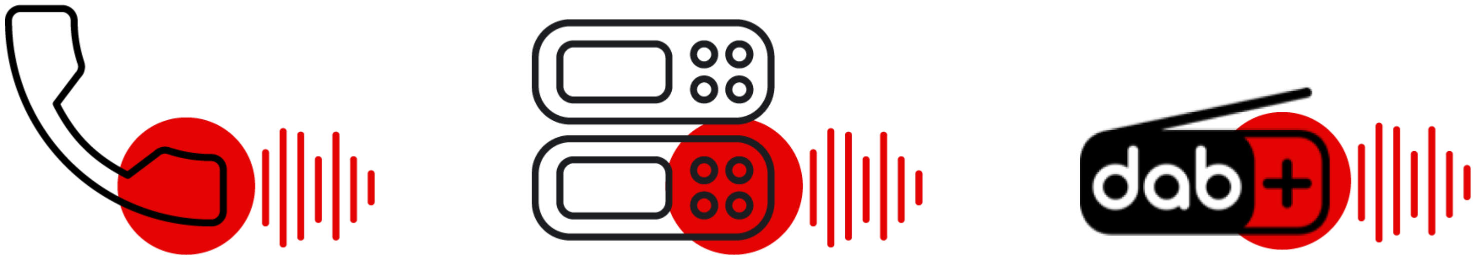
AVT Audio Video Technologies



MAGIC AC1 Go

Configuration Guide

Version: 2.412 (19. December 2023)

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Contents

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MAGIC AC1 Go

Overview

- Hardware
- Audio Codecs
- Operation Modes

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Hardware - Front

- Half width 19" x 1U system with external power supply
- 19" mounting brackets included
- Illuminated graphic display with 160 x 32 pixels & front keypad.
  - For configuration and operation

Hardware - Rear

- Earthing screw
- External Power Supply (+12 V DC / 15 W)
- RS232 socket with 2 RS232 data interfaces
- TTL/Relay socket
  - 4 x TTL Input / Output
  - 2 x Relay

- 1 x LAN Interface
- 1 x Stereo Audio Input / Output
  - Configurable as analogue interface or AES/EBU interface

- Standard
    - Signalling
        - Audio over IP (AoIP)
        - Direct IP connection
    - Codecs
        - G.711
        - G.722
        - OPUS
        - FLAC
        - PCM 16/20/24 Bit
        - ISO/MPEG-2 Layer 2
    - Up to five simultaneous PC control connections

- Optional
    - Codecs
        - ISO/MPEG-2 Layer 3
        - AAC-LD, AAC-ELD
        - AAC-LC
        - HE-AAC v1/v2
    - AES67 (4 channels)
    - MAGIC System Manager Integration
    - MAGIC PhonerSet

Features

	ISO/MPEG1/2 Layer 2/3	AAC-LD AAC-ELD	HE-AAC-LC HE-AAC-V1/2	G.722	G.711	PCM16/20/24	OPUS	FLAC
Algorithm	Sub-Band, Psychoacoustics	Sub-Band, Psychoacoustics	Sub-Band, Psychoacoustics	ADPCM	PCM	N/A	SILK CELT	PCM
Structure	Frames	Frames	Frames	Bytes	Bytes	Bytes	Frames	Frames
Bandwidth	20-Hz … 20-kHz	20-Hz … 20-kHz	20-Hz … 20-kHz	50-Hz … 7-kHz	300-Hz … 3.4-kHz	20-Hz … 20-kHz	20-Hz…20-kHz	20-Hz…20-kHz
Sampling Rates (kHz)	MPEG1: 32, 48 MPEG2: 16, 24	16, 24, 32, 48	LC: 32, 48 V1/V2: 16, 24	16	8	32, 48	max. 48 dynamic	48
Data rates (kbit/s)	Typical: 128…192kbit/s	Typical: LD: 64… 128 ELD: 48…136	Typical: LC: 96 V1: 80 V2: 64	48, 56, 64	56, 64	512…2304	16…384	Typical: 300 … 2000
Delay	Layer 2: 150 ms Layer 3: 250 ms	LD: < 50 ms ELD: < 30 ms	LC: 350 ms V1/V2: 450 ms	< 10 ms	< 5 ms	< 3 ms	<30 ms	

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Audio Codecs

- Audio transmission via UDP packets using the RTP protocol
- Numbering of the packets for the receiver
- Configurable packet size
- The packet size is variable if it is not determined by the codec that is used.
- Automatically adapting or fixed decoder buffer
- Max. Buffer size: 2 seconds
- Mitigation of lost packets by repeating packets on the receiving side
- QoS (DiffServ) function to prioritize audio packets
- Remote GPIO: Transmission of GPI states between AVT audio codecs.
- Secure Streaming Function
  - Sending two streams with identical content
  - A delay between the streams can be configured
  - Streams can be sent via separate VLANs

- IP Leased Line Dial Up
  - Bidirectional point-to-point connection
  - IP address of the receiver is entered when connecting
  - Compatible with MAGIC ACip3 and Luci Live
- IP Leased Line Standard
  - Bidirectional point-to-point connection
  - IP address of the receiver is entered in the configuration
  - Auto connect after power up
  - Compatible with MAGIC AC1 XIP and MAGIC ACip3
- IP Leased Line Extended
  - Unidirectional point-to-point connection
  - IP addresses of the receivers are entered in the configuration
  - Auto connect after power up
  - Independent configuration of the audio decoder
  - Simulcast streaming to up to 5 decoders
  - Compatible with MAGIC ACip3
  - Raw stream compatibility mode available

Audio Video Technologies

IP Leased Line

- Establish a connection
  - by dialling a phone number via a SIP server
  - by entering the IP address of the receiver via Direct SIP
- Support for up to 5 SIP provider profiles which are active simultaneously
- Uses SDP to negotiate the coding algorithm
- Audio transmission via RTP over UDP
- Numbered packets allow for packet loss detection
- The packet size is variable if it is not determined by the codec used.
- Max. buffer size: 2 seconds
- Mitigation of lost packets by repeating packets on the receiving side
- Support for the STUN protocol (Session Traversal Utilities for NAT)
  - Simplified: Conversion of the internal IP address into an external IP address
- Supports Quality of Service (DiffServ) to prioritize audio and SIP independently
- Interworking with SIP telephones, in SD quality or HD-Voice
- Secure Streaming Function
  - Sending two streams with identical content
  - A delay between the streams can be configured

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MAGIC AC1 Go

First Start

- Connecting the PC software

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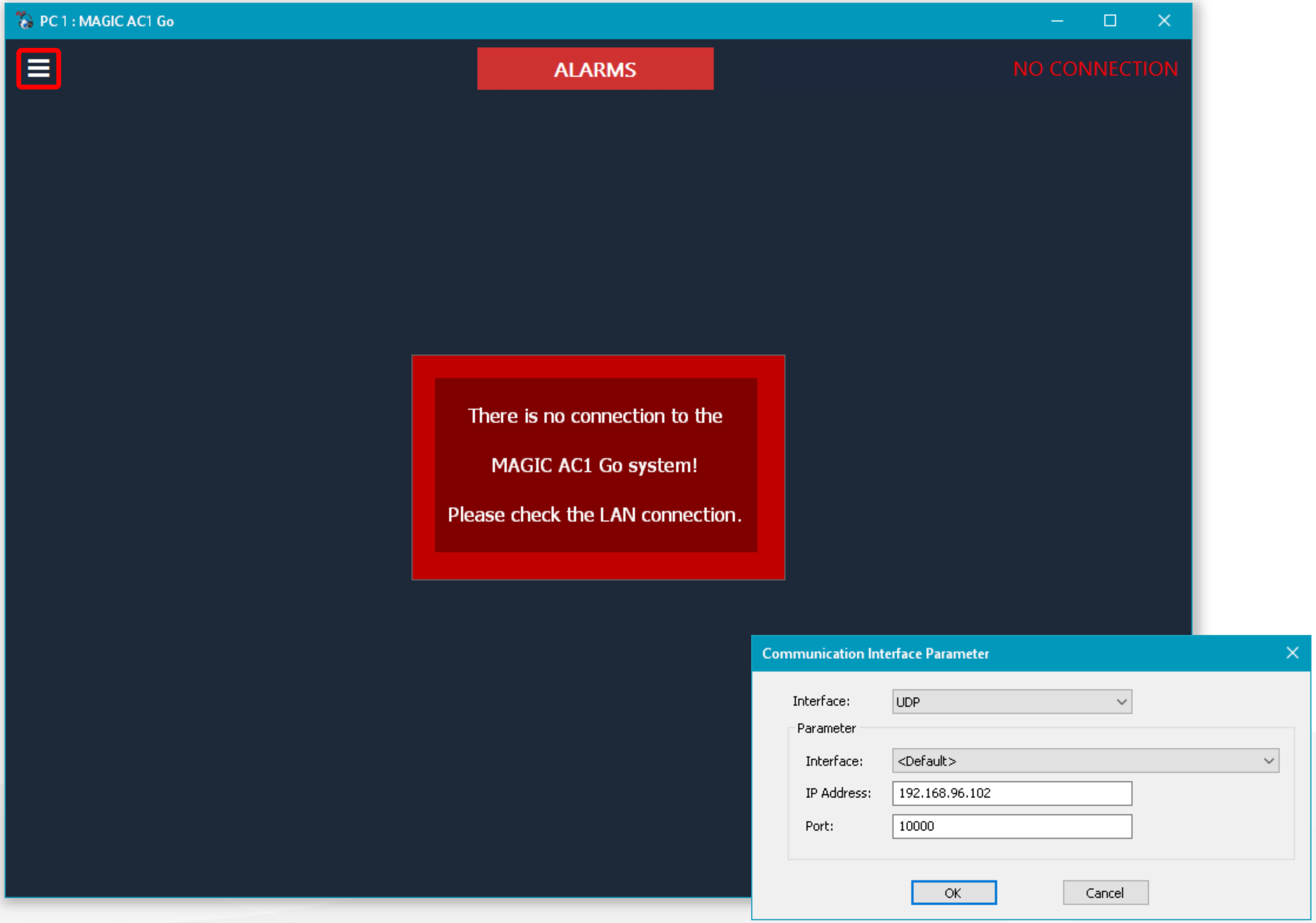
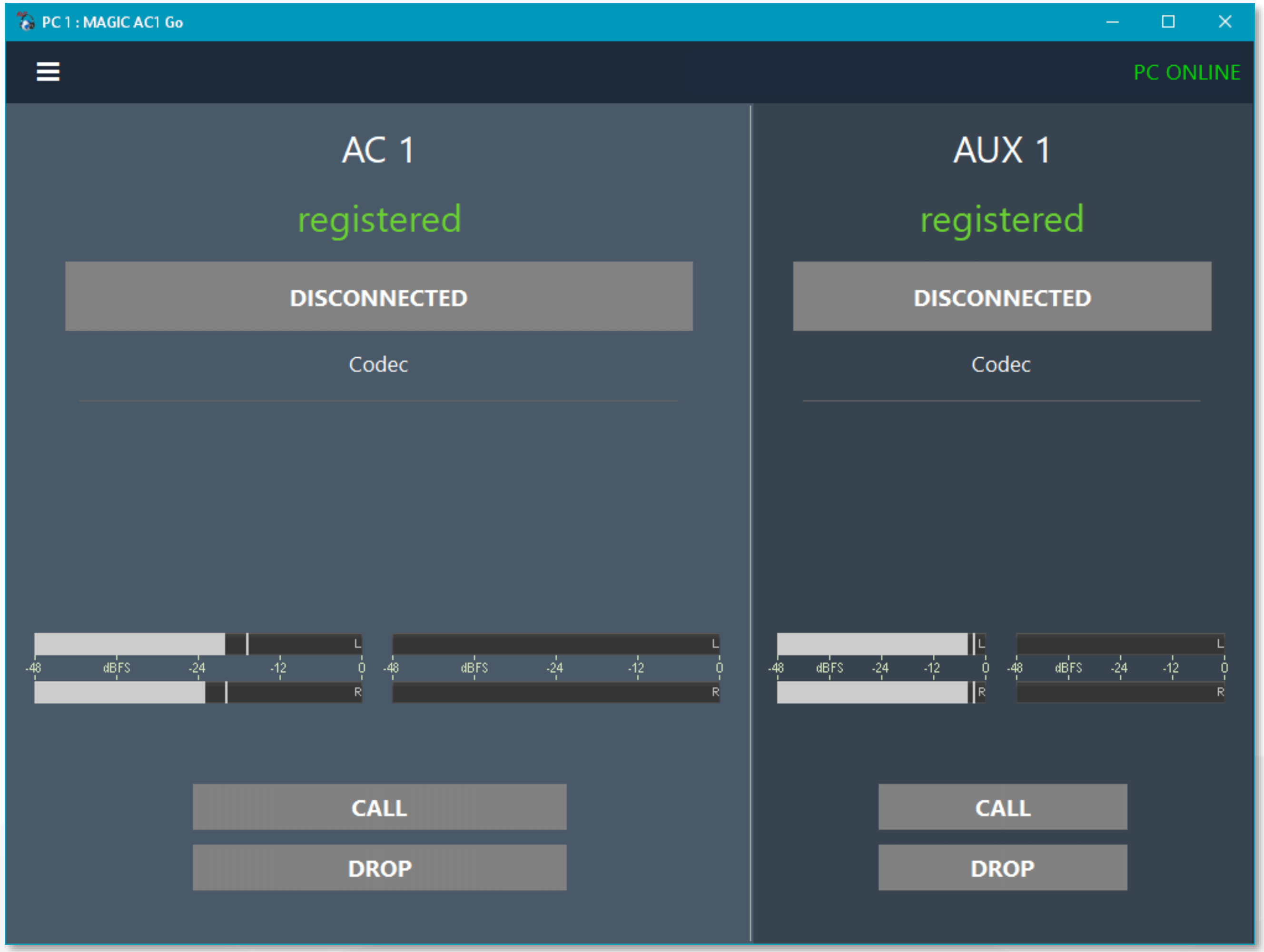
- Start the MAGIC AC1 Go PC software with administrator rights.
  - Right-click on the icon on the desktop and select “Run as Administrator”.
- Click on the menu button in the upper left corner.
- Open **Configuration > Control Interface**.
- Enter the IP address of the device in the "IP Address" field
- Defaults:
  - Interface: UDP
  - Interface: <Default>
  - IP: 192.168.96.102
  - Port: 10000



Audio Video Technologies

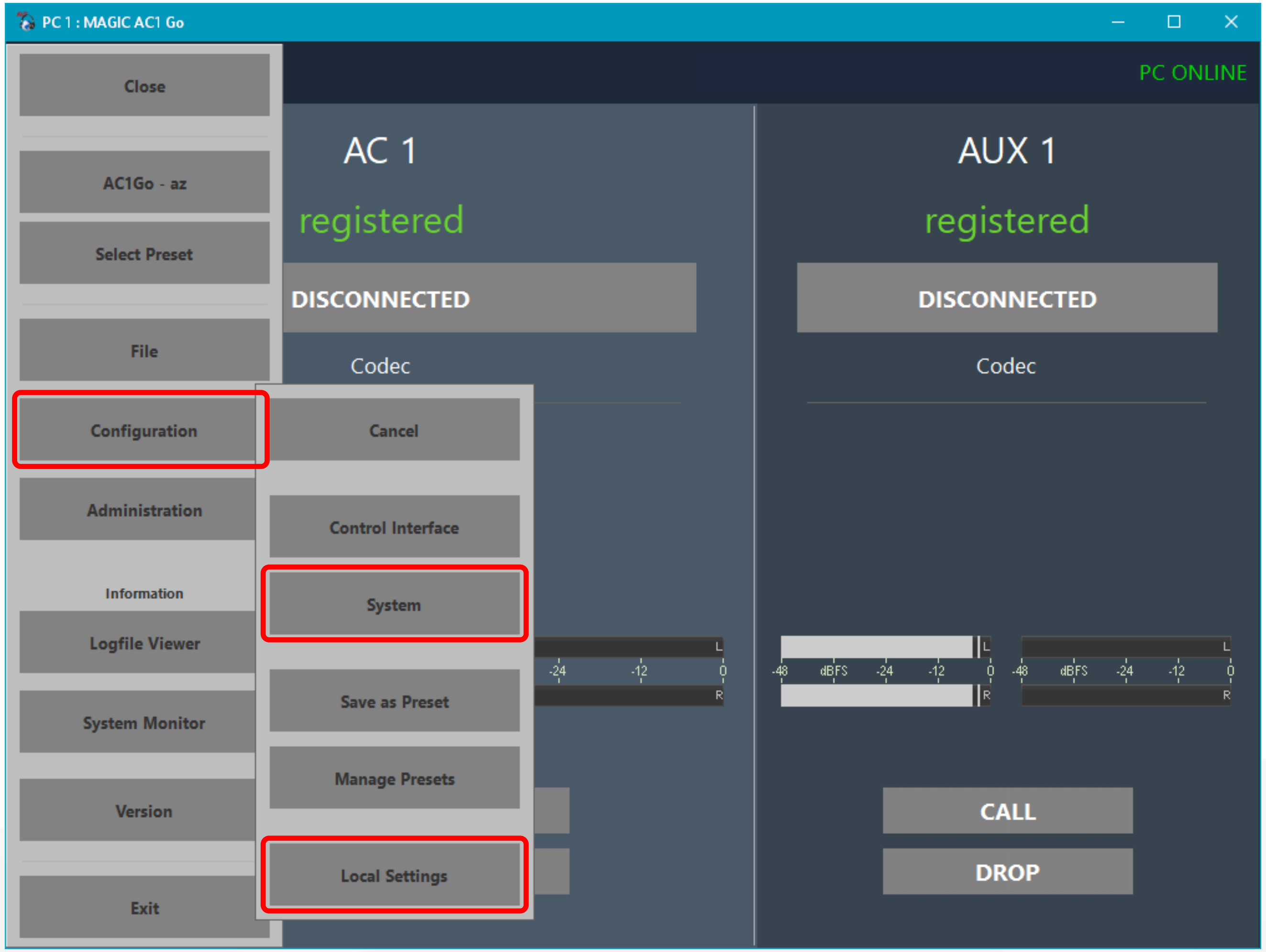
First Start

- "PC ONLINE" in the top right corner, indicates that the PC software is connected to the unit.
- The number of channels displayed depends on the configuration.
- The codec channel AC 1 provides high quality audio transmission.
- The AUX 1 channel provides audio transmission with SD and HD-voice quality.



Main Panel

- Go to **Menu > Configuration > System** to enter the system configuration.
  - The system configuration is stored on the MAGIC AC1 Go unit.
- Go to **Menu > Configuration > Local Settings** to change the appearance of the PC software.
  - The local settings are stored on the PC.



MAGIC AC1 Go

Configuration

- Local Settings

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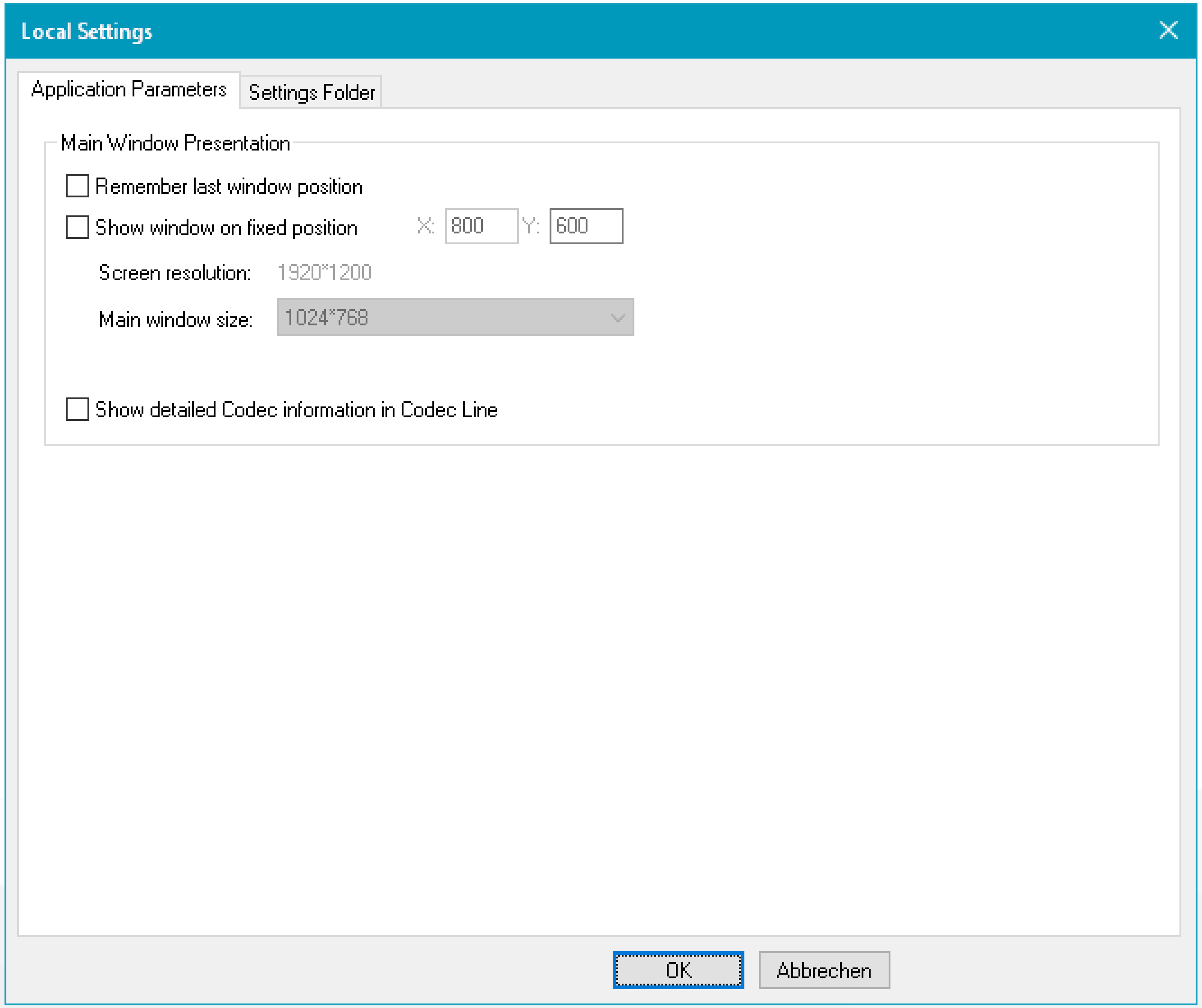
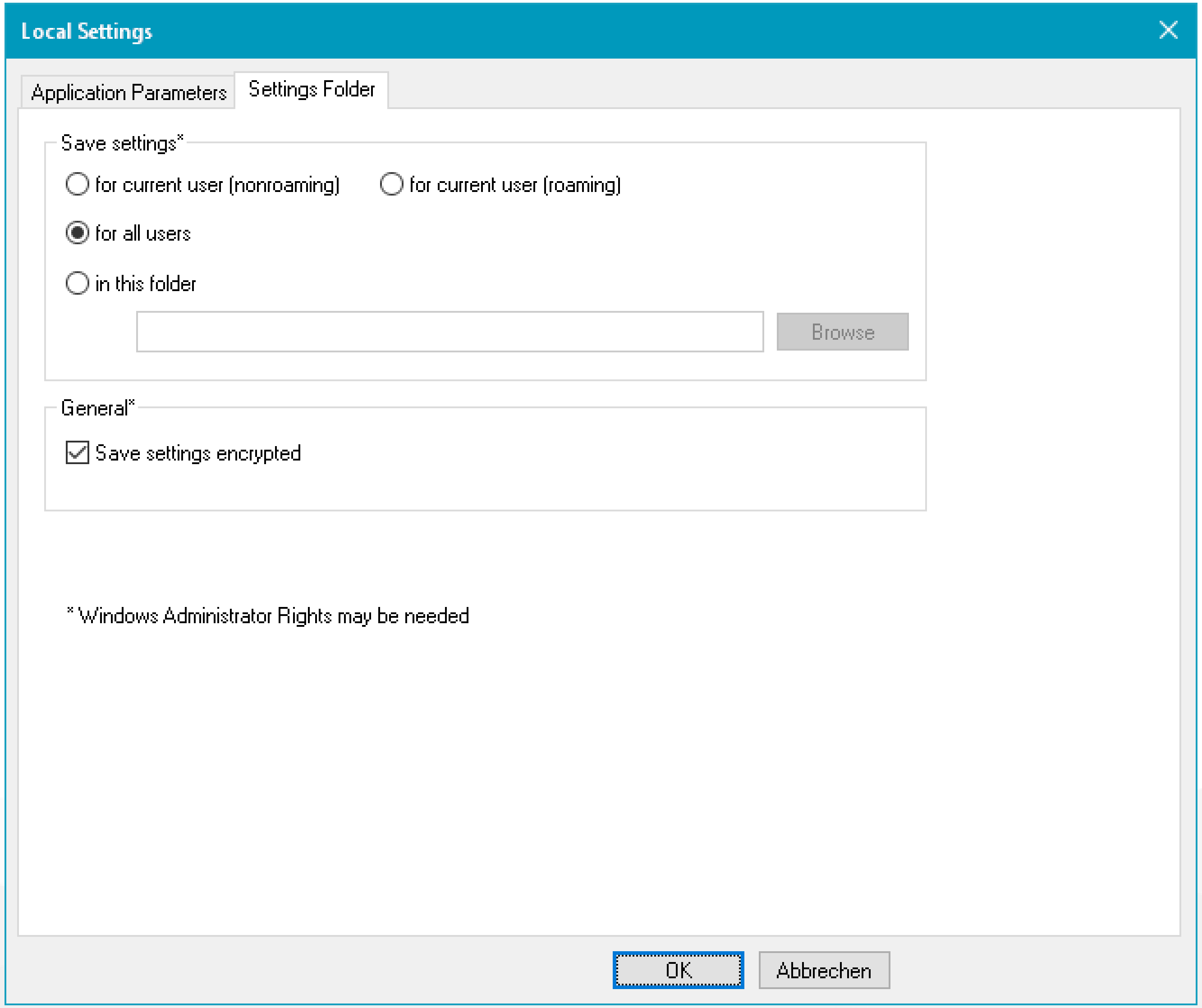


Application Parameters (1)

- Define the appearance of the PC software on the APPLICATION PARAMETERS configuration page.
- MAIN WINDOW PRESENTATION: Define position and size of the main window of the PC software.
  - REMEMBER LAST WINDOW POSITION: The last window position of the main window is stored when closing the app. The next time the app is started, the window is displayed in the same place if this option is enabled. Otherwise, it is displayed in the top left corner of the main screen.
  - SHOW WINDOW ON FIXED POSITION: Define the position of the top left corner of the application window on the screen when the application starts. The top left corner of the main screen has the coordinates X=0, Y=0.
  - SCREEN RESOLUTION: Displays the screen resolution of the monitor the application is displayed on as it is provided by the operating system. This may be different from the resolution defined in the graphic card driver due to the high DPI scaling feature of the operating system.
  - MAIN WINDOW SIZE: Select one of the predefined windows sizes.
    - The window size can also be adjusted by dragging the frame of the main window with the mouse.
- SHOW DETAILED CODEC INFORMATION IN CODEC LINE: Enable this option to show detailed information about the codec used for audio transmission of the codec channel (AC). Otherwise, a simpler status information is shown. More details are displayed when clicking on the simple status information.

Application Parameters (2)



Settings Folder (1)

Settings Folder (2)

- Define the storage location of the local settings on the SETTINGS FOLDER page.
- Local settings include all settings in the LOCAL SETTINGS window as well as the settings under MENU > CONFIGURATION > CONTROL INTERFACE.
- The selected storage location determines which Windows user rights are required to change the local settings.
- Changing these settings may require administrator rights.
- Select a settings location:
  - LOCAL SETTINGS VALID ONLY FOR CURRENT USER: The settings are saved in the user directory of the logged-in Windows account. User rights are sufficient to change the local settings. (%APPDATA%\LOCAL\AVT\MAGIC THipPro LITE)
  - FOR ALL USERS: All users of the PC use the same settings. Windows Administrator rights are required to change the local settings. (%PROGRAMDATA%\AVT\MAGIC THipPro LITE)
  - STORE SETTINGS IN CUSTOM PATH: The settings are saved in an adjustable folder path. The required user rights are determined by the file's properties. The path is saved in the settings.ini file in the installation directory.
- STORE SETTINGS ENCRYPTED: Enable to encrypt the content of the local settings file.

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Configuration

- Operation Settings

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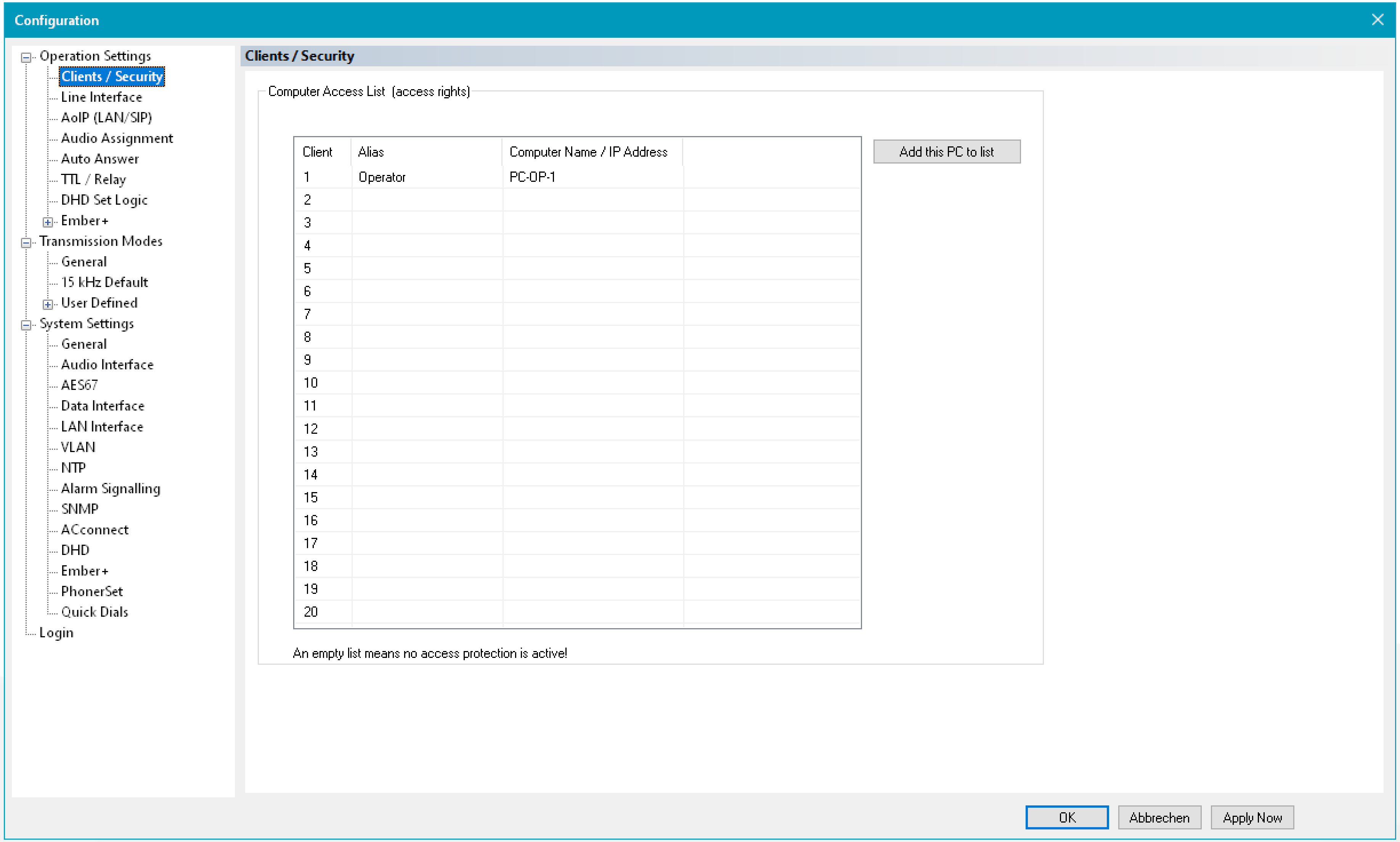

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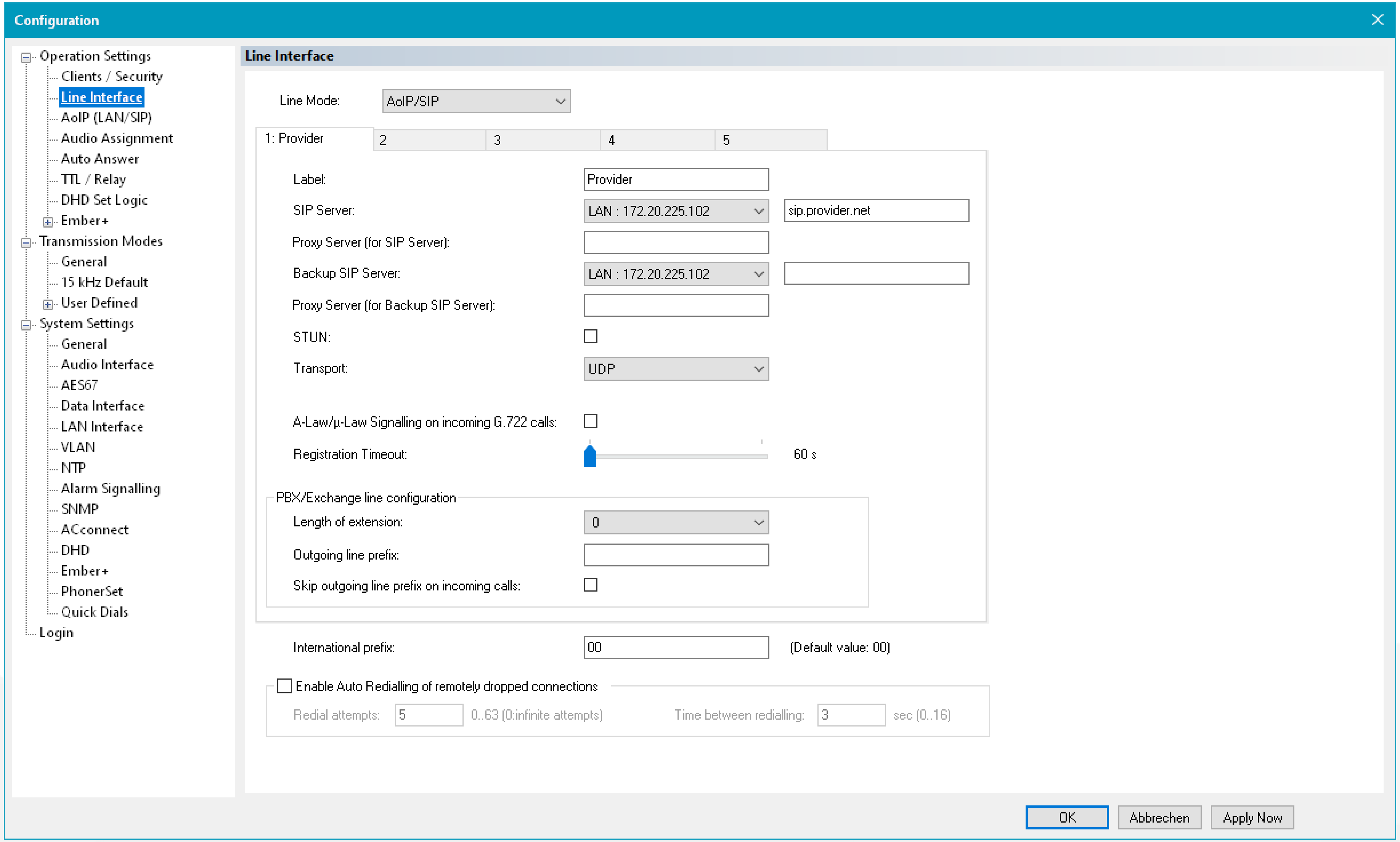
Clients / Security (1)

- When a preset is created, only the settings in the OPERATION SETTINGS branch are saved in the preset. These settings affect the audio transmission and can therefore be flexibly reloaded via presets.
- Access to the device via PC software can be restricted on the CLIENTS / SECURITY settings page.
- COMPUTER ACCESS LIST:
  - If the list is empty, the system can be accessed from any PC.
  - If at least one client is entered, only PCs that are in the list can access the system.
- CLIENT: Internal ID of each PC.
- ALIAS: A freely selectable designation for the PC.
- COMPUTER NAME / IP ADDRESS: This entry is used to check the access rights of a client to the system. Either the Windows computer name or the IP address of the PC can be entered.
- ADD THIS PC TO LIST: Adds the Windows computer name of the PC currently used for configuration to the list. Only applicable if the PC is not yet included in the list.

