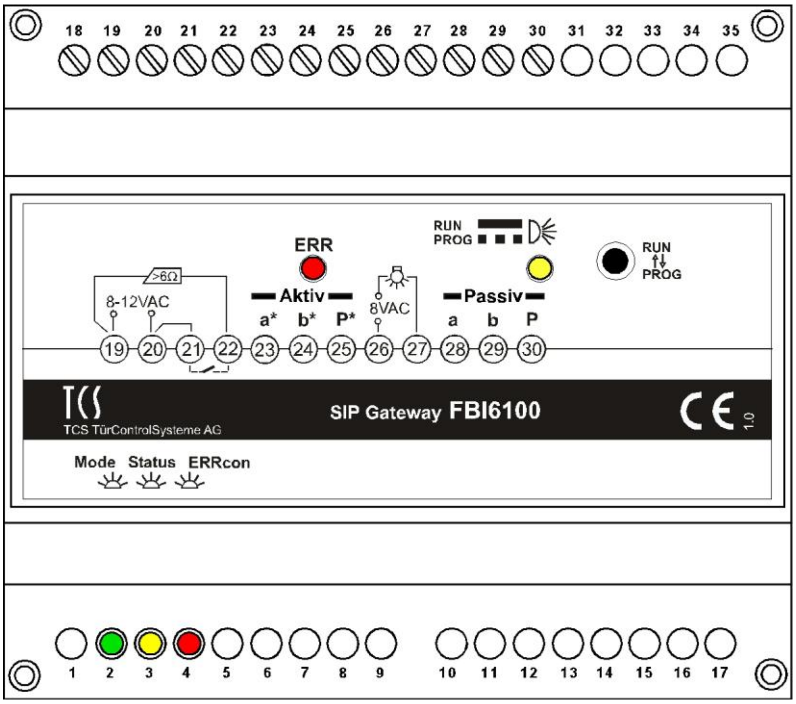
User manual

SIP-Gateway

for 115 call destinations **FBI6100-0400**

for 10 call destinations **FBI6101-0400**



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Approvals and certificates

The SIP-Gateway has a CE-test certificate.
TCS TürControlSysteme AG is certifictaed according to ISO 9001:2008.

General

Basics on VoIP and SIP

The transmission of voice and image via ethernet and IP is more and more replacing analog and ISDN-based transmission techniques.
Thus, voice is transferred into digital signals (*VoIP = VoiceOverIP*) and transported via the IP network in a special type of protocol, such as **SIP** (*Session Initiation Protocol)*. When an IP camera is used, the digitized video image data is also transmitted via the same network.

SIP versus H.323
In addition to SIP-systems, also systems with protocol **H.323** are used. Simplified, this method can also be called *ISDN over IP*. However, this method requires a highly efficient hardware and has some disadvantages regarding firewalls and grid integration.
The SIP-Gateway exclusively supports the forward-looking SIP-protocol.

Advantages of SIP-protocols
Companies with already existing IT-networks and structured cabling gain essential advantages by using this new technique. Existing IT-ressources can be used, an additional supply network is not necessary.
When no supply connection exists, indoor stations can also be operated via **W-LAN** (IP radio connection).

SIP-server / SIP-provider
The *SIP-server* realises the arrangement between the individual participants. Therefore, each participant resp. each end device has to register at a SIP-server with its user data. After a successful arrangement, the communication data is automatically transmitted directly from participant to particiant. In turn, the end of the communication is transmitted to the SIP-server.
SIP-server and *SIP-provider* differ only in their shape and location.
A SIP-server is an IP telephone system or a telephone system software which is installed on a PC. Those are existing physically in the network.
A SIP-provider only makes the SIP-Server functionality available via an internet connection. The SIP-provider is mostly operated by an internet provider and is located there.

Arrangement and connection variants for calls

IP telephone systems

Many manufacturers of telephone systems offer devices with SIP-ability.
There are different variants depending on the number of users and different communication quantities. These are ususally fitted with integrated hardware to lead over to S2M (primary rate access), ISDN or analogue connections.

SIP-enabled DSL-router (VoIP-router)

For applications with 1 to 5 users, depending on the number of telephone conversations also more or less users are possible, SIP-enabled DSL-router are used (e.g. AVM Fritzbox with VoIP). Those usually also have an integrated telephone system, thus enabling the transition to an ISDN or an analogue connection.

Local SIP-server software

These software solutions can be operated on any PC.
For example, the 3CX-SIP server software (aslo avaialble as freeware) is working directly under windows. An alternative is Asterisk, which can be used on a Linux PC.
The software itself resp. the performance and load of the PC limits the max. number of users. With the corresponding hardware, the transition to an ISDN or an analogue connection is possible.

SIP-provider, public VoIP-provider

Depending on bandwidth of the existing internet connection, also pure web services can be used as SIP-server. SIP-Gateway and IP telephone are connected by a public provider (usually free of charge). Transitions from SIP to land line and mobile phone networks are offered and deducted to favourable conditions.
Thus, the front-door station and your IP telephone become a telephone that can be used all over the world.
Even switching functions, e.g. to release doors, can be triggered on the way.
Many telecommunication providers, who offer a combined DSL and telephone connection, usually make the telephone connection available as a pure VoIP connection (DSL-telephony). Those are mostly working according to SIP-standard.

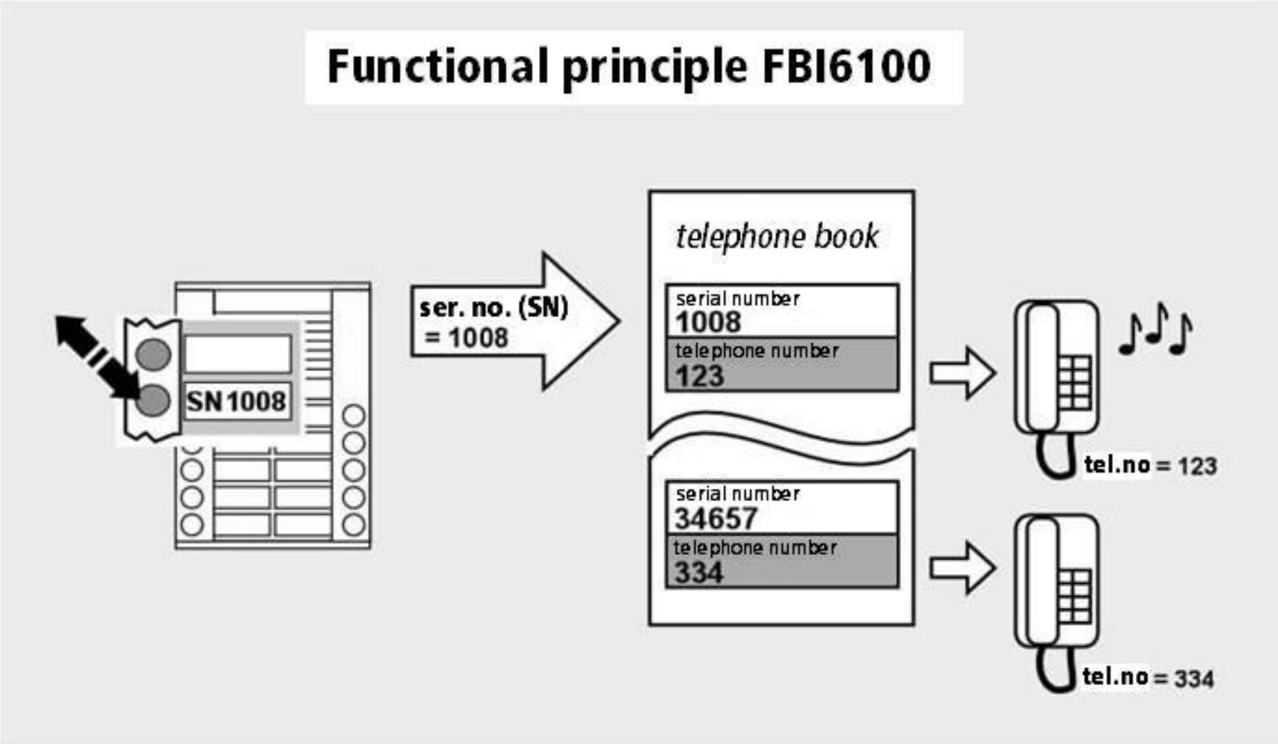
Direct connection (front-door station – switch – IP telephone)

The direct connection is the simplest application without a SIP-server, e.g. in a residential building.
The SIP-Gateway is directly calling the IP address of an IP telephone. Calls are only possible within a local network.
The used telephone has to support the direct calling of IP addresses in order to call the SIP-Gateway or to receive calls.

Product description

Functional principle

The SIP-Gateway is an interface for the VoIP-connection from TCS front-door stations via the *Session Initiation Protocol* (SIP). The SIP-Gateway is acting as a client in the SIP-network. Thus, VoIP-connections to other SIP-clients can be established via the SIP-Gateway.



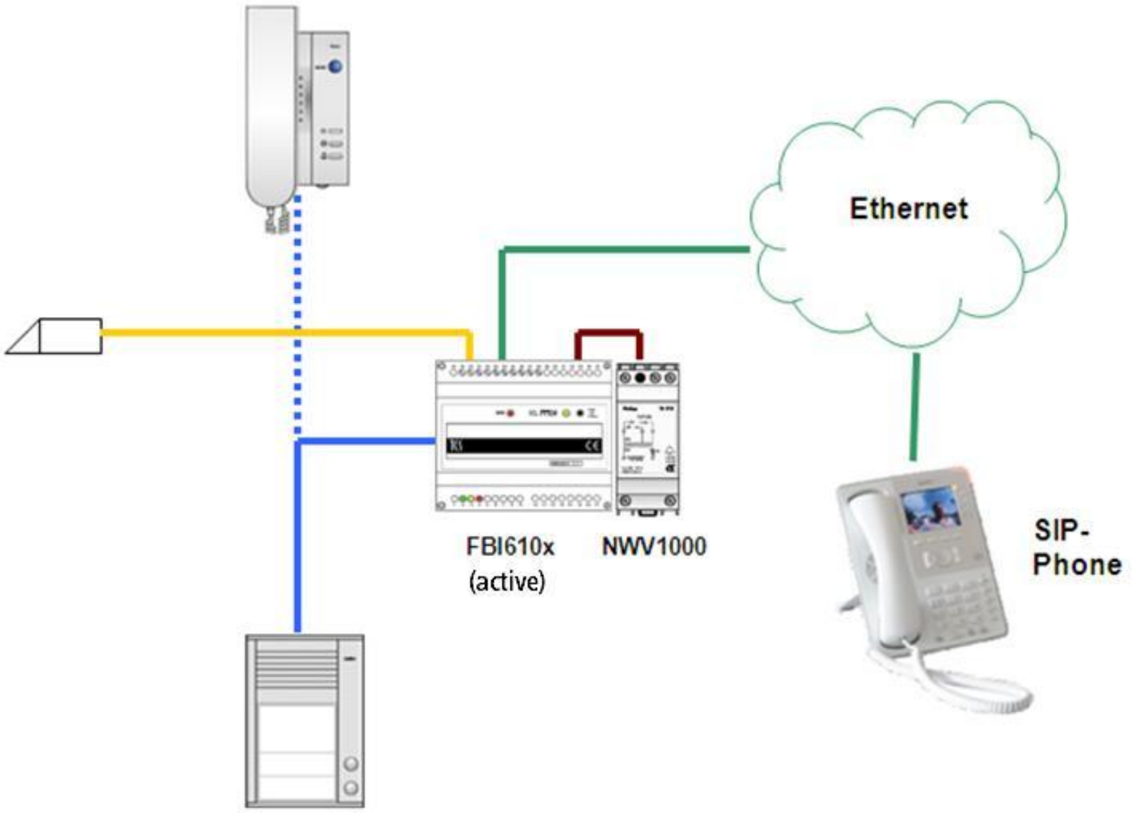
By using the programmed call buttons of the connected door station, a call to an allocated remote station can be triggered. For this, the SIP-Gateway manages a corresponding list of telephone numbers (telephone book) which are assigned to TCS:BUS serial numbers. From the remote station, the communication can be terminated early via DTMF-commands during a conversation. It is also possible to trigger the door opener or operate the light switch relay.

Operating modes at the TCS:BUS

The SIP-Gateway can be used in 2 operating modes at the TCS:BUS: *ACTIVE* and *PASSIVE*. The terminal groups for both operating modes are separately located from one another.

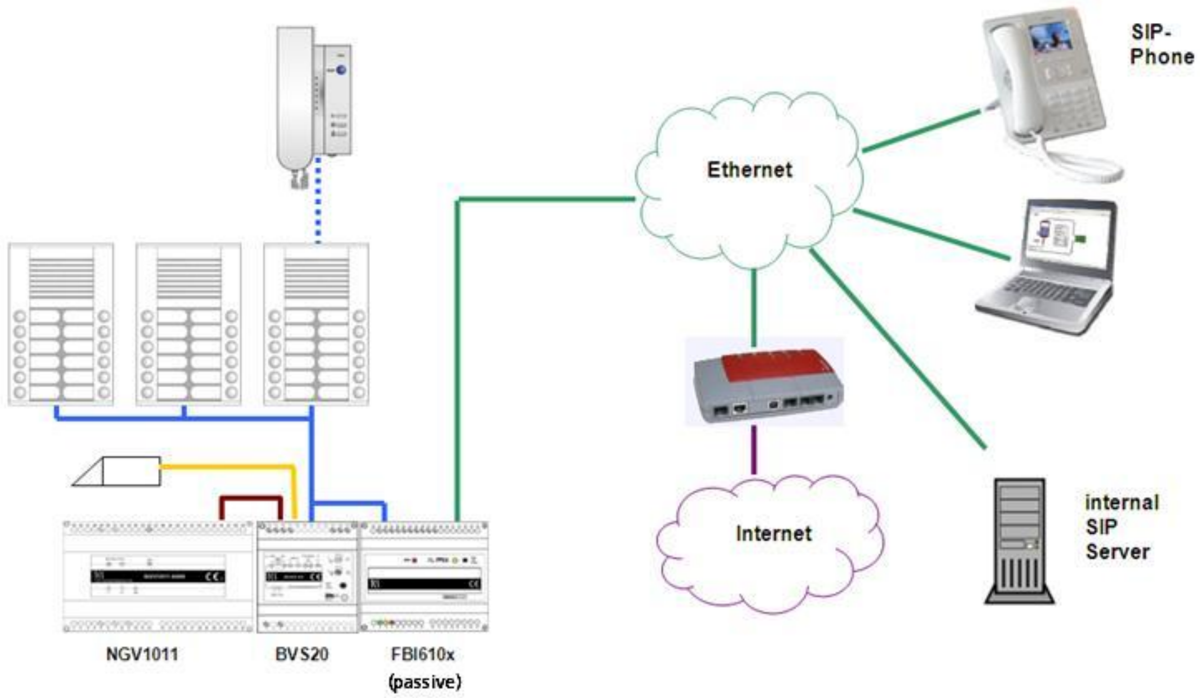
Operating mode *ACTIVE*

Within the operating mode *ACTIVE, the S*IP-Gateway supplies a BUS line where max. one audio front-door station (incl. extensions) and 3 audio indoor stations can be operated. The supply of the SIP-Gateway and the TCS:BUS with the connected devices is realised by a bell transformer 8-12V AC (NWV1000-0400). In this operating mode, a front-door station can be connected to a network with a minimum of effort. The selection of the connectable devices must be realised in accordance with the max. quiescent current output of the FBI610x.



