

MAGIC THipPro Intercom

Configuration Guide

Version 3.070 (18 September 2019)
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MAGIC THipPro

Intercom

Hardware,
Interfaces,
Options



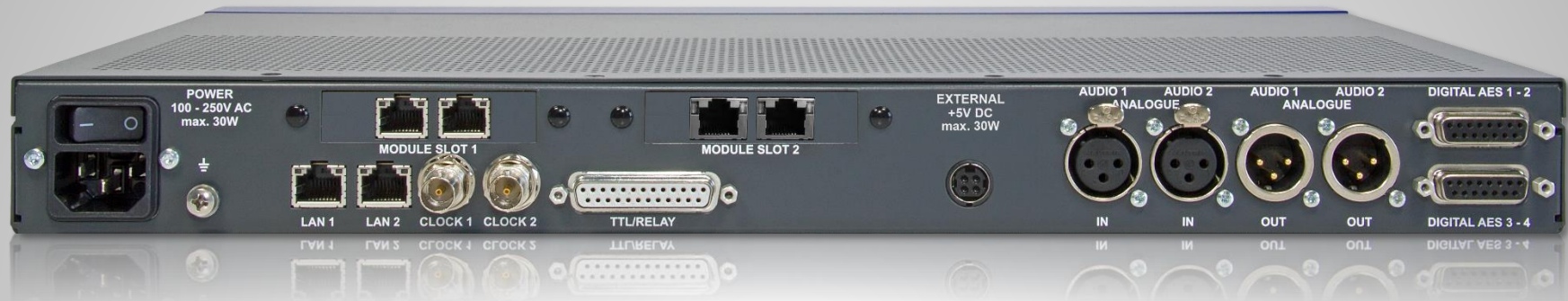
- Five Status-LEDs
 - POWER, SYNC, ALARM, INFO 1, INFO 2
- Illuminated graphic display with 160 x 32 pixels & front keypad
 - For basic settings and status displays, but not for operation.

Front View



- Two independent LAN interfaces for control and VoIP applications.
 - Word clock input/output
 - Programmable GPIO interface
 - 8 x TTL input/output
 - 8 x Relays (8 x normally open contact)
 - Redundant Power Supply (optional requires hardware version 4.0)
- 2 x analogue Audio input/output
 - Je 2 x XLR (female/male)
 - 4 x AES Audio input/output
 - 8 x digital audio cable (2 x Sub-D 15-polig)
 - 2 x Slot for DANTE/LAN modules

Rear view without modules



- Slot 1: LAN 3/4
 - 2 additional Ethernet interfaces
 - Flexible use e.g. for control, VoIP, Ember+, DHD Set Logic, SNMP
- Slot 2: DANTE
 - 32 channels
 - 2 Ethernet interfaces
 - The classic audio interfaces can still be used.

Available modules

- 19" housing x 1 U
- Without fan for silent operation
- Low power consumption of typ. 15 W
- Optional redundant power supply (requires Hardware Version 4.0)
- Two system variants
 - 8 caller lines, expandable to 16 caller lines
 - 16 caller lines
- 12 audio interfaces
 - 2 x analogue input/output
 - 8 x digital input/output (4 x AES)
 - 2 x handset/headset
- Digital signal processing for each line
 - Echo canceller to eliminate echoes with a runtime of up to 120 ms (Echo Tail Time)
 - AGC Automatic Gain Control
 - Expander for noise suppression
 - Send Level Booster
- Programmable GPIO for mixing console control and external

signalling

- DHD Set Logic
 - 96 inputs/outputs
- Ember+ provider
 - 96 inputs and 96 outputs
 - Keypad, phone number, name
- Ember+ Consumer
 - Connection to 2 Ember+ providers
 - 20 functions per consumer
- DTMF generator for the transmission of DTMF tones
- VLAN support
- QoS support
- integrated SIP monitor
- Audio test panel with signal generator

Features

- *MAGIC THipPro Intercom PC Software*

- Single user license (1 licence included, optionally up to 20 per system).
- Up to 10 systems with 160 lines can be displayed simultaneously.
- Grouping of the systems on up to 10 pages.
- Resolution independent design.
- Supported operating systems
 - Windows 7, 8.1, 10 (32/64 Bit)
- Free colour design

- 6 workplace configurations
- 8 or 16 VoIP lines
- HD Voice
 - Significantly better voice quality with G.722 codec.
- Pretalk via USB Headset
- AES67
 - Audio over IP
 - 8 Channels (in addition to hardware audio interfaces).

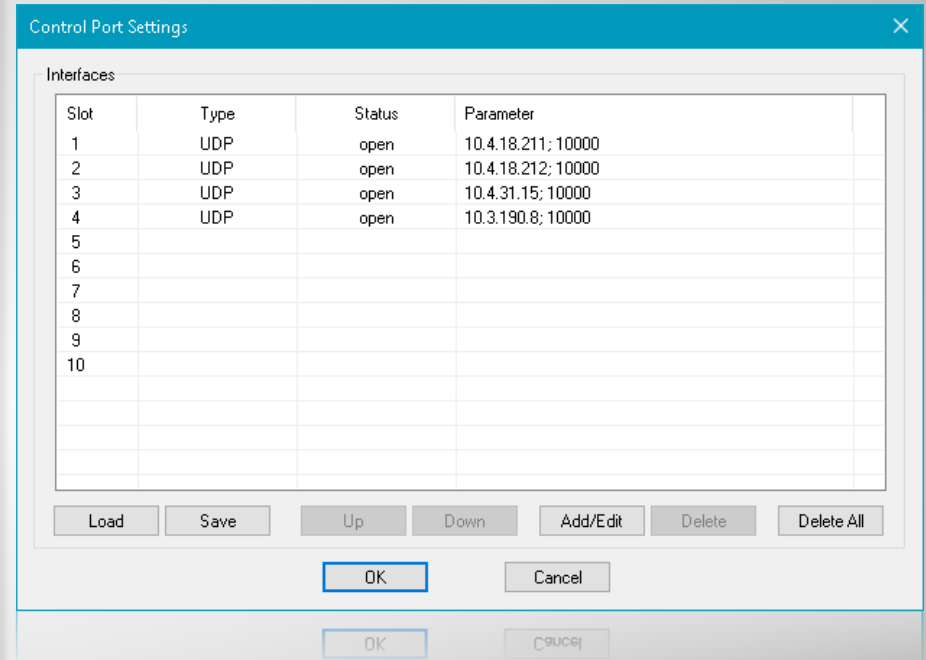
Available options

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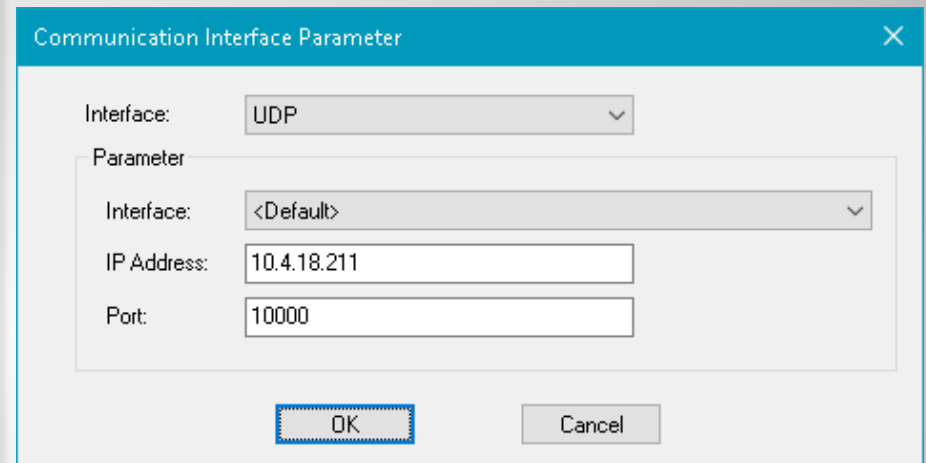
PC Software Operation

- Install the *MAGIC THipPro Intercom Software* from the USB stick with administrator rights on your PC and then start the software with administrator rights.
 - From Windows 7 and higher via the context menu "Run as administrator", even if you are currently logged on as administrator.
- Under MENU → CONFIGURATION → CONTROL INTERFACE, enter the systems to be controlled.
- Add systems by double-clicking a line or use the ADD/EDIT key.
- Remove a system using DELETE.
- DELETE ALL removes all entries from the list.
- The order of the entries can be changed via UP and DOWN. The sequence determines the order of the systems in the main window.



Starting the PC Software

- **INTERFACE:** The connection can only be established via the LAN interface (UDP).
- **PARAMETERS:**
 - **INTERFACE:** If the PC has several network interfaces, specify here which of them is to be used.
 - **IP ADDRESS:** IP address of the *THipPro Intercom* device.
 - **PORT:** UDP port of the control connection on the *THipPro Intercom* device. The default value is 10 000.
- Press the right telephone key on the device twice to display the currently assigned IP address of the system,.
- The network settings can be adjusted on the device under MENU → SYSTEM SETTINGS → LAN SETTINGS.



Communication Interface Parameter

Interface: UDP

Parameter

Interface: <Default>

IP Address: 10.4.18.211

Port: 10000

OK Cancel

Connection parameters

- The software can connect to up to 10 *THipPro Intercom* systems simultaneously.
- The *THipPro Intercom* systems can be distributed over up to 10 pages.
- The pages are selected via the tabs (❶) below the menu bar.
- Depending on the screen resolution, up to 10 *THipPro Intercom* systems with 160 lines can be displayed simultaneously on one page.
- On the right-hand side you will find:
 - (❷) Telephone book, Redial list and the input field for the telephone number.
 - (❸) Keys for the Preallocation of lines.
 - (❹) Quick dial keys and call history list. Click on the list title to switch between the two views.
 - (❺) The WORKPLACE SELECTION key to switch the view to an alternative workplace configuration.



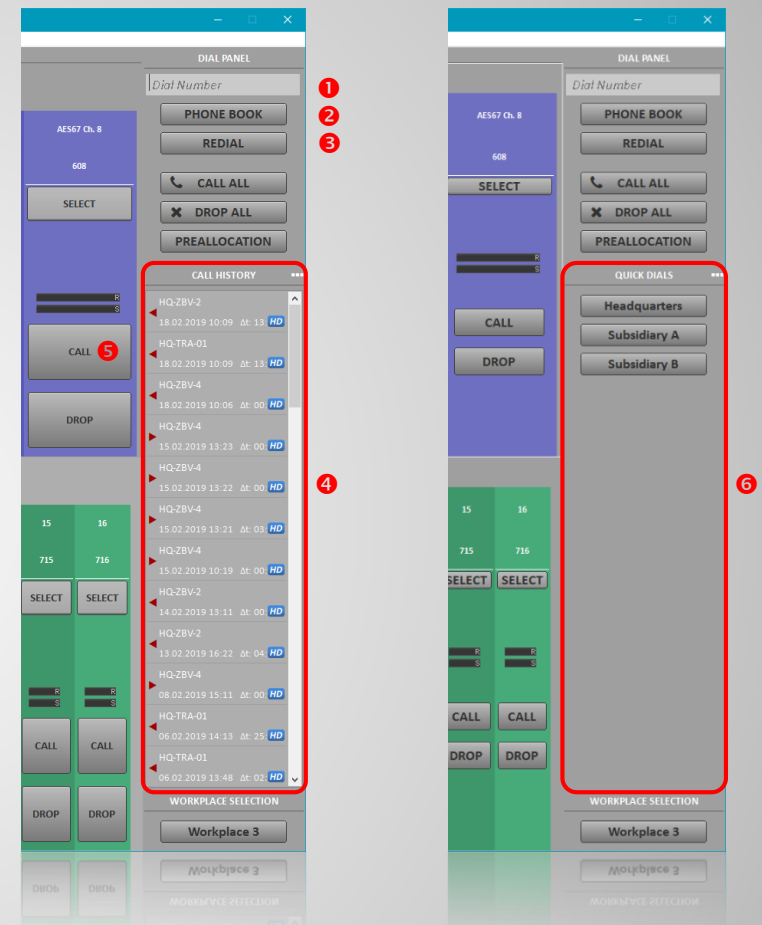
Main Window

- The title bar can be hidden using LOCAL SETTINGS → APPLICATION PARAMETERS.
- Clicking the menu button (❶) displays the menu as a sidebar.
- Without the title bar, the window can no longer be moved with the mouse.
- To temporarily show the title bar, press the SCROLL LOCK key on the keyboard.



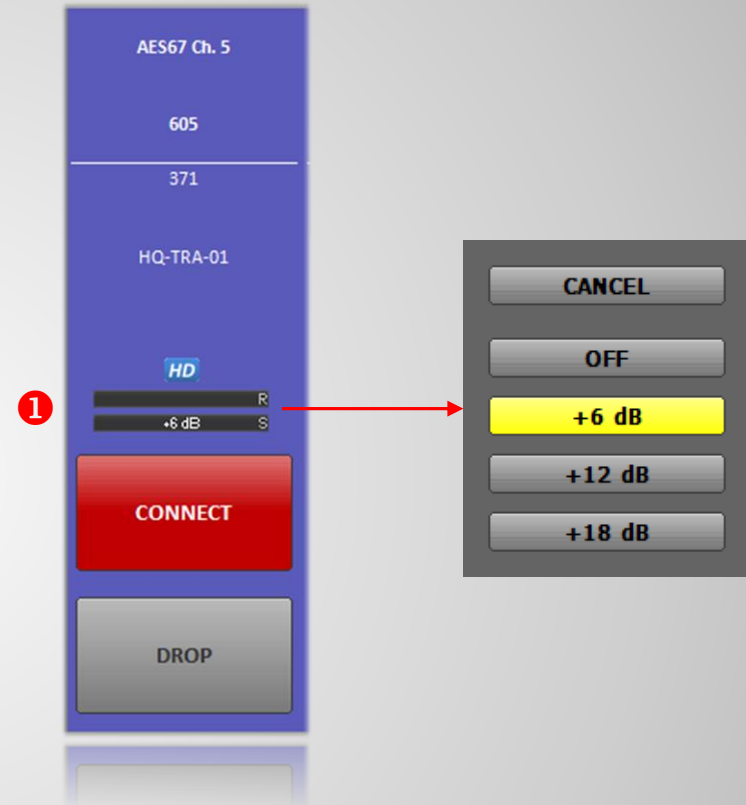
Borderless Main Window

- A telephone connection can be established in various ways:
 - Enter the phone number in the phone number field (❶) and press the CALL key (❺) on the desired line.
 - Select a phone book entry (❷). The phone number is then transferred to the phone number field. Press the CALL key (❺) on the desired line.
 - Select a redial number (❸) and press the CALL key (❺) on the desired line.
 - Select a number from the CALL HISTORY list (❹) and press the CALL key (❺) on the desired line.
 - Press the CALL key (❺) on the desired line. A dialling window opens where you can select a directory entry or enter a telephone number.
 - Select a quick dial key (❻) and press the CALL key (❺) on the desired line.



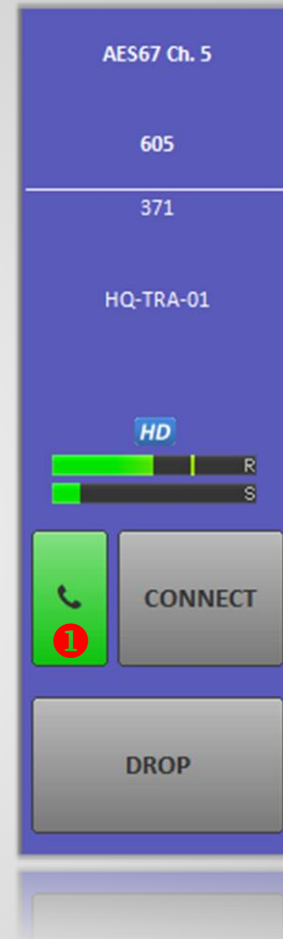
Call set-up

- A click on the level indicator (①) opens the send level booster.
 - The level of the outgoing signal can be increased in three steps.
 - The individual levels are configurable (SYSTEM CONFIGURATION → SIGNAL PROCESSING).
 - The currently set gain is highlighted in yellow.
- The currently set gain is permanently displayed in the level meter.
- Whether the gain is reset or kept after a connection is terminated is configurable.



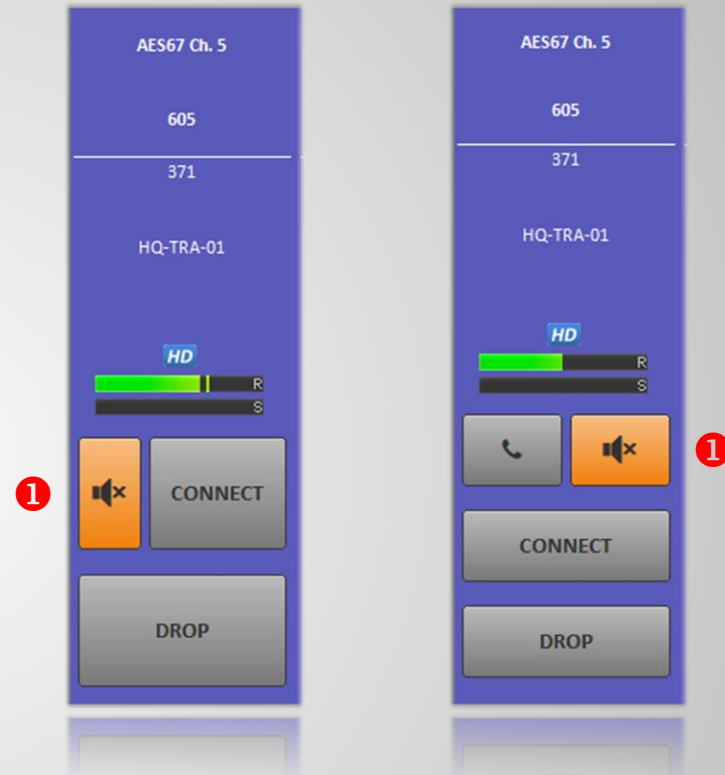
Level booster

- A key (❶) for local communication with the subscriber can be displayed for all lines (activate under CONFIGURATION → LOCAL SETTINGS → PRETALK STREAMING).
- This pretalk is conducted via the PC's sound card.
- The audio data streams are transferred between the PC and the *THipPro Intercom System* via LAN.



