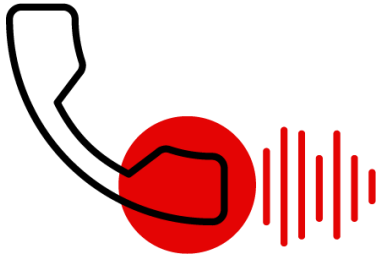

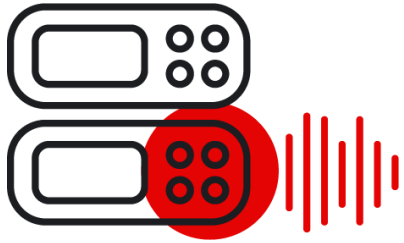

MAGIC TH1 potsGo / MAGIC TH1 ipGo

Quick Guide

Version V1.701 (06 March 2023)

info@avt-nbg.de
www.avt-nbg.de

Content

- First Steps
- Operation
- Settings
- Administration
- Telephone Control Codes

MAGIC TH1Go

First Steps

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- Connect your MAGIC TH1 system via the LAN interface to your IP network
- The easiest way to obtain a valid IP address is to activate DHCP in the system via the front panel
  - Press the softkey MENU and select LAN INTERFACE > DHCP and set the checkbox ENABLE
  - Press the DROP key and save the configuration
  - The assigned IP address is displayed when pressing the DROP key again
- Alternatively, you can set a static IP address, subnet mask, etc. manually under the LAN INTERFACE menu item
- Open now your web browser and enter the IP address of the MAGIC TH1 system
  - You can also use the system name TH1go instead of the IP address if DNS is available
- If everything is correct the main window appears

MAGIC TH1Go

Operation

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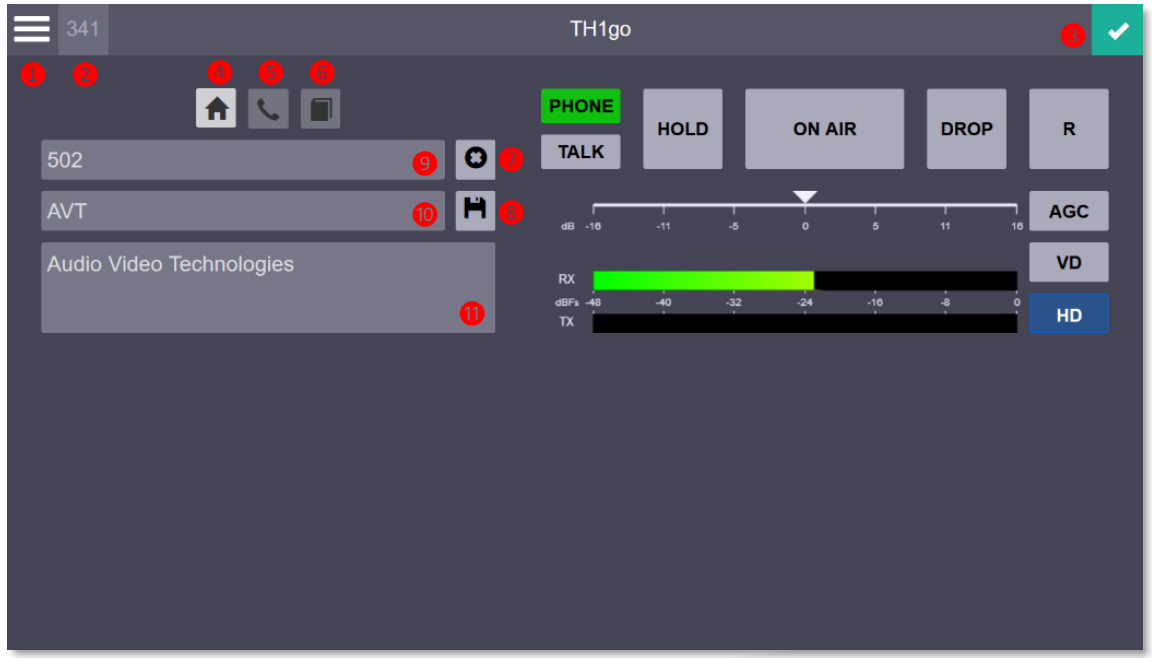
Opens
Configuration Menu

Connection
status of
MAGIC TH1

Main Window

- ❶ Menu
- ❷ Own SIP number
- ❸ Connection & system status
- ❹ Caller information
- ❺ Dial keypad
- ❻ Telephone book
- ❼ Delete caller information
- ❽ Save caller information
- ❾ Phone number of caller
- ❿ Name of caller
- ⓫ Information of caller



Operating Elements (1)

- ❶ System Name
- ❷ PRETALK – Telephone in use
- ❸ PRETALK – Audio XLR in use
- ❹ HOLD
- ❺ ON AIR
- ❻ Disconnect line
- ❼ Transfer call
- ❽ AGC Automatic Gain Control on/off
- ❾ Level adjustment
- ❿ Level meter
- ⓫ VD Voice Disguise on/off
- ⓬ Audio in HD Voice (7-kHz) received

Operating Elements (2)

MAGIC TH1Go

Settings

- General
- Line Interface
- Audio
- Operation
- Signal Processing
- LAN Interface
- VLAN
- GPIO (TTL/Relay)

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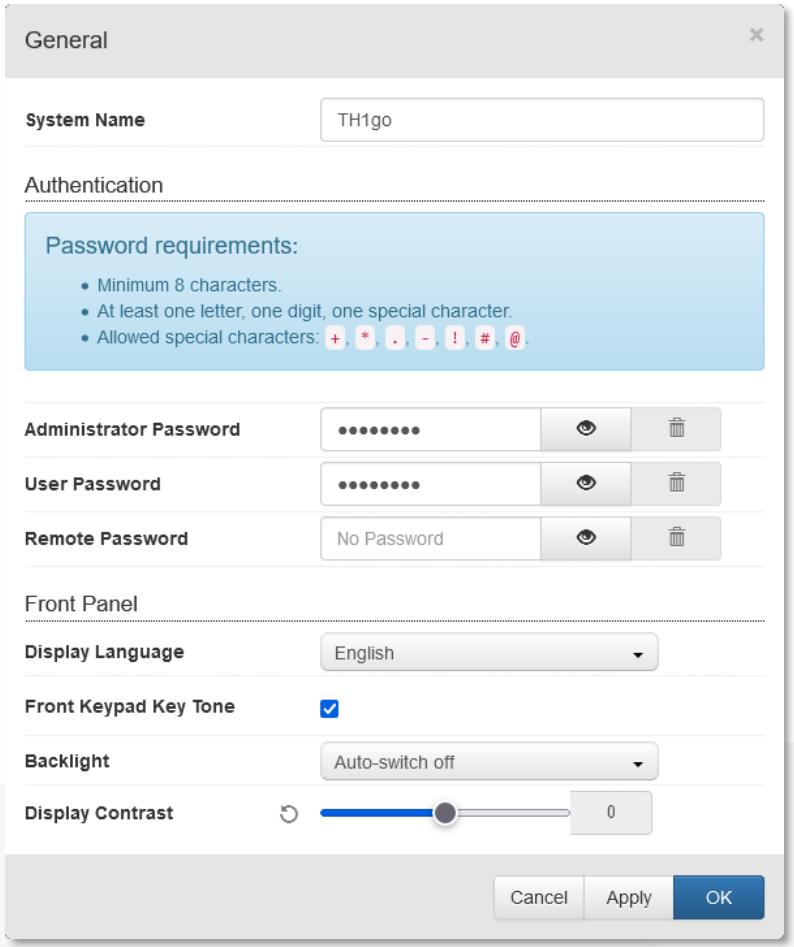
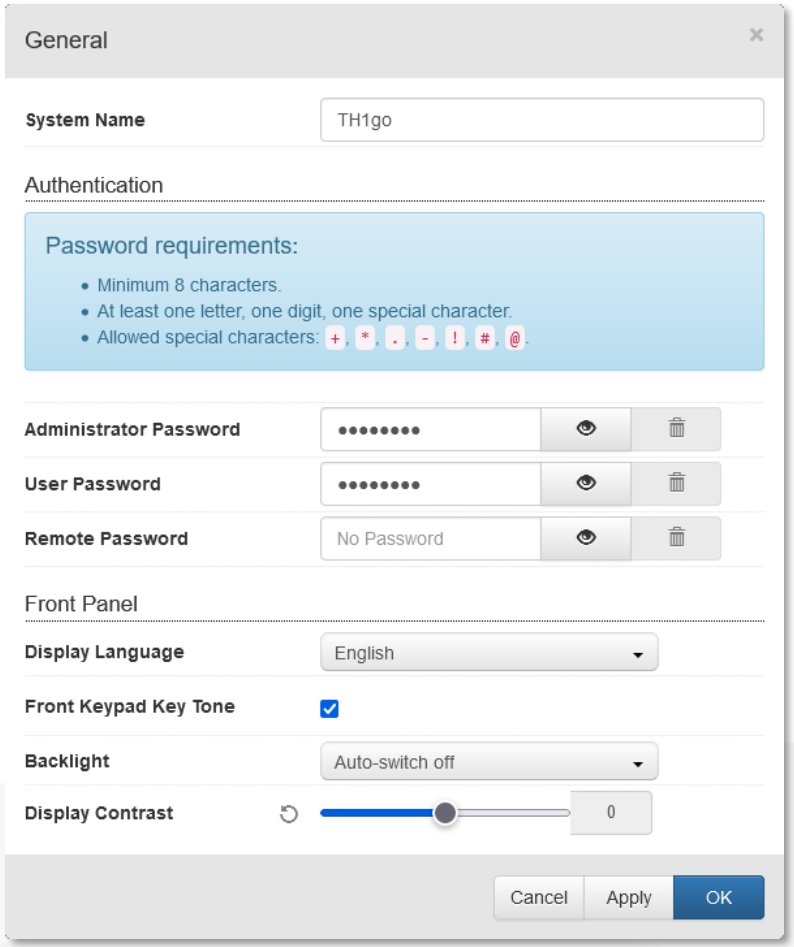
- System Name
  - The System Name is displayed in the main window which is helpful if you have more than one TH1 system in use
  - The system can be also used instead of the IP address in a web browser (default: TH1go)
- Authentication
  - Administrators may restrict access to the systems via passwords. Setting or changing a password requires you to confirm the changes at the front display of the unit within 5 minutes.
  - Administrator Password: Protects the configuration.
  - User Password: Lets the user load presets via the optional PC software.
  - Remote Password: Required to control or configure the unit from an external network. Controlling the unit from the network where the unit is located or from trusted networks is not protected by passwords. (Trusted networks are configured under LAN Interface.)