Operating mode *PASSIVE*

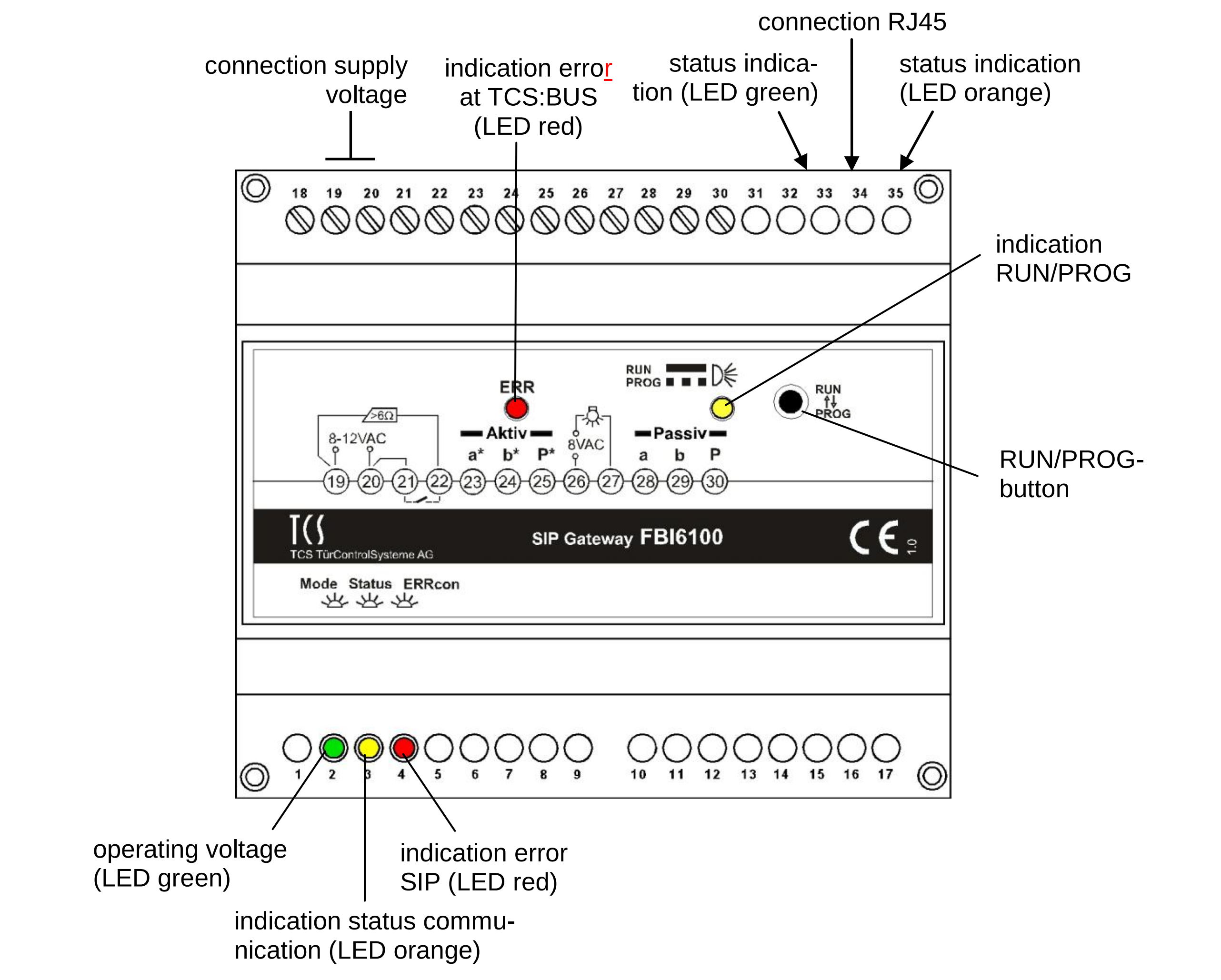
Within the operating mode *PASSIVE,* the TCS:BUS is realised via an additional supply and control unit. The SIP-Gateway is only a passive user at this BUS and is also supplied by the TCS:BUS via the P-terminal.
This operating mode is selected when the SIP-Gateway is integrated, e.g. into a video system, a system with several front-door stations resp. the system, which has to be supplied exceeds the output of the SIP-Gateway.



Ethernet (network)

In order to function, the SIP-Gateway needs an Ethernet connection. In the simplest case, the ethernet can be realised via a standard DSL-router. In this case, a SIP-server in the internet of any provider can be used as a server. Alternatively, the SIP-Gateway can also be connected directly to a VoIP telephone system.
In larger systems, a local server within the intranet can also be used as SIP-server. In this case, an internet connection is not necessarily required.
As indoor stations, SIP-IP-telephones, PC (or panel PC) with VoIP software (softphone) and conventional telephones (e.g. in VoIP telephone systems or VoIP adapter) can be used.
If the router has a WLAN function, resp. if corresponding access-points are integrated into the network, also wireless IP telephones can be used, if they're supporting the SIP-protocol.

Device overview



Connections, display and operating elements

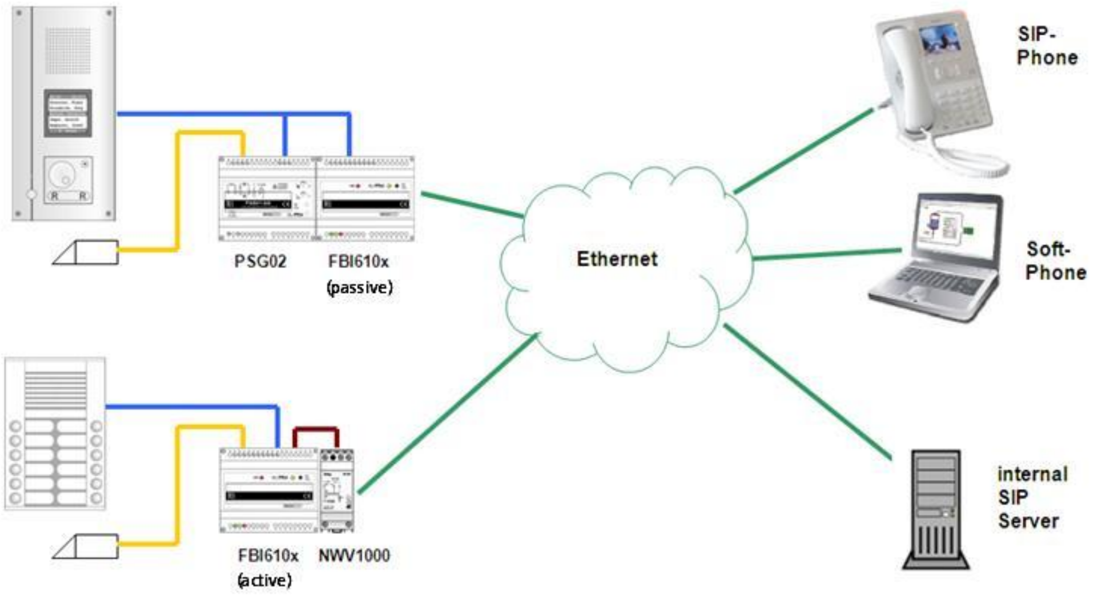
	name	function
	connection supply voltage	• **only operating mode ACTIVE:** 8 - 12 V AC (from bell transformer NWV1000-0400)
	a*/b*/P* connection TCS:BUS ACTIVE	**Only for operating mode ACTIVE**! The SIP-Gateway provides the supply voltage for a BUS line. • In this operating mode, a front-door station can be connected to a network with a minimum of effort. • An audio front-door station (incl. extensions) and 3 audio indoor stations can be operated at the BUS line. • The selection of the connectable devices must be realised in accordance with the max. quiescent current output of the FBI610x. • terminals are short-circuit protected

	a/b/P connection TCS:BUS PASSIVE	**Only for operating mode PASSIVE**! The supply voltage for the TCS:BUS is provided by an additional supply and control unit. • This operating mode is selected, when the SIP-Gateway is integrated, e.g. into a video system, a system with several front-door stations resp. the system, which has to be supplied exceeds the output of the SIP-Gateway. • terminals are short-circuit protected
	door release relay (potential-free relay contact – closing contact)	• door release voltage via jumper terminal 20 to 21 • 12 V, 50/60 Hz / 2 A (for door opener not less than 6 Ohm) • adjustable door release time: 0 s to 99 s • function can be adjusted via web interface
	internal light switch relay (potential-free relay contact - closing contact)	to control automatic light switching units, connectable are: • automatic light switching units (max. permitted contact load capacity of 24 V DC / 1 A) • staircase light switching unit FNA1000 (or TZ1-SG) (with 8 - 24 V AC) • relays • adjustable light switch time: 0 s to 99 s • function can be adjusted via web interface
	RJ45-socket	• connection for network (PC/laptop)
	RUN/PROG-push button	**Only for operating mode ACTIVE**! • switch the system mode: operating mode - programming mode
RUN PROG	indication RUN/PROG (LED orange)	• indication of the system mode: **is ON**: operating mode **blinking**: programming mode
ERR	Indication error at TCS:BUS (LED red)	• **flashes**: error in connection to the TCS:BUS **flash OFF**: error in the network connection (see page 35)
2	indication operating voltage (LED green)	• **is ON**: operating voltage exists
3	indication status communication (LED orange)	• indication communication or communication establishment FBI610x to external user
4	indication error SIP (LED red)	• **is ON**: start-up (ca. 40 s) or no connection to SIP-server (see page 35)
	status indication (LED green)	• **blinking:** if data is transmitted or received via LAN • **is ON**: connection FBI610x with network exists
	status indication (LED orange)	• **is ON** simultaneously with status indication green**:** no connection to the network

Application / installation examples

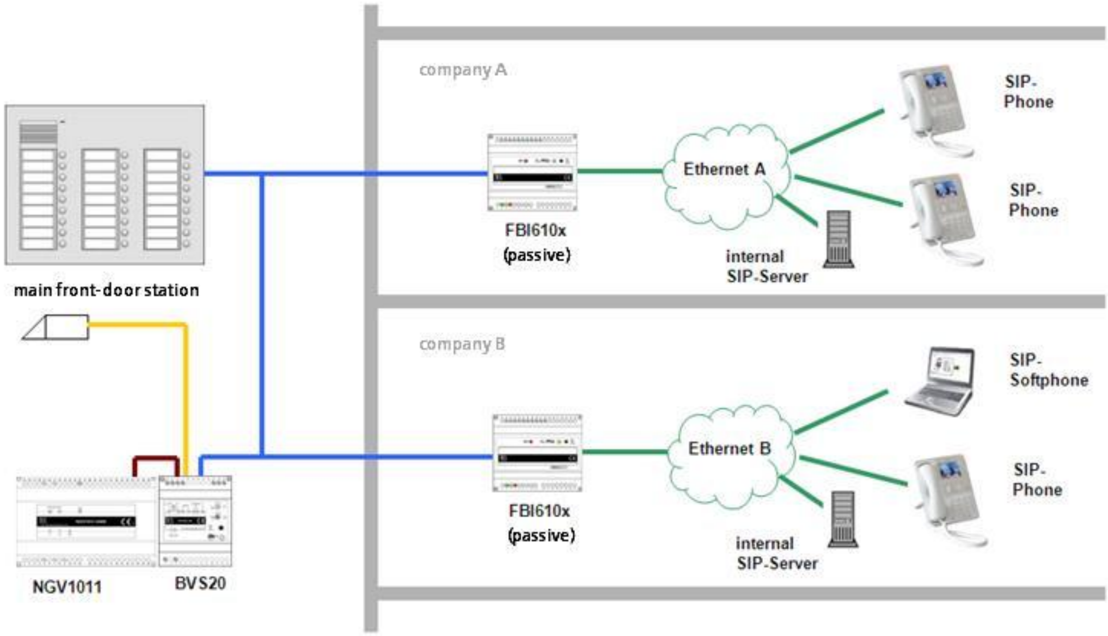
Installation with a network (industrial or residential building)

Every TCS front-door station is connected via a SIP-Gateway with the network.
Front-door stations can establish parallel communications.