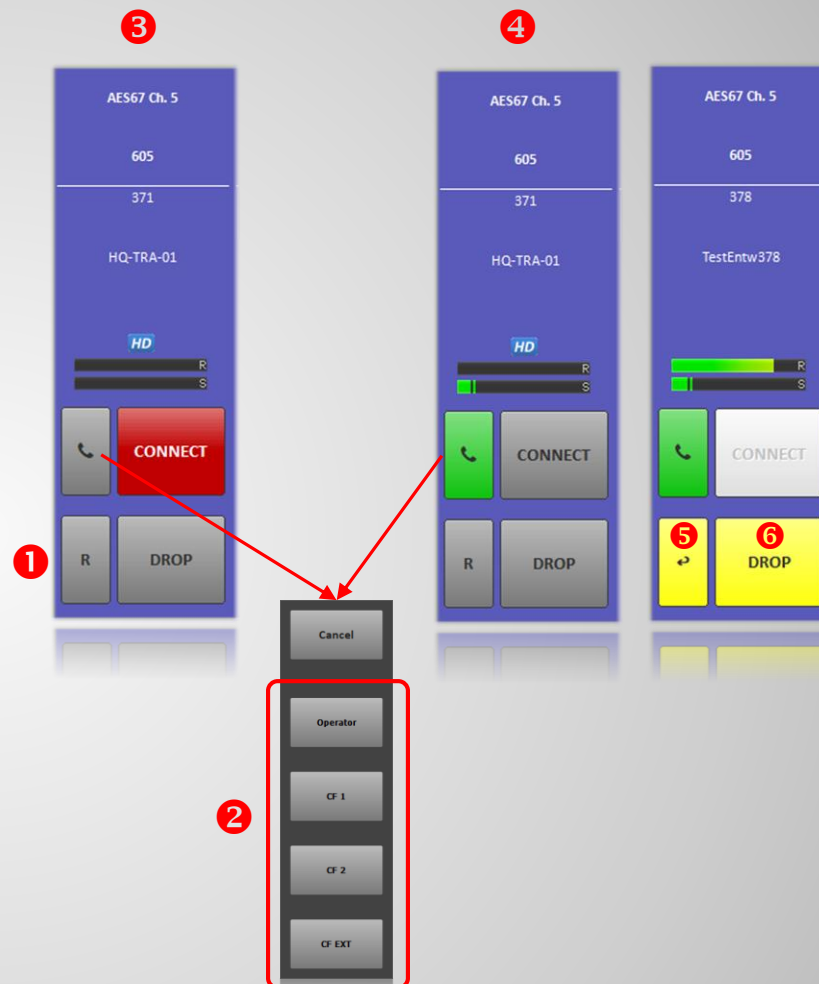
Pretalk

- A key (❶) to hold the connection can be displayed for all lines (activate under SYSTEM CONFIGURATION → INTRO / HOLD SIGNAL).
- The position of the key depends on whether the pretalk function is activated.
- The received audio signal will no longer be output to the audio interface.
- The following signals may be played to the caller:
 - Intro: A short announcement, individual for each line stored on the THipPro Intercom system.
 - External: A signal fed via a separate audio interface.
 - Line: The signal from the audio interface assigned to the telephone line.



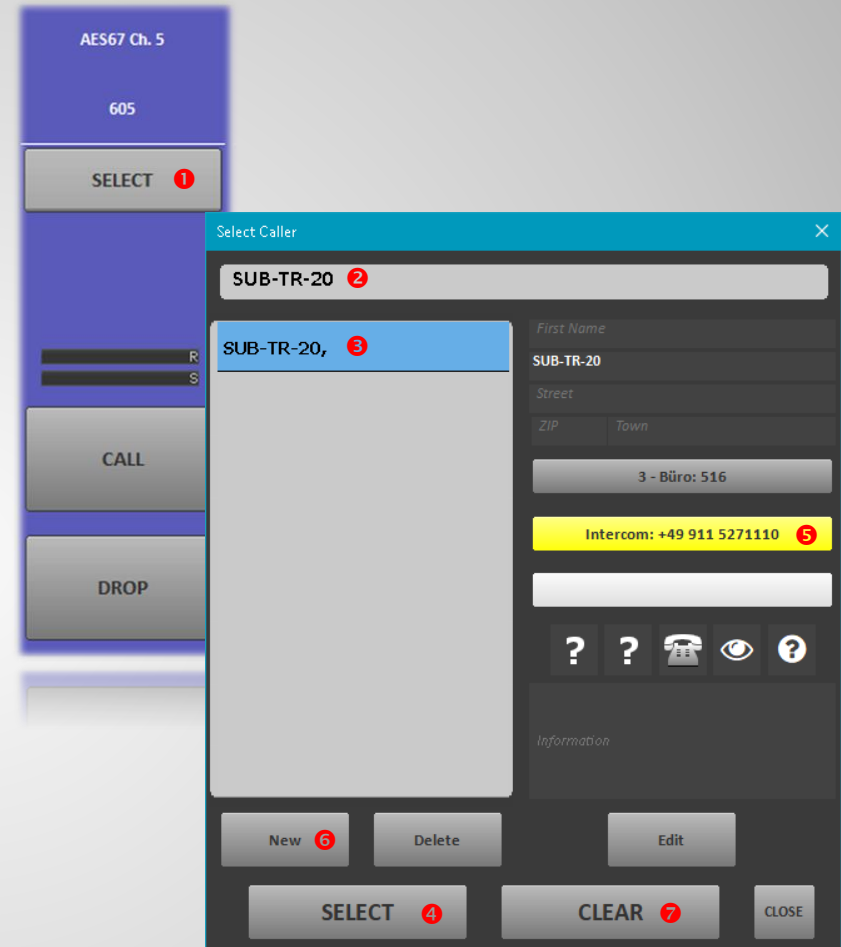
Hold

- An R key (①) can be displayed for all lines to transfer calls to up to 10 predefined destinations (②) (enable under CONFIGURATION → LOCAL SETTINGS → QUICK DIALS / CALL TRANSFER).
- In the CONNECT (③) state, the connection is transferred directly (blind call transfer).
- In the PRETALK (④) state, the desired party is notified (attended call transfer).
 - The symbol of the R key changes to an arrow (⑤). The call can be retrieved.
 - The DROP key (⑥) completes the call transfer.



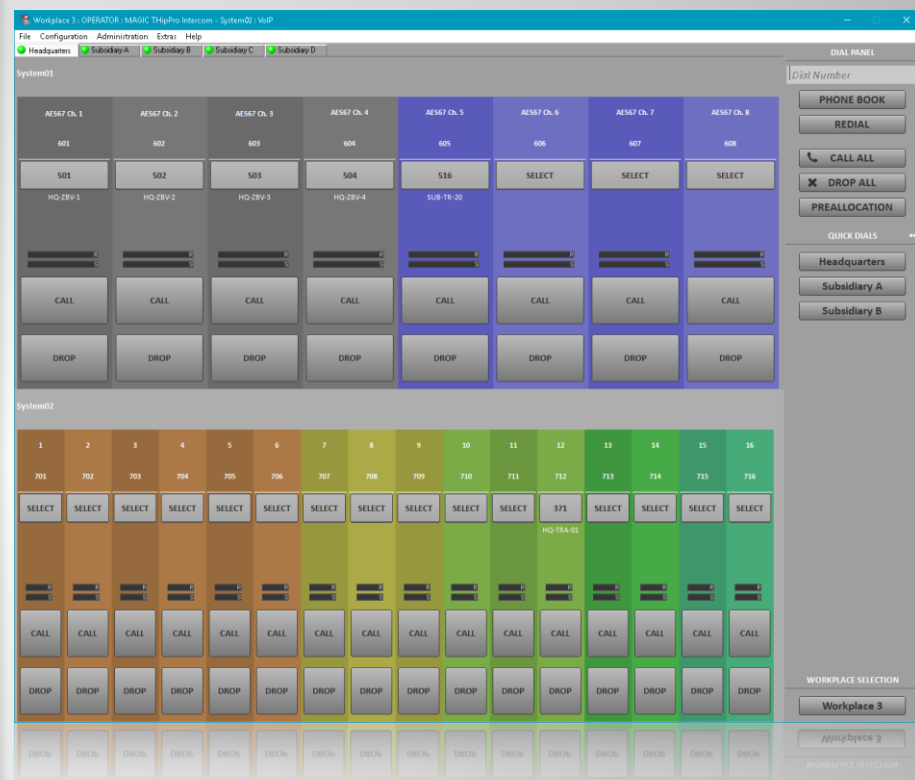
Call Transfer

- A telephone connection can also be prepared and established at a later time.
- The SELECT (❶) key opens the phone book:
 - An entry can be searched for (❷) and selected (❸) here and the preallocation can be completed by pressing the SELECT key ❹().
 - If several numbers are stored for an entry, the preallocation is completed by clicking on the desired number (❺).
 - If the desired number is not yet stored in the phone book, a new entry must be created using the NEW (❻) key.
 - The CLEAR key (❼) deletes the preallocation on this line.
- Preallocation is also possible without a database:
 - The SELECT (❶) key opens a keypad.
 - Here you can enter a number and complete the preallocation by pressing the SELECT key.
 - The CLEAR key (❼) deletes the preallocation on this line.



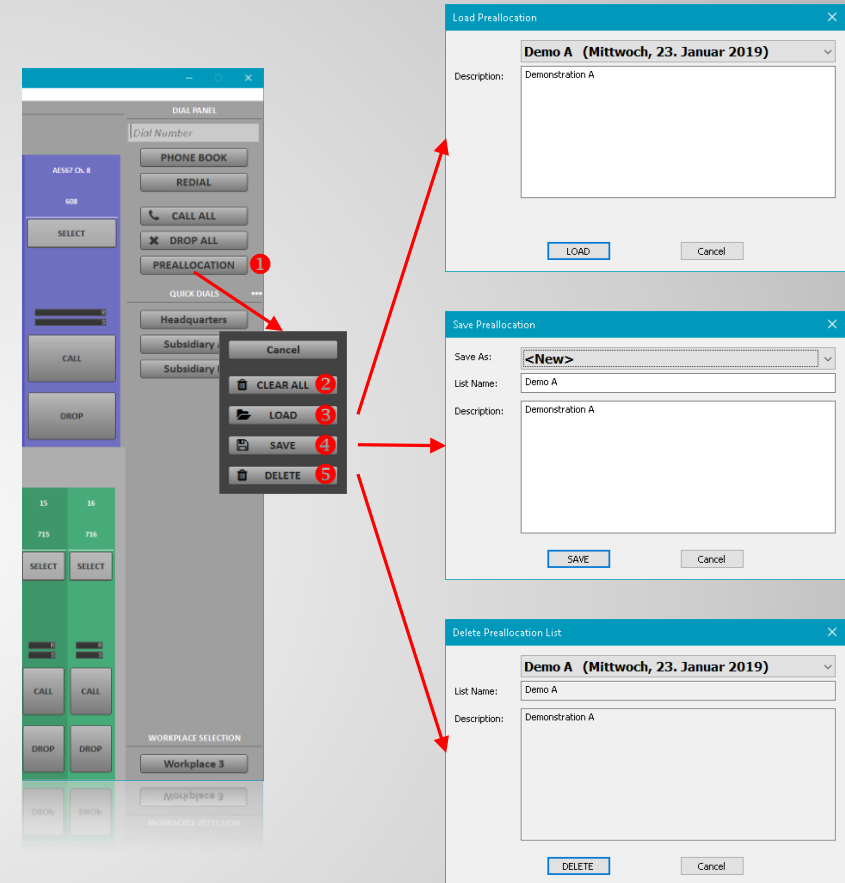
Preallocation of lines (1)

- The CALL ALL (❶) key is used to establish telephone connections on all preallocated lines of all *THipPro Intercom* systems.
- The DROP ALL (❷) key terminates all active telephone connections on all *THipPro Intercom* systems.



Preallocation of lines (2)

- The preallocation of all lines of all *THipPro Intercom* systems can be stored in a preallocation record in the database.
- Each *THipPro Intercom* system must be assigned an individual system index under SYSTEM CONFIGURATION → GENERAL → SYSTEM → INDEX.
- Preallocation records are managed via the PREALLOCATION key (❶):
 - CLEAR ALL (❷) clears the preallocations of all *THipPro Intercom* systems.
 - LOAD (❸) displays all preallocation records stored in the database.
 - SAVE (❹) stores the lines currently preallocated on the *THipPro Intercom* systems as preallocation records in the database. Select <New> under SAVE AS to create a new record. Select an existing record under SAVE AS to overwrite it.
 - Individual preallocation records can be deleted from the database using DELETE (❺).



Preallocation of lines (3)

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Local Settings

These settings are stored on the PC. Each instance on each PC stores its own record. The storage location depends on the SETTINGS FOLDER setting in the LOCAL SETTINGS (see Tips & Tricks).

- All Settings under CONFIGURATION → LOCAL SETTINGS are stored on the PC.
- On page APPLICATION PARAMETER the appearance, notifications and path of the log files are configured.
- MAIN WINDOW SIZE: The size of the program window in pixels.
 - AUTO: The window size is automatically set to full screen resolution at program start.
 - CUSTOM: The window size can be specified pixel by pixel under CUSTOM WINDOW SIZE.
 - Some frequently used window sizes can be selected directly.
- LAYOUT: Arrangement of the *THipPro Intercom* systems on the main window:
 - COLUMNS: Number of systems next to each other.
 - ROWS: Number of systems on top of each other.
 - HIDE UNITS WITHOUT CONFIGURED IP ADDRESS: Gaps in the list of systems under CONTROL INTERFACE can optionally be skipped.
- SYSTEM APPEARANCE: Arrangement of the caller lines of the individual *THipPro Intercom* devices:
 - ROWS PER SYSTEM: One or two row display of a *THipPro intercom* system.
 - CALLER LINES PER ROW: Number of caller lines of a *THipPro Intercom* System per line.
 - UNIFORM WIDTH: All lines have the same width.
 - MIN X: At least X caller lines per row.
 - 16: 16 caller lines per row
 - SPACE BETWEEN ROWS: Height of the horizontal space between two rows.

Application Parameters (1)

- **SYSTEM APPEARANCE:** Arrangement of the caller lines of the individual *THipPro Intercom* devices:
 - **ROWS PER SYSTEM:** One or two row display of a *THipPro intercom* system.
 - **CALLER LINES PER ROW:** Number of caller lines of a *THipPro Intercom* System per line.
 - **UNIFORM WIDTH:** All lines have the same width.
 - **MIN X:** At least X caller lines per row.
 - **16:** 16 caller lines per row
 - **SPACE BETWEEN ROWS:** Height of the horizontal space between two rows.
- **SIDE BAR SIZE:** Width of the sidebar of the main window in pixels.
- **SHOW TITLE BAR:** Displays the classic Windows title bar along with the menu bar. Otherwise, the title bar will disappear, and the menu will be offered in a sidebar that can be opened by pressing the menu button in the upper left corner.
- **SHOW WINDOW ON FIXED POSITION:** The window is displayed at the given position (in pixels) on the screen when the program starts.
- **PAGE LABELS:** Labelling of the tabs in the main window.

The screenshot shows the 'Local Settings' dialog box with the following sections and options:

- Application Parameters:** Quick Dials / Call Transfer, PRETALK Streaming, Settings Folder
- Main Window Presentation:**
 - Screen resolution: 1920*1170
 - Main window size: Custom (dropdown)
 - Custom window size: 1400 * 1100
 - Layout:
 - Columns: 1 (dropdown)
 - Rows: 2 (dropdown)
 - ☐ Hide units without configured IP address
 - Side-Bar size: 200
 - ☒ Show title bar (temporarily toggle with [Scroll-Lock] key)
 - ☐ Show window on fixed position X: 0 Y: 0
- System Appearance:**
 - Rows per System: 1 (dropdown)
 - Caller lines per row: min. 8 (dropdown)
 - Space between rows: 0
- Page labels:**
 - Page 1: Headquarters
 - Page 2: Subsidiary A
 - Page 3: Subsidiary B
 - Page 4: Subsidiary C
 - Page 5: Subsidiary D
 - Page 6: (empty)
 - Page 7: (empty)
 - Page 8: (empty)
 - Page 9: (empty)
 - Page 10: (empty)
- Play Wave File:**
 - ☐ On Incoming Call: Ring.wav (Browse)
 - ☐ On Remote Drop: C:\Users\Public\AVT\MAGIC THipPro Intercom\Warning %d.wav (Browse)
 - ☐ On Local Drop: %d in the wave file name will be replaced by the line index (1..16) of the line which was dropped.
- Logfile:**
 - Logfile Folder: C:\Users\Public\AVT\MAGIC THipPro Intercom\Logfiles (Browse) (Open)
 - File Name Format: <folder>\log-<day of month>.txt (default) (dropdown)

Buttons at the bottom: OK, Abbrechen, OK, Abbrechen.

Application Parameters (2)

- **PLAY WAVE FILE:** The PC software can play audio files in WAVE format to signal the following events:
 - **ON INCOMING CALL:** Activates the playback of an audio file when a call arrives on a line of the *THipPro Intercom* system.
 - **ON REMOTE DROP:** Activates the playback of an audio file when the telephone connection has been terminated by the remote side.
 - **ON LOCAL DROP:** Enables playback of an audio file when the telephone connection has been disconnected from the local side.
 - A file name can be specified for incoming and outgoing calls respectively. If the file name or path contains the symbol "%d", a separate audio file will be played for each line. In the example, Warning_%d.wav expands to Warning_1.wav for line 1, Warning_2.wav for line 2, and so on.

Application Parameters (3)

- **LOGFILE:** A log file is saved on the PC for each day of the month. After one month the files are overwritten.
 - **LOGFILE FOLDER:** The storage path of the log files can be selected via BROWSE.
 - **OPEN** opens the current log file.
 - **FILE NAME FORMAT:** The PC name can be included in the file name of the log file, e.g. if all log files are stored in a central location.

