ATEUS® - VoiceBlue

User Manual

Version: 1.0

Dear customer,

Let us congratulate you on having purchased the ***ATEUS*® - VoiceBlue** system. This new product has been developed and manufactured to provide the maximum utility value, quality and reliability. We hope you will be fully satisfied with our ISDN GSM gateway satisfied for a long time.

! Important !

- The manufacturer constantly improves the firmware contained in the product. The ISP technology (In System Programming) used therein helps you load the latest version into the VoiceBlue gateway using a common PC anytime. For the latest firmware version including all necessary details see **www.2n.cz** and for instructions see the ***"Firmware Upgrade"*** section hereof. We recommend you to apply the latest version to avoid problems that have been eliminated and acquire new functions free of charge.

- To set the VoiceBlue parameters using your PC, you need the "VoiceBlue – Software". For the latest version of this programming tool see **www.2n.cz**.
- You also find the latest PDF version of this manual on **www.2n.cz**. You are recommended to use it especially for firmware upgrade, which provides explanations of new functions.
- Check your delivery for completeness according to the packing list mentioned below and study this manual carefully before installing this product. The manufacturer shall not be responsible for damage caused by using this product in contradiction with this 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 basic installation purposes. Therefore, pay attention to information which subsections are necessary for you and which are not in order to save time.
- Preliminary information on functions that are not yet available are highlighted or printed in grey instead of black.

Packing List

ATEUS ® **- VoiceBlue** including **accessories** contains the following components:

Item	Number of Pieces	Note
VoIP GSM gateway – model according to the order number, see the type label on the GSM gateway back side	1	
Mains adapter	1	1)
Serial laplink cable	1	
Ethernet cable 4-wire (RJ-45)	1	
Antenna with SMA connector	1	
Wall mounting rack	1	
Dowels	2	
Bolts	2	
Warranty certificate	1	
Software on a CD-ROM	1	2)

Notes:

1) *According to the particular type*
2) *Enclosed software:*
   - *Configuration software*
   - *Driver for USB connection*
   - *User Manual*
   - *CE and EMC certificates*

CONTENTS

1. INTRODUCTION

1.1. Purpose

- ***ATEUS*® - VoiceBlue** is designed for any VoIP networks supporting the SIP signalling protocol. It allows for direct calling into the GSM network.
- The voice mode, i.e. outgoing and incoming calls, is the basic function. The gateway is equipped with all functions required for this use and provides the highest comfort in this mode.
- In addition, it enables to receive and send SMS messages.
- No additional equipment is needed for operation. The system can be installed by a layman too (the main parameters can be set comfortably using the configuration software according to the installation instructions included in section 8.4.3, page 26). Default values are set for all programmable parameters. You can make calls the moment you connect the gateway to the Ethernet, connect antennas and the power supply, and install the SIM cards (your VoIP interface configuration must correspond to your VoIP PBX requirements).

1.2. Basic Description of ATEUS® - VoiceBlue

- By connecting the ***ATEUS*® VoiceBlue** to your VoIP PBX you can make outgoing calls to a mobile network directly, **thus eliminating the cost of the PTSN – GSM connection**. Mobile telephone calls from your field staff will be cheaper too.
- You can use **the most advantageous rate of your GSM provider** for the GSM gateway because the calls of all your GSM gateway users are added together.
- You can use SIM cards of different providers and select the **LCR** (Least Cost Routing) in your gateway thanks to the intelligent routing feature.
- If you have SIM cards, the GSM gateway can use free minutes of your free SIM cards first.
- If you use **an answering machine,** you have to pay for the GSM message delivery but if you use your VoIP PBX VoiceMail, you pay nothing for **voice message listening and message reading**.
- You can bar selected numbers in your GSM gateway. **You will not pay for the call you have barred.**
- You can get time and duration records for selected calls. This helps you **identify why your call costs are higher than you expected.**
- You get an almost unlimited storage capacity for your call records thanks to a **replaceable Compact Flash** memory.
- The GSM gateway also generates **statistics** on incoming and outgoing calls automatically.
- **You can use both VoIP and GSM network services conveniently.**
- The **intelligent routing of incoming calls** accelerates connection of incoming calls and provides a higher calling comfort.
- The **Intelligent callback function** enables your staff to call at the account of the GSM gateway SIM cards.
- Voice DISA message including simple recording and editing of invitation message.
- Web interface for SMS sending.
- Optional use of conditioned and unconditioned call forwarding.
- CLIP option.

1.3. Further Advantages and Applications

- You are not exposed to the direct effect of the RF electromagnetic field as the case is with mobile phones.
- You need not use a serial cable for VoiceBlue. All functions are available via Ethernet and USB interfaces.
- You can implement the VoiceBlue gateway into your company's unified messaging using the proper external software (or the internal web server).

1.4. Safety Precautions Related to rf Radiation

! It is prohibited to use any transmitters, i.e. the VoiceBlue gateway too, in areas where explosives are used, such as quarries.

! It is forbidden to use mobile phones and thus the VoiceBlue gateway too at petrol stations.

! The VoiceBlue gateway may affect sensitive life-saving devices in medical centers. So it is prohibited to use mobile phones and the VoiceBlue gateway here.

! In general, any prohibition regarding mobile phones based on RF energy radiation applies to the VoiceBlue gateway.

! Where necessary, the GSM gateway may be installed at a safe distance (in the neighboring building, e.g.) and it must be prohibited to carry an Ethernet line from the VoiceBlue gateway.

! Although the VoiceBlue gateway is not intended for airplanes or cars, all relevant prohibitions and regulations also apply to the VoiceBlue system.

2. PRESENTATION OF ATEUS® - VOICEBLUE

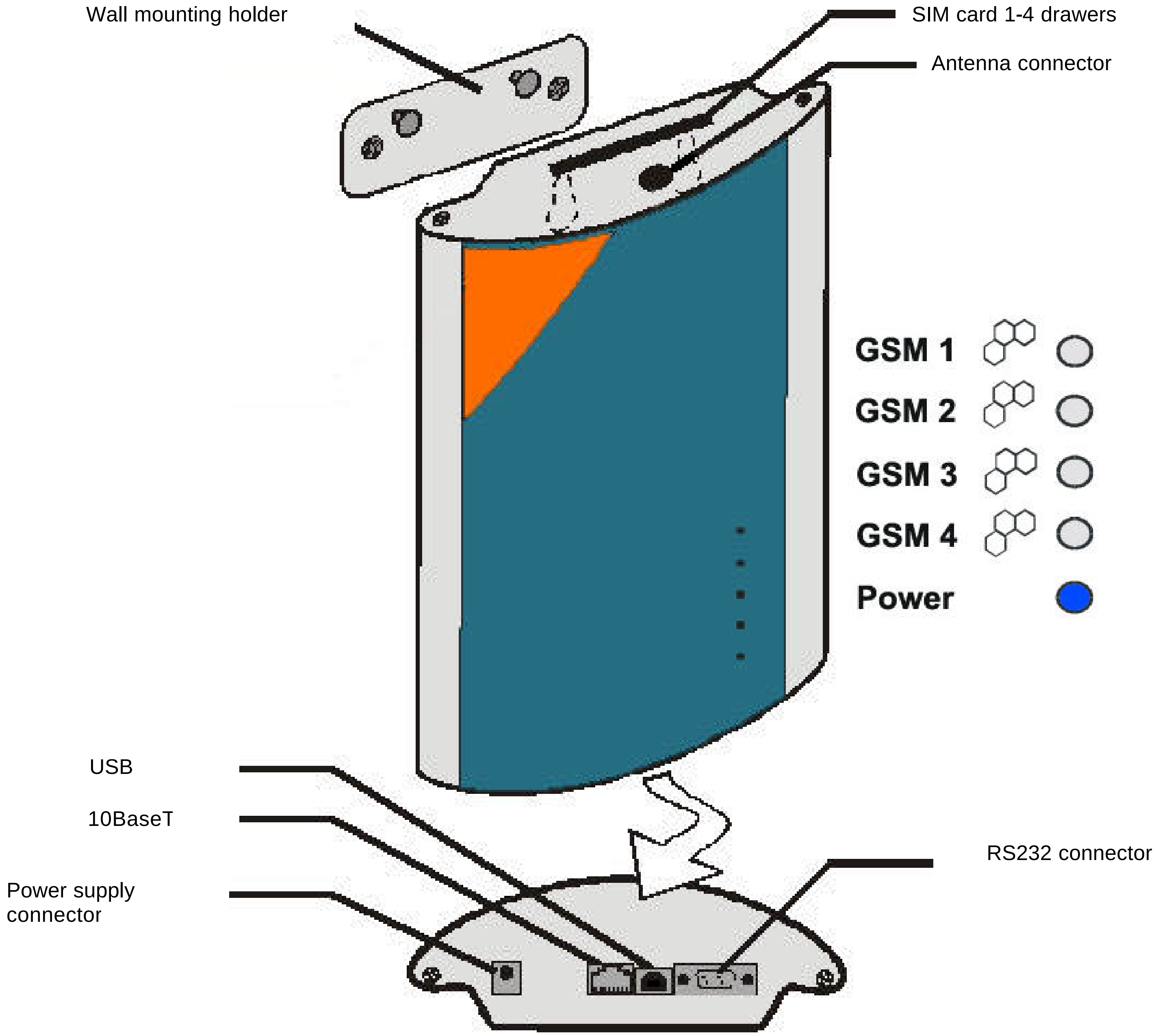


***Fig.1**: Description of external connectors, LEDs and functional components of the VoiceBlue gateway*

2.1. Quick Installation

- **Proper mounting** - ATEUS® - VoiceBlue is designed for vertical walls. Fit the included wall-mounting holder and hang the gateway on it. For the prescribed working position and more details refer to Section 3.1.

- **Cable connection** – connect the gateway to your VoIP PBX using the Ethernet cable. For more details on correct wiring refer to Section 3.7.

- **Antenna connection** – connect an internal antenna cable or an external antenna cable leading from a place with a good GSM signal to the SMA connector (see 3.2).

- **PC connection** – connect the gateway to your PC to configure the VoiceBlue gateway using the configuration software. To do so, use an RS232 crossed cable (the same as for PC-PC interconnection) (see 3.6), or a USB cable.

- **Power supply** – the delivery includes a mains adapter. Plug in the adapter connector into the gateway and put the power adapter in a mains socket. The gateway goes on immediately (see 3.5). (This does not apply to the 19´ rack version where power is fed along a bus).

- **SIM card insertion** – to insert a SIM card, press the yellow reader microbutton with a suitable tool to make the small drawer slide out a little. Remove the drawer, insert the SIM card and replace the drawer (see 3.3 and 3.4).

- **Configuration software installation** – run an installation file from the installation CD on your gateway-connected PC and install the VoiceBlue configuration software (see 10.1).

- **Configuration software** – run the VoiceBlue program installed, set the communication parameters of the serial channel to which the gateway is connected. Establish communication between your PC and the gateway (see page 26 section 8.4).

- **ATEUS® - VoiceBlue configuration** – now set parameters to determine your gateway's behaviour using the configuration software. They include, e.g., VoIP parameters, basic GSM parameters and tariff metering, routing, restrictions, rates and system parameters. Select the required function parameters and transfer the configuration data to the gateway through the serial channel. For more details on the configuration software refer to Section 10.3, page 29. For the basic installation instructions see section 8.4.3, page 26.

