MAGIC THipPro Intercom

Configuration Guide

Version 3.070 (18 September 2019) © 2019, AVT Audio Video Technologies GmbH



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



MAGIC THipPro Intercom

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.

Slot	Туре	Status	Parameter
1	UDP	open	10.4.18.211; 10000
2	UDP	open	10.4.18.212; 10000
3	UDP	open	10.4.31.15; 10000
4	UDP	open	10.3.190.8; 10000
5			
6			
7			
8			
9			
10			
Load	Save	Up	Down Add/Edit Delete All
		ОК	Cancel

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 Int	erface Parameter		×
Interface: Parameter	UDP	~	
Interface:	<default></default>		\sim
IP Address:	10.4.18.211		
Port:	10000		
	OK	Cancel	
	OV		

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 (**0**) 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:
 - (②) Telephone book, Redial list and the input field for the telephone number.
 - (**⑤**) Keys for the Preallocation of lines.
 - (④) Quick dial keys and call history list. Click on the list title to switch between the two views.
 - (G) The WORKPLACE SELECTION key to switch the view to an alternative workplace configuration.

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																QUICK DIALS
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Main Window



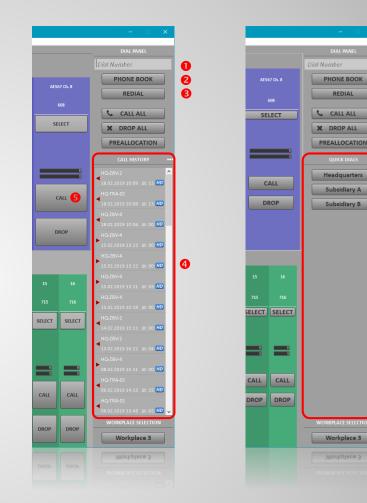
- The title bar can be hidden using LOCAL SETTINGS → APPLICATION PARAMETERS.
- Clicking the menu button (①) displays the menu as a sidebar.
- Without the title bar, the window can no longer be moved with the mouse.
- To temporarily show the title bar, press the SCROLL LOCK key on the keyboard.



Borderless Main Window



- A telephone connection can be established in various ways:
 - Enter the phone number in the phone number field (①) and press the CALL key (⑤) on the desired line.
 - Select a phone book entry (2). The phone number is then transferred to the phone number field. Press the CALL key (3) on the desired line.
 - Select a redial number (③) and press the CALL key (⑤) on the desired line.
 - Select a number from the CALL HISTORY list (④) and press the CALL key (⑤) on the desired line.
 - Press the CALL key (⑤) 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 (③) and press the CALL key (⑤) on the desired line.

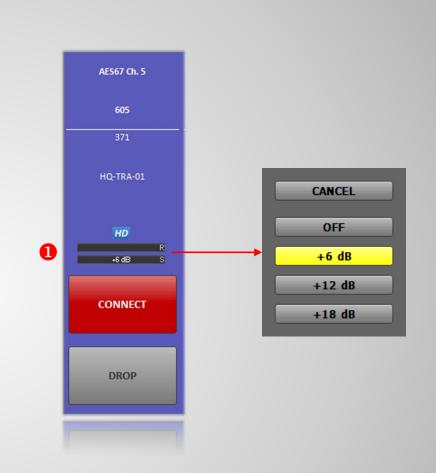


Call set-up



6

- 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 (●) 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



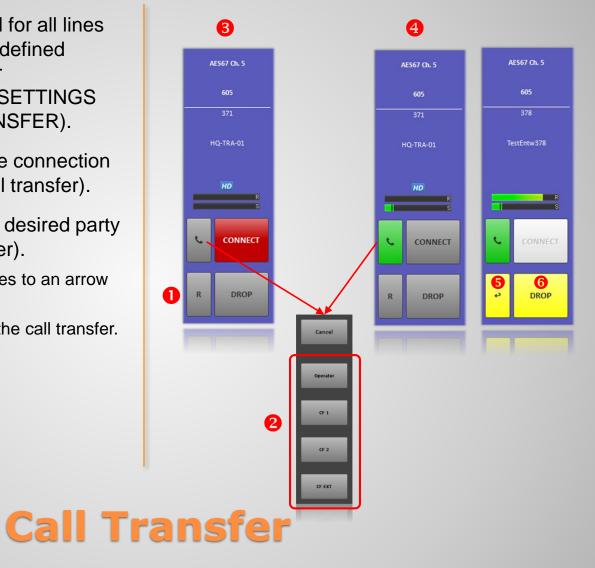
- A key (●) to hold the connection can be displayed for all lines (activate under SYSTEM CONFIGURATION → INTRO / HOLD SIGNAL).
- The position of the key depends on whether the pretalk function is activated.
- The received audio signal will no longer be output to the audio interface.
- The following signals 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.







- An R key (●) can be displayed for all lines to transfer calls to up to 10 predefined destinations (●) (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
 (G). The call can be retrieved.
 - The DROP key (**(**) completes the call transfer.



- A telephone connection can also be prepared and established at a later time.
- The SELECT (**0**) key opens the phone book:
 - An entry can be searched for (2) and selected (3) here and the preallocation can be completed by pressing the SELECT key 3().
 - If several numbers are stored for an entry, the preallocation is completed by clicking on the desired number (⁶).
 - If the desired number is not yet stored in the phone book, a new entry must be created using the NEW (③) key.
 - The CLEAR key (②) deletes the preallocation on this line.
- Preallocation is also possible without a database:
 - The SELECT (①) key opens a keypad.
 - Here you can enter a number and complete the preallocation by pressing the SELECT key.
 - The CLEAR key (②) deletes the preallocation on this line.



Preallocation of lines (1)



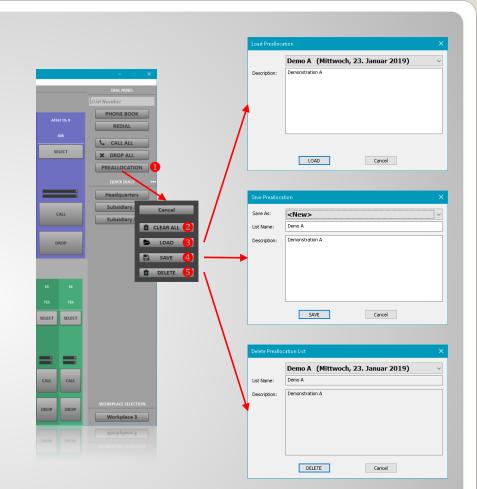
- The CALL ALL (①) key is used to establish telephone connections on all preallocated lines of all *THipPro Intercom* systems.
- The DROP ALL (2) key terminates all active telephone connections on all *THipPro Intercom* systems.

	e 3 : OPERATO		HipPro Interco Extras Help	m - System02	: VoIP											- • ×
			ubsidiary B	Subsidiary C	C Subsid	kary D										DIAL PANEL
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60									505		506		607			REDIAL
		_						_				_				CALL ALL
50			02 28V-2		281/3		-281-4		516	SE	LECT	SE	LECT	SI	LECT	X DROP ALL
	84-1							SUB								PREALLOCATION
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R S	CALL	CALL DROP	CALL	CALL DROP	CALL	CALL	CALL	CALL	CALL	CALL	CALL	CALL	CALL	CALL DROP		

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 (①):
 - CLEAR ALL (②) clears the preallocations of all *THipPro* Intercom systems.
 - LOAD (③) displays all preallocation records stored in the database.
 - SAVE (④) 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 (^(G)).