Clients / Security (2)



Line Interface (1)

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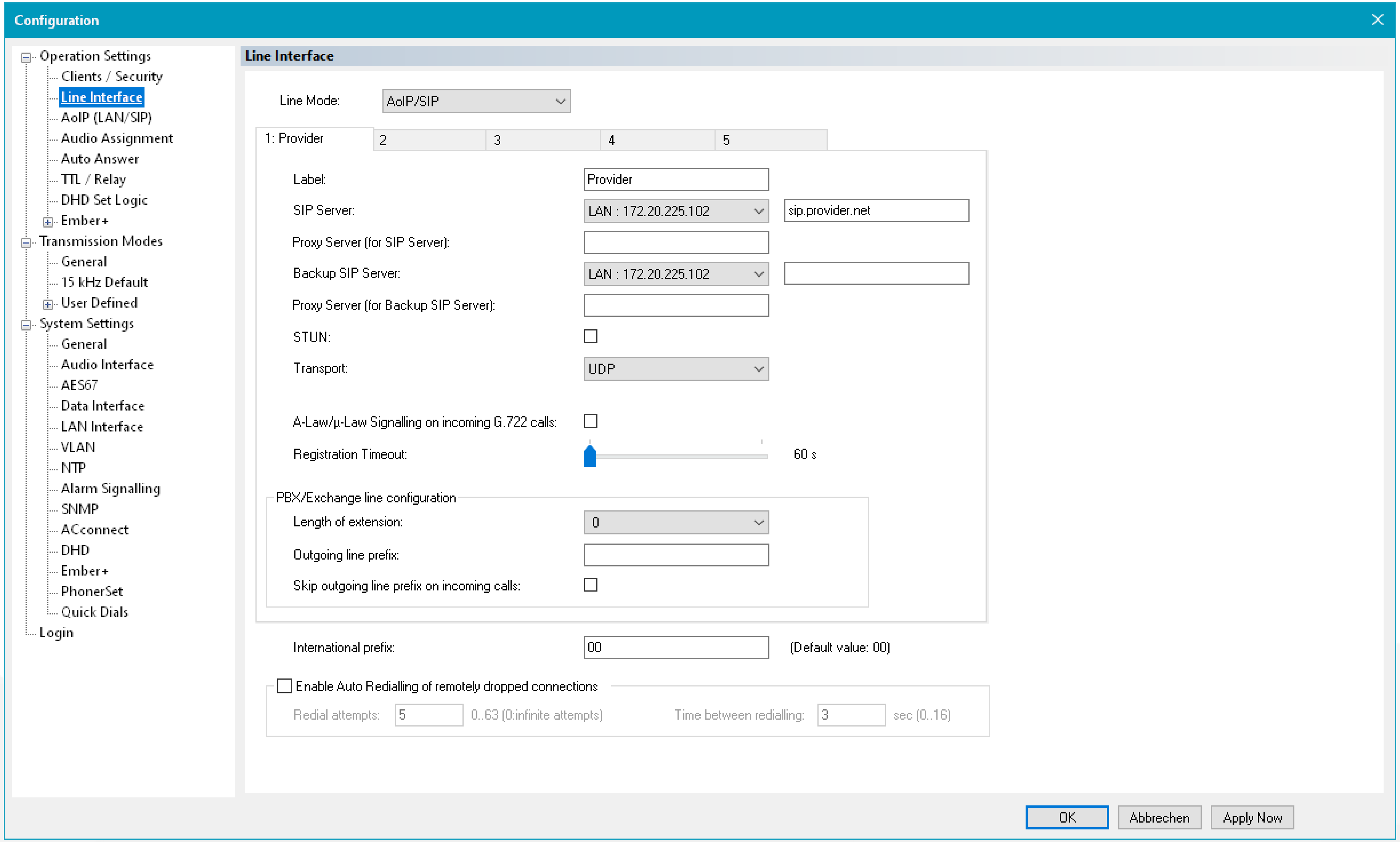
Line Interface (2)

- On the LINE INTERFACE setting page, the operating mode and the connection parameters are configured.
- LINE MODE: Sets the operating mode.
  - AoIP/SIP
    - The unit can register with a SIP server. The connection can be established by calling a phone number when using a SIP server.
    - The unit can register with up to five accounts simultaneously. Incoming calls on any of these SIP accounts can be accepted. An account must be selected when dialling out.
    - By entering the IP address of the receiver, the connection is established via Direct SIP without the need for a SIP server.
    - It is recommended to use an AOIP/SIP provider, or a self-operated SIP server. Telephony providers usually don’t support high quality audio codecs.
    - Using an account with telephony providers gives access to the public telephone network in SD- and HD-Voice quality.
    - The IP address of the receiver must be entered when establishing the connection.

- IP Leased Line Dial Up
  - Bidirectional point-to-point connection
  - IP address of the receiver is entered when connecting
  - Compatible with MAGIC ACip3 and Luci Live
- IP Leased Line Standard
  - Bidirectional point-to-point connection
  - IP address of the receiver is entered in the configuration
  - Auto connect after power up
  - Compatible with MAGIC AC1 XIP and MAGIC ACip3
- IP Leased Line Extended
  - Unidirectional point-to-point connection
  - IP addresses of the receivers are entered in the configuration
  - Auto connect after power up
  - Independent configuration of the audio decoder
  - Simulcast streaming to up to 5 decoders
  - Compatible with MAGIC ACip3
  - Raw stream compatibility mode available



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Line Interface – AoIP/SIP (1)

- In AOIP/SIP mode, the device uses the SIP protocol to negotiate the coding algorithm and establish the connection.
- The device can register with up to five SIP accounts on up to five different SIP servers
- When registered with a SIP server, a connection can be established by dialling a phone number.
- By entering an IP address in the dialling window, the connection is established via Direct SIP with no need for a SIP server.
- It is recommended to use an AOIP provider, or a self operated SIP server. Telephony providers usually don't support high quality audio codecs.
- Using an account with telephony providers gives access to the public telephone network in SD- and HD-Voice quality.
- TABS 1-5: There is a tab for each SIP server. The account credentials must be entered on the **AoIP (LAN / SIP)** configuration page.

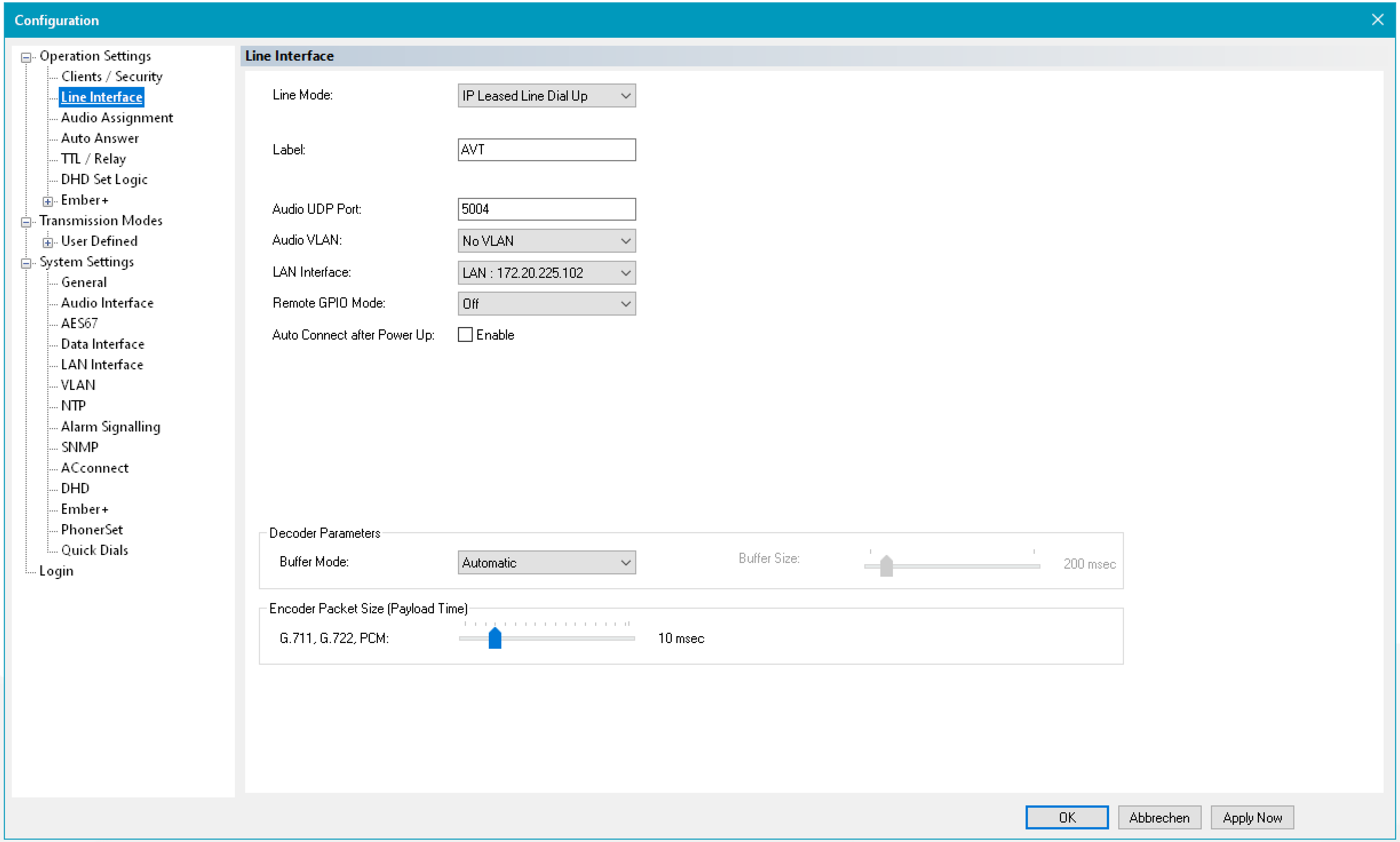
- LABEL: Enter any text. The label is displayed on other configuration pages, the main panel and the dialling window.
- SIP SERVER
  - Select which IP address / VLAN of the device is to be used for the connection to the SIP server.
  - Enter the IP address or the host name of the SIP server.
- PROXY SERVER (FOR SIP SERVER): If the respective account requires the use of a proxy server, enter it here.
- BACKUP SIP SERVER: See SIP SERVER
- PROXY SERVER (FOR BACKUP SIP SERVER): See PROXY SERVER (FOR SIP SERVER)

Line Interface AoIP/SIP (2)

- STUN: If the provider requires the use of a STUN server, activate this option. The STUN server is entered on the configuration page LAN INTERFACE.
- TRANSPORT: Select whether the SIP telegrams are to be transmitted via UDP or TCP. The protocol is specified by the provider.
- A-LAW/µ-LAW SIGNALLING ON INCOMING G.722 CALLS: Activate this setting if audio is missing or dropouts occur when forwarding calls or receiving forwarded calls. This problem may occur if one of the endpoints is not HD Voice (G.722)-capable, but the PBX is not aware of this.
- REGISTRATION TIMEOUT: By default, the device renews the SIP registration every 60 seconds to check if the SIP server is still available. Increase the interval if the SIP server rejects the interval as too short.
- PBX/EXCHANGE LINE CONFIGURATION: The device can distinguish between internal and external calls when operating on a PBX.
    - LENGTH OF EXTENSION: Number of digits of internal call numbers.
    - OUTGOING LINE PREFIX: Digits inserted before the number when dialling an external number.
    - SKIP OUTGOING LINE PREFIX ON INCOMING CALLS: Some PBXs signal the number of the caller including the prefix digits, others do not. Set this option so that the phone numbers of incoming calls are displayed in the PC software without the prefix.
- INTERNATIONAL PREFIX: Prefix digits for dialling international telephone numbers. (Default: "00", do not set to "+"!)

Line Interface AoIP/SIP (3)

- ENABLE AUTO REDIALLING OF REMOTELY DROPPED CONNECTIONS: If a connections is dropped by the other endpoint the device tries to establish the connection again.
    - REDIAL ATTEMPTS: Up to 63 reconnection attempts can be configured. Set to 0 for infinite redialling.
    - TIME BETWEEN REDIALLING: Timespan between a failed attempt and the next attempt in seconds (max. 16 s).



Line Interface – IP Leased Line Dial Up (1)

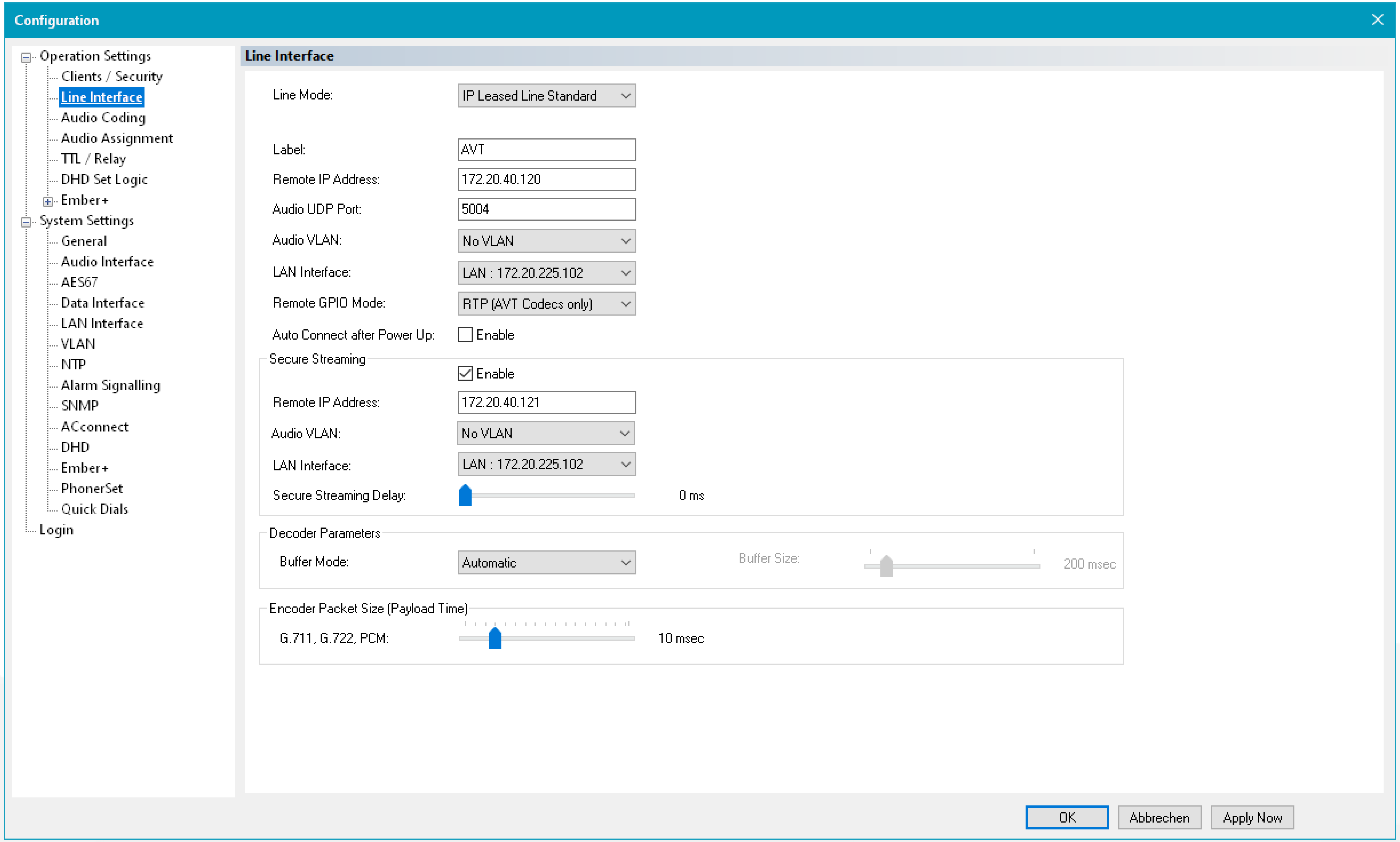
- In the IP LEASED LINE DIAL UP mode, audio is transmitted directly between two devices.
- This mode is compatible with MAGIC ACip3 and Luci Live.
- A connection is established by entering the IP address of the other device in the dialling window.
- The remote device recognises the incoming audio data stream and adopts the source IP address and port for sending its own audio stream.
- LABEL: Enter any text. The label is displayed in the main panel of the PC software above the channel.
- AUDIO UDP PORT: Set the UDP port used for sending and receiving audio data.
- AUDIO VLAN: Select which VLAN is used for the audio transmission. The VLAN must be defined beforehand on the VLAN configuration page.
- LAN INTERFACE: Define which IP address of the device is to be used for the audio transmission.
- REMOTE GPIO MODE: When connected to a MAGIC ACip3 or MAGIC AC1 Go, the states of the GPIO inputs can be transmitted to the remote device. Specify how the data is transmitted.
  - OFF: No transmission.
  - RTP (AVT CODECS ONLY): The states of the inputs are transmitted in the RTP data stream.
  - PAD (AVT CODECS ONLY): The states of the inputs are fed into the coding algorithm as PAD. This is only supported with MPEG.

Line Interface – IP Leased Line Dial Up (2)

- DECODER PARAMETERS: In the receiver, the data packets are buffered to compensate for transmission jitter (= temporal fluctuations in the reception of packets).
  - BUFFER MODE: The receive data buffer offers two modes.
    - AUTOMATIC: The device monitors the jitter of the received data packets and automatically adjusts the size of the receive data buffer to minimise delay while avoiding packet loss.
    - FIXED: The buffer size can be set to a fixed value under BUFFER SIZE. A larger buffer results in a higher audio delay.
  - BUFFER SIZE: Set a receive data buffer size between 5 ms and 2 s.

- ENCODER PACKET SIZE (PAYLOAD TIME): Some encoding algorithms do not mandate a specific size for the IP packets. For the following codecs, the packet size can be set here in milliseconds.
  - G.711, G.722, PCM
  - The device automatically reduces the packet size if the maximum packet size (MTU) of an Ethernet frame is exceeded.

Line Interface – IP Leased Line Dial Up (3)

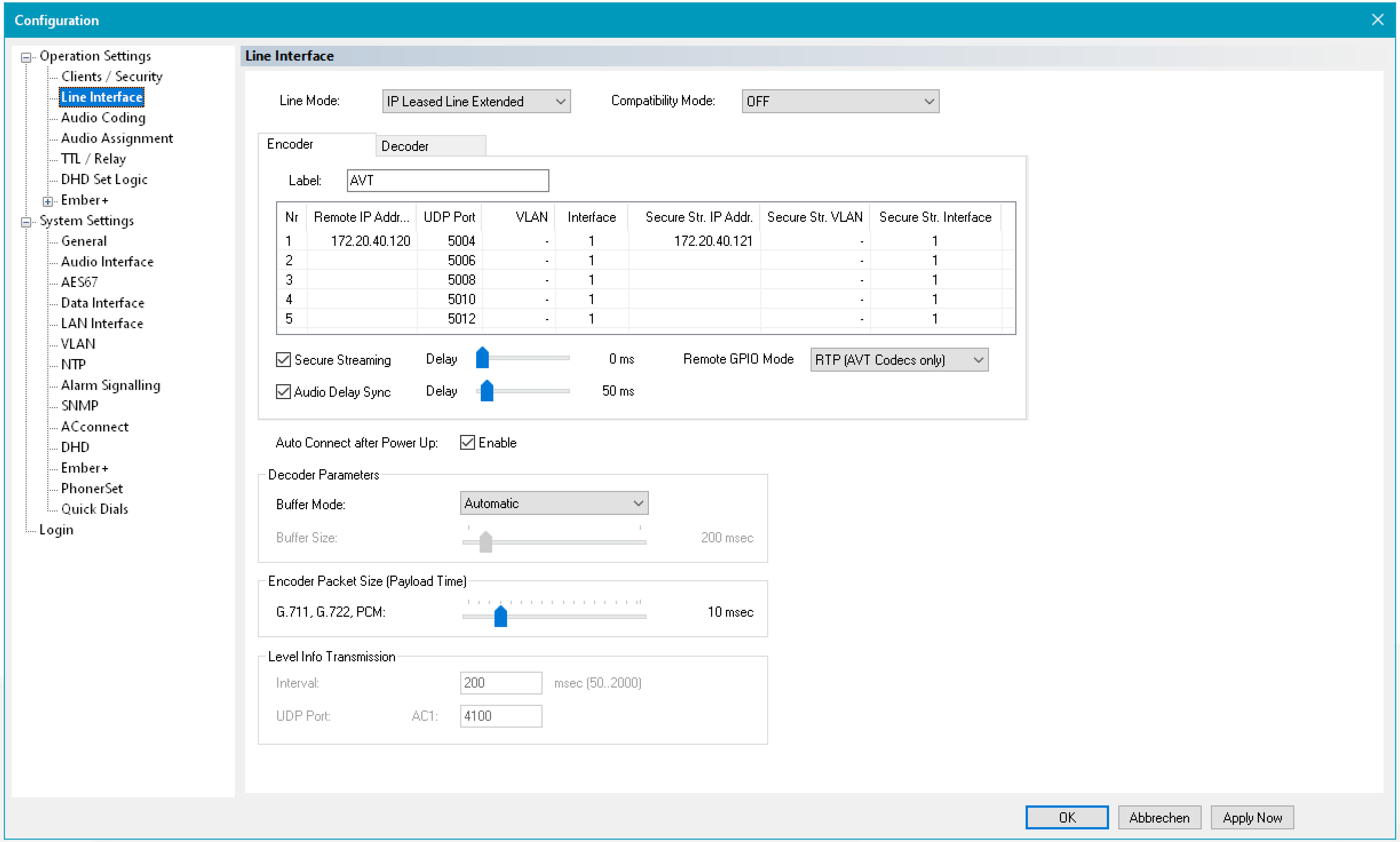


Line Interface – IP Leased Line Standard (1)

- In the IP LEASED LINE DIAL STANDARD mode, audio is transmitted directly between two devices.
- This mode is compatible with MAGIC AC1 XIP and MAGIC ACip3.
- A connection can be established automatically after power-up or using the Connect buttons.
- LABEL: Enter any text. The label is displayed in the main panel of the PC software above the channel.
- REMOTE IP ADDRESS: IP address of the remote device. The encoded data is sent to this address. Incoming data packets are only accepted from this address.
- AUDIO UDP PORT: UDP port used for sending and receiving audio data.
- AUDIO VLAN: Select which VLAN is used for the audio transmission. The VLAN must be defined beforehand on the VLAN configuration page.
- LAN INTERFACE: Define which IP address of the device is to be used for the audio transmission.
- REMOTE GPIO MODE: When connected to a MAGIC AC1 XIP, MAGIC ACip3 or MAGIC AC1 Go, the states of the GPIO inputs can be transmitted to the remote device. Specify how the data is transmitted:
  - OFF: No transmission.
  - RTP (AVT CODECS ONLY): The states of the inputs are transmitted in the RTP data stream.
  - PAD (AVT CODECS ONLY): The states of the inputs are fed into the coding algorithm as PAD. This is only supported with MPEG.
- AUTO CONNECT AFTER POWER UP: Audio transmission starts automatically after the device has been powered up.

- SECURE STREAMING: Sending two streams with identical content. The second stream can be delayed and routed via a different VLAN. Secure Streaming only works between AVT codecs.
    - ENABLE: Enable Secure Streaming in the transmit and receive direction.
    - REMOTE IP ADDRESS: IP address of the remote device for the second stream. The encoded data is sent to this address. Incoming data packets are only accepted from this address.
    - AUDIO VLAN: VLAN for the transmission of the second stream. The VLAN must be defined beforehand on the VLAN configuration page. If the audio transmission is not assigned to a VLAN, select NO VLAN.
    - LAN INTERFACE: Define which IP address of the device is to be used for the audio transmission.
    - SECURE STREAMING DELAY: Time delay for the second audio streams. This increases the overall delay of the transmission.
- DECODER PARAMETERS: In the receiver, the data packets are buffered to compensate for transmission jitter (= temporal fluctuations in the reception of packets).
    - BUFFER MODE: The receive data buffer offers two modes.
        - AUTOMATIC: The device monitors the jitter of the received data packets and automatically adjusts the size of the receive data buffer to minimise delay while avoiding packet loss.
        - FIXED: The buffer size can be set to a fixed value under BUFFER SIZE. A larger buffer results in a higher audio delay.
    - BUFFER SIZE: Set a receive data buffer size between 20 ms and 2 s.
- ENCODER PACKET SIZE (PAYLOAD TIME): Some encoding algorithms do not mandate a specific size for the IP packets. For the following codecs, the packet size can be set here in milliseconds.
    - G.711, G.722, PCM
    - The device automatically reduces the packet size if the maximum packet size (MTU) of an Ethernet frame is exceeded.

Line Interface – IP Leased Line Standard (3)



Line Interface – IP Leased Line Extended (1)

- IP LEASED LINE EXTENDED: Allows sending of up to five streams via unidirectional point-to-point connections. All streams transport identical content (simulcast).
  Reception of one independent stream.
  - Compatible with MAGIC ACip3
- A connection can be established automatically after power-up or using the Connect buttons.
- ENCODER and DECODER are configured on separate tabs.
- COMPATIBILITY MODE: Enable connections to otherwise incompatible endpoints.
  - OFF: Use standard mode
  - AC1 XIP MODNET IP: Compatible with AC1 XIP Modnet devices in IP mode.
  - RAW RTP/MPEG: Special application.
- ENCODER settings
  - LABEL: Enter any text. The label is displayed in the main panel of the PC software above the channel.
  - REMOTE IP ADDRESS: IP address of the remote device. The encoded data is sent to this address.
  - AUDIO UDP PORT: UDP port used for sending audio data.
  - AUDIO VLAN: Select which VLAN is used for the audio transmission. The VLAN must be defined beforehand on the VLAN configuration page.
  - LAN INTERFACE: Define which IP address of the device is to be used for the audio transmission.

Line Interface – IP Leased Line Extended (2)

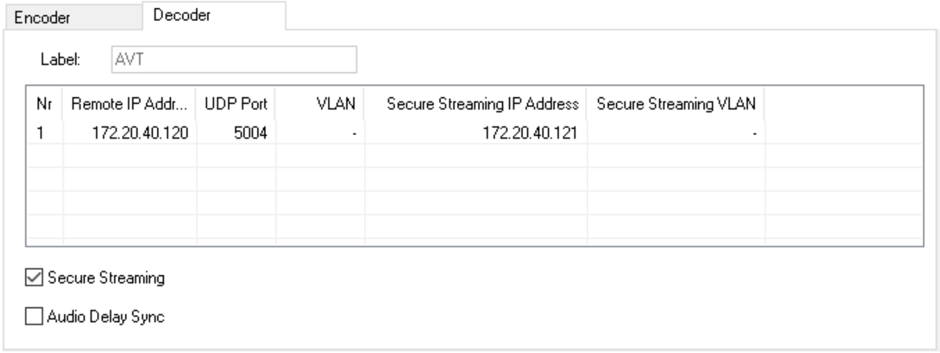
Line Interface – IP Leased Line Extended (3)

- SECURE STREAMING: Sending two streams with identical content. The second stream can be delayed and routed via a different VLAN. Secure Streaming only works between AVT codecs.
- SECURE STR. IP ADD.: IP address of the remote device for the second stream. The encoded data is sent to this address.
- SECURE STR. VLAN: VLAN for the transmission of the second stream. The VLAN must be defined beforehand on the VLAN configuration page. If the audio transmission is not assigned to a VLAN, select NO VLAN.
- SECURE STR. INTERFACE: Define which IP address of the device is to be used for the audio transmission.
- SECURE STREAMING: Enable Secure Streaming for transmission.
  - DELAY: Time delay for the second audio streams. This increases the overall delay of the transmission.
- AUDIO DELAY SYNC: This function makes it possible to output the audio data of this encoder at the decoder locations at synchronously and thus compensate for transmission delays of different lengths of the individual tracks. For this, the internal clocks of the encoders and decoders must be synchronised via NTP. Each transmitted data packet is time stamped, with the current time plus the set audio delay. The decoder waits until its current time matches the time stamp of a received packet and then decodes the content.
  - DELAY: Set this value to the maximum expected transmission delay so that even the most distant decoder receives the audio data stream in time and can play it out synchronously. You can check the fill level of the receive buffers at the decoders in the SYSTEM MONITOR under RX JITTER BUFFER.

Line Interface – IP Leased Line Extended (4)

- REMOTE GPIO MODE: When connected to a MAGIC AC1 XIP, MAGIC ACip3 or MAGIC AC1 Go, the states of the GPIO inputs can be transmitted to the remote device. Specify how the data is transmitted:
  - OFF: No transmission.
  - RTP (AVT CODECS ONLY): The states of the inputs are transmitted in the RTP data stream.
  - PAD (AVT CODECS ONLY): The states of the inputs are fed into the coding algorithm as PAD. This is only supported with MPEG.