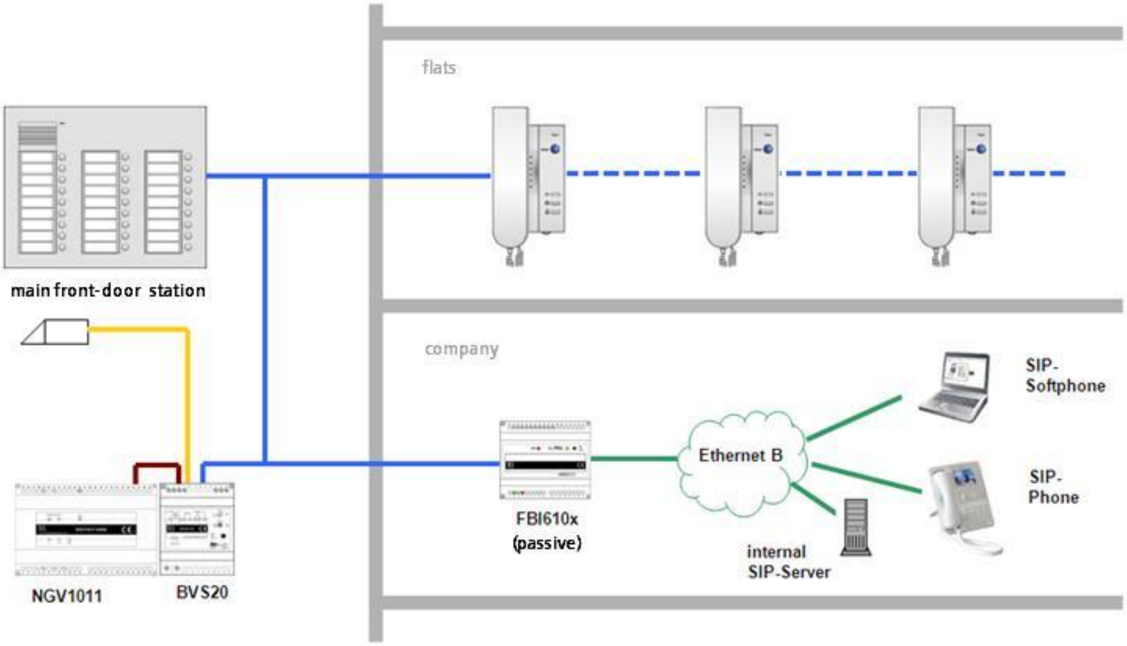
Installation with separated networks (office building)

The TCS:BUS with one or several front-door stations is connected via a SIP-Gateway with the network. The networks stay separated from one another.



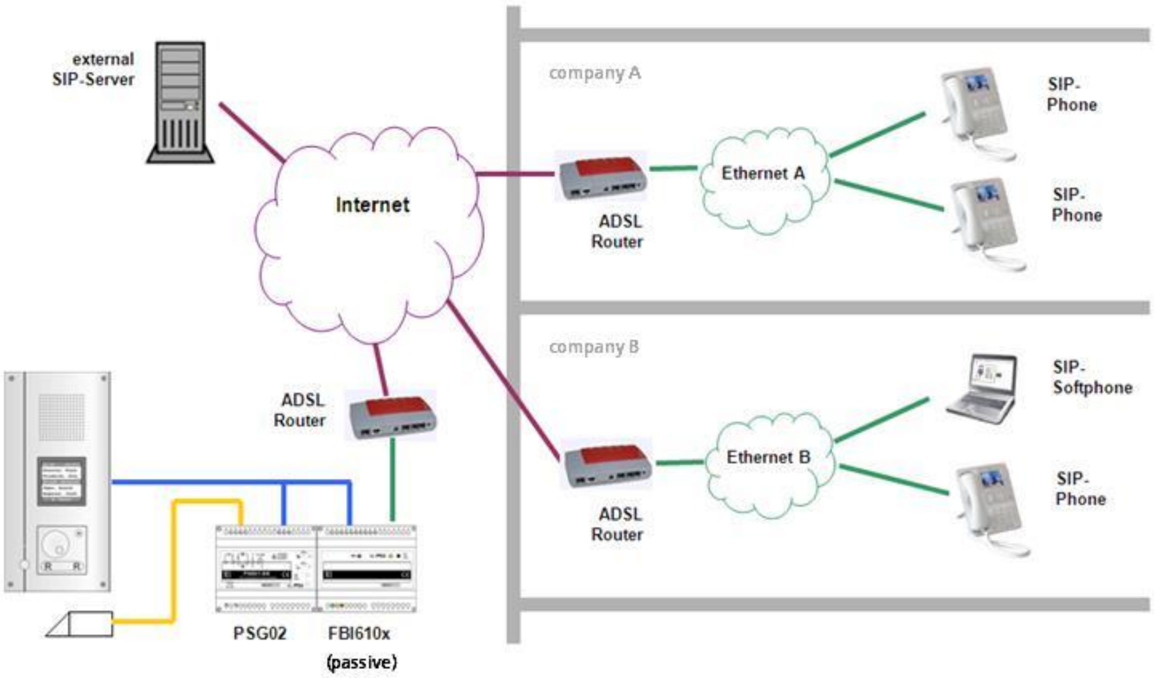
Installation with indoor stations and network (mixed systems)

SIP-Gateway and indoor stations are operated in one installation.



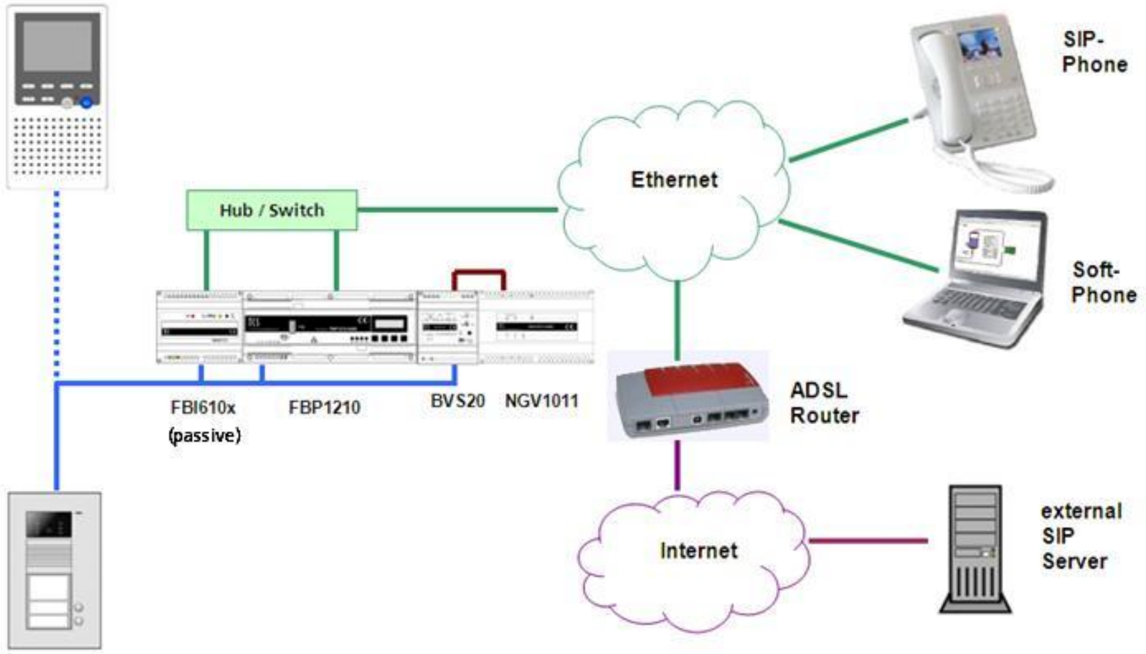
Installation at separate locations (internet connection)

The internet is used to transmit the communication data, SIP-Gateway and call destination are located in different networks.



Installation with TCS:Server (in combination with video)

Combination of TCS:Server and SIP-Gateway, the video image is transmitted parallel to the speech.



Commissioning

Pre-configuration

In order to connect to the SIP-Gateway after the installation, an IP address in the address area of your network has to be assigned to the gateway.
For configuring the SIP-Gateway (web interface), the PC/laptop must have a network connection. Further, it has to be located in the same subnet like the SIP-Gateway. If your network and the SIP-Gateway are working in the 192.168.1-address area (C-net), you can access it directly (as long as another device isn't using the IP address of the SIP-Gateway).
Otherwise, the PC/laptop has to be adjusted to a temporary IP address within the address area of the FBI610x to establish a connection with the SIP-Gateway.
In case that several FBI610x, which are in delivery condition, are to be installed in a network, the devices have to be commissioned successively and a free IP address has to be assigned.
In delivery condition the pre-settings at the FBI610x are:

IP address: 192.168.1.200, net mask: 255.255.255.0

To set/configure the network connection, please use the network installation assistant or the help of your operating system (keyword: *setting the network*).

Example: network configuration under Windows XP

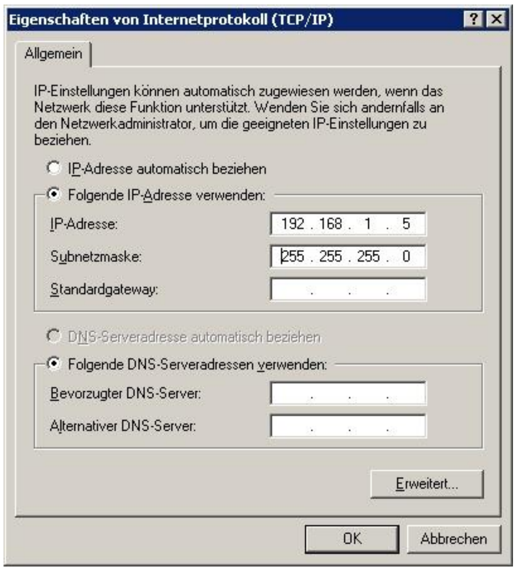
Open *network and DFÜ connections* by one of the two ways:

1. Click with the right mouse button on the icon *network environment* on your desktop.
   Chose *characteristics*.
   or
2. Click on the *Start*-button (task bar of your desktop) > *settings* > *control panel* > *network connectivity*.

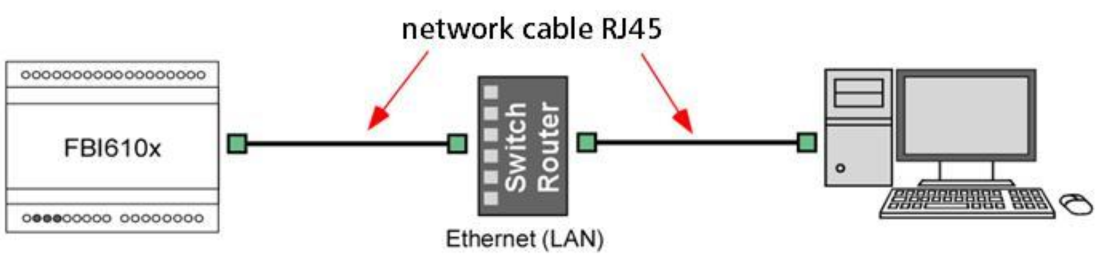
- Click in the menu with the right mouse button on *LAN connection*.
- Click on *settings*.

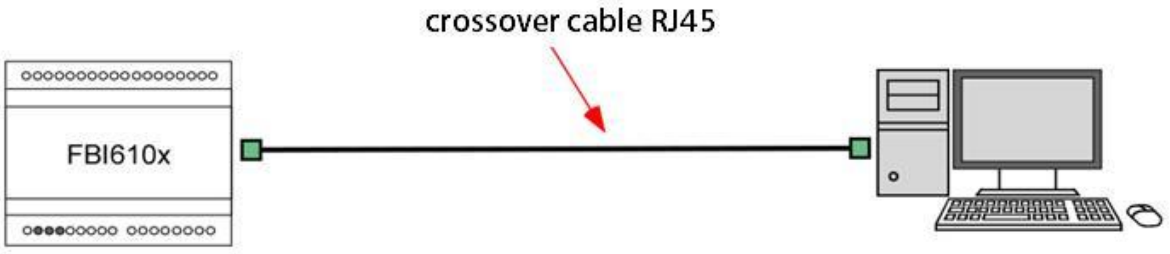
- Click on *internet protocol (TCP/IP)* (see illustration).
- Click on *settings*-button.

- Activate: *use following IP address*.
- Enter an IP address, which only differs in the last digit from the one of the TCS:Server, e.g. 192.168.1.5.
- Enter the number of the subnet mask: 255.255.255.0.
- Confirm with the *OK*-button.



Establish a network connection for configuration





- establish a connection from PC/laptop via switch or router to SIP-Gateway.
  or
- connect the SIP-Gateway directly with a PC/laptop via a crossover cable (twisted-pair-cable).

Connection / Commissioning

!	Mounting, installation and commissioing have to be realised by a qualified electrician!

- Connect the SIP-Gateway according to the supplied product information and put it into operation.

Configuration: introduction

Starting conditions

- Connection of the SIP-Gateway with an ethernet 10/100 LAN.
- The SIP-Gateway is supplied with voltage.

Start configuration menu

- Open an internet browser.
- Enter the IP address of the SIP-Gateway into the address line: http://192.168.1.200
- The start page is called up.

Configuration: telephone book

Telephone book

In this menu, a TCS:BUS serial number is connected with a SIP-telephone number, to be called.

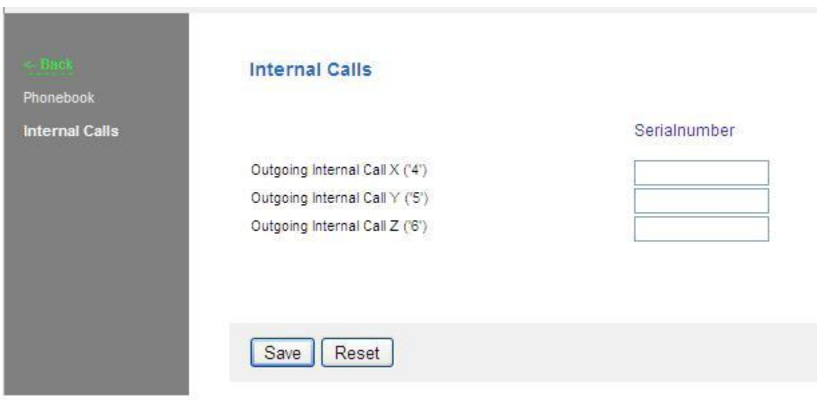
- Register under *telephone book*.
- Enter your required call destinations for your connected front-door stations.
- Click on *Save*.