Preallocation of lines (3)



MAGIC THipPro Intercom

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.

oplication Parameters	Quick Dials / Call Transfer	PRETALK Streaming	Settings Folder			
Main Window Present	ation					
Screen resolution:	1920*1170					
Main window size:	Custom	~	Custom window size:	1400 * 1100		
Layout			System Appearance			
Columns:	1 ~ B	ows: 2 🗸 🗸	Rows per System:	1 ~		
Hide units with	nout configured IP address		Caller lines per row:	min. 8 🗸 🗸		
			Space between rows:	0		
Side-Bar size:	200					
	porarily toggle with [Scroll-Li	ockl keu)				
Show window on		Y: 0				
		1. 0				
Page labels		age 4: Subsidiary C				
Page 1:			Page 7:	Page 10:		
Page 2:	Subsidiary A Pa	age 5: Subsidiary D	Page 8:			
Page 3:	Subsidiary B Pa	age 6:	Page 9:			
Play Wave File						
On Incoming Call:	Ring.wav			Browse		
On Remote Drop:	C:\Users\Public\AVT\MA	DIC THE Dis laters and	\	Browse		
On Local Drop:	%d in the wave file name w	-				
	%u in the wave life halfle w	ili be replaced by the lif	ne index (116) of the line w	nich was uroppeu.		
Logfile						
Logfile Folder:	C:\Users\Public\AVT\MA	GIC THipPro Intercom	Logfiles	Browse	Open	
File Name Format:	<folder>\log-<day mont<="" of="" td=""><td>n>.txt (default)</td><td>~</td><td></td><td></td><td></td></day></folder>	n>.txt (default)	~			
			_			
			L	OK Abbrechen		

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.

al Settings		
plication Parameters	Quick Dials / Call Transfer PRETALK St	treaming Settings Folder
Main Window Presen	tation	
Screen resolution:	1920*1170	
Main window size:	Custom	✓ Custom window size: 1400 * 1100
Layout		System Appearance
Columns:	1 ~ Rows: 2	✓ Rows per System: 1 ✓
Hide units wit	hout configured IP address	Caller lines per row: min. 8 ~
		Space between rows: 0
Side-Bar size:	200	
🗹 Show title bar (ter	nporarily toggle with [Scroll-Lock] key)	
Show window on	fixed position 🛛 👋 0 🖓 10	
Page labels		
Page 1:	Headquarters Page 4: Subs	sidiary C Page 7: Page 10:
Page 2:	Subsidiary A Page 5: Subsi	sidiary D Page 8:
Page 3:	Subsidiary B Page 6:	Page 9:
Play Wave File		
On Incoming Call	Ring.wav	Browse
🗌 On Remote Drop	C:\Users\Public\AVT\MAGIC THipPro I	Intercom\Warning &d.way Browse
On Local Drop:		d by the line index (116) of the line which was dropped.
Logfile		
Logfile Folder:	C:\Users\Public\AVT\MAGIC THipPro I	Intercom\Logfiles Dpen
	<folder>\log-<day month="" of="">.txt (default)</day></folder>) ~
File Name Format:		
File Name Format:		
File Name Format:		OK Abbrechen
File Name Format:		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.

Local Settings			
Application Parameters (Quick Dials / Call Transfer PRETALK Streaming	3 Settings Folder	
Main window size: Layout Columns: Hide units with Side-Bar size:	1920°1170 Custom 1 Rows: 2 out configured IP address 200	Custom window size: System Appearance Rows per System: Caller lines per row: Space between rows:	1400 * 1100 1 ~ min. 8 ~ 0
Show title bar (tem	porarily toggle with [Scroll-Lock] key) ixed position X: 0 Y: 0		
Page labels Page 1: Page 2: Page 3:	Headquaters Page 4: Subsidiary C Subsidiary A Page 5: Subsidiary D Subsidiary B Page 6:	Page 7: Page 8: Page 9:	Page 10:
Play Wave File			
On Remote Drop:	Ring.wav C:\Users\Public\AVT\MAGIC THipPro Intercom %d in the wave file name will be replaced by the I		Browse Browse ch was dropped.
Logfile Logfile Folder: File Name Format:	C:\Users\Public\AVT\MAGIC THipPro Intercom	∖Logfiles ✓	Browse Open
			OK Abbrechen
			0K Abbrechen

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.

Settings		
ation Parameters	Quick Dials / Call Transfer PRETALK Streaming Settings Folder	
ain Window Present	ation	
creen resolution:	1920*1170	
ain window size:	Custom v Custom window size: 1400 *	1100
Layout	System Appearance	
Columns:	1 V Rows: 2 V Rows per System: 1	\sim
Hide units with	out configured IP address Caller lines per row: min. 8	\sim
	Space between rows: 0	
de-Bar size:	200	
Show title bar (tem	porarily toggle with [Scroll-Lock] key)	
Show window on f		
ae labels		
Page 1:	Headquarters Page 4: Subsidiary C Page 7:	Page 10:
Page 2:	Subsidiary A Page 5: Subsidiary D Page 8:	-
Page 3:	Subsidiary B Page 6: Page 9:	
Fage 5.	Tage 5.	
ay Wave File		
] On Incoming Call:	Ring.wav B	rowse
] On Remote Drop:	C:\Users\Public\AVT\MAGIC THipPro Intercom\Warning %d.wav	rowse
] On Local Drop:	%d in the wave file name will be replaced by the line index (116) of the line which was dropped	d.
qfile		
ogfile Folder:	C:\Users\Public\AVT\MAGIC THipPro Intercom\Logfiles B	rowse Open
e Name Format:	<folder>\log-<day month="" of="">.txt (default)</day></folder>	
	OK A	Abbrechen
	OK 2	/pprechen

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 D	lials		🗹 Ena	ible Call Transfer —	
	Name	Number		Name	Number
1:	Headquarters	300	1:	Operator	502
2:	Subsidiary A	401	2:	CF 1	551
3:	Subsidiary B	402	3:	CF 2	552
4:			4:	CF EXT	09115271110
5:			5:		
6:			6:		
7:			7:		
8:			8:		
9:			9:		
10:			10:		
11:				lee "Blind Call Trans	fer" instead of "Attended Call Transfer"
12:					("Attended Call Transfer" is still used)
13:					
14:					
15:					
16:					
17:					
18:					
19:					
20:					
	L				
				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.

ication Parameters	Quick Dials / Call Transfer PRETALK Streaming Settings Folder
Enable PRETALK	Streaming
idio Input:	Primärer Soundaufnahmetreiber v 🗌 Stream test signal to MAGIC THipPro
Audio Output:	Primärer Soundtreiber V Play test signal on audio output
	OK Abbrechen
	OK Abbrechen
	OK Abbrechen
	OK YPprechen
	OK Abbrechen

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.

al Settings		
plication Parameters Quick Dials / Call Transfer PRETALK Streaming Setting	ps Folder	
Save settings*		
for current user (nonroaming)		
○ for all users		
O in this folder (from settings.ini) O in this folder (from command line argume	nt /ini_file)	
	Browse	
General*		
Save settings encrypted		
* Windows Administrator Rights may be needed		
* Windows Administrator Rights may be needed		
* Windows Administrator Rights may be needed		
* Windows Administrator Rights may be needed		
* Windows Administrator Rights may be needed		
" Windows Administrator Rights may be needed		
* Windows Administrator Rights may be needed		
* Windows Administrator Rights may be needed		
* Windows Administrator Rights may be needed		
" Windows Administrator Rights may be needed		
" Windows Administrator Rights may be needed		
* Windows Administrator Rights may be needed		
" Windows Administrator Rights may be needed		
" Windows Administrator Rights may be needed	0K Abbrechen	
* Windows Administrator Rights may be needed	OK Abbrechen	

Settings Folder



MAGIC THipPro Intercom

System Configuration



- The settings can be found in the CONFIGRATION 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 unter 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.





- 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.

tem01 - Configuration			
Global Settings	Login		
Clients / Security	USER		
General Client Workplace Assignment			
- Client Workplace Assignment Client Workplace Restrictions	Password:	•••••	
- Database			
Operation Settings	Confirm Password	•••••	
- Workplace Definition			
- Audio Line			
- Intro / HOLD Signal	ADMINISTRATOR		
- Signal Processing	Password:	•••••	
Line Labels	r deeword.		
Auto Answer	Confirm Password:	••••••	
i∎- GPIO			
System Settings			
General			
Line Interface			
- Caller Line Grouping			
- VoIP (LAN/SIP)			
Audio Interface			
PRETALK Streaming			
AES67 LAN Interface			
- DHD Audio Matrix			
- Ember+			
SNMP			
Login			
cogin			
nt ID: 1 Workplace: 3			OK Abbrechen Apply Now
it ID: 1 Workplace: 3			OK Abbrechen Apply Now

Login

MAGIC THipPro Intercom

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.