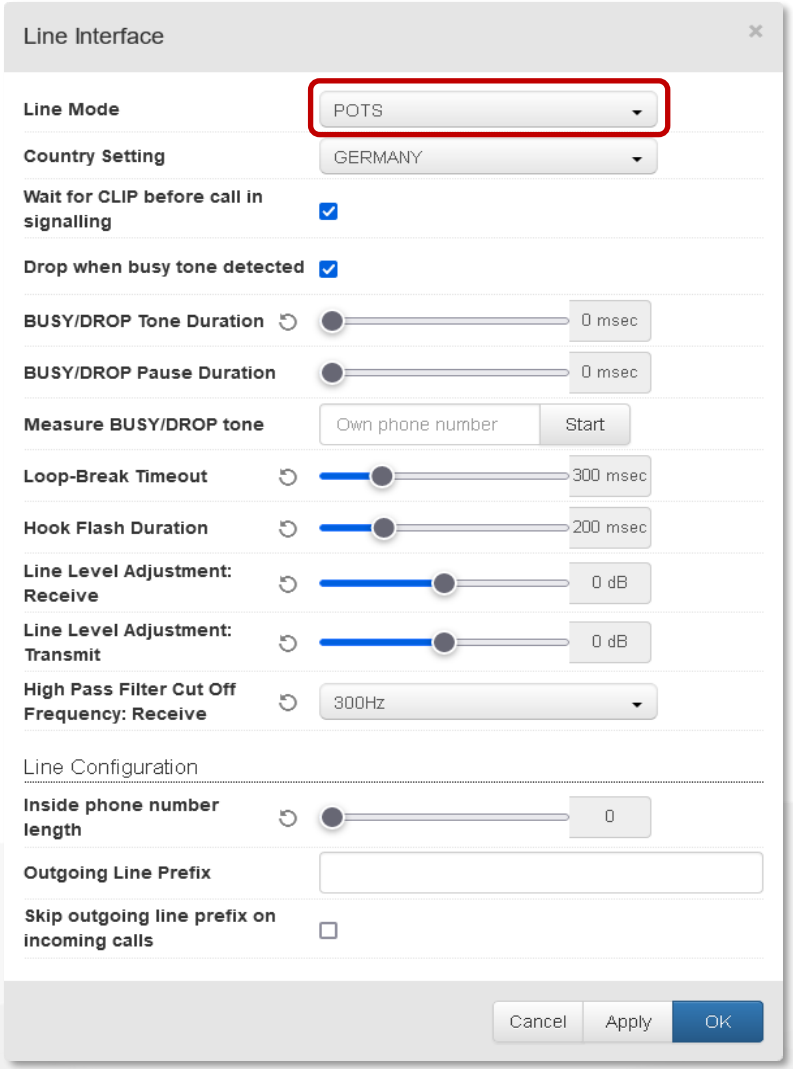
General Settings (1)

- Display Language
  - Select English or German
- Front Keypad Key Tone
  - If Front Keypad Key Tone is selected a tone is generated after pressing a button on the key panel
- Backlight
  - One of the following modes can be selected:
    - Permanent On
    - Auto-switch off: on only during key entries
- Display Contrast
  - The display contrast can be adjusted in a range between -4 … +4



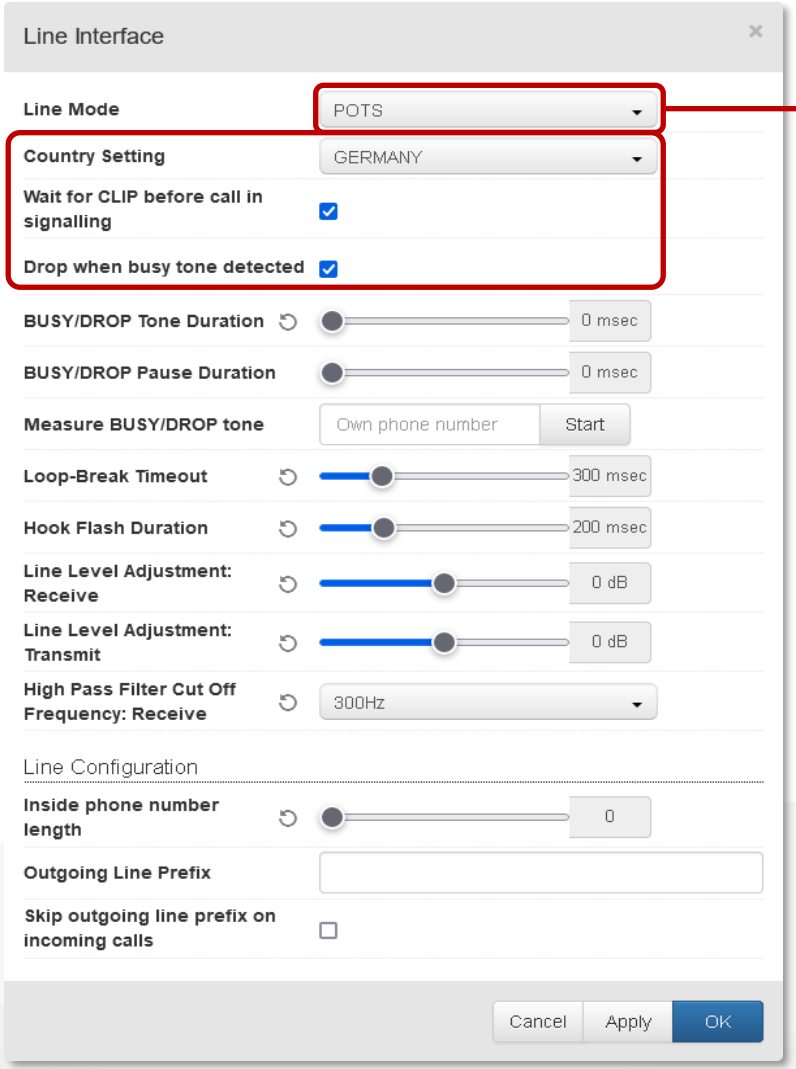
General Settings (2)

- Under ☰ → Line Interface the line interface settings can be configured
- Select under Line Mode which interface you want to use
  (the availability depends on your system and enabled options)
  - POTS
  - VoIP



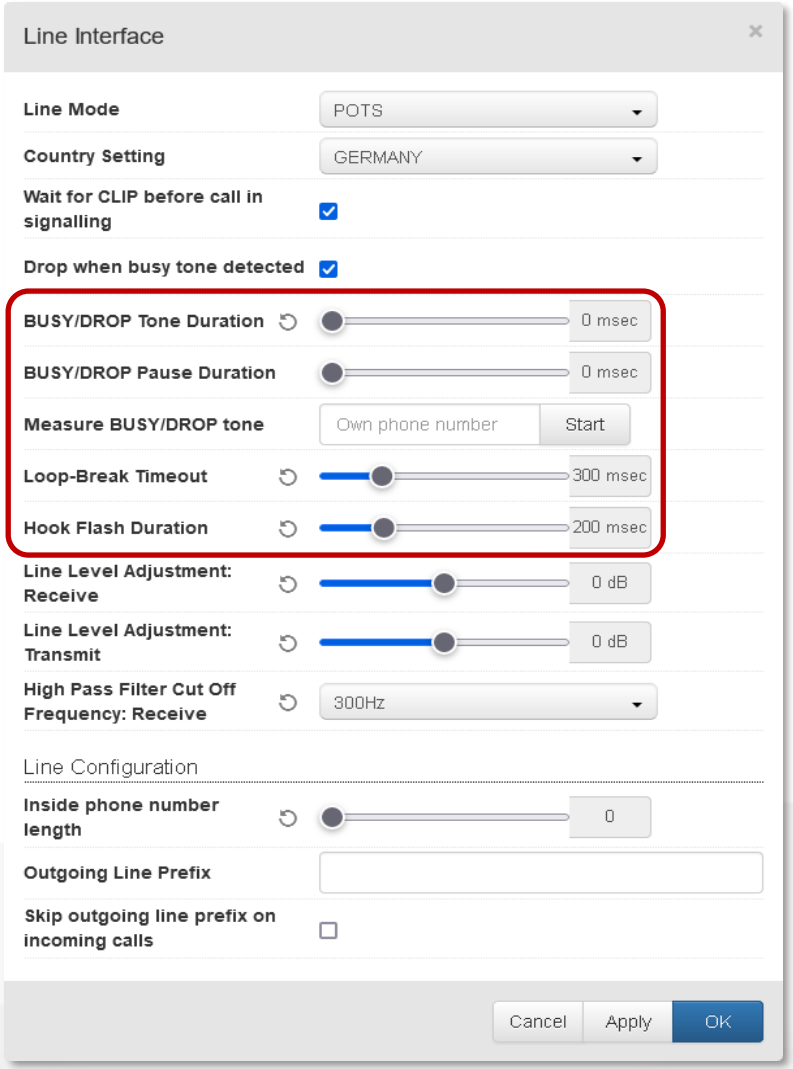
Line Interface Settings

- Your Country
  - If you use landlines: Country of your location
  - If you use a PBX: Try first your country or alternatively the country of origin of the PBX
    - Note that this setting is essential for correct operation of the built-in digital echo canceller. When you hear echoes, the cause is usually a maladaptation of the impedance, which is determined by the country setting.
- Wait for CLIP before call in signalling
  - If selected, an incoming call is not displayed until the caller's phone number has been recognized (CLIP function). If no phone number is signaled, the call is displayed after a short time.
- Drop when busy tone detected
  - Automatic drop after detection of the busy tone
    - This function requires to Start the function **Measure BUSY/DROP tone**



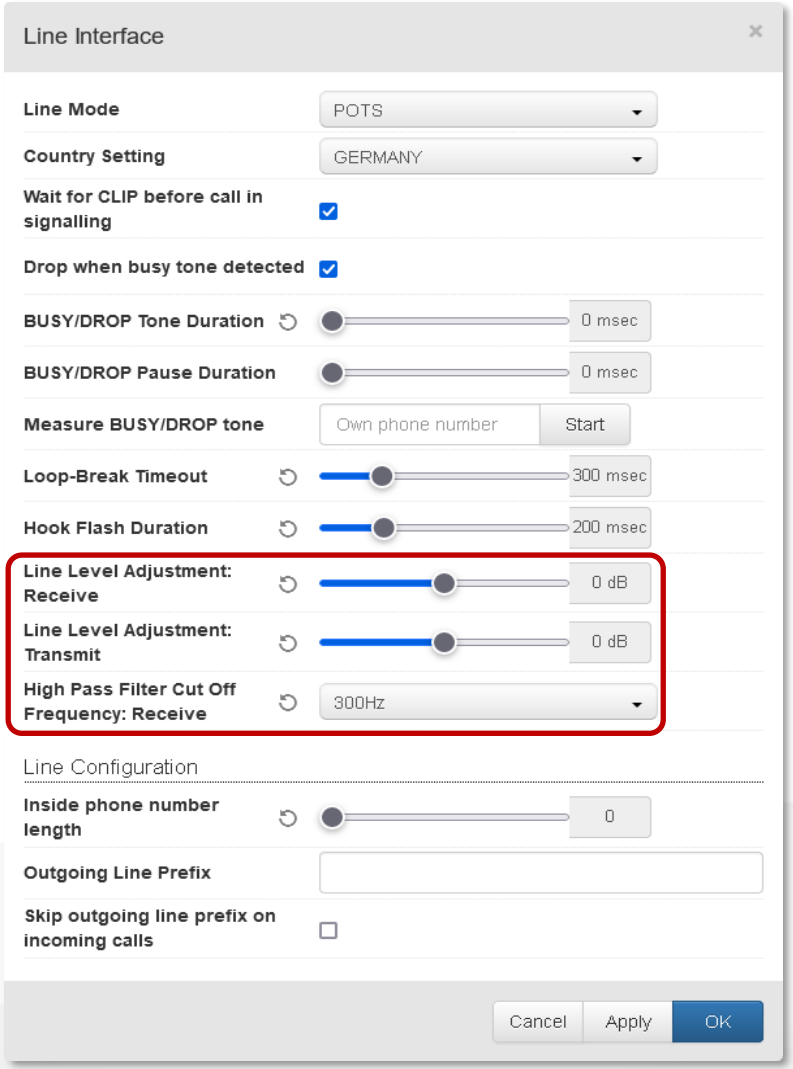
POTS Mode (1)

- The values for BUSY/DROP Tone Duration and BUSY/DROP Pause Duration can be set manually, but we strongly recommend using the built-in measurement function.
- Measure BUSY/DROP tone
  - Enter your own phone number and press Start
  - The values for BUSY/DROP Tone Duration and BUSY/DROP Pause Duration are automatically adjusted
  - With that settings the system recognises when a caller drops the line or if the line is busy
- Loop-Break Timeout
  - The loop current can be interrupted for that time period without interrupting the connection
- Hook Flash Duration
  - The length of the pulse which is necessary for call forwarding. It is also an interruption of the loop current