! The button *reset* resets the content to the last stored condition. No factory setting is loaded.

name	This text is to inform the electrician resp. to display the allocation of call and serial numbers to a resident or a flat. There is no indication of this text at a different location. The field is mandatory.
number	Enter the SIP-telephone number that is to be dialed without special characters. In case of a direct connection, the IP address of the call destination is entered here.
serial number	TCS:BUS serial number, which has to be programmed at the front-door station. The serial number has max. 6 digits. Shorter serial numbers are entered and indicated without prefixed zeros.
incomming	Decision, how to react in case of an incomming call from a certain number. The following can be selected: reject, automatic acceptance, automatic acceptance with autoplay of a recorded message (the caller is hearing) and manual acceptance. The additional option ”beep“ ensures that the dialogue partners are hearing a signal tone after the audio connection is established.
outgoing	Decision, if in case of outgoing calls, a recorded message (e.g. location of the indoor station) should be played for the called dialogue partner.
group-ID	Via the group-ID, 10 different call chains can be defined. Telephone book entries with the same ID are handled as call chain. If an entry of this group is called by a front-door station and the call is not accepted, all entries with the same ID will be called one after the other.

Internal calls

- Enter the reqested internal call destinations, between which you want to choose when calling from the telephone to the SIP-Gateway.
- Click on *Save*.



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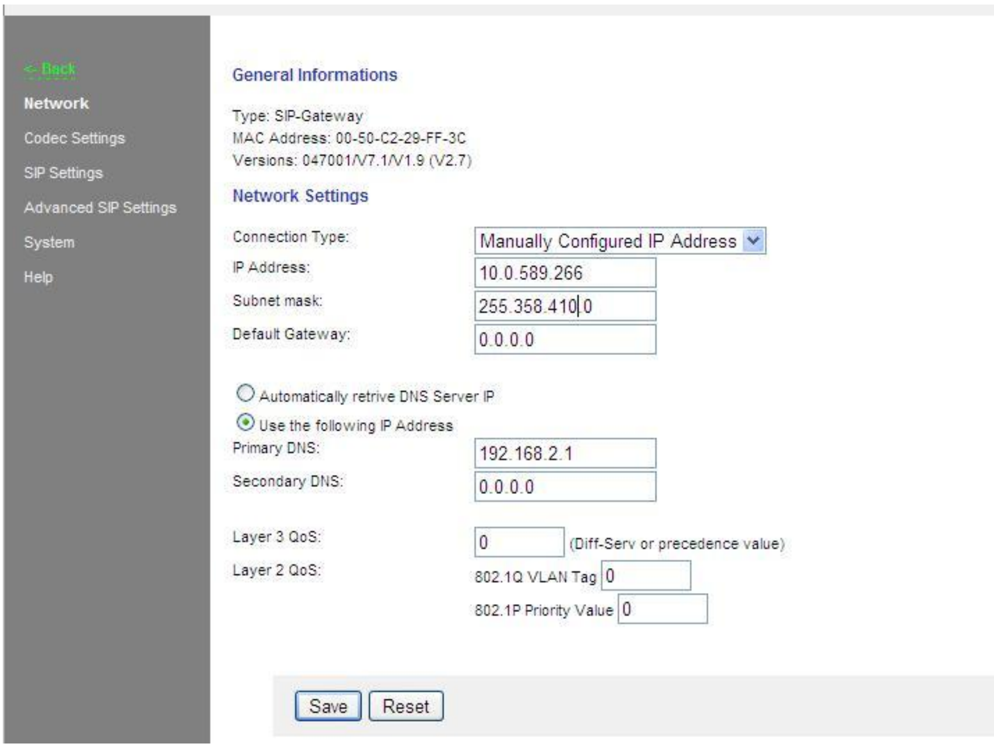
Outgoing internal call	To the serial numbers which are registered there, an internal call can be sent from the connected SIP-telephone via the DTMF commands '4', ‚5' or '6'. If nothing is entered, instead of the serial number with max. 6 digits, the corresponding internal call is deactivated.

Configuration: VoIP settings

network	The corresponding entries for the network in which the SIP-Gateway will be integrated have to be implemented here.
voice	Selection and setting options for the used method to code and decode the network voice transmission. Modifications, if they are necessary, should only be carried out by experienced system administrators.
SIP-settings	The access data of the SIP-server / SIP-provider has to be entered here.
extended SIP-settings	Extended settings, if necessary. Modifications should only be carried out by experienced system administrators.
system	system settings such as password, time zone etc.

Network



- Click on *Save* to save the settings.

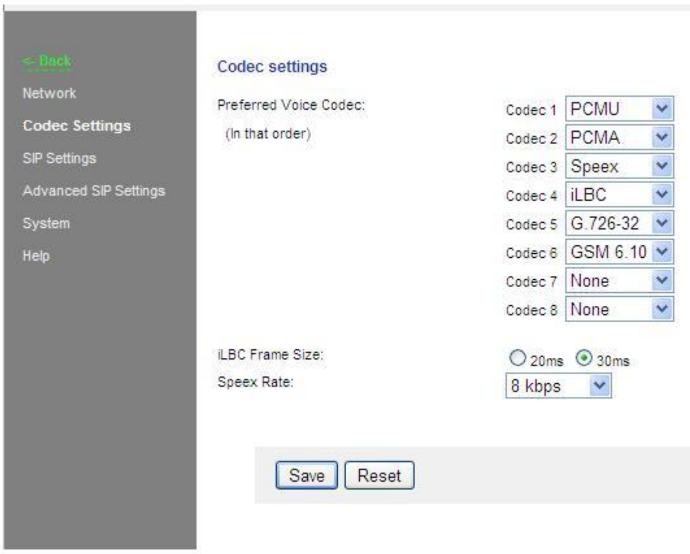
!	The button *reset* resets the content to the last stored condition. No factory setting is loaded.

general information	The MAC-address and information on hard- and software versions of the SIP-Gateway are stored here.
network settings	**type of connection**: The network settings can be entered manually, if *manually configured IP address* is selected, or received automatically via a DHCP (Dynamic Host Configuration Protocol) server. **IP address**: Here you can assign an idle IP address of your network to the SIP-Gateway. ! If the IP address has been modified manually, the web browser may not be able to display the page with the feedback. If this is the case, enter the new IP manually in the browser to get to the start page again. If necessary, also change the address area of your PC/laptop.
subnet mask	The subnet mask informs the SIP-Gateway about the size of the subnet, in which it is located. Enter the subnet mask of your network here.
standard Gateway	A gateway is a transition point between different networks (e.g. ethernet to internet via a router). Clients of a network send their packages to this IP address, if the destination address is outside of the network. Enter the IP address, to which requests from differing IP adddresses should be sent here.
retrieve DNS server IP automatically	The Domain Name System (DNS) is one of the most important services within the internet. It translates e.g. the names of a webseite into an IP address. If this option has been chosen, the fields for primary and secondary DNS will be highlighted with a grey background and the IP addresses will be received automatically from the DHCP-server.
use the following address	If this option is selected, addresses have to be assigned manually for primary and secondary DNS. Enter the IP address of the DNS-server resp. of your internet gateway (e.g. router).
layer 3 QoS / layer 2 Qos	Quality of Service (QoS) prioritises resp. parameterises the data traffic of the voice transmission in the network / internet. ! The settings for QoS should only be adjusted by experienced system administrators.

Voice

!	Modifications, if these are necessary, should only be carried out by experienced system administrators.

- Click on *Save* to save the settings.



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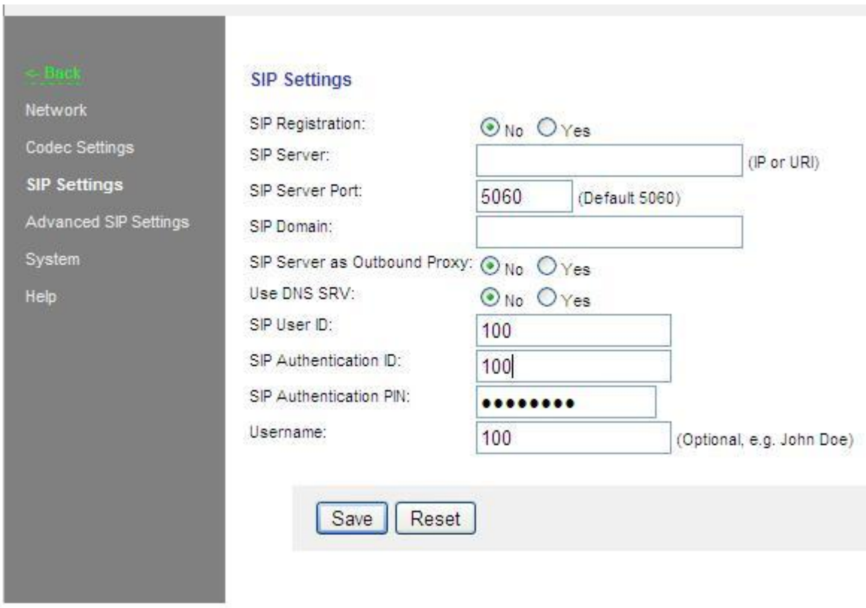
voice settings	To transmit the voice at the SIP-Gateway, the following voice codecs can be used. These are used in sequence when etsablishing a connection. That means that the first codec, which matches with the opposite partner, will be used.
preferred voice codec	• **PCMU** (G.711 µ-Law) is standard for digital communication in Europe. Excellent voice quality, but a very high data volume of ca. 80 to 100 kbit/s. • **PCMA** (G.711 A-law) is standard for digital communication in North America and Japan. Excellent voice quality, but also a very high data volume of ca. 80 to 100 kbit/s. • **Speex** is optimised for voice transmission but also very scalable. In this case, only the data volume is scalable. The standard setting of 8 kbit/s should be sufficient for a clear communication. The loss of data packages causes little resp. no problems. • **iLBC** (Internet Low Bitrate Codec) has been especially designed for voice transmission via IP networks. The iLBC causes a data volume of ca. 14 kbit/s (20 ms frame size) or 16 kbit/s (30 ms frame size) and is robust against the loss of data packages. • **G.726-32** Causes a data volume of ca. 32 kbit/s with a moderate voice quality. • **GSM 6.10** originates from mobile communications. The GSM 6.10 causes only a low data volume with a quality that is barely acceptable.
frame size / Speex Rate	This settings only refer to the codecs *iLBC* and *Speex*.

SIP-settings



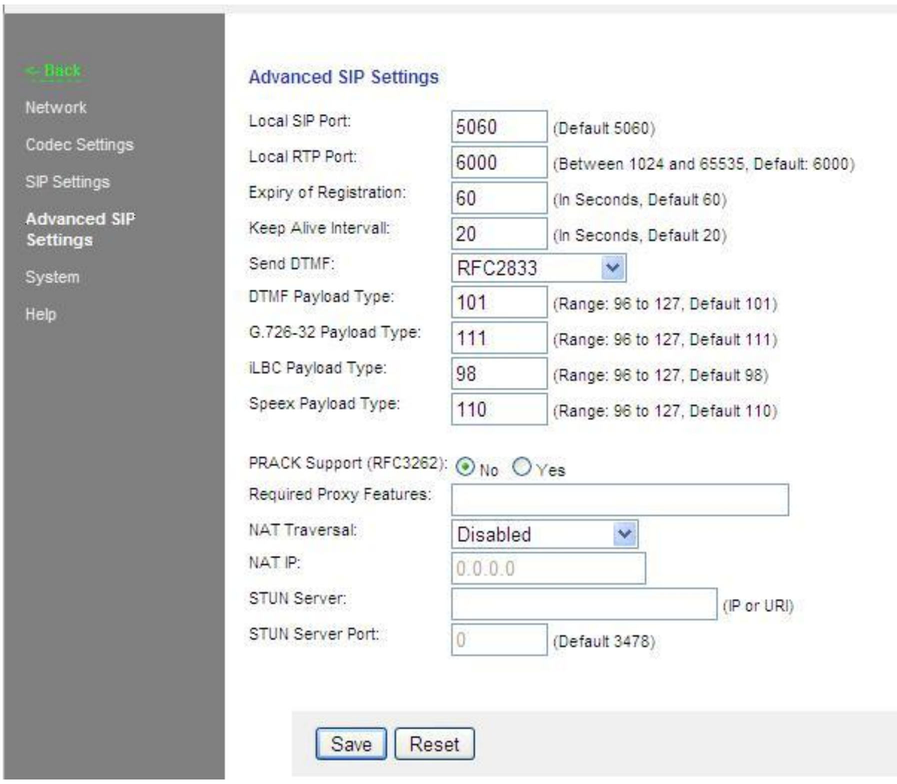
- Click on *Save* to save the settings.

!	The button *reset* resets the content to the last stored condition. No factory setting is loaded.

SIP-registration	indicates, whether the indoor station should register at the SIP-server. If this option is not activated, only direct connections (IP to IP) are possible. In this case, the IP address of the remote station has to be registered in the telephone book. A connection via call numbers is not possible.
SIP-Server and SIP-Server port	The IP address or URL of the SIP-provider / SIP-server has to be registered here. The port number indicated, to which port the server is reacting.
SIP-domain	The SIP-domain is used to disconnect the participants. It is used together with the number in the SIP-protocol to establish the connection (e.g. 1234@sipgate.de). Without a SIP-domain, a connection via call number is not possible. Instead of registering a name, also the IP address of the SIP-server can be entered here.
SIP-server as outbound proxy	Use the SIP-server as proxy for outgoing conversations. However, the SIP-server has to support this. Thus, phoning through a NAT firewall is possible.
Use DNS SRV	Use the DNS server entry in roder to reach a participant within the SIP-domain.
SIP user ID	This is the ID within a SIP domain, which is used to identify the intercom station. In case of an incoming call, the allocation is realised via the SIP user ID. That means that a caller is transmitting the call request as ”<call number>@sipdomain.de“ or as ”<user-ID>@sipdomain.de“.
SIP authenti-cation-ID and SIP-authentica-tion-PIN	User name and password for the registration at the SIP-server.
User name	This field is only informative and has no special function. You can register e.g. the location of the front-door station.