Global Settings	lients / Securi	ty			
Clients / Security General	- Computer Acc	ess List (access rights	1		
General Client Workplace Assignment	Intercom CI				
Client Workplace Assignment	macomen	GIRO			
Database	Client	Alias	Computer Name / IP Address	Add this PC to list	
- Operation Settings	1	OPERATOR	10.4.18.51		
- Workplace Definition	2	ICC-TRL-001	ICC-TRL-001		
- Audio Line	3	ICC-TRL-002	ICC-TRL-002		
- Intro / HOLD Signal	4	ICC-TRL-003	ICC-TRL-003		
- Signal Processing	5	100-1112-003	100-110-003		
- Line Labels					
- Auto Answer	6				
GPIO	7				
System Settings	8				
General	9				
Line Interface	10				
Caller Line Grouping	11				
VoIP (LAN/SIP)	12				
Audio Interface	13				
PRETALK Streaming	14				
AES67	15				
LAN Interface	16				
VLAN	17				
DHD Audio Matrix	18				
Ember+	19				
- SNMP					
Login	20				
		list means no access			
	An empty	list means no access	protection is active		
rkplace: 3				OK Abbred	chen Apply Now
rkplace: 3				OK Abbred	then Apply Now
diplace: 3					

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.

Global Settings General Clients / Security General Client Workplace Assignment Client Workplace Restrictions Database	General					
	Workplace Names and Colors		Button Labels	Button Labels		
	Name	Colour	Button	Label		
	Workplace 1	Text	Intro	0.0	•	
eration Settings	· · · · · · · · · · · · · · · · · · ·	Text	HOLD	u(×	-	
Workplace Definition Audio Line	Workplace 2		PRETALK	L.	•	
Intro / HOLD Signal	Workplace 3	Text	CONNECT	CONNECT	•	
ignal Processing ine Labels			CALL	CALL	•	
Auto Answer	Workplace 4	Text	CALLING	CALLING	•	
GPIO tem Settings	Workplace 5	Text	ACCEPT	ACCEPT	-	
General	Workplace 6	Text	DROP	DROP	-	
Line Interface Caller Line Grouping	Workplace 6	lext	Call Transfer	R	•	
VoIP (LAN/SIP)			Preallocate Caller	SELECT	-	
Audio Interface PRETALK Streaming AES67 LAN Interface VLAN DHD Audio Matrix	Preallocated Lines	numbers other than the preallocate				
Adio Interface Adio Interface LAN Interface VUAN DHD Audio Matrix Ember + SIMMP		numbers other than the preallocate				

General



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

Clients / Security General <mark>Client Workplace Assignment</mark>	Audio Assignment			
- Client Workplace Restrictions	Client	Workplace Access		
Database	1: OPERATOR	1,2,3,4	•	
 Operation Settings Workplace Definition 	2 : ICC-TBL-001	1	•	
Audio Line Intro / HOLD Signal	3: ICC-TBL-002	2.3	•	
Signal Processing				
Line Labels	4 : ICC-TRL-003	4	•	
Auto Answer GPIO				
System Settings				
General				
Line Interface Caller Line Grouping				
Caller Line Grouping VolP (LAN/SIP)				
- Audio Interface				
PRETALK Streaming				
AES67 LAN Interface				
- DHD Audio Matrix				
Ember+				
L. SNMP				
Login				
	Caution: Invalid settings are red	Settings for this	client have gray background colour.	
lient ID: 1 Workplace: 3				OK Abbrechen Apply Now
				DK Abbrechen Apply Now
lient ID: 1 Workplace: 3				OV Abbrehen Andrikten

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.

obal Settings	Client Workplace Rest	rictions									
Clients / Security General											
- Client Workplace Assignment Client Workplace Restrictions	Client	Long Click Dura	ion	Drop All		Call All		Preallocate Calle	Ħ	Preallocation Loa	±7
Database	1: OPERATOR	1000 msec	•	allowed	-	allowed		allowed	•	allowed	•
peration Settings Workplace Definition	2 : ICC-TRL-001	1000 msec	•	allowed	•	allowed	•	allowed		allowed	•
- Audio Line	3: ICC-TRL-002	1000 msec		allowed	-	allowed	•	allowed		allowed	-
Intro / HOLD Signal Signal Processing	4 : ICC-TRL-003	1000 msec		allowed	-	allowed	-	allowed	-	allowed	
Signal Processing Line Labels	4.100 1112 000			aloved	•	aloneu	-	diowed	•	dioweu	
Auto Answer											
- GPIO /stem Settings											
- General											
Line Interface											
Caller Line Grouping											
VoIP (LAN/SIP)											
Audio Interface											
PRETALK Streaming AES67											
- LAN Interface											
- VLAN											
DHD Audio Matrix											
- Ember+		<									>
SNMP ogin											
		S	ettings for th	is client have g	ay backgr	ound colour.				Default Settings	
): 1 Workplace: 3								0	К	Abbrechen	Apply N
1 Workplace: 3									K	Abbrechen	Apply N

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.

Global Settings Clients / Security	Client Workplace Rest	rictions									
General Client Workplace Assignment <mark>Client Workplace Restrictions</mark>	Client	Long Click Duration		Drop All		Call All		Preallocate Calle	1	Preallocation Loa	dZ
Database	1: OPERATOR	1000 msec	•	allowed		allowed	•	allowed	-	allowed	•
Operation Settings	2 : ICC-TRL-001	1000 msec	•	alowed	•	alowed	-	alowed	F	allowed	•
- Audio Line	3 : ICC-TRL-002	1000 msec	•	allowed	•	allowed	•	allowed	-	allowed	•
Intro / HOLD Signal Signal Processing	4 : ICC-TRL-003	1000 msec	-	allowed	•	allowed	•	allowed	-	allowed	•
Line Labels Acto Answer General Line Interface Coller Line Grouping VolP (LAN/SP) Addio Interface PRETALK Streaming Actor Actor Actor VAN DHD Audio Matrix Ember + SMMP		¢						-			>
in		Settin	gs for this	client have gr	ay backgi	ound colour.				Default Setting:	:
nt ID: 1 Workplace: 3								01	ĸ	Abbrechen	Apply
ID: 1 Workplace: 3								0)	K	Abbrechen	Apply
(D. 1)/(-f-f3											

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.

Global Settings	Database							
Clients / Security General	1: 10.4.18.5\SQLEXPRES	S AVTIntercom						
Client Workplace Assignment								
- Client Workplace Restrictions	Database							
Database	SQL Server:	10.4.18.5\SQLEXPRES	s				Test	Open Connection
Operation Settings	Database:	AVTIntercom						
Workplace Definition Audio Line								
Intro / HOLD Signal	User:	IntercomUser						
- Signal Processing	Password	•••••						
- Line Labels	Network Library:	Default					~	
Auto Answer	Network Library.	Derault					~	
. GPIO	Telephone Book							
System Settings General	Phone Number Types	Connection Types	Mode	Resolution	Sampling Rate	Data Rate	Secure Stream	Remot
- Line Interface	Other	Telephone	•		0.000	0.01010		
Caller Line Grouping	Mobile	Telephone	•					
- VoIP (LAN/SIP)	Office	Telephone						
- Audio Interface	Private	Telephone	•					
PRETALK Streaming	Intercom	Telephone	•					
AES67		rooprono						
LAN Interface								
VLAN DHD Audio Matrix	Add Phone Nun	nberType Sa	ave					
- Ember+	Location Details							
SNMP	Country:	Deutschland (+49)	~	Country C	ode: +49			
- Login	City:	Nümberg	~	Area Cod		_		
	City:			1				
	Label for in-house calls:	<intern></intern>		Disabl	e Number Formatti	ng		
						OK.	Abbrechen	Apply Now
nt ID: 1 Workplace: 3								
at ID: 1 Workblace: 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.

Client Workplace Assignment	1: 10.4.18.5\SQLEXPRES	S AVTIntercom						
Client Workplace Restrictions	☑ Database							
Database	SQL Server:	10.4.18.5\SQLEXPRESS					Tes	/Open Connection
peration Settings Workplace Definition	Database:	AVTIntercom						
Workplace Definition Audio Line								
Intro / HOLD Signal	User:	IntercomUser						
Signal Processing	Password:	•••••						
Line Labels Auto Answer	Network Library:	Default					\sim	
Auto Answer GPIO								
/stem Settings	Telephone Book							
General	Phone Number Types	Connection Types	Mode	Resolution	Sampling Rate	Data Rate	Secure Stream	n Remot
- Line Interface	Other		•					
Caller Line Grouping VoIP (LAN/SIP)	Mobile		•					
- Audio Interface	Office		•					
- PRETALK Streaming	Private		•					
AES67	Intercom	Telephone	*					
LAN Interface								
VLAN	Add Phone Num	ber Type Sav	e .					
DHD Audio Matrix			~					
Ember+	Location Details							
- SNMP	Country:	Deutschland (+49)	~	Country C	ode: +49			
gin	City:	Nümberg	~	Area Code	s 911			
	Label for in-house calls:	<intern></intern>		Dieable	e Number Formatti	20		
	Laber for in-house calls.	Cirkent/			e recimber r cimaca	ig		
):1 Workplace: 3								
. 1 Wolkplace. 3						OK	Abbrechen	Apply Now
: 1 Workplace: 3						OK	Abbrechen	Apply Now
1.1 Mediation 2								

Database (2)



MAGIC THipPro Intercom

System Settings

Settings that <u>cannot be</u> 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.