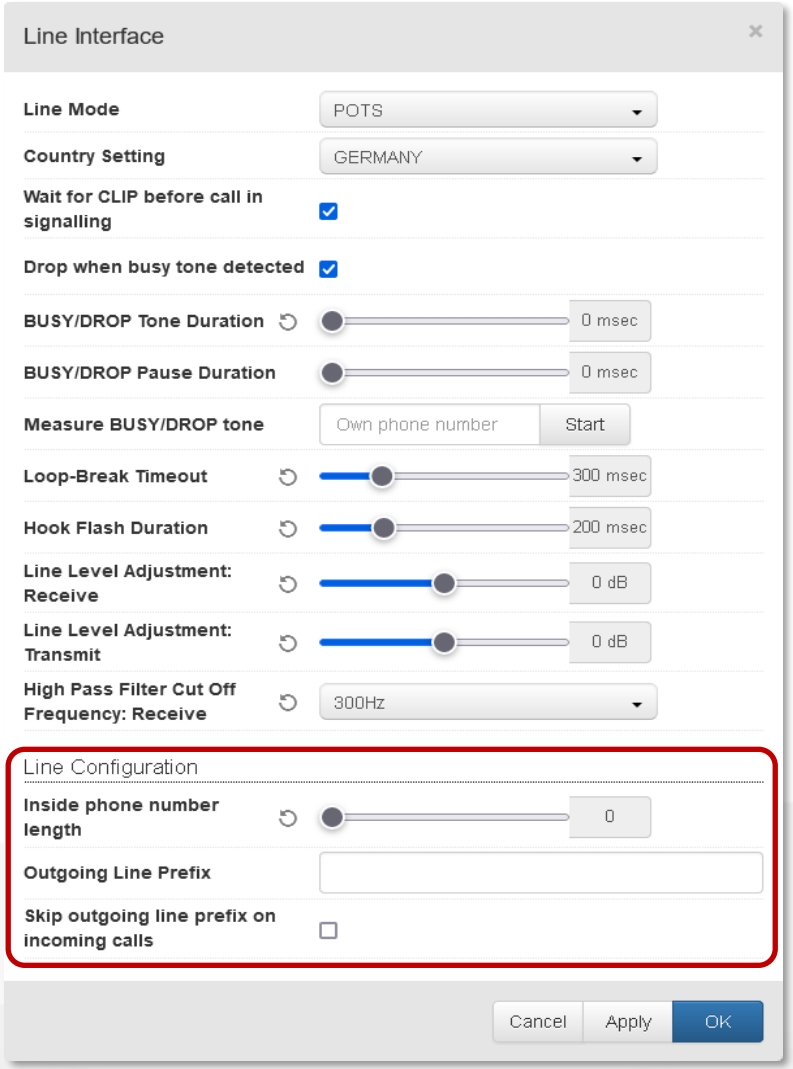
POTS Mode (2)

- Line Level Adjustment: Receive
  - Receive Line Level can be adjusted from -9 to +9 dB
    - Change this setting only if the caller level is always too low
- Line Level Adjustment: Transmit
  - Transmit Line level can be adjusted from -9 to +9 dB
    - Change this setting only if the callers can always hear you poorly.
- High Pass Filter Cut OFF Frequency: Receive
  - High pass filter for the reduction of disturbances
  - If you hear a humming noise on the line, set a higher cut-off frequency (e.g., 100 Hz)



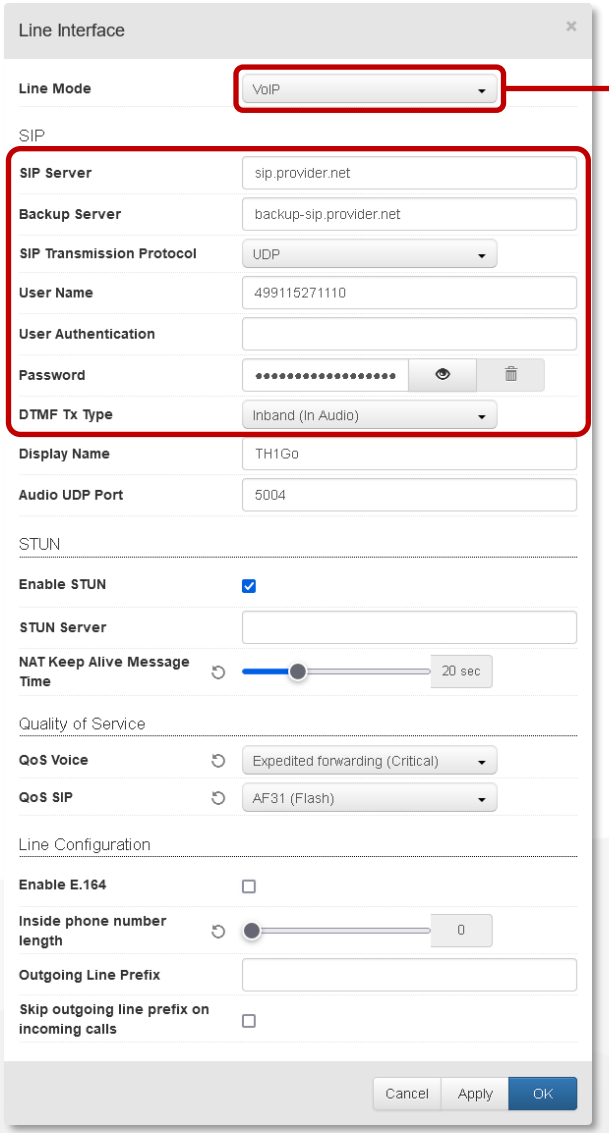
POTS Mode (3)

- Inside Phone number length
  - The outgoing line prefix is dialled automatically if you dial a number that is longer than the number length specified here
  - If you work with landlines, please select 0
- Outgoing Line Prefix
  - If the system is connected to a PBX, you can enter the Outgoing Line Prefix, e.g., 0, which is automatically added by the unit when you dial an external number
- Skip outgoing line prefix on incoming calls
  - If the phone number is displayed in the software with outgoing line prefix, the outgoing line prefix is deleted, and the displayed telephone number can be saved directly to the phone book



POTS Mode (4)

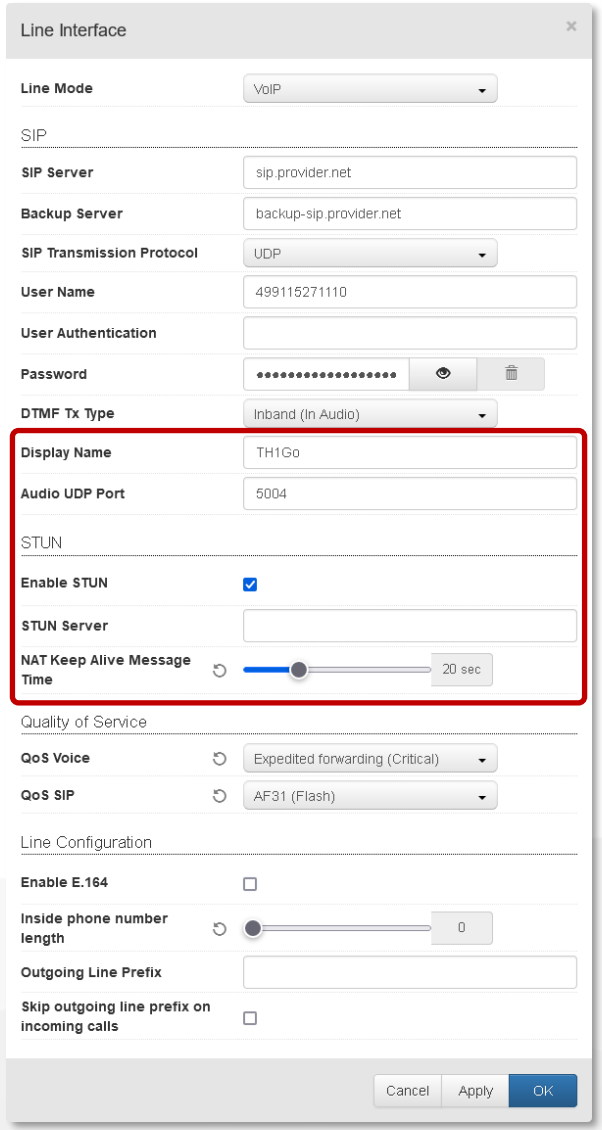
- SIP Server
  - Enter the name or IP address of the SIP Server and optionally a Backup Server.
    - A name requires a valid DNS entry under LAN settings
- SIP Transmission Protocol
  - Select UDP or TCP/IP according to the requirements of the VoIP provider
- User Name
  - Enter your User Name (typically the phone number) provided by the VoIP provider
- User Authentication
  - Enter the User Authentication provided by the VoIP provider or leave it blank if it is not specified
- Password
  - Enter Password provided by the VoIP provider
- DTMF Tx Type
  - Select Inband (In Audio) or RTP (RFC2833)



VoIP mode selected

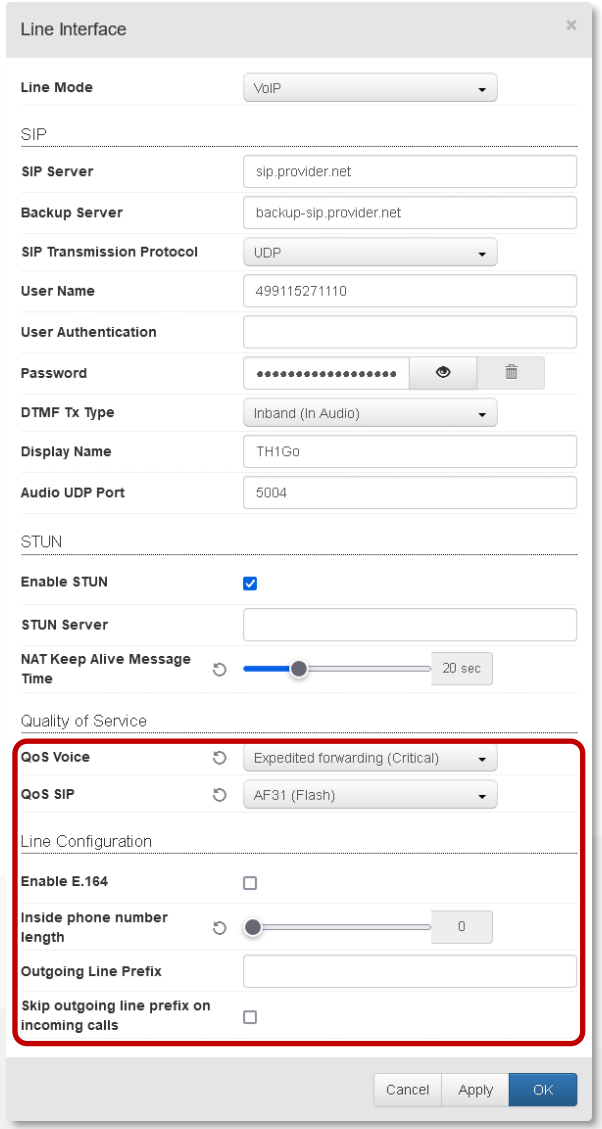
VoIP Mode (1)

- Display Name
  - The display name is displayed on the caller's phone. However, it might be overwritten by the PBX or the provider.
- Audio UDP Port
  - Change only if necessary
- STUN
  - Enable STUN only if required by the VoIP provider. In this case a STUN Server must be configured
- STUN Server
  - Enter name or IP Address of the STUN Server
    - A name requires a valid DNS entry under LAN settings
- NAT Keep Alive Message Time
  - Time to open the STUN Server for communication