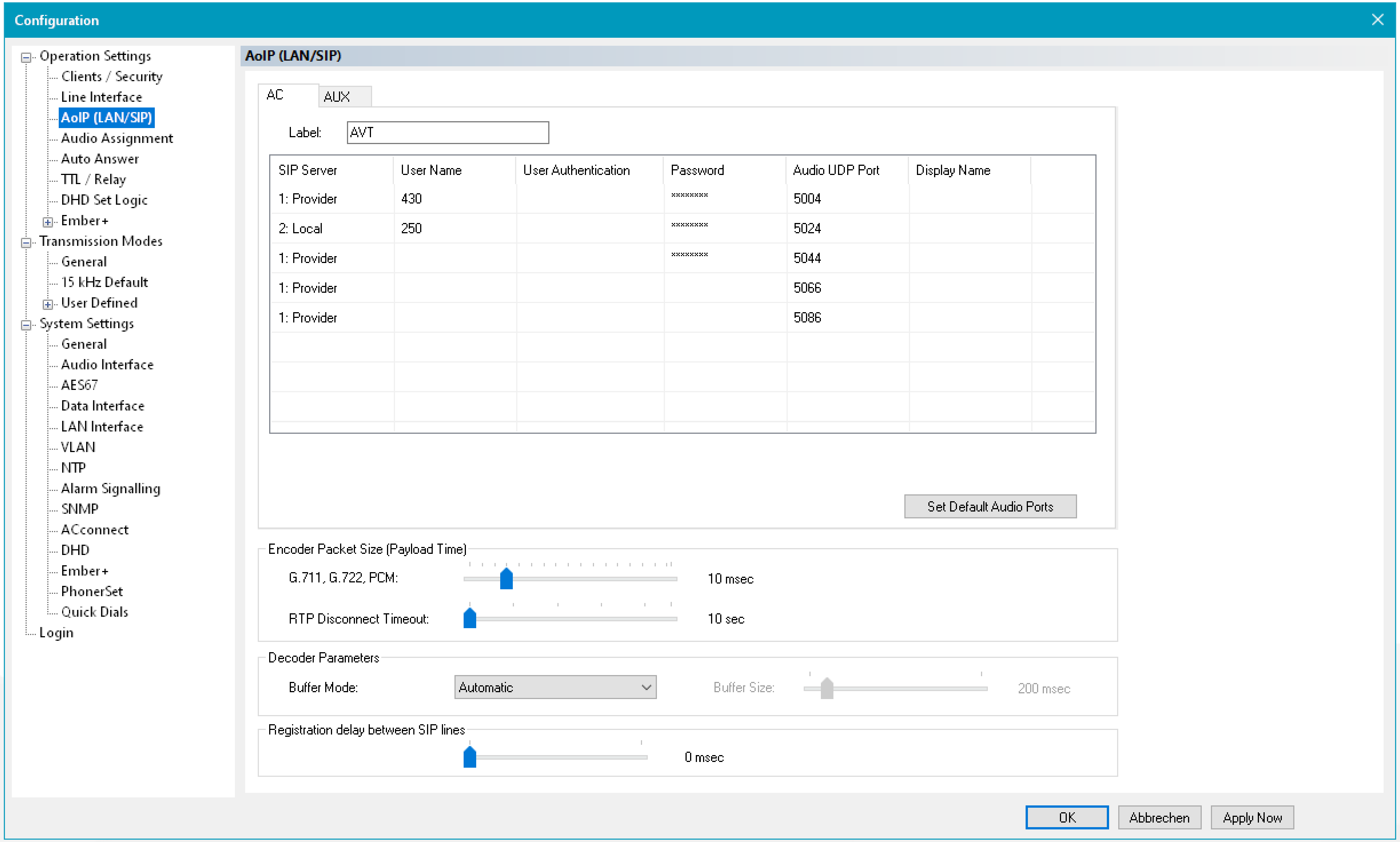
DECODER SETTINGS



- LABEL: Must be configured on the ENCODER tab.
- REMOTE IP ADDRESS: IP address of the remote device. Incoming data packets are only accepted from this address.
- UDP PORT: UDP port used for receiving audio data.
- VLAN: Select which VLAN is used for the audio reception. The VLAN must be defined beforehand on the VLAN configuration page.
- SECURE STREAMING: Receiving two streams with identical content. The second stream can be delayed and routed via a different VLAN. Secure Streaming only works between AVT codecs.
- SECURE STREAMING IP ADDRESS: IP address of the remote device for the second stream. Incoming data packets are only accepted from this address.
- SECURE STREAMING VLAN: VLAN for the reception of the second stream. The VLAN must be defined beforehand on the VLAN configuration page. If the audio transmission is not assigned to a VLAN, select NO VLAN.
- SECURE STREAMING: Enable Secure Streaming for reception.
- AUDIO DELAY SYNC: Enable for reception. See Encoder settings for further information.

- AUTO CONNECT AFTER POWER UP: Audio transmission starts automatically after the device has been powered up.
- DECODER PARAMETERS: In the receiver, the data packets are buffered to compensate for transmission jitter (= temporal fluctuations in the reception of packets).
  - BUFFER MODE: The receive data buffer offers two modes.
    - AUTOMATIC: The device monitors the jitter of the received data packets and automatically adjusts the size of the receive data buffer to minimise delay while avoiding packet loss.
    - FIXED: The buffer size can be set to a fixed value under BUFFER SIZE. A larger buffer results in a higher audio delay.
  - BUFFER SIZE: Set a receive data buffer size between 20 ms and 2 s.
- ENCODER PACKET SIZE (PAYLOAD TIME): Some encoding algorithms do not mandate a specific size for the IP packets. For the following codecs, the packet size can be set here in milliseconds.
  - G.711, G.722, PCM
  - The device automatically reduces the packet size if the maximum packet size (MTU) of an Ethernet frame is exceeded.
- LEVEL INFO TRANSMISSION: In COMPATIBILITY MODE: RAW RTP/MPEG, the audio level at the encoder input can be transmitted to the remote device.
  - INTERVAL: Set the time interval at which the level information is sent.
  - UDP PORT: Set the UDP port for sending the level information.

Line Interface – IP Leased Line Extended (5)



Audio
Video
Technologies

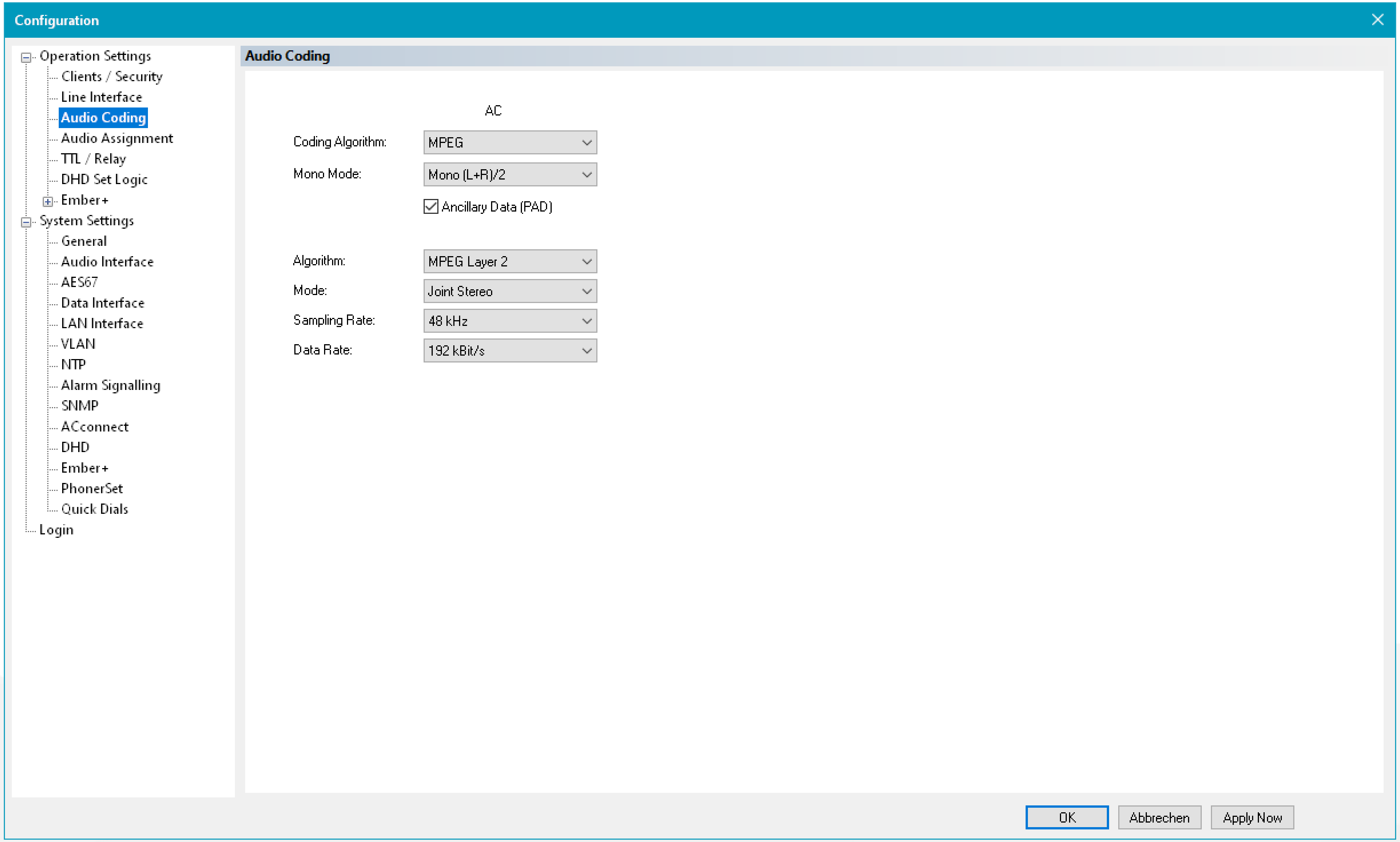
AoIP (LAN/SIP) (1)

AoIP (LAN/SIP) (2)

- On the AOIP (LAN/SIP) page, the SIP accounts are set if the operating mode is set to AoIP/SIP.
- The available channels are shown on tabs.
  - AC: Codec channel
  - AUX: Command channel
- LABEL: Enter any text. The label is displayed in the main window of the PC software above the channel.
- ACCOUNT TABLE: Up to five AoIP/SIP accounts can be assigned to each channel. Incoming calls are signalled from each account. If there is a connection on one account, the other accounts are reply with busy. For outgoing calls, an account must be selected when dialling.
- SIP SERVER: Select one of the predefined SIP servers. Configure the SIP servers on the LINE INTERFACE page.
- USER NAME: Enter the Username of the SIP account as specified by the provider.
- USER AUTHENTICATION: Enter the User Authentication of the SIP account as specified by the provider.
- PASSWORD: Enter the password for the SIP account as specified by the provider.
- AUDIO UDP PORT: Define the UDP port which the device uses to send the audio data.
- DISPLAY NAME: Enter any text. This text is displayed at the remote device. (Might be overwritten by the PBX or the provider.)
- SET DEFAULT AUDIO PORTS: Resets the local audio ports to the default values.

- ENCODER PACKET SIZE (PAYLOAD TIME): Some encoding algorithms do not mandate a specific size for the IP packets. For the following codecs, the packet size can be set here in milliseconds.
  - G.711, G.722, PCM
  - The device automatically reduces the packet size if the maximum packet size (MTU) of an Ethernet frame is exceeded.
- RTP DISCONNECT TIMEOUT: The time after which the device automatically terminates a connection if no audio data packets are received.
- DECODER PARAMETERS: In the receiver, the data packets are buffered to compensate for transmission jitter (= temporal fluctuations in the reception of packets).
  - BUFFER MODE: The receive data buffer offers two modes.
    - AUTOMATIC: The device monitors the jitter of the received data packets and automatically adjusts the size of the receive data buffer to minimise delay while avoiding packet loss.
    - FIXED: The buffer size can be set to a fixed value under BUFFER SIZE. A larger buffer results in a higher audio delay.
  - BUFFER SIZE: Set a receive data buffer size between 20 ms and 2 s.
- REGISTRATION DELAY BETWEEN SIP LINES: During start-up, the device simultaneously sends SIP registration telegrams for each AoIP/SIP account to the SIP server(s). If this is overwhelming a SIP server this setting can be used to introduce a delay between the registration attempts of the AoIP/SIP accounts.

AoIP (LAN/SIP) (3)



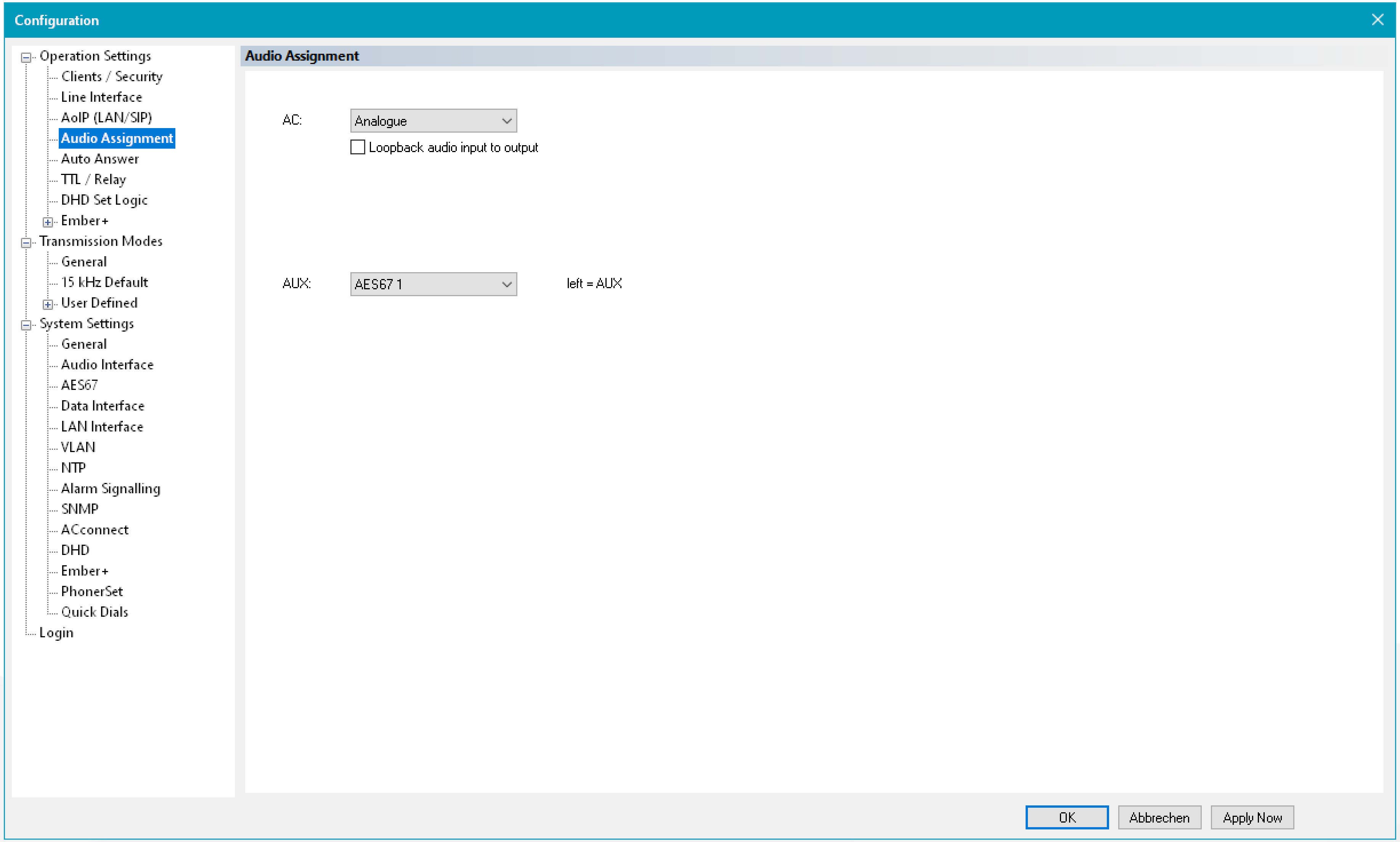
Audio Coding (1)

Audio Coding (2)

- On the AUDIO CODING page, the coding algorithms are set when the operating mode is set to IP LEASED LINE STANDARD / EXTENDED.
- CODING ALGORITHM: Select one of the installed codecs:
  - G.722, PCM, OPUS, FLAC, MPEG (some optional).
- MONO MODE: Select how the audio input signal is used when the algorithm is set to mono:
  - MONO LEFT: Only the left channel of the input signal is used.
  - MONO RIGHT: Only the right channel of the input signal is used.
  - MONO (L+R)/2: Left and right channels of the input signal are mixed.
- ANCILLARY DATA (PAD): Some codecs support transmission of data along with the encoded audio within the stream. If enabled, the device routes data provided via the DATA Interface (RS232) to the encoder. The data rate used for PAD reduces the data rate of the audio signal.
- ALGORITHM: If the MPEG codec is selected, you can choose between the available variants here (some optional):
  - L2: Layer 2 (MP2)
  - L3: Layer 3 (MP3)
  - AAC LD: Advanced Audio Coding Low Delay
  - AAC ELD: Advanced Audio Coding Enhanced Low Delay
  - AAC LC: Advanced Audio Coding Low Complexity
  - HE-AAC v1: High Efficiency Advanced Audio Coding with Spectral Band Replication (SBR)
  - HE-AAC v2: High Efficiency Advanced Audio Coding with Spectral Band Replication (SBR) and Parametric Stereo (PS)

- MODE: Select how the input signal is to be encoded:
  - STEREO: The codec treats the signal as left and right channel of a stereo signal.
  - JOINT STEREO: The codec may convert right and left channel to a mid/side signal if the available bandwidth is limited.
  - DUAL CHANNEL: The codec treats left and right channels as separate signals.
  - MONO: The codec is fed a single-channel signal that is formed according to the MONO MODE setting.
- SAMPLING RATE: Select a sampling rate of 48 kHz or 32 kHz. A higher sampling rate results in a higher audio bandwidth. However, not every codec can use this at all data rates. (Not available for all codecs).
- DATA RATE: A higher data rate provides better audio quality but requires higher transmission bandwidth. (Not available for all codecs.)
- BIT RESOLUTION: A higher bit depth allows for a higher dynamic range but requires a higher transmission bandwidth. (Not available for all codecs).
- BITRATE: If the data rate cannot be set directly but results from the available parameters, it is shown here for information.

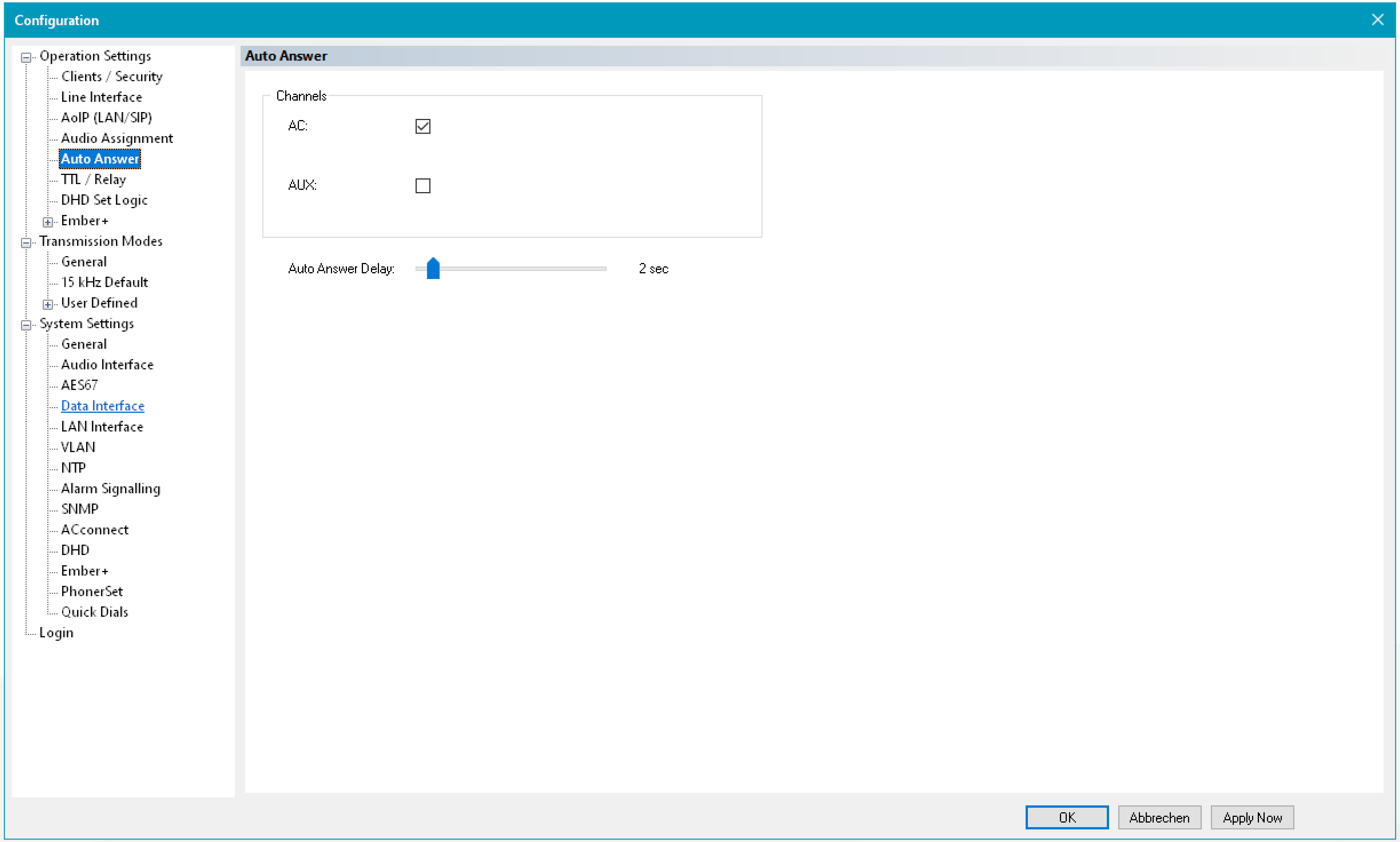
Audio Coding (3)



Audio Assignment (1)

Audio Assignment (2)

- The audio interfaces of the device are assigned to the codec channel (AC) and the AUX channel on the AUDIO ASSIGNMENT page.
- AC: Select one of the available stereo audio interfaces.
- LOOPBACK AUDIO INPUT TO OUTPUT: Activate this option to loop the unencoded audio input signal internally back to the audio output.
- AUX: Select one of the available stereo audio interfaces for the AUX channel. Only the left channel of that audio interface is used.
- The following audio interfaces are available:
  - OFF: The channel is not used and is not displayed on the user interface.
  - AES: AES/EBU stereo interface XLR
  - ANALOGUE: Analogue stereo interface XLR
  - AES67 1: Audio over IP stereo interface 1 (optional)
  - AES67 2: Audio over IP stereo interface 2 (optional)
- It is not possible to use the analogue audio interface and the AES/EBU interface at the same time, as they share the same connectors on the unit.

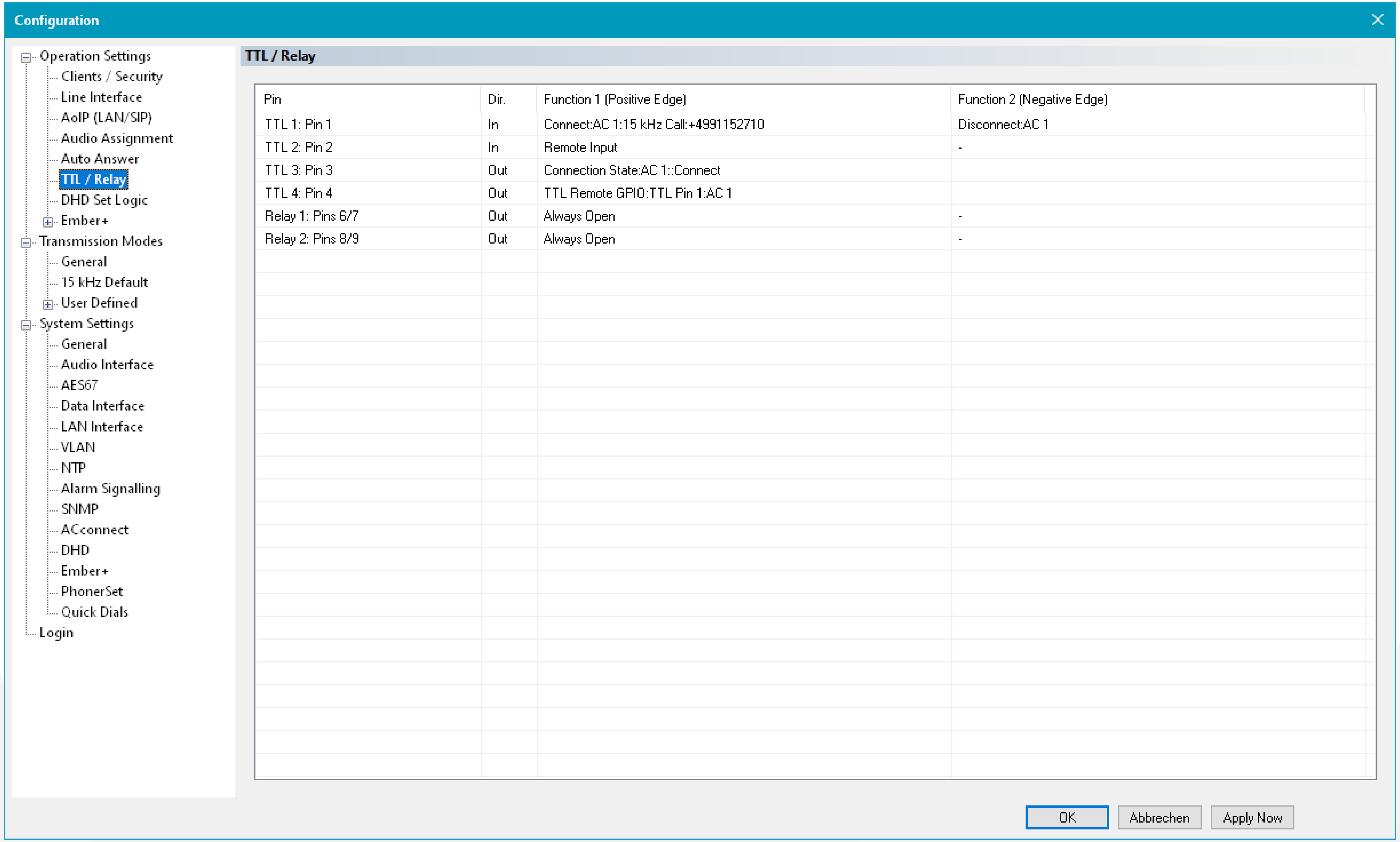


Audio Video Technologies

Auto Answer (1)

Auto Answer (2)

- The unit can automatically accept calls which is configured on the AUTO ANSWER page.
- Available in AoIP/SIP and IP Leased Line Dial Up mode
- Auto Answer can be switched on separately for each channel.
- AUTO ANSWER DELAY: Define the time period after which the call is automatically accepted. During this time, the incoming call is signalled normally on the user interface.



Audio Video Technologies

GPIO (1)

- GPIOs (General Purpose Input Output) offer a way of exchanging information with other systems.
- The functions of each GPIO can be configured on the TTL / RELAY, DHD SET LOGIC and EMBER+ configuration pages.
- The following GPIO variants are available:
- TTL: 4 hardware pins via the TTL/Relay connector
  - Each TTL can be operated as input or output.
  - Each TTL pin is internally connected to a pull-up resistor which means the default state is "high".
  - Connect the respective pin to GND (Pin 5 of the TTL/Relay connector) to generate a "low" input signal.
- RELAY: 6 relays that can only be used as outputs.
- DHD SET LOGIC: Uses an IP-based protocol to transmit control signals and status messages via Ethernet. There are 64 GPIOs available, each of which can be used as input or output. Must be enabled and configured under **System Settings > DHD** first.
- EMBER+: Uses an IP-based protocol to transmit control signals and status messages via Ethernet. An end device can act as a provider or consumer in a connection.
  - INPUT: 64 inputs are available. (as Ember+ provider)
  - OUTPUT: 64 outputs are available. (as Ember+ Provider)
  - CONSUMER FUNCTIONS: Allows the transmission of text information via the Ember+ protocol. 10 functions are available per consumer. (as Ember+ Consumer)

GPIO (2)

MAGIC AC1 Go

Configuration

- Transmission Modes

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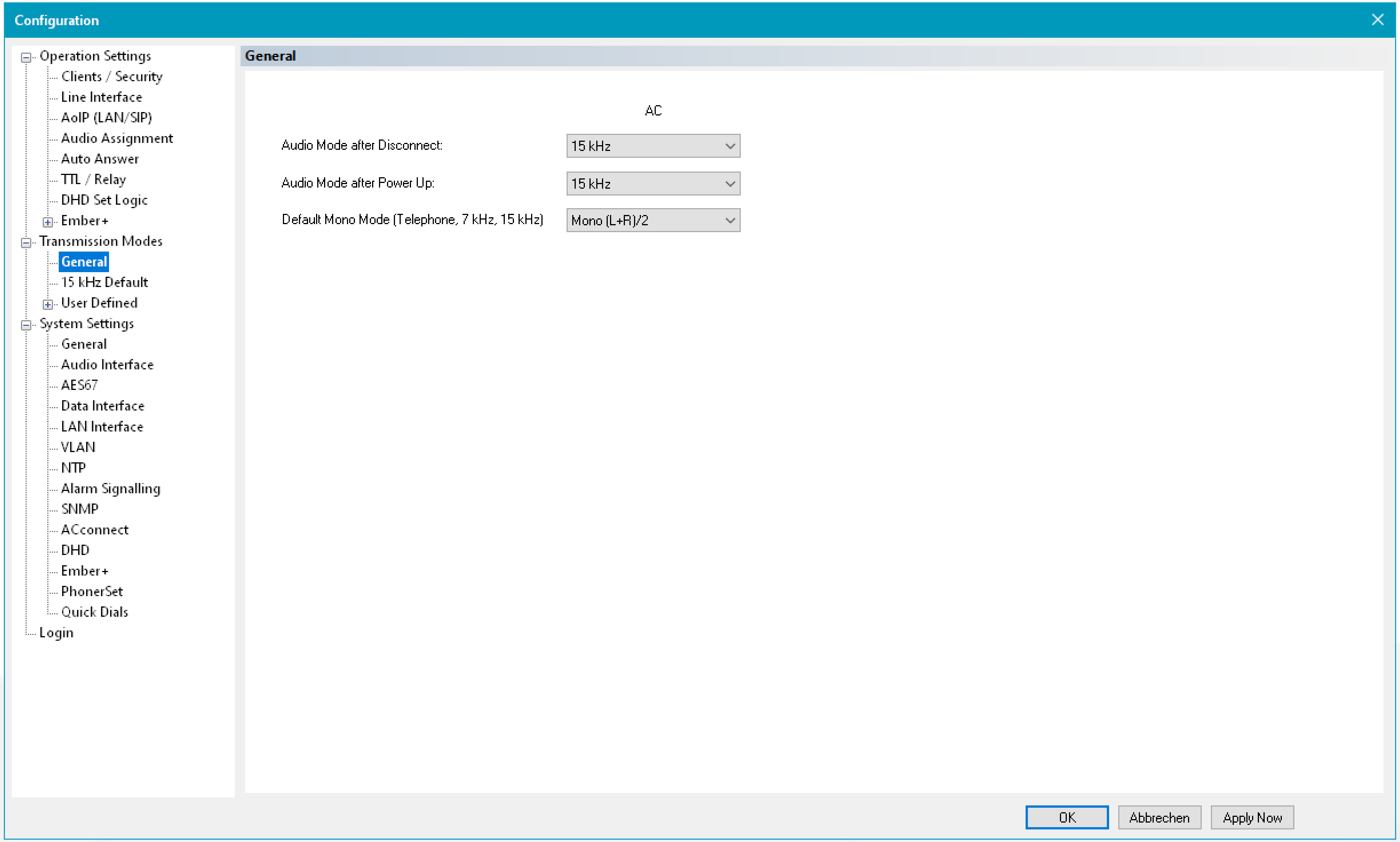


Transmission Modes - General (1)

- In the AoIP/SIP and IP Leased Line Dial Up operating mode, the codec used must be selected with the desired parameters each time a connection is established.
- To simplify the selection when establishing a connection, the desired codecs must be preconfigured.
- For that, codec, algorithm and other parameters are configured and stored as a TRANSMISSION MODE.

- There are three predefined transmission modes that are always available when dialling out :
  - **TELEPHONE**: G.711, 64 kbit/s
  - **7 KHZ**: G.722, 64 kbit/s
  - **15 KHZ Default**: configurable under **Transmission Modes > 15 kHz Default**.
- Custom transmission modes can be configured and stored under **Transmission Modes > User Defined**.