System01 - Configuration		
Global Settings - Clenth / Security - Clenth / Security - Clenth / Security - Clenth / Workplace Assignment - Clenth / Workplace Retrictions Database Operation Settings - More / HOLD Signal - Signal Processing - Into / HOLD Signal - Signal Processing - Unit Leads - Audio Line - Staff - Retro - Staff - Clenth / Security - Security	General Display Larguage English Front Kaypad Kay Tone Display Backgit: On Contrast Backgit:	
lient ID: 1 Workplace: 3		OK Abbrechen Apply Now
Sent ID: 1 Workplace: 3		OK Abbrechen Apply Now

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.

Global Settings	Line Interface	
Clients / Security General Client Workplace Assignment Client Workplace Restrictions Database	General Line Mode: ValP (LAN/SIP) Disp not answered incoming/outgoing calls after 90 seconds	
Operation Settings Workplace Definition Audio Line Intro / HOLD Signal	Channels 1 2 3 4 5 6 7 8 9 10 11 12 13 14 115 16	
Signal Processing Line Labels Auto Answer B- GPIO	Inchase Lines Image: Control of the contr	
 System Settings General General Caller Line Grouping Vol9 (LAN/SP) Audio Interface PRETALK Streaming ALSA LAN Interface DHD Audio Matrix DHDR++ SSNAP Login 		
nt ID: 1 Workplace: 3	OK Abbrechen Apply N	WW

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																	
Clients / Security General Client Workplace Assignment	Name	Colour	Channe]	
- Client Workplace Restrictions			1	2 3	4	5 6	7 8	9	10	11 1	2	13	14	15	16			
Operation Settings	Line Group 1	Text			v								- I					
Workplace Definition Audio Line	Line Group 2	Text				~ ~	~	1				F I	- I					
Intro / HOLD Signal Signal Processing	Line Group 3	Text								~	•	L L	- r					
- Line Labels - Auto Answer	Line Group 4	Text										L L	- r					
GPIO System Settings	Line Group 5	Text											- 1					
General Line Interface	Line Group 6	Text											- 1					
- Caller Line Grouping - VolP (LAN/SIP)	Line Group 7	Text											- r		—		1	
Audio Interface	Line Group 8	Text											- 1					
PRETALK Streaming AES67 LAN Interface VLAN	Line Group 9	Text											- 1					
	Line Group 10	Text											- r					
DHD Audio Matrix Ember+	Unassigned											•	~	~	~			
Login																		
	CF: Call Forwarding Lir	ne																
t ID: 1 Workplace: 3												01	(1	Abbrechen	A	pply No	v
k ID: 1 Workplace: 3											E	Ok	(1	Abbrechen	V	pply No	
		_																
	Backgrou	ind colour:																
		T	ext															
	Luminanc	e differenc	e: 15	i														
			_		-													
	Text colo		White															
	O BI	аск 🔍	white															

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.

Clients / Security General	Line	LA		SIP Server	ν		Backup Server	TCP	STUN	Liser Name		Password	Audio Port (Displaye
Client Workplace Assignment	Line 1		_				10.2.141.8		STUN	601	User Authenti	Password	Audio Port (Displaye
- Client Workplace Restrictions		1	-		2		1		_			2001		
- Database Operation Settings	Line 2	1		10.2.140.8	2	_	10.2.141.8			602			5005	
- Workplace Definition	Line 3	1		10.2.140.8	2		1			603		3008	5006	
Audio Line	Line 4	1			2		1			604		2003	5007	
Intro / HOLD Signal	Line 5	1	-	10.2.140.8	2	-	10.2.141.8			605		8008	5008	
Signal Processing Line Labels	Line 6	1	•	10.2.140.8	2	-	10.2.141.8			606		NOR	5009	
- Auto Answer	Line 7	1	•	10.2.140.8	2	•	10.2.141.8			607		1008	5010	
🛓- GPIO	Line 8	1	•	10.2.140.8	2	-	10.2.141.8			608		2005	5011	
System Settings General	Line 9	1	-	10.2.140.8	2	-	10.2.141.8			609		2004	5012	
- Line Interface	Line 10	1	-	10.2.140.8	2	-	10.2.141.8			610		1001	5013	
Caller Line Grouping <mark>VoIP (LAN/SIP)</mark> Audio Interface PRETALK Streaming	Line 11	1	-	10.2.140.8	2		10.2.141.8		í.	611		1008	5014	
	Line 12	1	-	10.2.140.8	2		10.2.141.8		Ē	612		1008	5015	
	Line 13	1		10.2.140.8	2		10.2.141.8	Ē	<u> </u>	613		2005	5016	
AES67	Line 14	1		10.2.140.8	2		1			614		1001	5017	
LAN Interface VLAN	Line 15	1	_	10.2.140.8	2		1	-	-	615		1001	5018	
- DHD Audio Matrix	Line 16	1		10.2.140.8	2		1		-	616		3008	5019	
Ember+	Line to		Ľ	10.2.140.0	2		10.2.141.0		J	010				
Login	- VolP Par	ramete	a -							Registration			Set Defau	It Audio Por
	Paylo	oad Ti	me:		- b -			10 ms	ec	Delay betwee	en SIP lines: 0	msec (04	000)	
		Law/	μ·La	w Signalling on in	ncoming	G.72	2 calls			Timeout	60	sec (60	500)	
	U	se firs	t coc	lec of SDP audio	codec	list a:	: default							

VoIP (LAN/SIP) (1)



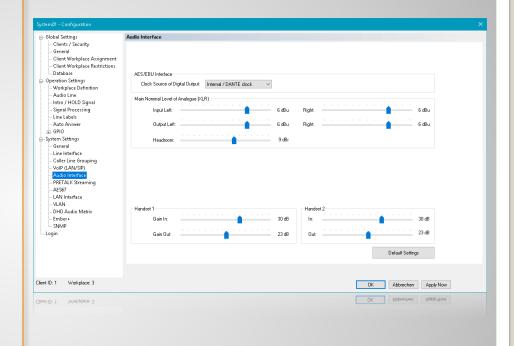
- A low PAYLOAD TIME reduces the audio delay, but increases the gross data rate a little.
- A-LAW/U-LAW SIGNALLING ON INCOMING G.722 CALLS / USE FIRST CODEC OF SDP AUDIO CODEC LIST AS DEFAULT: Functions to avoid errors in the implementations of some SIP servers. Normally disabled.
- Under REGISTRATION 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.

ne ine 1 ine 2	LAN 1		SIP Server	LA	N	Backup Server	VoiP (LAN/SIP)													
ne 2	1	-				Backup Server	TCP	STUN	User Name	User Authenti	Password	Audio Port (Displaye							
		101	10.2.140.8	2	•	10.2.141.8			601		ROOM	5004								
	1	•	10.2.140.8	2	•	10.2.141.8			602		8008	5005								
ne 3	1	-	10.2.140.8	2	•	10.2.141.8			603		1001	5006								
ne 4	1	•	10.2.140.8	2		10.2.141.8		Ē	604		8008	5007								
ne 5	1	-	10.2.140.8	2		10.2.141.8		Ē	605		1001	5008								
ine 6	1	-	10.2.140.8	2		10.2.141.8	_	, 	606		1000	5009								
ne 7	1	_		2	•	10.2.141.8	_	, 	607		1001	5010								
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ne 13 1 v 10.2140.8 2 v ne 14 1 v 10.2140.8 2 v ne 15 1 v 10.2140.8 2 v ne 15 1 v 10.2140.8 2 v afP Parametr Paylood Time v v v v afLawly-Lew Signaling on incoming 6 72 v v v	ne 6 1 v 10.2140.8 2 v 10.2141.8 ne 7 1 v 10.2140.8 2 v 10.2141.8 ne 7 1 v 10.2140.8 2 v 10.2141.8 ne 9 1 v 10.2140.8 2 v 10.2141.8 ne 9 1 v 10.2140.8 2 v 10.2141.8 ne 10 1 v 10.2140.8 2 v 10.2141.8 ne 11 1 v 10.2140.8 2 v 10.2141.8 ne 12 1 v 10.2140.8 2 v 10.2141.8 ne 13 1 v 10.2140.8 2 v 10.2141.8 ne 14 1 v 10.2140.8 2 v 10.2141.8 ne 15 1 v 10.2140.8 2 v 10.2141.8 ne 15 1 v 10.2140.8 2 v 10.2141.8 ne 15 1 v 10.2140.8 2 v 10.2141.8	ne 6 1 1 10.2140.8 2 10.2141.8 ne 7 1 10.2140.8 2 10.2141.8 ne 8 1 10.2140.8 2 10.2141.8 ne 9 1 10.2140.8 2 10.2141.8 ne 10 1 10.2140.8 2 10.2141.8 ne 11 1 10.2140.8 2 10.2141.8 ne 12 1 10.2140.8 2 10.2141.8 ne 14 1 10.2140.8 2 10.2141.8 ne 15 1 10.2140.8 2 10.2141.8 ne 14 1 10.2140.8 2 10.2141.8 ne 15 1 10.2140.8 2 10.2141.8 ne 16 1 10.2140.8 2 10.2141.8 ne 16 1 10.2140.8 2 10.2141.8 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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.