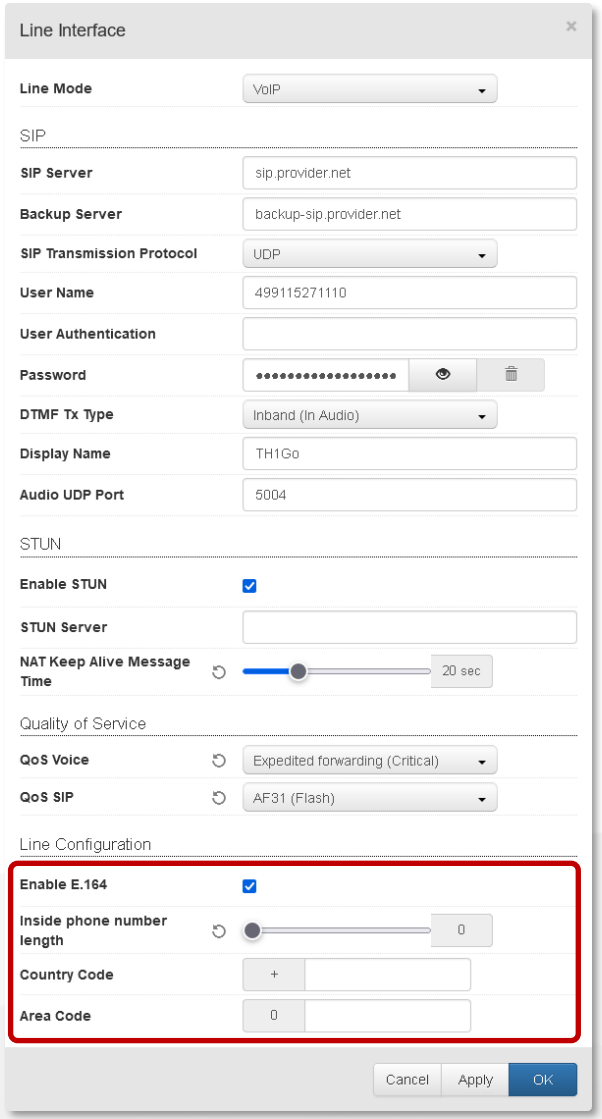
VoIP Mode (2)

- QoS Voice and QoS SIP
  - If your network offers Quality of Service (QoS), please enter the matching values under QoS Voice and QoS SIP.
- If E.164 is not enabled
  - Inside Phone number length
    - The outgoing line prefix is dialled automatically if you dial a number that is longer than the number length specified here
    - If you work with landlines, please select 0
  - Outgoing Line Prefix
    - If the system is connected to a PBX, you can enter the Outgoing Line Prefix, e.g. 0, which is automatically added if you dial an external number
  - Skip outgoing line prefix on incoming calls
    - If the phone number is displayed in the software with outgoing line prefix, the outgoing line prefix is deleted and the displayed telephone number can be saved directly to the phone book



VoIP Mode (3)

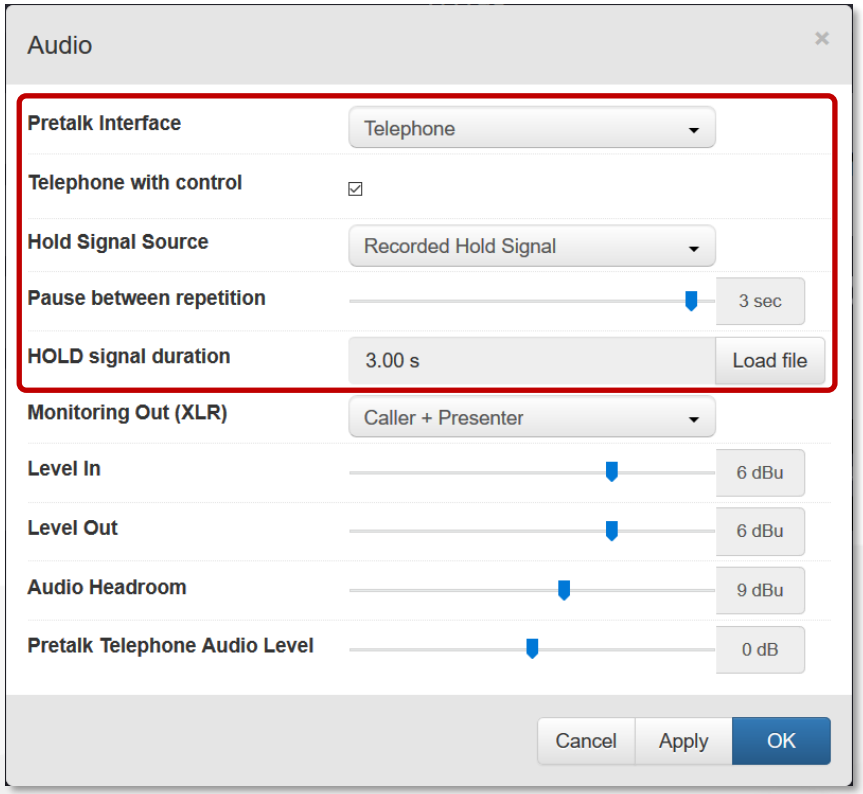
- Enable E.164 if E.164 number formatting is required by the provider.
  - Inside Phone number length
    - The outgoing line prefix is dialled automatically if you dial a number that is longer than the number length specified here
    - If you work with landlines, please select 0
- Country Code
  - Enter the local country code (e.g., 49 for Germany)
- Area Code
  - Enter the local area code if available



VoIP Mode (4)

- Pretalk Interface
  - Select here the desired Pretalk Interface:
    - Not used
    - Telephone
    - Pretalk (XLR)
    - Telephone + Pretalk (XLR)
- Telephone with control
  - When enabled, the TH1 sends the callees phone number and the line status of the TH1 to the display of the telephone. (Telephone must support FSK.)
- Hold Signal Source
  - The HOLD signal can be generated from the following sources:
    - ON AIR in (XLR)
    - HOLD in (XLR)
    - Recorded Hold Signal (Upload a preproduced audio file via the LOAD FILE button below.)
      Format of the file must be WAV/8-kHz/Mono/16 Bit
- Pause between repetition
  - If Recorded Hold Signal is set as HOLD signal source, a Pause between repetitions can be defined

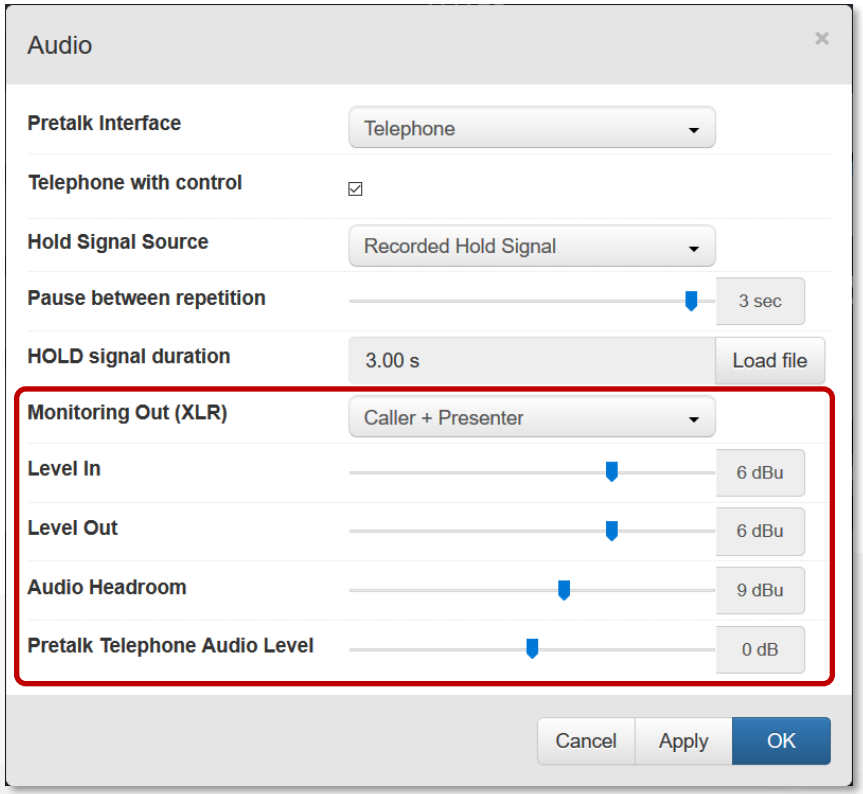
- HOLD Signal duration
  - After loading the Recorded Hold Signal file via the Load file button, the HOLD signal duration will be displayed in seconds



Audio Interfaces (1)

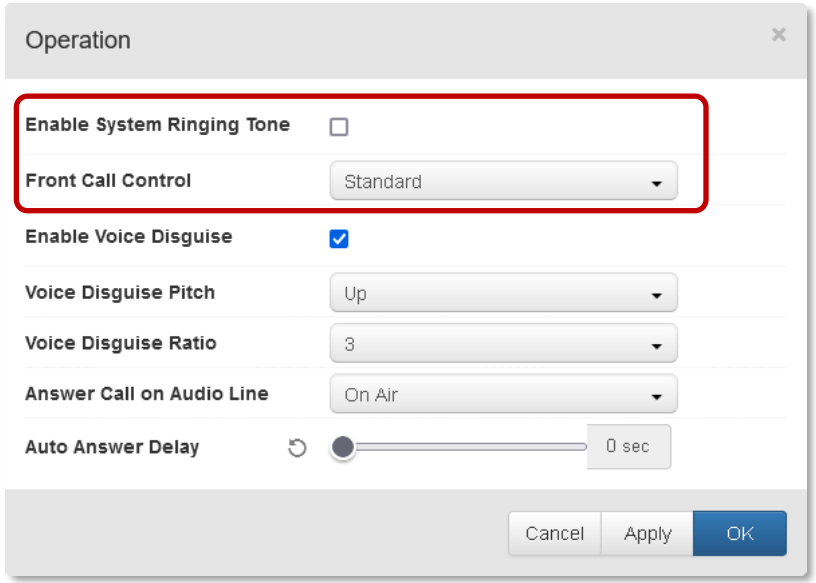
- Monitoring Out (XLR)
  - The monitoring output is only available if no Pretalk (XLR) is configured for the Pretalk Interface
  - You can choose between the following settings
    - Off
    - Caller
    - Caller + Presenter
- Level In
  - Nominal Level of the analogue input
  - Adjustable in a range from -3 to +9 dBu (default: +6 dBu)
- Level Out
  - Nominal Level of the analogue output
  - Adjustable in a range from -3 to +9 dBu (default: +6 dBu)
- Audio Headroom
  - Headroom of the analogue audio interface
  - Adjustable in a range from 0 to 15 dBr (default: +9 dBr)

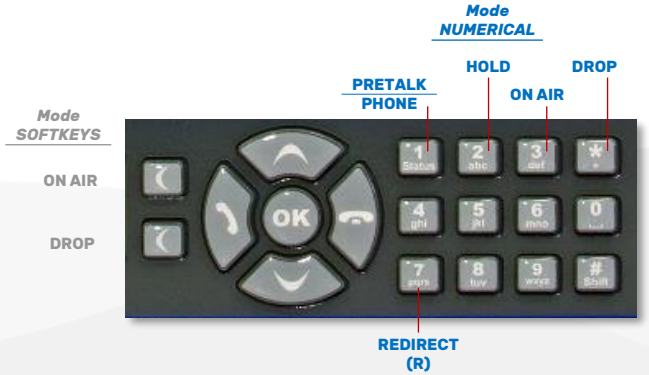
- Pretalk Telephone Audio Level
  - Audio Level to the optional connected POTS telephone
  - Adjustable in a range from -12 to +12 dB



Audio Interfaces (2)