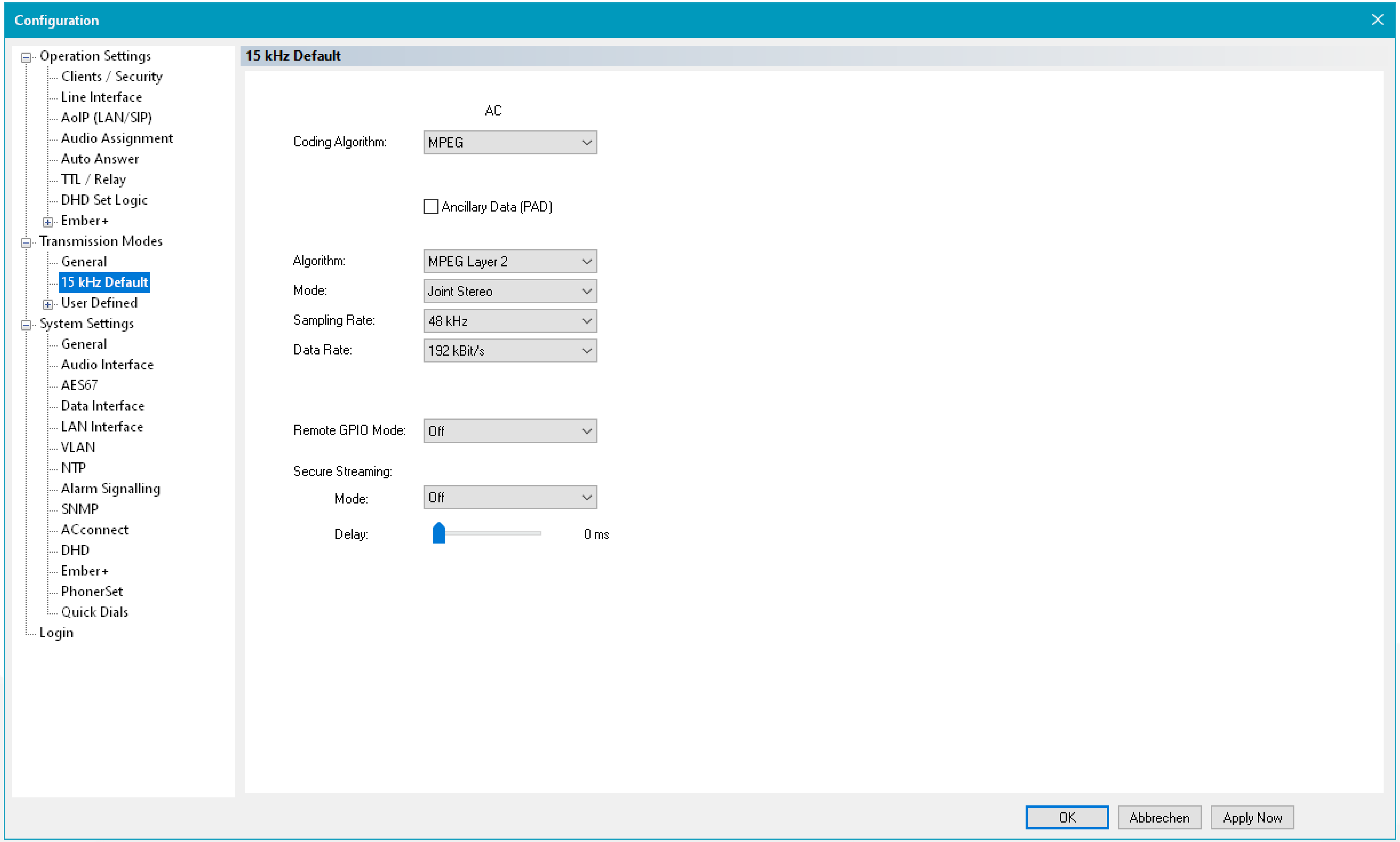
AVT Audio Video Technologies

Transmission Modes - General (2)

Transmission Modes - General (3)

- The TRANSMISSION MODES - GENERAL page provides basic settings.
- AUDIO MODE AFTER DISCONNECT: Select which transmission mode is set when a connection is terminated. This mode is preset the next time a call is set up. The mode can then be changed.
- AUDIO MODE AFTER DISCONNECT: Select which transmission mode is set when a connection is terminated. This mode is preset the next time a call is set up. The mode can then be changed.
- AUDIO MODE AFTER POWER UP: Select which transmission mode is set when the device is started. This mode is preset the first time a call is set up. The mode can then be changed.
- DEFAULT MONO MODE (TELEPHONE, 7 KHZ, 15 KHZ): Select how the audio input signal is used when a standard transmission mode uses a mono signal:
  - MONO LEFT: Only the left channel of the input signal is used.
  - MONO RIGHT: Only the right channel of the input signal is used.
  - MONO (L+R)/2: Left and right channel of the input signal are mixed.

Audio Video Technologies



Transmission Modes – 15 kHz Default (1)

- On the TRANSMISSION MODE - 15 KHZ DEFAULT page, the coding algorithms for the predefined 15 kHz mode are set.
- CODING ALGORITHM: The following codecs are available:
  - G.722
  - MPEG
  - PCM
  - OPUS
  - FLAC
- ANCILLARY DATA (PAD): Some codecs support transmission of data along with the encoded audio within the stream. If enabled, the device routes data provided via the DATA Interface (RS232) to the encoder. The data rate used for PAD reduces the data rate of the audio signal.
- ALGORITHM: Select an algorithm if the MPEG codec is selected above. (Availability depends on the activated software options.)
  - L2: Layer 2 (MP2)
  - L3: Layer 3 (MP3)
  - AAC LD: Advanced Audio Coding Low Delay
  - AAC ELD: Advanced Audio Coding Enhanced Low Delay
  - AAC LC: Advanced Audio Coding Low Complexity
  - HE-AAC v1: High Efficiency Advanced Audio Coding with Spectral Band Replication (SBR)
  - HE-AAC v2: High Efficiency Advanced Audio Coding with Spectral Band Replication (SBR) and Parametric Stereo (PS)

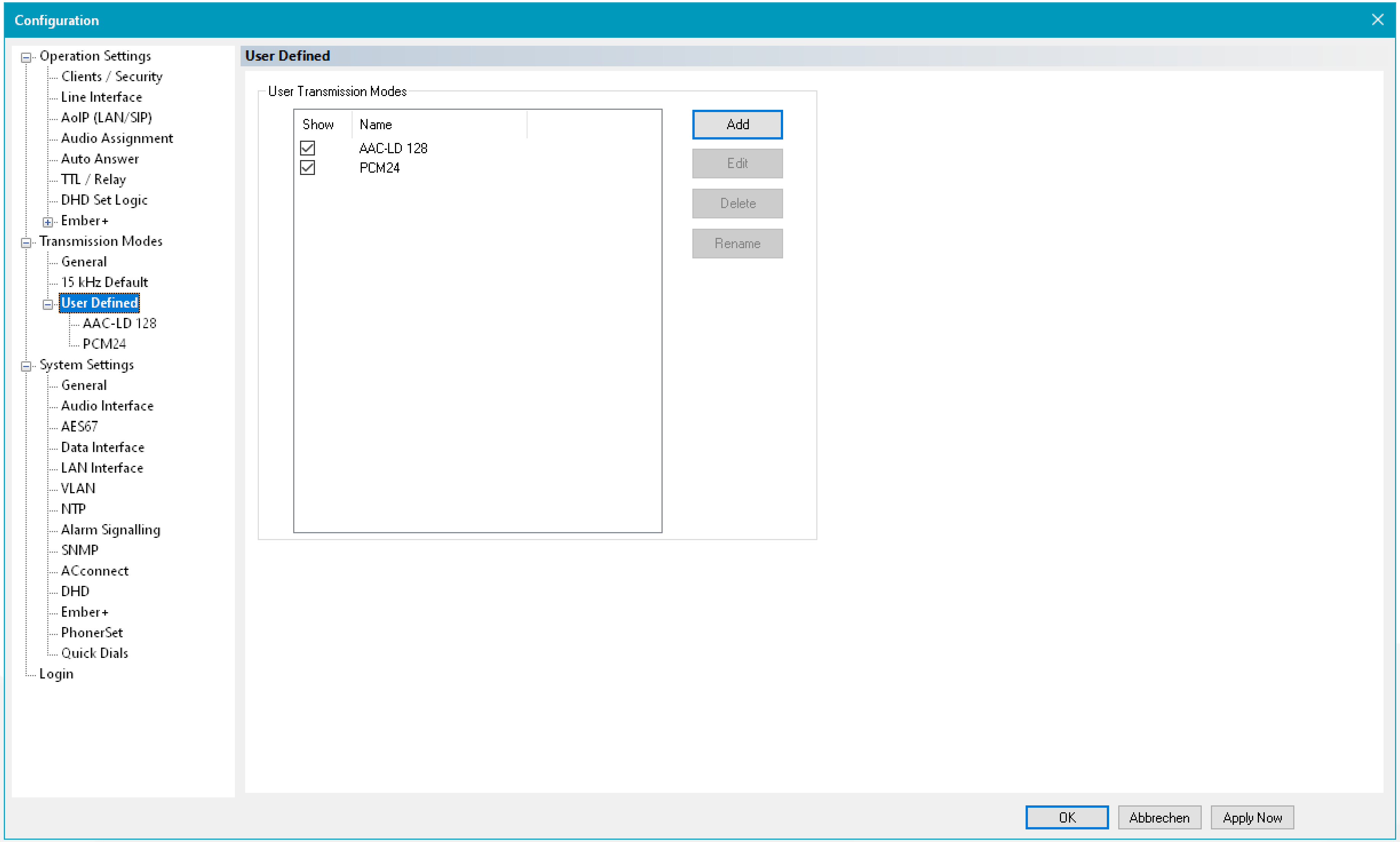
Transmission Modes – 15 kHz Default (2)

- MODE: Select how the input signal is to be encoded:
  - STEREO: The codec treats the signal as left and right channel of a stereo signal.
  - JOINT STEREO: The codec may convert right and left channel to a mid/side signal if the available bandwidth is limited.
  - DUAL CHANNEL: The codec treats left and right channels as separate signals.
  - MONO: The codec is fed a single-channel signal that is formed according to the MONO MODE setting.
- SAMPLING RATE: Select a sampling rate of 48 kHz or 32 kHz. A higher sampling rate results in a higher audio bandwidth. However, not every codec can use this at all data rates. (Not available for all codecs).
- DATA RATE: A higher data rate provides better audio quality but requires higher transmission bandwidth. (Not available for all codecs.)
- BIT RESOLUTION: A higher bit depth allows for a higher dynamic range but requires a higher transmission bandwidth. (Not available for all codecs).
- BITRATE: If the data rate cannot be set directly but results from the available parameters, it is shown here for information.
- REMOTE GPIO MODE: When connected to a MAGIC AC1 XIP, MAGIC ACip3 or MAGIC AC1 Go, the states of the GPIO inputs can be transmitted to the remote device. Specify how the data is transmitted:
  - OFF: No transmission.
  - RTP (AVT CODECS ONLY): The states of the inputs are transmitted in the RTP data stream.
  - PAD (AVT CODECS ONLY): The states of the inputs are fed into the coding algorithm as PAD. This is only supported with MPEG.

Transmission Modes – 15 kHz Default (3)

- SECURE STREAMING: Sending two streams with identical content. Secure Streaming only works between AVT codecs.
  - MODE:
    - ON: The unit sends two streams and expects to receive two streams. If no second stream is received, an alarm is raised.
    - OFF: The unit does not send or expect to receive a second stream.
    - AUTO: The unit sends a second stream when the connection is established. If a second stream is received within in the first 10 seconds into the connection it will continue to send the second stream. Otherwise sending of the second stream is stopped.
  - DELAY: Time delay for the second audio stream. This increases the overall delay of the transmission.

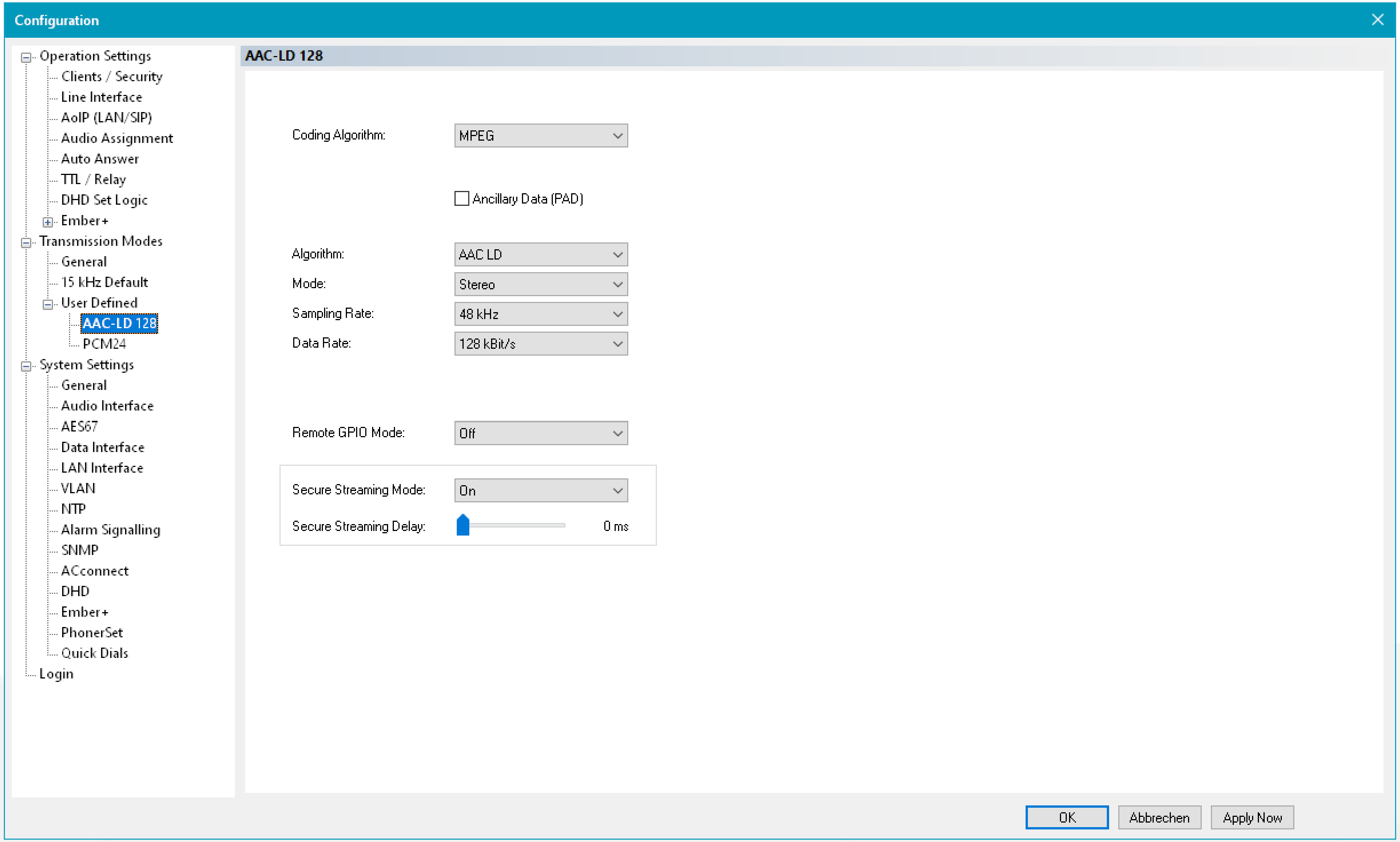
Transmission Modes – 15 kHz Default (4)



Transmission Modes – User Defined (1)

- On the TRANSMISSION MODE - USER DEFINED page, custom transmission modes can be defined.
- USER TRANSMISSION MODES: The user-defined transmission modes are listed here.
  - SHOW: Enabled transmission modes are available for selection when establishing a connection.
  - NAME: Name of the transmission mode.
- ADD: Adds a new transmission mode.
- EDIT: Opens the selected transmission mode for editing. A transmission mode can also be selected for editing in the tree on the left.
- DELETE: Deletes the selected transmission mode.
- RENAME: Opens a window for editing the name of the selected transmission mode.

Transmission Modes – User Defined (2)



User Defined Transmission Mode (1)

User Defined Transmission Mode (2)

- On each USER DEFINED TRANSMISSION MODE page, the parameters of a transmission mode can be set.
- CODING ALGORITHM: The following codecs are available:
  - G.722
  - MPEG
  - PCM
  - OPUS
  - FLAC
- ANCILLARY DATA (PAD): Some codecs support transmission of data along with the encoded audio within the stream. If enabled, the device routes data provided via the DATA Interface (RS232) to the encoder. The data rate used for PAD reduces the data rate of the audio signal.
- ALGORITHM: Select an algorithm if the MPEG codec is selected above. (Availability depends on the activated software options.)
  - L2: Layer 2 (MP2)
  - L3: Layer 3 (MP3)
  - AAC LD: Advanced Audio Coding Low Delay
  - AAC ELD: Advanced Audio Coding Enhanced Low Delay
  - AAC LC: Advanced Audio Coding Low Complexity
  - HE-AAC v1: High Efficiency Advanced Audio Coding with Spectral Band Replication (SBR)
  - HE-AAC v2: High Efficiency Advanced Audio Coding with Spectral Band Replication (SBR) and Parametric Stereo (PS)

User Defined Transmission Mode (3)

- MODE: Select how the input signal is to be encoded:
  - STEREO: The codec treats the signal as left and right channel of a stereo signal.
  - JOINT STEREO: The codec may convert right and left channel to a mid/side signal if the available bandwidth is limited.
  - DUAL CHANNEL: The codec treats left and right channels as separate signals.
  - MONO: The codec is fed a single-channel signal that is formed according to the MONO MODE setting.
- SAMPLING RATE: Select a sampling rate of 48 kHz or 32 kHz. A higher sampling rate results in a higher audio bandwidth. However, not every codec can use this at all data rates. (Not available for all codecs).
- DATA RATE: A higher data rate provides better audio quality but requires higher transmission bandwidth. (Not available for all codecs.)
- BIT RESOLUTION: A higher bit depth allows for a higher dynamic range but requires a higher transmission bandwidth. (Not available for all codecs).
- BITRATE: If the data rate cannot be set directly but results from the available parameters, it is shown here for information.
- REMOTE GPIO MODE: When connected to a MAGIC AC1 XIP, MAGIC ACip3 or MAGIC AC1 Go, the states of the GPIO inputs can be transmitted to the remote device. Specify how the data is transmitted:
  - OFF: No transmission.
  - RTP (AVT CODECS ONLY): The states of the inputs are transmitted in the RTP data stream.
  - PAD (AVT CODECS ONLY): The states of the inputs are fed into the coding algorithm as PAD. This is only supported with MPEG.

- SECURE STREAMING: Sending two streams with identical content. Secure Streaming only works between AVT codecs.
  - MODE:
    - ON: The unit sends two streams and expects to receive two streams. If no second stream is received, an alarm is raised.
    - OFF: The unit does not send two streams nor expects to receive a second stream.
    - AUTO: The unit sends a second stream when the connection is established. If a second stream is received within in the first 10 seconds into the connection it will continue to send the second stream. Otherwise sending of the second stream is stopped.
  - DELAY: Time delay for the second audio stream. This increases the overall delay of the transmission.

Audio Video Technologies

User Defined Transmission Mode (4)

MAGIC AC1 Go

Configuration

- System Settings

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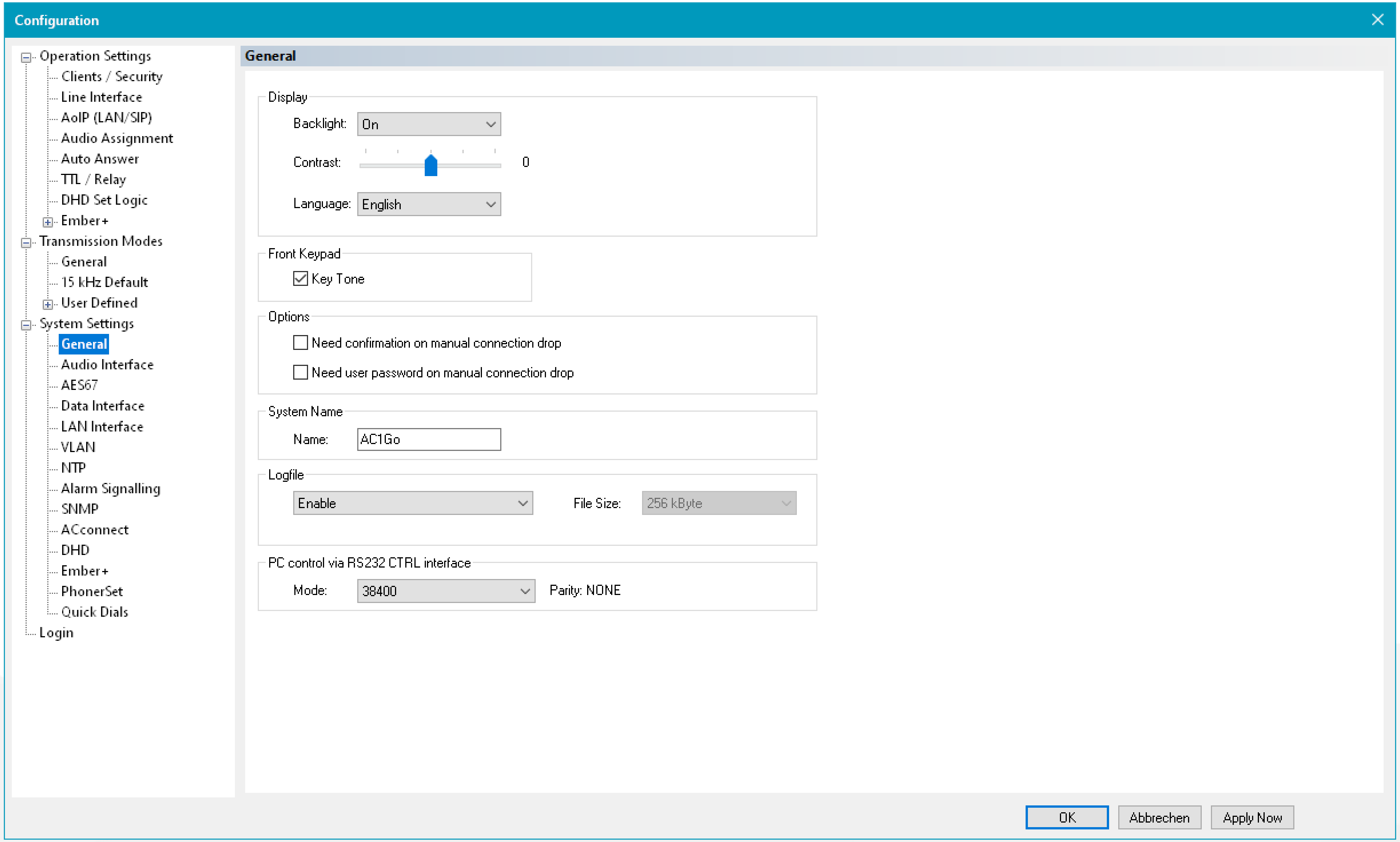


Audio Video Technologies

General (1)

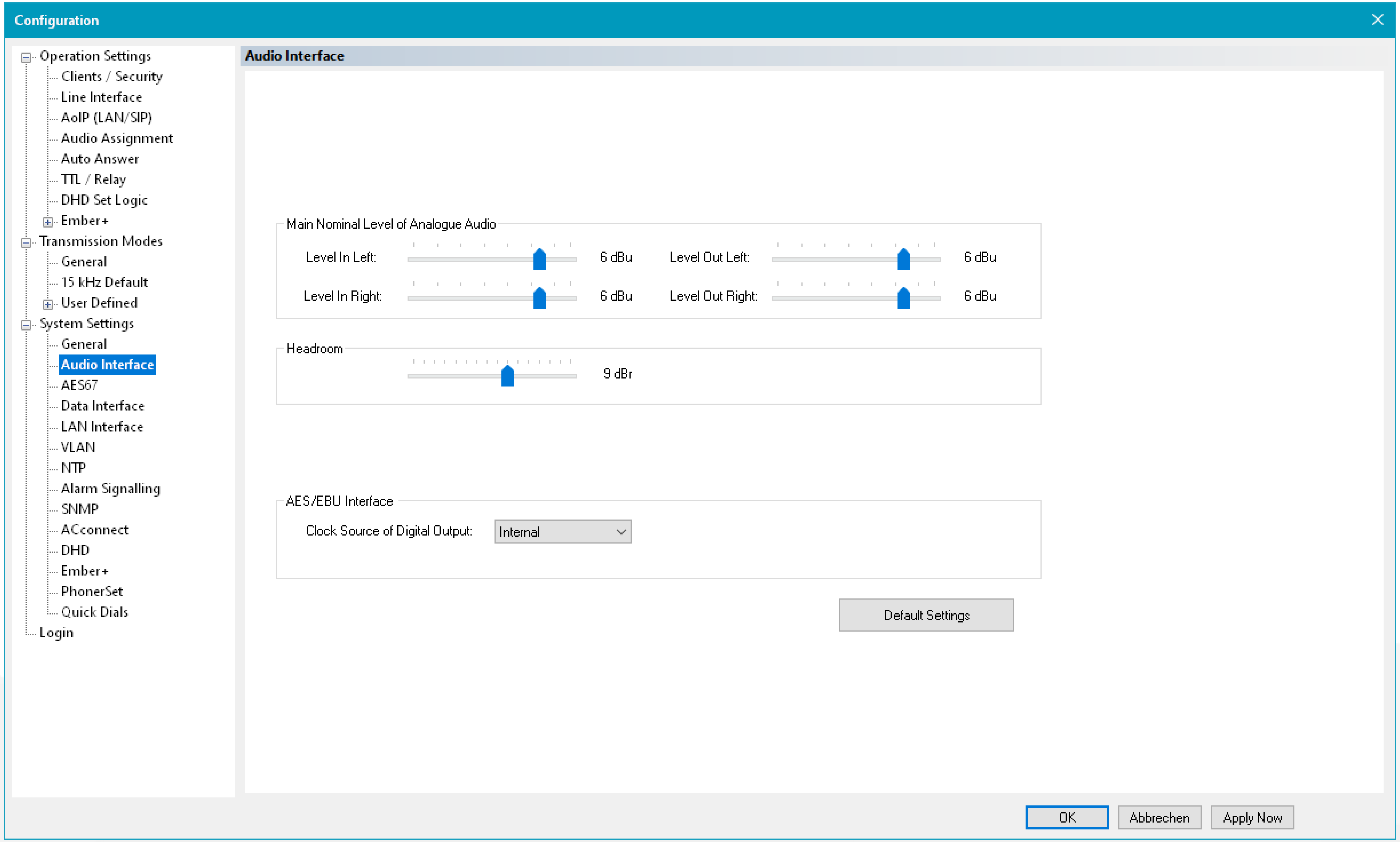
- Settings in the SYSTEM SETTINGS branch cannot be saved in presets. These settings concern the hardware or the connection to the direct environment. Flexible switching via presets seems therefore less useful.
- DISPLAY: The front display of the device is configured here.
  - BACKLIGHT: The backlight of the front display can be operated in two modes:
    - AUTO: The backlight is switched on when a key is pressed on the unit. After a few seconds, the backlight switches off again.
    - ON: The backlight is permanently on.
  - CONTRAST: Set the value for the contrast of the front display so that the display can be read best.
  - LANGUAGE: The front display of the device supports two languages.
    - ENGLISH
    - GERMAN
- FRONT KEYPAD: The front panel keyboard is configured here.
  - KEY TONE: Activates the key click of the unit keyboard.
- OPTIONS
  - NEED CONFIRMATION ON MANUAL CONNECTION DROP: If enabled, the user must confirm a warning message when he wants to terminate a connection via the user interface.
  - NEED USER PASSWORD ON MANUAL CONNECTION DROP: If enabled, the user must enter the user password when he wants to terminate a connection via the user interface. The user password is set up on the LOGIN configuration page.
- SYSTEM NAME: Enter any text. The system name is displayed in the user interface or in the MAGIC System Manager.

AVT Audio Video Technologies

General (2)

- LOGFILE: The unit can write changes to the system status to an internal log. Two settings are available:
  - DISABLE: No log is recorded.
  - ENABLE: The log file is recorded in the internal memory. If the file is full, it is overwritten piece by piece from the beginning.
- PC CONTROL VIA RS232 CTRL INTERFACE
  - MODE: Select a baud rate for the RS232 PC interface.

General (3)

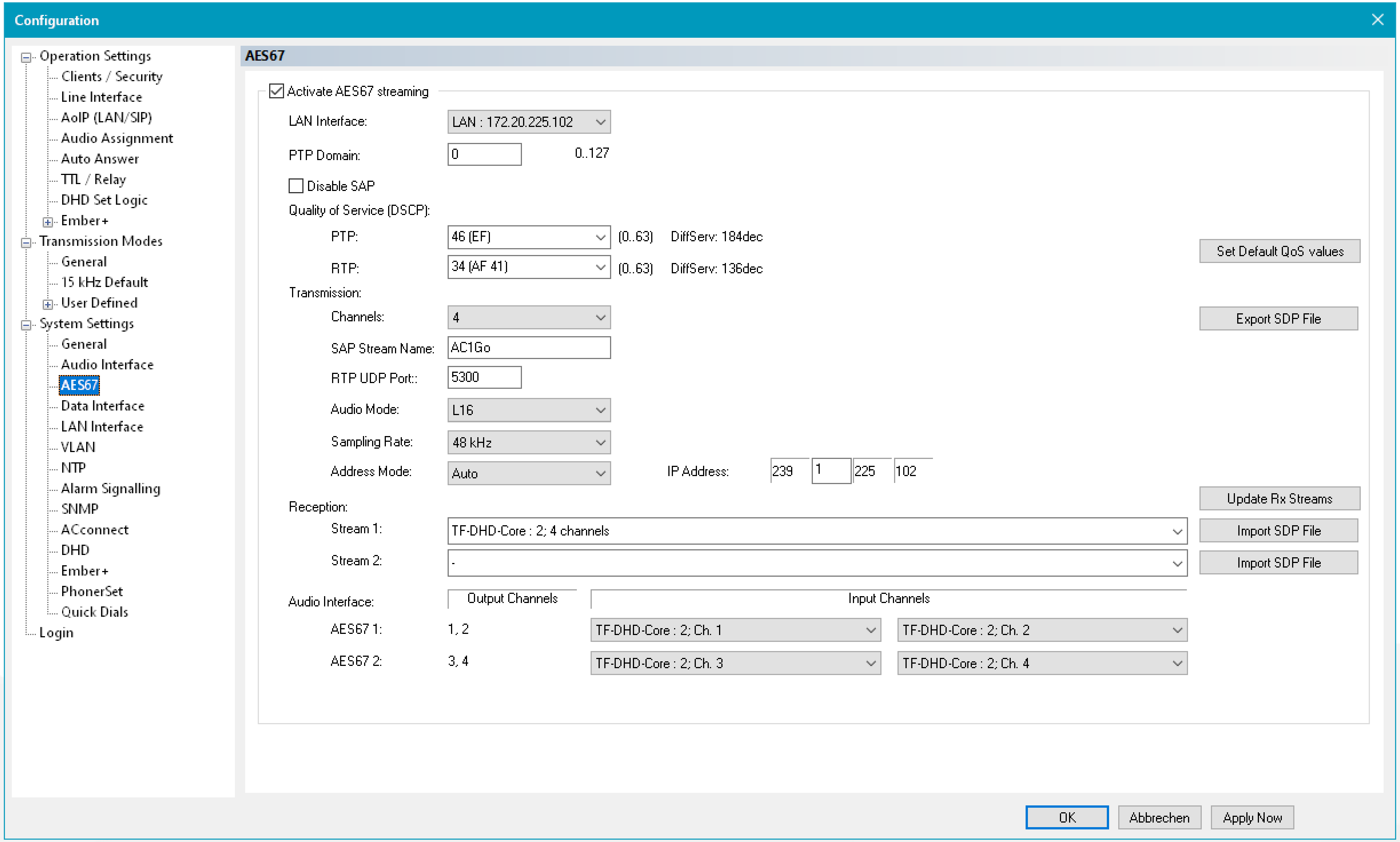


Audio Video Technologies

Audio Interface (1)

- The basic parameters of the audio interfaces of the device are configured on the AUDIO INTERFACE page.
- MAIN NOMINAL LEVEL OF ANALOGUE AUDIO:
  - LEVEL IN LEFT / RIGHT: Set the sensitivity of the audio inputs. Decreasing the value increases the audio level and vice versa. (Default setting: 6 dBu)
  - LEVEL OUT LEFT / RIGHT: Adjust the gain of the audio outputs. Decreasing the value lowers the audio level and vice versa. (Default setting: 6 dBu)
- HEADROOM: This value lowers the audio level of analogue input signals to create headroom for internal audio processing. Otherwise, the audio signal may clip. The analogue output signal is amplified by the headroom.
  The headroom is fixed at 20 dBr.
- AES/EBU INTERFACE
  - CLOCK SOURCE OF DIGITAL OUTPUT: There are sample rate converters only on the AES/EBU input. The AES/EBU output runs at system clock. Systems connected to the device via AES/EBU may require sample rate converters on their inputs.
  - This setting determines the clock to which the system's audio clock is synchronised. Some of the options may not be available depending on the line mode or Audio over IP settings.
    - INTERNAL: The digital output is synchronised to the system's internal clock.
    - RECOVERED: The digital output is synchronised to the clock applied to the AES/EBU input.
- DEFAULT SETTINGS: This button resets all settings on this page to the factory defaults.