Global Settings	PRETALK Stree	ning	
 Client / Security General Client Workplace Assignment Client Workplace Retrictions Database Operation Settings Workplace Definition Audo Inter Signal Processing Linte J MOLD Signal Signal Processing Operation Settings Operation Settings	P Steam 1	UDP Port 5200	Set Default Audo Ports
t ID: 1 Workplace: 3			OK. Abbrechen Apply Now
			OK. Abbrechen Apply Now
int ID: 1 Workplace: 3			OK Abbrechen Annie Now

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 ≈ 1.4 Mbit/s x Channels (at L24 and 48 kHz)
- LAN INTERFACE: Network interface used for AES67.
- CHANNELS: Number of channels needed.

ystem01 - Configuration	1007			>
Global Settings	AES67			
- General	Activate AES67 streaming			
Client Workplace Assignment Client Workplace Restrictions	LAN Interface:	LAN 2 : 10.2.142.39 V		
Database	Channels:	8 ~		
Operation Settings Workplace Definition	Transmission:			
- Audio Line	SAP Stream Name:	System01		
Intro / HOLD Signal Signal Processing	RTP UDP Port::	5300		
- Line Labels	Audio Mode:	L16 ~		
Auto Answer GPIO	Sampling Rate:	48 kHz ~		
System Settings			239 0 142 39	
General	Address Mode:	Auto V IP Address:	239 0 142 39	
- Line Interface	Reception:			
Caller Line Grouping VoIP (LAN/SIP)	Stream 1:	AVIOUSB-AVT1: 8 channels	~	Update Bx Streams
- Audio Interface	Stream 2:			
PRETALK Streaming	Stream 2.	AVIOUSB-AVT2; 8 channels	~	
AES67				
LAN Interface VLAN				
- DHD Audio Matrix	Quality of Service (DiffServ):			
Ember+				
- SNMP	PTP:	239 (0255) EF DSCP: 59dec	Set Default OoS values	
Login	BTP:	175 (0255) EF DSCP: 43dec	Set Default GoS Values	
ient ID: 1 Workplace: 3			OK	Abbrechen Apply Now
ient ID: 1 Workplace: 3			OK	Abbrechen Apply Now