3. Basic ATEUS® - VoiceBlue Installation Instructions

3.1. Proper Mounting

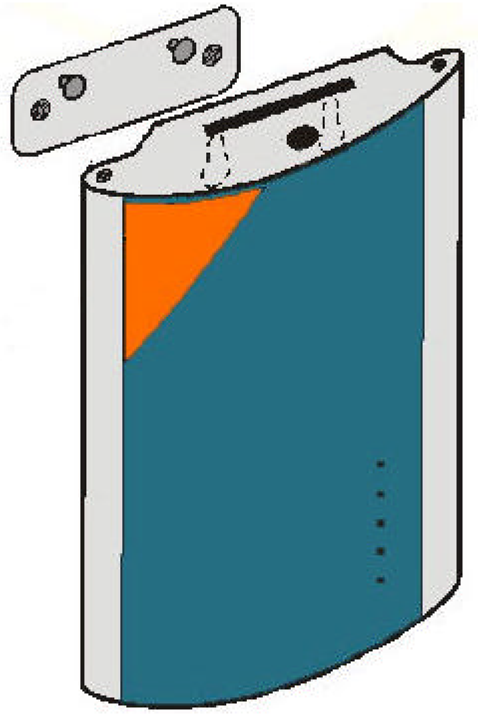


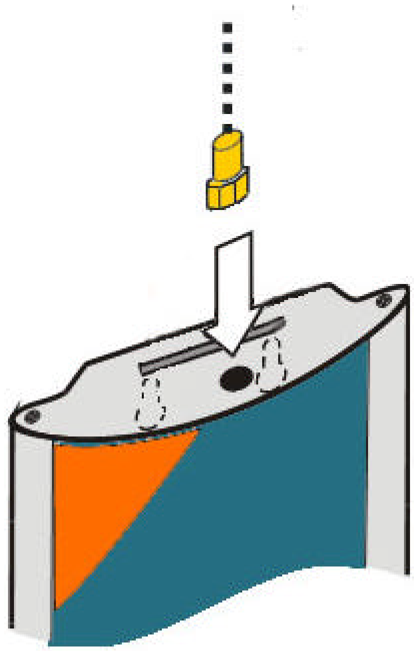
Fig.2: *VoiceBlue working position*

- The ***ATEUS®* - GSM VoiceBlue** gateway is designed for wall mounting. The prescribed working position is shown in Fig. 2.
- It is possible to operate the VoiceBlue gateway in another working position, e.g. on a desk, for a short time only – for quick testing purposes in the service centre, for example.
- The allowed range of working temperatures and relative humidity values are included in the "*Technical Parameter"* section.
- Any exceeding of the operating temperature does not affect the VoiceBlue gateway function immediately but may result in more rapid ageing and lower reliability.
- The VoiceBlue gateway is intended for indoor use. It may not be exposed to rain, flowing water, condensed moisture, fog, mist, etc..
- The VoiceBlue gateway may not be exposed to aggressive gas, acid vapours, solvents, etc. or aggressive liquids. During cover cleaning, for example.
- The VoiceBlue gateway is not designed for environments with high vibrations such as means of transport, machine rooms, etc..
- Free space has to be left under and over the VoiceBlue gateway for cables and agitated air to remove operational heat.
- Install the VoiceBlue gateway on a place with a good GSM signal.
- It is not recommended to install the VoiceBlue gateway or the antenna near television, broadcasting or similar RF-sensitive devices because it may affect their function negatively.
- As a source of RF energy emission, the VoiceBlue gateway antenna should not be located close to human bodies. The hazard is higher than with mobile telephones because the gateway usually radiates energy more often if it is used by many people.

3.2. External Antenna Connection

Connect the external antenna cable to the SMA antenna connector. The antenna should be installed on a site with a good GSM signal and mounted vertically. For antenna and cable parameters refer to the *Technical Parameters*. Tighten the antenna connector **gently** with your hand, never use a wrench. For the signal strength see page 32, section 10.4.4 .

Antenna



3.3. SIM Card Setting and Installation

GSM provider and SIM card type selection

To use the VoiceBlue gateway you need a SIM card of any of the GSM providers working in the band of 900 MHz or 1800 MHz. ***ATEUS***® **- VoiceBlue** works with all SIM cards that are supported by Siemens mobile telephones and other mobile equipment. To make sure that your provider‘s SIM card is supported by your ISDN GSM gateway, try to use the SIM card in your Siemens mobile telephone (e.g. C35).

PIN entering disable (optional)

PIN entering is set by default in the ***ATEUS***® **- VoiceBlue** gateway. You can disable this setting on your SIM card (insert the SIM card in any common mobile telephone). If you do so, it is irrelevant how VoiceBlue has been programmed or if there is a PIN code in the VoiceBlue memory or not. If you do not do so, your VoiceBlue lights a LED requesting the correct PIN, which must be identical for all SIM cards inserted.

Note: *If a PIN-active SIM card is inserted and no PIN code is entered, the LED indicator of this GSM nodule flashes red in the ratio of 1:1.*

GSM network service setting (mailbox, call forwarding)

You have to decide before installing the SIM card whether you want to use the **forwarding of incoming calls** function available in GSM networks (call forwarding in the case of busy line, absence, unavailability…). It is mostly advantageous to disable all such call forwarding statuses in cooperation with the PBX (especially to the provider‘s mailbox), or use an answering machine. Where multiple GSM gateways are connected to one PBX, you can redirect calls in case of busy line from one PBX to another, and so on. (Call forwarding activation and deactivation are more advantageous with the SIM card inserted in a standard GSM telephone).

Roaming settings (calling via foreign networks)

Roaming is disabled in the ***ATEUS***® **- VoiceBlue** gateway by default (can be set in the configuration program - see section 10.5.1.5, page 37). One usually does not travel with the VoiceBlue gateway and there is risk of foreign network log-in and dramatic cost increase in border areas in the case of local GSM network failure. To select roaming and prefer some networks, complete the list of preferred GSM networks using a mobile telephone and select the roaming function while programming.

3.4. SIM Card Insertion in ATEUS® - VoiceBlue

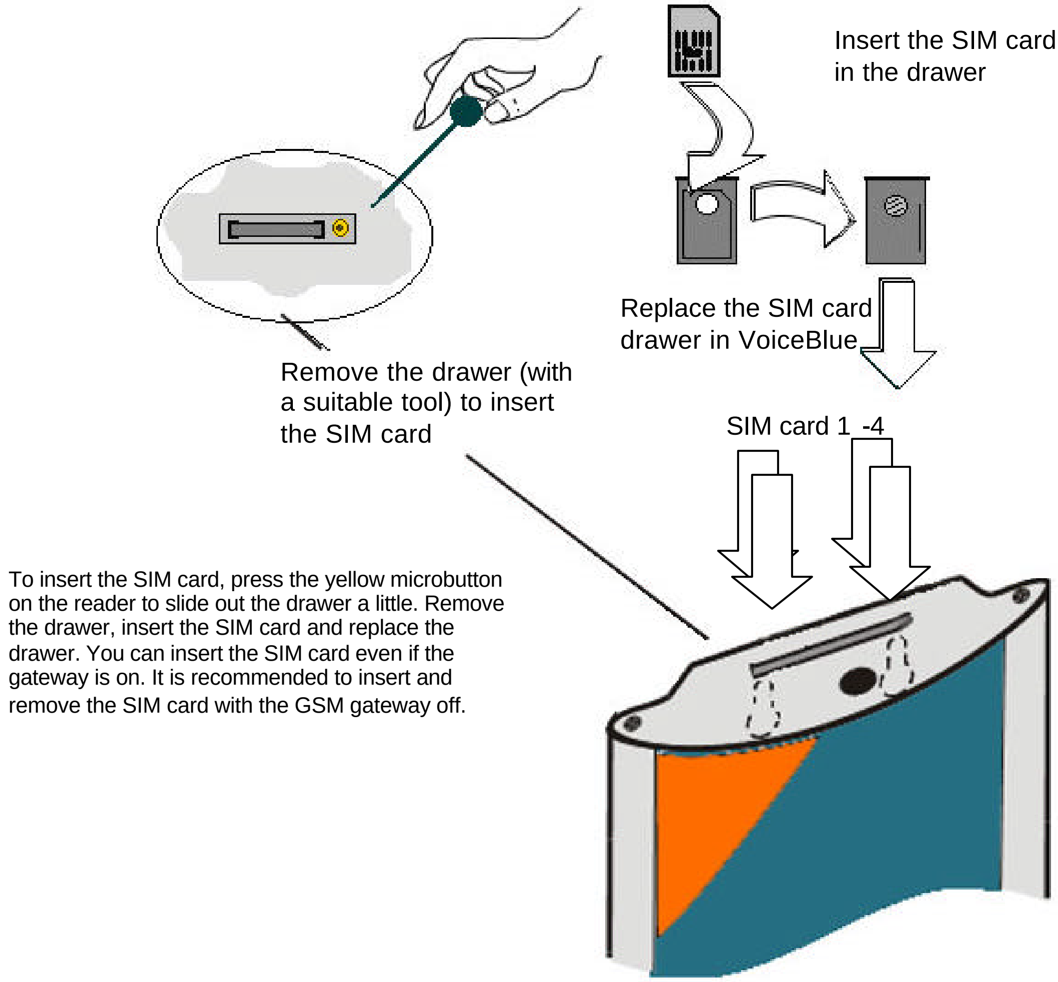


Fig.4: *SIM card insertion in VoiceBlue*

3.5. Power Supply Connection

- Make sure that the mains voltage value is in compliance with the data shown on the product label.
- Use the mains adapter, which is included in your VoiceBlue gateway delivery.*
- Make sure that you have connected the antenna. Connecting the power supply with a disconnected antenna may damage the GSM module transmitter!
- Connect the mains adapter connector to the gateway (see the *Bottom Face Connectors* – Fig.1). The green "Power" LED must go on in a while.

**This does not apply to the GSM gateway rack version where power is fed along a bus.*

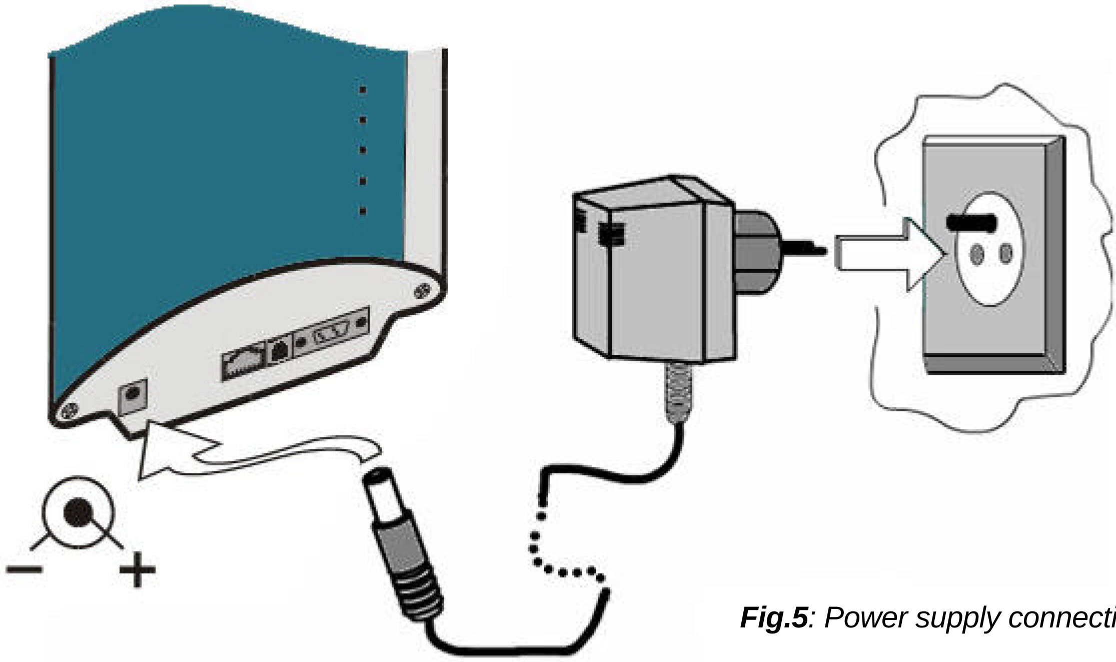
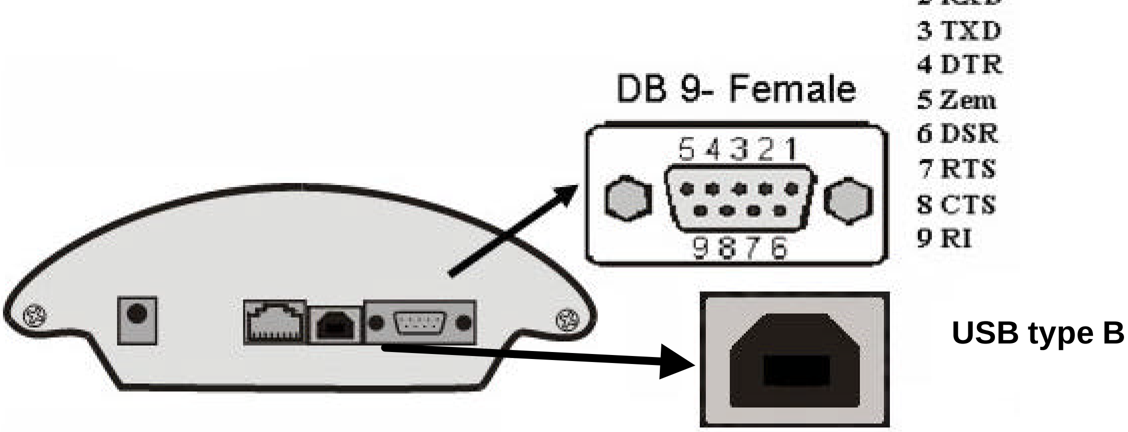


Fig.5: *Power supply connection*

3.6. PC Connection

For primary setting, you can connect your new GSM gateway to your PC using a serial crossed cable (laplink), or a USB cable (for controllers see the CD enclosed). The maximum serial cable length is several metres. In case you plan to use a more than 5 metres‘ long USB cable, we recommend you to use USB repeaters.