Audio Interface (2)



Audio Video Technologies

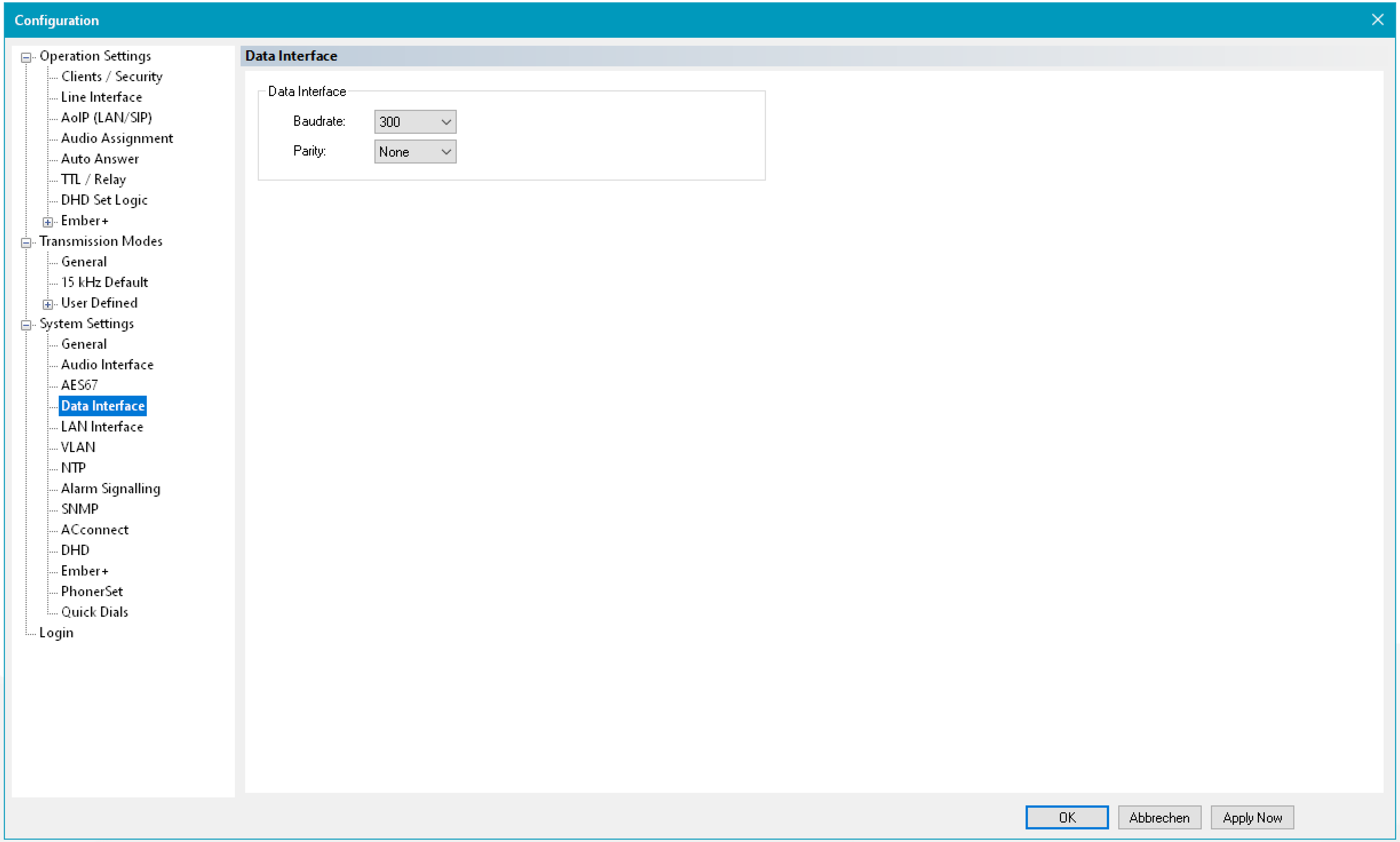
AES67 (1)

AES67 (2)

- AES67 is an open standard for Audio over IP developed by the Audio Engineering Society (AES). It sets a minimum standard for audio streaming and synchronisation in Audio over IP networks such as Dante, RAVENNA, Livewire+ and others.
- The optional AES67 functionality of the device is configured on the AES67 page.
- LAN INTERFACE: Select the IP address of the unit to be used for AES67.
- CHANNELS: Set the number of AES67 mono audio channels.
- TRANSMISSION: This section defines the AES67 stream that the unit sends.
  - SAP STREAM NAME: Enter any text. The stream is announced on the network using that name.
  - RTP UDP PORT: Set the local UDP port for the AES67 stream. (Default: 5300)
- AUDIO MODE: The audio signal is PCM encoded. The bit depths can be set here:
  - L16: 16 bit/sample
  - L24:24 bit/sample (Recommended for connections to Dante devices.)
- SAMPLING RATE: Set the sampling frequency of the audio signal.
  - 32 kHz
  - 48 kHz
- ADDRESS MODE: AES67 streams are multicast on the network. There are two modes for entering the multicast IP address:
  - MANUAL: You can freely enter the complete IP address.
  - AUTO: The IP address is derived from the IP address of the LAN interface of the device. Only the second byte of the IP address must be entered manually.
  - IP ADDRESS: Enter the entire IP address or the second byte of the IP address, depending on the ADDRESS MODE set.

- RECEPTION: The unit can subscribe to two AES67 streams with up to 8 mono audio channels each. The desired audio channels of the streams are then selected under AUDIO INTERFACE. The unit receives the stream descriptions via SAP of all streams announced in the network and lists them.
  - STREAM 1/2: Select one of the listed streams to subscribe to it.
  - UPDATE RX STREAMS: The list of detected streams is rebuilt.
- OUTPUT CHANNELS: The channels of the transmitted stream are permanently assigned to AES67 audio interfaces.
  - AES67 1: Channels 1 and 2 of the transmitted AES76 stream are assigned to the audio interface AES67 1.
  - AES67 2: Channels 3 and 4 of the transmitted AES76 stream are assigned to the audio interface AES67 2.
- INPUT CHANNELS: Audio channels of received streams can be freely assigned to the AES67 audio interfaces.
- QUALITY OF SERVICE (DSCP): The unit supports Quality of Service via DSCP (Differentiated Service Code Point). It uses the Differentiated Services (DiffServ) field in the IP header for this purpose. QoS must be activated in the network. The DiffServ values of the individual priority levels are defined and evaluated in the network.
  - PTP: Enter the DiffServ value for clock distribution via PTP here.
  - RTP: Enter the DiffServ value for AES67 audio transmission via RTP here.

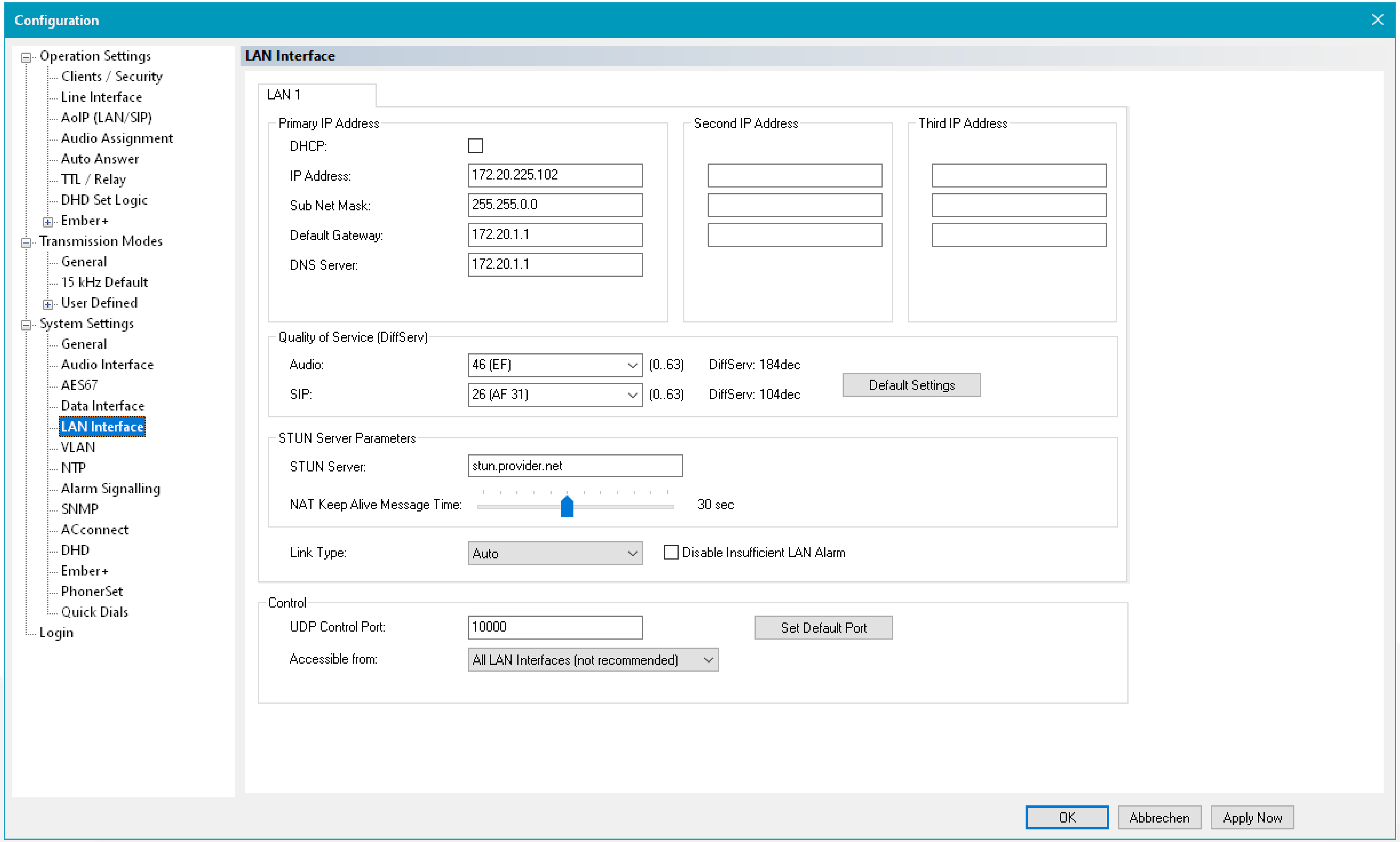
AES67 (4)



Data Interface (1)

Data Interface (2)

- Some codecs support transmission of data along with the encoded audio within the stream.
- If enabled, the device routes data provided via the DATA Interface (RS232) to the encoder.
- The data rate used for PAD reduces the data rate of the audio signal.
- The decoder retrieves the PAD from the stream and outputs it on the RS232 interface.
- BAUD RATE: The baud rate can be set in fixed steps between 300 and 38400.
- PARITY: Set how the parity bit is used in RS232 transmission:
  - NONE: No parity bit
  - EVEN
  - ODD



LAN Interface (1)

- The LAN interfaces of the device are configured on the LAN INTERFACE page.
- DHCP: Enable DHCP if the unit must obtain its IP address from a DHCP server. The obtained IP address is displayed on the front display of the unit by navigating to the home screen and then pressing the HANG UP key.
- A fixed address can be set using the following fields:
  - IP ADDRESS
  - SUBNET MASK
  - DEFAULT GATEWAY
  - DNS SERVER: Only available for the primary IP address. The DNS Server is only required to resolve the host name of SIP servers. If a SIP server is specified via the host name, the primary IP address must be used for AoIP/SIP.
- QUALITY OF SERVICE (DSCP): The device supports Quality of Service via DSCP (Differentiated Service Code Point). It uses the Differentiated Services (DiffServ) field in the IP header. QoS must be activated in the network. The DiffServ values of the individual priority levels are defined and evaluated in the network.
  - AUDIO: DiffServ value for audio transmission via AoIP/SIP or IP Leased Line.
  - SIP: DiffServ value for the AoIP/SIP connection setup.

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LAN Interface (2)

- STUN SERVER PARAMETERS: The use of a STUN server may be required by the AoIP/SIP provider. Only one STUN server is required, even if the device connects to multiple providers. Enable STUN for each SIP server individually on the LINE INTERFACE configuration page in AoIP/SIP mode.
  - STUN SERVER: IP address or host name of the STUN server.
  - NAT KEEP ALIVE MESSAGE TIME: Set the time interval at which the device periodically sends keep-alive packets to the SIP server. This allows routers and firewalls to keep the SIP communication ports open. This allows the SIP server to inform the unit of incoming calls. Keep alive packets are only sent if STUN is enabled for the respective SIP account. (Default: 20 seconds)
- LINK TYPE: Set how the unit's Ethernet connection to the network switch is established.
  - AUTO: The device automatically detects the link type. The System Monitor should display the status of the LAN interface at 100 Mbps, full.
  - TYPE 1/2: If the status of the LAN interface in the System Monitor is not 100 Mbps, full, even though the switch supports this mode, test whether Type 1 or Type 2 solves the problem.
  - DISABLE INSUFFICIENT LAN ALARM: For audio transmission, the LAN interface should be operated in 100 Mbps, full mode to prevent audio interference. Therefore, the unit raises the Insufficient LAN alarm if this mode is not established.

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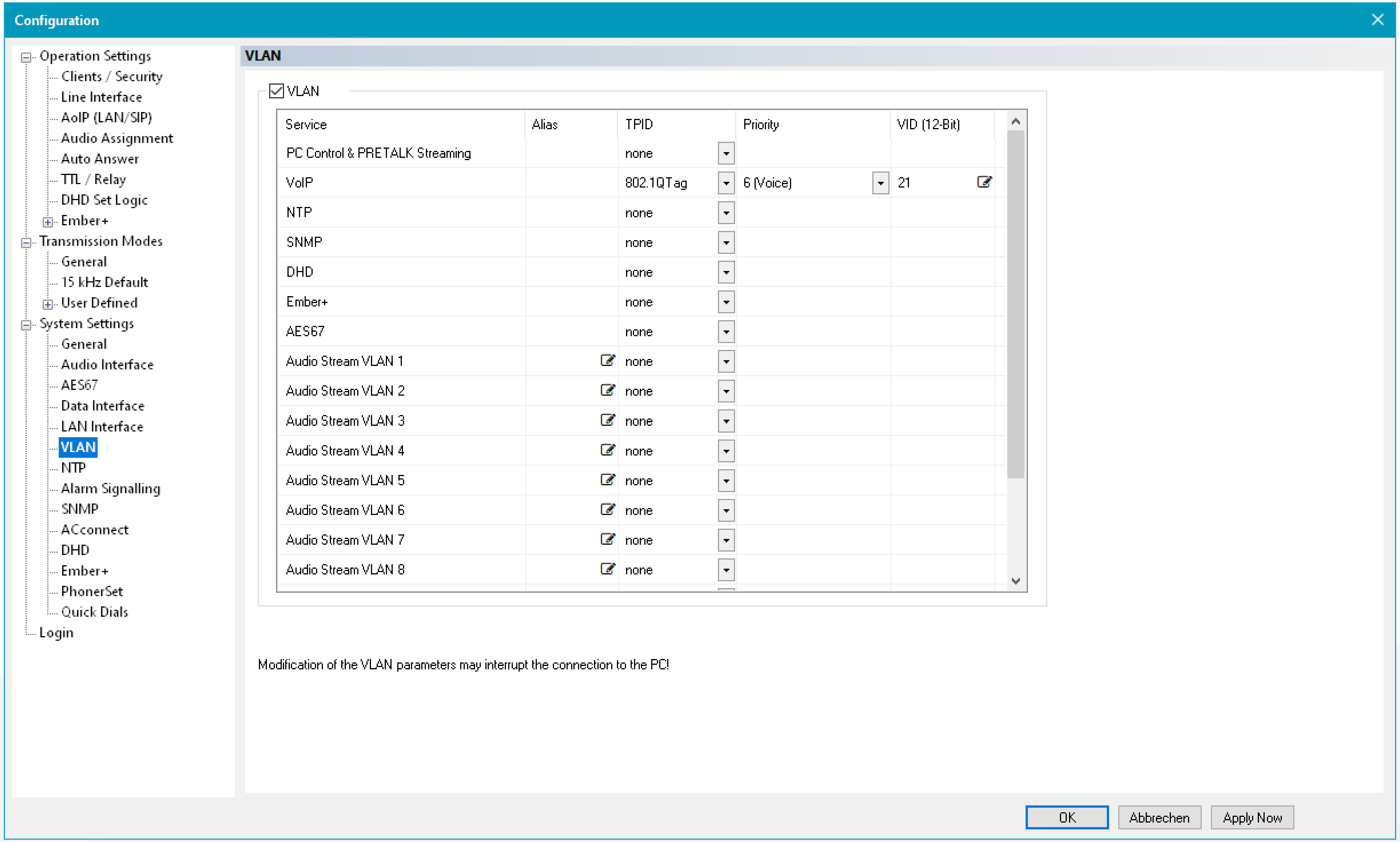
LAN Interface (3)

- CONTROL: Configure the connection of the PC software to the unit.
    - UDP CONTROL PORT: UPD port on the device to which all PCs connect.
    - ACCESSIBLE FROM: Limit access to the device to a specific IP address.
        - ALL LAN INTERFACES: Control is possible via all IP addresses. (not recommended)

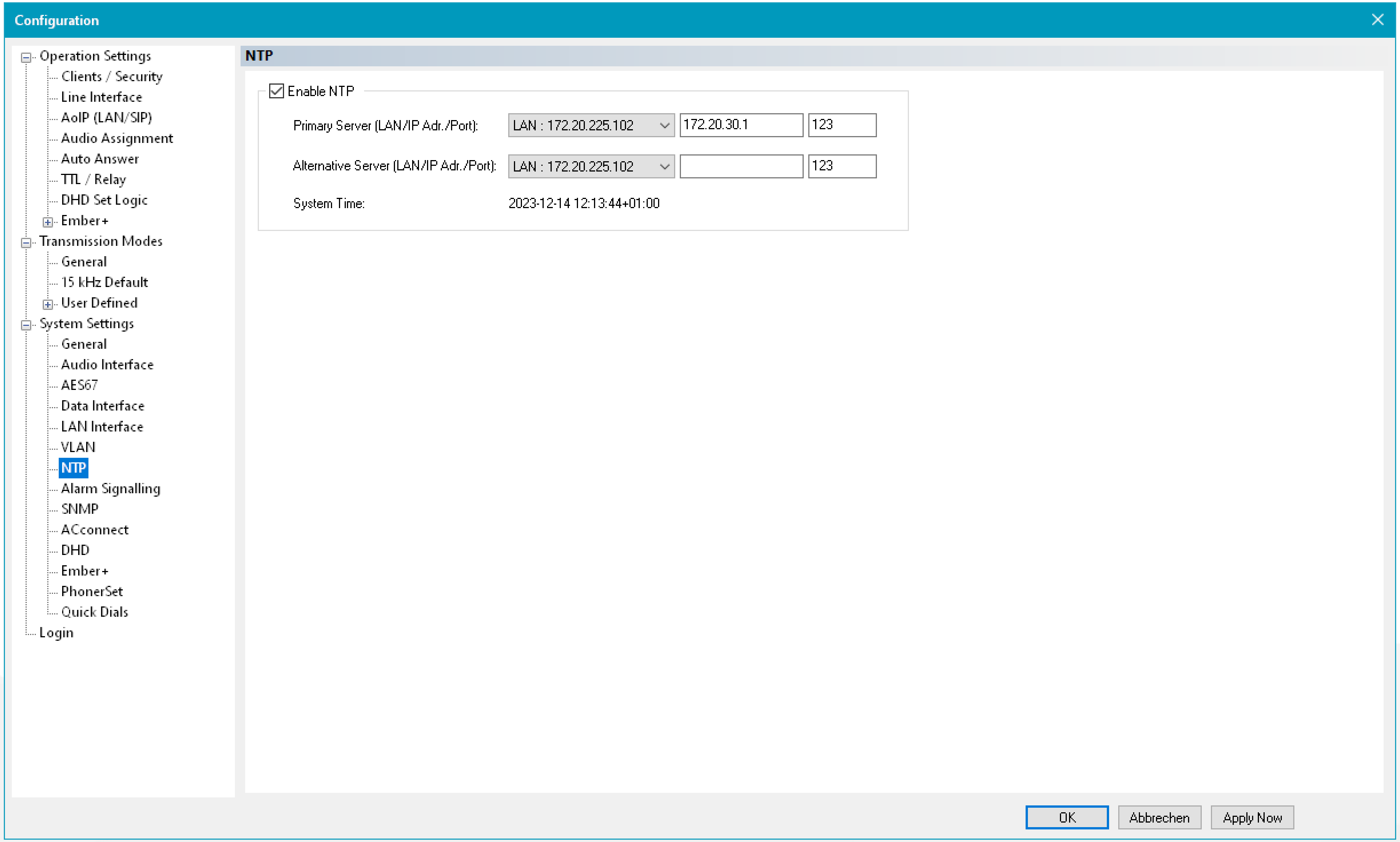
LAN Interface (4)


Configuration
Operation Settings
Clients / Security
Line Interface
AoIP (LAN/SIP)
Audio Assignment
Auto Answer
TTL / Relay
DHD Set Logic
Ember+
Transmission Modes
General
15 kHz Default
User Defined
System Settings
General
Audio Interface
AES67
Data Interface
LAN Interface
VLAN
NTP
Alarm Signalling
SNMP
AConnect
DHD
Ember+
PhonerSet
Quick Dials
Login
VLAN
VLAN
Service
Alias
TPID
Priority
VID (12-Bit)
PC Control & PRETALK Streaming
none
VoIP
802.1QTag
6 (Voice)
21
NTP
none
SNMP
none
DHD
none
Ember+
none
AES67
none
Audio Stream VLAN 1
none
Audio Stream VLAN 2
none
Audio Stream VLAN 3
none
Audio Stream VLAN 4
none
Audio Stream VLAN 5
none
Audio Stream VLAN 6
none
Audio Stream VLAN 7
none
Audio Stream VLAN 8
none
Modification of the VLAN parameters may interrupt the connection to the PC!
OK
Abbrechen
Apply Now

VLAN (1)

VLAN (2)

- The device supports tagged VLANs.
- The VLAN page is used to configure the VLAN parameters for individual services / protocols.
- VLAN: The VLAN function can be globally activated or deactivated. This setting is also possible via front display. E.g., if access to the system via LAN is no longer possible due to a misconfiguration.
- SERVICE: All services provided by the unit which support VLANs are listed here.
- ALIAS: for the AUDIO STREAM VLANs, any text can be entered here to help assign the correct VLANs on other configuration pages.
- TPID: Enables or disables VLAN for a specific service.
  - NONE: VLAN not enabled.
  - 802.1QTag: VLAN enabled.
- PRIORITY: The priority can be set from 0 = lowest priority to 7 = highest priority. It is recommended to set the priority to 6 for services that transmit audio (VoIP, AES67, audio streams).
- VID (12-bit): VLAN ID for the service.



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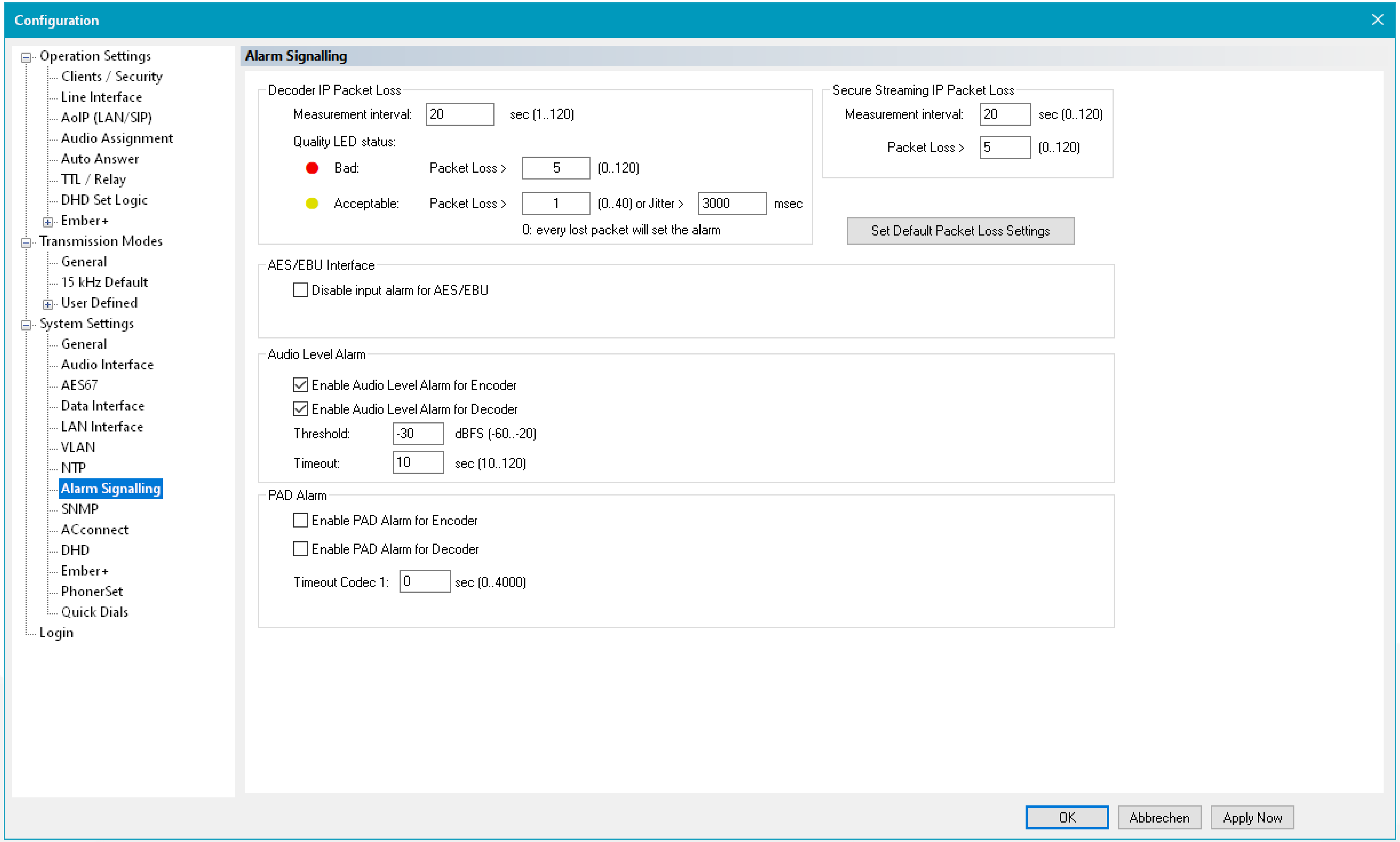
NTP (1)

- The device can synchronise itself with an external clock via NTP (Network Time Protocol).
- This allows log file entries to be time-stamped.
- Using NTP, the delays of different transmission lines can also be compensated for with the AUDIO DELAY SYNC function in the IP Leased Line Extended mode. For this, NTP must be activated on the encoder and on all decoders.
- ENABLE NTP: Enables the NTP on the unit.
- Two servers can be entered.
  - PRIMARY: If available, this server is used.
  - ALTERNATIVE: If the PRIMARY server is not available, the unit switches to this server.

- The following parameters must be set for each server:
  - LAN: IP address of the unit to be used to connect to the NTP server.
  - IP ADDRESS: Enter the IP address of the NTP server. Host names are not supported.
  - PORT: Enter the port of the NTP service on the server. (Default: 123)

NTP (2)



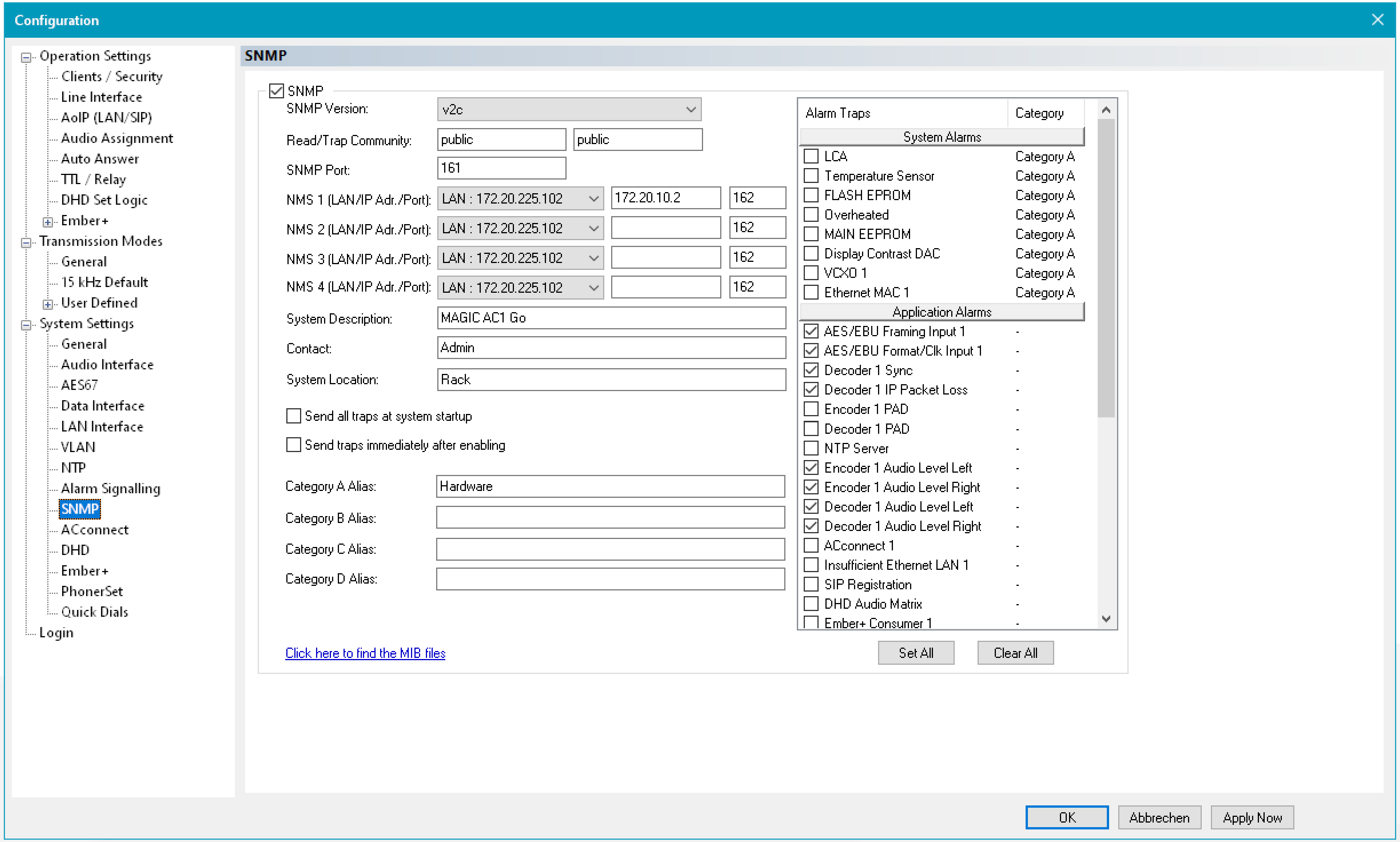
Alarm Signalling (1)

- Alarm parameters can be configured on the ALARM SIGNALLING page.
- DECODER IP PACKET LOSS: On the user interface, under RTP AUDIO QUALITY, the quality of the received stream is signalled for each codec channel via an LED.
  - ●: Good
  - ●: Acceptable
  - ●: Poor
- In addition, an alarm is set in the event of poor quality. The threshold values for the individual levels can be set here.
  - MEASUREMENT INTERVAL: Sets the measurement interval for assessing the quality of the stream. During this period, the lost packets (packet loss) are counted and the highest jitter value that occurred is recorded.
  - BAD: The threshold value for packet loss, above which the stream quality is classified as poor.
  - ACCEPTABLE: The threshold values for the packet loss as well as the maximum jitter value from which the stream quality is classified as acceptable.
- SECURE STREAMING IP PACKET LOSS: A separate threshold for the alarm can be set for the second stream when secure streaming is activated:
  - MEASUREMENT INTERVAL: Sets the measurement interval for assessing the quality of the stream. During this period, the lost packets (packet loss) are counted.
  - PACKET LOSS: The threshold value for packet loss above which the stream quality is classified as poor.

Alarm Signalling (2)

Alarm Signalling (3)

- AES/EBU INTERFACE: The device raises an alarm if there is no signal at the AES/EBU inputs. For example, if an AES/EBU interface is only used to output an audio signal, the alarm can be disabled.
  - DISABLE INPUT ALARM FOR AES/EBU
- AUDIO LEVEL ALARM: The device monitors the audio levels of all encoders and decoders.
  - ENABLE AUDIO LEVEL ALARM FOR ENCODER / DECODER: Enable the alarm for each direction separately.
  - THRESHOLD: If this level in dBFS is not reached for the set period, the respective alarm is raised.
  - TIMEOUT: If the set level is not reached for this period, the respective alarm is raised.
- PAD ALARM: The device monitors the data transmission via PAD.
  - ENABLE PAD ALARM FOR ENCODER / DECODER: The alarm can be enabled separately for each PTY direction.
  - TIMEOUT CODEC 1: The alarm is set if no PAD data has been transmitted during this time period.