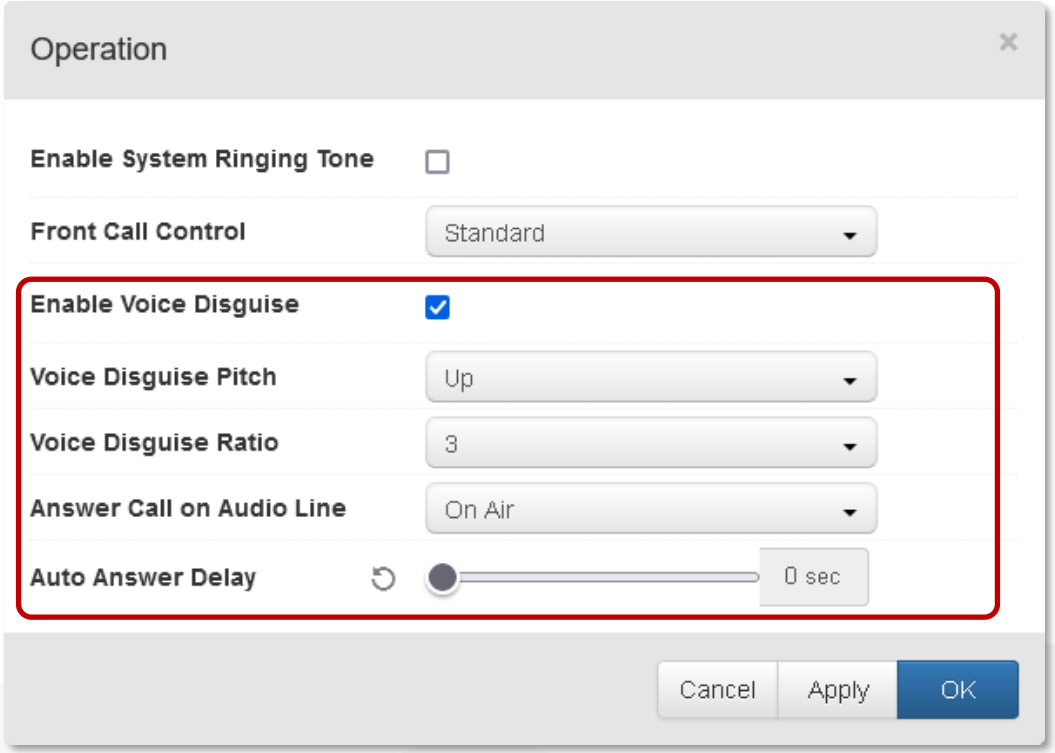
- Enable System Ringing Tone
  - If activated, the system internal buzzer signals incoming calls
- Front Call Control
  - Switching the line status (e.g., Pretalk, Hold, On Air etc.) via the front keypad can be configured in three different ways:
  - Standard
    - Standard function as labelled on the keys
  - Numerical Keypad
    - No connection → Standard function as labelled on the keys
    - Connection established:
      - 1 → PRETALK/PHONE
      - 2 → HOLD
      - 3 → ON AIR
      - 7 → Redirect Call (R) → DROP
  - Softkeys
    - The two softkeys allows a direct switching between ON AIR and DROP





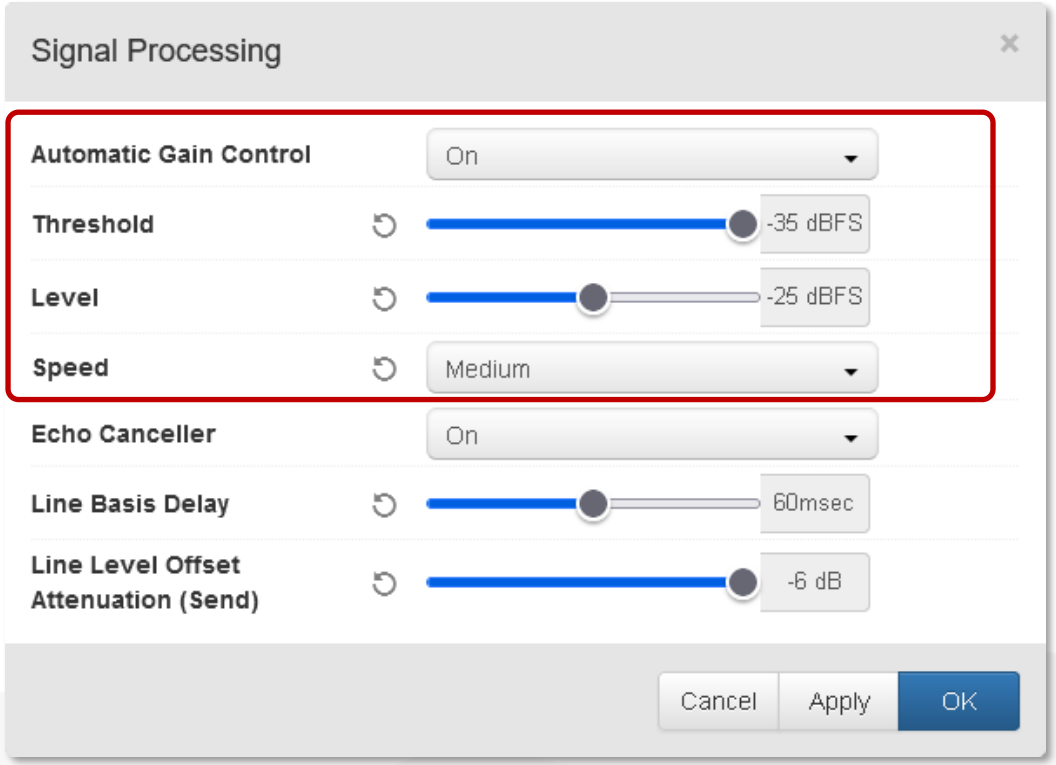
Operation (1)

- Enable Voice Disguise
  - If activated, Voice Disguise is enabled
- Voice Disguise Pitch
  - As pitch can be selected
    - Up for a shift to higher frequencies
    - Down for a shift to lower frequencies
- Voice Disguise Ratio
  - The strength of the pitch shift can be adjusted in the range from 1 (low) ... 4 (high)
- Auto Answer Call on Audio Line
  - This activates automatic call answering for incoming calls. You can choose:
    - Not used to disable the function
    - Switch caller to HOLD
    - Switch caller to ON AIR
- Auto Answer Delay
  - This function can be used to delay automatic call answering in the time range of 0 ... 31 seconds. During this time, an incoming call is signaled as usual by ringing and/or flashing



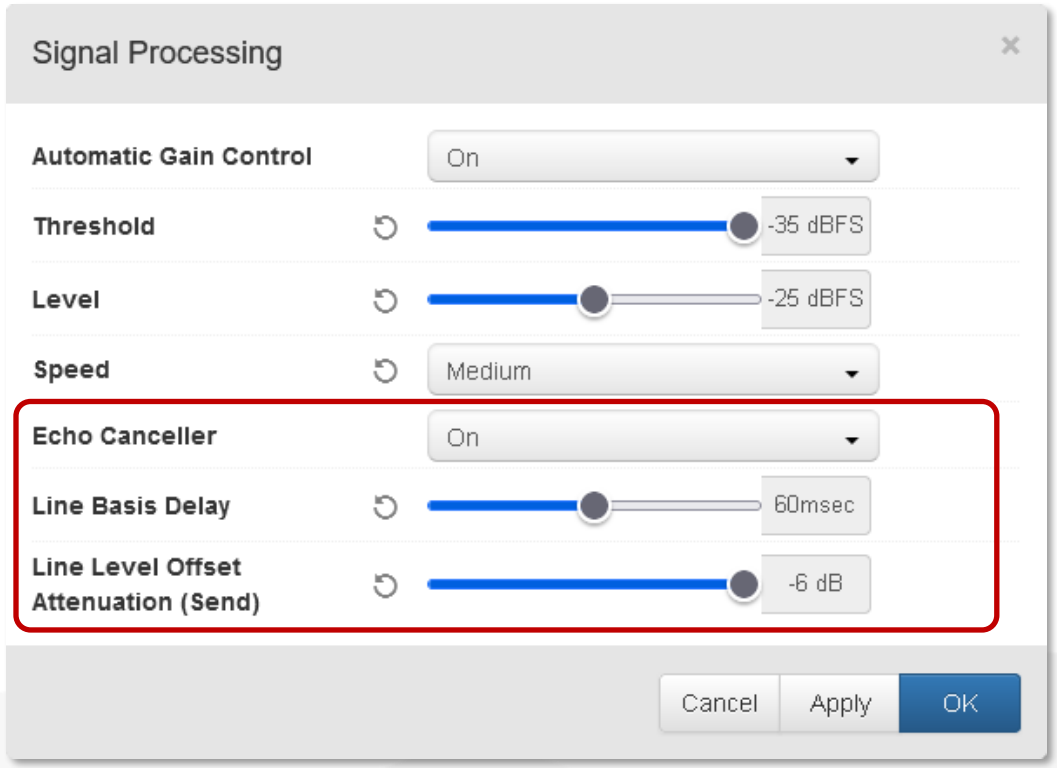
Operation (2)

- Automatic Gain Control
  - If the integrated Automatic Gain Control is switched On, the function can be configured via three parameters
  - The Threshold value determines the level when the AGC should start work
    - Range: -45 .... -35 dBFS
    - Recommended: -40 dBFS
  - The Level parameter defines the target level
    - Range: -30 .... -20 dBFS
    - Recommended: -25 dBFS
  - The Speed selection defines how fast the AGC should work. Values are:
    - Slow, Medium, Fast
    - Recommended: Medium or Slow



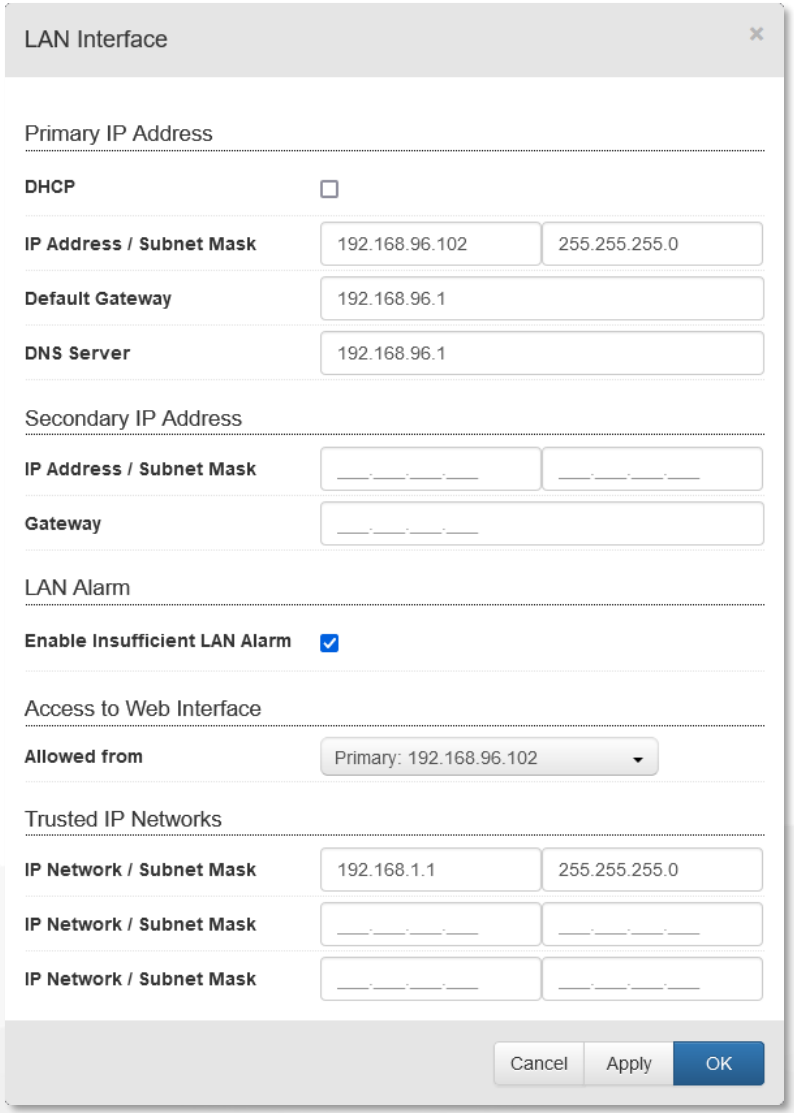
Signal Processing (1)

- Echo Canceller
  - The integrated Echo Canceller can be switched On or Off
    - Especially callers still using POTS lines can cause a strong echo. Therefore, we always recommend to activate the Echo Canceller.
    - The maximum echo tail time is approx. 120 msec. In this time window the echo canceller can reduce the echo.
- Line Basis Delay
  - If the system is connected to a PBX or callers are forwarded via an external platform, the Line Basis Delay of the (local) system can be set as the basis value so that a possible echo falls within the time window of 120 msec in any case.
    - Range: 0 … 120 msec
    - Recommended: Landline = 0 msec, PBX > 40 msec
- Line Level Offset Attenuation (Send)
  - The send level can be reduced in a range from -6 dB … -15 dB
    - If you have problems with echoes you should try to reduce this level slightly