Fig.6: *PC connection of VoiceBlue gateway via RS232*



3.7. *Ethernet Connection*

VoiceBlue is connected to your Ethernet network like any other terminal (PC, VoIP terminal) with a straight Ethernet cable (except for direct interconnection of two identical units where a crossed cable must be used)..

Fig.9: *View of 10BaseT connector*

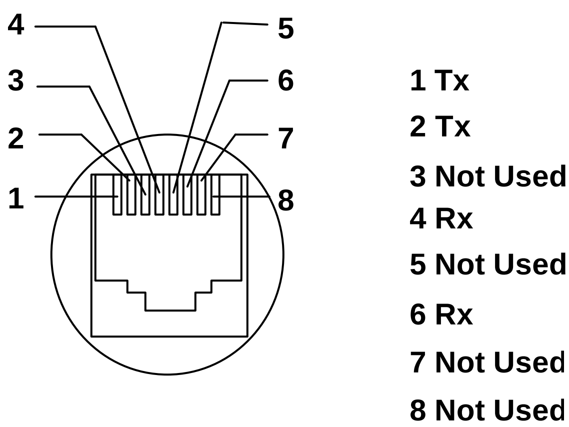
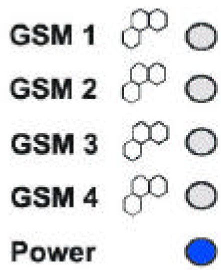


Fig.10: Meanings of RJ45 contacts

3.8. LED Indicators

3.8.1. Basic Function of LEDs

Fig.12: LED indicators



LED GSM statuses

Each module has a LED of its own to indicate its status.

Module status	LED GSM 1 - GSM 4	Colour / Status
Module not available	GSM 1 and/or GSM 2	Red is on
SIM card not inserted	GSM 1 and/or GSM 2	Red is on
Module is being initialised	GSM 1 and/or GSM 2	Green flashes slowly 1:3
Module is logging in	GSM 1 and/or GSM 2	Green flashes quickly 1:1
Module has logged in	GSM 1 and/or GSM 2	No light
Module has an active call	GSM 1 and/or GSM 2	Green is on

4. VOICE TRANSMISSION VIA IP

4.1. Voice Coding

Voice and signal communication channels are strictly separated in the VoIP network. The RTP (Realtime Transport Protocol) is mostly used for voice transmission in modern VoIP networks. Its task is to transmit data (voice) from the source to the destination at real time. So-called codecs are used to save the data bandwidth by reducing the transmission rate using a complex algorithm. The level of compression used by the codec affects the quality of the transmitted voice. This means that the wider the data bandwidth (the higher the transmission rate), the higher the voice transmission quality. The voice transmission quality is measured by the MOS (Mean Opinion Score) where 1 means the worst and 5 the best quality. For a list of VoiceBlue-supported codecs see the table below:

Table of VoiceBlue-supported codecs and comparison of MOS with transmission rate requirements:

Standard	Algorithm	Transmission Rate [kbps]	MOS
G.711	PCM	64	4.1
G.726	ADPCM	32	3.85
G.729	CS-ACELP	8	3.92
G.723.1	ACELP	5.3	3.65

To achieve a high-quality voice transmission, it is important to maintain the required transmission rate during the whole connection and a short, identical time for transmission of one data packet.

4.2. Establishing Connection

The codec to be used for voice transmission is selected for call establishing automatically. ATEUS - VoiceBlue is ready to use codecs included in Section 12, page 49. It depends on your VoIP network which coding system shall be selected. ATEUS - VoiceBlue is designed primarily for connection to corporate VoIP networks and tries to comply with the opposite side‘s codec. If the codec is incompatible with the VoiceBlue system, the call is rejected.

IETF SIP and ITU-T H.323 protocols are most frequently used for calling and signalling in the world. The ATEUS - VoiceBlue gateway applies the SIP (Session Initiation Protocol) for signalling. The SIP uses the following components:

- UAC (User agent client) – client in the terminal that initiates SIP signalling
- UAS (User agent server) – server in the terminal that responds to the SIP signalling from the UAC
- UA (User Agent) – SIP network terminal (SIP telephones, or gateway to other networks), contains UAC and UAS
- Proxy server – receives connection requests from the UA and transfers them to another proxy server if the particular station is not in its administration.
- Redirect server – receives connection requests and sends them back to the requester including destination data instead of sending them to the calling party

- Location Server – receives registration requests from the UA and updates the terminal database with them.

All server sections (Proxy, Redirect, Location) are typically available on a single physical machine called proxy server, which is responsible for client database maintenance, connection establishing, maintenance and termination, and call directing.

The VoIP-GSM Ateus VoiceBlue gateway is ALWAYS UA – has the same functions as a VoIP telephone – receives call requests and directs calls to the GSM networks (according to the internal LCR).

There are no SIP-defined server sections in the Ateus VoiceBlue gateway.

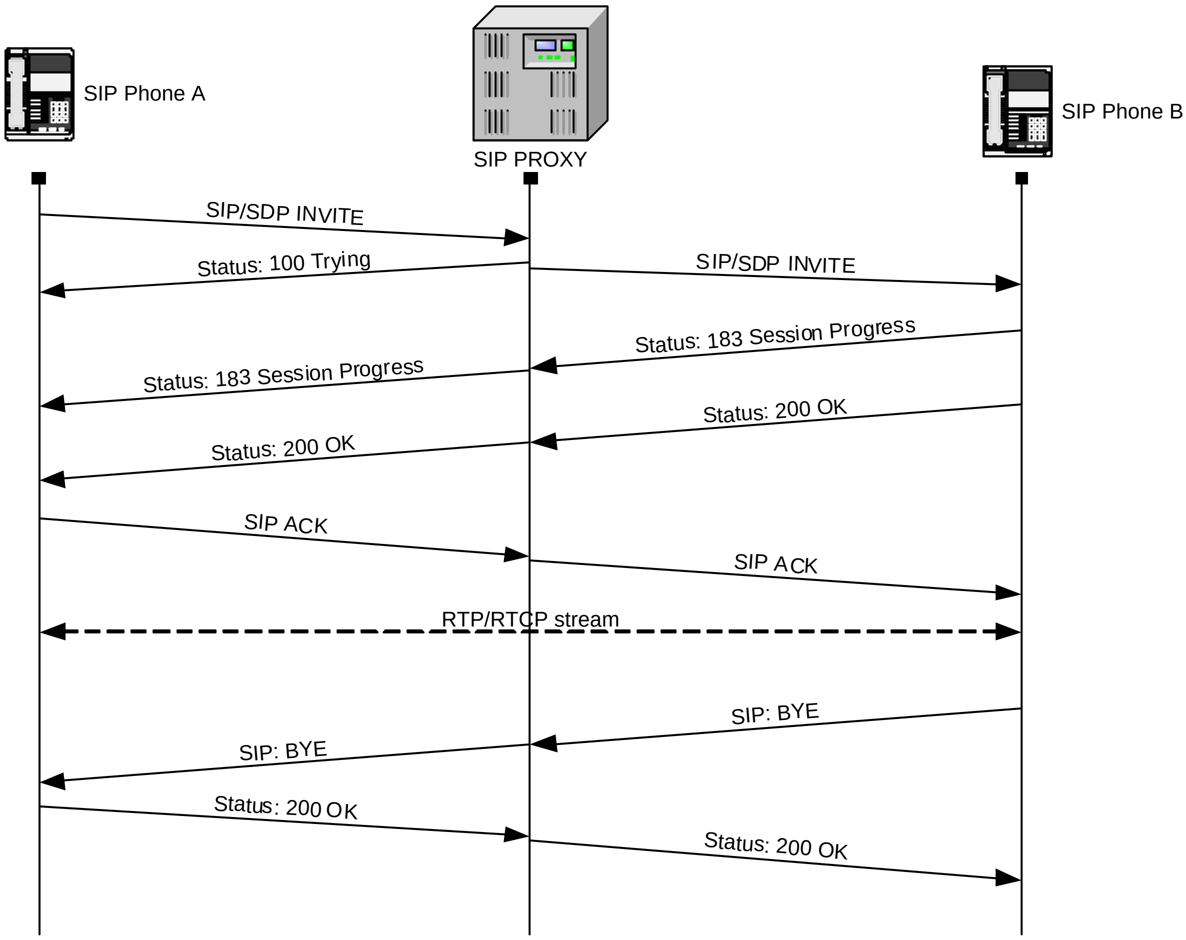
Basic messages sent in the SIP environment:

- INVITE – connection establishing request
- ACK – acknowledgement of INVITE by the final message receiver
- BYE – connection termination
- CANCEL – termination of non-established connection
- REGISTER – UA registration in SIP proxy
- OPTIONS – inquiry of server options

Answers to SIP messages are in the digital format like in the http protocol. Here are the most important ones:

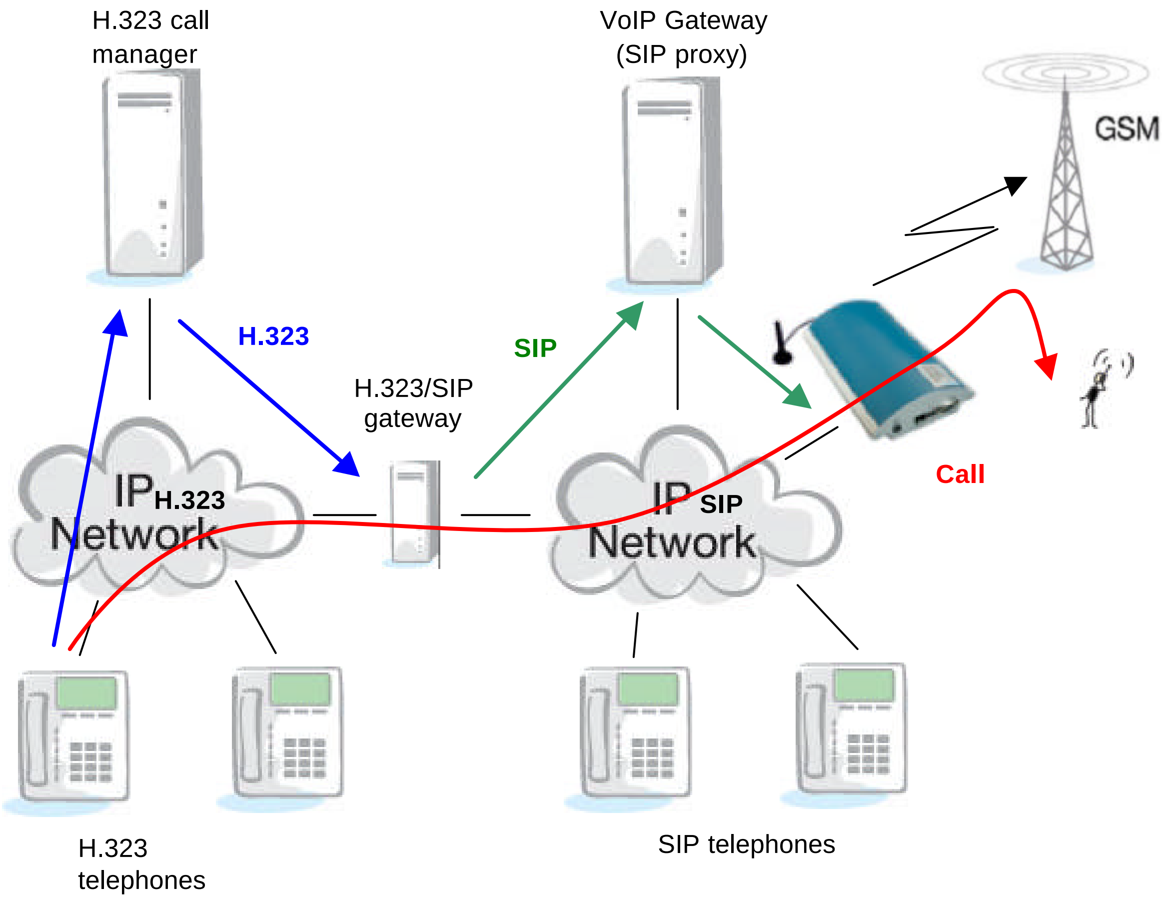
- 1XX – information messages (100 – trying, 180 – ringing, 183 - progress)
- 2XX – successful request completion (200 – OK)
- 3XX – call forwarding, the inquiry should be directed elsewhere (302 – temporarily moved, 305 – use proxy)
- 4XX – error (403 – forbidden)
- 5XX – server error (500 – Server Internal Error, 501 – not implemented)
- 6XX – global failure (606 – Not Acceptable)

Connection establishing and terminating procedures in the SIP proxy server environment:



SIP and H.323 Interconnection

Terminals working with the SIP can communicate with terminals working with the H.323 protocol using the SIP/H.323 gateway. This gateway transfers signalling messages of the two protocols. Since both the SIP and H.323 protocols use the RTP for multimedia data transmission, they can communicate directly with each other through the gateway once the connection is established. ATEUS - VoiceBlue can be implemented in the existing H.323 environment using the SIP/H.323 gateway.