Audio Video Technologies

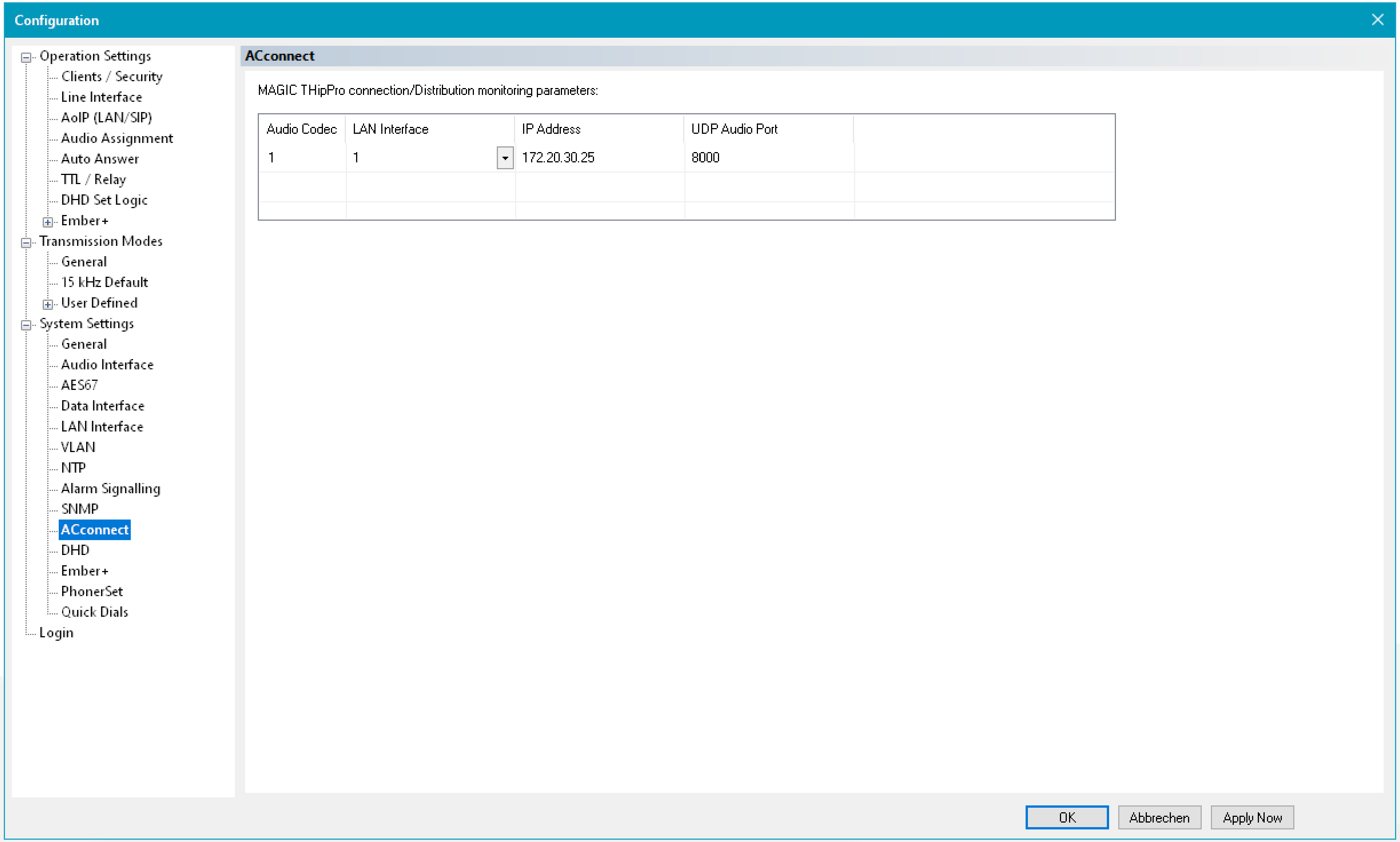
SNMP (1)

SNMP (2)

- The device can be monitored via a network management system. The parameters for that are configured on the SNMP page.
- The information accessible via SNMP can only be read.
- The device responds to Get requests and sends traps.
- Find the MIB for the device in the installation directory of the PC software.
- SNMP VERSION: Select the SNMP version. SNMPv1 and SNMPv2c are supported.
- READ / TRAP COMMUNITY: Enter a string for the READ community and a string for the TRAP community.
- SNMP PORT: Enter the local UDP port for receiving SNMP requests and sending SNMP responses. The remote port is derived from the received SNMP requests. (Default: 161)
- NMS 1-4: The unit can send traps to up to four Network Management Systems. Traps are sent when the following parameters are set for an NMS:
  - LAN: The IP address of the device used to send SNMP traps.
  - IP ADDR: The destination IP address.
  - PORT: The destination UDP port. (Default: 162)
- SYSTEM DESCRIPTION: A device description. This string is part of the standard MIB.
- CONTACT: The person responsible for this device. This string is part of the default MIB.
- SYSTEM LOCATION: Physical location of the device. This string is part of the standard MIB.

- SEND ALL TRAPS AT SYSTEM STARTUP : Enable this option to send all traps when the device has finished booting.
- SEND TRAPS IMMEDIATELY AFTER ENABLING : Enable this option to send a trap immediately after it has been activated in the configuration.
- CATEGORY A-D ALIAS: To reduce the number of traps, several traps can be combined into one category. If an alarm is set which is contained in the category, the trap of this category can be sent. The name of the category can be configured here.
    - The category traps can be activated at the end of the ALARM-TRAPS list.
- ALARM-TRAPS: All available traps are listed in this table.
    - Select traps which should be sent.
    - Click in the CATEGORY column to assign a trap to a category.
- MISCELLANEOUS TRAPS:
    - CHANNEL 1 RTP QUALITY
    - COLD START: The SNMP agent is being reinitialised.
    - AUTHENTICATION FAILURE: An unknown community string was used in a request.

SNMP (3)


Configuration
Operation Settings
Clients / Security
Line Interface
AoIP (LAN/SIP)
Audio Assignment
Auto Answer
TTL / Relay
DHD Set Logic
Ember+
Transmission Modes
General
15 kHz Default
User Defined
System Settings
General
Audio Interface
AES67
Data Interface
LAN Interface
VLAN
NTP
Alarm Signalling
SNMP
ACconnect
DHD
Ember+
PhonerSet
Quick Dials
Login
ACconnect
MAGIC THipPro connection/Distribution monitoring parameters:
Audio Codec
LAN Interface
IP Address
UDP Audio Port
1
1
172.20.30.25
8000
OK
Abbrechen
Apply Now

ACconnect (1)

- The unit can be integrated into a MAGIC THipPro system via the ACconnect function.
- In this case, the unit is configured and controlled via the MAGIC THipPro software.
- The ACCONNECT page allows you to disconnect from a MAGIC THipPro by deleting the IP address of the MAGIC THipPro from the IP ADDRESS column.

Audio
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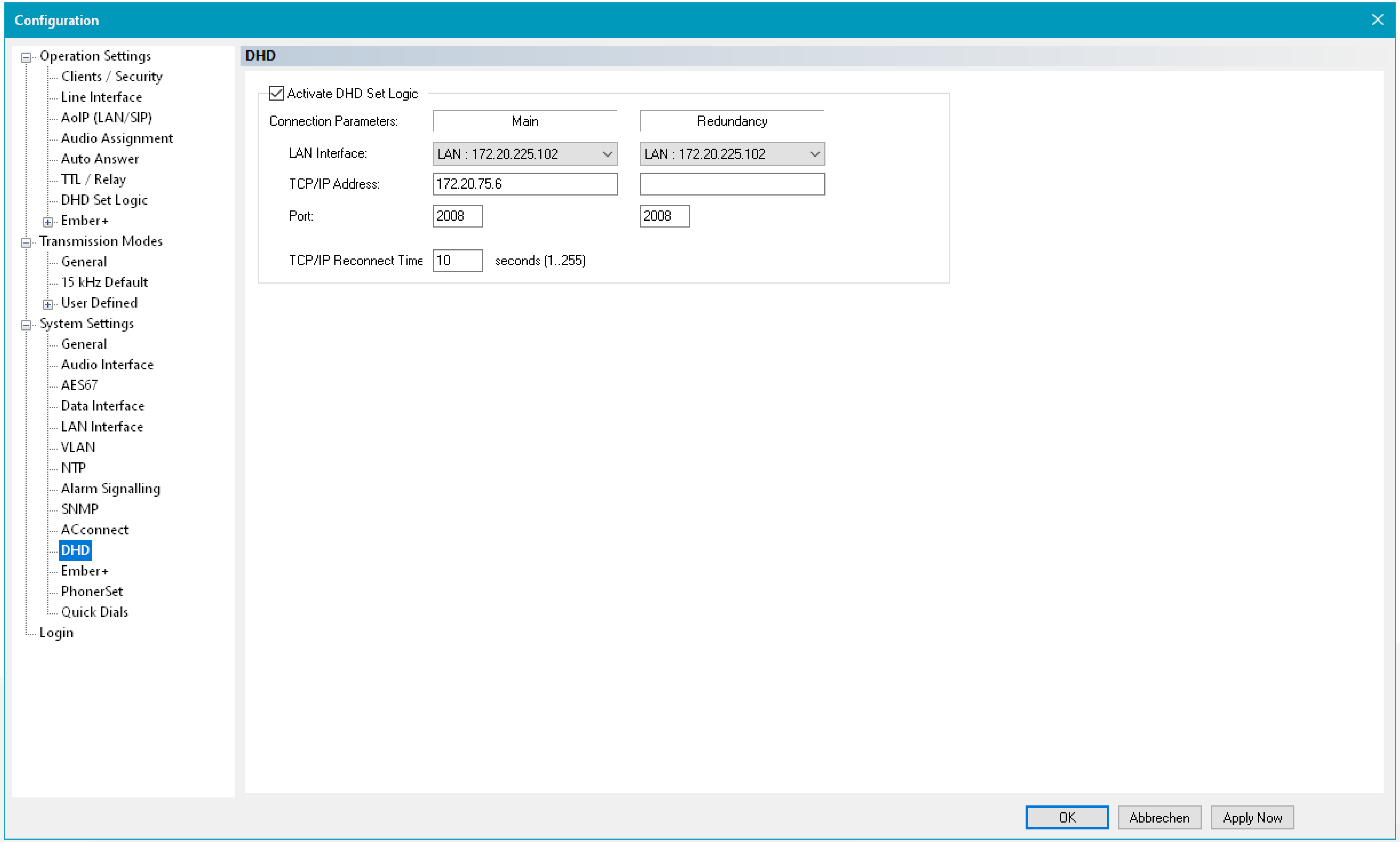
ACconnect (2)

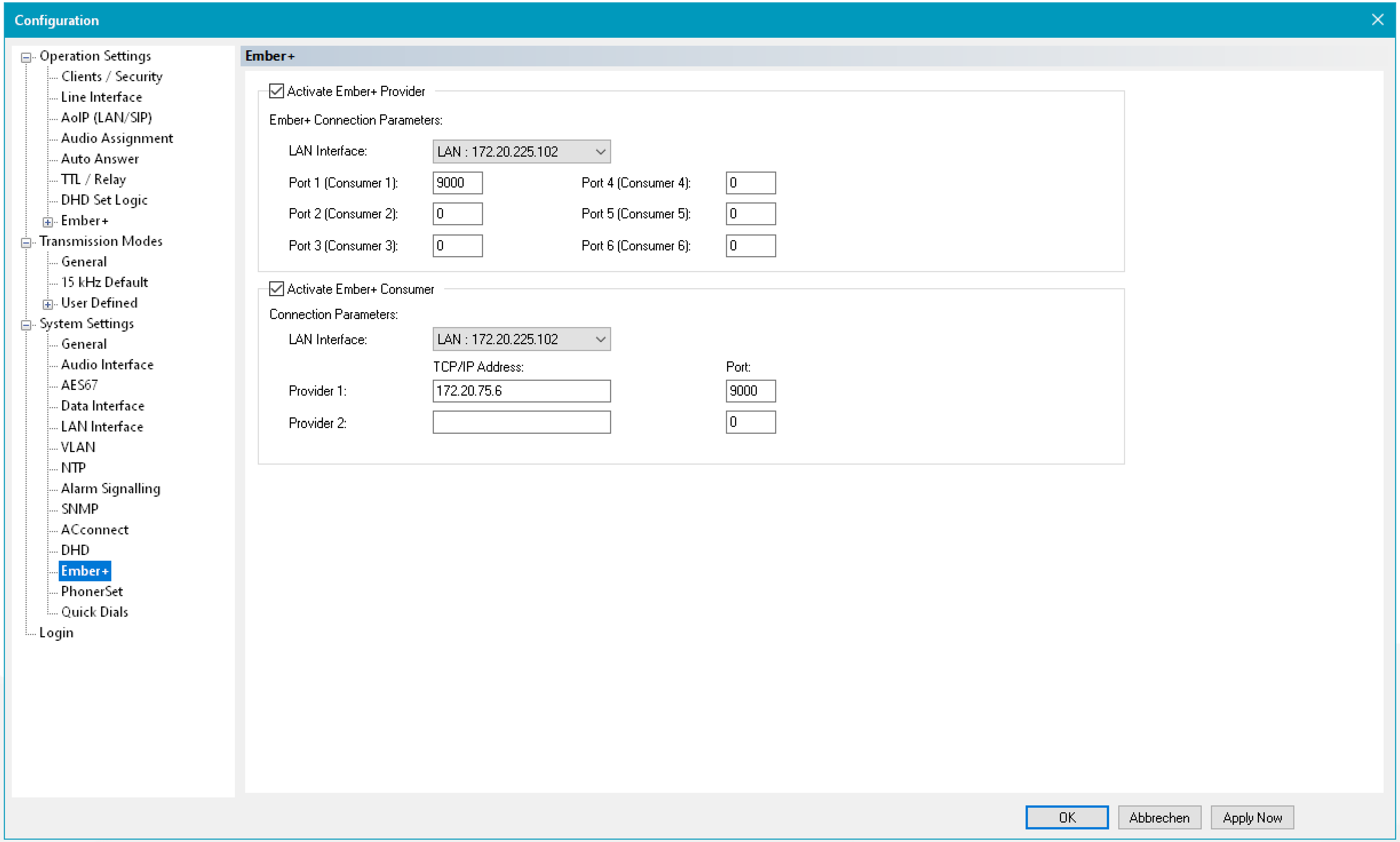


Audio Video Technologies

DHD (1)

- The unit supports DHD-ECP (DHD-External Control Protocol) for communication with a DHD core.
- The connection parameters are configured on the DHD page.
- If DHD is enabled, the DHD Set Logic GPIOs are available.
- ACTIVATE DHD SET LOGIC: Activates the DHD communication protocol.
- The unit always tries to connect via the MAIN connection. While it is not available, the unit connects via the REDUNDANCY connection.
- LAN INTERFACE: Select the IP address of the unit through which the connection to the DHD core is made.
- TCP/IP ADDRESS: Enter the IP address of the DHD Core.
- PORT: Enter the port via which the DHD Core can be reached.
- TCP/IP RECONNECT TIME: Specifies the time interval between TCP connection requests to the DHD core in seconds. The range is 1 - 255 seconds. (Default: 10 seconds)

DHD (2)

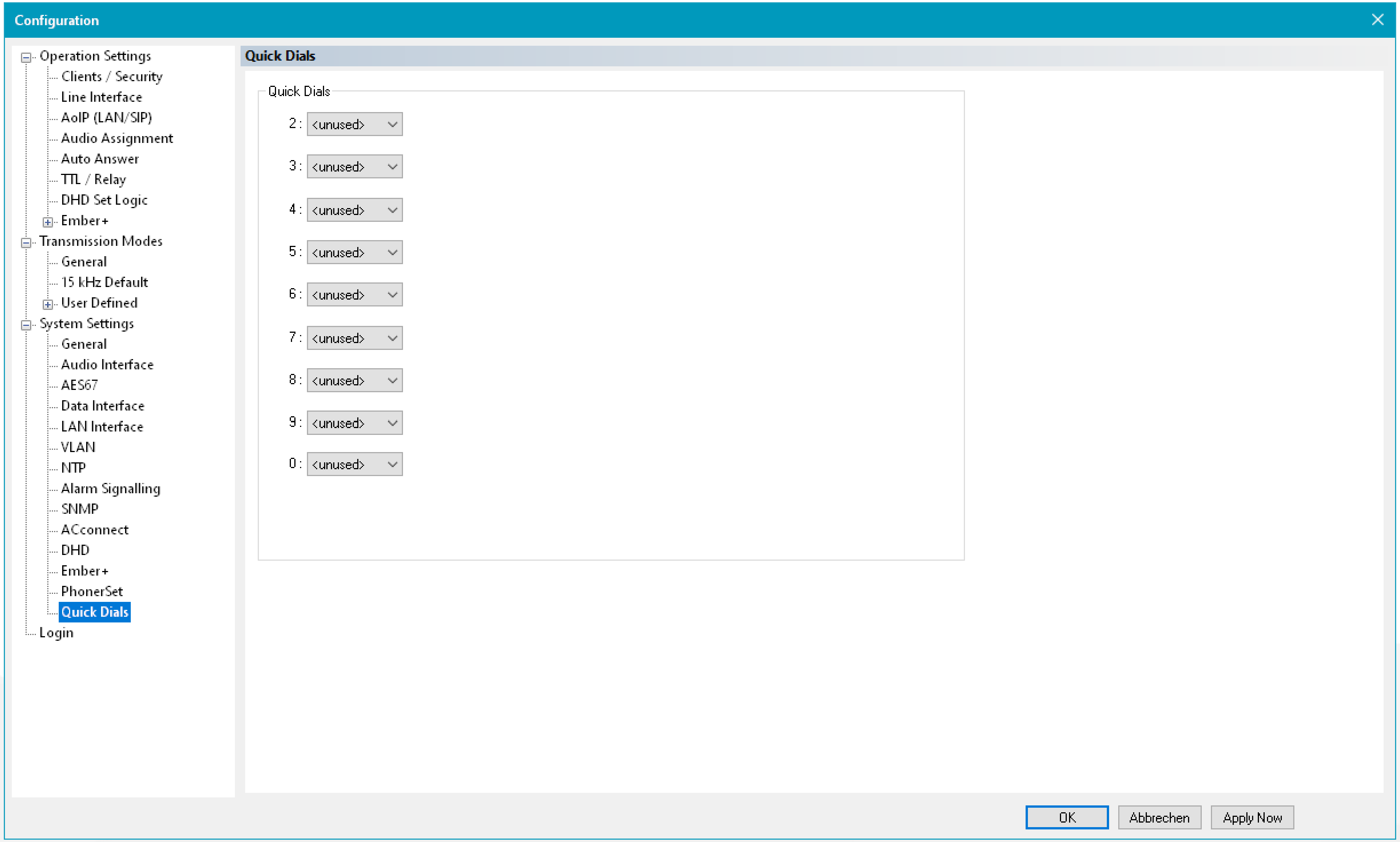


AVT Audio Video Technologies

Ember+ (1)

Ember+ (2)

- The unit supports the Ember+ protocol in both the provider role and the consumer role. Both roles can be active at the same time.
- GPIOs are available via the Ember+ Provider of the unit.
- Text information (e.g., phone numbers) can be exchanged via the Ember+ Consumer of the unit.
- The parameters for communication via Ember+ are configured on the EMBER+ page.
- ACTIVATE EMBER+ PROVIDER: Activates the Ember+ provider of the unit.
- LAN INTERFACE: Select the IP address of the unit to which the Ember+ consumers of other devices can connect.
- PORT N (CONSUMER N): Up to 6 consumers can connect to the unit simultaneously. Each consumer must use an individual port. Unused ports should be set to '0'. (Default: 9000, 9001, 9002, ...)
- ACTIVATE EMBER+ CONSUMER: Activates the Ember+ consumer of the unit. The unit can connect to two providers simultaneously.
- LAN INTERFACE: Select the IP address of the unit through which it connects to the Ember+ providers of other devices.
- TCP/IP ADDRESS / PORT: Enter the IP address and port of each provider to which the unit should connect.



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Quick Dials (1)

Quick Dials (2)

- The number keys on the front of the unit can also be used as quick-dial keys.
- Press and hold a number key to trigger a quick-dial function.
- The quick-dial functions are configured on the QUICK DIALS page.
- A function can be defined for each number key:
  - <UNUSED>: not defined.
  - PRESET: Load a preset
    - All available presets are listed for selection during configuration.

MAGIC AC1 Go

Configuration

- Login

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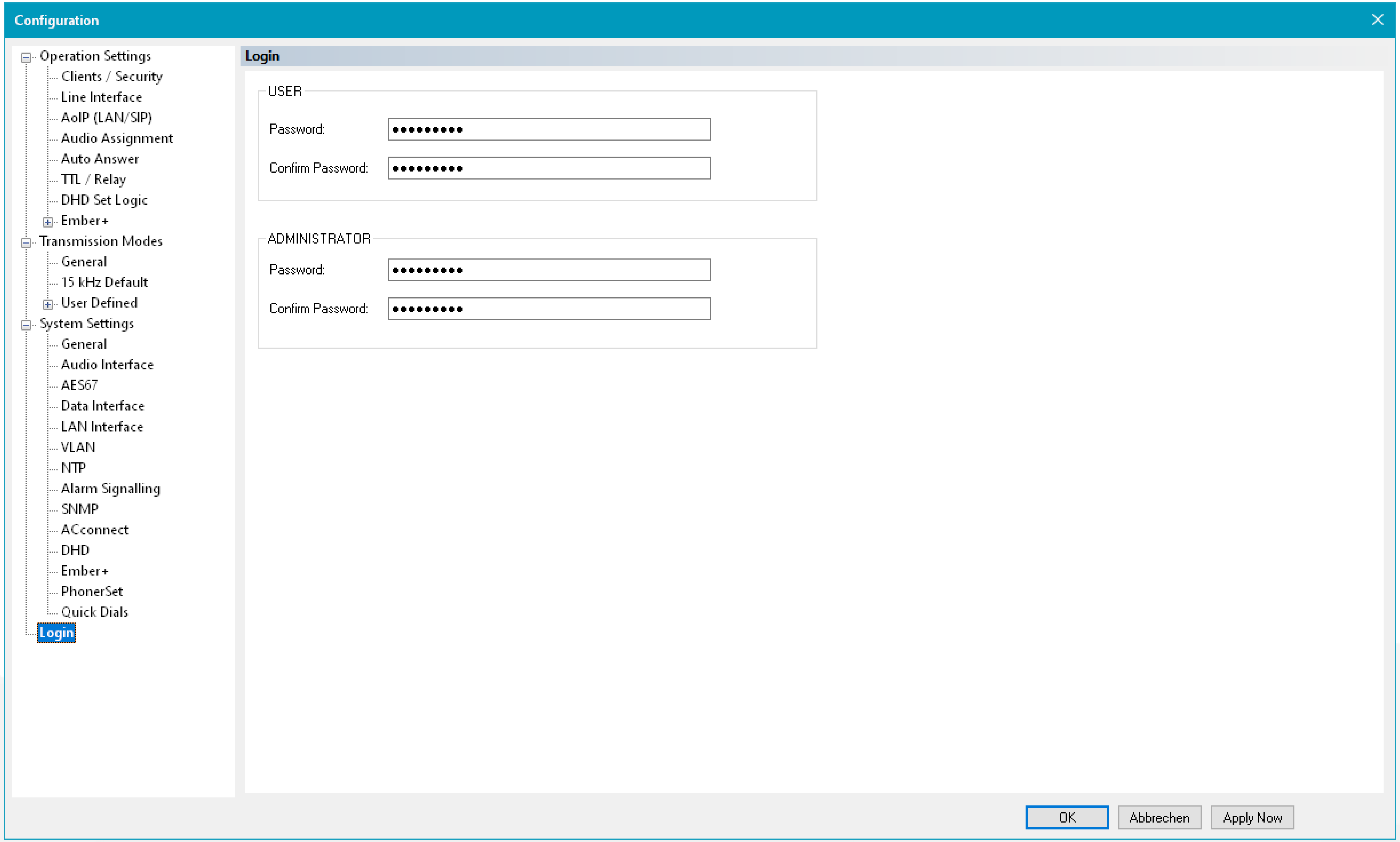

AVT Audio Video Technologies

Login (1)

Login (2)

- Passwords can be configured to protect the configuration of the device from unauthorised access. These are set on the LOGIN page.
- There are two levels of permissions:
  - ADMINISTRATOR: If an administrator password is set, the password is required for the following operations :
    - Viewing and changing the configuration
    - Importing configurations
    - Managing presets
    - Loading presets
    - Opening the system panel
    - Accessing the file system
    - Updating the firmware
    - Resetting to factory settings
  - USER: If a user password is set, the password allows presets to be loaded.
    - If only a user password is set, the user password acts as the administrator password.
- Note: To gain access to the configuration without the password, the unit must be reset to factory settings via the front display.
- Note: The LOCAL SETTINGS are not protected by these logins, but by the permissions of the operating system. (See Local Settings > Settings Folder for further information.)

MAGIC AC1 Go

Main Panel

- AoIP/SIP

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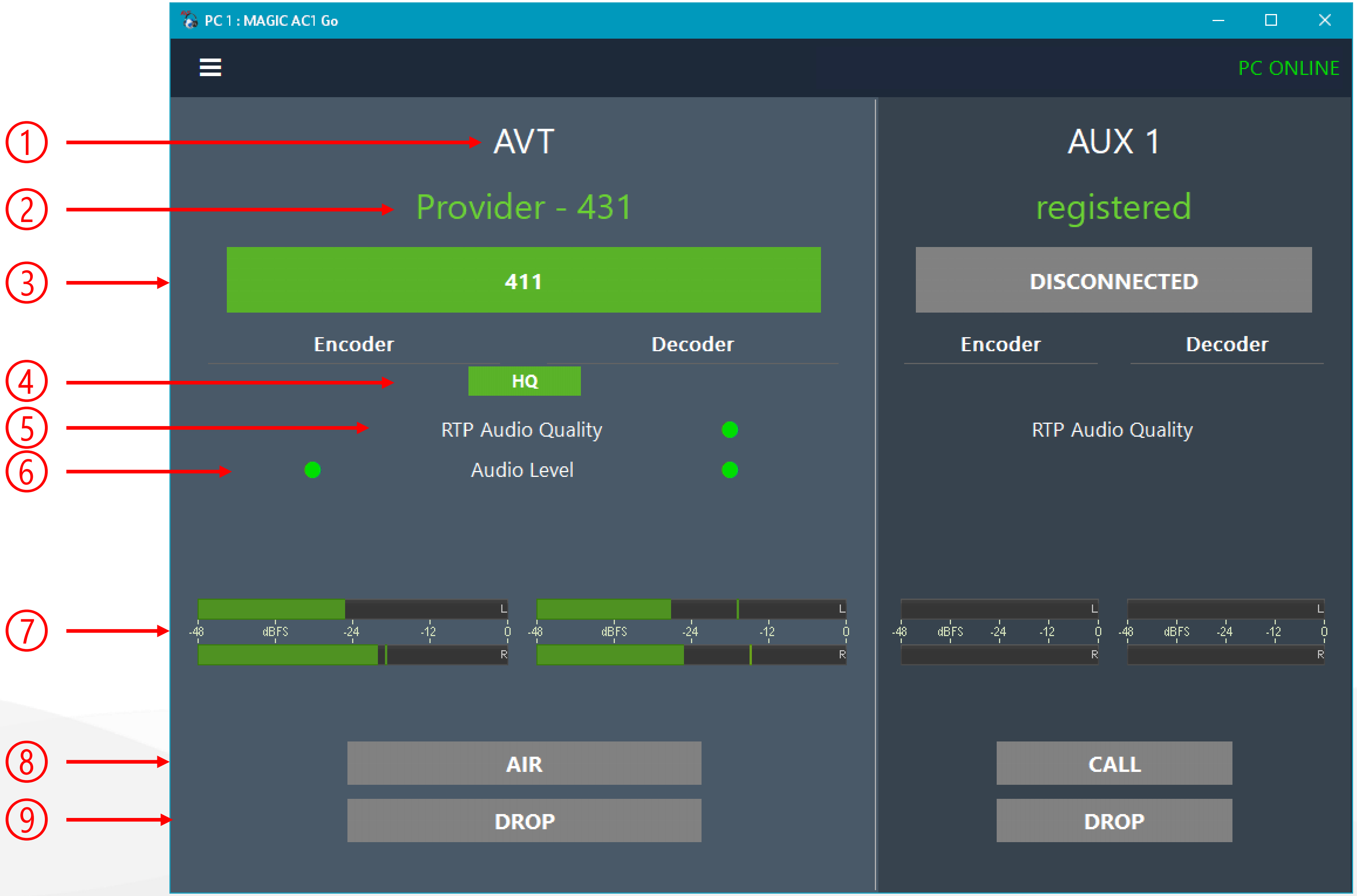


Main Panel – AoIP/SIP (1)

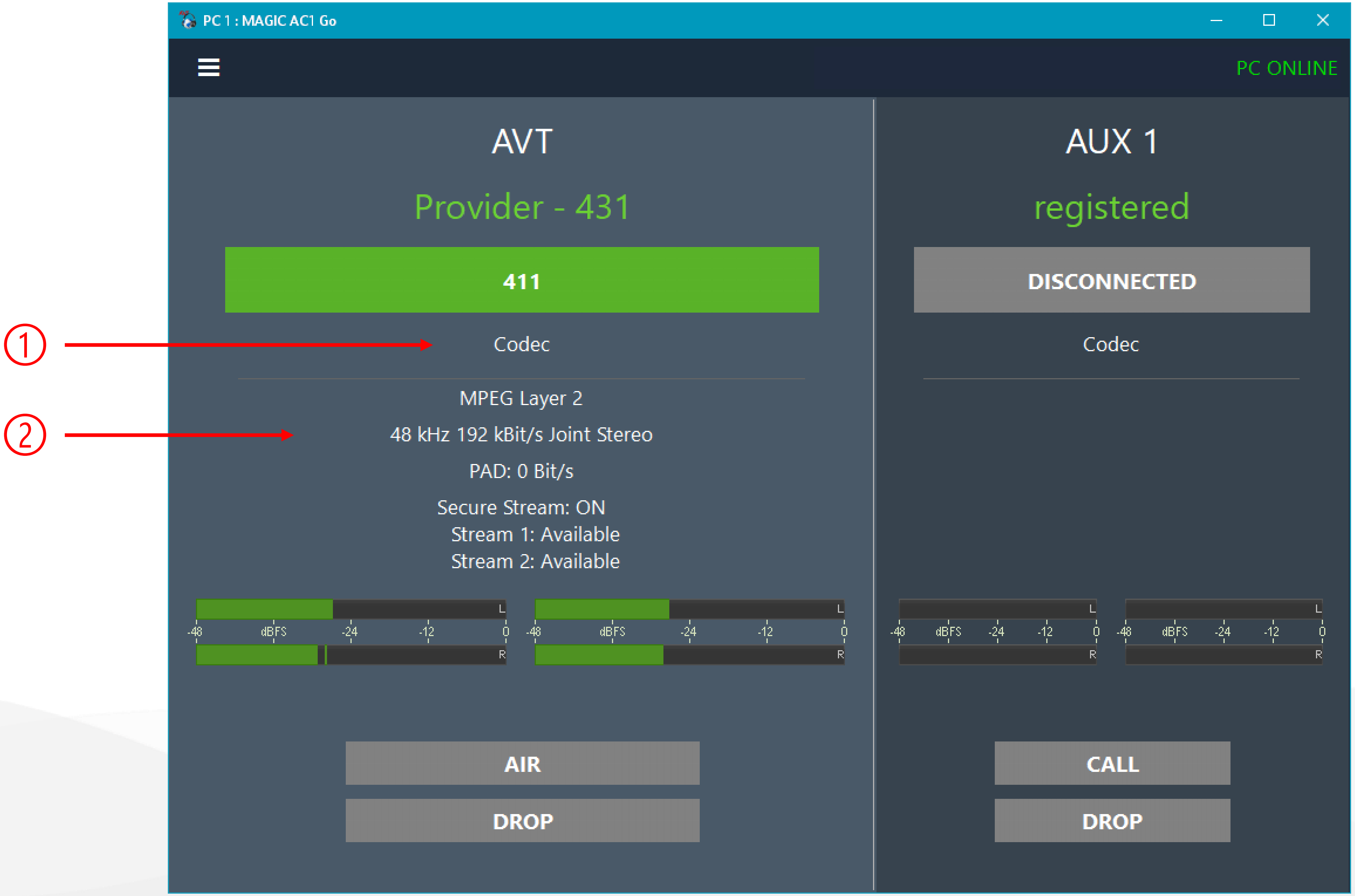
Main Panel – AoIP/SIP (2)

- In the AoIP/SIP operating mode, the codec channel and AUX channel are displayed in the main window.
- The following information is displayed:

1. The label of the channel.
2. SIP account information.
   - If the channel is connected, the SIP account which is used for the connection is displayed.
   - If the channel is disconnected the SIP registration status off all configured accounts of the channel is displayed via a status text and the font colour of that text:
     - REGISTERED: (green) All SIP accounts are registered.
     - REGISTERED: (yellow) At least one SIP account could not register with the SIP server.
     - INVALID: (red) None of the SIP accounts could register with the SIP server.
   - Click on the text to get more information about the available SIP accounts and their registration status.
   - In case of registration errors, you can get more information under System Monitor > SIP Monitor.
3. Channel Status: Shows the connection status of the channel in different colours. During a connection, number, name or IP address of the remote party is displayed.
   - Grey: There is no connection.
   - Green: The connection is established.
4. Encoder and decoder status.
   - The audio quality is displayed via a symbol:
     - HQ: A high-quality codec is used.
     - HD: The G.722 codec is used.
     - SD: The G.711 codec is used.
   - Click on the symbol to get information about
     - codec
     - algorithm
     - data rate
     - PAD
     - Secure Streaming

5. The RTP Audio Quality shows the quality of the received data stream in the colours green, yellow and red. The threshold values of the individual levels are set on the **Alarm Signalling** configuration page.
6. Status of the Audio Level monitoring. The thresholds are set on the **Alarm Signalling** configuration page.
7. The Level Meters of Encoder on the left and Decoder on the right show the current audio level in dBFS. During a connection, the level is displayed in green.
   If there is no connection the Encoder level is displayed in grey.
8. CALL: Press the button to establish a connection. When there is an incoming call, the key flashes yellow.
9. DROP: Press the button to terminate the connection or reject an incoming call. When there is an incoming call, the button flashes yellow.