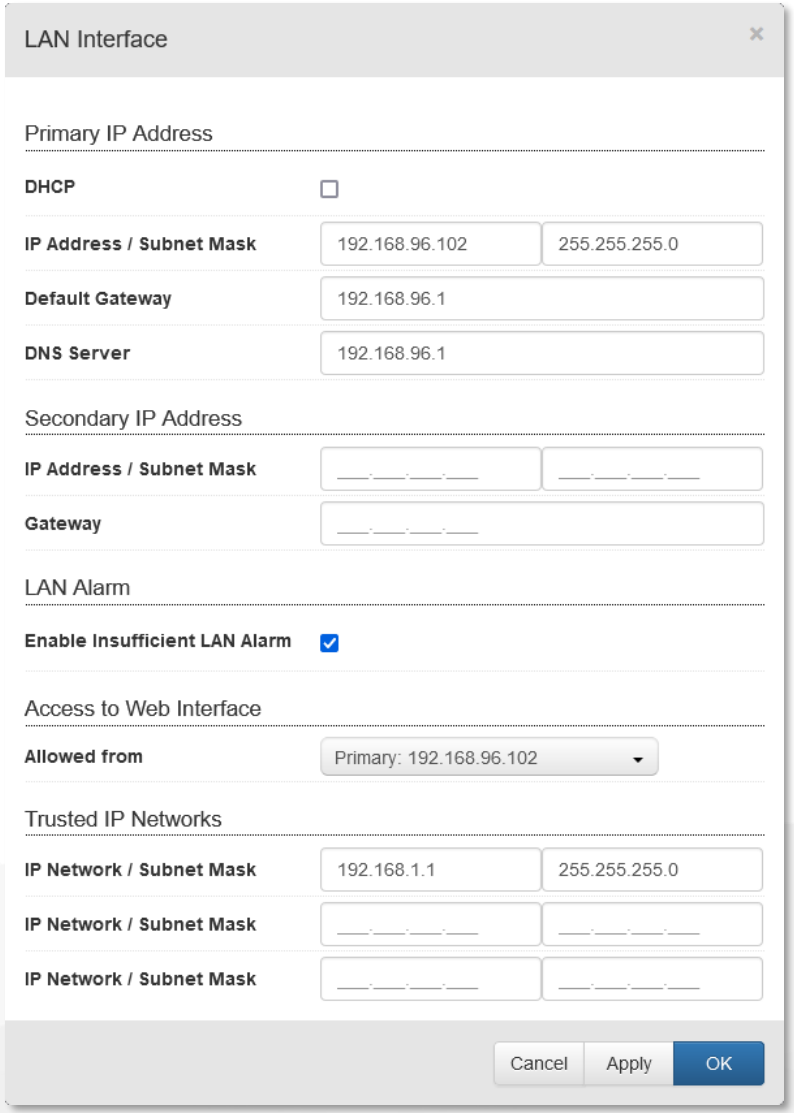
Signal Processing (2)

- For easy setup you can use DHCP to obtain all IP parameters for the Primary IP Address automatically
  - The assigned IP address is visible by pressing the DROP key on the front panel of the system
- Alternatively you can assign all parameters manually
- Primary IP Address
  - Please enter IP Address, Subnet Mask, Default Gateway and DNS Server
  - Note that VoIP must use the Primary IP Address network
- Additionally you can enter a Secondary IP Address if you want to separate e.g. the VoIP and the control network
  - Please enter IP Address, Subnet Mask and Gateway
- LAN Alarm
  - Activate Enable Insufficient LAN Alarm to get an alarm if LAN interface is too slow.

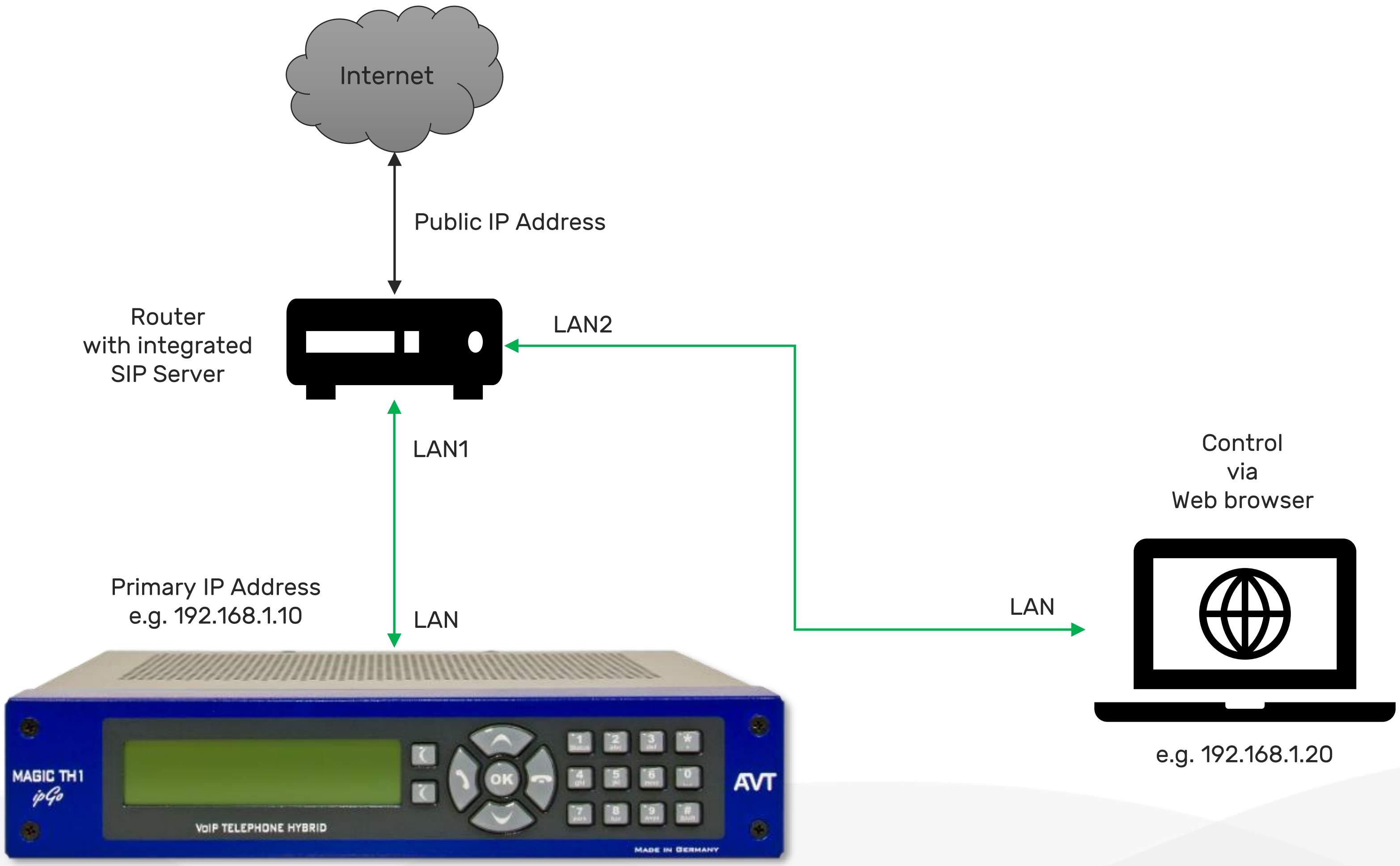


LAN Interface (1)

- Access to Web Interface
  - We recommend to use different IP address ranges for VoIP and Control – if your router supports that – to be sure that nobody can easily login into your system via the VoIP network
    - Therefore the primary network must be used for VoIP. You should use the secondary network for control
- Trusted IP Networks
  - By default the unit allows access to the web interface from the local network only.
  - PCs located in trusted networks can access the unit as if they where in the local network.
  - Accessing the unit from unknown networks requires the user to enter the remote password.



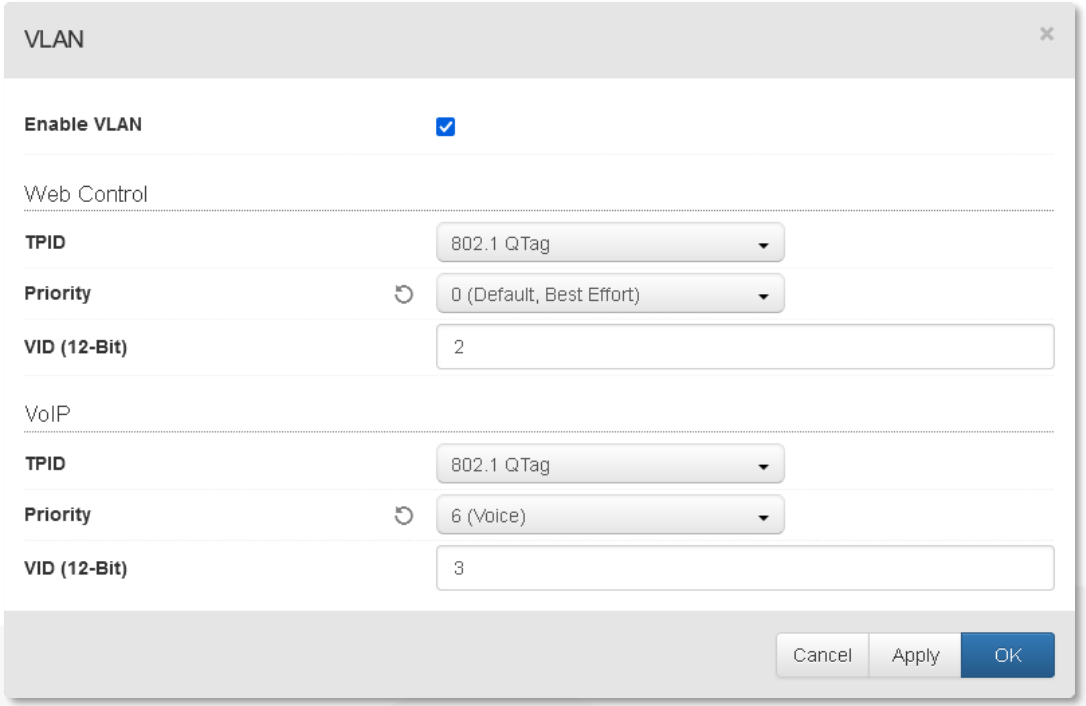
LAN Interface (2)