Extended SIP-settings

! Modifications, if they are necessary, should only be carried out by experienced system administrators.



- Click on *Save* to save the settings.

! The button *reset* resets the content to the last stored condition. No factory setting is loaded.

local SIP-port	Via the indicated port, the SIP-protocol, which is responsible for the administration of the SIP-connections, is handled.
local RTP-port	The data transfer of the audio data in real-time is realised via the indicated port.
expiry of the regist-ration	This setting indicates the intervals of the registration renewal at the SIP-server.
maintenance inter-vall	Indicates the intervals to send an empty RTP-data package to the SIP-server in order to keep the RTP-port open with the help of a NAT firewall resp. a router.
transmit DTMF	Here, the method is selected to realise the DTMF signalling: • Inband Audio – DTMF tones are transmitted as audio data • SIP-Info – DTMF digits are transmitted as SIP-protocol • RFC2833 – DTMF digits are transmitted via the RTP-protocol
type of user data (DTMF, G.726-32, ILBC, Speex)	The user data type should be left on standard settings. There is no special use for the user.
PRACK support (provisional acknowledge)	If this setting is activated, certain signallings within the SIP-protocol are stored.
required proxy settings	Characteristics which the Proxy server has to control.

NAT Traversal	If the SIP-Gateway is located behind a NAT firewall resp. a router, the way how the SIP-Gateway is identifiing it’s public IP address is determined here. This can be done with the option *use NAT IP* in case of a fixed IP address (e.g. a leased line) or in case of a dynamic assignment via a STUN server. In case of working exclusively within a local network, this option can be deactivated.
NAT-IP	Indicates the IP address of the SIP-Gateway from the perspective of the internet (WAN address). In case of dynamic assignment, this should happen automatically via a STUN server.
STUN server and STUN server port	IP or URL of the server to identify the current public IP address of the SIP-Gateway, and its port number.

System

- Click on *Save* to save the settings.

! The button *reset* resets the content to the last stored condition. No factory setting is loaded.

administrator password	Access password for the configuration of the SIP-Gateway via the web interface. The adjusted standard password ex works is ”1234“. You can also remove the password, if there is no need for security. **!** Please note that without a password, anybody can manipulate the system!
syslog server IP	The IP address can be used to forward system information to a syslog server. If the IP 0.0.0.0 is set here, the mode is deactivated.
SNTP server and time zone	The time zone for the system time and a server, which is needed to load the current time, can be indicated here. The time is related to the standard time GMT and only indicates the time without considering summer and winter time. In order to consider summer and winter time, the option *adjust time* has to be activated.

Configuration: hardware settings

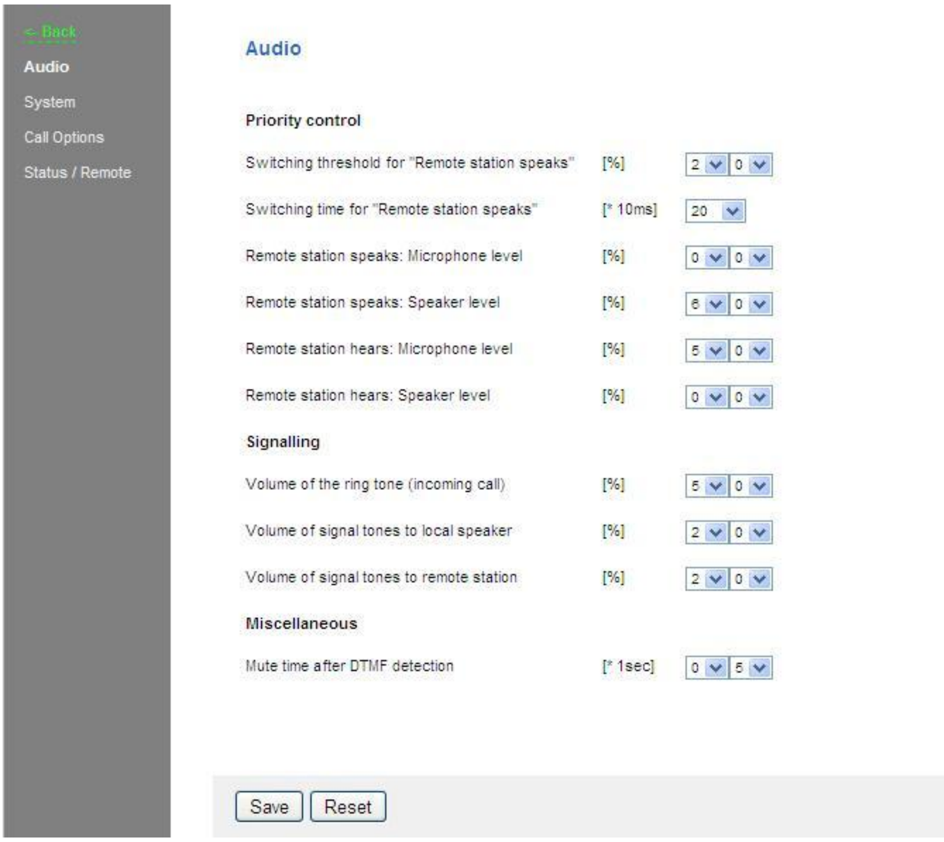
Audio	Adjustments for the prioritiy control when connected with another communication station for the loudspeaker and microphone level, as well as the change-over threshold and change-over period. Signalling settings for the volume of the ring tone and the signal tones. Modifications, if necessary, should only be carried out by experienced system administrators.
System	Settings for function and ON period of relay 1 and 2, the door release signalling, the period of the ID announcement, the format of the door release protocol and the function of internal calls to the SIP-Gateway.
Conversation	Settings for the length of call, for the call establishment and redialing, for the expectation of an acknowledgement via DTMF signs, for the number of repeating chain calls and to listen when the connection is established at the TCS:BUS.
Status /	Settings for the status and the ports for the remote operation and the status message. Path to the video source for Snom8xx VoIP-telephones.

Audio

! Modifications, if necessary, should only be carried out by experienced system administrators.

To suppress feedback noises and reduce echos, the SIP-Gateway automatically recognises whether the intercom station is communicating at the moment or not. The speech direction is unlocked and the opposite direction is quieted accordingly. Thus, it is switched between the pairs *remote station is speaking* and *remote station is listening*.
Settings regarding above mentioned can be adjusted here.

! The pre-adjusted parameters ex works are optimal for the different TCS:BUS front-door tations.
Make a note of the adjusted values before making modifications!



- Click on *Save* to save the settings.

> **!** The button *reset* resets the content to the last stored condition. No factory setting is loaded.

Priority control

change-over threshold for "remote station is speaking"	Sensitivity of the change-over to *remote station is speaking*. If the connected SIP-participant falls below this value, the audio signals are transmitted by the TCS:BUS. Note that the volume does not remian constant during the communication.
change-over period for "remote station is speaking"	Switchback delay after shortfall of the change-over threshold. If there is no communication from the TCS:BUS after the expiry of the set time, the audio signals are transmitted by the SIP-telephone.
remote station is speaking: microphone and loudspeaker lever	Microphone and loudspeaker level, if the change-over is activated, the audio signals are transmitted from the SIP-telephone to the TCS:BUS. Thus, the microphone level should be lower than in the state *remote station is hearing* and the loudspeaker level should be higher.
Remote station is hearing: microphone and loudspeaker level	Microphone and loudspeaker level, if the change-over is not activated, the audio signals are transmitted from the TCS:BUS to the SIP-telephone. Thus, the loudspeaker level should be lower than in the state *remote station is speaking* and the microphone level should be higher.

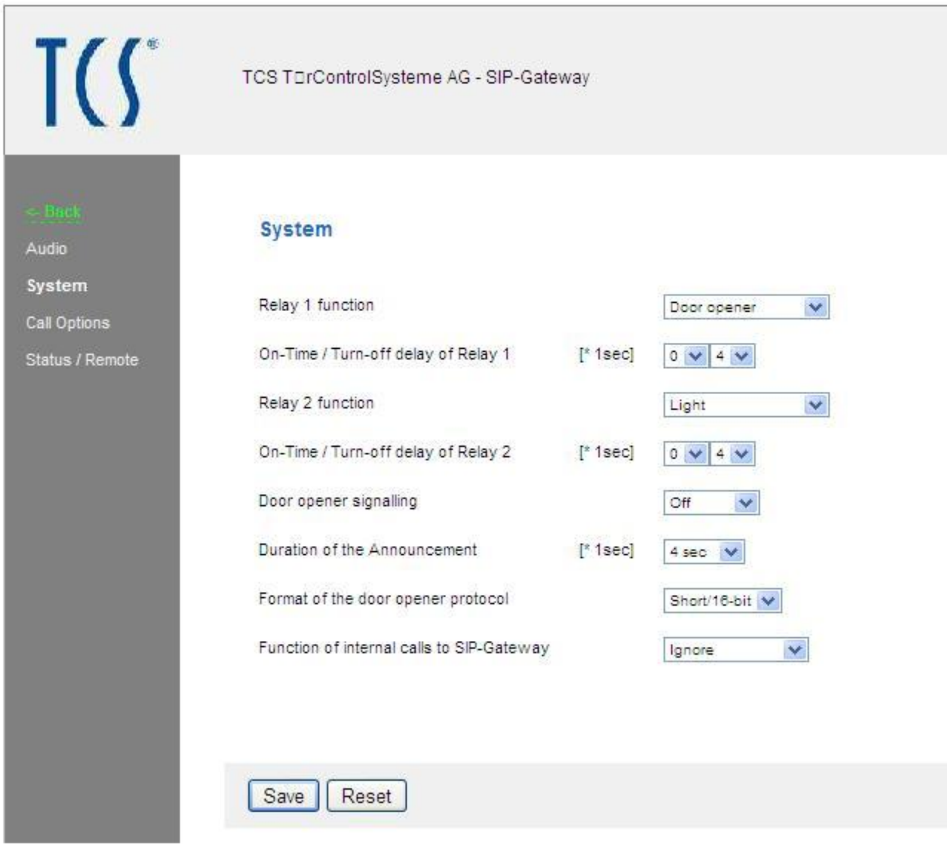
Signalling

volume of the ring tone when a call comes in	Volume for playing the call signalling. The volume of the dial tone and the signal of a busy line when establishing the call is half of the adjusted value above.
volume of the signal tones in the loudspeaker	Volume of the signal tones which the participant is hearing (locally) at the intercom station.
volume of the signal tones to the remote station	The setting determines the volume of the signalling tones which the participant is hearing at the remote station (DTMF acknowledgement, confirmation of the acknowledgement, door release and error tone).

Other

muting period after DTMF-detection	As soon as the 2nd DTMF digit has been entered, the loudspeaker at the front-door station is switched into mute. If the code is entered correctly, the muting is deactivated immediately after the positive acknowledgement. If a wrong code is entered or the entry is interrupted, the deactivation of the muting is realised after the muting period has expired.

System

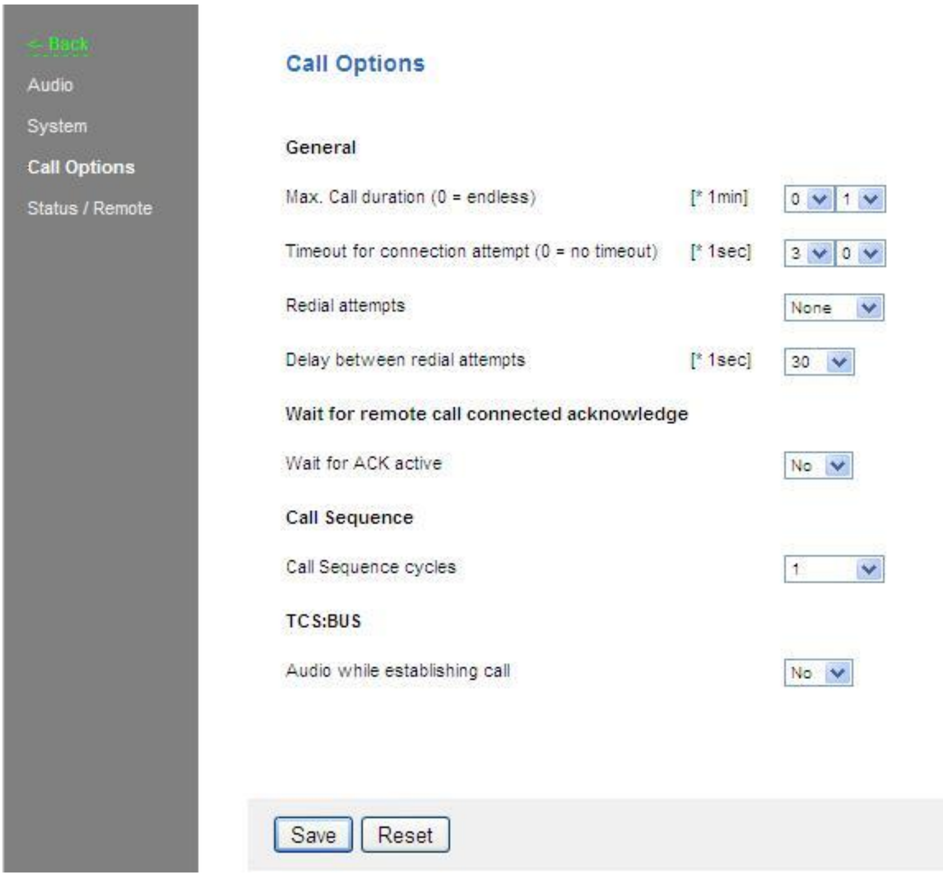


- Click on *Save* to save the settings.