The screenshot shows the 'Local Settings' dialog box with the 'Application Parameters' tab selected. The settings are organized into several sections:

- Main Window Presentation:**
 - Screen resolution: 1920*1170
 - Main window size: Custom (dropdown)
 - Custom window size: 1400 * 1100
- Layout:**
 - Columns: 1 (dropdown)
 - Rows: 2 (dropdown)
 - ☐ Hide units without configured IP address
- System Appearance:**
 - Rows per System: 1 (dropdown)
 - Caller lines per row: min. 8 (dropdown)
 - Space between rows: 0
- Side-Bar size:** 200
- ☒ Show title bar (temporarily toggle with [Scroll-Lock] key)
- ☐ Show window on fixed position X: 0 Y: 0
- Page labels:**
 - Page 1: Headquarters
 - Page 2: Subsidiary A
 - Page 3: Subsidiary B
 - Page 4: Subsidiary C
 - Page 5: Subsidiary D
 - Page 6: (empty)
 - Page 7: (empty)
 - Page 8: (empty)
 - Page 9: (empty)
 - Page 10: (empty)
- Play Wave File:**
 - ☐ On Incoming Call: Ring.wav (Browse)
 - ☐ On Remote Drop: C:\Users\Public\AVT\MAGIC THipPro Intercom\Warning %d.wav (Browse)
 - ☐ On Local Drop: %d in the wave file name will be replaced by the line index (1..16) of the line which was dropped.
- Logfile:**
 - Logfile Folder: C:\Users\Public\AVT\MAGIC THipPro Intercom\Logfiles (Browse)
 - File Name Format: <folder>\log-<day of month>.txt (default) (dropdown)
 - Buttons: OK, Abbrechen, Open

Application Parameters (4)

- Up to 20 quick dial entries can be configured.
 - NAME: Labelling of the quick dial key.
 - NUMBER: Telephone number.
 - The quick dial keys are displayed in the sidebar of the main window.
- Calls can be transferred to up to 10 different numbers.
 - **ENABLE CALL TRANSFER:** This activates the call transfer function. An R key is displayed on each line in the main window.
 - **NAME:** You can optionally specify a name for a transfer destination.
 - **NUMBER:** Telephone number.
- **USE BLIND CALL TRANSFER...:** Sets the SIP method used to transfer a connection.
 - Anyhow, an attended call transfer is only performed if the line is in Pretalk.

Local Settings

Application Parameters Quick Dials / Call Transfer PRETALK Streaming Settings Folder

Quick Dials

	Name	Number
1:	Headquarters	300
2:	Subsidiary A	401
3:	Subsidiary B	402
4:		
5:		
6:		
7:		
8:		
9:		
10:		
11:		
12:		
13:		
14:		
15:		
16:		
17:		
18:		
19:		
20:		

☒ Enable Call Transfer

	Name	Number
1:	Operator	502
2:	CF 1	551
3:	CF 2	552
4:	CF EXT	09115271110
5:		
6:		
7:		
8:		
9:		
10:		

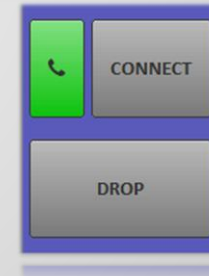
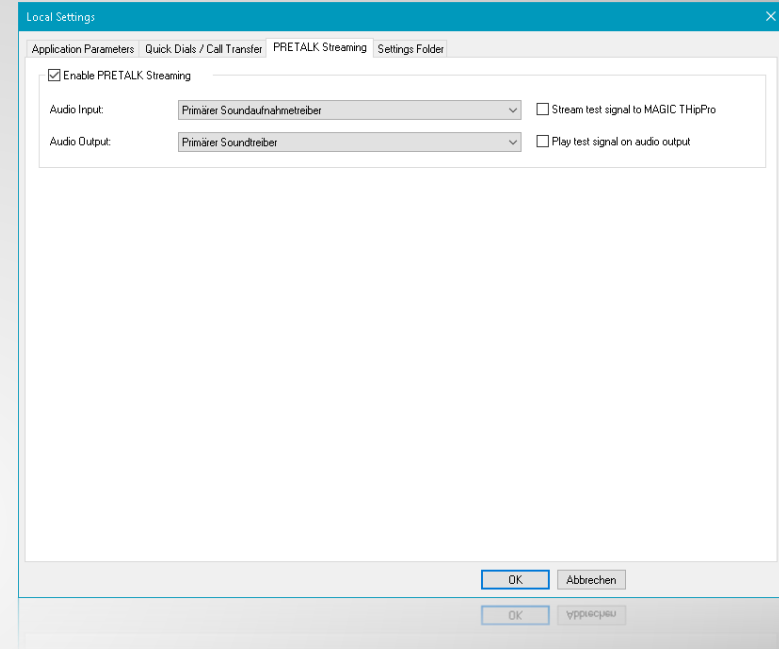
☐ Use "Blind Call Transfer" instead of "Attended Call Transfer"
(In case of PRETALK "Attended Call Transfer" is still used)

OK Abbrechen

OK Abbrechen

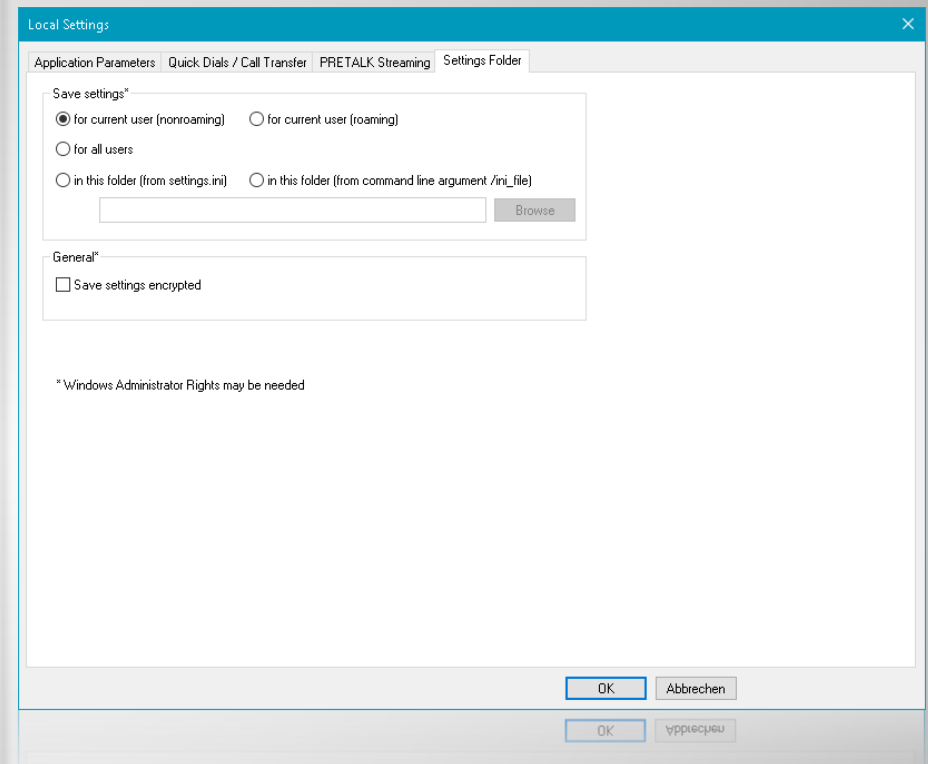
Quick Dials / Call Transfer

- The PC software allows the user to speak directly to the remote station (software option PRETALK STREAMING required).
- This pretalk is conducted via the PC's sound card.
- The audio data streams are transferred between the PC and the *THipPro Intercom System* via LAN.
- Configure the function on the PRETALK STREAMING page:
 - **ENABLE PRETALK STREAMING:** A pretalk key is displayed on all lines.
 - **AUDIO INPUT:** Select an audio recording device from the PC.
 - **STREAM TEST SIGNAL TO MAGIC THIPPRO:** Sends a test signal to the remote terminal to check the network connection to the THipPro intercom system. A telephone connection must be established in PRETALK mode.
 - **AUDIO OUTPUT:** Select an audio output device from the PC.
 - **PLAY TEST SIGNAL ON AUDIO OUTPUT:** Plays a test signal on the PC's audio output device to check the network connection from the *THipPro Intercom system*. A telephone connection must be established in PRETALK mode.



Pretalk Streaming

- Determine how the settings are stored on the local PC on page SETTINGS FOLDER:
 - FOR CURRENT USER: Each user has separate settings and can change them themselves.
 - FOR ALL USERS: All users of the PC use identical settings. Administrator rights are needed to change them.
 - IN THIS FOLDER: The settings are saved to a file in the specified location.
- SAVE SETTINGS ENCRYPTED: The local settings are stored encrypted.
- If you want to create a backup of the local settings, the command *showprofilepath* under ADMINISTRATION → SYSTEM PANEL shows where the file can be found.



Settings Folder

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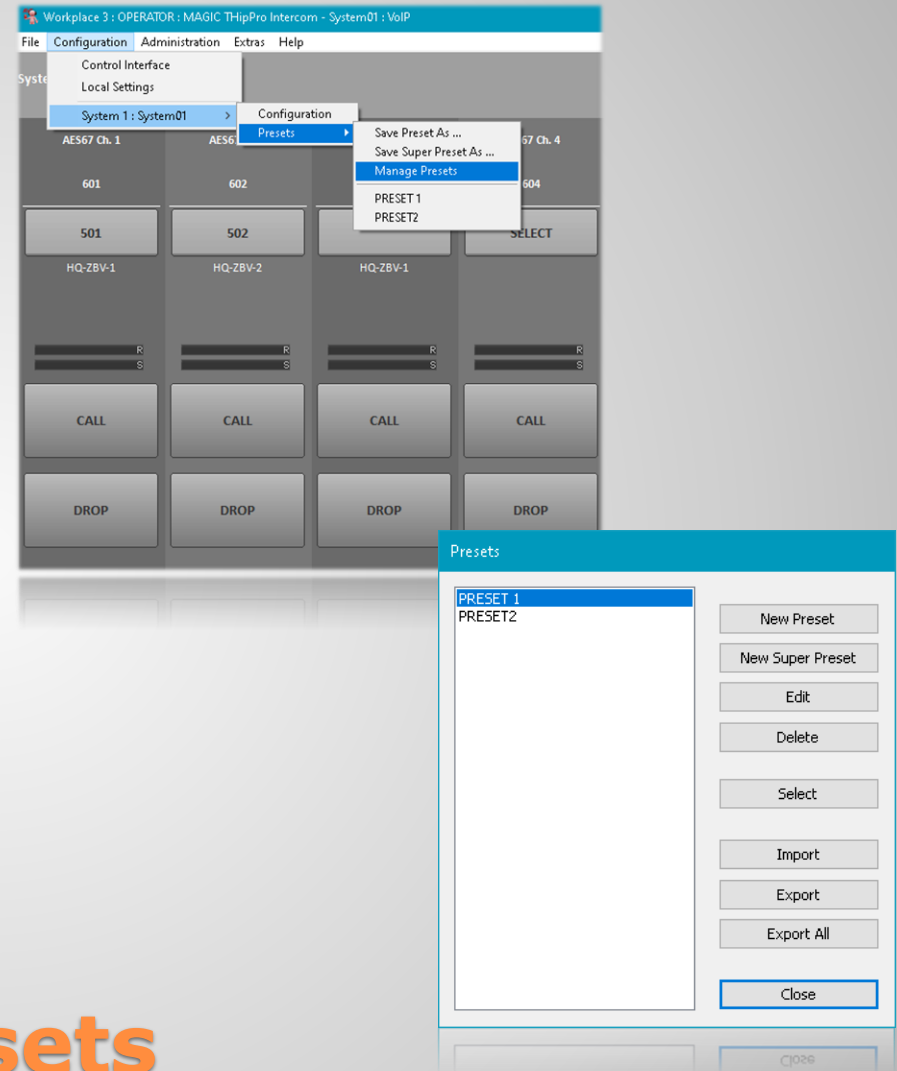
Intercom

System Configuration

- The settings can be found in the CONFIGURATION menu. There is a submenu for each connected *THipPro Intercom* device. CONFIGURATION opens the settings of the respective system:
 - These settings are stored on the device.
 - All settings under GLOBAL SETTINGS and OPERATION SETTINGS can be saved as PRESET.
 - A SUPER PRESET contains all settings under GLOBAL SETTINGS, OPERATION SETTINGS and SYSTEM SETTINGS.
- PRESETS and SUPER PRESETS are stored on the device and can be managed and loaded via CONFIGURATION → SYSTEM X → PRESETS.

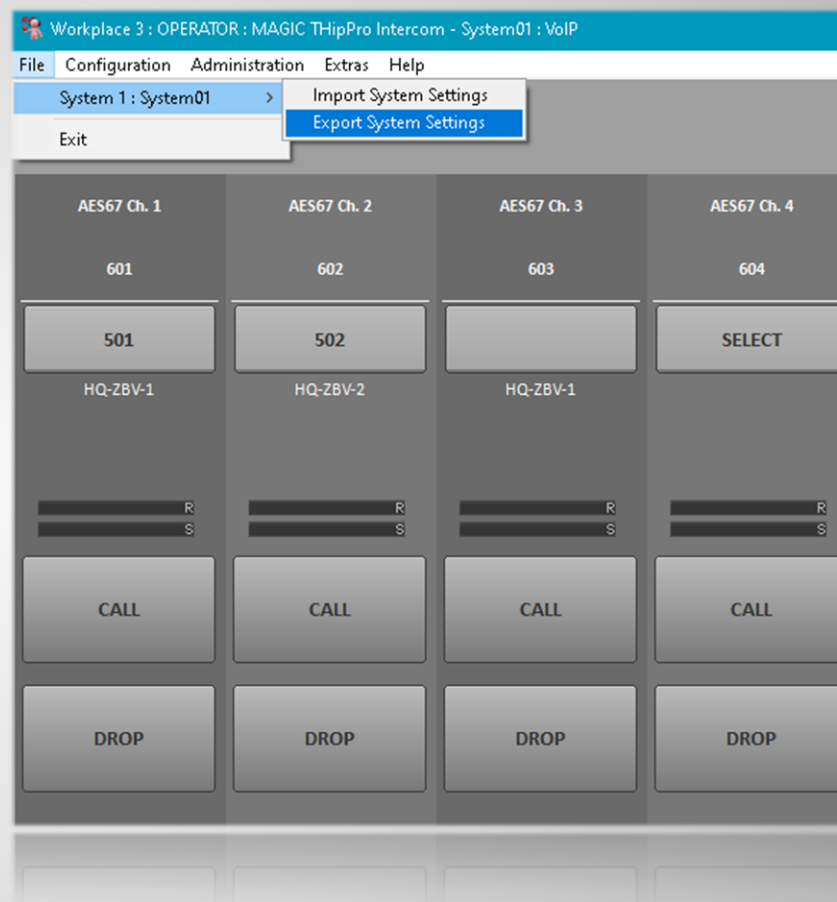
General

- Presets are managed via CONFIGURATION → SYSTEM X → PRESETS:
 - SAVE PRESET AS:** The current configuration of the OPERATION SETTINGS branch is saved as a Preset. The name is freely selectable (max. 16 characters).
 - SAVE SUPER PRESET AS:** The entire current system configuration is saved as a super Preset. The name is freely selectable (max. 16 characters).
 - MANAGE PRESETS:** Displays a list of all stored Presets and offers additional management functions:
 - NEW PRESET:** Creates a new Preset based on the current configuration.
 - NEW SUPER PRESET:** Creates a new super Preset based on the current configuration.
 - EDIT:** Opens the selected Preset for editing.
 - SELECT:** Activates the selected Preset.
 - IMPORT:** Imports a Preset stored on the PC.
 - EXPORT:** Saves a selected Preset to the PC.
 - EXPORT ALL:** Saves all Presets to the PC.
 - List of Presets:** The menu also displays all available Presets. Click on a Preset to activate it.



Presets

- The system configuration can be stored in a file under FILE → SYSTEM X → EXPORT SYSTEM SETTINGS.
- To restore a backup, open the file using FILE → SYSTEM X → IMPORT SYSTEM SETTINGS.
- It is also recommended to recreate the backup file after a firmware update, as it cannot be guaranteed that old backups are compatible with the latest firmware.
 - In such a case, the device would first have to be downgraded to the software version with which the backup file was created.



Backup and restore

- The local settings are automatically protected when working under a user account while FOR ALL USERS is selected under CONFIGURATION → LOCAL SETTINGS → SETTINGS LOCATION.
- To also protect the system settings, a password must be set under LOGIN.
- Two levels are available:
 - ADMINISTRATOR: Log in with this password to access all functions and settings.
 - USER: Log in with this password to load presets and switch between workplaces.
- Note: If you have forgotten the administrator password, the device can only be unlocked by resetting to factory settings.

The screenshot shows a software window titled "System01 - Configuration". On the left is a tree view with categories: Global Settings (Clients / Security, General, Client Workplace Assignment, Client Workplace Restrictions, Database), Operation Settings (Workplace Definition, Audio Line, Intro / HOLD Signal, Signal Processing, Line Labels, Auto Answer), GPIO, and System Settings (General, Line Interface, Caller Line Grouping, VoIP (LAN/SIP), Audio Interface, PRETALK Streaming, AES67, LAN Interface, VLAN, DHD Audio Matrix, Ember+, SNMP). The "Login" option under System Settings is highlighted. A "Login" dialog box is open in the foreground, containing two sections: "USER" and "ADMINISTRATOR". Each section has "Password:" and "Confirm Password:" fields, all masked with asterisks. At the bottom of the dialog are "OK", "Abbrechen", and "Apply Now" buttons. Below the dialog, the main window shows "Client ID: 1" and "Workplace: 3".