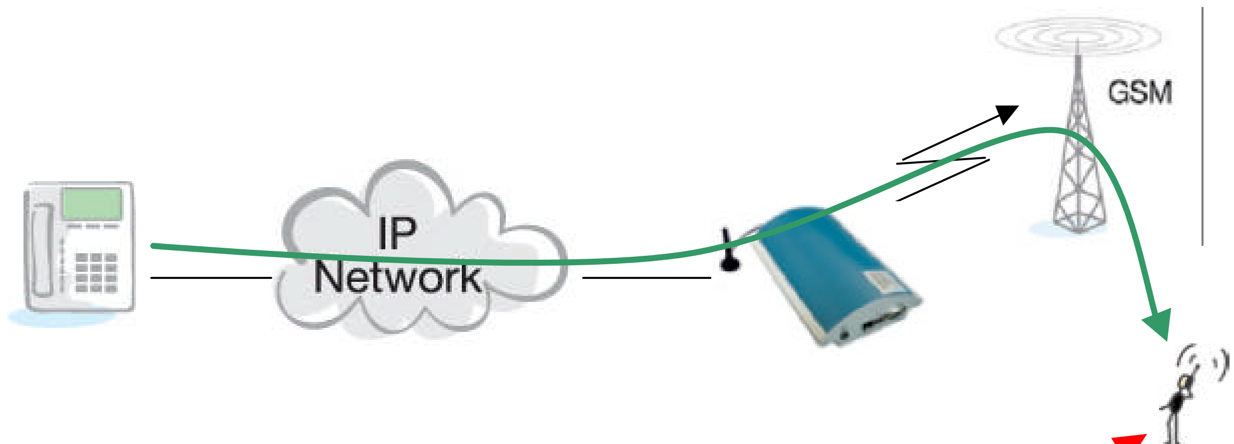


5. VoiceBlue Connection Options

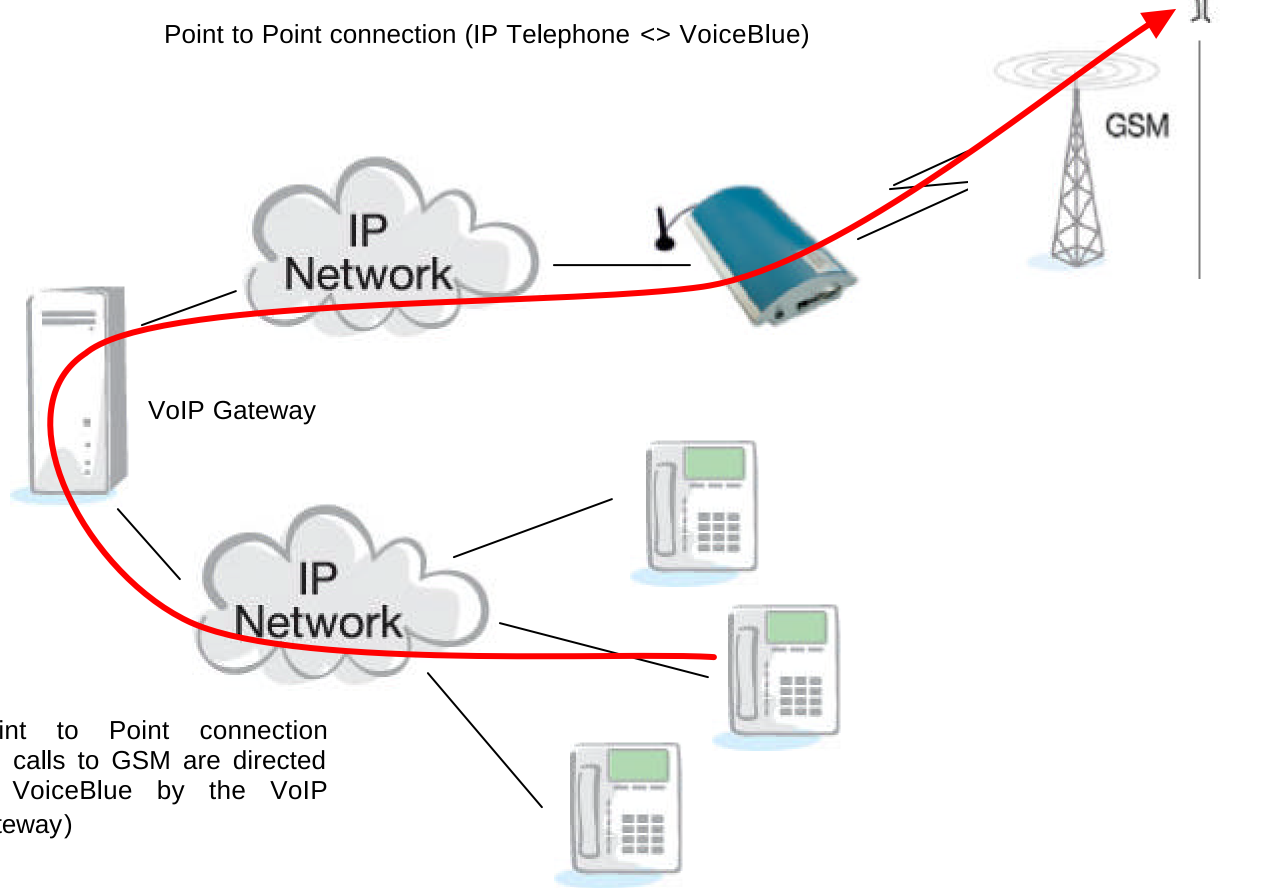
Ateus VoiceBlue can be employed in two modes, either Point-to-Point or Point-to-Multipoint, with the SIP Proxy server.

5.1. Point-to-Point Mode

In the Point-to-Point mode, VoiceBlue can communicate with one SIP VoIP telephone only, which is often used for testing purposes before VoIP implementation or combination with some other SIP VoIP equipment. In this mode, it is either possible to set the IP address of the opposite equipment as the proxy server in both units, or, if an intelligent VoIP gateway is used, resend some calls to the VoiceBlue IP address directly. The other side's IP is always set as the proxy server IP in VoiceBlue in the P-T-P mode.



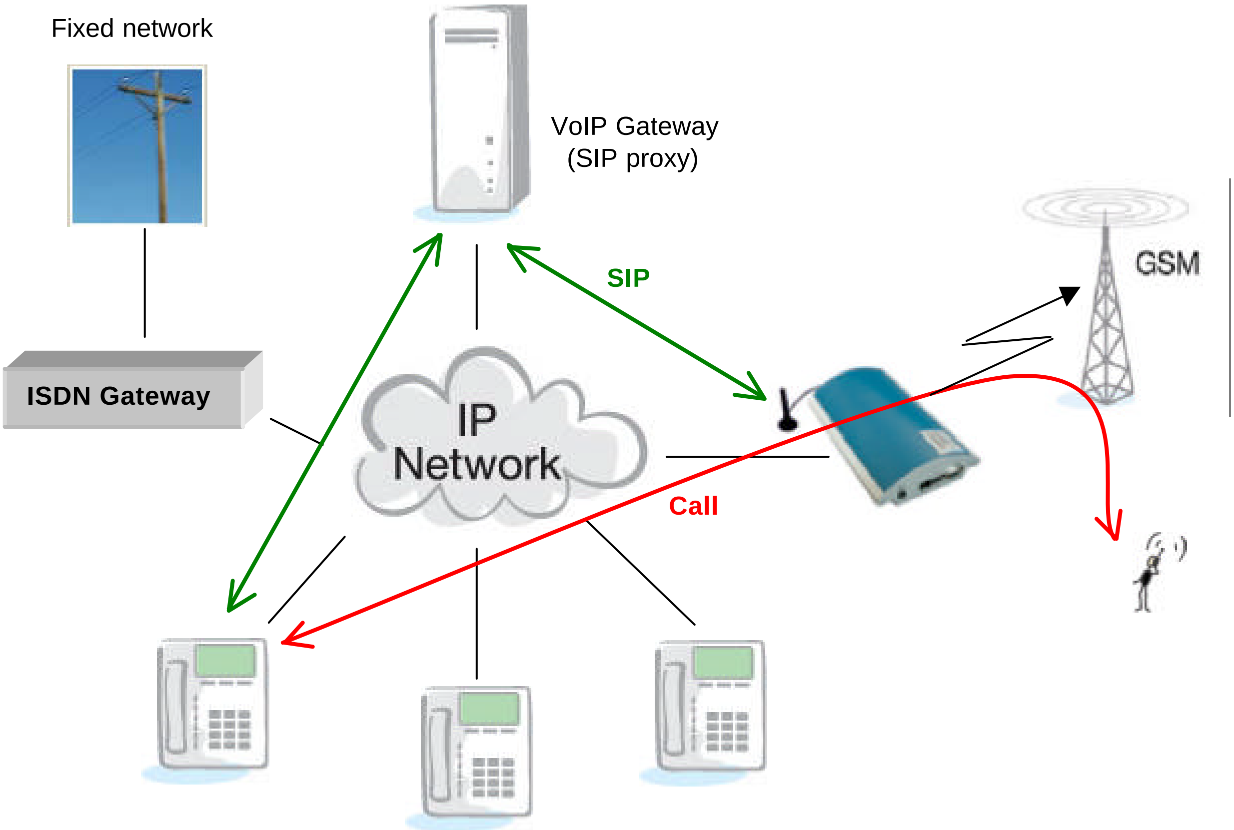
Point to Point connection (IP Telephone <> VoiceBlue)



Point to Point connection (all calls to GSM are directed to VoiceBlue by the VoIP gateway)

5.2. Point-to-Multipoint Mode

Speaking of the Point to Multipoint mode we think of the classic layout of a distributed VoIP network with one or more SIP Proxy servers (VoIP gateway). The SIP proxy server is a software version of a PBX (or a standard PBX extended with VoIP services), which is responsible for all VoIP signalling services. In this mode, multiple source units (e.g. VoIP telephones) and multiple destination units (e.g. VoiceBlue) can be used. An internal routing algorithm (LCR) of your SIP proxy is used in this mode to route calls going to GSM and other networks. GSM calls can be routed via VoiceBlue gateways connected. All signal communication (SIP) is controlled by the SIP proxy server here and the subsequent voice stream is executed in the RTP Point-to-Point mode.



6. Routing Principle

An incoming call from the VoIP port is routed according to the internal routing algorithm (LCR) to an arbitrary GSM port. An incoming call is routed directly to the defined SIP address or the DISA tone dial-in is activated. Routing can also be made on the basis of the CLIP (incoming call number).

The internal routing algorithm distinguishes the type of outgoing calls, the current time rate, day of the week, or free minutes on GSM providers‘ SIM cards, and routes outgoing calls according to these settings.

If an outgoing call is routed via a busy port, the next available port is used (depending on the configuration). In case none of the allowed outgoing ports is free, the outgoing call is rejected. In the latter case, your SIP proxy can either use another gateway or terminate the unsuccessful call.

6.1. Outgoing Call via VoIP GSM Gateway

6.1.1. Routing to GSM Network

The moment a VoIP subscriber seizes the line and dials a number, your SIP proxy routes the call to the VoiceBlue gateway. In case the outgoing route is defined for the outgoing prefix in the VoiceBlue LCR, the connection request is accepted and the gateway tries to establish connection through the defined GSM module(s). In case the call has an unknown prefix or all outgoing routes are occupied, VoiceBlue rejects the call request. However, it depends on your SIP proxy configuration whether calls intended for GSM networks shall be directed to VoiceBlue or not.

