ALPHATECH TECHNOLOGIES s.r.o.
Jeremenkova 88, Praha 4, Czech Republic
www.alphatechtechnologies.cz

BlueGate SIP 1

VoIP GSM GATE

Installation and setup guide V 1.0

1. General description

1.1 Technical parametres

Dimensions	100 x 130 x 37 mm
Weight	850 g
Operating position	various
Operating condition	temperature: +5°C ÷ +40° C
	humidity: 10% ÷ 80% při 30° C
Power supply	9-15V dc or 8-12 Vac, 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…
- 1 VoIP channel (1 IP adress)
- 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
- Connection with DTMF dial in
- connection directly to the operator or designated line
- connection to operator after time out (adjustable) for direct dialing in
- transmit of calling part number (CLIP)
- incoming call restriction from GSM network
- outgoing call restriction from GSM network
- Smart Callback – automatic incoming calls routing up CLIP
- Echo canceller – switching ON/OFF
- 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

1.2 Installation

Insert SIM cards.

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

SIM holders will be released after pressing of yellow button

Connect GSM antenna
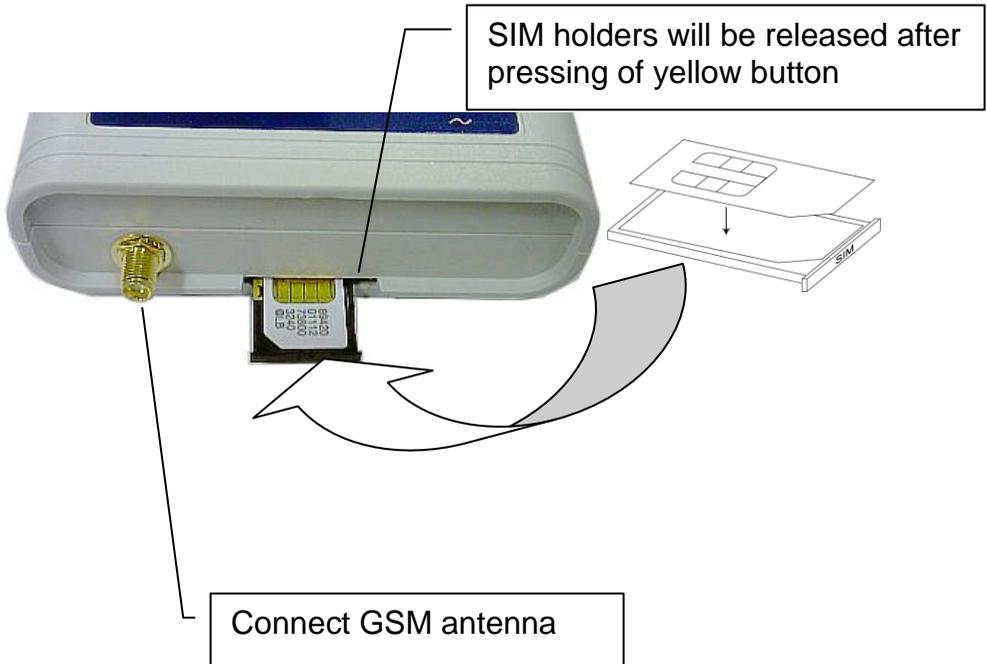
Finally, connect the power adapter

Connect the LAN cable
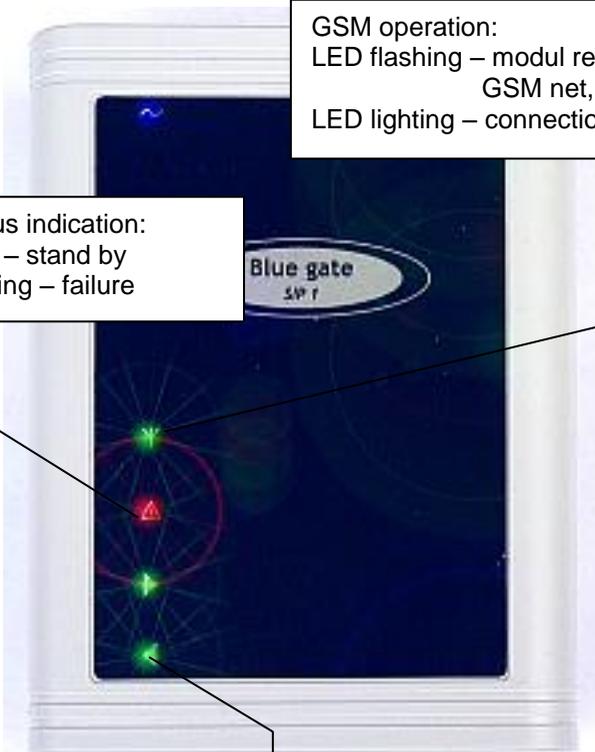
GSM operation:
LED flashing – modul registrate to GSM net, stand by
LED lighting – connection (call)

Error status indication:
LED is off – stand by
LED flashing – failure

Blue gate
SIP 1

Incoming and outgoing call

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On PC which is connected to LAN, you can control connected Gateway via various web broser. Insert IP address of gate (default 192.168.1.250) and you can continue in setting of parametrs.