Login

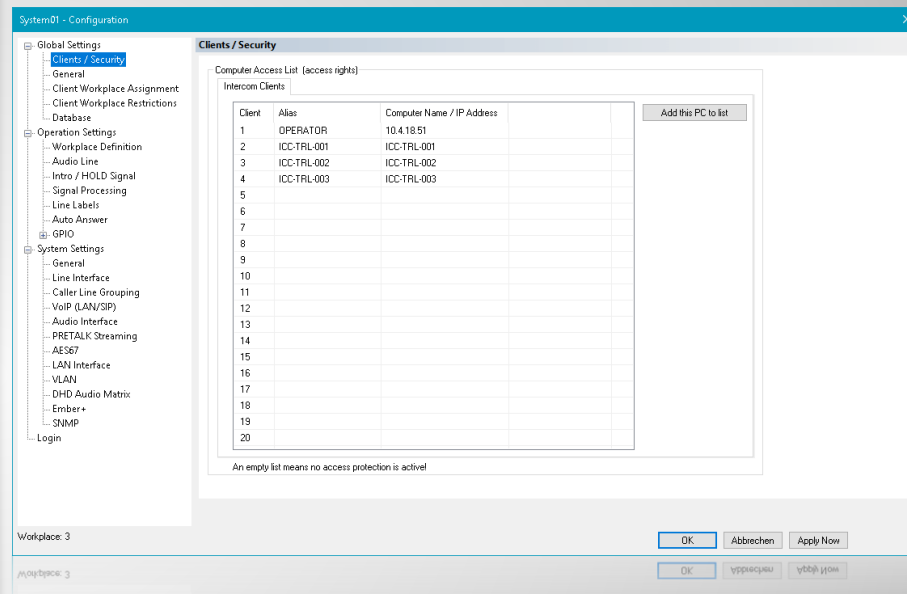
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Global Settings

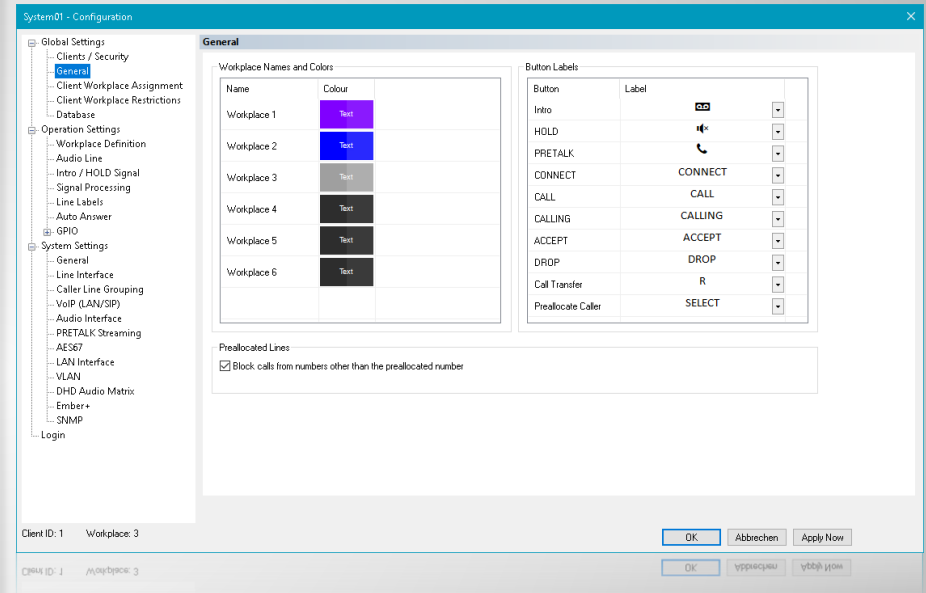
These settings are identical for all connected systems. The PC software only reads the GLOBAL SETTINGS of the first available THipPro Intercom device and displays them for all subsequent devices. If the configuration window is closed with OK, the settings are saved on all connected devices.

- Optionally, up to 20 PCs can connect to a *THipPro Intercom* system.
- If the INTERCOM CLIENTS list on the CLIENTS/SECURITY page is empty, any PC can connect to the system.
- As soon as an entry exists, access protection is active.
 - All PCs in the list can connect directly to the system
 - On all other PCs, the administrator password must be entered when establishing a connection.



Clients / Security

- Basic settings are made on the GENERAL page.
- Each workplace configuration can be assigned a name and a background colour of the main window.
- Customize the buttons off the main window under BUTTON LABELS.
- If BLOCK CALLS FROM NUMBERS OTHER THAN THE PREALLOCATED NUMBER is activated the *THipPro Intercom* Systems only accept calls from the respective preallocated number.



General

- On the CLIENT WORKPLACE ASSIGNMENT page, the WORKPLACE ACCESS column specifies which workplace configurations the respective clients can access.

The screenshot shows the 'System01 - Configuration' window with the 'Client Workplace Assignment' tab selected. The left sidebar lists various configuration categories, with 'Client Workplace Assignment' highlighted. The main area displays a table for assigning workplace access to clients.

Client	Workplace Access
1 : OPERATOR	1,2,3,4
2 : ICC-TRL-001	1
3 : ICC-TRL-002	2,3
4 : ICC-TRL-003	4

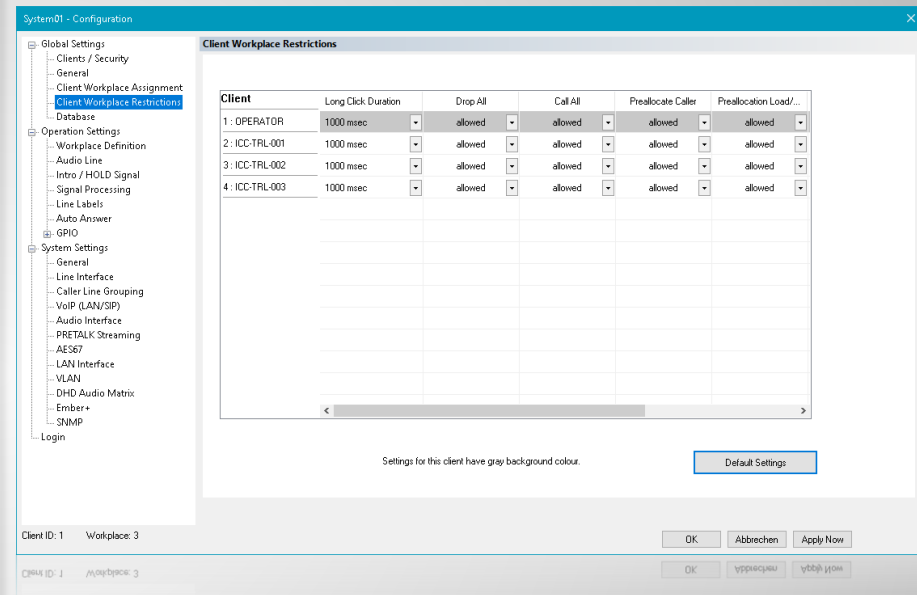
Caution: Invalid settings are red. Settings for this client have gray background colour.

Client ID: 1 Workplace: 3 [OK] [Abbrechen] [Apply Now]

Client ID: 1 Workplace: 3 [OK] [Abbrechen] [Apply Now]

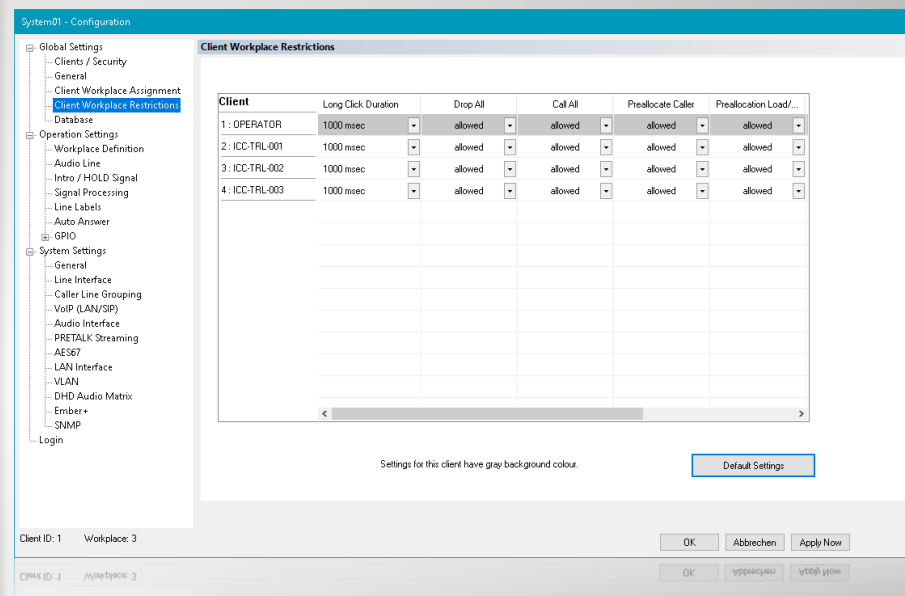
Client Workplace Assignment

- On the CLIENT WORKPLACE RESTRICTIONS page, the permissions of individual clients can be configured centrally.
- For each client, each authorization can be set to one of two values:
 - ALLOWED: The client is allowed to use this function.
 - FORBIDDEN: The client is not allowed to use this function.
- LONG CLICK DURATION: Special functions of some keys are triggered by a long keystroke. The duration from which a keystroke is recognized as a long keystroke can be set here.



Client Workplace Restrictions (1)

- The following permissions are available:
 - DROP ALL: Hang up all lines.
 - CALL ALL: Establish telephone connections on all preallocated lines.
 - PREALLOCATE CALLER: Preallocate lines.
 - PREALLOCATION LOAD / SAVE / DELETE: Load, save and delete preallocation records.
 - VIEW CALL HISTORY: Show call history list on main panel.
 - MANAGE PRESETS: Manage presets.
 - LOAD SUPER PRESET: Load Super Presets.
 - LOAD PRESET: Load presets.



Client Workplace Restrictions (2)

- The phone book is configured on the DATABASE page.
- The phonebook is provided via a Microsoft SQL database.
 - Information on installing a free Microsoft SQLExpress database can be found in the download area of our website under QUICK GUIDES in the document SQL SERVER 2012 INSTALLATION.
- Enable the DATABASE and enter the following information:
 - SQL SERVER: IP address or computer name of the PC on which the SQL Server is installed and - if available - the SQL Server instance.
 - DATABASE: Name of the database
 - USER
 - PASSWORD
 - NETWORK LIBRARY: Default
 - You can test the database connection via TEST/OPEN CONNECTION.

System01 - Configuration

Database

1. 10.4.18.5\SQLEXPRESS:AVTIntercom

☒ Database

SQL Server: 10.4.18.5\SQLEXPRESS Test/Open Connection

Database: AVTIntercom

User: IntercomUser

Password: *****

Network Library: Default

Telephone Book

Phone Number Types	Connection Types	Mode	Resolution	Sampling Rate	Data Rate	Secure Stream	Remot...
Other	Telephone						
Mobile	Telephone						
Office	Telephone						
Private	Telephone						
Intercom	Telephone						

Add Phone Number Type Save

Location Details

Country: Deutschland [+49] Country Code: +49

City: Nürnberg Area Code: 911

Label for in-house calls: intern ☐ Disable Number Formatting

Client ID: 1 Workplace: 3

OK Abbrechen Apply Now

Database (1)

- Under TELEPHONE BOOK you can define the PHONE NUMBER TYPES that shall be available in the telephone book, e.g. private, mobile, office, etc.
- If you make a change, save the data with SAVE
- To enable automatic area code evaluation, enter your location information (LOCATION DETAILS):
 - The import of a valid area code database is required (see document SQL SERVER 2012 INSTALLATION).
 - Select your COUNTRY.
 - Enter your CITY.
 - Please check that your COUNTRY CODE and AREA CODE are correct.
 - Specify a LABEL FOR IN-HOUSE CALLS.
 - If you have problems with number formatting, choose DISABLE NUMBER FORMATTING.
 - Only applicable if area code database is used.

System01 - Configuration

Database

1. 10.4.18.5\SQL&PRESS.AVTIntercom

☒ Database

SQL Server: 10.4.18.5\SQL&PRESS

Database: AVTIntercom

User: IntercomUser

Password: *****

Network Library: Default

Test/Open Connection

Telephone Book:

Phone Number Types	Connection Types	Mode	Resolution	Sampling Rate	Data Rate	Secure Stream	Remot...
Other	Telephone						
Mobile	Telephone						
Office	Telephone						
Private	Telephone						
Intercom	Telephone						

Add Phone Number Type Save

Location Details:

Country: Deutschland (+49) Country Code: +49

City: Nürnberg Area Code: 911

Label for in-house calls: cintern

☐ Disable Number Formatting

Client ID: 1 Workplace: 3

OK Abbrechen Apply Now

Database (2)

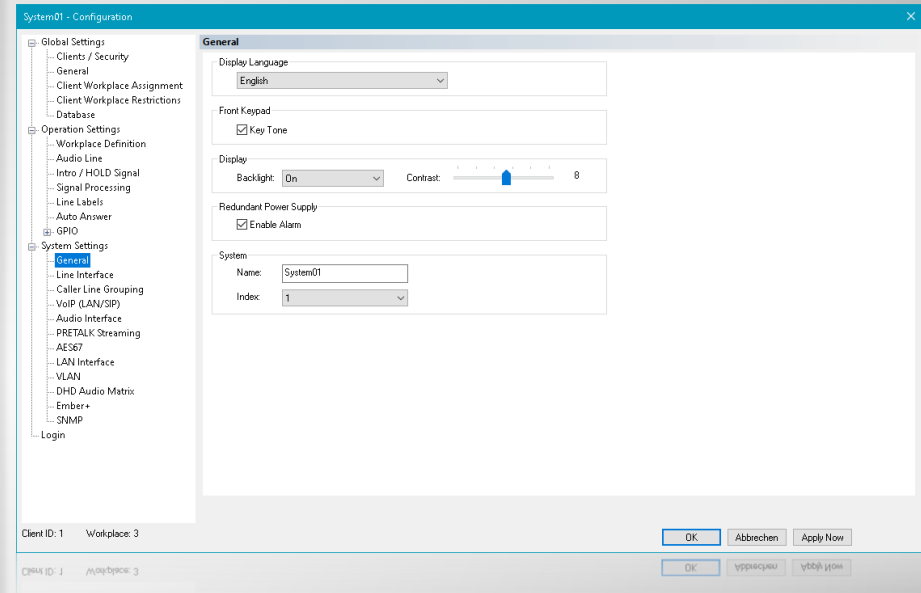
MAGIC THipPro

Intercom

System Settings

Settings that cannot be loaded using a PRESET.

- The language for the front display can be set under DISPLAY LANGUAGE.
- KEY TONE activates the key click on the front keypad.
- Backlight and contrast of the front display are set under DISPLAY.
- Under NAME you can enter a device name, which is also displayed on the main window.
- REDUNDANT POWER SUPPLY → ENABLE ALARM activates the alarm window if one of the two power supplies (optional from hardware version 4.0) fails.
- Assign an individual INDEX to every *THipPro Intercom* system to be able to save preallocation records.



General

- LINE MODE: *THipPro intercom* systems are only available with VoIP interface.
- DROP NOT ANSWERED CALLS AFTER 90 SECONDS terminates a connection if it is not established after 90 seconds at the latest.
- On the LINE INTERFACE page, in the INHOUSE LINES line, mark all lines which are connect to a PBX.
- ANONYMOUS CALLING suppresses the own phone number for outgoing calls. Attention: Some SIP servers then refuse registration.
- Under PBX/EXCHANGE LINE CONFIGURATION, automatic outside line access is configured:
 - INTERNATIONAL PREFIX is the prefix for international calls.
 - NATIONAL PREFIX is the prefix for calls to other national prefix areas.
 - LENGTH OF EXTENSION is the number length of internal extensions.
 - OUTGOING LINE PREFIX is the prefix for external calls.
 - PBX NUMBER is the local number of the PBX.
 - SKIP OUTGOING LINE PREFIX ON INCOMING CALL: Activate this function if the PBX signals numbers of incoming calls including the prefix for external calls.
 - ENTER OUTGOING LINE PREFIX ON MANUAL CALLS prompts the user to enter the prefix for external calls when dialling manually.
 - ANONYMOUS CALL SIGNALLING: Character string used by your PBX to signal anonymous calls.
 - IGNORE SIP DISPLAY NAME OF CALLER discards the caller's supplied display name.

System01 - Configuration

Line Interface

General

Line Mode: VoIP (LAN/SIP)

☐ Drop not answered incoming/outgoing calls after 90 seconds

	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
In-house Lines	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>
Anonymous Calling	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

PBX/Exchange line configuration

International prefix: 00 (Default value: 00)

National prefix: 0 (Default value: 0)

Length of extension: 3

Outgoing line prefix: 0

PBX number: 5271

☐ Skip outgoing line prefix on incoming calls

☐ Enter outgoing line prefix on manual calls

☐ Anonymous Call signalling

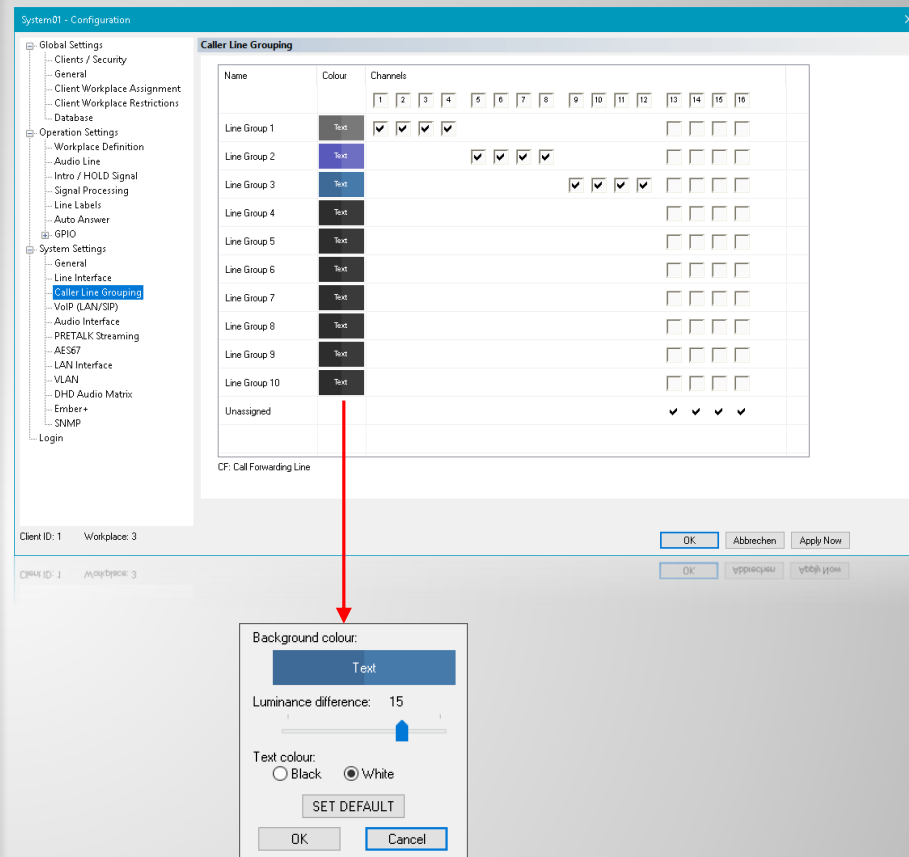
☐ Ignore SIP Display Name of caller

Client ID: 1 Workplace: 3

OK Abbrechen Apply Now

Line Interface

- Under CALLER LINE GROUPING lines can be combined into groups.
 - Up to 10 line groups can be set up per system.
 - All unassigned lines remain in the standard UNASSIGNED line pool.
 - You can assign a NAME to a line group, which is available under LINE LABELS.
- In the COLOUR column, each group can be assigned an individual colour.