AVT Audio Video Technologies

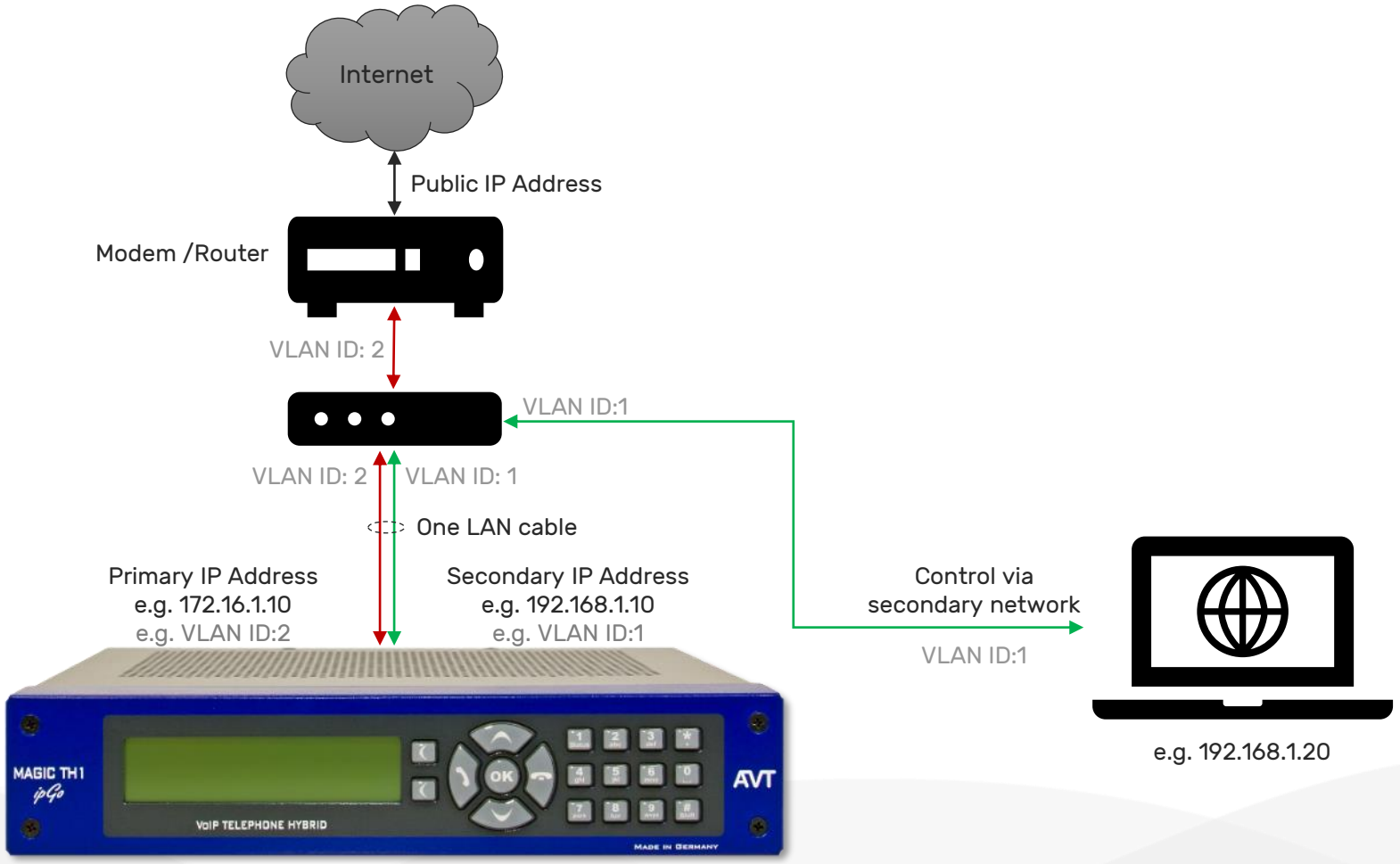
Typical Application

- As already mentioned, we recommend to separate the VoIP network from the Control network for VoIP applications in general
- An additional possibility is to use VLANs if your switch/router supports that
- The configuration is done under VLAN
- Before you Enable VLAN, please make sure that your administrator has configured the Switch Port correspondingly
  - Activate the 802.1Q Tagging for Web and VoIP services
  - Please select the desired Priority (Standard for VoIP = 6 Voice)
  - Under VID you need to enter the VLAN IDs which has been allocated in the switch/router
- If you did something wrong in the configuration and you have no longer access to your system you can switch-off the VLAN support via the front panel of the system



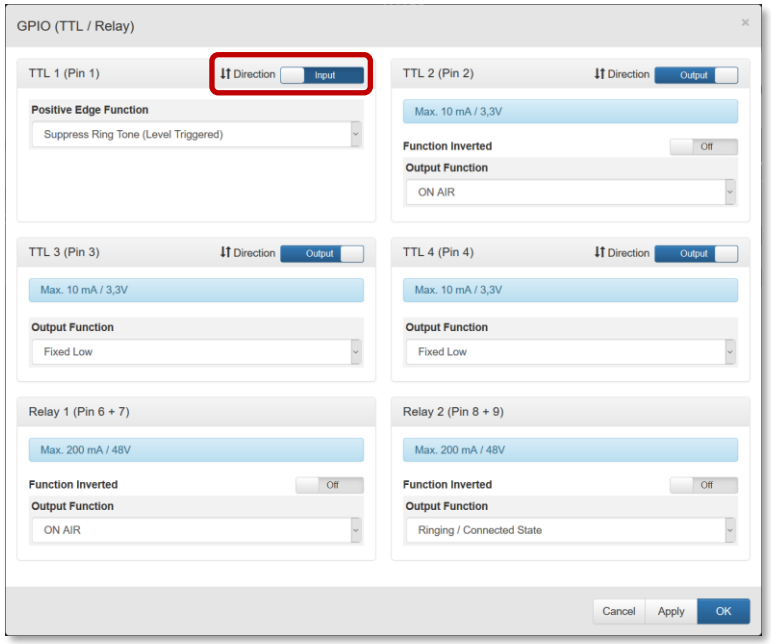
Audio Video Technologies

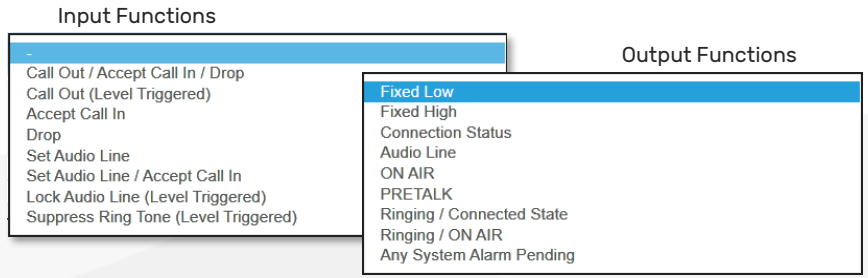
VLAN Configuration (1)



VLAN Configuration (2)

- MAGIC TH1 Go provides the following GPIO contacts
  - 4 x programable TTL in-/outputs
  - 2 x relay outputs
- TTL pins can be configured as Input or Output
  - Various functions are offered
    - Inputs
      - An input is high active by default. Set an input to GND
        (Pin 5) to trigger a switching operation
    - Outputs max. current 10 mA
      - max. voltage 3.3V
- Relay contacts
  - Various output functions are offered
    - max. current 200 mA
    - max. voltage 48V
- Please notice the maximum permissible currents and voltages





GPIO (TTL/Relay)

MAGIC TH1Go

Administration

- Registration
- Firmware Download
- System Panel
- Set Factory Settings
- Restart Device
  About

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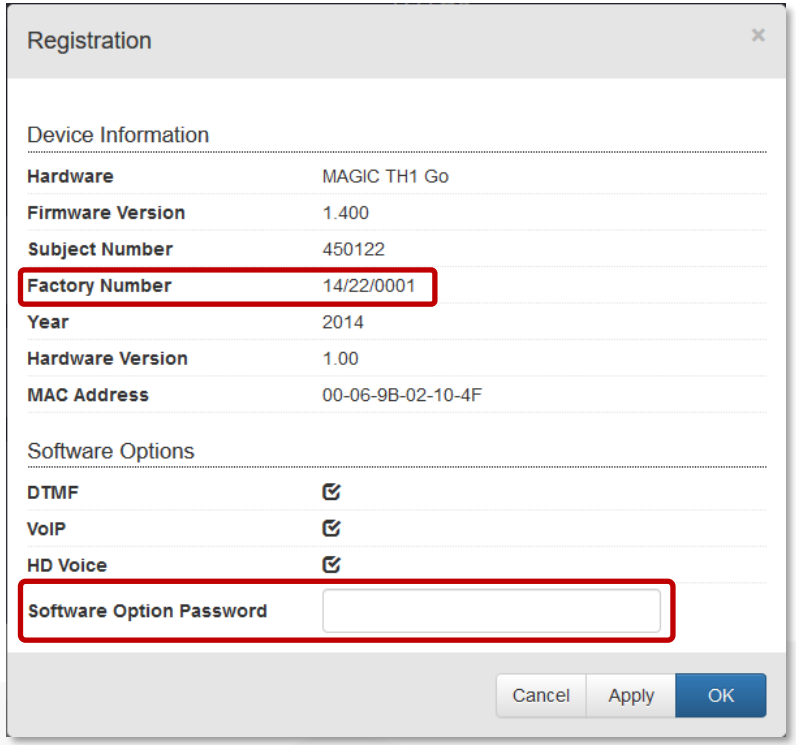
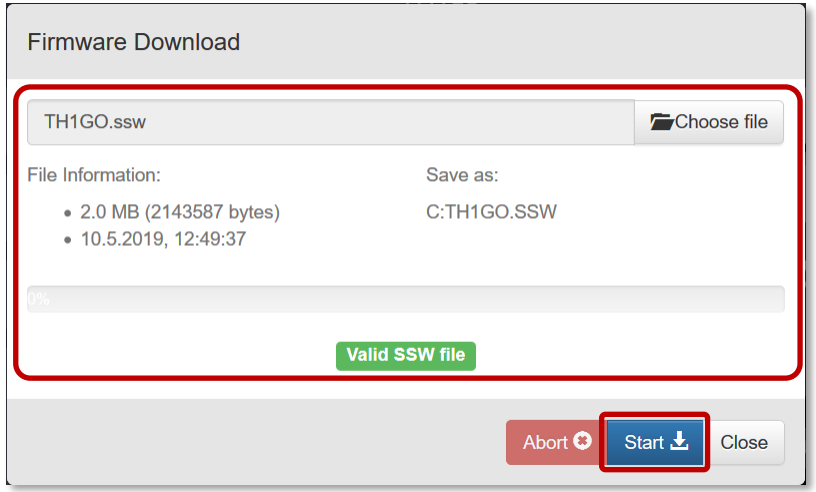
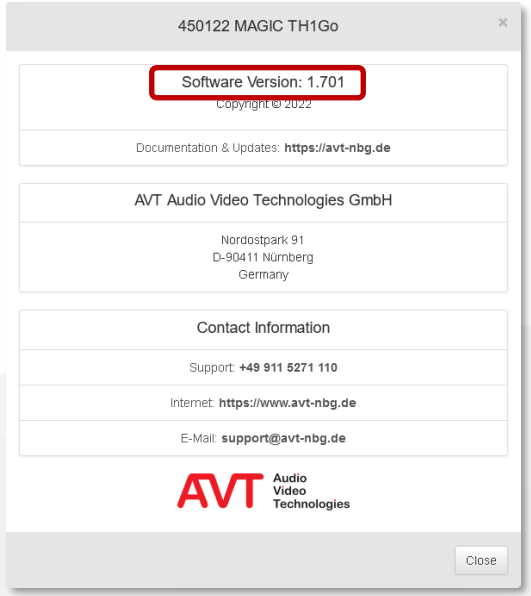
- Via Menu → Registration you will find the available Software Options for the system
  - The marked Options are already enabled in your system
- You will need the Factory Number of the device if you want to purchase additional software options later
- You will receive a licence with password, which you must enter under Software Option Password to activate the ordered software option