1.3 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. switch is for default GSM parameters. For normal Gateway operation set this switch to ON position.

Note: ON position is on the right side

1.4 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 internal or external SIP server. 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.
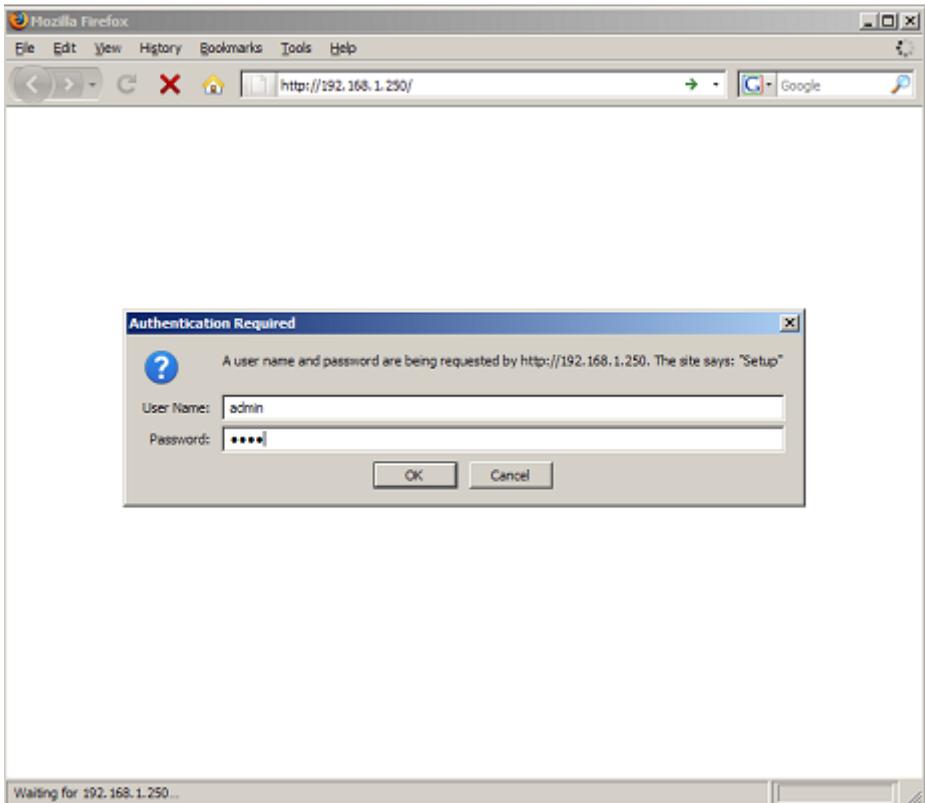
1.5 language option
Language - can be chosen in the menu on the left panel.
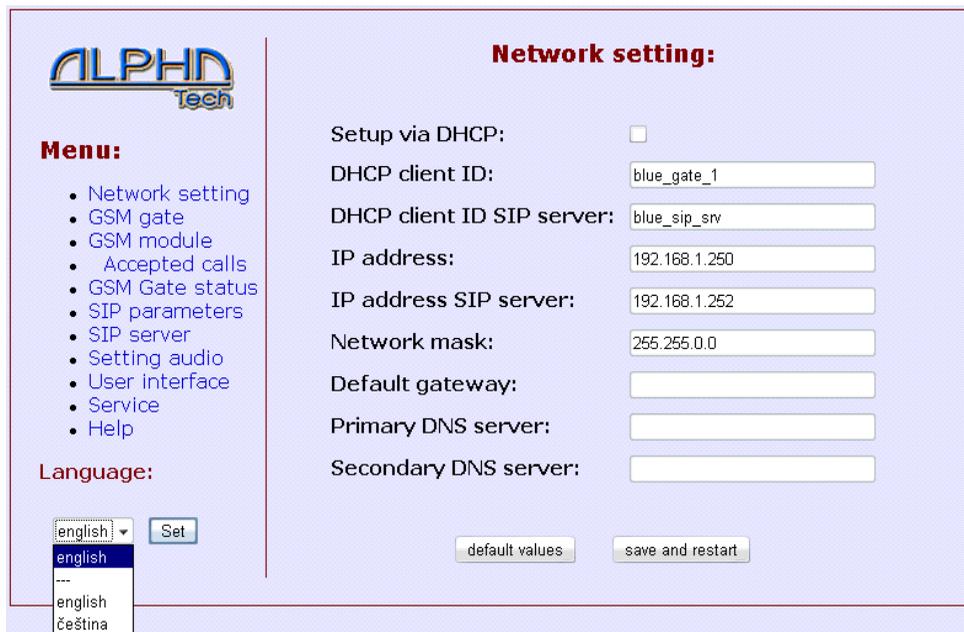
Network setting:

Setup via DHCP:	☐
DHCP client ID:	blue_gate_1
DHCP client ID SIP server:	blue_sip_srv
IP address:	192.168.1.250
IP address SIP server:	192.168.1.252
Network mask:	255.255.0.0
Default gateway:	
Primary DNS server:	
Secondary DNS server:	

Menu:
- Network setting
- GSM gate
- GSM module
- Accepted calls
- GSM Gate status
- SIP parameters
- SIP server
- Setting audio
- User interface
- Service
- Help

Language:

english ▼   Set
english
---
english
čeština

default values   save and restart

2. Gateway setup

After making changes click on a **save and restart** or **save changes** button. For default data restoring click on a **default values** button.

2.1 Network setting

- **Setup via DHCP** - it is possible to use DHCP service or you can enter IP address manualy.
- **DHCP client ID** - host identification for DHCP server address allocation
- **IP address** - address of GSM gates. In P2P mode make a call to this address. If use internal or external SIP server, make calls to address of SIP server.
- **IP address SIP server** – is used for internal SIP server. If it's enabled all clients (phones) make calls via this address.
- **Network gateway and DNS** - servers are needed only if you make connections or register via internet. If GSM Gateway is used only on local network, leave them empty. Network gateway is not address of internal or external SIP server usually.

Network setting:

Setup via DHCP:	☐
DHCP client ID:	blue_gate_1
DHCP client ID SIP server:	blue_sip_srv
IP address:	192.168.1.250
IP address SIP server:	192.168.1.252
Network mask:	255.255.0.0
Default gateway:	
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Language:

english ▾ Set

[default values] [save and restart]

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2.2 GSM Gate

- **DISA** - if a number of digits in the direct dialling code is greater than zero, the callers may dial directly using tone selection to the required extension.
- **Wait** - sets, how many seconds the automatic operator waits for direct dialling code selection. If it is not selected, the pre-selected number of the operator is dialed again.
- **Erase Clip** - 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).
- **Echo canceller** - this feature restricts ECHO.
- **Add zero before CLIP** - it adds 0 before each incomming number.

2.3 GSM Gate - module

- **PIN** - set up the PIN of SIM card in modul. When you use SIM card without PIN then this window is not used..
- **Subscriber number** - here is set a customer number of operator which will be dialed by Gateway at an incoming calling.
- **Volume GSM, Volume VoIP** - allows call volume adjusting in both direction.
- **Incoming calls, Outgoing calls** - permission of incoming/outgoing calls
- **Smart callback** - store all outgoing calls which has been missed or refused. When called part is calling back then the call is automatically routed to extension which made the call. When connection was succesfull data are erased from memory.
- **0, CLIR** - it adds 0 or code for Switch OFF outgoing CLIP before each outgoing dialled number
- **Roaming** - enable Roaming.

2.4 Accepted calls

If the table is not filled, all calls are enabled. Up to 12 blocks of permitted calls can be selected. 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 digits from the right when is 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!

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Permitted calls - module:

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Language:

english ▼   Set

Called number

	min	max
1.		
2.		
3.		
4.		
5.		
6.		
7.		
8.		
9.		
10.		
11.		
12.		

default values   save changes

2.5 GSM Gate status

- **Operator** – name of network operator
- **Signal strength** - 113 to -99 dBm is very bad signal, -98 to -83 dBm is bad signal, -82 to -71 dBm is good signal and -70 to -51 dBm is very good signal.
- **Bit error rate** - less value means the signal is better.
- **IMEI** - unique number assigned by GSM part producer.
- **GSM Firmware version** - name of SW version for GSM part.