Caller Line Grouping

- On the VOIP (LAN/SIP) page the access data of the VoIP lines are configured, please also refer to the tips & tricks at the end of this document.
- The following parameters are available for each line:
 - LAN: Network interface to be used.
 - SIP SERVER: IP address or URL of the SIP server, DNS-SRV is also supported.
 - LAN: Network interface to be used for the redundant SIP server.
 - BACKUP SERVER: IP address of the redundant SIP server. This parameter is ignored for DNS-SRV.
 - TCP: Switching between UDP and TCP as SIP transport protocol.
 - STUN, USER NAME, USER AUTHENTICATION, PASSWORD: SIP account credentials.
 - AUDIO PORT: Local UDP port for RTP audio transmission.
 - DISPLAYED NAME: Any text transmitted to and displayed by the other party.

Line	LAN	SIP Server	LAN	Backup Server	TCP	STUN	User Name	User Authentication	Password	Audio Port	Display Name
Line 1	1	10.2.140.8	2	10.2.141.8	<input type="checkbox"/>	<input type="checkbox"/>	601		***	5004	
Line 2	1	10.2.140.8	2	10.2.141.8	<input type="checkbox"/>	<input type="checkbox"/>	602		***	5005	
Line 3	1	10.2.140.8	2	10.2.141.8	<input type="checkbox"/>	<input type="checkbox"/>	603		***	5006	
Line 4	1	10.2.140.8	2	10.2.141.8	<input type="checkbox"/>	<input type="checkbox"/>	604		***	5007	
Line 5	1	10.2.140.8	2	10.2.141.8	<input type="checkbox"/>	<input type="checkbox"/>	605		***	5008	
Line 6	1	10.2.140.8	2	10.2.141.8	<input type="checkbox"/>	<input type="checkbox"/>	606		***	5009	
Line 7	1	10.2.140.8	2	10.2.141.8	<input type="checkbox"/>	<input type="checkbox"/>	607		***	5010	
Line 8	1	10.2.140.8	2	10.2.141.8	<input type="checkbox"/>	<input type="checkbox"/>	608		***	5011	
Line 9	1	10.2.140.8	2	10.2.141.8	<input type="checkbox"/>	<input type="checkbox"/>	609		***	5012	
Line 10	1	10.2.140.8	2	10.2.141.8	<input type="checkbox"/>	<input type="checkbox"/>	610		***	5013	
Line 11	1	10.2.140.8	2	10.2.141.8	<input type="checkbox"/>	<input type="checkbox"/>	611		***	5014	
Line 12	1	10.2.140.8	2	10.2.141.8	<input type="checkbox"/>	<input type="checkbox"/>	612		***	5015	
Line 13	1	10.2.140.8	2	10.2.141.8	<input type="checkbox"/>	<input type="checkbox"/>	613		***	5016	
Line 14	1	10.2.140.8	2	10.2.141.8	<input type="checkbox"/>	<input type="checkbox"/>	614		***	5017	
Line 15	1	10.2.140.8	2	10.2.141.8	<input type="checkbox"/>	<input type="checkbox"/>	615		***	5018	
Line 16	1	10.2.140.8	2	10.2.141.8	<input type="checkbox"/>	<input type="checkbox"/>	616		***	5019	

VoIP Parameter

Payload Time: msec

☐ A-Law/u-Law Signalling on incoming G.722 calls

☐ Use first codec of SDP audio codec list as default

Registration

Delay between SIP lines: msec (0, 4000)

Timeout: sec (60, 500)

Client ID: 1 Workplace: 3

OK Abbrechen Apply Now

Set Default Audio Ports

VoIP (LAN/SIP) (1)

- A low PAYLOAD TIME reduces the audio delay, but increases the gross data rate a little.
- A-LAW/U-LAW SIGNALLING ON INCOMING G.722 CALLS / USE FIRST CODEC OF SDP AUDIO CODEC LIST AS DEFAULT: Functions to avoid errors in the implementations of some SIP servers. Normally disabled.
- Under REGISTRATION the registration on the SIP server can be controlled:
 - DELAY BETWEEN SIP LINES: Some SIP servers refuse registration if all lines of the *ThipPro intercom* system log in at the same time. Here you can set a time interval between the lines.
 - TIMEOUT: Time until the line is re-registered.

System01 - Configuration

Global Settings

- Client / Security
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 - Client Workplace Restrictions
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- Operation Settings
 - Workplace Definition
 - Audio Line
 - Intro / HOLD Signal
 - Signal Processing
 - Line Labels
 - Auto Answer
- GPIO
- System Settings
 - General
 - Line Interface
 - Caller Line Grouping
 - VoIP (LAN/SIP)**
 - Audio Interface
 - PRETALK Streaming
 - AESE7
 - LAN Interface
 - VLAN
 - DHD Audio Matrix
 - Ember+
 - SNMP
- Login

VoIP (LAN/SIP)

Line	LAN	SIP Server	LAN	Backup Server	TCP	STUN	User Name	User Authent...	Password	Audio Port (...)	Display...
Line 1	1	10.2.140.8	2	10.2.141.8	<input type="checkbox"/>	<input type="checkbox"/>	601		***	5004	
Line 2	1	10.2.140.8	2	10.2.141.8	<input type="checkbox"/>	<input type="checkbox"/>	602		***	5005	
Line 3	1	10.2.140.8	2	10.2.141.8	<input type="checkbox"/>	<input type="checkbox"/>	603		***	5006	
Line 4	1	10.2.140.8	2	10.2.141.8	<input type="checkbox"/>	<input type="checkbox"/>	604		***	5007	
Line 5	1	10.2.140.8	2	10.2.141.8	<input type="checkbox"/>	<input type="checkbox"/>	605		***	5008	
Line 6	1	10.2.140.8	2	10.2.141.8	<input type="checkbox"/>	<input type="checkbox"/>	606		***	5009	
Line 7	1	10.2.140.8	2	10.2.141.8	<input type="checkbox"/>	<input type="checkbox"/>	607		***	5010	
Line 8	1	10.2.140.8	2	10.2.141.8	<input type="checkbox"/>	<input type="checkbox"/>	608		***	5011	
Line 9	1	10.2.140.8	2	10.2.141.8	<input type="checkbox"/>	<input type="checkbox"/>	609		***	5012	
Line 10	1	10.2.140.8	2	10.2.141.8	<input type="checkbox"/>	<input type="checkbox"/>	610		***	5013	
Line 11	1	10.2.140.8	2	10.2.141.8	<input type="checkbox"/>	<input type="checkbox"/>	611		***	5014	
Line 12	1	10.2.140.8	2	10.2.141.8	<input type="checkbox"/>	<input type="checkbox"/>	612		***	5015	
Line 13	1	10.2.140.8	2	10.2.141.8	<input type="checkbox"/>	<input type="checkbox"/>	613		***	5016	
Line 14	1	10.2.140.8	2	10.2.141.8	<input type="checkbox"/>	<input type="checkbox"/>	614		***	5017	
Line 15	1	10.2.140.8	2	10.2.141.8	<input type="checkbox"/>	<input type="checkbox"/>	615		***	5018	
Line 16	1	10.2.140.8	2	10.2.141.8	<input type="checkbox"/>	<input type="checkbox"/>	616		***	5019	

Set Default Audio Ports

VoIP Parameter

Payload Time: msec

☐ A-Law/u-Law Signalling on incoming G.722 calls

☐ Use first codec of SDP audio codec list as default

Registration

Delay between SIP lines: msec (0, 4000)

Timeout: sec (60, 500)

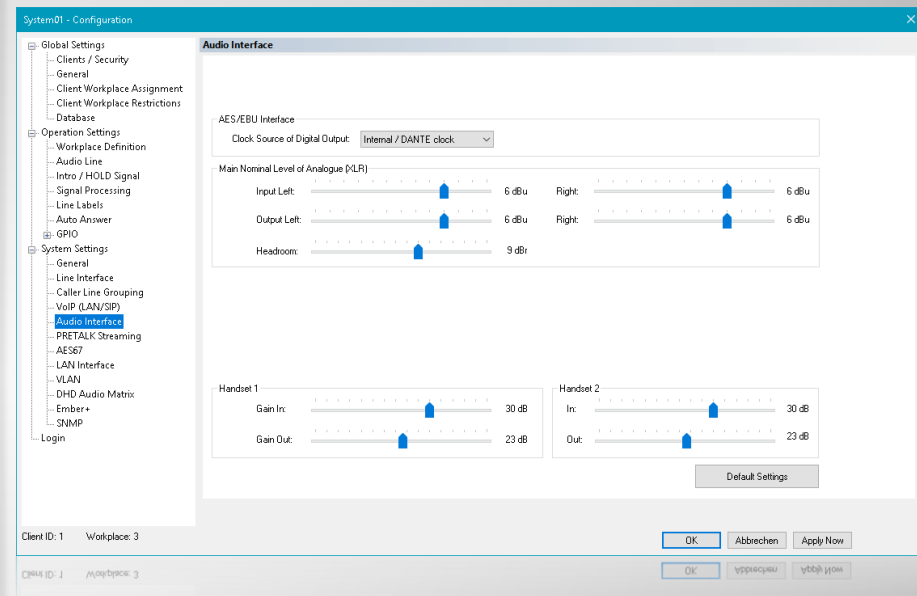
Client ID: 1 Workplace: 3

OK Abbrechen Apply Now

OK Abbrechen Apply Now

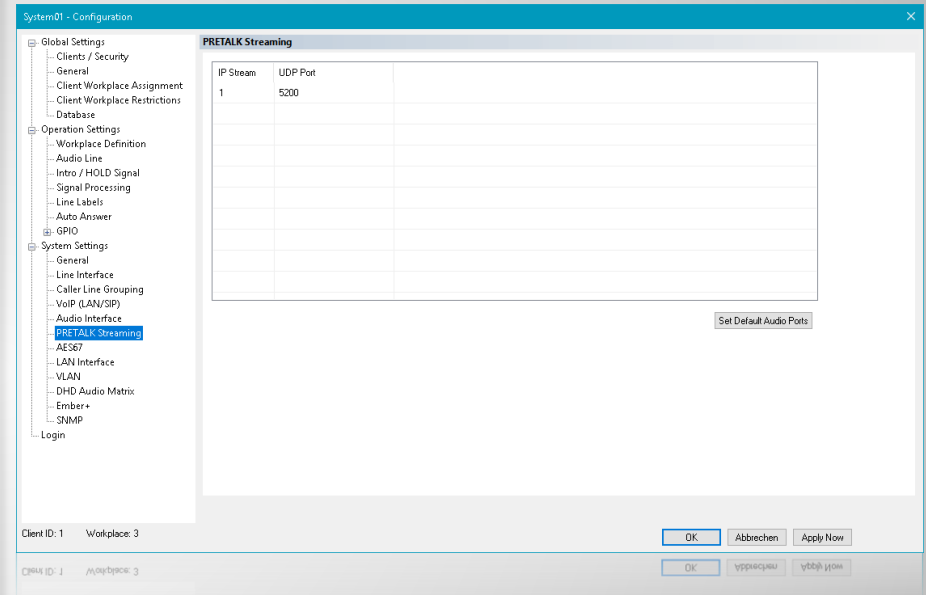
VoIP (LAN/SIP) (2)

- On the AUDIO INTERFACE page, the clock source for the digital audio outputs is set under AES/EBU INTERFACE. If *DANTE* is used, the digital outputs always run synchronously to the *DANTE* clock.
- Under MAIN NOMINAL LEVEL OF ANALOGUE (XLR) the nominal level of the analogue audio interface is set.
- Input and output gain can be set separately for both HANDSETS.



Audio Interface

- On the PRETALK STREAMING page, the UDP audio ports are configured for transferring audio data between *THipPro Intercom* and PC.

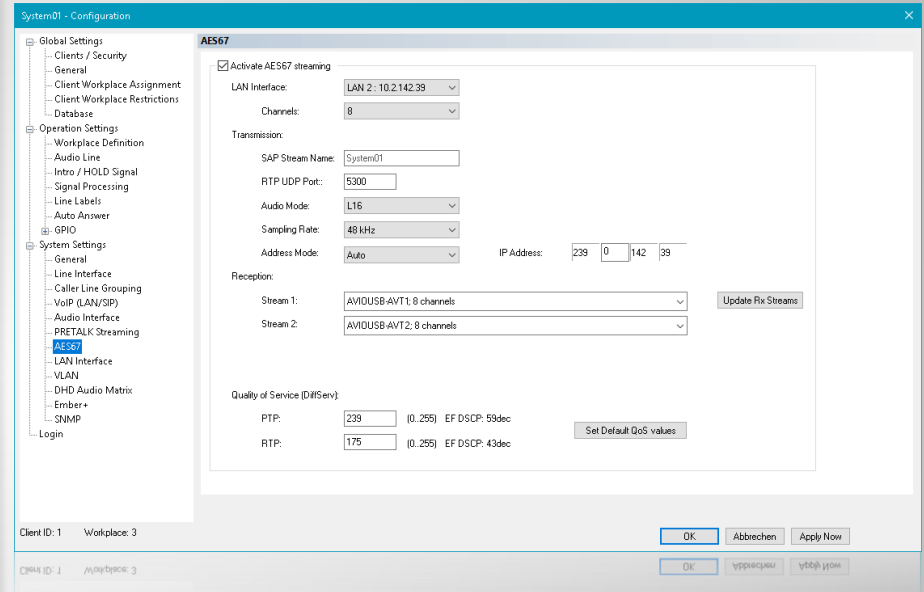


Pretalk Streaming

- The AES67 page is used to configure the transmission of audio data streams over the network (software option AES67 required).
- **ACTIVATE AES67 STREAMING** turns the function on.
 - When activated for the first time, existing AES67 streams are automatically searched for. The process may take some time.
- AES67 streams are distributed via multicast. So that systems do not receive all multicast streams in the network, the switches must support IGMP snooping.
 - The RX data rate in the system monitor shows you quickly whether IGMP is working correctly: RXRate \approx 1.4 Mbit/s x Channels (at L24 and 48 kHz)
- **LAN INTERFACE:** Network interface used for AES67.
- **CHANNELS:** Number of channels needed.

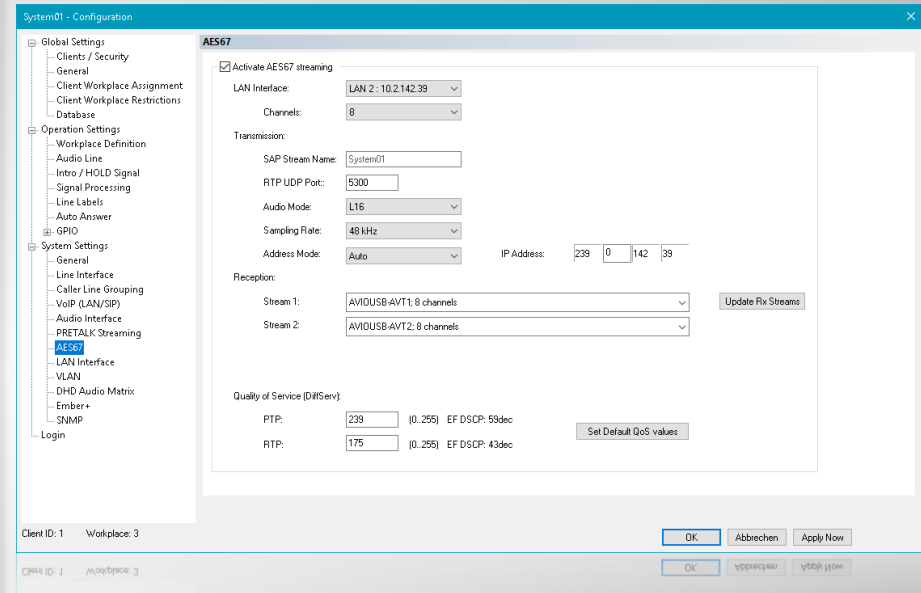
AES67 (1)

- **TRANSMISSION:** Configuration of the TX audio data streams:
 - **SAP STREAM NAME:** Name of the stream identifying it in the network.
 - **RTP UDP PORT**
 - **AUDIO MODE:** L16/L24
 - **SAMPLING RATE:** 32 kHz / 48 kHz
 - **ADDRESS MODE:** Select whether the multicast IP address should be assigned automatically or set manually.
 - **IP ADDRESS:** Multicast IP address



AES67 (2)

- **RECEPTION: Selection of the RX audio data streams:**
 - Filling the list may take some time.
 - The list can be updated using **UPDATE RX STREAMS**.
- **QUALITY OF SERVICE (DIFFSERV)**
 - The values for RTP and PTP must be identical throughout the AES67 network.



AES67 (3)

- Configure the network interfaces on the LAN INTERFACE page. The Device has two LAN interfaces, which can be extended to four via LAN 3/4 module.
- You can assign two additional IP addresses per interface for utilisation in VLANs.
- Configure the STUN server if it is required by the VoIP provider. Check which LAN interface the VoIP line is assigned to.
- Under QUALITY OF SERVICE the DiffServ parameters of the network can be configured.
- For safety reasons, the PC access to the system should be restricted to one interface under ACCESSIBLE FROM in the CONTROL/PRETALK STREAMING section. It can be set on the front display under MENU → SYSTEM SETTINGS → LAN SETTINGS → CTRL LAN INTERFACE as well.