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

General Cience Workplace Assignment Cience Workplace Restrictions Database Operation Settings 	Clent Workplace Assignment	Global Settings	AES67			
Client Workplace Assignment Client Workplace Assignment Client Workplace Assignment Client Workplace Restrictions Channels: 0 Database Operation Sattings Control Line Lable Sap Processing Line Lable Sap Processing Control System Statings Society Control Cont	Client Workplace Assignment Client Workplace Retrictions Database Pertent Workplace Retrictions Charnels: Database Pertent Workplace Retrictions Charnels: Dar	Clients / Security	- Activate AES67 streaming			
Client Workplace Retrictions Unamelia: Unamelia: Unamelia: Operations Strings Operations SAP Stream Name: Suptem01 Intro / HOLD Signal SAP Stream Name: Suptem01 Intro / HOLD Signal Signal Processing Audo Mode: L16 Intro / HOLD Signal Signal Processing Audo Mode: L16 Intro / HOLD Signal Signal Processing Audo Mode: L16 Intro / HOLD Signal Signal Processing Audo Mode: L16 Intro / HOLD Signal Signal Processing Audo Mode: L16 Intro / HOLD Signal Collent Line Intrafice Audo Mode: L16 Intro / HOLD Signal Contentine Stream T: Auto Device IP Address: 239 142 39 Audo Interface Receptor: Update Rx Streamelia: Update Rx Streamelia: Update Rx Streamelia: Intro / HOLD Signal Losi Interface Voip (Ax/S1) Stream T: Av10USB AVT12 & channels: Update Rx Streamelia: Interface ULINI Interface Usage of Service (DWServ) Interface Interface Interface Interface Interface <	Image: Section Workplace Restrictions Image: Section Settings Optimizer Sampling Rate: B Owners::::::::::::::::::::::::::::::::::::					
Database Operation Stim Operation Stim Audio Line Audio Line Sap Steam Name Sap Steam Name Steam Name	Outsabase Channels: 0 Operation Strings Transmission: Audio Line SAP Shean Name: System Name: Audio Line SAP Shean Name: System Name: Signal Processing Audio Node: LIE Auto Answer Audio Node: LIE Strings Addees: Mode: Audio Of Caller Inter Stronging Addees: Mode: Audio Vold (ANVSP) Stream I: AMDUSP AVT1: Stream Strings Addees: Mode: Audio Vold (ANVSP) Stream I: AMDUSP AVT2: Stream Strings Addees: Mode: Vold (AnvSP) Vold (ANVSP) Stream I: AMDUSP AVT2: Stream I: AMDUSP AVT2: channels Vold (AnvSP) Vold (AnVSP) Stream I: AMUOUSP AVT2: channels Vold (AnvSP) Stream I: Outedy of Service (Diffserv) Interface Vold (AnvSP) Set Default (Oos voldee: Stream I: T75 (0.255) EF DSCP. 53dec Set Default (Oos voldee: Cogin RTP: T75 (0.255) EF DSCP. 43dec </td <td></td> <td>LAN Interface:</td> <td></td> <td></td> <td></td>		LAN Interface:			
Workplace Definition SAP Steam Name: System 01 - Audio Line Sagnal Processing Bit P UDP Prot: S300 Line Isbels Audo Mode: L16 Sampling Rate: 48 HHz GR00 Sampling Rate: 48 HHz PAddees: 238 142 33 General General Adde Stress Mode: Audo P Addees: 238 142 33 - Coller Line Grouping Steam 1: AVIOUSB AVT1: B charmels Update Rx Steams Update Rx Steams - Audio Interface - Natio Interface Steam 2: AVIOUSB AVT2: Scharmels Steam - DHD Audio Matrix - Dually of Service [DHServ] Steam 2: (0.255) EF DSCP. 59dec - Sthort - - TT75 (0.255) EF DSCP. 59dec Set Default Ood values - Up Audio Matrix - Dually of Service [DHServ] - TT75 (0.255) EF DSCP. 59dec - Sthort - - TT75 (0.255) EF DSCP. 59dec - Set Default Ood values	- Workplace Definition Hardmetator - Audio Into / HOLD Signal SAP Sheam Name: Syntem01 - Signal Processing Att PUOP Port: - General Adda Mode: - General Adda Mode: - Off Addess: 233 0 - Off Addess: 233 0 - General Adda Mode: - Off Addess: 233 0 - Off Addess: Addess Mode: - Off Addess: Addess Mode: - Off Addess: Stream 1: - Off Addess: Stream 2: - Off Addess: Stream 2: - Off Addess Stream 2: - Off Addess Guadey of Service (DiffServ) - Inster+ - T5 - Songher - T5 - Ogin RTP: 175 - Modesse: Set Default God values - Off Addio Matrix Off Addesse - Ogin RTP: 175 <td< td=""><td> Database</td><td>Channels:</td><td>8 ~</td><td></td><td></td></td<>	Database	Channels:	8 ~		
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Intro / HOLD Signal ATP UDP Port: 3300 Signal Processing Audo Mode: L16 Line Labels Audo Mode: L16 GRU Sameping Rate: 48 HHz Collect Line Grouping Addeess Mode: Audo Voir (ANXSP) Addeess Mode: Audo - Audio Intrafrace Stream 1: AVIOUSB AVT1 8 channels Update Rx Streams - Audio Intrafrace Stream 2: AVIOUSB AVT2 8 channels Update Rx Streams - Madio Intrafrace Stream 1: AVIOUSB AVT2 8 channels Update Rx Streams - Madio Intrafrace Stream 1: AVIOUSB AVT2 8 channels Update Rx Streams - Madio Intrafrace Stream 1: AVIOUSB AVT2 8 channels Update Rx Streams - Madio Intrafrace Stream 1: AVIOUSB AVT2 8 channels Update Rx Streams - Unit Notrifice Update Rx Streams RTP: 175 (0.255) EF DSCP: 53doc - Stream 1: T75 (0.255) EF DSCP: 53doc Stream 1: Apply Now	- Index / HOLD Signal RTP UDP Port: 5300 - Signal Porcessing					
Signal Processing BTP UDP Port: 5300 Line Interface Audo Mode: L15 Option Sampleng Pade: 491Hz Caller Line Interface Audo Mode: L16 Caller Line Interface Reception: Update Rx Streams Caller Line Interface Stream 1: AVIOUSB AVIT 1: 8 channels Update Rx Streams PMET ALK Streaming Stream 2: AVIOUSB AVIT 2: 8 channels Update Rx Streams PMET ALK Streaming Stream 2: AVIOUSB AVIT 2: 8 channels Update Rx Streams UANI Interface PIF: 175 (0.255) EF DSDP: 59dec Stopp: BTP: 175 (0.255) EF DSDP: 59dec	- Signal Processing RTP UOP Pore: 5300 - Mark Advances Audo Mode: L16 - Unic Labels Audo Mode: L16 - General Somping Rate: 484 Hz - General Audo Mode: L16 - General Receptor: Update Rx Stream: - Vall (LAN/SDP) Stream 1: AVIDUSB AVT1; 8 channels Update Rx Stream: - MAI Interface Stream 2: AVIDUSB AVT2; 8 channels V - MAI Interface Stream 2: AVIDUSB AVT2; 8 channels V - MAI Interface Stream 2: (0.255) EF DSCP-53dec Set Default OoS values - Login RTP: 175 (0.255) EF DSCP-43dec Mode Acept Nove		SAP Stream Name:	SystemUT		
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Add Answer GPIC GPIC System Statings General Line Interface PGITALK Streaming PGITALK Streaming PGIT	Adda Answer Sampling Rate: 49 kHz GF00 Sampling Rate: 49 kHz Strings Addees: Mode: Auto General In: Interface Under Rick Reception: Caller Line Grouping Stream 1: Addies: Mode: AU0USB:AVT1: 8 charmels Under Rick Stream 2: Addies: Mode: AU0USB:AVT2: 8 charmels Undate Rick Stream 2: Addies: Mode: Class (Difference) UNIN Guadity of Service (Difference) OHD Audio Matrix Guadity of Service (Difference) Singer PTP: 238 Ogin RTP: 175 (0.255) EF DSCP: 53dec ent ID: 1 Workplace: 3 Workplace: 3 Motephase: 3		Audio Moder	116		
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PRETAIL Streaming Interface LAN Interface Quality of Service (DI/Serv): Strike PTP: Strike PTP: Login RTP: NMP PTP: Login RTP: No Abbrechen Apply Now	PERTAK Streaming Interface - WAN Out Dudio Matrix - Orb Dudio Matrix Quality of Service (DifServ) - SNMP PTP. - Login RTP. 175 (0.255) EF DSCP: 59dec Set Default DoS values		Stream 2	MROUGE MC2-9 abarrate		
Login RTP: 175 (0.255) EF DSCP: 33dcc Set Default 0.05 values	- Login RTP: 175 (0.255) EF DSCP: 53dec - Login RTP: 175 (0.255) EF DSCP: 43dec - Login RTP: 175 (0.255) EF DSCP: 43dec		ordaniz.	AVIOUSDIAV12, 6 channels	~	
UAN Dub Audio Matrix Guality of Service (DHServ): Embert- ShMrp PTP. 233 (0.255) Login RTP: 175 (0.255) RTP: 175 (0.255) EF DSCP. 43dec	Login Cushy of Service (DiffServ) Ember+ SNAP PTP: 233 (0.255) EF DSCP: 53dec RTP: 175 (0.255) EF DSCP: 43dec ent ID: 1 Workplace: 3 OK Abbrechen Apply Now					
LD-DH Audio Matrix Ember* Login RTP: 123 (0.255) EF DSCP: 53dec set Default DoS values AlD: 1 Workplace: 3 OK Abbrechen Apply Now	Chi Dudio Matrix Cuality of Service [DHServ] SNMP PTP: 238 (0.255) EF DSCP: 59dec Set Default DoS values RTP: 175 (0.255) EF DSCP: 43dec					
Login PTP: 23 (0.255) EF DSCP: 53doc RTP: 175 (0.255) EF DSCP: 43doc MD:1 Workplace: 3 OK Abbrochen Apply Now	Login RTP: 175 (0.255) EF DSCP: 53dec Login RTP: 175 (0.255) EF DSCP: 43dec					
Login RTP: 175 (0.255) EFDSCP: 43dec Set Default OoS values xt(D:1 Woltplace: 3 DK Abbrechen Apply Nov	Login RTP: 175 (0.255) EF DSCP 43dec Set Default DoS values		Quality of Service (DiffServ)			
RTP: 175 (0.255) EF DSCP: 43dec	ent ID: 1 Workplace: 3 OK Abbrechen Apply Nov	SNMP	PTP:	239 (0.255) EF DSCP: 59dec		
nt ID: 1 Workplace: 3 DK Abbrechen Apply Now	ert ID: 1 Workplace: 3 DK Abbrechen Apply Now	Login	DTD.	175 (0.2EE) EE DOOD 434	Set Default QoS values	
				(0.233) EI D3CL 40080		
		ant ID: 1 Workplace: 2				
OK WODIECUEU WITH MODE	ert ID: 1 Wolkplace 3 DK Abbrechen Apply Now	encip. 1 workpiece: 3			OK	Abbrechen Apply Now
VID-1 Workplace 3 Apple Low		ent ID: 1 Workplace: 3			OK	Abbrechen Apply Now
# ID: 1 Workplace: 3 Approximate Arealy Mount						

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.