Note: *If the GSM gateway is busy, the SIP proxy can either give the caller the busy tone - - - -, or the PBX selects another connection type (there may be more gateways than one connected to the PBX).*

6.1.2. LCR (Least Cost Routing)

The LCR function is enabled on the VoIP port. The Network lists include prefixes (up to 8 prefix groups can be defined). In addition, you can sort out GSM modules into up to 4 GSM groups using the configuration software. Select the outgoing GSM group(s) in the LCR (Least Cost Routing) chart to be used if the outgoing call prefix matches one of the Network lists. You can set a number of free minutes for each GSM group for each SIM card. In this case, you can give priority to SIM cards with the highest number of free minutes in the GSM group. An adjustable accuracy of deduction of free minutes and automatic recovery of the free minute credit on a defined day (applicable to SMS too) are a matter of course. If there are more GSM groups defined for one prefix in the LCR chart, the gateway attempts to establish the call using the highest-priority group and, if unsuccessful (all GSM modules of the group are occupied), the gateway attempts to establish the call using the next group automatically. Having exhausted all GSM groups defined, the gateway rejects to establish the call.

6.1.3. Establishing Connection

You do not pay for your connection until the called party answers the call. The GSM network signals this moment (connect) and the GSM gateway transfers this information to the SIP proxy. You can send the so-called connecting tone to the called party in outgoing GSM calls*.

**GSM modules of the TC35i version can activate/deactivate a connecting tone, which fills in the silence between the request sent to the GSM network and the ringing tone.*

6.2. Incoming Call from GSM Network to VoiceBlue Gateway

Below is an incoming call sequence. The actions are arranged as they occur. All of these services can be enabled and disabled selectively.

1. CALLBACK

If the CALLBACK function is activated and the incoming call identifies itself with a CLIP matching the CALLBACK CLIPs, the GSM gateway does not receive the call and lets it ring. When the ringing is terminated, the GSM gateway calls back to the GSM network (the CLIP number) and, when the GSM side answers the call, starts to play back its DISA voice message allowing for the DTMF dial-in.

2. CLIP Routing

Having find out that the incoming call number (CLIP) matches a number included in the "Autorouting" chart, the gateway connects this call directly to the VoIP destination selected.

3. DISA DTMF Dial-In

If the DISA DTMF dial-in function is enabled and an invitation message (or another dialtone) has been recorded, this message is played back for every incoming call except for calls described in items 1 and 2 and the DTMF dial-in is awaited for a defined timeout. Having received a sufficient number of digits (see the Setting), the gateway gets through to the SIP proxy inquiring of the defined IP destination.

Should a number with a lower count of digits than defined be accepted, the dialling should be terminated with a #. If the DISA rejects a pre-programmed number or the # character, no connection is established. Therefore, enable the operator service (see below) while activating the DISA service.

4. Operator

The gateway supports dialling to the operator(s). Hence, you can set different numbers for the GSM ports to be dialled-in to the VoIP network. There may be multiple operator numbers for each GSM incoming destination. In case the SIP proxy designates the first-dialled destination as unavailable, the gateway attempts to connect the call using the next one. For example, the gateway can work with the operator service in the incoming mode only. In that case, all incoming calls are directed to the operator number.

6.3. DISA Message Recording

A DISA message can be recorded in the gateway using the configuration tool enclosed or a terminal. The voice file must have defined parameters (included in the table). The file can also be recorded using a terminal with the aid of the XMODEM transmission protocol.

Acoustic format	Wav
Sampling Frequency	8kHz
Count of channels	1 (mono)
Codec	ISDN Alaw

7. ANTENNA SPLITTER

7.1. Splitter Description

The antenna splitter is a passive unit suitable for GSM gateways. It combines 4 antenna inputs into one antenna output. The antenna splitter saves antenna cables, the number of outdoor antennas and mounting space on the roof.
As a passive unit it inserts loss between the antenna and equipment.

7.2. Technical Parameters of Antenna Splitters

Parameters	Value	Note
Connector type		
Output antenna connector	SMA type, female	
RF parameters		
Impedance	50 OHM	
Frequency	850 – 1900 MHz	
Insertion loss	< 8 dB	
Isolation between two channels	> 20 dB	

7.3. Discreet Antenna

Discreet antenna basic parameters:

Type	Car antenna
Frequency	900/1800 MHz
Gain	3dB
Cable	Small coax cable 174A (5 m)
Connector	SMA (male)

Discreet antenna

8. INSTALLATION

8.1. Firmware Version

Before installing your VoiceBlue gateway please upload the new firmware into the gateway. You find the current firmware version and all necessary software on an enclosed CD or on our website www.2N.cz .

FIRMWARE uploading procedure:

- Connect your PC to VoiceBlue via RS232/USB.
- Prepare the firmware-containing file into a folder selected by you (*Pxxxx-V-xx.xx.xx.bin.*
- Run the VoiceBlue program.
- Select the “Upload firmware” item in the “Gateway control” menu.
- Select the file containing new firmware (Pxxxx-V-xx.xx.xx.bin).
- The program now automatically uploads your new firmware (it takes about 2 minutes).
- The gateway is reset during this procedure (thus discontinuing all current calls and SMS). Do not interrupt the program during this procedure to avoid errors in firmware uploading – otherwise the gateway might stop working!

Should the firmware uploading procedure be terminated, please make cool reset of the gateway and re-load the VoiceBlue program and try to upload the firmware again.

8.2. Installation Conditions

The following system installation conditions must be met:

- appropriate location (enough free space);
- GSM signal intensity (minimum signal level**: –80db**); you can use the NETmonitor on a mobile phone (e.g. Nokia, Siemens) as a GSM signal intensity meter;
- un-overloadable GSM cells to which the GSM gateway modules are logged-in. Please keep in mind that full traffic means up to 30 calls at the same time (according to the gateway configuration!);
- no strong electromagnetic radiation on the system installation site;
- no strong reflections on the antenna installation place;
- the VoIP connection must be configured properly and meet related recommendations.

8.3. Potential GSM Network Problems

The VoiceBlue works reliably even under a 100% load. GSM networks may cause the following problems:

1. The GSM modules cannot log in, log in slowly or log out occasionally. This problem can have two causes:
   - The GSM signal level is low – the recommended minimum signal level is approximately **-80dB**, if lower, the GSM antenna location must be changed.
   - The GSM cell to which the modules try to log in is overloaded – the GSM antenna location must be changed or decrease number of GSM channels which are using problem GSM network.
2. One of the GSM modules is permanently logged-out or fails to receive incoming calls – this problem indicates a GSM network overload due to heavy traffic. You can eliminate this problem by the above mentioned solutions or extend the “relaxing time“ – i.e. the delay between two calls via one GSM module (the recommended value is 2 seconds). In case any of the GSM modules cannot log in to the network even after reset, your GSM provider may have located the SIM card but refused to log it in because either too many calls are made using this card, or the SIM card keeps logged -in to the same GSM cell and the same GSM module for too long. This problem can be solved by occasional exchange of SIM cards between modules.

8.4. Main Installation

- Place the gateway into an environment that corresponds with its working conditions.
- Configure the gateway properly using the configuration software included.
- The gateway mains supply must be backed-up and overvoltage-protected (a line-interactive or on-line UPS is recommended).
- For a more comfortable gateway administration, it is advisable to have one of available remote control tools on site (Ethernet).

8.4.1. Control Ways

The system can be supervised and controlled locally or remotely as follows:

A. Local control using a PC connected by a standard full crossed serial cable.

B. Local control using a PC connected by a USB 1.1 cable.

C. Remote control over the IP network using a standard Telnet protocol.

8.4.2. Configuration Ways

The system can be configured in any of the following two ways:

A. using extended AT commands (refer to Appendix A);

B. using the VoiceBlue program..

A PC can be used for both the ways, either locally (via a serial cable), or remotely with a modem, using the Telnet protocol over IP, or via the B-channel.

Since VoiceBlue is constantly being upgraded, you are advised to use the latest VoiceBlue software firmware version.

8.4.3. Your First Installation (Quick Step-by-Step Guide)

1/ Connect the gateway with the PC via an RS232/USB cable.
2/ Install&Run the VoiceBlue program from attached CD.
3/ Open the COMMUNICATION parameters in VoiceBlue program and select the proper COM port for VoiceBlue connection
4/ PROGRAMMING, SYSTEM Section (for description see page 35, Section 10.5.1.1):

- Add the IP address and IP mask of the VoiceBlue system (in case you want to make remote control via the IP);
- Enter the PIN code (in case the PIN code is enabled on your SIM cards) **!!! WARNING – IN CASE YOU HAVE PIN-ACTIVE SIM CARDS ALL THE SIM CARDS MUST HAVE AN IDENTICAL PIN CODE AS DEFINED IN THE VoiceBlue !!!**

5/ PROGRAMMING, ETHERNET Section (for description see page 36, Section 10.5.1.2):

- Set right settings of SIP signalization. Please do not remember that for right function you should modify settings in your SIP.

6/ PROGRAMMING, GSM SETTINGS (for description see page 36, Section 10.5.1.3):

- Divide the GSM modules (cards) into outgoing and incoming GSM groups according to your installation requirements.

8/ PROGRAMMING, NETWORK LIST Section (for description see page 39, Section 10.5.1.7):

- In case you use multiple GSM providers please delete all of the eight Network lists and add GSM provider 1 prefixes to Network list 1, GSM operator 2 prefixes to Network list 2, etc… Please remember to add the correct length of the dialled number for each prefix.

9/ PROGRAMMING, LCR TABLE Section (for description see page 39, Section 10.5.1.8):

- Here you have the possibility to define at what time which prefixes (saved in network lists) are to be routed to which GSM group. You can select more GSM groups in one line – the gateway attempts to use another GSM group in case the first GSM group is unavailable or busy. Moreover, you can set more attempts for one GSM group > for increase ASR…

10/ Open the CONTROL, DATE AND TIME item in the top menu of the GATEWAY CONTROL to set time and date in the gateway.

11/ Open the CONTROL, CHANGE USER NAME AND PASSWORD items in the top menu of the GATEWAY CONTROL to change access rights for remote control via Telnet or PRI ISDN.

12/ Save the configuration to the PRIGW.

13/ Plug in the SIM cards, connect the antennas and restart the gateway.

If you had a problem with the gateway function please read this manual carefully and check all parameters. You can also contact us by e-mail (techsupport@2n.cz). Please describe the problem in detail and attach the gateway configuration file (config.cfg) in your e-mail message.

9. How to connect VoiceBlue via USB

You can make connection between your PC and VoiceBlue via USB 1.1 connection. A Drivers for OS Windows 98SE/ME/2000/XP you find on attached CD. The main installation procedure is very easy:

- insert the CD to PC
- connect USB cable to PC and then to the VoiceBlue gateway
- the Windows OS automatically recognize new hardware on USB BUS and ask you for right drivers.
- The drivers you find on the CD in following path:
  CDROM:\VoiceBlue\USB drivers
- OS automatically recognize right driver and install it
- OS can ask you for drivers for USB COM port, these drivers are in same directory on the CD.

Please do not install different USB drivers for the gateway connection
Microsoft does not certify that you are using OS Windows XP, please click to OK in case of warning that driver.

After successful installation you can communicate with VoiceBlue via virtual serial port (default number is 3).

10. VOICEBLUE SOFTWARE USER MANUAL

10.1. VoiceBlue Program Installation

The ***ATEUS®* - *VoiceBlue*** delivery includes an installation CD containing the ISDN GSM program installation data. After you insert the CD in your PC CD-ROM, an introductory page is displayed showing an overview of 2N products. Select VoiceBlue. Now click on the VoiceBlue program installation in the new window. A simple installation guide will help you install the software.

10.2. VoiceBlue Program Running

Once the installation is completed, start the program by clicking on the "VoiceBlue program" in your PC program menu, clicking on the icon on the desktop, or running the "***VoiceBlue program.exe***" file, which you find on the appropriate location installed by you with the aid of any explorer or file browser.

10.3. Connection with VoiceBlue

Once the program has been installed, set the gateway communication. To do this, select the "Setting > Communication" menu items. The basic setting is shown in the figure. For more details see section **10.4.5.**

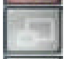
The "Gateway > Connect gateway" menu or the "Connect gateway" icon on the button bar can be used for gateway connection and gateway–PC communication. Select the "Modem" function only for remote gateway supervision using a connected modem. You can use a standard RS232* interface or Ethernet 10baseT (Telnet) for the gateway connection.