System01 - Configuration

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 - Signal Processing
 - Line Labels
 - Auto Answer
 GPIO
 System Settings
 - General
 - Line Interface
 - Caller Line Grouping
 - VoIP (LAN/SIP)
 - Audio Interface
 - PRETALK Streaming
 - AES67
 LAN Interface
 - VLAN
 - DHD Audio Matrix
 - Embser+
 - SNMP
 Login

LAN Interface

LAN 1 LAN 2

Primary IP Address
 IP Address: 10.4.18.211
 Sub Net Mask: 255.255.255.0
 Default Gateway: 10.4.18.1
 DNS Server: 10.4.18.2

Second IP Address
 IP Address:
 Sub Net Mask:
 Default Gateway:
 DNS Server:

Third IP Address
 IP Address:
 Sub Net Mask:
 Default Gateway:
 DNS Server:

STUN Server Parameters
 STUN Server:
 NAT Keep Alive Interval: 20 sec (5, 60)

Quality of Service (DiffServ)
 Voice: 184 (0, 255) EF DSCP: 46dec
 SIP: 104 (0, 255) EF DSCP: 26dec
 Default Settings

Link Type: Auto ☐ Disable Insufficient LAN Alarm

Control/PRETALK Streaming
 UDP Control Port: 10000 Set Default Port
 Accessible from: LAN 1: 10.4.18.211

Client ID: 1 Workplace: 3 OK Abbrechen Apply Now

Client ID: 1 Workplace: 3 OK Abbrechen Apply Now

LAN Interfaces

- Enable the VLAN functionality on the VLAN page.
- To assign a service to a VLAN select 802.1QTAG in the TPID column. Select NONE to disable VLAN for this service.
- Select the desired priority. 6 (VOICE) is the default for VoIP.
- ENTER VLAN ID in the VID column.
- The IP address of the device in the VLAN is selected on the configuration page of the respective service (VoIP, SNMP, DHD, ...).

System01 - Configuration

VLAN

☒ VLAN

Service	TPID	Priority	VID (128K)
PC Control & PRETALK Streaming	none		
VoIP	802.1QTag	6 (Voice)	30
SNMP	802.1QTag	0 (Default, Best Effort)	31
DHD	none		
Ember+	none		
Remote Light Protocol	none		
AES67	none		

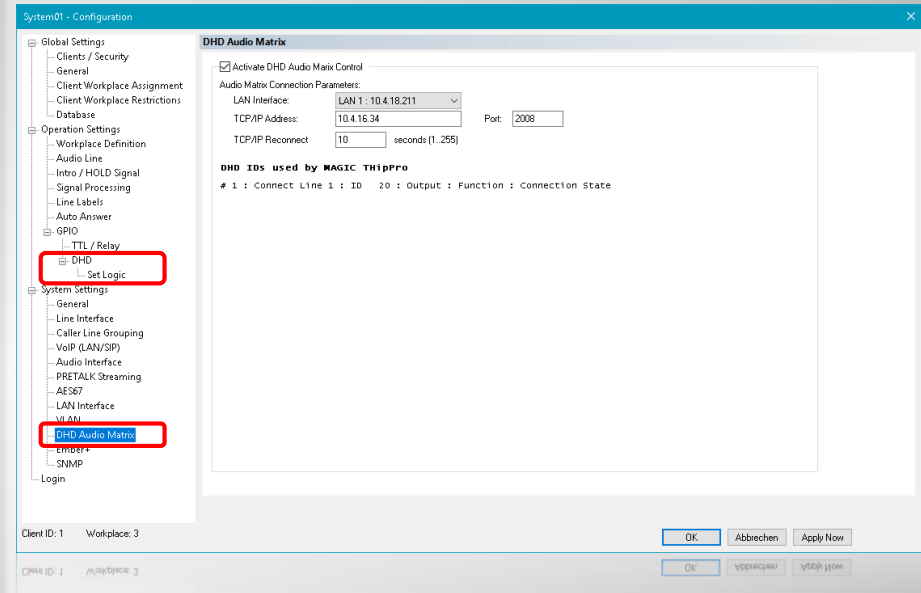
Modification of the VLAN parameters may interrupt the connection to the PCI

Client ID: 1 Workplace: 3

OK Abbrechen Apply Now

VLAN

- On the DHD AUDIO MATRIX page, control and signalling via DHD Set Logic is activated.
- Under LAN INTERFACE, select the LAN interface of the *THipPro Intercom* used for the connection to the DHD core.
- Enter the IP address of the DHD core under TCP/IP ADDRESS.
- The functions and signals are configured on the GPIO → DHD → SET LOGIC page.
- More detailed information can be found in the download area of our website under QUICK GUIDES in the section EMBER+ & DHD SET LOGIC.



DHD Audio Matrix

- On the EMBER+ page, control and signalling via Ember+ is configured. The *THipPro Intercom* can provide the roles PROVIDER and CONSUMER.
- ACTIVATE EMBER+ PROVIDER: Enables the Ember+ provider role:
 - Select the LAN INTERFACE of the *THipPro Intercom*, via which the Ember+ consumers establish a connection to the system.
 - Enter EMBER+ TCP PORTS (default ports: 9000 - 9007) for up to eight Ember+ consumers.
 - The functions and signals are configured on the GPIO → EMBER+ → INPUT / OUTPUT pages.
 - The Ember+ parameter tree also provides functions that can be called directly by consumers.
- ACTIVATE EMBER+ CONSUMER: Enables the Ember+ consumer role:
 - Select the LAN INTERFACE of the *THipPro Intercom*, used for connecting to Ember+ providers.
 - PROVIDER: IP address and TCP port of up to two Ember+ providers.
 - The functions and signals are configured on the page GPIO → EMBER+ → CONSUMER FUNCTIONS.
- More detailed information can be found in the download area of our website under QUICK GUIDES in the section EMBER+ & DHD SET LOGIC.

System01 - Configuration

Global Settings

- Clients / Security
 - General
 - Client Workplace Assignment
 - Client Workplace Restrictions
- Database
- Operation Settings
 - Workplace Definition
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 - Intro / HOLD Signal
 - Signal Processing
 - Line Labels
 - Auto Answer
- GPIO
 - TTL / Relay
 - Ember+**
 - Input
 - Output
 - Consumer Functions
- System Settings
 - General
 - Line Interface
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 - VoIP (LAN/SIP)
 - Audio Interface
 - PRETALK Streaming
 - AESE7
 - LAN Interface
 - VLAN
 - DHD Audio Matrix
 - Ember+**
 - SNMP
 - Login

Ember+

☒ Activate Ember+ Provider

Ember+ Connection Parameters:

LAN Interface: LAN 1: 10.4.18.211

Port 1 (Consumer 1):	9000	Port 4 (Consumer 4):	0	Port 7 (Consumer 7):	0
Port 2 (Consumer 2):	9001	Port 5 (Consumer 5):	0	Port 8 (Consumer 8):	0
Port 3 (Consumer 3):	0	Port 6 (Consumer 6):	0		

☒ Activate Ember+ Consumer

Connection Parameters:

LAN Interface: LAN 1: 10.4.18.211

Provider 1: TCP/IP Address: 10.4.18.212 Port: 9010

Provider 2: TCP/IP Address: Port: 0

Client ID: 1 Workplace: 3

OK Abbrechen Apply Now

Ember+

- SNMP must be activated to integrate the *THipPro Intercom* into a network management system.
- Up to four destinations can be configured.
- The desired ALARM TRAPS can be selected individually, or four categories can be assigned in order to minimise the number of messages in the control centre.
- The necessary MIBs can be found in the installation directory of the *THipPro Intercom* PC software.

The screenshot shows the 'System01 - Configuration' window with the 'SNMP' tab selected. The left sidebar lists various configuration categories, with 'SNMP' highlighted. The main area contains the following settings:

- ☒ **SNMP**
- SNMP Version: v2c
- Read/Trap Community: public / public
- SNMP Port: 161
- NMS 1 (LAN/IP Adr./Port): LAN 1: 10.4.18.211 / 0
- NMS 2 (LAN/IP Adr./Port): LAN 1: 10.4.18.211 / 0
- NMS 3 (LAN/IP Adr./Port): LAN 1: 10.4.18.211 / 0
- NMS 4 (LAN/IP Adr./Port): LAN 1: 10.4.18.211 / 0
- System Description: THipPro Intercom 01
- Contact:
- System Location:
- ☐ Send all traps at system startup
- ☐ Send traps immediately after enabling
- Category A Alas:
- Category B Alas:
- Category C Alas:
- Category D Alas:

On the right, there are two lists of alarm traps with checkboxes:

- System Alarms:** LCA, Time Keeper, Temperature Sensor, FLASH EEPROM, Overhead, MAIN EEPROM, Display Contrast DAC, VDO, VDO Port, Ethernet MAC 1, Ethernet MAC 2, Ethernet MAC 3, Ethernet MAC 4, DSP 2 Boot, DSP 2 Access, Slot 1 Access, Slot 2 Access, Slot 3 Access, Module Assembly, DANTE Module Access.
- Application Alarms:** AES/EBU Framing Input 1, AES/EBU Framing Input 2, AES/EBU Framing Input 3, AES/EBU Framing Input 4.

At the bottom right of the window are buttons for 'Set All' and 'Clear All'. The bottom status bar shows 'Client ID: 1' and 'Workplace: 3'.

SNMP

MAGIC THipPro

Intercom

Operation Settings

Settings that can be loaded using a PRESET.

- On the WORKPLACE DEFINITION page, the system's lines can be distributed across six workplaces.
- Each line can be assigned to several workplaces.

Workplace Definition

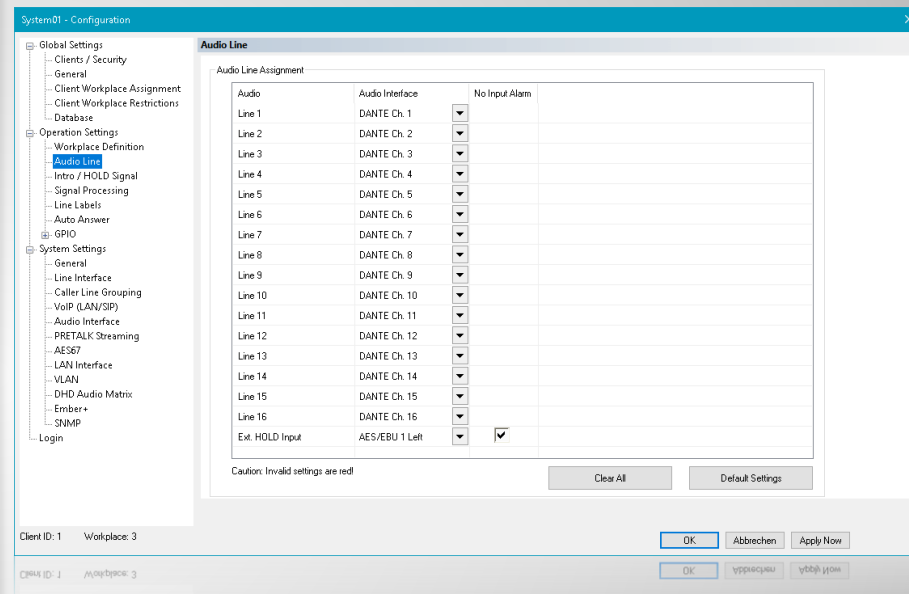
Name	Channels
	1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16
Line Groups	6 3 4
Workplace 1	<input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/>
Workplace 2	<input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/>
Workplace 3	<input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/>
Workplace 4	<input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/>
Workplace 5	<input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/>
Workplace 6	<input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/>

Client ID: 1 Workplace: 3

OK Abbrechen Apply Now

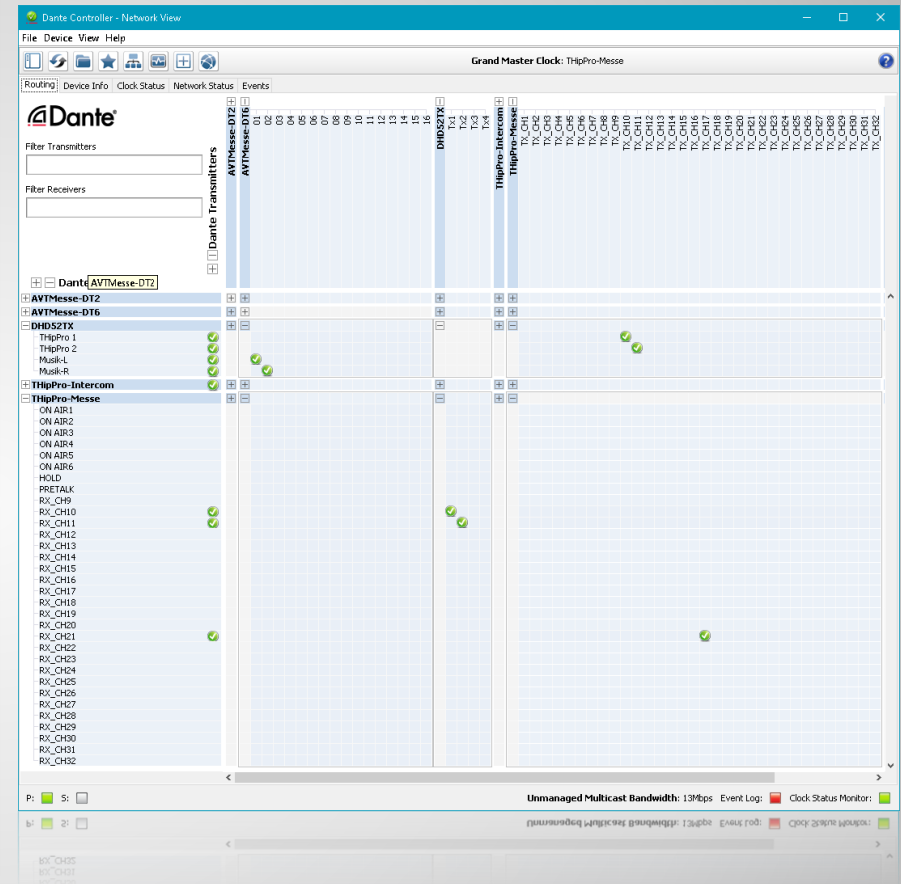
Workplace Definition

- On the AUDIO LINE page the audio interfaces are assigned to the telephone lines.
- It is possible to use the device audio interfaces and DANTE at the same time. However, DANTE and AES67 cannot be mixed.
- If an AES interface is only used as an output, the INPUT ALARM can be deactivated.



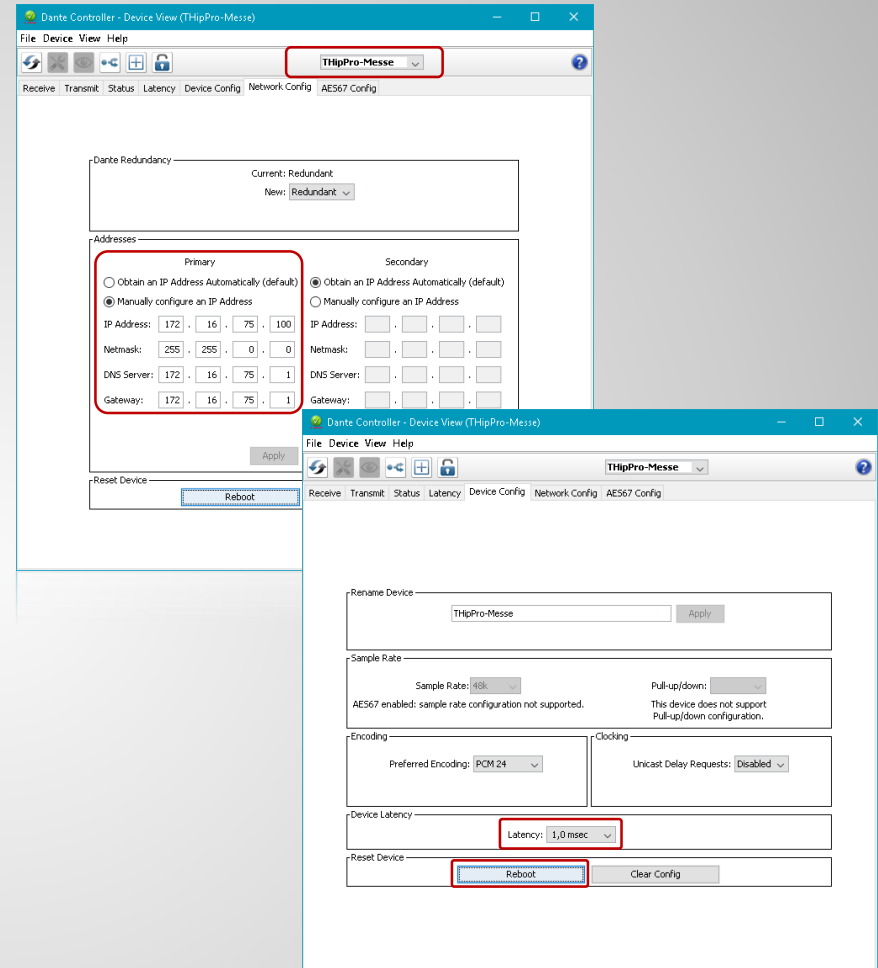
Audio Line

- After starting the DANTE CONTROLLER software, NETWORK VIEW - ROUTING automatically displays all devices that support the Dante protocol.
- The inputs and outputs of the systems can be assigned to each other via the matrix.