Main Panel – AoIP/SIP (3)



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Main Panel – AoIP/SIP (4)

- The detailed codec information can be permanently displayed on the main panel.
- This view is enabled under **Menu > Configuration > Local Settings > Application Parameters > Show detailed Codec information in Codec Line**

1. CODEC: Codec information shows details of encoder and decoder combined.
2. DETAILS: Shows the following information if available:
   - Codec
   - Algorithm
   - Sampling rate
   - Bitrate
   - Mode (Mono, Stereo)
   - PAD status and bitrate
   - Secure Stream mode
   - Status of stream 1 and stream 2 if Secure Streaming is enabled.

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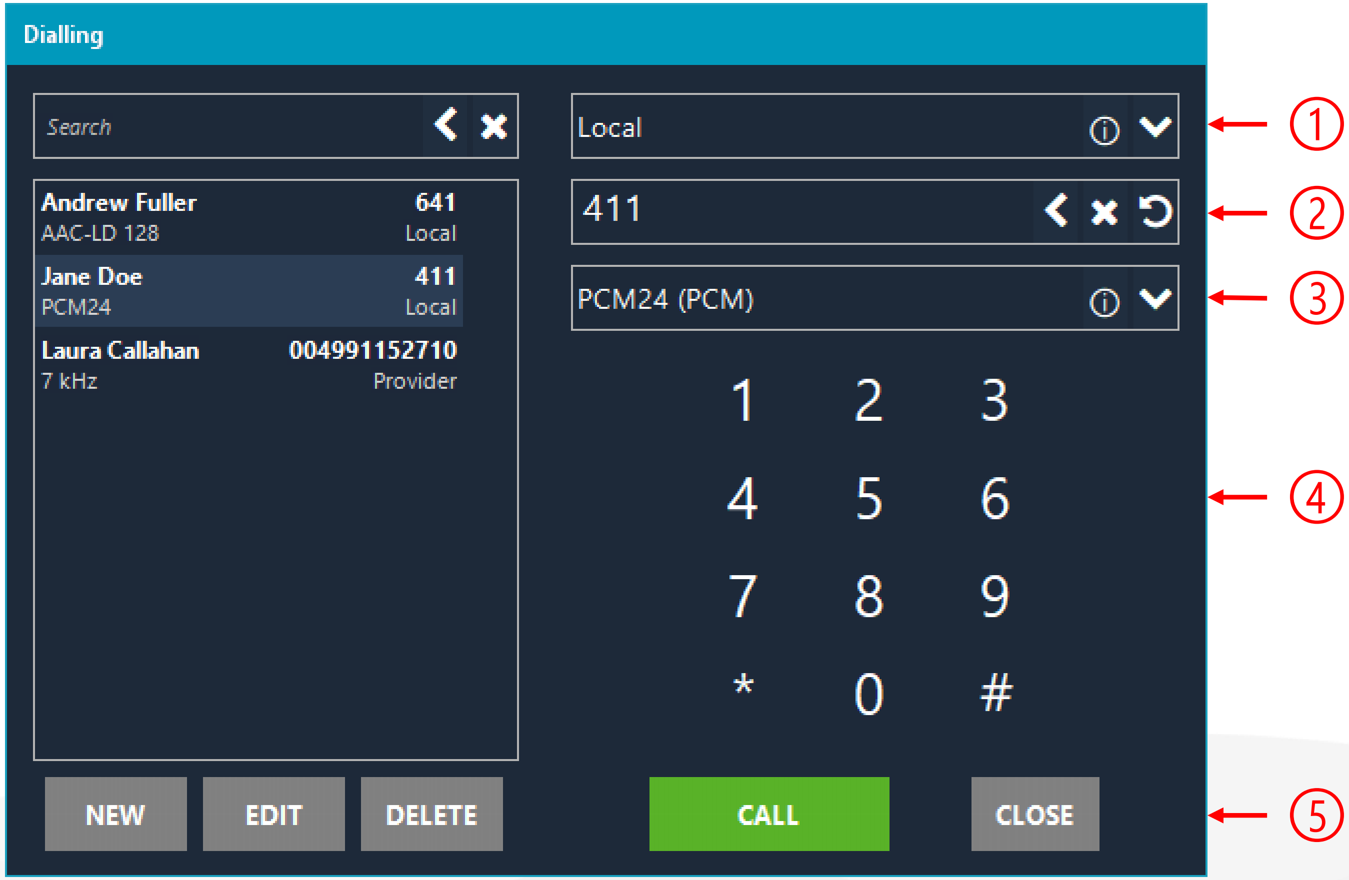
Main Panel – AoIP/SIP (5)

- Press the CALL button on the main panel to open the dialling window.

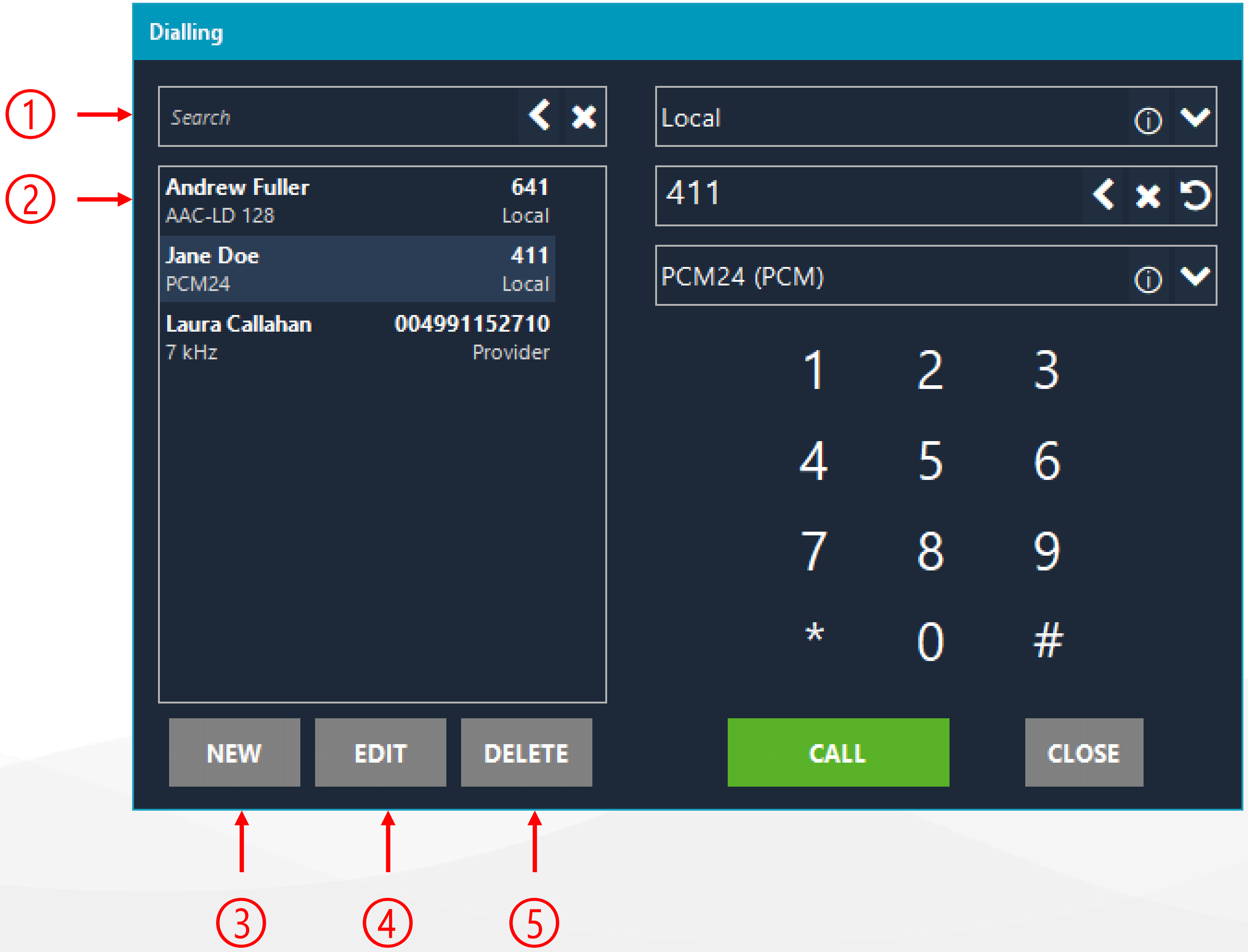
1. Select a SIP account
   - ⓘ Displays detailed SIP account information and status.
2. Enter a phone number when using the SIP account or an IP address for Direct SIP.
   - ↺ Open a list of last dialled numbers.
3. Select a transmission mode for the connection.
   - ⓘ Displays codec details of the selected transmission mode.
4. Number pad
5. CALL: Dials the number
   CLOSE: Closes the dialog



Dialling

- The internal phone book of the unit is displayed on the left of the dialling window.

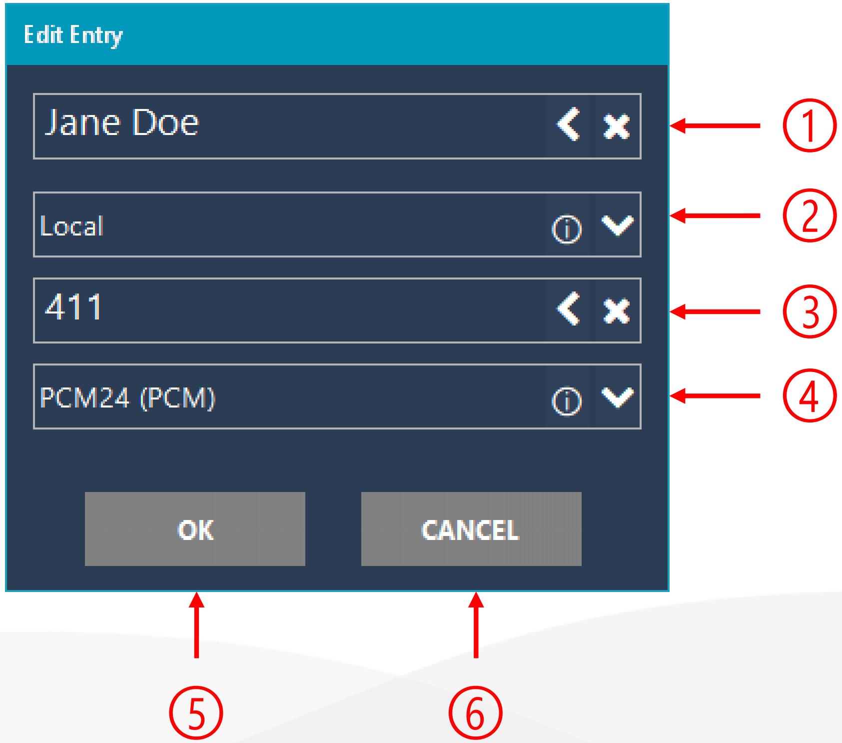
1. Search for an entry. Leave blank to display all phone book entries.
2. Search result, or list of all entries . Select an entry, to transfer it to the dialling parameters on the right.
3. Add a new phone book entry.
4. Edit the selected phone book entry.
5. Delete the selected phone book entry.



Phone book

- When adding a new entry or editing an existing entry, a new windows opens.

1. Enter the name.
2. Select a SIP account.
   - ⓘ Displays detailed SIP account information.
3. Enter the phone number or IP address.
4. Select a transmission mode.
   - ⓘ Displays codec details of the selected transmission mode.
5. Applies the changes and closes the window.
6. Discards the changes and closes the window.

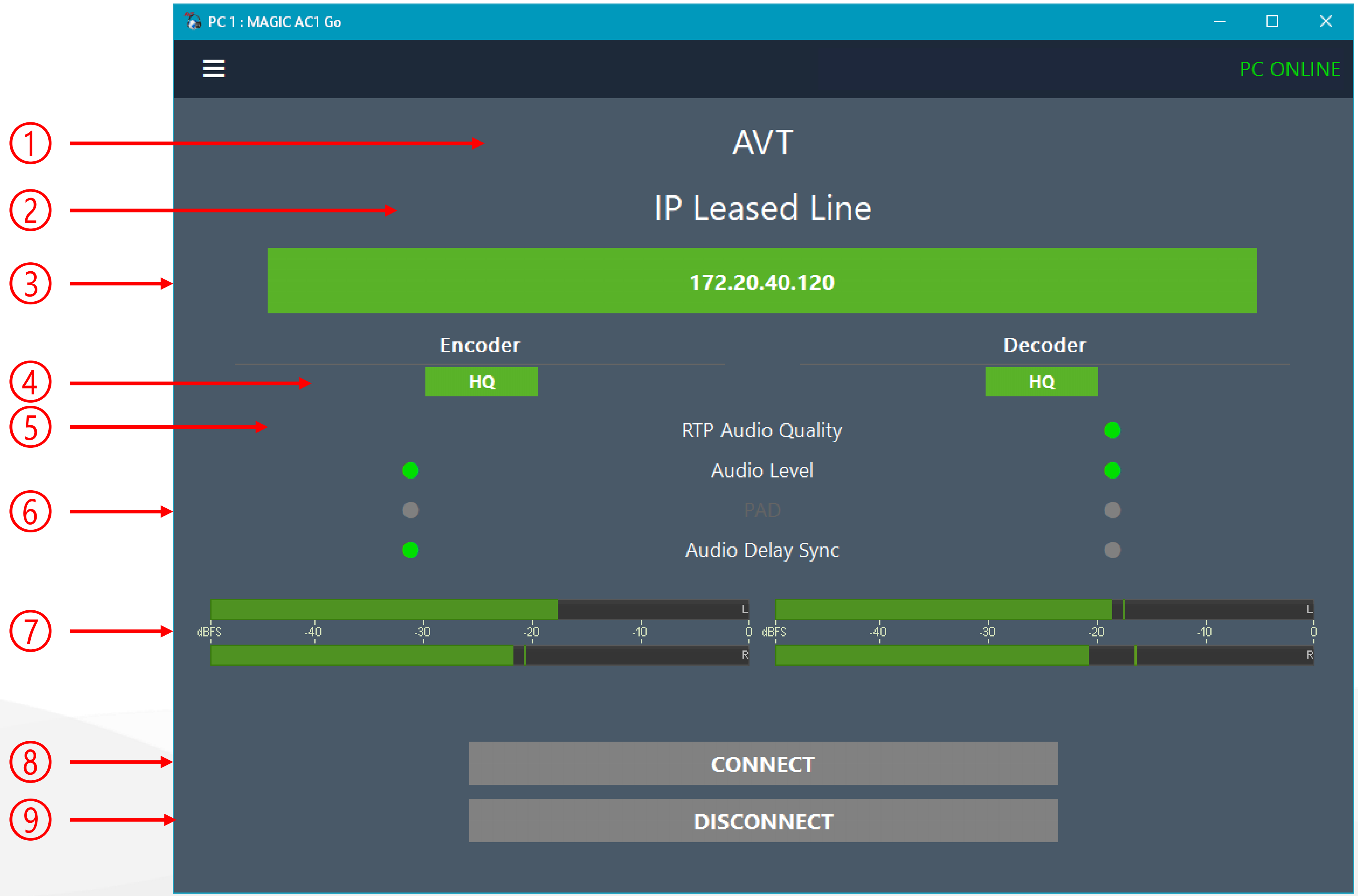


Phone book entry

MAGIC AC1 Go

Main Panel

- IP Leased Line



Audio Video Technologies

Main Panel – IP Leased Line (1)

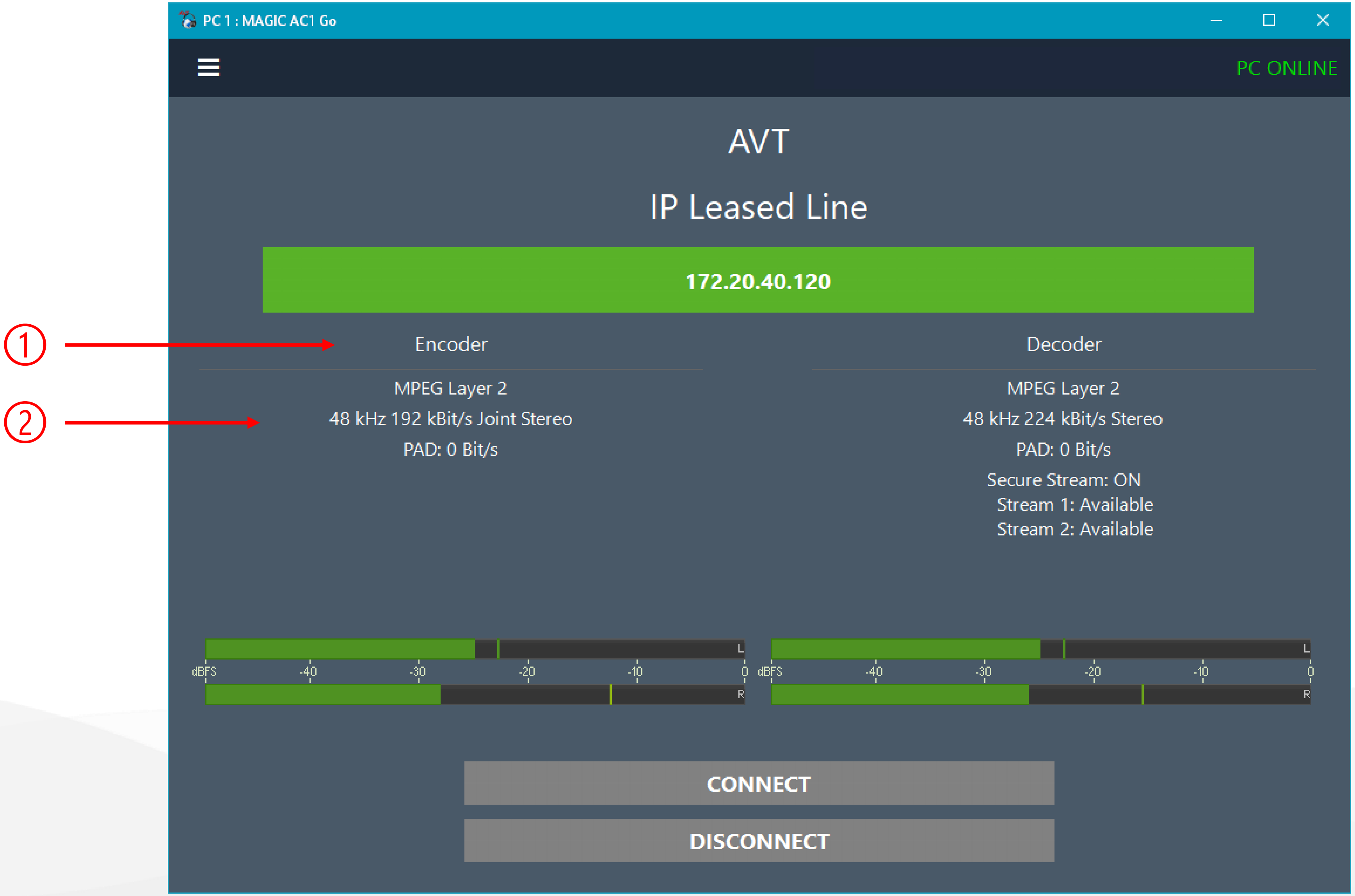
- In the IP Leased Line operating mode, the codec channels is displayed in the main window.
- The following information is displayed from top to bottom:

1. The label of the channel.
2. Operation Mode.
3. Channel Status: Shows the connection status of the channel in different colours. During a connection, the IP address of the remote party is displayed.
   - Grey: There is no connection.
   - Green: The connection is established.
4. Encoder and decoder status.
   - The audio quality is displayed via a symbol:
     - HQ: A high-quality codec is used.
     - HD: The G.722 codec is used.
     - SD: The G.711 codec is used.
   - Click on the symbol to get information about
     - codec
     - algorithm
     - data rate
     - PAD
     - Secure Streaming
5. The RTP Audio Quality shows the quality of the received data stream in the colours green, yellow and red. The threshold values of the individual levels are set on the **Alarm Signalling** configuration page.

Main Panel – IP Leased Line (2)

6. More status information:
   - Status of the Audio Level monitoring which can be configured on the **Alarm Signalling** configuration page.
   - Status of the PAD monitoring which can be configured on the **Alarm Signalling** configuration page.
   - Status of the Audio Delay Sync of Encoder and Decoder. Only available for the Decoder if IP Leased Line Extended is used.
7. The Level Meters of Encoder on the left and Decoder on the right show the current audio level in dBFS. During a connection, the level is displayed in green.
   If there is no connection the Encoder level is displayed in grey.
8. CONNECT: Press the button to establish the connection.
9. DROP: Press the button to terminate the connection.

Main Panel – IP Leased Line (3)



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Main Panel – IP Leased Line (4)

- The detailed codec information can be permanently displayed on the main panel.
- This view is enabled under **Menu > Configuration > Local Settings > Application Parameters > Show detailed Codec information in Codec Line**

1. ENCODER / DECODER: Codec information is shown for encoder and decoder separately.
2. DETAILS: Shows the following information if available:
   - Codec
   - Algorithm
   - Sampling rate
   - Bitrate
   - Mode (Mono, Stereo)
   - PAD status and bitrate
   - Secure Stream mode
   - Status of stream 1 and stream 2 if Secure Streaming is enabled.

Main Panel – IP Leased Line (5)

MAGIC AC1 Go

Operation

- Multi Control Software

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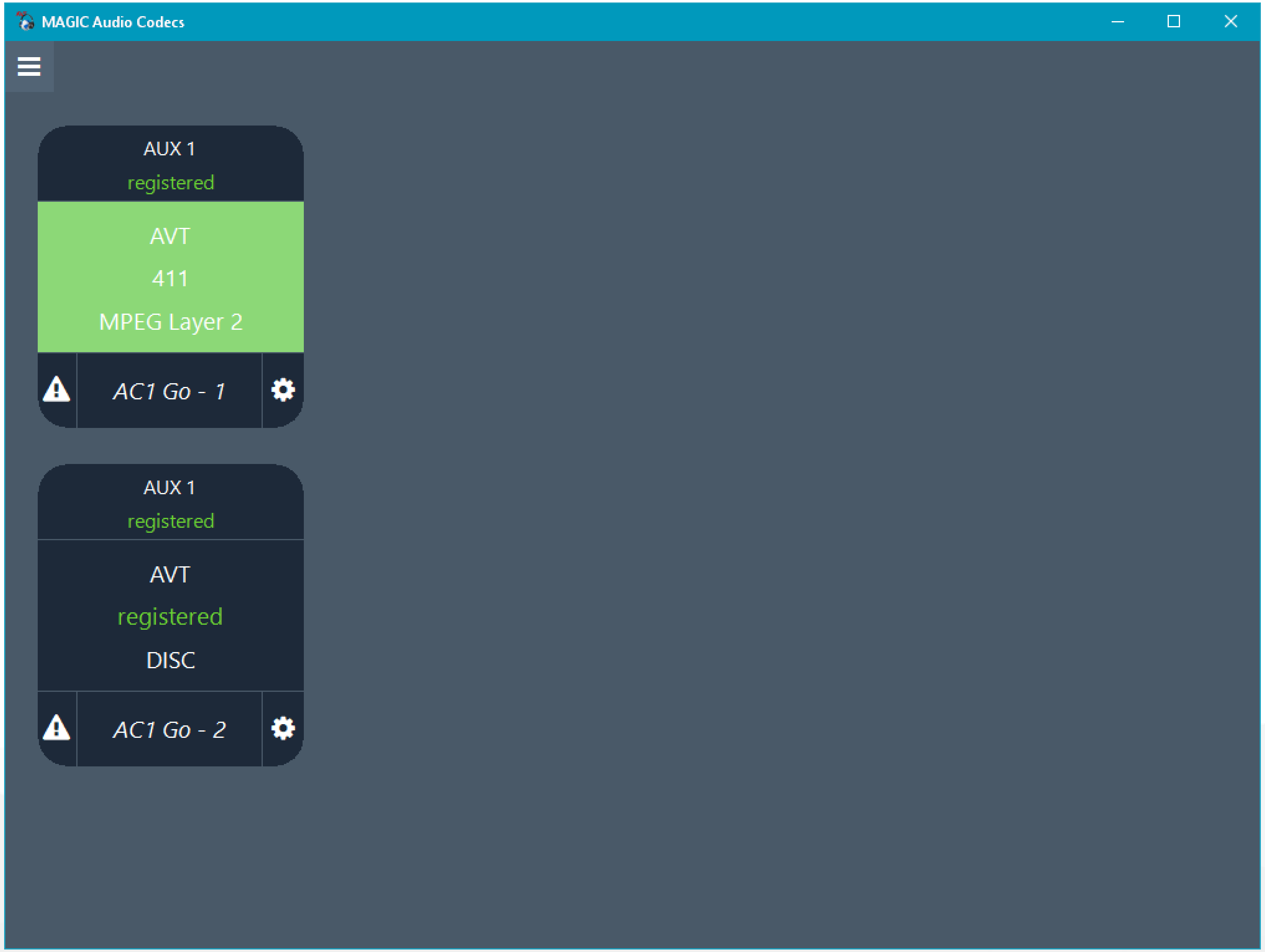

Multi Control Software (1)

- The MAGIC AC1 Go - Multi Control Software is included in the standard software installation.
- Up to 99 MAGIC AC1 Go devices can be monitored and operated.
- New devices can be added via **Menu (☰) > Configuration > Control Interface**.
  - Administrator rights are required. Right -click on the of the MAGIC AC1 Go Multi Control Software and select “Run as administrator”.
- Each device is displayed as a tile which displays information about:
  - SIP registration status
  - Connection status
  - Alarm status
- Click on a tile to open the standard operation panel of a device.
- Click on the configuration icon (⚙) to open the configuration of a device.

Multi Control Software (2)

MAGIC AC1 Go

Maintenance

- Backup / Restore
- Firmware Update
- Registration
- Software Options
- System Monitor
- SIP Monitor
- Logfile

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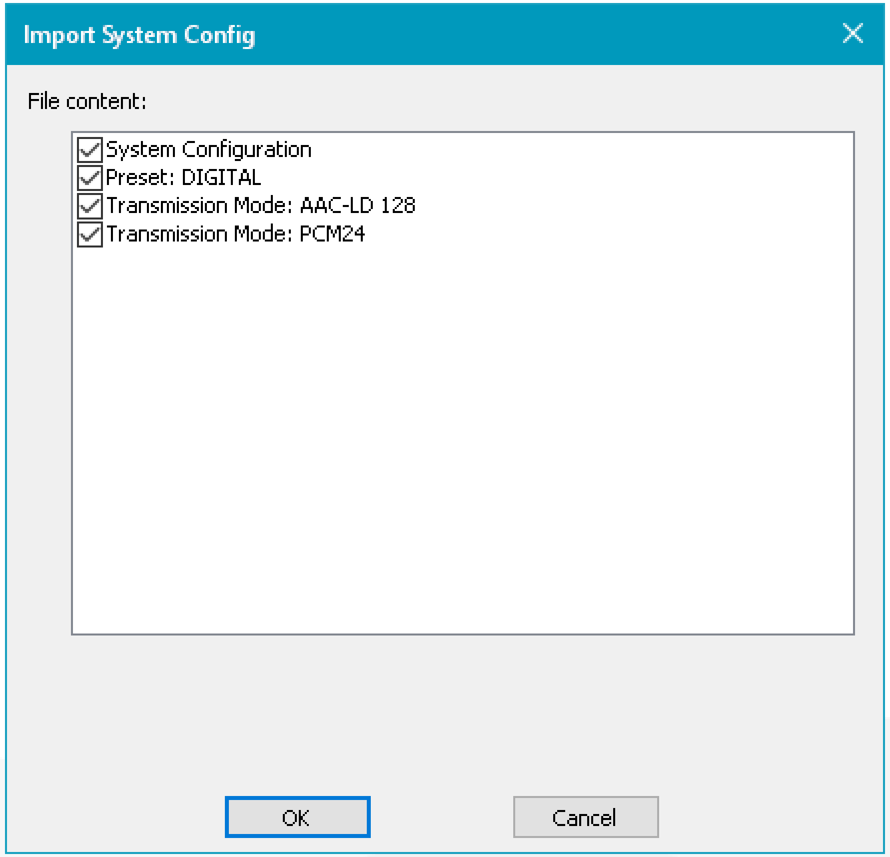
- The system configuration can be saved to a PC via **Menu > File > System Settings > Export**.
- The backup file contains:
  - The current configuration of the unit
  - Presets
  - Transmission modes
- To restore a configuration, open **Menu > File > System Settings > Import**. After selecting the backup file, you may define which parts of the backup shall be restored.
- After loading a file to be imported, a window opens in which you can select which parts of the data are to be restored.



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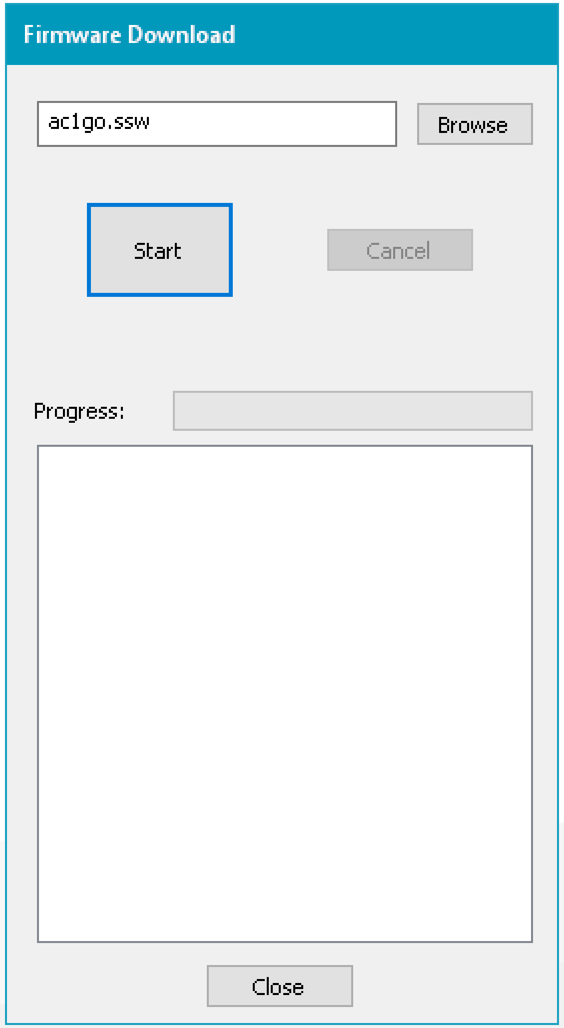
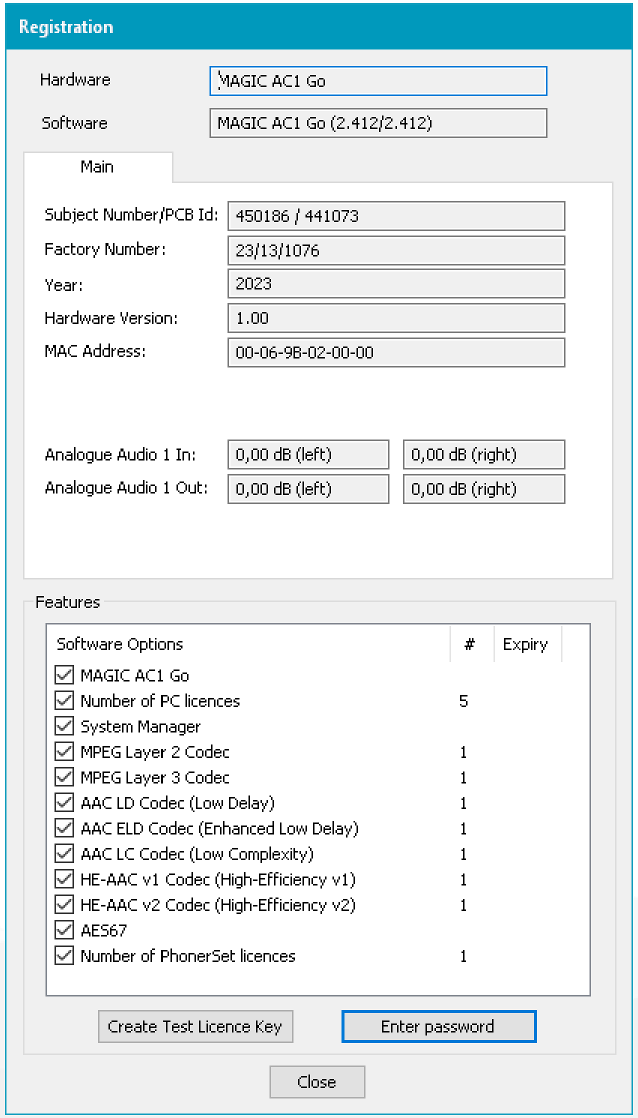
Backup / Restore

- The latest version of the AC1 Go software is freely available on our website.
- The installation package contains the PC software and the matching firmware.
- Update process:
  - Install the new PC software version.
  - Start the firmware update under **Administration - Firmware Download**.
  - Clicking **Start** automatically loads the appropriate firmware from the installation directory and starts the update.
  - The device restarts automatically to complete the update.