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Language:

english ▼ Set

GSM Gate status:

Operator:	T-Mobile CZ
Signal strength:	-51 dBm
Bit error rate:	0.2 %
IMEI:	357820022453708
GSM Firmware version:	VER6.0

2.6 Number translation

The gate must be set to the P2P mode. In this mode it is important to set a translation of numbers into IP addresses. Item „Number" is a phone number, IP address is an address of the selected phone.

2.7 SIP parameters

The module must be set to the mode with SIP server. Next, internal SIP server must be deactivated. After the saving of entered data, registration attempt is executed (if name is not empty) and result is shown. If the registration is not successfull, you will see the reason in Registration log in menu Services

- **SIP proxy server** - enter IP address or the name and port (if differ from default 5060) of SIP server to send registration or route calls.
- **SIP registrar server** - it isn't necessary fill if it is identical with SIP proxy server
- **Name and Password** - are not mandatory but must be set exactly same as on SIP server machine.
- **Use (180 Ringing)** - signalling of incomming call by default Ringing.
- **Use (183 Session progress)** – the change on Session progress is added for SIP proxy servers which require it.
- **Enable Simmetric RTP** - added for SIP proxy servers which require it.
- **Send DTMF as SIP INFO** - Outband DTMF as SIP INFO.
- **Send DTMF according to RFC2833** - Outband DTMF in "RTP channel ".

2.8 SIP Server (internal)

A module must be set to the mode with SIP server.

- **Enable SIP server** - activation/deactivation of the internal SIP server
- **Server name (realm)** - the name is sent from SIP server to client as the SIP server name at make registration or call.
- **Prefix** - 1 to 3 digit number which is prepended before called GSM number from IP phone. If leave it empty, call via appropriate GSM Module will not be possible.
- **Client numbers and passwords** - must be entered the same as in client settings, in other case client will not be able to register itself on internal SIP server and make a call.

2.9 Setting audio
In case of poor audio quality try another codec priority. In menu 'Setting audio' choose G711µ as first, G711a as second, ... Check client (IP Phone) audio settings. At least one audio codec must be chosen.
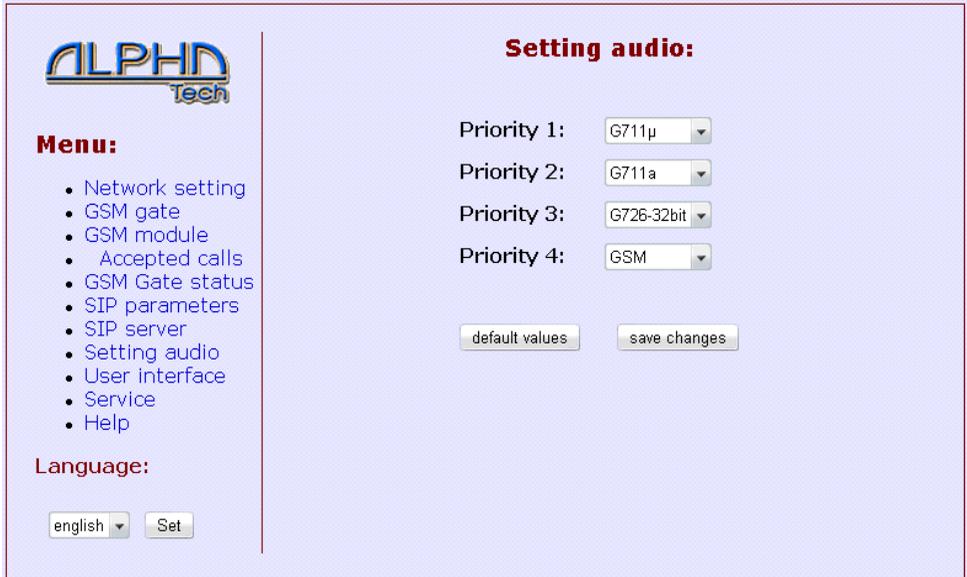
2.10 User interface

- **Web interface tcp port** - possibility change Web interface network port.
- **Enable telnet** - possibility disable / enable access over telnet.