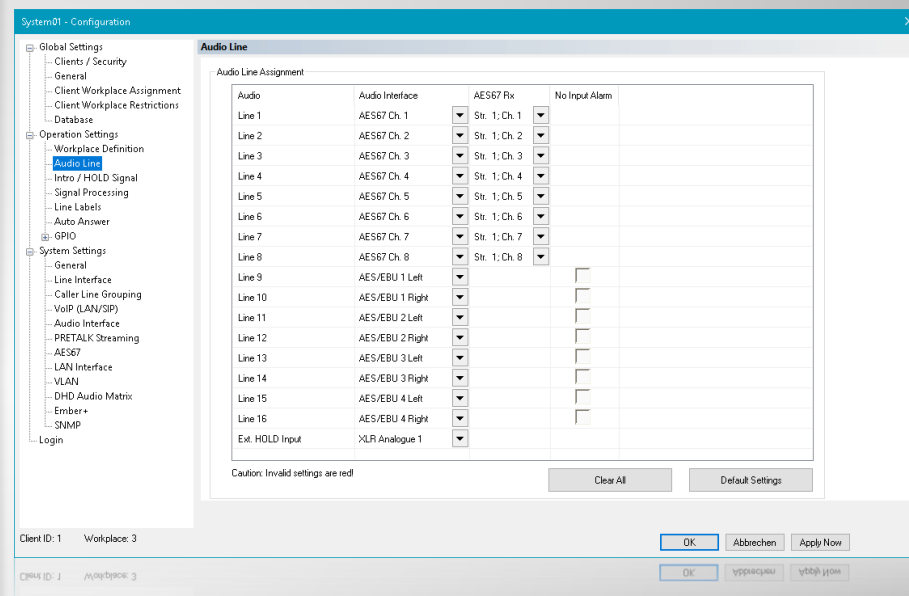
Audio Line / DANTE (1)

- In the DANTE CONTROLLER software, the Ethernet interfaces can be configured in the DEVICE VIEW under NETWORK CONFIG, if necessary.
 - Assign IP address automatically (default setting)
 - Manual adjustment
- It is also essential to correctly configure the maximum expected latency in the network, which should be identical for all Dante devices.
- Attention: After a REBOOT MAGIC THipPro may have to be switched off/on if a DSP alarm appears in the display.



Audio Line / DANTE (2)

- If AES67 audio data streams are used, the AES67 TX channels must be assigned to the lines in the AUDIO INTERFACE column.
- In the AES67 RX column, the AES67 RX channels are assigned to the lines.
- It is possible to use device audio interfaces (analogue and digital) and AES67 simultaneously. However, DANTE and AES67 cannot be mixed.



Audio Line (AES67)

- Automatic announcements and the HOLD function are configured on the Intro / HOLD Signal page.
- In the HOLD SIGNAL area, SIGNAL SOURCE defines what the caller hears when the line is held.
 - NOT USED: The hold function is not available. The corresponding key in the main window is not displayed.
 - INTRO LOOP: The announcement is played in an endless loop.
 - ASSOCIATED LINE: The caller hears the signal of the audio interface assigned to the telephone line.
 - <AUDIO INTERFACE>: The caller hears a signal which is fed to a separate audio interface (to be set under AUDIO LINE).
- PLAY INTRO WHEN SWITCHING TO HOLD: If a call is switched to HOLD, the announcement starts playing.
- NUMBER OF ANNOUNCEMENTS: Number of repetitions of the announcement (1...4).
- PAUSE TIME AFTER ANNOUNCEMENT: Pause after each announcement (0...7 seconds).

System01 - Configuration

Intro / HOLD Signal

☒ Play Intro after connection established
☐ Skip Intro when call is accepted manually

Number of Announcements: 2
 Pause time after announcement: 1 second

HOLD Signal
 Signal Source: Associated Line
☒ Play Intro when switching to HOLD
 Number of Announcements: 1
 Pause time after announcement: without pause

Speech Synthesis
 Generate Intros from Intro Info
 Generate Intros from Line Labels

Stored Intros

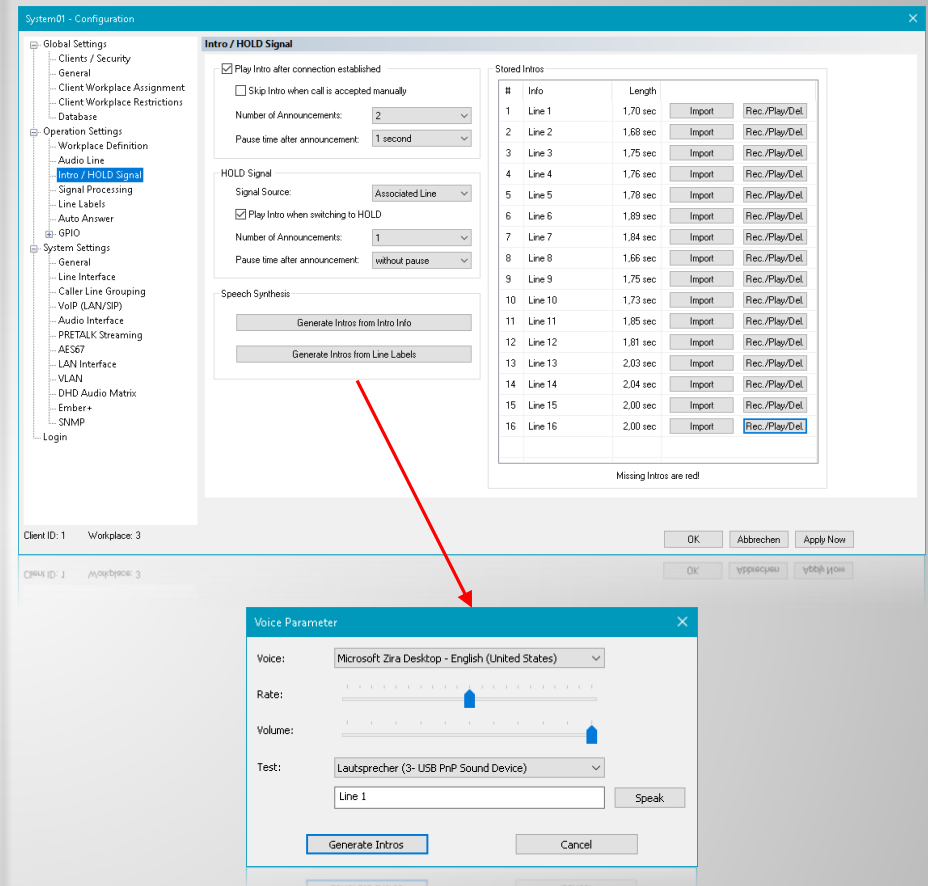
#	Info	Length	Import	Rec./Play/Del
1	Line 1	1.70 sec	Import	Rec./Play/Del
2	Line 2	1.68 sec	Import	Rec./Play/Del
3	Line 3	1.75 sec	Import	Rec./Play/Del
4	Line 4	1.76 sec	Import	Rec./Play/Del
5	Line 5	1.78 sec	Import	Rec./Play/Del
6	Line 6	1.89 sec	Import	Rec./Play/Del
7	Line 7	1.84 sec	Import	Rec./Play/Del
8	Line 8	1.66 sec	Import	Rec./Play/Del
9	Line 9	1.75 sec	Import	Rec./Play/Del
10	Line 10	1.73 sec	Import	Rec./Play/Del
11	Line 11	1.85 sec	Import	Rec./Play/Del
12	Line 12	1.81 sec	Import	Rec./Play/Del
13	Line 13	2.03 sec	Import	Rec./Play/Del
14	Line 14	2.04 sec	Import	Rec./Play/Del
15	Line 15	2.00 sec	Import	Rec./Play/Del
16	Line 16	2.00 sec	Import	Rec./Play/Del

Missing Intros are red

Client ID: 1 Workplace: 3 OK Abbrechen Apply Now

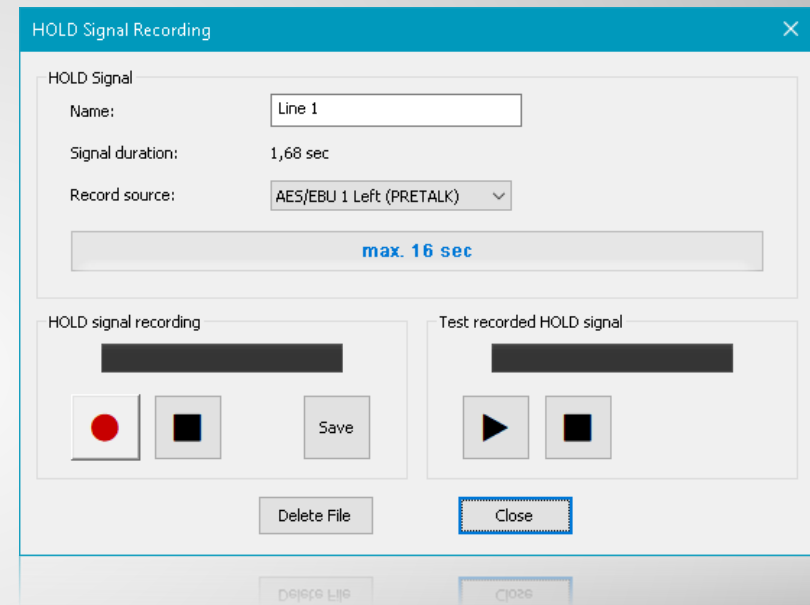
Intro / HOLD Signal (1)

- PLAY INTRO AFTER CONNECTION ESTABLISHED: The device plays an announcement to the caller after the connection has been established:
 - SKIP INTRO WHEN CALL IS ACCEPTED MANUALLY: The announcement is not played if a call was answered manually.
 - NUMBER OF ANNOUNCEMENTS: Number of repetitions of the announcement (1...4).
 - PAUSE TIME AFTER ANNOUNCEMENT: Pause after each announcement (0...7 seconds).
 - STORED INTROS: A separate announcement can be stored on the device for each line.
 - #: Number of the line.
 - INFO: Info on the currently stored greeting. This text can be automatically converted into an audio file via speech synthesis (SPEECH SYNTHESIS).
 - LENGTH: Length of the announcement.
 - IMPORT: A pre-produced announcement can be imported in the formats wav or mp3. REC/PLAY/DEL: Opens a window to record, play and delete announcements.
 - SPEECH SYNTHESIS: The announcements can be created automatically by the voice synthesis function contained in the Windows operating system.
 - GENERATE INTROS FROM INTRO INFO: The text in the INFO column is converted into an audio file for each line and stored on the system.
 - GENERATE INTROS FROM LINE LABELS: The text defined on the LINE LABELS page for each line is converted into an audio file and stored on the system.



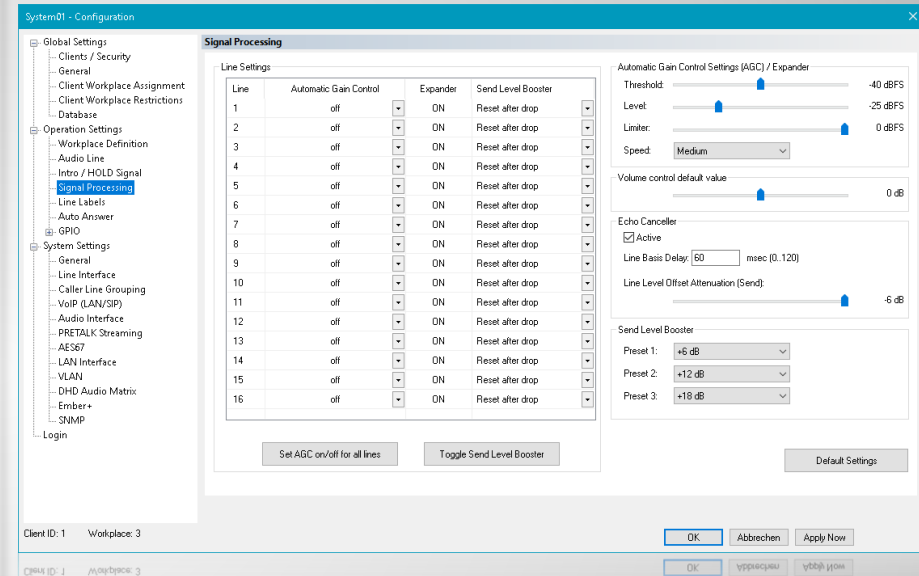
Intro / HOLD Signal (2)

- The REC/PLAY/DEL key opens a window in which the announcement of a line can be recorded, played back, saved or deleted.
- NAME: About the announcement.
- SIGNAL DURATION: Length of the announcement.
- RECORD SOURCE: Audio interface of the THipPro intercom system from which recording is being made. This interface is also used to listen to the recording.
- Progress bar: An announcement can be a maximum of 16 seconds long.
- HOLD SIGNAL RECORDING:
 - Level display.
 - Record button: Starts recording.
 - STOP: Stops the recording.
 - SAVE: Saves the announcement in the device.
- TEST RECORDED HOLD SIGNAL:
 - PLAY: Starts playback of the greeting.
 - STOP: Stops the playback of the greeting.
 - DELETE FILE: Deletes the announcement from the system.



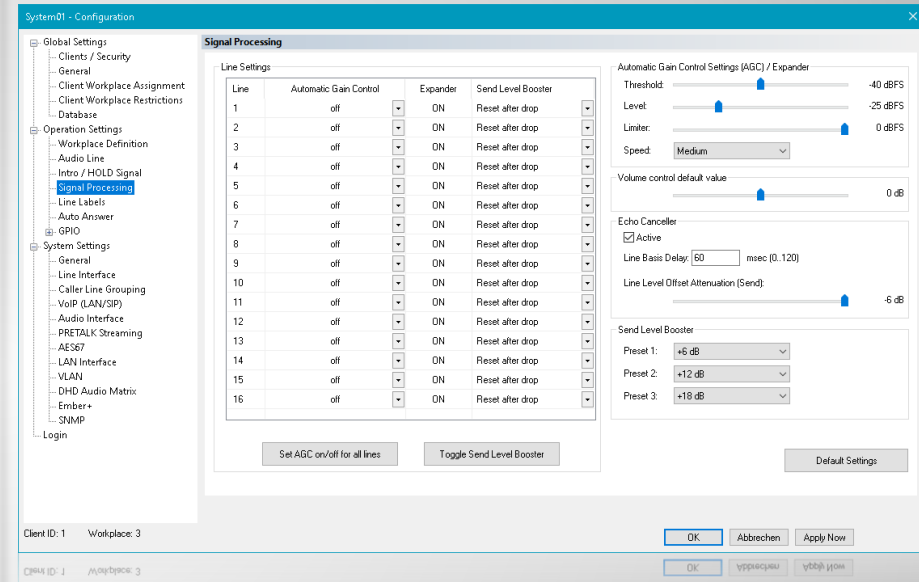
Intro / HOLD Signal (3)

- Audio processing is configured on the SIGNAL PROCESSING page.
- AGC (AUTOMATIC GAIN CONTROL) equalises the volume of the received audio signal.
- The EXPANDER is enabled to eliminate background noise.
- The SEND LEVEL BOOSTER raises the level of the outgoing audio signal on the telephone line. The behavior can be configured for each line:
 - RESET AFTER DROP: The gain is reset after hanging up.
 - KEEP BOOST VALUE: The gain is retained for the next connection.
- SEND LEVEL BOOSTER defines three gain levels that are available in the main window after clicking on the level meter.



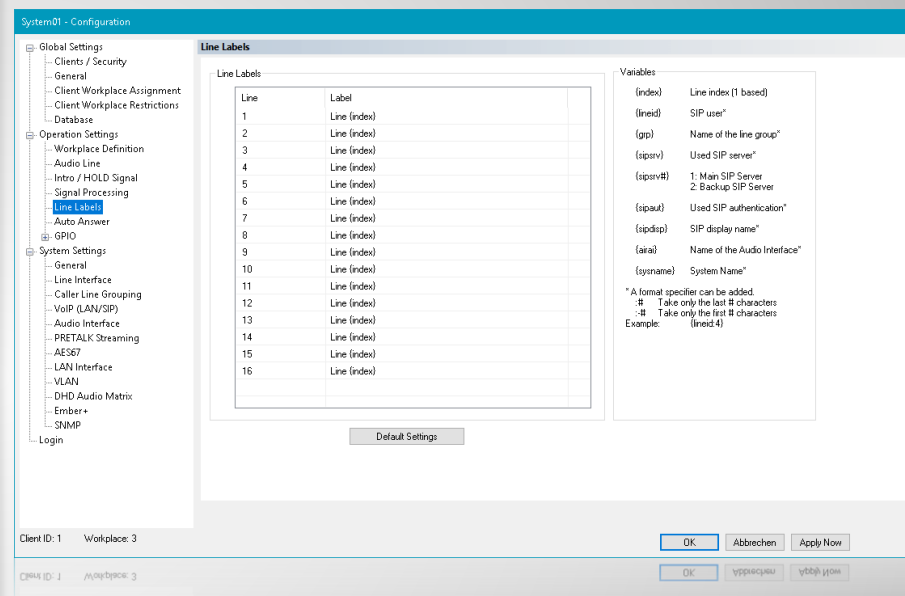
Signal Processing (1)

- **AUTOMATIC GAIN CONTROL SETTINGS (AGC) / EXPANDER** is used to configure the behavior of AGC and Expander.
 - **THRESHOLD:** Below this threshold the expander attenuates the signal to suppress background noise. Above this threshold, the AGC alters the audio signal.
 - **LEVEL:** This is the average level at which the AGC tries to bring the received audio signal. The AGC can dynamically adjust the gain from -16 dB to +16 dB.
 - **LIMITER:** Regardless of the AGC speed, the limiter can quickly lower the level above this threshold.
 - **SPEED:** The AGC speed can be set from SLOW to VERY FAST.
- **VOLUME CONTROL DEFAULT VALUE** raises or lowers the level of the audio signal received from all telephone lines.
- The **ECHO CANCELLER** circuit eliminates received line echoes.
 - Echoes received with a delay of 0 - 120 ms can be suppressed by the Echo Canceller.
 - Especially with VoIP, however, longer delays can occur. In this case, the working range of the echo canceller can be shifted via the LINE BASIS DELAY.



Signal Processing (2)

- On the LINE LABELS page, the headings of the lines on the main window are defined.
- The following placeholders are available:
 - {index}: consecutive line number
 - {lineid}: SIP User
 - {grp}: Line group name
 - {sipsrv}: SIP Server
 - {sipsrv#}: Index of the active SIP Servers (1=Main; 2=Backup)
 - {sipaut}: SIP Authentication
 - {sipdisp}: SIP display name of the line
 - {airai}: Audio interface used
 - {sysname}: System name
 - The length of the label can be restricted:
 - :-# Only the first # characters are displayed.
 - :# Only the last # characters are displayed.



Line Labels

- The Auto Answer mode can be activated for all lines or only for selected lines on the AUTO ANSWER page.
- ANSWER CALL ON determines how the call is accepted:
 - ASSOCIATED LINE: The call is switched through directly.
 - HOLD: The connection is put in hold.
 - LINE GROUP SELECTION: For each line group it is possible to choose between ASSOCIATED LINE and HOLD.
- Set the delay with which the call is automatically accepted under AUTO ANSWER DELAY.