In case that you are connected via standard serial cable you have to set transmission rate to **57600 bps**.

Note: *If you have more gateways than one, then follow the gateway communication instructions included in Section 10.4.2.*

10.4. Description of icons on main window

For easier controlling of VoiceBlue GSM gateway have VoiceBlue program modern graphical interface. On table below you have description of all icon buttons which you can find on this start window::

Icon	Description
	End of VoiceBlue program
	Minimalization of the program
	Helo
	Settings of VoiceBlue program
	Switching of language
	On-line control items
	Program parameters
	On-line monitoring of the gateway
	Reset of the gateway
	Selection of the gateway from list of gateways
	Communication settings

10.4.1. Control menu, File

Using this menu, you can work with the **config.cfg** gateway configuration file or default configuration file, i.e. load, save, ... It contains an item for program end too.

- **Load –** loads the last-saved configuration file from the VoiceBlue program directory. Or requires loading of default settings upon the first run.
- **Save –** saves the current settings into a file in the VoiceBlue program directory (or in a folder defined in the List of gateways – refer to 10.4.2).
- **Load from –** loads a file from a folder selected by you.
- **Save as –** saves a file into a folder selected by you.
- **Default settings –** loads the preset default configuration.
- **Program end –** terminates the program.

10.4.2. Control menu, Gateway

Used for connecting / disconnecting the gateway. You can select a gateway from the List of gateways if you use remote control.

- **Connect gateway –** connects VoiceBlue with your PC and establishes mutual communication via an RS232 serial interface or using the Telnet protocol.

  Note: The gateway must be connected and the "Comm. settings " items have to be selected properly.
- **Disconnect gateway –** disconnects the gateway and discontinues its communication with your PC.

  If you communicate with more gateways than one, you are advised to have each configuration file in a different directory to avoid unintentional rewriting of the configuration file with another gateway configuration. To do this, use the following menu:

10.4.3. GATE menu

- **Select gateway from list** – select a gateway (directory with the configuration file) to be connected to the PC.
- **Connect gateway from list** – get connected to the selected gateway.
- **List of gateways** – edit the list of gateways including directories.
- **Terminal –** select the command-type control and work with the VoiceBlue (refer to section 11.1, page 41).

10.4.4. Control and Monitor menu

Contains commands for VoiceBlue (available only if the VoiceBlue gateway is connected).

- **Diagnostics –** information on boards (GSM modules and PRI board), contains 9 cards whose numbers correspond to those of VoiceBlue positions (00=PRI board).

Ethernet – contains information on statuses VoIP interface.

GSM – includes information on statuses and types of GSM modules on the GSM board.

- **Layer 2,3** – status of the module communication layers.
- **GSM network** – name of the network to which the module is currently logged in.
- **Network ID –** ID code of the network (MCC+MNC) to which the module is currently logged in.
- **GSM cell –** identification number of the GSM cell to which the module is currently logged in.

Description of displayed numbers:

NETCELL: A,BBB,CCC,DDDDD

A = status :
0 - not registered
1 - registered to home network
2 - not registered, but ME is searching for a new operator
3 - registration denied
4 - unknown
5 - registered to roaming network
BBB = first byte of LAC (location area code) in dec format
CCC = second byte of LAC in dec format
DDDDD = cell ID

- **GSM group –** number of the current GSM group.
- **Active SIM –** number of the active SIM card on this GSM module.
- **Module ID -** international identification code of the GSM module (IMEI).
- **Rev ID –** GSM module firmware revision number.
- **SIM Number –** SIM card serial number.
- **ID SIM card–** international identification code (IMSI) or SIM serial number (SCID) of the SIM card.
- **Signal intensity –** the current signal level in the network to which the module is logged in (minimum value = -113 dB – the module is logged out).
- **Block** button – blocking of the selected GSM module (unavailable for outgoing and incoming calls).
- **Down** button – used for blocking a module after the current call end.
- **Reset** button – hardware reset of a board or GSM module.

- **Info on current calls –** information on currently made calls. This information can be arranged according to the GSM module.
- **Connection status –** status of all possible remote/local VoiceBlue control ports.
- **Buffer status –** current buffer status (CDR). The maximum capacity is 511 calls.
- **Tracing –** item for storing traces from IP/GSM layers.
- **GSM Monitor info –** with this feature you can download information to the GSM cell where the GSM module is connected.

  Description of response parameters from a TC35i module:

Chann	ARFCN (Absolute Frequency Channel Number) of the BCCH (THC) carrier. If chann is h BTS supports hopping during connection.
Rs	RSSI (Received signal strength) of the BCCH carrier from 0 to 63. The indicated value is composed of the measured value in dBm plus an offset. This is in accordance with a formula specified in 3GPP TS05.08
dBm	Receiving level of the BCCH carrier in dBm
PLMN	PLMN ID code
LAC	Location area code (HEX)
Cell	Cell ID code (HEX)
NCC	PLMN colour mode
BCC	Base station colour mode
PWR	Maximal power level used on RACH channel in dBm or current power level
RXLev	Minimal receiving level (in dBm) to allow registration
C1	Base station selection coefficient
TS	Timeslot number
timAdv	Timing advance in bits
Q	Receiving quality (0-7)
Chmod	Channel mode (S_HR: Half rate, S_FR: full rate, S_EFR: Enhanced full rate)

- **Export statistics –** used for storing and clearing of statistic data from the VoiceBlue gateway connected. For a detailed description of statistic data refer to page 47.
- **Data into gateway –** sends and saves the configuration file into the gateway, the program then resets the gateway – for internal configuration updating – and saves the config.cfg file into the VoiceBlue program folder (or a folder pre-defined by you if you use the "Multi-gateway menu“ option).
- **Reset –** resets the gateway and initialises all VoiceBlue boards (the gateway communication is not discontinued but all current calls and ready-to-send SMS are terminated!).
- **Factory reset –** resets the gateway, selecting the factory (default) settings. The gateway communication is not discontinued upon this command.
- **Upload firmware** – uploads the firmware into the gateway.

FIRMWARE uploading procedure:

- Prepare the firmware-containing file into a folder selected by you (*Pxxxx-V-xx.xx.xx.bin).*
- Select and open the "Upload firmware" file.
- The program now automatically uploads your new firmware – the gateway is reset during this procedure (thus discontinuing all current calls and SMS). Do not interrupt the program during this procedure to avoid errors in firmware uploading – otherwise the gateway might stop working!

Should the procedure of uploading firm ware be terminated, please reset the gateway, re-load the PRIGW program and try to upload the firmware again.

CAUTION!: Be sure to use original and undamaged firmware files for firmware upload to avoid gateway function problems! For the latest firmware ve rsion see our websites (**www.2n.cz**).

- **Time and date –** sets time and date in the gateway.
- **Change username and password –** sets username and password for access to VoiceBlue via TELNET.
- **Download trace –** saves gateway operation and error records on the disk.
- **Save call data –** saves records on calls on the disk (the records remain in the VoiceBlue gateway).
- **Save call data and delete –** saves records on calls on the disk and clears the call memory.

10.4.5. Setting Menu

Contains communication (**see Section**) and language settings.

Communication

- **Direct to COM port –** program communication through an RS232 serial interface.
- **Modem –** communication through a connected modem.
- **TCP/IP –** communication through the TELNET protocol via Ethernet.
- **Transmission rate –** the recommended and default value is 57.600 bps.

10.4.6. Help Menu

Contains a help to the PRIGW program, instructions for help use and details on the program version.

10.5. Configuration

As already mentioned, the program includes the "Topics" and "Alphabetical glossary" folder menus. These menus contain identical items (as shown in the figures below) and it depends on the user which of them he or she chooses for easy orientation. You can set ISDN GSM gateway parameters in these menus.

10.5.1. Programming

10.5.1.1 System Settings

This window allows you to set the basic parameters of VoiceBlue.

- **IP address –** IP address of the VoiceBlue Ethernet port.
- **IP mask –** VoiceBlue port IP address mask.
- **Init sequence for modem –** an AT command sent by VoiceBlue via COM2 when it detects a connected modem.
- **Record calls –** records information on calls.
    - **No calls –** the gateway does not record any call information.
    - **Successful outgoing –** the gateway records information on all successfully connected outgoing calls.
    - **Successful outgoing+incoming –** the gateway records information on all successfully connected outgoing+incoming calls.
    - **All outgoing –** the gateway records information on all outgoing calls.
    - **All outgoing+incoming –** the gateway records information on all outgoing and incoming calls.
- **Record monitor info –** if this feature is activated, the gateway saves the NETmonitor information too in the CDR line. With this feature activated you have only a half of the memory for the CDR !!!
- **PIN code -** the PIN code that the gateway tries to enter if a PIN-active SIM card is inserted (this item has no meaning if a SIM card without an active PIN is used).
- **Gate ID –** a code that is added to the call data recording line. With this code you can identify more easily the VoiceBlue gateway on which the CDR was generated (in case you collect CDRs from multiple VoiceBlue gateways).
- **Dial for far control –** number for remote control via the DATA B-channel (this feature is optional).
- **Automatic logout of GSM modules -** this function is used for an automatic log-out of modules at a defined time. If logged-in modules are occupied by a call, logged-out modules log in to the network automatically. When the traffic decreases, the modules log out at random intervals.
    - **Enable automatic logout –** Enable / disable of this function.
        - **Logout Hour –** time when modules start to log out randomly from the GSM network.
        - **Login Hour –** time when modules start to log in to the GSM network.
        - **Min. logged-in modules –** the minimum number of GSM modules that remain logged-in.
- **Automatically move to summer/winter time –** for automatic summer/winter time switching.

10.5.1.2 ***Ethernet***

- Mode/Protocol – Select signalisation protocol
- Day of delete statistics – Day of automatic deleting of the statistics of VoIP interface
- Voice parameters, First and last RTP port – you have to set right range of used ports for RTP protocol (protocol for voice transmit).
- IP addresses – settings for SIP proxy server and for NAT firewall.
- SIP registration – additional settings for successful log-in to you SIP proxy server.

10.5.1.3 ***GSM Basic settings***

- **Number of digits dialled from VoIP –** sets outgoing VoIP calls into GSM networks.
  - **Maximum –** the maximum number of digits to be dialled into a GSM network. Any dialling longer than or equal to this parameter is dialled automatically (without waiting time).
  - **Minimum –** the minimum number of digits to be dialled into a GSM network.
  - **Waiting for next digit –** time (in seconds) for VoiceBlue to wait for the next dialled digit. After this timeout, the number is dialled automatically into the GSM network.
- **Call delay -** the minimum timeout between the end of a call and the beginning of another outgoing call for one GSM module (incoming and outgoing calls are not rejected during this timeout). The optimum time is 2 seconds. Unless absolutely necessary, do not change the default value to avoid system instability.
- **DTMF number –** sensitivity of the DTMF receiver from the GSM network. The "delay" parameter means 10*miliseconds added to the default value of 20ms. The total time is the minimum delay between two DTMF chars. Example: DTMF number=30 delay=30*10+20=320

10.5.1.4 ***GSM, Assignment to Groups***

The table of GSM module assignments to Outgoing (VoIP>GSM) and Incoming (GSM>VoIP) GSM groups.

10.5.1.5 ***GSM, Outgoing Groups***

Settings for outgoing GSM groups (and SIM cards*). In this section you can set all parameters for each SIM card for outgoing calls from the ISDN PRI to GSM networks.

Day of delete statistics in group – the xth day of a month on which the GSM group statistics is to be deleted.

Delay for send CONNECT (s) – delay for sending the CONNECT message to the ISDN PRI interface (in case VoiceBlue builds an ISDN-GSM call).

Timeout for send ALERTING(s) – timeout for sending the ALERTING message to the ISDN PRI interface (in case VoiceBlue builds an ISDN-GSM call); 0= switched off.

Roaming enabled for network – an international ID of the GSM network (MCC+MNC) that is enabled for roaming. Please leave it free if you don't want to enable roaming.

CLIR – This parameter determines whether a subscriber number on the gateway SIM card shall be dialled into the GSM network or not. All VoiceBlue outgoing calls present the calling line numbers as inserted in the VoiceBlue SIM card. Technically, it is impossible to transfer number information from the ISDN into the GSM network (the opposite function is supported, i.e. all ISDN subscribers can see the GSM calling number). Hence, it is mostly purposeful to restrict the calling line identification (activate CLIR) to avoid problems with backcalls to the gateway in the case of a missed GSM call, for example. Select this parameter for each GSM group separately.