!	The button *reset* resets the content to the last stored condition. No factory setting is loaded.

function of relay 1 function of relay 2	Hereby, a function is assigned to each relay within the SIP-Gateway. (pre-set relay 1: door opener, pre-set relay 2: light) • **door opener:** The relay is activated by pressing the door release function at the connected SIP-telephone or by a door release protocol at the TCS:BUS. 32-bit door release protocols are treated as door release or light protocol, according to the respective acknowledgement. • **camera:** The relay is activated when the call is established by the intercom station and during communication. • **light:** The relay is activated by pressing the light switch function at the connected SIP-telephone or by a light switch protocol at the TCS:BUS. 32-bit-door release protocols are treated as door release or light protocol, according to the respective acknowledgement. • **interference:** The relay is activated, when there is no interference. It drops out, if there is no network connection, no connection to the SIP-server (only, if the registration at the server has been activated) and in case of power failure. • **remote control:** An automatic function has not been assigned to the relay and it can be remotely operated via UDP without the influence of internal control processes. • **binary actuator:** The relay is activated by pressing the annunciator function at the connected SIP-telephone.
cyclic duration / switch-off delay	Depending on the assigned function, the setting either is a cyclic duration (door opener, light, annunciator) or a switch-off delay (camera). In case of a fault alarm function, the change-over is realised immediately, if a new condition is recognised. When under remote operation, the sent cyclic duration is relevant. *The switch time has to be set by selecting the digits from 0 to 99 seconds.The left selection field adjusts the two-digit seconds, the right one the one-digit seconds.*
door release signalling	After pressing the door opener, an acoustic signal is emitted for ca. 4 seconds at the front-door station. This can be useful e.g. if an opener for direct-current operation is used because no electro-acoustic signalling is realised.
duration of the ID announcement	The duration of the deposited announcement text has to be set here.
format of the door release protocol	• **short/16bit:** A 16bit-door release protocol is sent to the TCS:BUS, if the door release function has been activated via DTMF at the connected SIP-telephone. • **long/32bit**: A 32bit-door release protocol is sent to the TCS:BUS, if the door release function has been activated via DTMF at the connected SIP-telephone.
function of internal calls to the SIP-Gateway function of internal calls to SIP-Gateway	• **ignore**: Incoming calls at the TCS:BUS will not be processed by the FBI610x. • **as door call**: Incoming internal calls are treated just like incoming door calls. For this, the serial number must be deposited in the telephone book and a SIP-telephone number has to be assigned to this number • **as floor call**: Incoming internal calls are signalled separately. When accepting the call, the triggering of the door release function at the connected SIP-telephone triggers the transmission of a control function X with the corresponding serial number of the used storage space. The control function number complies with the internal address of the received internal call. Together with a BRE2, this function can be reasonably used. (floor push button at BRE2 sensor / door opener with external transformer at the BRE actuator)

Conversation



- Click on *Save* to save the settings.

!	The button *reset* resets the content to the last stored condition. No factory setting is loaded.

General

max. length of call	Call duration limit. After the determined period is expired, a forced interruption of the connection is realised by the SIP-Gateway, independent from the max. length of call of other devices within the system. Before the separation, an indication tone sounds at the connected telephone. For no limitation of the communication time, the time has to be set to 0.
max. period for call establishment	Determines the max. period to wait for a connection. This parameter can also be adjusted at a SIP-provider resp. a SIP-server. If the call is not accepted, the dialing attempt is cancelled.
break between redialing	Break between dialing attempts. In case of a cascade call there will only be a break if a redialing is realised. In case of a dial number change, the dialing attempt is realised immediately!
redialings	Number of attempts until the dialing is cancelled. If the intercom station is busy, a new dialing attempt is realised after the set break.

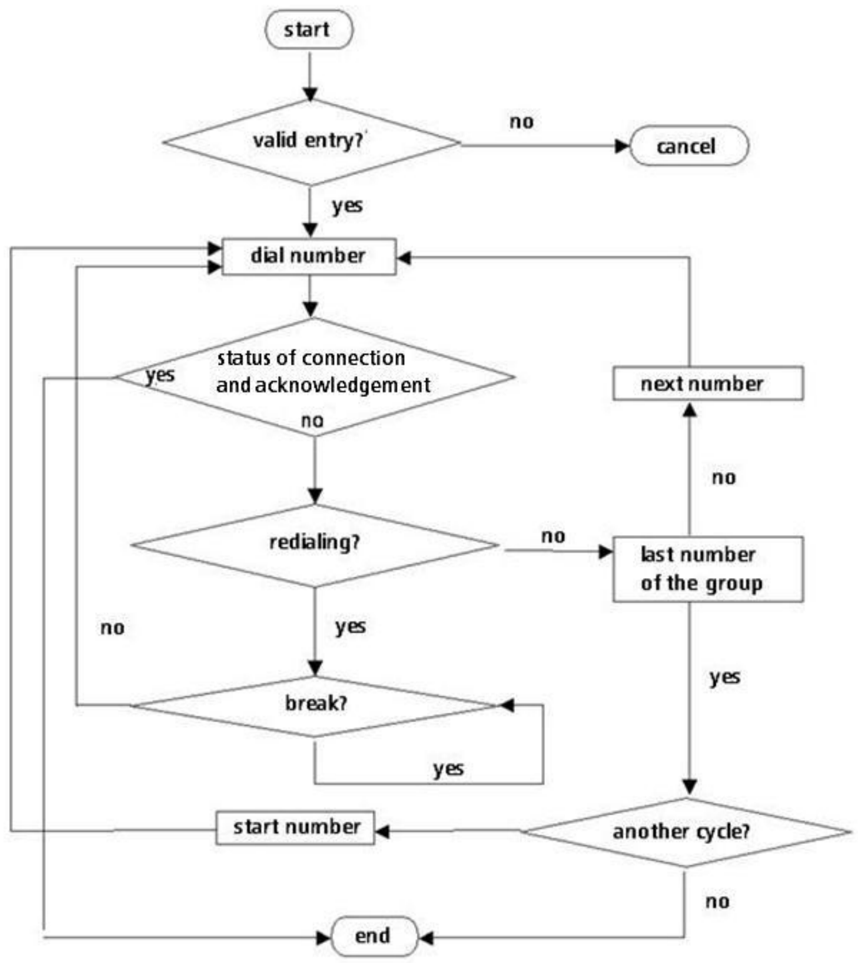
Acknowledgement expectation

expecting the acknowledgement	If an explicit acknowledgement of a call acceptance is required, it can be realised by this function. The participant who is called has to confirm the call acceptance with the button ’7’. Otherwise, the connection is interrupted after 10 seconds and a new call-attempt is started.

Cascade call

cycles for cascade calls	If the entries in the telephone book have been divided into groups (group-ID), the numbers within a group are dialed one after another until a connection is realised, an acknowledgement for the call acceptance is made or the cycle for cascade calls is run through accordingly. The order of a cascade call is determined by the fixed order of the entries in the telephone book. Under *cycles of cascade calls* you can adjust how often a cascade call should be run through.

function of the cascade call



TCS:BUS

listen when connection is establishing	If this function is activated, the Gateway automatically establishes a connection to the calling front-door / indoor station after having receicved a call protocol with a known serial number. The communication time set in the front-door / indoor station starts precisely at this moment. Via the loudspeaker of the front-door station, all acoustic signallings of the Gateway are reproduced during the establishment of the connection.

Status / remote operation

- Click on *Save* to save the settings.

! The button *reset* resets the content to the last stored condition. No factory setting is loaded.

authentication code	For remote operation, the code to be entered here has to match with the code, specified in the protocol.
IP address	Indicates the destination address to which the status messages should be sent. Usually this is the broadcast address. Thus, the messages are sent to all participants in the subnet. This is only useful if several participants of a network have to receive the messages. Otherwise only indicate a direct destination.
status port (send)	Identification number of the application. Thus, special basic conditions (firewall etc.) can be considered.
remote operation port (receive)	Identification number of the application. Thus, special basic conditions (firewall etc.) can be considered.
status	If this function is activated, status messages are sent. If this option is deactivated, no acknowledgement is sent, even when using the remote operation function! The status messages and the used protocol are further described in chapter 0.
remote operation	If this function is activated, the relays of the SIP-Gateways can be remotely operated via the network. Although remote operation works when remote operation has been activated, but without activated status messages, no acknowledgement is sent upon a command for remote operation.

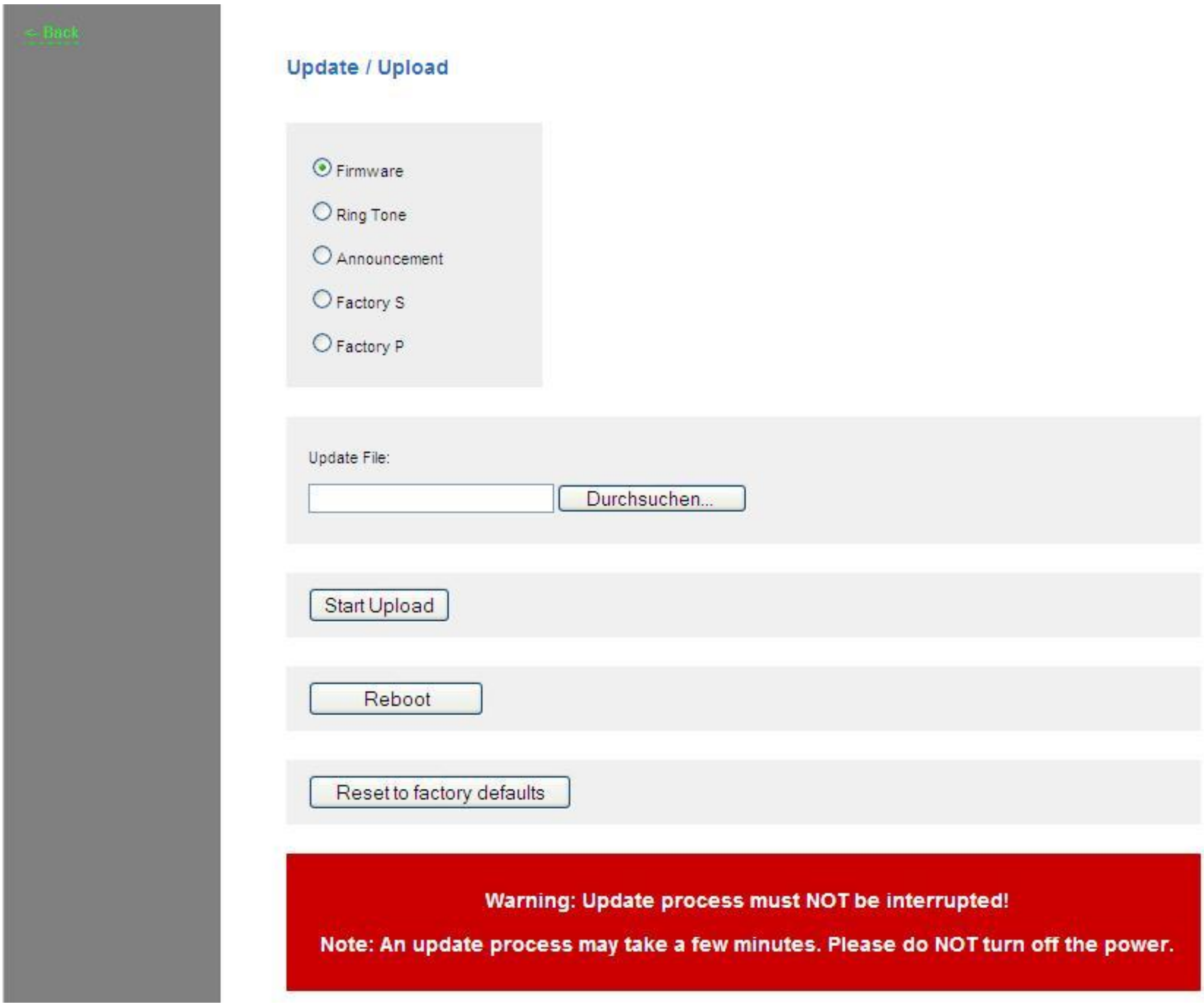
SIP-NOTIFY

URL for the snom-XML file	If together with a SIP-Gateway, e.g. a TCS:Server has been installed in the system, an image of the corresponding front-door station can be displayed when the call has been made to a Snom VoIP-telephone of the series 8xx. Therefore, a separated file stored on the TCS:Server (from SW-version 1.2.1.2 on), has to be executed. The path for URL to the Snom-XML file has to be registered here. Please find detailed instructions in the application example under www.tcsag.de

Configuration updates

! Modifications, if necessary, should only be carried out by experienced system administrators.

Updates can be loaded for the firmware, the ring tone and for the announcement (information, the person who called is hearing).



The following file types are predefined:

firmware:	xxxx.bin (firmware file for the SIP-Gateway)
ring tone:	xxxx.dat (sound file)
announcement:	xxxx.dat (sound file)
factory settings S:	xxxx.dat upload for setup data
factory settings P:	xxxx.dat upload from telephone book data

If necessary, the corresponding update files are made available by TCS.

Firmware

With an update of the firmware you can get the latest version of the software for the SIP-Gateway.

For an update, the latest version of the firmware ia available upon request.

- Just send us an Email.
- In order to send you the suitable firmware, we need the product name, the serial number and the information on hard and software versions (see *network > general information, page 17*)

Email : ***hotline@tcsag*.de**

After you have received the latest version of the firmware from us, you can load the update.
If it is also necessary to update the micro controller, our hotline will help with the required instructions.

Updating the firmware:

1. Start the configuration menu.
2. Chose the updates, enter password and log in.
3. Chose the firmware.
   Data to be transferred: indicate the path with the stored file xxx.bin, send by us.
4. Start transmission and wait until the automatic restart of the SIP-Gateway which is realised after the transmission is finished (< 2 min).

   Notes

!	The SIP-Gateway restarts automatically after a voltage interruption, even when it was OFF before the interruption.

!	Switching-off the power supply during a recording process or a software update can cause loss of data or damage of the device.

5. *You have sucessfully updated the firmware*.

Factory settings S / factory settings P

(restore backup)

In order to load saved device settings / saved telephone book data into the SIP-Gateway again, proceed as follows:

1. Start the configuration menu.
2. Chose *updates*, enter password and log in.

3. Chose *factory setting S (or P)*.
   Data to be transferred: indicate the path to upload the setup data (or to upload the telephone book data) settings.dat (or phonebook.dat).
4. *Start transmission* and wait until the automatic restart of the SIP-Gateway which is realised after the transmission is finished (< 2 min).