Firmware-Update



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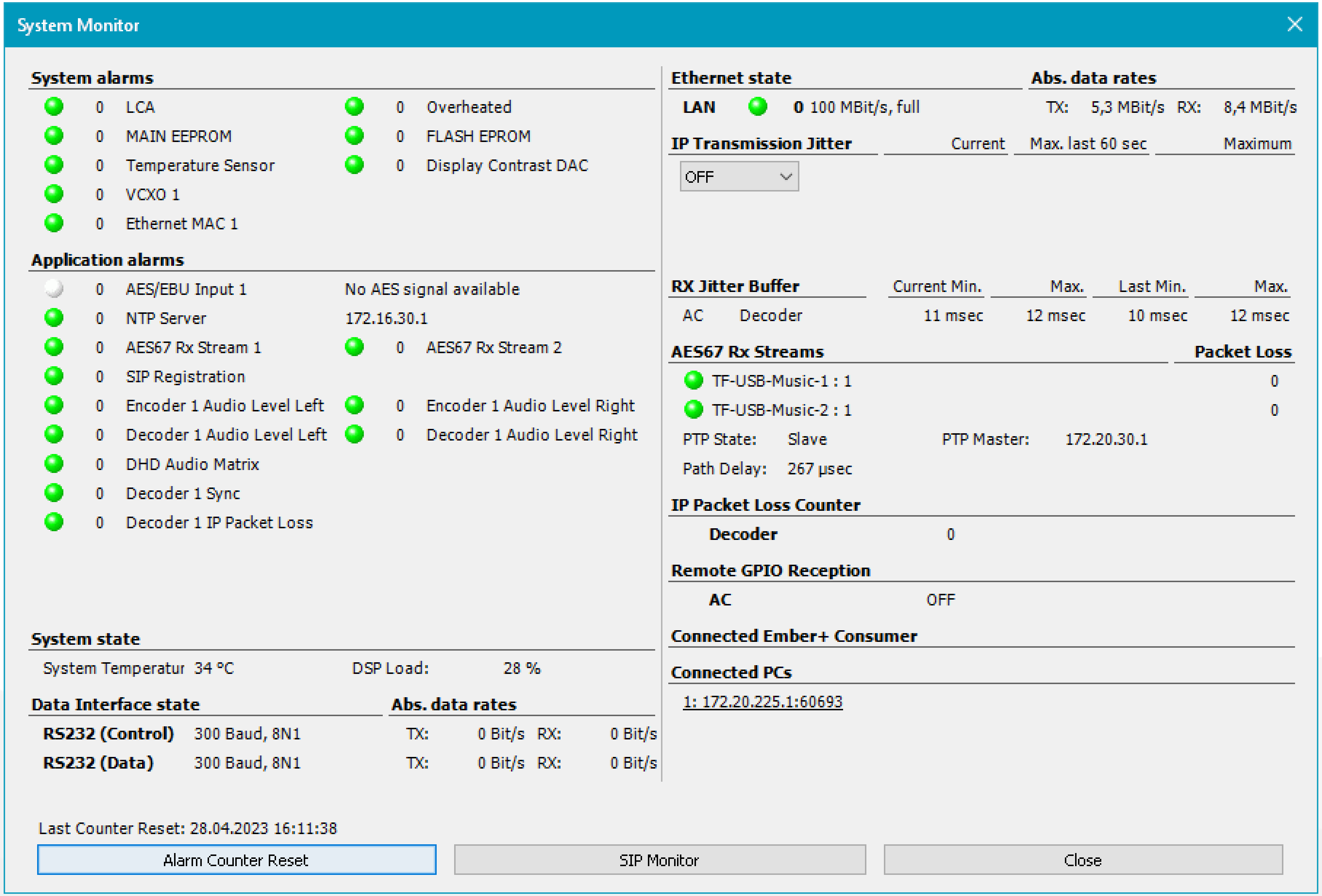
Registration and Software Options (1)

Registration and Software Options (2)

- Under ADMINISTRATION - REGISTRATION, the hardware information and software licences are displayed.
- HARDWARE: The name of the product.
- MAIN: Hardware information of the main board.
  - SUBJECT NUMBER: Our product ID.
  - FACTORY NUMBER: The serial number..
  - YEAR: The year of manufacture.
  - MAC ADDRESS: The MAC address of the LAN interface.
  - ANALOGUE AUDIO IN/OUT: The calibration values of the analogue audio interface.
- FEATURES: List of all available software options.
  - Activated options are ticked.
  - For options that can be activated more than once, the number is shown to the right.
- CREATE TEST LICENCE KEY: If you like to test a feature before buying it, create a test licence key and send it to us along with the factory number of the unit.
  - The test licence will work for a limited time.
  - The timer only counts when the unit is running.
- The test licence enables all available features.
- ENTER PASSWORD: Software options can be activated later. To do so, contact AVT with the SUBJECT NUMBER and the FACTORY NUMBER. You will receive a licence password which you can enter into the system here.



System Monitor (1)

System Monitor (2)

- Under MENU - SYSTEM MONITOR an overview of the system status is shown.
- The status of the individual components is indicated by a coloured LED:
  - ● (red): The alarm is active. There is an error.
  - ● (yellow): The alarm is active, but not relevant in the current operating mode.
  - ● (green): No alarm. The function works as expected.
- An error counter is displayed next to each alarm LED, showing how often the error occurred, since the unit was started, or the error counters were reset.
  - The counter can be reset by pressing the ALARM COUNTER RESET button.
  - The time of the last reset is displayed above the key.
- If the AoIP/SIP mode is enabled, the status of each SIP account is displayed when pressing the SIP MONITOR button.
- SYSTEM ALARMS: These alarms affect the hardware components of the device. An active alarm means that the component in question has failed temporarily or permanently. An alarm may also indicate a problem with the control of the component in the firmware.
- APPLICATION ALARMS: These alarms are used to monitor the ongoing operation of the unit.
  - AES/EBU INPUT: The unit detects whether a valid AES/EBU signal is present at the input. More information is displayed to the right of the alarm.
  - AES67 RX STREAM: Shows whether the configured AES67 streams are being received.
  - SIP REGISTRATION: Shows whether all AoIP channels could be registered with the SIP server.

- ENCODER / DECODER AUDIO LEVEL: Indicates whether the corresponding audio signal has fallen below the set threshold for the set period. This alarm must be activated in the settings.
- DHD AUDIO MATRIX: Shows whether the unit was able to establish a connection to the configured DHD core.
- EMBER+ CONSUMER: Shows whether the unit was able to establish a connection to the configured Ember+ provider.
- DECODER SYNC: Indicates whether the decoder can decode the incoming data stream.
- DECODER IP PACKET LOSS: Shows whether the number of packet losses exceeds the threshold for BAD stream quality as configured under **Alarm Signalling**.
- SYSTEM STATE: General information about the system state is displayed here.
  - SYSTEM TEMPERATURE: Displays the temperature measured in the enclosure. Above 57°C, the OVERHEATED alarm is triggered. The temperature should always be kept below 50°C. To do this, it is necessary to leave at least one unit of free space above the device in the rack so that the heat can escape via the ventilation grille in the enclosure lid.
- DSP LOAD: Shows the current load of the DSP in percent.
- DATA INTERFACE STATE: Shows the status of the RS232 data interfaces.
  - ABS. DATA RATES: Shows the current data rates of the RS232 interfaces in the transmitting direction (TX) and receiving direction (RX).

System Monitor (3)

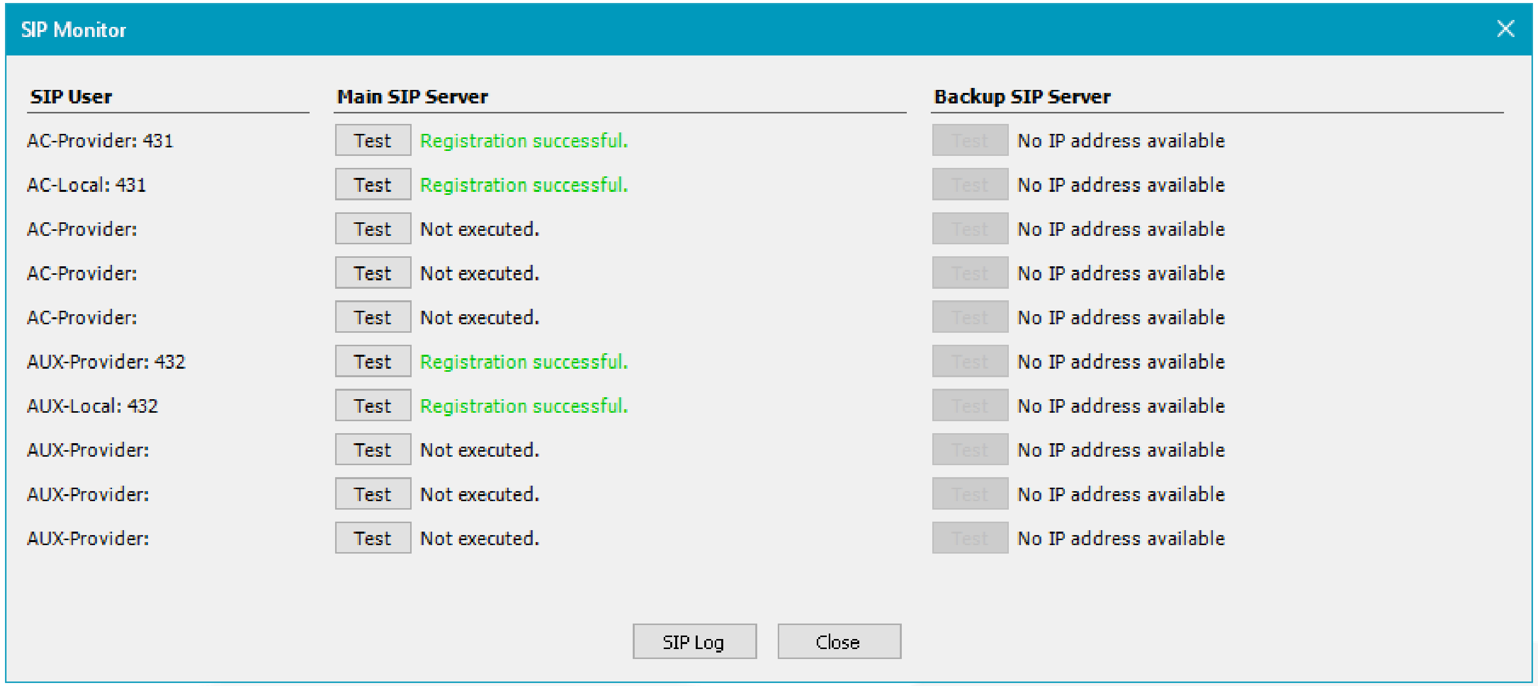
- ETHERNET STATE: Shows the status of the LAN interface.
  - LED = Link Status.
    - ● (green): A network cable is connected and the connection to the switch is established.
    - ● (grey): No network cable is connected or the connection to the switch is disturbed.
- Operating mode of the LAN interface:
  - An interface used for audio transmission must be set to 100 Mbps, full to avoid drop-outs.
- ABS. DATA RATES: Shows the current data rates of the LAN interface in the transmitting direction (TX) and receiving direction (RX).
- IP TRANSMISSION JITTER: Provides a jitter measurement for the stream of the codec channel.
  - Select the Decoder to display the measurement.
  - CURRENT: Current jitter value. Updated every two seconds.
  - MAX. LAST 60 SEC: Highest jitter value in the last 60 seconds.
  - MAXIMUM: Highest jitter value since the start of the measurement. This value can be reset via the MAX. JITTER RESET key.
- RX JITTER BUFFER: Shows the fill levels of the receive buffer for the codec channel.
  High fill levels can be due to high jitter, active Secure Streaming or active Audio Delay Sync.
  - CURRENT MIN: Shows the minimum fill level in the last two seconds.
  - MAX: Shows the maximum fill level in the last two seconds.
  - LAST MIN: Shows the minimum fill level in the last 60 seconds.
  - MAX: Shows the maximum fill level in the last 60 seconds.

System Monitor (4)

System Monitor (5)

- AES67 RX STREAM: Shows the status of the configured RX AES67 streams.
  - LED = reception status:
    - 🟢 (green): Audio data is being received.
    - 🔴 (red): No audio data is being received.
  - PACKET LOSS: Shows the number of packet losses of the stream since the beginning of the measurement.
  - PTP STATE: Shows the state of the PTP clock synchronisation. The unit can only operate as a slave.
  - MASTER: Shows the IP address of the PTP master.
  - PATH DELAY: Shows the time delay of the connection to the PTP master.
- IP PACKET LOSS COUNTER: Shows the number of packet losses of the receive streams of the codec channel since the device was started or the alarm counters were reset.
- REMOTE GPIO RECEPTION: Shows whether the transmission of GPIO states is activated.
- CONNECTED EMBER+ CONSUMER: Lists all Ember+ consumers that are connected to the Ember+ provider of the unit.
- CONNECTED PCS: Lists all PCs that are connected to the unit.
- CONFIGURATION DIALOGUE OPEN AT: The system settings cannot be opened on more than one PC at the same time. This line shows on which PC the configuration is open.

- The SIP registration status of all accounts is displayed under **System Monitor > SIP Monitor**.
- TEST: Start registration for that account.
- The registration status is displayed with troubleshooting information in case of an error.



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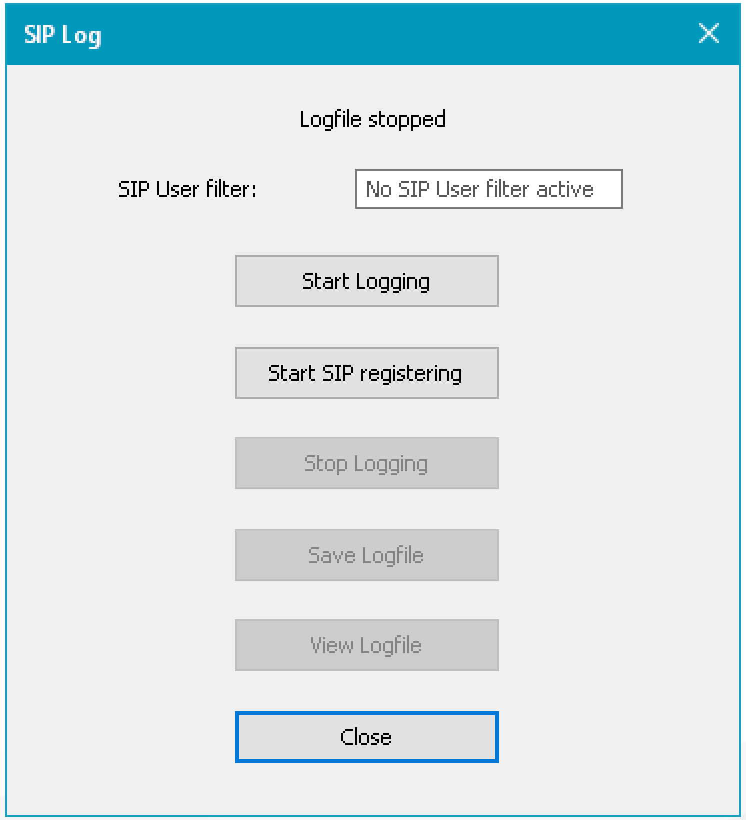
SIP Monitor



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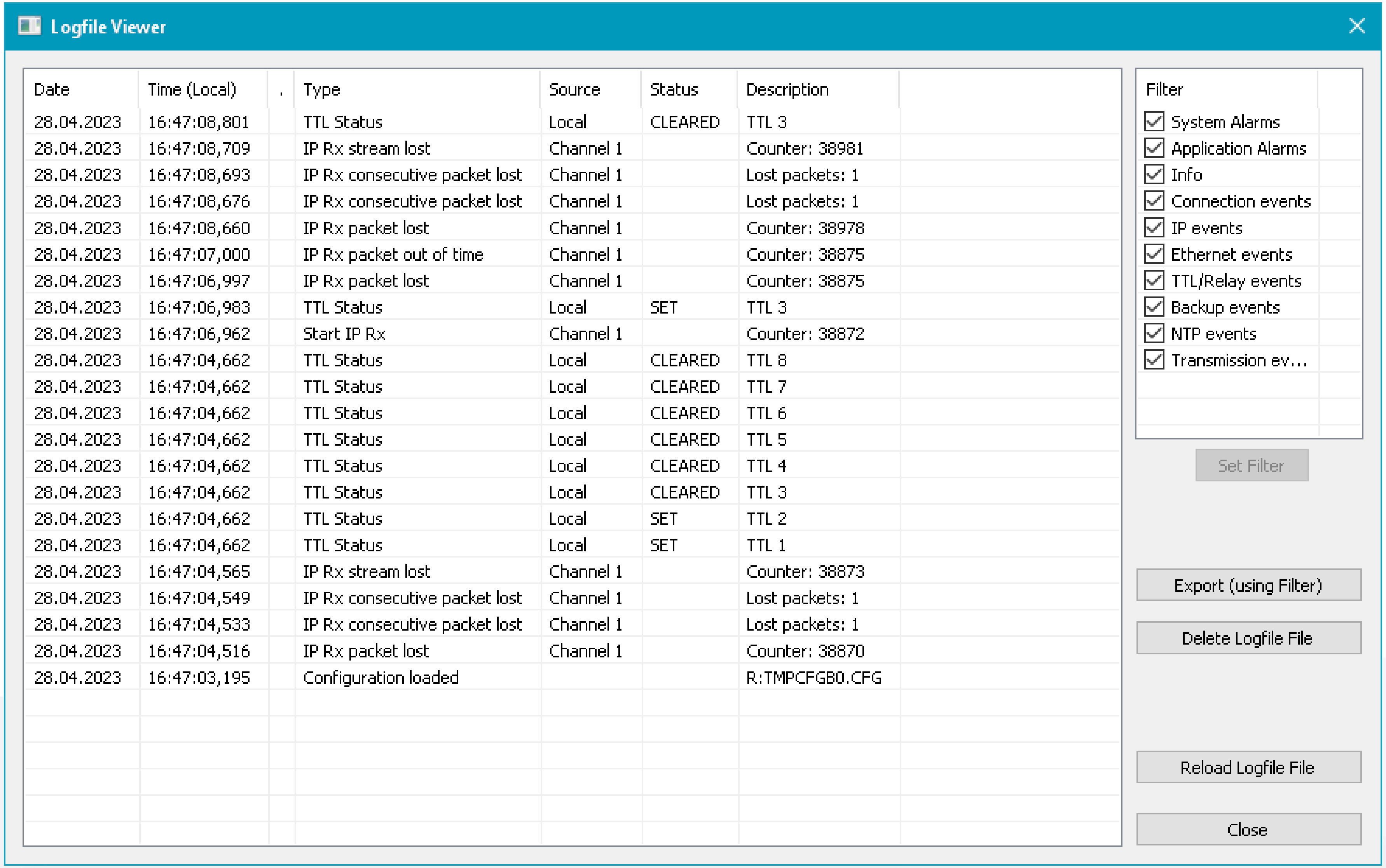
SIP Log (1)

SIP Log (2)

- The SIP communication of the unit can be recorded under **System Monitor > SIP Monitor > SIP Log**.
- The logfile is stored in the internal flash memory of the device. Since the memory space is limited, the logging should not be active for more than one hour.
- SIP USER FILTER: Optionally, enter a SIP user name to log only the SIP communication of one SIP account.
- START LOGGING: Click to start recording the SIP messages.
- START SIP REGISTERING: This can be used to record the SIP registration process. Click while the logging is active.
- STOP LOGGING: Click to stop recording the SIP messages.
- VIEW LOGFILE: Open the logfile in a text editor on the PC.
- SAVE LOGFILE: Click to save the logfile on the PC.



Date	Time (Local)	.	Type	Source	Status	Description
28.04.2023	16:47:08,801		TTL Status	Local	CLEARED	TTL 3
28.04.2023	16:47:08,709		IP Rx stream lost	Channel 1		Counter: 38981
28.04.2023	16:47:08,693		IP Rx consecutive packet lost	Channel 1		Lost packets: 1
28.04.2023	16:47:08,676		IP Rx consecutive packet lost	Channel 1		Lost packets: 1
28.04.2023	16:47:08,660		IP Rx packet lost	Channel 1		Counter: 38978
28.04.2023	16:47:07,000		IP Rx packet out of time	Channel 1		Counter: 38875
28.04.2023	16:47:06,997		IP Rx packet lost	Channel 1		Counter: 38875
28.04.2023	16:47:06,983		TTL Status	Local	SET	TTL 3
28.04.2023	16:47:06,962		Start IP Rx	Channel 1		Counter: 38872
28.04.2023	16:47:04,662		TTL Status	Local	CLEARED	TTL 8
28.04.2023	16:47:04,662		TTL Status	Local	CLEARED	TTL 7
28.04.2023	16:47:04,662		TTL Status	Local	CLEARED	TTL 6
28.04.2023	16:47:04,662		TTL Status	Local	CLEARED	TTL 5
28.04.2023	16:47:04,662		TTL Status	Local	CLEARED	TTL 4
28.04.2023	16:47:04,662		TTL Status	Local	CLEARED	TTL 3
28.04.2023	16:47:04,662		TTL Status	Local	SET	TTL 2
28.04.2023	16:47:04,662		TTL Status	Local	SET	TTL 1
28.04.2023	16:47:04,565		IP Rx stream lost	Channel 1		Counter: 38873
28.04.2023	16:47:04,549		IP Rx consecutive packet lost	Channel 1		Lost packets: 1
28.04.2023	16:47:04,533		IP Rx consecutive packet lost	Channel 1		Lost packets: 1
28.04.2023	16:47:04,516		IP Rx packet lost	Channel 1		Counter: 38870
28.04.2023	16:47:03,195		Configuration loaded			R:TMPCFGB0.CFG

Logfile Viewer (1)

Logfile Viewer (2)

- Status changes of the unit can be recorded in an internal log file.
- The log file is configured on the GENERAL settings page.
- In order to be able to evaluate the log file, the Logfile Viewer, which is part of the PC software, can be used.
- Open the Logfile Viewer via **Menu > Logfile Viewer**.
- The individual events are displayed line by line in the table on the left.
  - DATE: The date on which the event was recorded. The correct date is only displayed if the unit has a connection to an NTP server.
  - TIME (LOCAL): The time at which the event was recorded. The correct time is only displayed if the unit has a connection to an NTP server. The time is converted to the time zone of the PC by the PC software.
  - DURATION [d] hh:mm:ss,msec: Duration for which the event was active in hours : minutes : seconds , milliseconds. This information is only available if it is relevant.
  - TYPE: A short information about the type of event.
  - SOURCE: Source of the event, e.g., Channel, LAN.
  - STATUS: Shows the new state of the component.
    - OK: An alarm is reset.
    - ERROR: An alarm has occurred.
    - SET / CLEARED: TTL status
    - LAN mode (speed / duplex)
  - DESCRIPTION: A short text describing the event in more detail or providing additional information.
- FILTER: The list of events can be filtered. Check the boxes of the categories which should be displayed.
  - SET FILTER: The new filter settings are not applied to the table until the SET FILTER key has been pressed.

- EXPORT (USING FILTER): Exports the content of the list as it is displayed to a file on the PC.
    - The file can be stored as text file (*.log) or XML file (*.xml) by selecting the respective type in the Save As Dialog.
- DELETE LOGFILE: Deletes the log file on the unit. A confirmation message is displayed before the file is deleted.
- RELOAD LOGFILE FILE: Loads the latest state of the logfile.

Logfile Viewer (3)

MAGIC AC1 Go

Front Display

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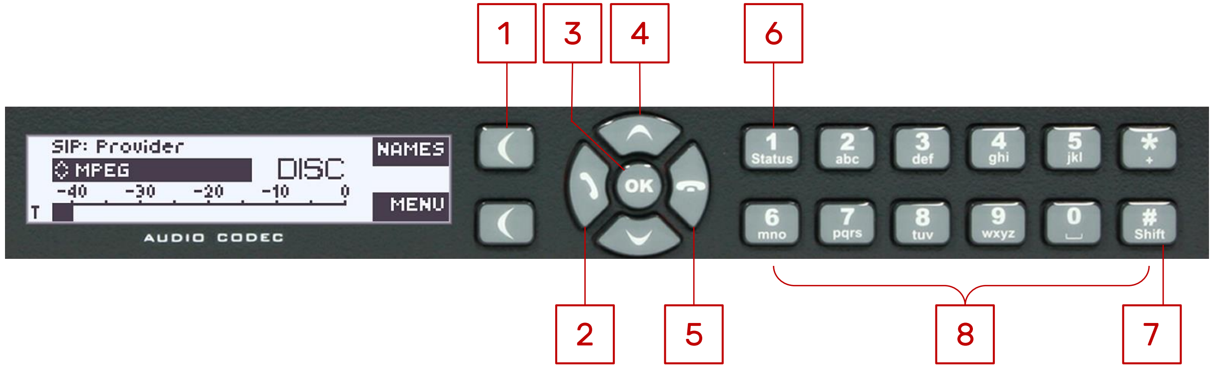
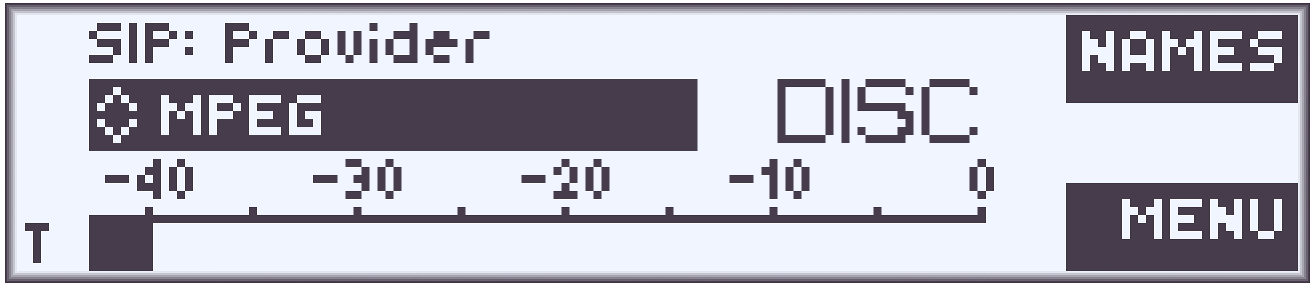
www.avt-nbg.de

- 1: Softkeys
  - The function is displayed to the left of the buttons on the screen.
- 2: Call
- 3: OK
- 4: Arrow Keys
  - Navigating in menus and lists.
  - Changing audio levels.
- 5: Hang up
  - Disconnect the connection.
  - Go directly to the start screen.
- 6: Status
  - Detailed connection status
- 7: ‘#’
  - SIP server selection
- 8: Alphanumeric keypad
  - Enter a phone number.
  - Enter text.



Front panel operation

- The top line shows the operating mode and in case of “SIP” the selected SIP server.
    - The '#' key changes the SIP server.
- The second line shows the current transmission mode. This can be changed with the arrow keys.
- The softkey "Names" opens the internal telephone book.
- The "Menu" softkey opens the system configuration menu.
- Enter a telephone number via the numeric keypad to switch to the dialling screen.
- Press "OK" to go to the Connection screen.



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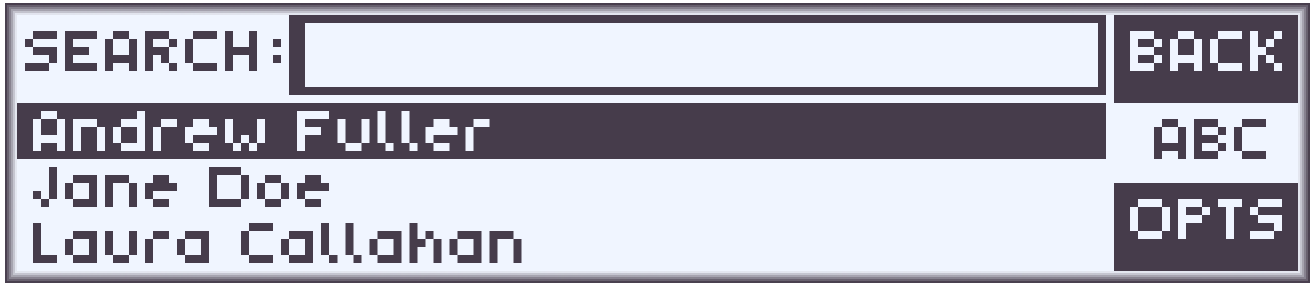
Front - Start screen

- All entries of the internal telephone book are displayed in alphabetical order.
- Enter a character string via the numeric keypad to filter the list.
- Display, add, edit and delete entries via the "Opt." softkey.
- Press the “CALL" key to dial the number of the selected contact.



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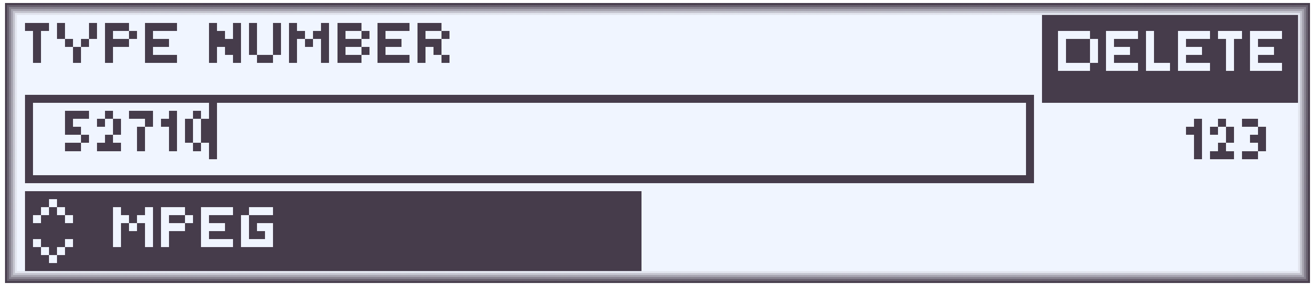
Front – Phone Book

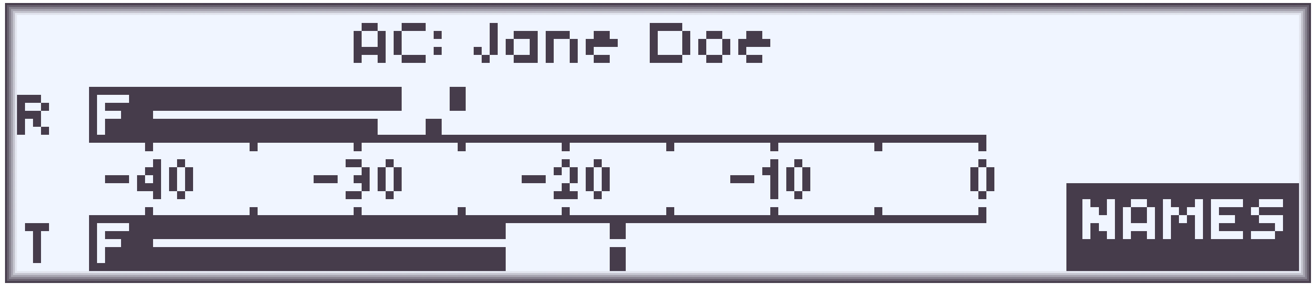
- The SIP server cannot be changed on the dialling screen. This must be done beforehand on the start screen.
- Starting to type a number on the start screen, opens the dialling screen.
- Select one of the configured transmission modes via the arrow keys.
- Press CALL to establish the connection.



Front – Dialling a number

Front - Connection Status

- During a connection, the audio levels for transmit (T) and receive (R) are displayed.
- Press and hold "OK" to switch between codec channel and AUX channel.
- Press the "1/Status“ key to display the codec status.
- Step through the status screens via the lower softkey (>>).
  - Encoder status
  - PAD TX / RX
  - Alarms
  - Decoder IP statistic
  - LAN status



MAGIC AC1 Go

Interfaces

- TTL
- Data / Control

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TTL / Relay Interface

- Connector: SUB-D (9-pin / female)

Pin	Signal	Electrical Characteristics
1	TTL 1 IN/OUT	Capacity of the TTL inputs/outputs: Maximum voltage: 3.3 V Maximum current: 10mA
2	TTL 2 IN/OUT	
3	TTL 3 IN/OUT	
4	TTL 4 IN/OUT	
5	GND	
6	Relay 1a	Capacity of the relays: Maximum voltage: 48V Maximum current: 200mA
7	Relay 1b	
8	Relay 2a	
9	Relay 2b	

RS232 1/2 Interface

- Connector: SUB-D (9-pin / female)

Pin	Signal		Electrical Characteristics
1		not used	Type: DCE Level: V.24 Data rate: 38400 Baud Range: max. 15 m Protocol: 1 Start bit, 8 Data bits, 1 Stop bit DCE = Data Communication Equipment, to connect a PC a 1:1 cable is required
2	TXD 1	OUT (PC control)	
3	RXD 1	IN (PC control)	
4		not used	
5	GND	GND	
6		not used	
7	RXD 2	IN (Data)	
8	TXD 2	OUT (Data)	
9		not used	

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