- **Default -** settings according your GSM provider's network.
- **Enabled (CLIP-off)** –** the CLIR function is activated, no numbers are sent to the GSM network. **CAUTION!** Your GSM provider must support **and activate** this service! If not, the number will still be sent to the network and outgoing calls will be impossible with some providers.
- **Disabled (CLIP-on) -** the CLIR function is disabled, numbers are sent. Your GSM provider must support and activate this service! If not, the number will still be sent to the network and outgoing calls will be impossible with some providers.

**Some GSM networks (SIM cards) do not support CLIR activation via GM22 GSM modules, in that case you can see: Network: clir-err in the status window.

Max. of called minutes – call time limit for one month for one SIM card.
Number of SMS messages – number of SMS messages sent via a SIM card per month.
Day of restore limits – day on which the "call time limit" and "SMS" counts are reset to zero (1 to 31).
First count – length of the first count. After this timeout, VoiceBlue uses the "next count" parameter to calculate the length of the call (1 to 250 seconds).
Next count – length of the next counts (1 to 250 seconds).

The First count and Next count parameters are used for calculating the real length of the call from the viewpoint of the GSM provider.

Example 1: If your calls are charged on the second basis, set both parameters to 1.
Example 2: If your calls are charged on the minute basis for the first minute after answering and then by seconds, set the first count to 60 and the next count to 1.
A proper setting of these parameters helps you keep an actual record of minutes spent and charged for each SIM card. These parameters are used for limit counts and statistics (not for the CDR).

10.5.1.6 *GSM, Incoming groups*

Settings for incoming calls from GSM networks to the VoIP interface.

Incoming calls to VoIP– sets how VoiceBlue shall process incoming calls from GSM networks and route them to the PRI ISDN interface.

Mode – sets how VoiceBlue shall answer incoming calls from GSM networks.

- **Reject incoming calls -** by selecting this item you bar GSM incoming calls (the calling subscriber gets the busy tone*).
- **Ignore incoming calls –** by selecting this item you ignore GSM incoming calls (but the calling subscriber gets the ringing tone).
- **Receive incoming calls + voice message –** incoming calls are routed to the ISDN interface according to the following parameters.
- **Receive incoming calls + dialtone** – incoming calls are routed to the ISDN interface.
- **After ring callback / Refuse –** activated callback (for CLIP it is saved in the Autorouting/Callback table). Other calls are rejected.
- **After ring callback / Ignore** - activated callback (for CLP it is saved in the Autorouting/Callback table). Other calls are ignored.
- **Report to PC + voice message –** external callback (for callback centre) with a saved voice message.
- **Report to PC + dial tone -** external callback (for callback centre) with the dialtone.

Minimum digits in DTMF – minimum count of DTMF digits in DTMF dial-in.

Maximum digits in DTMF – maximum count of digits to be DTMF-dialled for incoming calls. An ISDN call is made automatically with the currently selected DISA (or DTMF prefix if necessary) after the last DTMF dialling.

Timeout while inputting DTMF digits(s) – time (in seconds) for VoiceBlue to wait for the first and next DTMF digits. After this timeout, either the DTMF number received so far is dialled into the ISDN or, if no DTMF digit is dialled, a number is dialled from the “List of dialled numbers”.
List of called numbers - list of numbers to be dialled automatically gradually (in the case of unavailability or busy) if no DTMF dialling (DISA) is made.

Prefix before DISA preselection – prefix added before the DTMF dial-in. For example: the DTFM digits received are 487 and this prefix is 6655. Hence, the gateway sends number 6655487 to the PRI ISDN interface.

CLIP - with this parameter you can replace the “+“ chars with different digit(s) in the CLIP. In case you leave this parameter blank, the gateway only removes “+“ from the CLIP.

Looping of DISA message – sets an automatic repetition of the DISA voice message in case of incoming DTMF dial-in.

10.5.1.7 ***GSM, Network List***

A list of prefixes for GSM providers.

Table of replaced prefixes – prefixes to be replaced (in VoIP – GSM calls) with other prefixes (e.g. +420 replaced with 0)*.

*make this change before you find the right prefix in the table of prefixes!

Table of prefixes – prefixes called from the VoIP. By filling-in this table you can set the length of the called number. If you do not complete this parameter, the VoiceBlue gateway uses the "Default number of digits" setting.
Network number – ID code of the GSM network for these prefixes*

*this parameter is used in higher firmware versions for pseudotariffication. Please fill-in this parameter.

-

10.5.1.8 ***GSM, LCR***

A table of outgoing Least Cost Routing (LCR) options. Every outgoing call VoIP interface is routed to the GSM according to this table. VoiceBlue checks line by line during every call and if the called prefix matches any item in the network list and the current call meets the time limits, the call is routed via the defined GSM group(s) or PRI 2 interface.

Network list number – number of the "Network list"
Time limitation – validity time of this line
Weekend – determines whether this line is valid on weekends and holidays.
Groups - GSM destination of outgoing calls for GSM group(s) (in case the first GSM group is unavailable, the call is re-routed via the next GSM group or cancelled) (without ACK setup).
Limit - maximum call length (in minutes). A call is disconnected after this timeout. The maximum value is 60 minutes. If you set a "0", the call length is unlimited (limited by the GSM network).

10.5.1.9 ***GSM, Autorouting / Callback Table***

A table for the Callback and Autorouting functions.
Callback – in case you activate callback in the GSM incoming group, you can add an authorised CLIP, which can use the callback function.
Autorouting – you can select a destination in the VoIP for each CLIP; these CLIPs are then automatically connected with the defined number on the VoIP.
Limit – maximum length of the call.

11. CONFIGURATION AND COMMUNICATION USING A STANDARD TERMINAL

You can communicate with the GSM gateway via an internal serial link or USB or TCP/IP connection.

The serial link communication requires a full crossed RS232 cable (laplink). The following communication parameters have to be selected too:

Communication rate	57600bps
Bits	8
Stop bits	1
Parity	None
Flow control	Hardware
File transmission protocol	XMODEM

The USB communication requires an active controller (enclosed), which installs a new virtual COM port in the Windows system to be used during communication.

The gateway only supports the TELNET protocol for TCP/IP communication (port 23).

The gateway behaves like an ANSI terminal with echo. Commands are text-entered and the Xmodem protocol is used for file transmission. An access password can be selected. The gateway requires the password with the “USER“ and “*PASSWORD“* prompts; if the password has been entered or is not required, the gateway sends the > prompt. The gateway uses an extended set of AT commands for configuration (see below).

11.1. Terminal Commands

By default, all commands start with AT.

A/	repeat last command (without AT)
I3	copyright & firmware
I4	serial number
&FRES	factory defaults & reset
&V	view active parameters (system)
&VE	view active parameters (ethernet)
&V0	view active parameters (common)
&V1	view active parameters (groups 1..4)
&V9	view active parameters (inc. groups)
&P	view pseudo params
&N#	view network params (net 1..8)
&NALL	view all network params
&A	view autorouting table <SPACEcontinue>
&R	view lcr-routing table <SPACEcontinue>
&TIN	view lan + groups + modules inc totals
&TOUT	view lan + groups + modules out totals
&G#=atcommand	send at command to gsm 1..4
&GALL=cmd	cmd for gate 1..4 (RESET,BLOCK,OFF)
&G#=cmd	cmd for gate 1..4 (RESET,BLOCK,OFF,ATBAUD)
&S	view lan + module status
&S=info	view all modules selected info (by at&S#)
&SL	view lan status
&S#	view one module status (1..4)
&QALL	view signal quality on all gsm
&Q#	view signal quality on gsm 1..4
&L	view logfile from recent <SPACEcontinue>
&C	view buffer (calls) from recent <SPACEcontinue>
&CR	read buffer from the oldest <#erase & continue>
&BSYS=cmd	cmd for system (RESET)
&X	view conn table/gsm (call states)
&X#	view call details/gsm (1..4)
&M0	disable ansi colors
&M1	enable ansi colors
&M9	start matrix
&U	view logged users

extended user commands

&EA	view arp table
&EA=cmd	cmd for arp table (RESET)

disk commands (under development)

&DI	view cfc disk info
&DC=cmd	cmd for cfc (RESET,FLUSH)
&DD[=dirname]	open/close dir ;list directory
&DF[=filename]	open/close file ;view ascii file

system params:

%S70=iii.iii.iii.iii	IP address
%S71=iii.iii.iii.iii	IP mask
%S72=iii.iii.iii.iii	IP router
%S91=buf,id	cdr mode (b0=outg, b1=inc, b2=failed, b3=moninfo)
	unit id (0=off, 1..255)
%S92=rep	report mode (b0=states, b1=tstamp, b2=smp,
	b3=lay2, b4=select)
%S98=pin	sim pin (max 7 digits)
%S99=dd.mm.yy.w/hh.mm.ss	set date/time (w=1..7 day of week)
%X20=mmdd,mmdd	date of hour+1,hour-1 time change (0=off,0101..1231)
%X80=login/pass	login name / password (max 15 chars all)

ethernet params:

%E00=xxx	protocol (==I00)
%E01=c1,c2,c3	codec list (18,8,0)
%E02=exp,rep	expire (sec >= 600), reattempt (sec >= 10) for sip.reg.
%E03=rtp1,rtp2	first (>=1024), last (>=first+10) RTP port, even only !
%E08=bits	bit1=ringing
%E09=day	day (0=off,1..31) of clearing stat
%E10=x.x.x.x	SIP proxy for calls IP-->GSM
%E11=x.x.x.x	SIP proxy for calls IP<--GSM
%E12=x.x.x.x	MGCP gateway
%E14=x.x.x.x	SIP registrar
%E16=x.x.x.x	NAT firewall
E80=name/pass	registration name / password (max 31 chars all)

group params:

%G00=xxxx	out.group numbers for g1..g4 (0=off, 1..4=group)
%G01=0,dspo,dspi	dsp output,input gain (1=-31dB, 32=0dB, 63=+31dB)
%G02=mode,atms,afms	tc35 mode (2,4)
	atms/afms gain (+5dB=3,+2.5dB=1,0dB=0,-2.5dB=2,-5dB=4)
%G06=mmdd,..mmdd	holiday list (0101=1st jan, 1231=31st dec)
%G07=mmdd,..mmdd	holiday list2
%G08=delay,min,max,tout	gsm call delay (0..10 sec), dial min/max (0..20)
	dial tout (0..20 sec)
%G09=scn	sim card number (0=imsi,1=scid)
%G#1=netid,clir,min,sms,day,sec,sec2,pseudo	sim #1 params
	netid (7 chars), clir (0=netw,1=on,2=off)
	min (0=0ff,1..65535 minutes), sms (0=off,1..65535)
	day (0=off,1..31), sec/2 (1..250), pseudo (0=off,1..8)
%G#9=ale,conn,disc,day,lim	alerting tout (0=off,1..20sec), conn delay (0..20sec)
	forced disc (0=off, 1=on sim limit, 2=on sim or time limit)
	day (0=off,1..31) of clearing group stat,

call durat. limit (0=off, 1..99 min)

%G90=xxxx inc.group numbers for g1..g4 (0=off, 1..4=group)
%G9#=mode,min,max,tout,day,dial,clip params #=1..4 for inc.groups 1..4
Mode (0=reject,1=ignore,2=ok-message,3=ok-tone)
min(0..20), max(0..20) tout (0..20 sec) dtmf dial-in
day (0=off,1..31) of clearing inc.group stat
dial prefix (max 15 ch), clip prefix (max 7 ch)
%G9#=xxx,xxx,xxx auto dials (max 63 chars) #=5..8 for inc.groups 1..4

pseudo params:

%P01=uuu/HH:MM, pseudo tariff 6x cents/until (cents=1..9999)
(until=00:00first item is all weekend)
(until=24:00last mandatory item)
%P02=uuu/HH:MM, pseudo tariff extension
%P03=uuu/HH:MM, pseudo tariff extension
%P04=uuu/HH:MM, pseudo tariff extension
%P05=uuu/HH:MM, pseudo tariff extension
%P06=uuu/HH:MM, pseudo tariff extension
%P07=uuu/HH:MM, pseudo tariff extension
%P09=mode,sec pseudo mode (0=off,1=cdr,2=cdr+isdn), isdn.sec (1..250)

network params:

%N#0=opx/npx, list of old/new main-prefixes (max 47 chars)
%N#1=pref/dig, list of prefixes/digits-to-end (max 63 chars)
%N#2=pref/dig, pref. list extension (max 63 chars)
%N#3=pref/dig, pref. list extension (max 63 chars)
%N#4=pref/dig, pref. list extension (max 63 chars)
%N#5=pref/dig, pref. list extension (max 63 chars)
%N#6=pref/dig, pref. list extension (max 63 chars)
%N#7=pref/dig, pref. list extension (max 63 chars)
%N#9=netid,max network id (7 chars), default max digits (0..20)

routing params:

%A##=clip/dial set autorouting item (ix 0..95)
clip (20), dial(20)
%R##=net,hh:mm/hh:mm/w+-,groups,lim set lcr-routing item (ix 0..63)
net (1..8), groups (max 7 chars=1..9)
call limit (0=off, 1..99min)

; totals
; first m,c,s inc. minutes,calls,sms
; second m,c,s out. minutes,calls,sms
; ri,ro redirected inc,out calls

%TL=m,c,m,c init minutes,calls in lan (0..65535)
%TG#=m,c,ri,ro init minutes,calls,rin,rout in group # (0..65535)

%TGALL=m,c,ri,ro	init minutes,calls,rin,rout in all groups (0..65535)
%TI#=m,c	init minutes,calls in inc.group # (0..65535)
%TIALL=m,c	init minutes,calls in all inc.groups (0..65535)
%TM#=m,c,s,m,c,s	init minutes,calls,sms in mod # (0..65535)
%TMALL=m,c,s,m,c,s	init minutes,calls,sms in all modules (0..65535)

service AT commands:

!V0FB=key	set system key
!RE	report errors only
!RR	start report layer2..4 on COM1 (from COM2)
!RX	stop report layer2..4 on COM1 (from COM2)
!R#	report messages 1=layer1..4 2=layer2..4 3=layer3..4 4=layer4
!L#	report lan 2=ip/arp..telnet 3=tcp/udp..telnet 4=telnet
!P	view process info
!Q	view system info
!SB##=ddd,ccc	callback to GSM/ccc (0..31,32=by prefix) & PRI/ddd (cyclic)
!M=cmd	cmd for message system (ERASE;RECORD;READ;WRITE)

spec gsm commands:

at&g##=at+cnum	view own number
at&g##=at+cpin="####"	set pin (before pin checking off !!!)
at&g##=at+clck="SC",0,"####"	pin checking off (####=PIN)
at&g##=at+clck="SC",1,"####"	pin checking on (####=PIN)
at&g##=at+cpin="****"	set pin (before pin changing pin !!!)
at&g##=at+cpwd="SC","****","####"	pin change (****=old, ####=new PIN)
at&g##=at+cacm?	accumulated call meter
at&g##=at+camm?	maximum call meter
at&g##=at+cpuc?	call meter currency/unit

*some of the above mentioned AT commands may not be available in the current firmware version.

11.2. Handling with SMS

Commands for SMS sending and receiving	
AT!G=A6	Start low-level control of SMS (can run on one port only)
AT!G=55	Stop low-level control on used port
	SMS control
AT^SX=ch	…(sms listing) request to list all SMS messages and status confirmations saved on SIM card. Possible answers:: *smserr (busy,list) or *smsinc (ix=1..255) for each saved SMS or status SMS , end of list or empty SIM card - *smsinc (ix=0).
AT^SR=ch,ix	…(sms read) request to read an SMS message or SMS status saved on SIM card. Possible answers: *smserr (busy,read) or *smspdu
AT^SD=ch,ix	…(sms delete) request to delete an SMS message (or SMS status message). Possible answers: *smserr (busy,delete) or *smsdel
AT^SM=ch,len,pdu,csum	…(sms to module) request to send a message via GSM module 0..31 or via any GSM module (ch=32). Possible answers: *smserr (busy,write) or *smsout
AT^SG=grp,len,pdu,csum	…(sms to group) request to send an SMS message via GSM group 1..8. Possible answers: *smserr (busy,write) or *smsout
	Messages from VoiceBlue unit
***smsinc: ch,ix,sts**	… SMS received and saved onto SIM card: • Ch …GSM module number 0..31 • Ix …index number of saved SMS messages 0..255 • Sts …status of SMS message
***smsrep: ch,ix**	…SMS status confirmation received and saved to SIM card (this message is only for GSM modules TC35 and GM47)
***smsout: ch,ix,ref**	…SMS message sent and not saved onto SIM card: Ref …reference number of sent SMS 0..255 (will be used in SMS status confirmation message)
***smspdu: ch,ix,sts,len,pdu,csum**	…content of SMS message or status confirmation: • Len …length of SMS message (number of bytes in pdu) • Pdu …content message in PDU format • Csum …Checksum of all PDU bytes (2 hexa digits) calculated without carry
***smsdel: ch,ix**	SMS message or status confirmation was deleted from position ix
***smserr: ch,ix,req,err**	response to error command: • Req …required GSM module or GSM group • Err …error code (6=busy, 40=write, 41=read, 42=delete, 43=list)

11.3. Operational Records (LOG)

Type	Text	Description
POWER	[Power on]	System switched on
	[Power off]	System switched off
	[Warm boot]	Restart of system, unknown cause
	[Watchdog]	Restart of system by watchdog
	[BKPT code]	CPU error: break code detected
	[Stack error]	CPU error: stock integrity failure
	[Divided by zero]	CPU error: dividing by zero
	[RETI code]	CPU error: illegal using of instruction reti
	[NMI intr]	CPU error: wrong interrupt
	[VOID intr]	CPU error: wrong interrupt
	[Upgrade reset]	Start of upgrade firmware procedure
	[Software reset]	Reset by AT commads (at&fres…)
INIT	Eeprom	Initialisation of eeprom (configuration)
	Flash	Initialisation of flash memory (firmware)
HW-ERR		(##…address of chip, RD…read value/WR…expected value)
	Codec ##,RD/WR	Error in initialisation of codec on GSM,AUX board
	COM2 #####,RD/WR	Error in initialisation of COM2 on AUX board
	Duart #####,RD/WR	Error in initialisation of serial controller on GSM board
	Hscx #####,RD/WR	Error in initialisation of HDLC controller on AUX board
SYSERR	User stack error!	SW error: stock integrity failure
BRDIN	#08 TYP STS	board inserted (number of gsm board, type, status)
BRDOUT	#08 TYP STS	board disconnected
BRDRES	#08 TYP STS	Reset of board by AT command
	ALL GSM RESET CMD	Reset of all GSM boards by AT command
	SYSTEM RESET CMD	Reset of system by command at&bsys
G2-ERR	ATD/ERROR init (g##)	Error of layer 2 isdn: restart of module g## after rejected command ATD by GSM network
	GSM Cause 150 (g##)	Error of layer 2 isdn: restart of module g## after cause 150 was received (call barred by GSM network)
G3-ERR	tout sts # (g##)	Error of Layer 3 isdn: timeout in status # on module g##
C4-ERR	tout sts # (p##/g##)	Error of Connecting layer 4: timeout in status # on call between channel p## and GSM module g##

11.4. Call Records – Description

Example of a successfully connected call:

** **31.07.02/11:07:53** **O-OK** **CAU-016** **aux/g02** **GRP-1** **0:23** **001:40** **00000.00**
1 **0608218005** **45456060** **1/8942019636000065750**

- **1. column**: **
- **2. column**: call start date/time
- **3. column**: call type
  - I-FD : unconnected incoming call attempt(will be implemented in higher firmware versions)
  - I-OK : successfully connected incoming call (will be implemented in higher firmware versions)
  - O-FD : unconnected outgoing call attempt
  - O-OK : successfully connected outgoing call
- **4. column**: CAUSE sent to VoIP

- **5. column:** number of used VoIP -channel/number of used GSM module
- **6. column:** used GSM group (C= callback)
- **7. column:** call establishing time
- **8. column**: call duration mmm:ss (max. 255:59) or error cause for unconnected calls
- **8. column**: call cost (will be implemented in higher firmware versions)
- **9. column** : gateway id (optional)
- **10. column**: called number
- **11. column:** caller number
- **12. column**: number of slot/IMSI* of used SIM card

11.5. Statistics - Description

[Group call statistics]

gsm	(reset)	minutes	hhhh:mm:ss	calls	reject	failed	c.offs	errors
#i1 inc	(31.12)	0	0:00:00	0	0	0	0	0
#i2 inc	(31.12)	0	0:00:00	0	0	0	0	0
#i3 inc	(31.12)	0	0:00:00	0	0	0	0	0
#i4 inc	(31.12)	0	0:00:00	0	0	0	0	0

gsm	(reset)	minutes	hhhh:mm:ss	calls	reject	failed	red.in	redout
#g1 out	(31.12)	0	0:00:00	0	0	0	0	0
#g2 out	(31.12)	0	0:00:00	0	0	0	0	0
#g3 out	(31.12)	0	0:00:00	0	0	0	0	0
#g4 out	(31.12)	0	0:00:00	0	0	0	0	0
#g5 out	(31.12)	0	0:00:00	0	0	0	0	0
#g6 out	(31.12)	0	0:00:00	0	0	0	0	0
#g7 out	(31.12)	0	0:00:00	0	0	0	0	0
#g8 out	(31.12)	0	0:00:00	0	0	0	0	0

[Statistics of incoming calls on all modules]

modules	brd	minutes	hhhh:mm:ss	calls	sms	minutes	hhhh:mm:ss	calls	sms
#00 #01	00	0	0:00:00	0	0	0	0:00:00	0	0
#02 #03	01	0	0:00:44	1	0	16	0:16:37	10	0
#04 #05	02	14	0:14:15	7	0	5	0:05:31	3	0
#06 #07	03	4	0:04:21	3	0	0	0:00:00	0	0
#08 #09	04	0	0:00:00	0	0	0	0:00:00	0	0
#10 #11	05	0	0:00:00	0	0	0	0:00:00	0	0
#12 #13	06	0	0:00:00	0	0	0	0:00:00	0	0

*each line is for two GSM modules

[Statistics of calls on GSM module #0]

sim/dir	net/grp	minutes	hhhh:mm:ss	calls	reject	failed	c.offs	sms
#1 inc	/1	14	0:14:15	7	0	2	9	0
#2 inc	/1	0	0:00:00	0	0	0	0	0
#3 inc	/1	0	0:00:00	0	0	0	0	0

#4	inc	/1	0	0:00:00	0	0	0	0	0

#1	out	/1	439	7:19:51	177	0	6	44	0
#2	out	/1	0	0:00:00	0	0	0	0	0
#3	out	/1	0	0:00:00	0	0	0	0	0
#4	out	/1	0	0:00:00	0	0	0	0	0

- Pri/grp : call type
- Reset : Last statistic reset date
- Minutes : Number of minutes
- Hhhh:mm:ss : Same number converted to time
- Calls : Number of calls
- SMS : Number of sent SMS messages
- Reject: number of unconnected calls (no free GSM module available - call rejected with cause 41(42))
- Failed: number of unconnected calls (rejected by GSM network)
- C.offs: number of unconnected calls (terminated by calling party)
- Errors: number of unconnected calls (wrong requests - disallowed prefix etc.)
- Red.in: number of connected calls (re-routed to this GSM group)
- Redout: number of connected calls (re-routed to another GSM group)

12. TECHNICAL PARAMETERS

GSM

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

Antenna

Frequency	900 / 1800 MHz
Impedance	50 Ω
Max. power output	2W
Antenna connector type	SMA (male)
Cable length	3-10 m or without cable

Power supply

Adapter	230 V±10%, 50/60 Hz / 12V DC
DC power	12 V DC / 1.2 A
Supply connector	DC jack, 2.1 mm
Lithium battery	CR2032

VoIP

Signalling	SIP
Count of channels	4
Codecs	G.711 PCM at 64 kbps
	G.726 and G.727 E-ADPCM at 16 to 40 kbps
	G.723.1 (optional) MP-MLQ/ACELP at 6.3/5.3 kbps
Echo cancellation	G.168-2000; max. echo length 25ms (15 ms for G.729A)
VAD/CNG	G.729B or G.729A coders
(Silence Suppression)	G.723.1A for G.723.1 coders
PCM Companding	A-law/u-law (selectable)

Interface

RS232	
Connector	D-Sub 9 contacts - slots
Transmission rate	57600 bit / s
USB	
Connector	B type
	USB 1.1
Ethernet	
	RJ45
	10BaseT

Remote supervision	Ethernet (telnet)

Others

Dimensions (without connectors)	250 x 150 x 50 mm
Operating temperature	0°C to 40°C
Relative humidity	5 to 95% (non condensing)