Registration

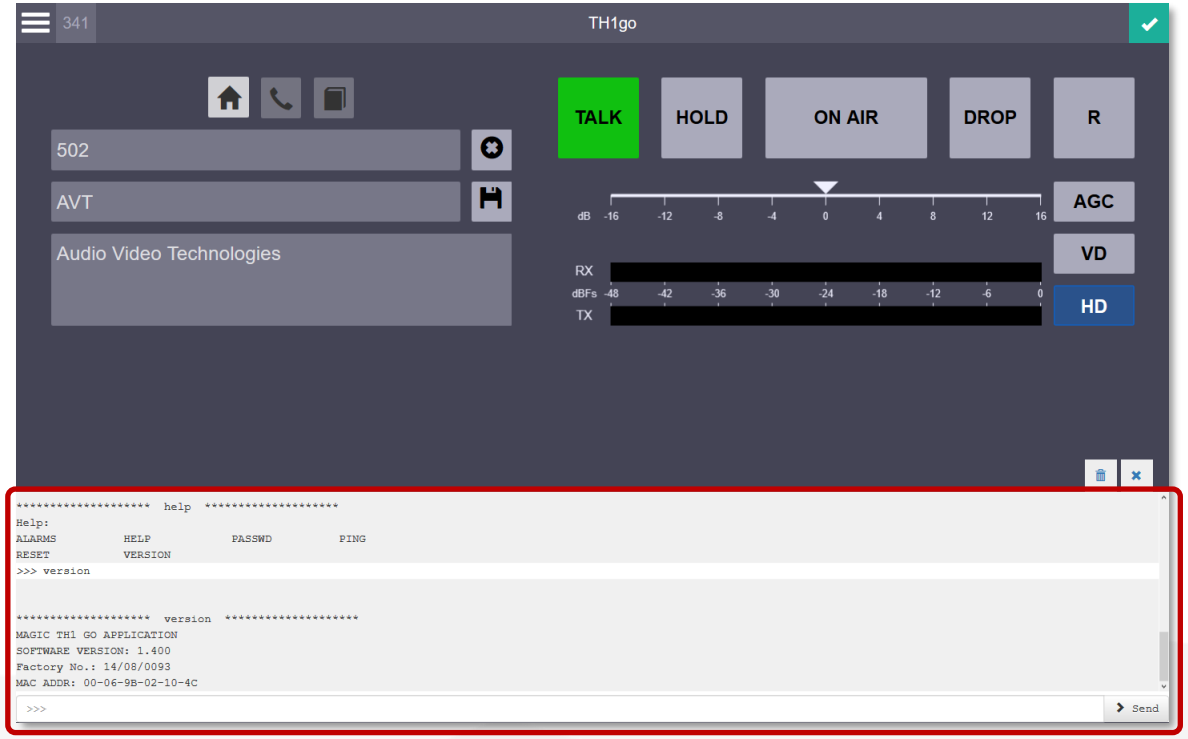
- New firmware can be downloaded from our Website under Downloads → Software
- The firmware file TH1GO.SSW must be downloaded to the MAGIC TH1Go via the Firmware Download menu
  - Press Choose File to select the downloaded firmware file
  - If the firmware file is valid, you see the size, the release date as well as the confirmation Valid SSW file
  - Press Start to update the system
  - Do not switch-off the system during a firmware update
- The new software version can be displayed via Menu → About





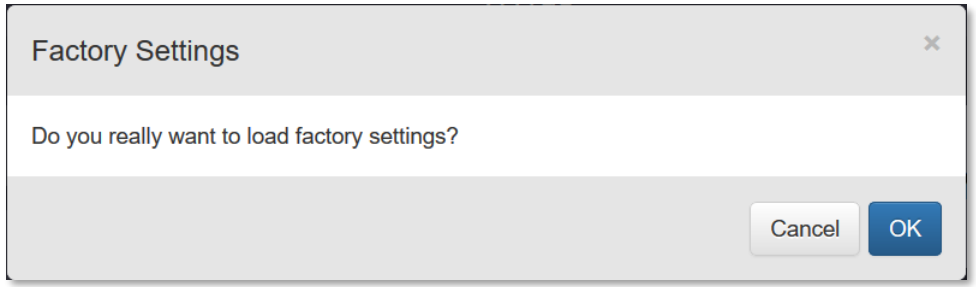
Firmware Download

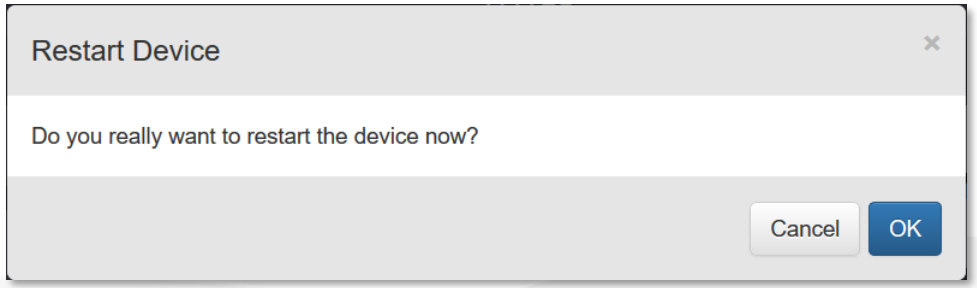
- Via Menu → System Panel a command window can be displayed, which offers special functions for e.g., extended troubleshooting
- The System Panel is displayed below the main window
  - Press 🗑 to clear the window
  - Press ✕ to close the command window
- By entering the “help” all commands available for users are displayed
  - ALARMS
    - To check alarms and reset alarm counters
  - PING
    - To check connections to e.g., SIP or DNS server via the system ethernet interface
  - RESET
    - To restart the system
  - VERSION
    - To show firmware version, factory number, MAC address



System Panel

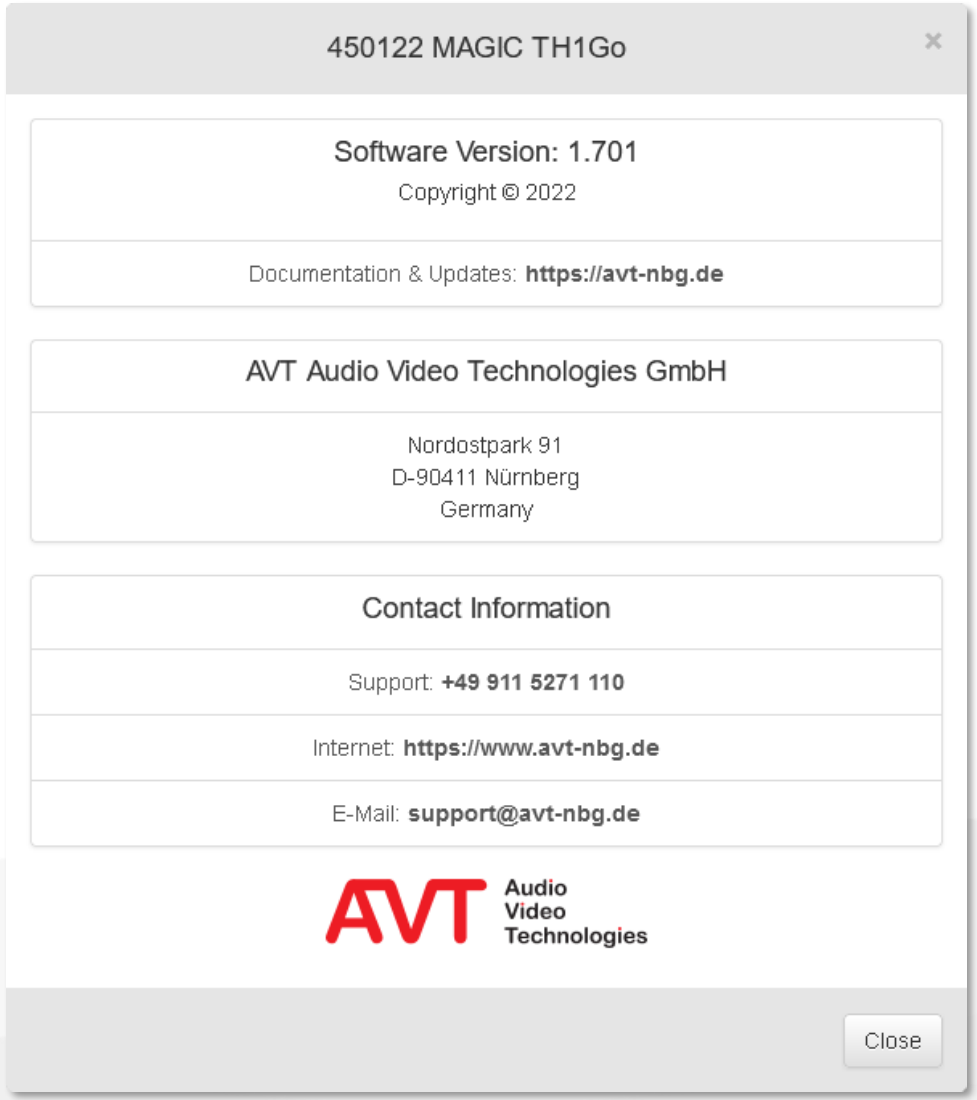
- Using Menu → Set Factory Settings, the system can be reset to the factory settings
  - For safety reasons, the action must be confirmed with OK
  - To be able to access the device via the web browser again, you must first reset the old IP address via the front keyboard (see Basic Configuration)
- Via Menu → Restart Device a soft restart start is initiated
  - For safety reasons, the action must be confirmed with OK





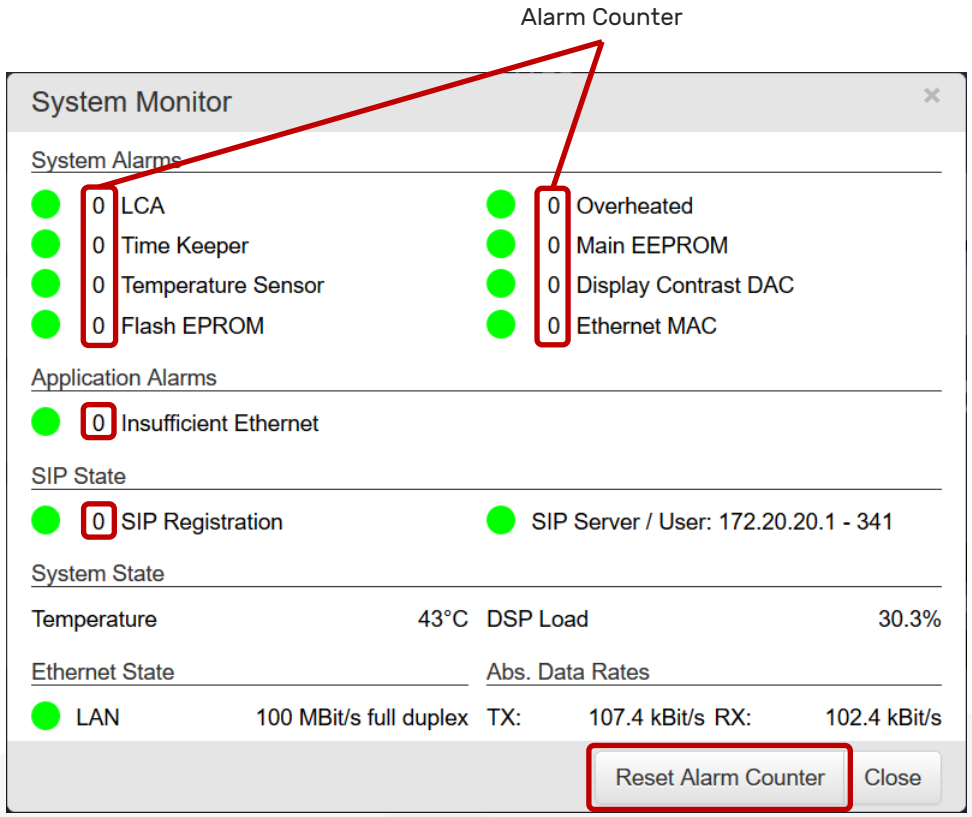
Factory Settings & Restart Device

- The About Panel displays the following information
  - Firmware version
  - AVT contact address
  - Support details
    - Phone number
    - Email address
  - Website



About Panel

- The system monitor displays all possible System and Application Alarms
  - To open the System Monitor, double-click the ☑ icon in the upper right corner of the PC software or select Menu → System Monitor
- Alarm indication
  - GREEN No alarm
  - RED Active alarm
  - BLUE Occurred alarm, but no longer active
- The alarm counter indicates the number of alarms that have occurred since the last time the Reset Alarm Counter button was pressed.
- The VoIP version displays additionally the SIP Registration and details of the SIP Server
- General information such as Temperature, DSP Load, Ethernet State and Data Rates are also displayed



System Monitor

MAGIC TH1Go

Tips & Tricks

- Telephone Control Codes

info@avt-nbg.de
www.avt-nbg.de

- The TH1 can be controlled via the connected telephone using DTMF tones.
- Press the following key combinations or store them as quick dials on the telephone:

#	CALL
*1	PRETALK
*2	ON AIR
*3	HOLD
*4	DROP
*5	AGC ON
*6	AGC OFF
*7	Voice Disguise ON
*8	Voice Disguise OFF
*9	LOCK

Telephone Control Codes

Audio
Video
Technologies

Support Hotline
+49 911 2571 110

Support-Portal
avt-nbg.zammad.com

Support E-Mail
support@avt-nbg.de

Support

info@avt-nbg.de
www.avt-nbg.de