be twork.	
ATERX - Client / Scurity - General - Client Workplace Arsignment - Client Workplace Restrictions - Database Image: Client Workplace Arsignment - Client Workplace Restrictions - Database Operation Settings - Operation Settings - Audio Inter - Audio Inter - Audio Inter - Audio Inter - Signal Processing - Signal Processing - Signal Processing - Audio Mode: Image: Client Workplace Arsignment - Signal Processing - Audio Inter - Signal Processing - Audio Mode: Image: Client Workplace - Audio Mode: Image: Client Wor	
ATE RX - General - General - General - General - General - General - General - General - Workplace Assignment - General - Workplace Definition - General - MAIDUSB AVT1: 8 charmets - General - MAIDUSB AVT1: 8 charmets - Workplace - WAIN Horefrace - WAIN - WAIN Merfarce - SNAPP - WOR PTP: 229 [0.259] EF DSCP: 58dec Set Default Go states	
OPE Operation Stiffings Charmeli: 8 Operation Stiffings SAP Steam Name: System01 Operation Stiffings Audo Mode: 115 Operation Stiffings Audo Mode: 116 Operation Stiffings Audo Mode: 118 Operation Stiffings Steam 1: AutoUDSRAVT18 channels Update Rx Stee Operation Stiffings Operation Stiffings Steam 2: AutoUDSRAVT2.8 channels V Operation Stiffings Operation Stiffings 0.250) EF DSCP: 59dec Sto Default GoS valage: </td <td></td>	
OP Outrative Refrictions Charmeli: B Operation Stiftings Transmission: - Audio Line SAP Steam Name: System01 - Audio Line SAP Steam Name: System01 - Audio Line SAP Steam Name: System01 - Audio Line Audio Mode: L15 - Audio Line foregoing Audio Mode: L16 - Audio Answer Sampling Rate: 481Hz - Other Mittings Audio Mode: L16 - Other Mittings Audio Mode: L16 - Other Mittings Audio Mode: L18 - Other Mittings Steem 1: AVIOUSB AVT12 8 channels Update Rx Stee - PRETALK Streaming Steem 2: AVIOUSB AVT2 8 channels V - Otho Audio Matrix Quality of Service (Diffserv) - - Finite** - Stode PTP: 233 (0.255) EF DSCP: 594ec	
OP Workplace Definition SAP Steen Name: System Name	
Audio Line SAP Steam Name System01 Support Signal Processing Support Signal Processing Support Suppor	
OP - Signal Processing RTP UDP Port: 5800 - Julia Jabetis - Auto Mode: LTB - Auto Annwer - Samping Rate: 48 kHz - System Settingsis - Addes: Mode: Auto - General - General - General - Unite Interface - Receptor: - Oller Line Stronging Stream 1: - Audio Interface - Streaming - Audio Interface - Streaming - Waldio Interface - WithOUSB AVTI2: 8 channels - Waldio Interface - WithOUSB AVTI2: 8 channels	
OP - Signal Processing - Audo Mode: 116 - Auto Answer - Samping Rate: 48 kHz - Orion Samping Rate: 48 kHz - Orion - Orion - Orion	
Addo Ansver CPIO Sanching Reic Addess Mode Addess Addess Mode Addess Addess Mode Addess Addess Mode Addess Addess	
OP System Settings Address Mode: Auto IP Address: 239 0 142 39 - General	
Openeral - General - Unit Interface Reception: - Caller Line Brouping Stream 1: - Audio Interface Stream 2: - Audio Interface Stream 2: - Audio Interface Stream 2: - Unit Interface Stream 2: - Unit Interface	
OC -Caller Line Grouping Stream 1: AVIOUSB-AVT1.8 channels Update Rx Stream -Audio Instrance -Audio Instrance AVIOUSB-AVT1.8 channels V -Audio Instrance -Audio Instrance AVIOUSB-AVT2.8 channels V -Det Audio Instrance -UAN Instrance V -UAN Instrance -UAN -UAN -UAN Instrance -UAN -UAN -UAN -UAN -UAN <td></td>	
OC -Audio Interface - PRETAL K Stearning - LAN Interface - UAN Interface - UAN Interface - UAN - DHD Audio Matrix - Ember + - SIMAP AVIOUSB AVT2:8 channels VOCK - UAN - UAN - DHD Audio Matrix - Ember + - SIMAP Quality of Service [DHServ) - SIMAP VOCK - SIMAP PTP. 233 [0.255] EF DSCP: 53dec	
PETALK Streaming AviDUSB AV12;8 channels AviDUSB AviDUS Avi	Rx Streams
De LAN Interface -VLAN -DFD Audio Matrix -Ember+ -SNMP PTP: 239 (0.255) EF DSCP: 584cc Set Default 005 values	
OC -VLAN -DHD Audio Matrix Quality of Service (D/I/Serv) -Ember+ SIMMP -SiMMP PTP: 239 (0.255) EF DSDP: 5940c Set Default GOS values	
Ember+ Guard of a service (unitserv) Standard a service (unitservice	
NOR PPP PTP. 233 (0.255) EF DSCP: 59dec	
VOĽK. Set Default QoS values	
Clert ID: 1 Workplace: 3 OK Abbrechen	schen Apply Now
Dent ID.1 Wokplace 3 DK Abbechen	chen Apply Now
Canado 1 Mondayan 2	



- 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	
 Global Settings Clients / Security General 	LAN I terface
- Usererai - Client Workplace Assignment - Client Workplace Restrictions - Database Operation Settings - Workplace Definition - Audio Line - Intro / HOLD Signal - Signal Processing	Dark 2 Second IP Address Trid IP Address IP Address 10.4.18.211
Line tabels - Auto Answer G (6P) - System Settings - General - Line Interface - Caller Line Grouping - VolP (LAN/SIP) - Audio Interface - PRETAL Streaming	STUN Server: STUN Server: NAT Keep Afrive Interval: 20 sec (5. 60) Quality of Service (DHServ) Voice: 184 (0. 255) EF DSCP. 46dec SIP: 104 (0. 255) EF DSCP. 26dec Default Settings
AES67 <mark>LAN Interface</mark> VLAN	Link Type: Auto Ulable Insufficient LAN Alam
- DHD Audio Matrix - Ember+ - SNMP - Login	Control/PRETALK Streaming UDP Control Port: 10000 Set Default Port Accessible from: LAN 1:10.418.211
Client ID: 1 Workplace: 3	OK Abbrechen Apply Now
Client ID: 1 Workplace: 3	OK Abbrichen Apply New

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, ...).

	VLAN				
Clients / Security General	VLAN				
	Service PC Control & PRETALK Streaming	TPID none	Priority	VID (12-Bit)	
Operation Settings	VolP			30	
Workplace Definition Audio Line	SNMP	802.1QTag •		31	
Intro / HOLD Signal	DHD	none -			
Signal Processing Line Labels	Ember+	none -			
Auto Answer	Remote Light Protocol	none -			
GPIO	AES67	none -			
System Settings General					
Line Interface					
Caller Line Grouping VoIP (LAN/SIP)			1		
- Login	Modification of the VLAN parameters may in	nterrupt the connection to th	e PCI		
nt ID: 1 Workplace: 3				OK Abbrec	hen Apply Now
nt ID: 1 Workblace: 3				OK Abbrec	





- 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.

System01 - Configuration	×
Global Settings ⊂ Global Settings – Glenet / Security – General – Clent Workplace Assignment – Clent Workplace Assignment – Clent Workplace Definition – Audio Line – Inter John Settings – Signal Processing – Line Labels – Auto Answer – Gignet Workplace – General – Line Interface – Caller Line Grouping – Vall (AN/SP) – Audio Interface – Caller Line Grouping – Vall (AN/SP) – Audio Interface – Caller Line Korsening – AESS7 – LAN Interface – Simp – Line Interface – Caller Line Grouping – Vall (AN/SP) – Audio Interface – Caller Line Grouping – ASS7 – LAN Interface – Line Interface – Caller Line Grouping – ASS7 – LAN Interface – Simp – Login	DHD Audio Marix Activate DHD Audio Marix Control Audo Marix Control Audo Marix Control LAN Indrades: AUAII 10418211 V TCP:AP Addres: 104.16.24 Pot 2000 TCP:AP Addres: 104.16.24 Pot 2000
Client ID: 1 Workplace: 3	OK Abbrechen Apply Now
Client ID: 1 Workplace: 3	OK. Aktendem Apply Now

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						×
Clients / Security General Client Workplace Assignment Client Workplace Excititions Database Operation Settings Workplace Definition Audio Line Intro / HOLD Signal Signal Processing	Ember+ ✓ Activate Ember+ Provid Ember+ Connection Param LAN Interface: Port 1 (Consumer 1): Port 2 (Consumer 2): Port 3 (Consumer 3):		Port 4 (Consumer 4) Port 5 (Consumer 5) Port 6 (Consumer 6)	0 0 0	Port 7 (Consumer 7): 0 Port 8 (Consumer 8): 0	
- Line Labels - Auto Answer - GPIO - TTL / Relay - Ember+ - Output - Consumer Functions - Sprem Settings - General	Activate Ember+ Consu Connection Parameters: LAN Interface: Provider 1: Provider 2:	ner LAN 1 : 10.4.18.211 TCP/IP Address: 10.4.18.212	~	Port 9010		
Line Interface Caller Line Grouping VolP (LAN/SP) Audio Interface PRETALK Streaming ASSF ALAN Interface VLAN DED Audio Mattice Ember Softmark Login						
Client ID: 1 Workplace: 3					OK Abbrechen Apply	Now
Client ID: 1 Workplace: 3					OK Abbrechen Apply	Now

Ember+

AVT Audio Video Technologies

- 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.

