls• GSM module 2• Accepted calls• GSM Gate status• SIP server• Setting audio• Service <p>Language:</p> <p>english <input type="button" value="Set"/></p>	<p>Permitted calls - module 1:</p> <table><tr><td>0.</td><td>from:</td><td><input type="text"/></td><td>to:</td><td><input type="text"/></td></tr><tr><td>1.</td><td>from:</td><td><input type="text"/></td><td>to:</td><td><input type="text"/></td></tr><tr><td>2.</td><td>from:</td><td><input type="text"/></td><td>to:</td><td><input type="text"/></td></tr><tr><td>3.</td><td>from:</td><td><input type="text"/></td><td>to:</td><td><input type="text"/></td></tr><tr><td>4.</td><td>from:</td><td><input type="text"/></td><td>to:</td><td><input type="text"/></td></tr><tr><td>5.</td><td>from:</td><td><input type="text"/></td><td>to:</td><td><input type="text"/></td></tr><tr><td>6.</td><td>from:</td><td><input type="text"/></td><td>to:</td><td><input type="text"/></td></tr><tr><td>7.</td><td>from:</td><td><input type="text"/></td><td>to:</td><td><input type="text"/></td></tr><tr><td>8.</td><td>from:</td><td><input type="text"/></td><td>to:</td><td><input type="text"/></td></tr><tr><td>9.</td><td>from:</td><td><input type="text"/></td><td>to:</td><td><input type="text"/></td></tr><tr><td>10.</td><td>from:</td><td><input type="text"/></td><td>to:</td><td><input type="text"/></td></tr><tr><td>11.</td><td>from:</td><td><input type="text"/></td><td>to:</td><td><input type="text"/></td></tr></table> <p><input type="button" value="default values"/> <input type="button" value="save changes"/></p>	0.	from:	<input type="text"/>	to:	<input type="text"/>	1.	from:	<input type="text"/>	to:	<input type="text"/>	2.	from:	<input type="text"/>	to:	<input type="text"/>	3.	from:	<input type="text"/>	to:	<input type="text"/>	4.	from:	<input type="text"/>	to:	<input type="text"/>	5.	from:	<input type="text"/>	to:	<input type="text"/>	6.	from:	<input type="text"/>	to:	<input type="text"/>	7.	from:	<input type="text"/>	to:	<input type="text"/>	8.	from:	<input type="text"/>	to:	<input type="text"/>	9.	from:	<input type="text"/>	to:	<input type="text"/>	10.	from:	<input type="text"/>	to:	<input type="text"/>	11.	from:	<input type="text"/>	to:	<input type="text"/>
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Permitted calls

Up to 12 blocks of permitted calls can be selected in each GSM module. The table of the permitted calls is accessible after clicking on the Accepted calls. If the table is not filled in, all calls are enabled. 1- up to 8-digit numbers can be written in the first column of the table. The program substitutes any digits instead of not filled in digits from the right which checking the enabled calls. This means, instead of writing the numbers 601,602,...609, it is possible to write only 60. If the operator has three successive numbers differing from each other only by the last digit, this calls can be written as a group on a single line: the first column contains up to 8-digit number of the first call (“from”) as in the case mentioned above and the last digit of the last call is stored in the second column (i.e. the last digit of “to”). A group may be created only within the range of one order of the last position - i.e. from 0 to 9! By proper filling in the tables of both modules (one module e.g. T-Mobile code, the other module Orange code), it is possible to achieve automatic re-routing of the outgoing call in the module allowing cheaper calling through the given operator (LCR).

GSM Gate status

Menu:

- Network setting
- GSM gate
- GSM module 1
- Accepted calls
- GSM module 2
- Accepted calls
- GSM Gate status
- SIP server
- Setting audio
- Service

Language:

english

GSM Gate status:

	Module 1	Module 2
Frequency channel:	76	76
Signal strength:	41	43
BCCH strength:	-69	-67
Country code:	230	230
Network code:	01	01
Area code:	434E	434E
Cell ID:	053B	053B
IMSI:	230013100153408	230012200142897
GSM Firmware version:	VER3.B	

when click on GSM Gate status then is open the window where are mentioned all parametres about GSM modules.

- **Frequenci Channel :** determine channel number of GSM net - frequency on which is GSM modul connected to BTS. The channels 0 - 124 are certained for GSM 900, channels 512 - 885 for GSM1800.
- **Signal Strength, BCCH strenght:** signal power is one of the most important parametr for voice connection quality.
 - o -113 to -99 dBm: very bad signal - it is impossible to use any services via GSM network.
 - o -98 to -83 dBm: bad signal - you can send SMS messages . Since level -87dBm you can establish voice connection
 - o -82 to -71 dBm: good signal - you can send SMS messages and make a calls. Data transmission CSD is not reliable
 - o -70 to -51 dBm: very good signal - you can use all services of GSM network without any restriction.
- **Network Code, Country Code, Area Code:** Network number (provider), where is GSM modul registrated (logged)
- **Cell ID:** BTS number, where is modul registrated.
- **IMSI:** *International Mobile Subscriber Identity*
- **GSM Firmware version:** SW version for GSM part.



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