Create a backup files

To save device settings and phonebook data of the SIP-Gateway, you have to create and execute a special batch file.

Under Windows XP/Vista/7 proceed as follows:

1. Open the editor notepad.
2. Copy the following text into the empty window:

```
:start
color 7
@echo off
md c:\FBI610x
cls
attrib -r c:\FBI610x\settings.dat
attrib -r c:\FBI610x\phonebook.dat
echo.
set /p var= Please entert he IP address of the SIP-Gateway:
cls
echo Bitte warten...
cls
if errorlevel 1 goto Error
tftp -i %var% get settings.dat c:\FBI610x\settings.dat
if errorlevel 1 goto Error
tftp -i %var% get phonebook.dat c:\FBI610x\phonebook.dat
if errorlevel 1 goto Error
color A
echo.
echo ****************************************************************************
echo   The backup of the SIP-Gteway has been successfully stored under C:\FBI610x\
echo ****************************************************************************
Pause
Exit
:Error
cls
color C
echo.
echo ****************************************************************************
echo        The entered IP address is wrong or cannot be reached!
echo ****************************************************************************
Pause
goto start
```

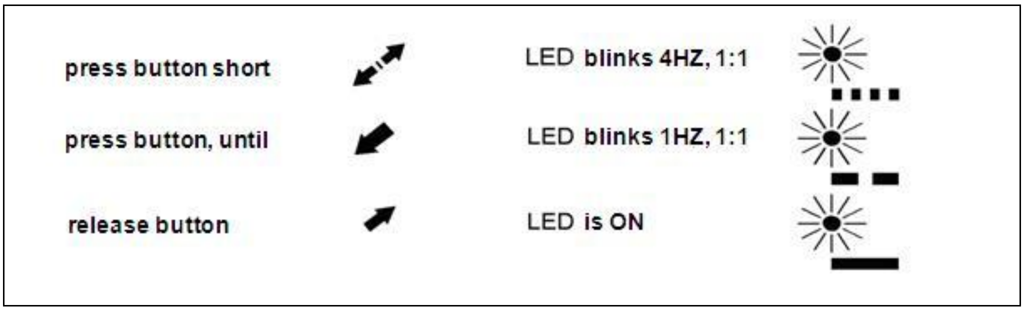
3. Save the file under “FBI610x.bat“ on your hard disk. Chose “all files“ under file type.
4. Execute the created file.

On your drive C: a folder “FBI610x“ has been created. This folder comprises 2 files: the device settings in ”settings.dat“ and the phonebook data in “phonebook.dat“.

- Save these files as backup.
- When the file is executed again, already existing files will be overwritten.

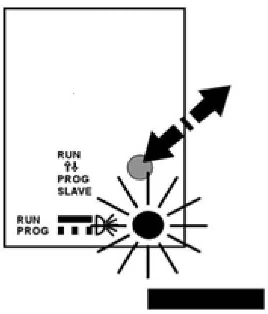
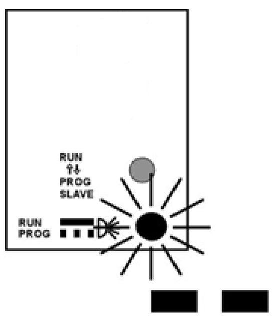
Manual adjustment via the RUN/PROG push button

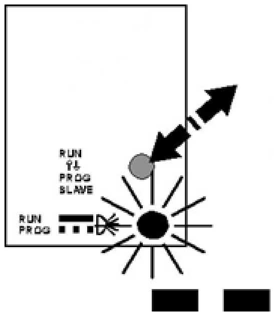
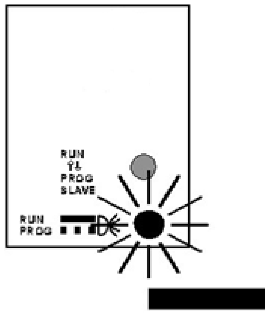
Legend for operating



Switch programming mode at TCS:BUS ON / OFF

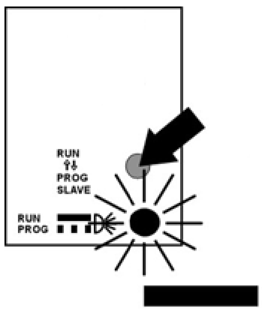
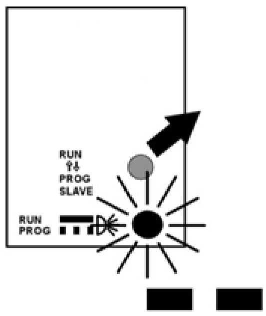
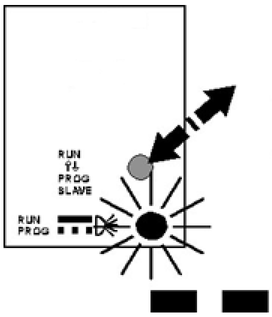
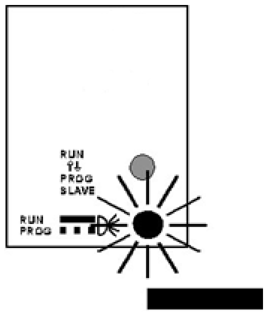
This function can be used, if the SIP-Gateway is operated in the ACTIVE mode. In the operating mode PASSIVE, this has to be executed at the power supply and control unit (see corresponding product information).

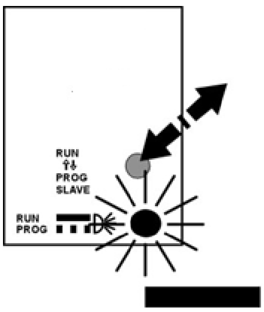
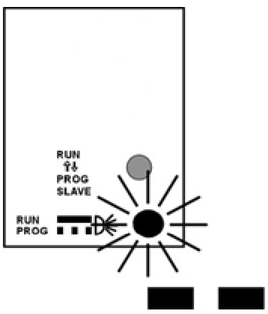
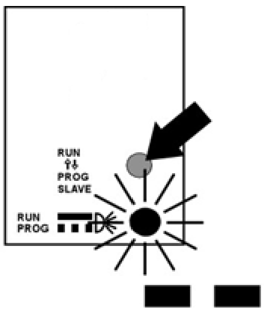
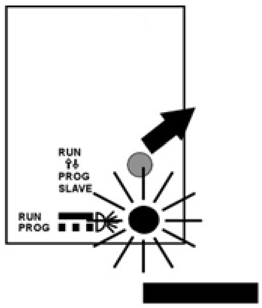
Switch ON the programming mode of the system	
 yellow LED is ON	 yellow LED blinks
press RUN/PROG button shortly	

Switch OFF programming mode of the system	
 yellow LED blinks	 yellow LED is ON
press RUN/PROG button shortly	

de- / activate light switch function in front-door stations

This function can be used, if the SIP-Gateway is operated in the ACTIVE mode. In the operating mode PASSIVE, it has to be executed at the power supply and control unit (see corresponding product information).

Activate light switch function in door stations			
 yellow LED is ON	 yellow LED blinks	 yellow LED blinks	 yellow LED is ON
press RUN/PROG button until…	…the LED blinks, then release	press RUN/PROG button shortly	

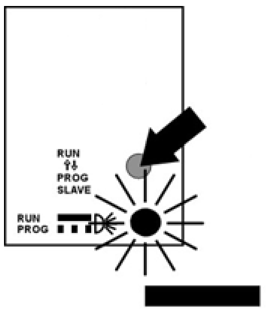
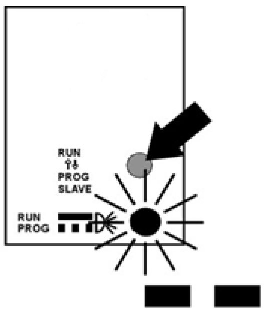
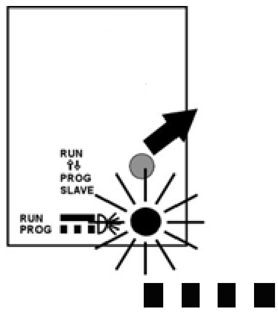
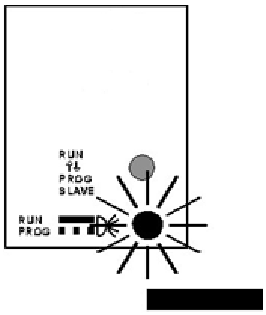
Deactivate light switch function in door stations			
 yellow LED is ON	 yellow LED blinks	 yellow LED blinks	 yellow LED blinks
press RUN/PROG button shortly		press RUN/PROG button until …	… the LED is ON, then release

Reset Reset factory settings

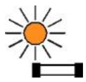
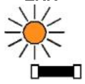
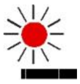
If the set IP address or the access password is unknown, the device can be restored to factory settings.

!	When loading the factory settings, all VoIP and hardware settings are reset. The phonebook entries remain unaffected.

The network factory settings are:
IP address: 192.168.1.200
subnet mask: 255.255.255.0
password: 1234

Load factory settings			
 yellow LED is ON	 yellow LED blinks	 yellow LED blinks fast	 yellow LED is ON
press and hold RUN/PROG button	press RUN/PROG bu-tuntil…	… the LED blinks fast, then release	factory settings are loaded

Error detection and display

error at the TCS:BUS **only operating mode PASSIVE:** a-wire is not connected / not supplied, short-circuit at a-b or a- and P-wire interchanged / short-circuited **only operating mode ACTIVE**: a-b or a-P short-circuited	ERR  **indication error flashes (1:7, 1 Hz)**	connect a-wire / check power supply, change a- and P-wire or eliminate short-circuit, device is in standby again
network error network connection incorrect or no connection to the SIP-server	ERR  **indication error flashes OFF (7:1, 1 Hz)**	check the network, check data transmission, check login data for SIP-server, device is in standby again
network error network connection incorrect or no connection to the SIP-server	 4 **indication error SIP ON ***	check the network, check data transmission, check SIP server, device is in standby again

***** *No error during the starting process, indication is ON and expices after ca. 40 s, if no error was detected.*

Status messages

Changes in status and the current status can be sent as status messages via an UDP data package from the SIP-Gateway. Therefore, status messages have to be activated in the configuration menu under *Status /* remote operation (see *Configuration: hardware settings* , page 23) and a receiver IP address has to be specified there.
This messages can be used e.g. to display the status of several SIP-Gateways in a central software supplied by the customer.

Protocol

An UDP data package is structured here as follows:

<sequence number>#<status>@<parameterbytes><checksum>

sequence number	Number of the current data set. Will always be enhanced plus 1 until 255 is reached and starts again at 0. Thus, a multiple receiving of a data set can be recognised. The sequence number consists of a 2 byte HEX-string (example: 01, FF, ...).
status	Indicates the current status resp. the type of the data message. The status consists of a 2 byte HEX-string.
parameter bytes	The parameter bytes are the supplement of the status bytes. They include detailed information (e.g. a call number, simplification of the status etc.) on the status byte. The parameter always consists of 24 digits (ASCII – no control characters!). Unused digits are filled with a blank.
checksum	The checksum is needed to control, if the data package contains the correct status data. A transmission protection is realised via the ethernet – transmission layer (CRC32). The checksum consists of a 2 byte HEX-string and is constructed via all data bytes as Addition Modulo 256.

Messages

status byte	**parameter bytes**	**description**
0x0A	<empty> or <XY>	standby (IDLE) XY = firmware version Atmega (2 digits HEX)
0x01	number of the caller	incoming call
0x05	number of the remote station	connection status
0x06	<empty>	dialling status
0x07	number oft he remote station	status of the call establishment
0x14	send identification	Max. 24 digits of the optional user name (SIP settings) are transmitted.
0x4C	byte0 = 0x31	login at the SIP-server executed successfully

Remote operation

Remote operation can be used to operate the relays (1 and 2).

Protocol

An UDP data package is structured as follows:

<identification><sender IP><seq. no.><outp. no.><period><password><checksum>

identification	Identification of the protocol: “BSREM“ (5 digits - ASCII).
sender IP	Includes the IP address of the sender as string made of Hex digits. 192.168.0.2 would be “C0A80002“.
seq. no. (sequence number)	To identify the package if several packages have been sent. Thus, also packages received twice can be recognised. The sequence number is represented hexa-decimal with two digits, scale 0 to 255 (00..FF).
outp. no. (output number)	Number of the output to be controlled. Permitted are the values 1 to 4. The number consists of only one Hex digit. 1 = door release relay 2 = light relay (note: special function within the web interface!) 3 = camera 1 (relay 1, if configured as camera) 4 = camera 2 (relay 2, if configured as camera)
period	Will be coded as two digit Hex string. 0 = off 1 = on (permanently) 2 ... 255 = switch-on period in seconds
password (authentication code)	Switching is only realised in accordance with the authentication code, which is registered in the configuration menu. Unused digits of the password have to be send as ’F’. The authentification code always consists of 4 digits. (DTMF characters ”0123456789*#“and ’F’)
checksum	The checksum is constructed same as the status messages.