System01 - Configuration

Auto Answer

☐ Auto Answer on all channels

Channel Index	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
Line Groups	1				2				3				4			
Auto Answer Channel	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>

Answer call on: Line Group selection

Auto Answer Delay: 1 sec

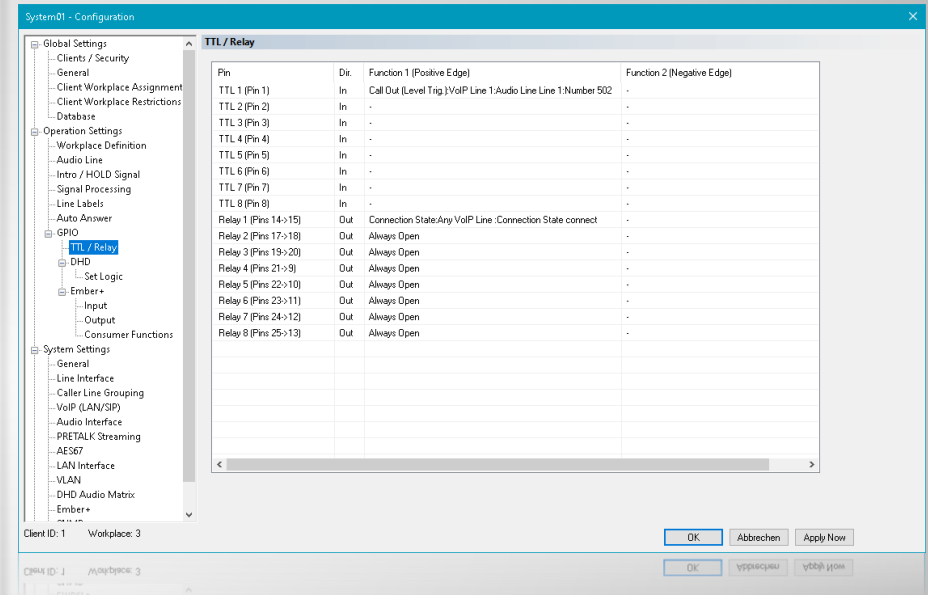
Line Gp. 1	Line Gp. 2	Line Gp. 3	Line Gp. 4	Line Gp. 5	Line Gp. 6	Line Gp. 7	Line Gp. 8
1	2	3	4	5	6	7	8
HOLD	HOLD	Associated Line	Associated Line	Associated Line	Associated Line	Associated Line	Associated Line

Client ID: 1 Workplace: 3

OK Abbrechen Apply Now

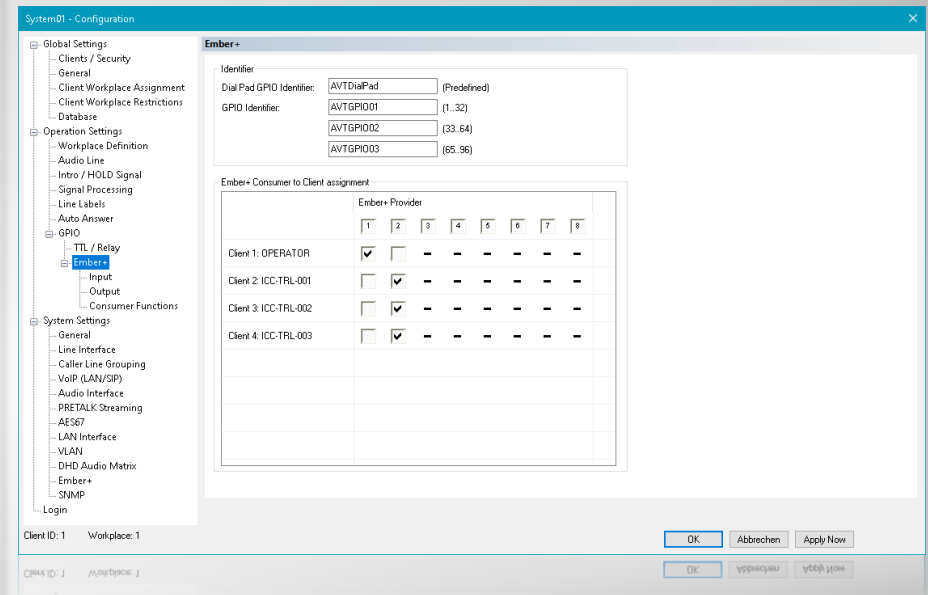
Auto Answer

- Under GPIO, functions for controlling the device and signals for displaying the system status can be configured.
- Functions and signals are available via TTL/Relay contacts as well as via “DHD Set Logic” and “Ember+”.
- The list shows an overview of the configured functions and signals.
- Double-clicking on a line opens the configuration of the GPIO.
- More detailed information can be found in the download area of our website under QUICK GUIDES in the section EMBER+ & DHD SET LOGIC. (These documents also describe all TTL/Relay functions and signals.)



GPIO

- If an external keypad is to be used via Ember+, a corresponding DIAL PAD GPIO IDENTIFIER must be defined.
 - The functions required to implement a keypad are already predefined and do not have to be created individually.
 - The individual dial keys are implemented as GPI functions.
- For further GPIO functions three GPIO blocks with 32 input and 32 output functions each are available for which the corresponding GPIO IDENTIFIERS must be assigned.
- EMBER+ CONSUMER TO CLIENT ASSIGNMENT: Assign client PCs to a provider to display a phone number entered via Ember+ in the PC software.



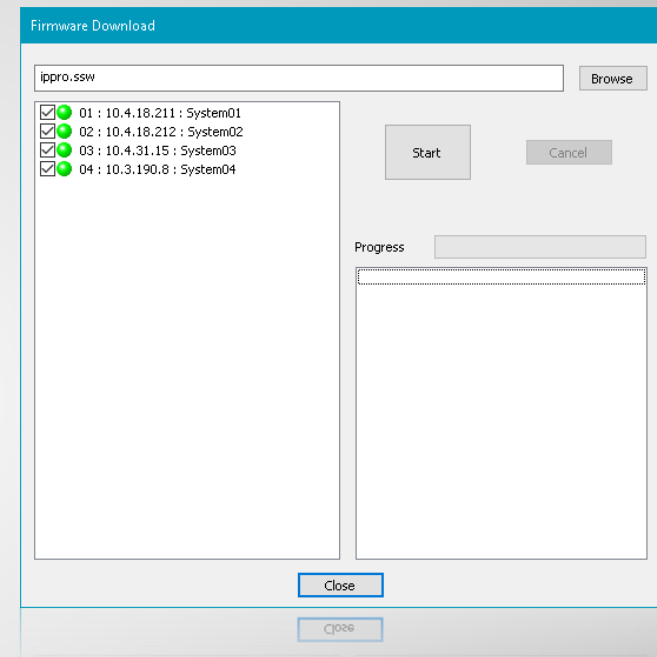
Ember+

MAGIC THipPro

Intercom

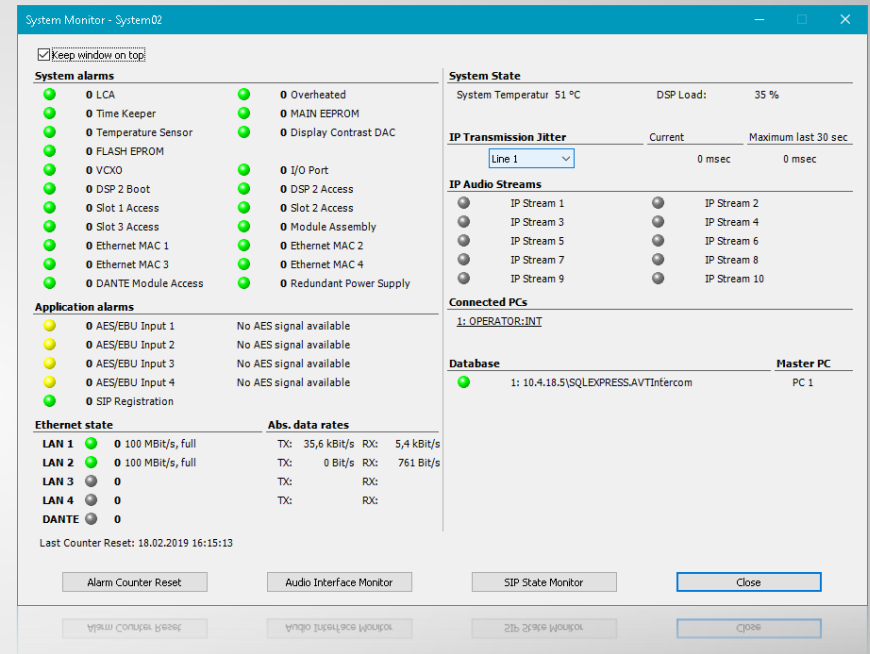
Extras

- The appropriate firmware is supplied with each PC software version and is stored in the installation directory of the application during installation.
- If the firmware version of a device does not match the PC software, a request to update the firmware appears when establishing a connection with this device.
- Via ADMINISTRATION → SYSTEM X → FIRMWARE DOWNLOAD the appropriate firmware can be loaded onto the *THipPro Intercom* System.
- A list of all connected systems is displayed. Check all devices to be updated.
- These devices will be updated after pressing the START button without further user interaction.



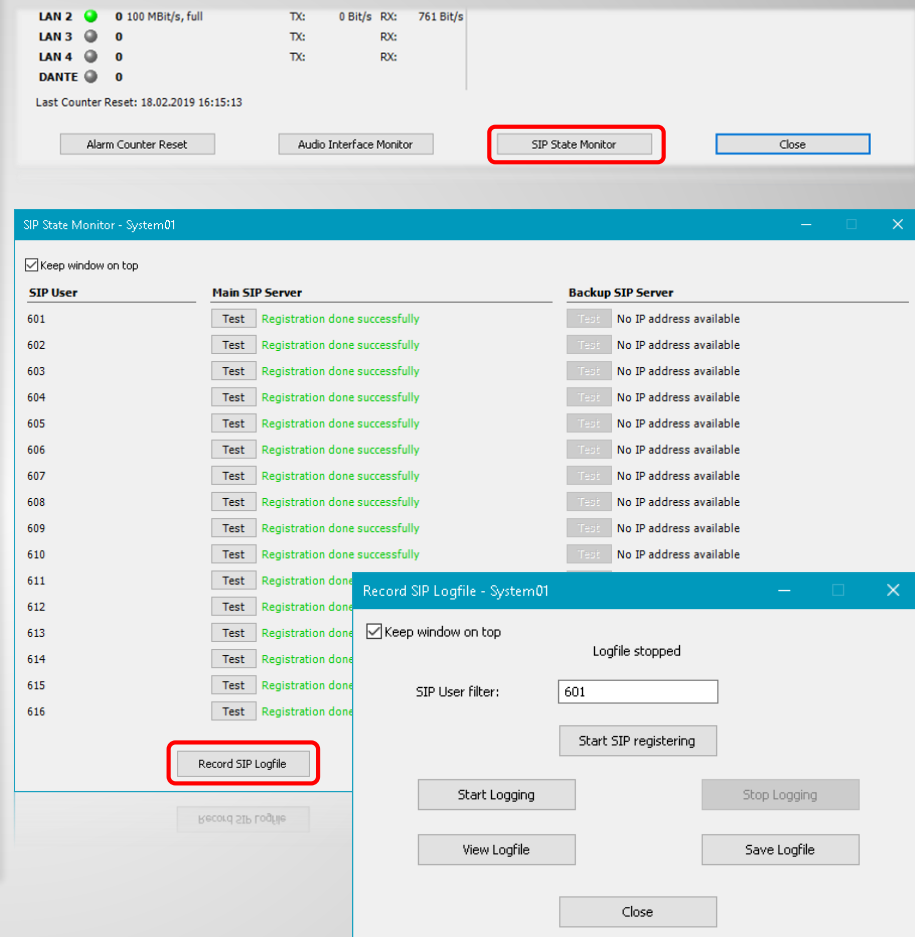
Firmware Update

- The detailed system status is displayed via EXTRAS → SYSTEM X → SYSTEM MONITOR:
 - Green LED: OK
 - Yellow LED: Warning
 - Red LED: Alarm
- For each alarm LED, an error counter provides information on the frequency of the error.
 - Use ALARM COUNTER RESET to reset the error counters.
- In addition, other important system information such as system temperature, processor load, network load, etc. is displayed.



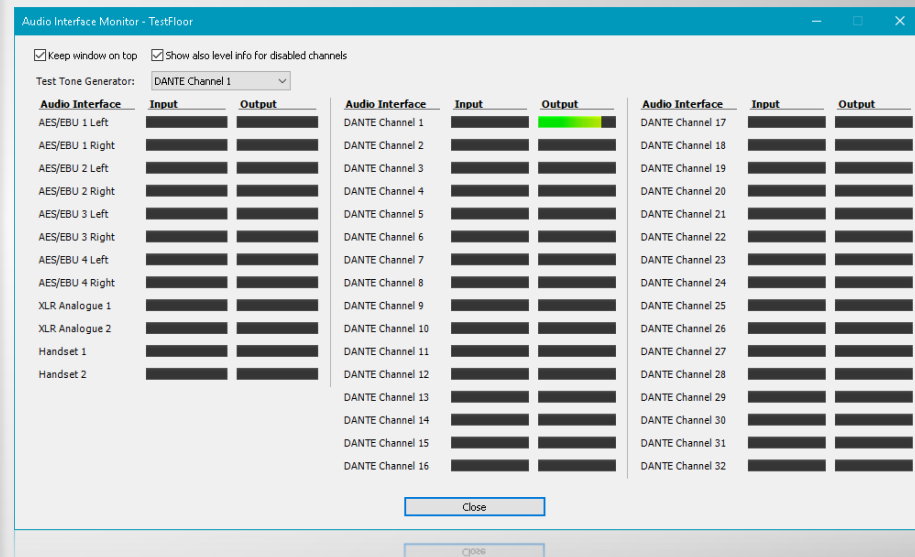
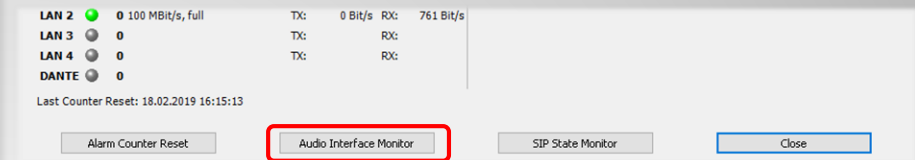
System Monitor

- Open the SIP STATE MONITOR by pressing the corresponding key in the SYSTEM MONITOR window.
- Here the SIP registration of all VoIP lines can be checked and tested.
- SIP communication can be recorded via RECORD SIP LOGFILE:
 - SIP USER FILTER is used to restrict logging to SIP packets containing the filter string (e.g. a phone number or the SIP USER NAME).
 - START LOGGING.
 - STOP LOGGING.
 - VIEW LOGFILE loads the log file from the device and displays it in an editor.
 - SAVE LOGFILE saves the file on the PC.
 - If you want to investigate problems with the SIP registration, press the START SIP REGISTERING key after starting the logging.



SIP Status Monitor

- Open the AUDIO INTERFACE MONITOR by pressing the corresponding key in the SYSTEM MONITOR window.
- All audio interfaces can be monitored here:
 - The input/output levels of all configured audio interfaces are displayed.
 - To display the audio levels of the currently unused audio interfaces, activate SHOW ALSO LEVEL INFO FOR DISABLED CHANNELS.
 - To output a test tone, select the desired audio interface under TEST TONE GENERATOR.



Audio Interface Monitor

- Via ADMINISTRATION → SYSTEM X → REGISTRATION you can check which SOFTWARE OPTIONS are available in your system.
- To activate optional system functionality, your you will be provided with a password key.
 - This key is calculated on the basis of the device FACTORY-NUMBER, which you need to send us together with the order.
 - Click ENTER PASSWORD to enter the key.
 - The option will then be marked as available in the list.
 - Restart the system to make sure the new functionality is fully operative.

The screenshot shows the 'Registration' window with the following fields and options:

- Hardware:** MAGIC THipPro
- Main:** DANTE, AEC 16 Module
- Subject Number:** 450063
- Factory Number:** 17/44/1193
- Year:** 2017
- Hardware Version:** 3.00
- MAC Address 1:** 00-06-9B-02-23-09
- MAC Address 2:** 00-06-9B-02-23-0A
- Modules:** free (dropdown), DANTE (dropdown), AEC16 (dropdown)
- Features:**
 - Software Options:**
 - ☒ MAGIC THipPro Intercom Basic
 - ☒ HD Voice
 - ☒ VoIP channels 16
 - ☒ Number of Intercom Client licenses 5
 - ☒ Number of PRETALK Audio Streams 1
 - ☒ AES67

Buttons: Enter password, Close, Close, Close

Registration

- HELP → About MAGIC *THipPro Intercom* displays the versions of the PC software and the firmware versions of the devices.



Version Information

MAGIC THipPro

Intercom

Annex

- Command line interface under ADMINISTRATION → SYSTEM PANEL:
 - Reset the system:
 - reset
 - PING using the desired network interface
 - ping [-i<LAN-IF>] [-n<IP-NETWORK>] [-v<VLAN-ID>] <IP-Address/Name>
 - LAN-IF: 1, 2, 3, 4 for LAN1, LAN2, LAN3, LAN4; Default: 1
 - IP-NETWORK: local IP address, 1:Primary, 2:Second, 3:Third; Default: 1
 - VLAN-ID: 0...4096; Default: 0
 - Display the storage location of the Local Settings:
 - showprofilepath

Tips & Tricks (1)

- SIP configuration:

- A non-standard SIP port is entered by appending the port to the SIP server with ':'.
▪ If the SIP provider requires the specification of a proxy server, it can be placed in front of the SIP server using '@':

- Example:

- Proxy server `fs1.ims.swisscom.ch`
- SIP server / Domain / Realm `swisscom.ch`
- SIP port `5070`
- → Configure under SIP Server
`fs1.ims.swisscom.ch@swisscom.ch:5070`

Tips & Tricks (2)

Web: www.avt-nbg.de

Email: support@avt-nbg.de

Phone: +49 911 5271-110

Support