User interface:

Web interface tcp port: `80`

Enable telnet: ☐

default values   save and restart

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Language:

english ▾ Set

2.11 Admin services

- **VoIP version** – name of SW version for VoIP part
- **Download log file** - if you are in troubles and need technical support, you will need this file. Follow these steps:
  1. press '**start enhanced log**' button
  2. do the problematic action, this action will be recorded step by step into log file
  3. click on **'Download log file'**, save the file and send it to technical support.
- **Show call log** - record of few last incomming and outgoing calls. In case of error, reason is also shown.
- **Show register log** - registration on SIP server, in case of error reason is shown. Registration is always done in two steps. Firstly a client sends the request to server and server responds with its realm, in second step the client sends identity based on realm of server and server responds with success or access denied.
- **Show VoIP log** - start VoIP monitor (online log file).
- **Time server** - IP adress of NTP server with time data. (If doesn't know address of NTP, you use * and system automatic choose acceptable.)
- **Syslog server** - computer able of receiving internal messages from device
- **Firmware upgrade** - customer is allowed to load new versions of firmware into device. Here you also load customizations. If upgrading firmware is in progress do not power off device, you risk the device gets unusable! After firmware upgrade the pressing of reset button is allways required.
- **Upload language** - possibility to supply custom language. Name of this file appears in Language combo box. HTML entities in file name are allowed. Language file must start with line cs or en - the language from which thist customization is derived.
- **Save configuration** - save actual setting of all gateway parameters to file
- **Upload configuration** - restores setting of all gateway parameters from file.
- **Service password** - possibility to change the manufacturer default password.

Admin services:

VoIP version: 1.53

Download log file   start enhanced log
Show call log
Show register log
Show VoIP log

Time server:

[   ] [GMT-13 ▾]   [save]

Syslog server:

[   ]   [save]

Firmware upgrade:

[   ] [browse]   [save]

Upload language:

[   ] [browse]   [save]

Save configuration:

[save]

Upload configuration:

[   ] [browse]   [save]

Service password:

[   ]   Retype password: [   ] [save]

[restart]

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Language:

[english ▾] [Set]

2.12 Help
Here you can find informations about Gateway configuration and problem solving

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3. Troubleshooting

Many of problems is possible to successfully solve by cooperation with technical support. In this case tech. support needs exact and clean description of your problem and log file downloadable from menu Support. Follow these steps:

1. press '**start enhanced log**' button
2. do the problematic action, this action will be recorded step by step into log file
3. click on **'Download log file'**, save the file and send it to technical support

3.1 Registration

- registration is unsuccessful
  - go to menu Service and click 'Show register log', record of registration attempts and results will appear
- registration is unsuccessful - in registration log is reason: Timeout
  - SIP server is unreachable, check SIP server address setting in menu 'SIP parameters'
  - check network connection and SIP server running
- registration is unsuccessful - in registration log is reason: 404 (Not found)
  - check IP address of SIP server, port and registration name in menu 'SIP parameters'
- registration is unsuccessful - in registration log is reason: 'Unauthorized' or 'Access denied'
  - registration is allways done in two steps, in first step client obtain server's realm and result is 'Unauthorized', in second step client sends authorisation and should be successful. Two step registration is OK.
  - registration name (number) and password are not mandatory, but must be set exactly same as on SIP server
  - look into SIP server machine log, you may find interesting things

- registration is unsuccessful - nothing helps
  - in menu Service download log file and send it with problem description to technical support

3.2 Call
- connection of call is not possible
  - in menu Service click on 'Show call log', window with call record and possible errors will be displayd
- called number in call log is not desired GSM number
  - called number must be same as desired GSM number, if it differs, your SIP server is misconfigured
- in call log I see 'Bypass SIP server'
  - fix properly client setting, it must call via SIP server, not GSM Gate IP address directly
  - try set IP address instead of host (domain) name for SIP server in menu 'SIP Setting'
- in call log I see 'Unsupported media type'
  - in 'Setting audio' choose as priority one codec G711μ, second G711a and so on.
  - check the client setting (phone), must have codecs G711μ or G711a enabled
- call is not possible, or is early lost, nothing helps
  - in menu Service download log file and send it with description to technical support

3.3 Audio
- poor quality
  - in 'Setting audio' choose as priority one codec G711μ, second G711a and so on
  - check the client setting (phone), must has codecs G711μ or G711a enabled
- poor quality persists
  - try other codec combinations
- audio setting is OK, but still poor quality
  - you may try record network traffic of call by analyzer named wireshark (download from www.wireshark.org)

- this record with description and log file from menu Service send to technical support

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Jeremenkova 88
140 00 Praha 4
Czech Republic
VAT: CZ27577350

Company is registered in the Commercial Register
administered by the Municipal
Court in Prague, Section C, Record 116886

Banking details:
Komerční banka, account No. 43-7671450207/0100
IBAN: CZ0801000000437671450207
SWIFT: KOMBCZPPXXX