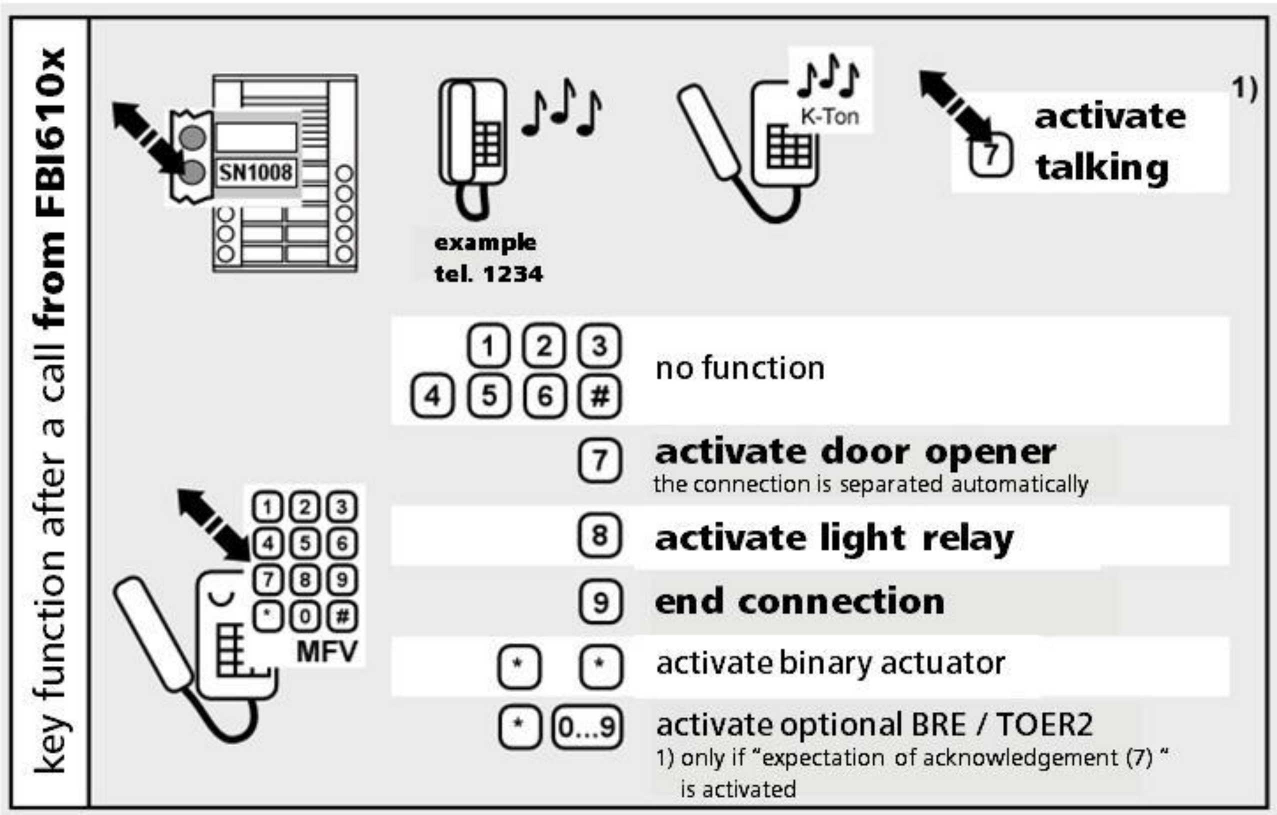
The SIP-Gateway sends an acknowledgement after the request has been successfully checked and diverted. The acknowledgement is only sent, if the status messages have been activated.
The data package has the status code 0x1E and the telecontroll data package, determinded above, is send back 1:1 as parameter (fills all 24 bytes of the parameter).

Operation

Door call from a front-door station

An incoming call is directly connected with the front-door station by accepting the call at the called telephone (delivery condition).

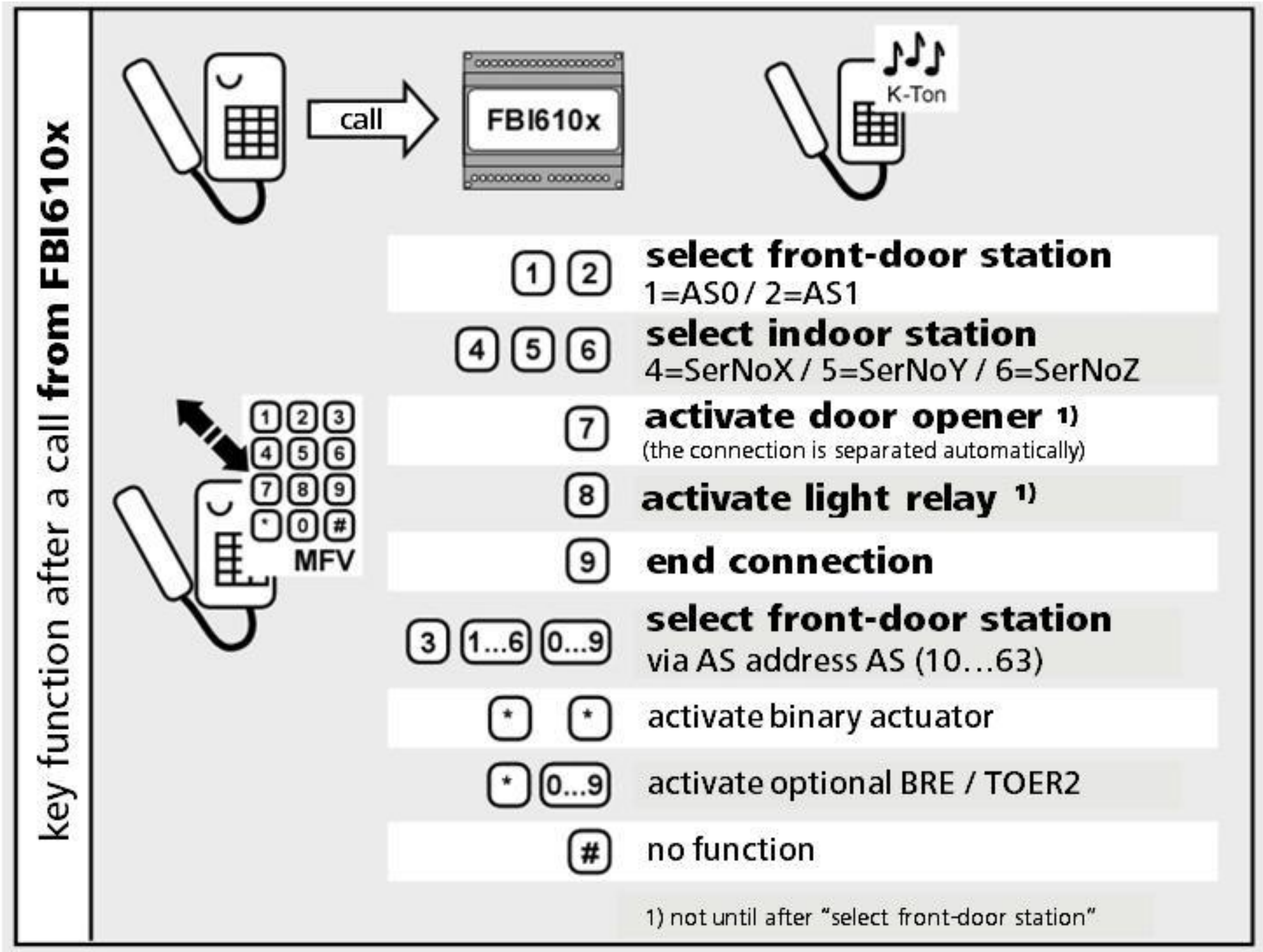
- If the option *expactation of acknowledgement ('7')* has been activated, press button 7 to accept the call.
- The control of the FBI610x is realised at the connected telephone via the MFV dial.



Call from a telephone to a front-door station

A telephone can only establish a connection to the FBI610x, if the incoming SIP telephone number is stored in the telephone book and the corresponding entry under *arriving* is <u>not</u> set on *reject*.

After the FBI610x has accepted the call, you have to chose the front-door station via telephone keypad. See illustration *chose front-door station*.



!	For using the binary output a configuration is necessary, see *settings hardware, system*.

Setup assistant for audio settings

Some audio settings can be changed during an ongoing conversation.
This is realised via special DTMF codes. However, these settings only serve to make the setup easier.
The parameters are temporary and are resetted to their original value after the conversation has been terminated. In case of detecting reasonable values, these can later be set permanently via the web configuration (see *Audio* page 23).

DTMF code

All DTMF commands for these functions have five digits. They are entered via the buttons of a telephone, which is connected to the SIP-Gateway. They start with zero (0), followed by a command digit, two digits for the parameter and end with a #-key.

DTMF code examples

command structure	**<zero>**	**<command>**	**<parameter>**	**<rhombus>**
example 1	0	2	65	#
example 2	0	7	33	#

> **!** To ensure that the setup assistant for changing the audio settings is working, you have to activate it with 0999#. In order to test the settings without delay, the DTMF muting is also deactivated .
> Further, there has to be an interval of one second between two identical digits to make sure that the second digit is recognized correctly! When entering a code consisting of different digits this will not be necessary.

Possible DTMF commands

command	**parameter area**	**description of the function**
9	[00...99]	special functions
	99	activate service mode
	00	deactivate service mode
1	[00...99]	remote station is listening: loudspeaker level
2	[00...99]	remote station is speaking: loudspeaker level
3	[00...99]	volume of the signallings in the local loudspeaker
4	[00...99]	volume of the signallings to the remote station
5	[00…99]	remote station is listening: microphone level
6	[00...99]	remote station is speaking: microphone level
7	[00...99]	switching threshold for ”remote station is speaking“
8	[00...20]	switching period for ”remote station is speaking“ [value * 100 ms]

FAQ

With what kind of system solutions has the SIP-Gateway been checked so far and is software-technically compatible?

1. STARFACE
   server is not answering to ”SIP-NOTIFY“ (Snom video display)
2. Sipgate.de
3. Sipcall-voip.de
4. 3CX
   (Free Edition: server is not answering to “SIP-NOTIFY“ (Snom video display)
5. Asterisk
6. accessVoIP
7. Asterisk-Cluster
8. FOXFON
9. EasyPBX
10. Swyx
11. Siemens HiPath
12. Octopus NetPhone
13. Cytel
    There are problems when transmitting the DTMF data. After starting the server, no or only the first DTMF comes through.
    The problem should be solved from version 4.0 Build 175a, at least partially, according to Cytel (and an internal test with an inofficial patch). The transmisson was working reliable via RFC2833. There are still problems when using SIP info.

14. Agfeo
    a) connection termination after ca. 10 seconds
       There can be 2 reasons:

       – *expectation of an acknowledgement* is activated and no acknowledgement DTMF is sent
       – server does not reply to SIP-NOTIFY (Snom video display)

       solutions:
       – deactivate expactation of acknowledgement or send acknowledgement signal before expiry of 10 seconds
       – remove NOTIFY-URL (thus, no NOTIFY wil be sent any longer) or use firmware as of version 6.1 (as of test version 6a0, the SIP-Stack is adapted).

    b) DTMF transmission problem

       At the moment (12/2009) these systems are not able to convert DTMF signals from an analogue participant into SIP-INFO or RFC2833.

       solution:
       Set the parameter *send DTMF* to *in audio data stream* under *settings VoIP > extended SIP*.

Note: This data is given without guarantee and may change at any time.

error	reason	solution
The LED *ERR* blinks.	Error at the TCS:BUS or within the network.	See section ”error detection and – display“
I cannot contact the SIP-Gateway with my browser, although a network connection to the SIP-Gateway is established.	The PC is not located in the same network segment as the server.	Check the network settings. Ask your administrator.
The IP address of the SIP-Gateway is unknown.	The IP address has been set to an unknown address (e.g. typing error while configuration).	Reset the IP address, by loading the factory setting.
The SIP-Gateway cannot be contacted under the known IP address.	There is an IP address conflict with another network participant (double assigned IP address).	Change the IP address of the SIP-Gateway by connecting yourself directly with the SIP-Gateway via a cable or separate the other network participant temporary from the network.
There are no settings shown on the web surface of the SIP-Gateway, menu items cannot be selected.	The used browser does not support JavaScript.	Activate JavaScript in the browser settings.
The programming mode of the TCS:BUS cannot be switched ON / OFF via the RUN/PROG push-button at the SIP-Gateway.	The SIP-Gateway is in operation mode PASSIVE. These settings are not possible in this operating mode.	Use the RUN/PROG push-button at your power supply and control unit.
The light switch function in the front-door station cannot be de- / activated via the RUN/PROG pushbutton at the SIP-Gateway.		Check, if the SIP-Gateway is operated in the right mode according to your installation.
Via the SIP-Gateway, no call can be established to a to a telephone, or call establishment is cancelled.	The SIP domain is wrong.	Please check the settings in the field “SIP Domain“ under “SIP settings”.
The SIP-Gateway is not able to register at the SIP server.	The field “SIP Domain“ under ”SIP settings“ is empty.	Enter the SIP domain into this field. Use the IP address when using a local SIP server resp. the absolute address / domain when using an external SIP server / SIP provider (e.g. www.example.com)
	The field “SIP Domain“ under “SIP settings“ is filled with an absolute address / domain (e.g. www.example.com), but there is no DNS server.	Check the settings to the DNS server under “network“. Enter the IP of your DNS server (e.g. the IP of the used router).
The entry of DTMF commands is without reaction at the SIP-Gateway / TCS:BUS	The used SIP server / SIP provider does not transmit the DTMF commands correctly or by another method.	Check under *settings VoIP extended SIP settings,* under the parameter *send DTMF* all possible transmission methods.
	The DTMF transmission at the used telephone is deactivated or in another method.	Check the settings of the telephone with the corresponding documentation.The parameter *send DTMF* under *settings VoIP > extended SIP settings* has to match in all devices.

A call from a telephone to a SIP-Gateway is cancelled / rejected.	The telephone number used to call the the SIP-Gateway, is not registered in the telephone book, the SIP-Gateway does not know the number.	Create an entry with the corresponding number in the telephone book of the SIP-Gateway.
	The SIP-Gateway has been configured to reject incoming calls with this number.	Don't set the settings at the corres-ponding telephone book entry under incoming calls to "reject".
	No name has been entered in the telephone book. This field has to be filled in.	Enter a name at the corresponding telephone book entry.
The connection from the tele-phone to the SIP-Gateway was established. While trying to es-tablish communication to an indoor or front-door station, the speech connection is interrupted immediately.	The called device at the TCS:BUS does not acknowl-edge the call, cannot be con-tacted.	Check the functionality of the called device. Check the set AS address / serial number.
When a door call is triggered, a negative acknowledgement tone sounds at the front-door station.	The SIP-Gateway already is in communication, maybe to another front-door station.	No error: The SIP-Gateway can esa-tablis max. one speech connection.

Service

For questions, please contact our technical support:
hotline@tcsag.de

Headquarter
TCS TürControlSysteme AG, Geschwister-Scholl-Str. 7, 39307 Genthin, Germany
Fon.: +49 (0) 3933/879910, FAX: +49 (0) 3933/879911, Mail: info@tcs-germany.de,
www.tcsag.de