Global Settings	SNMP			
Clients / Security				
General	SNMP SNMP Version:	v2c ~	[Category
Client Workplace Assignment			Alarm Traps	Category ^
Client Workplace Restrictions	Read/Trap Community:	public public	System Alarms	
Database	SNMP Port	161	LCA	
 Operation Settings 	NMS 1 (LAN/IP Adr./Port);	LAN 1 : 10.4.18.211 V 0	Temperature Sensor	
Workplace Definition	NMS 2 [LAN/IP Adr./Port]:	LAN 1:10.4.18.211 V 0	FLASH EPROM	
Audio Line			Overheated	
Intro / HOLD Signal	NMS 3 (LAN/IP Adr./Port):	LAN 1 : 10.4.18.211 V	MAIN EEPROM	
Signal Processing Line Labels	NMS 4 (LAN/IP Adr./Port):	LAN 1 : 10.4.18.211 V 0	Display Contrast DAC	
- Auto Answer	System Description:	THipPro Intercom 01		
GPIO	Contact		1/0 Port	
System Settings			Ethernet MAC 1	
- General	System Location:		Ethernet MAC 2	
- Line Interface	Send all traps at system s	tartup	Ethernet MAC 3	· ·
- Caller Line Grouping	Send traps immediately af		DSP 2 Boot	
- VoIP (LAN/SIP)	Send traps immediately and	ter enabling	DSP 2 Access	
- Audio Interface	Category A Alias:		Slot 1 Access	
PRETALK Streaming			Slot 2 Access	
AES67	Category B Alias:		Slot 3 Access	
LAN Interface	Category C Alias:		Module Assembly	
VLAN	Category D Alias:		DANTE Module Access	· ·
- DHD Audio Matrix	category o Alda.		Application Alarms	
Ember+			AES/EBU Framing Input 1	
- SNMP			AES/EBU Framing Input 2	
Login			AES/EBU Framing Input 3 AES/EBU Framing Input 4	
			MI AES/EBU Framind Induit 4	
			Set All D	lear All
			OK	Abbrechen Apply Now
ent ID: 1 Workplace: 3				

SNMP



MAGIC THipPro Intercom

Operation Settings

Settings that <u>can be</u> 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.

Global Settings	Workplace Definiti	Dn
Clients / Security General Client Workplace Assignment		
Client Workplace Restrictions	Name	Channels
Operation Settings		1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16
- Audio Line - Intro / HOLD Signal	Line Groups	5 3 4
- Signal Processing - Line Labels	Workplace 1	
Auto Answer GPIO	Workplace 2	
System Settings General	Workplace 3	
Line Interface Caller Line Grouping	Workplace 4	
VoIP (LAN/SIP) Audio Interface	Workplace 5	
PRETALK Streaming AES67 LAN Interface	Workplace 6	
- Ember+		
Login		
nt ID: 1 Workplace: 3		OK Abbrechen Apply Now
rt ID: 1 Workplace: 3		OK. Abbrechen Apply Now

Workplace Definition

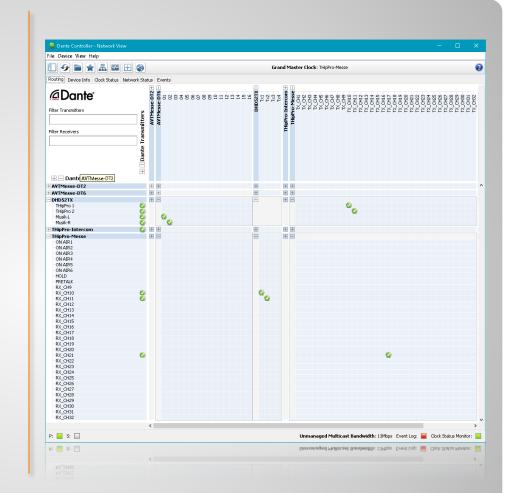


- On the AUDIO LINE page the audio interfaces are assigned to the telephone lines.
- It is possible to use the device audio interfaces and DANTE at the same time.
 However, DANTE and AES67 cannot be mixed.
- If an AES interface is only used as an output, the INPUT ALARM can be deactivated.

Global Settings	Audio Line							
General	Audio Line Assignment							
Client Workplace Assignment Client Workplace Restrictions	Audio	Audio Interface	No Input Alarr	n				
Database Operation Settings	Line 1	DANTE Ch. 1	• •					
- Workplace Definition	Line 2	DANTE Ch. 2						
Audio Line	Line 3	DANTE Ch. 3	•					
Intro / HOLD Signal Signal Processing	Line 4	DANTE Ch. 4	-					
- Line Labels	Line 5	DANTE Ch. 5	•					
Auto Answer	Line 6	DANTE Ch. 6	•					
🚋 GPIO System Settings	Line 7	DANTE Ch. 7	-					
General	Line 8	DANTE Ch. 8	-					
Line Interface	Line 9	DANTE Ch. 9	•					
Caller Line Grouping VoIP (LAN/SIP)	Line 10	DANTE Ch. 10	•					
- Audio Interface	Line 11	DANTE Ch. 11	•					
PRETALK Streaming AES67	Line 12	DANTE Ch. 12	-					
LAN Interface	Line 13	DANTE Ch. 13	-					
VLAN	Line 14	DANTE Ch. 14	-					
DHD Audio Matrix Ember+	Line 15	DANTE Ch. 15	•					
SNMP	Line 16	DANTE Ch. 16	•					
Login	Ext. HOLD Input	AES/EBU 1 Left	- V					
	Caution: Invalid settings a	re red		Clear All	Default Settings			
t ID: 1 Workplace: 3					OK Abbrechen Apply I	Now		
it ID: 1 Workplace: 3					OK Abbrechen Apply I	Now		

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.

Dante Controller - Device View (THipPro-Messe) le Device View Help	– O ×	
_		
♦ 🗶 🗠 🗄 🗄	ThipPro-Messe 🗸	
eceive Transmit Status Latency Device Config Network Con	nfig AE567 Config	
Dante Redundancy — Current: Re	dundant	
New: R	edundant 🗸	
Addresses		
Primary	Secondary	
 Obtain an IP Address Automatically (default) Manually configure an IP Address 	Obtain an IP Address Automatically (default) Manually configure an IP Address	
IP Address: 172 . 16 . 75 . 100	IP Address:	
Netmask: 255 , 255 , 0 , 0	Netmask:	
DNS Server: 172 . 16 . 75 . 1	DNS Server:	
Gateway: 172 . 16 . 75 . 1	Gateway:	
	👱 Dante Controller - Device View (THipPro-Messe)	– 🗆 X
Analy.	File Device View Help	
Apply	File Device View Help	Messe 🗸 😢
Reset Device Reboot		-
Reset Device	🖌 📰 🔤 < 🕀 🔓	-
Reset Device	🖌 📰 🔤 < 🕀 🔓	-
Reset Device	🖌 📰 🔤 < 🕀 🔓	-
Reset Device	Receive Transmit Status Latency Device Config Network Config AES67 Con	fo
Reset Device	Receive Transmit Status Latency Device Config Network Config AES67 Con	-
Reset Device	Receive Transmit Status Latency Device Config Network Config AES67 Con	fo
Reset Device	Receive Transmit Status Latency Device Config Network Config AES67 Con	fo Apply
Reset Device	Receive Transmit Status Latency Device Config Network Config AES67 Con Receive Transmit Status Latency Device Config Network Config AES67 Con Rename Device THipPro-Messe - Sample Rate Sample Rate: 105	fig
Reset Device	Receive Transmit Ratus Latency Device Config Network Config AES67 Con	fo Apply
Reset Device	Receive Transmit Status Latency Device Config Network Config AES67 Con	fig Apply Pull-up/down: This device does not support Pull-up/down configuration.
Reset Device	Receive Transmit Status Latency Device Config Network Config AES67 Con	fig
Reset Device	Receive Transmit Ratus Latency Device Config Network Config AES67 Con Receive Transmit Ratus Latency Device Config Network Config AES67 Con Rename Device THighthor Messe - Sample Rate - Sample Rat	fig Apply Pull-up/down: This device does not support Pull-up/down configuration.
Reset Device	Receive Transmit Status Latency Device Config Network Config AES67 Con Receive Transmit Status Latency Device Config Network Config AES67 Con Rename Device ThipPro-Messe Sample Rate Sample Rate Sample Rate Sample Rate Sample Rate Device Latency Device Latency	fig Apply Pull-up/down: This device does not support Pull-up/down configuration.
Reset Device	Rename Device Rename Device Rename Device ThipPro-Messe Sample Rate Sample Rate Sample Rate Sample Rate Sample Rate Device Latency Latency: 1,0 msec Latency: 1,0	fig Apply Pull-up/down: This device does not support Pull-up/down configuration.
Reset Device	Rename Device Receive Transmit Status Latency Device Config Network Config AES67 Con Receive Transmit Status Latency Device Config Network Config AES67 Con Rename Device ThipPro-Messe Sample Rate Sample Rate Sample Rate Sample Rate Cooking Preferred Encoding: PCM 24 Unc Device Latency: Latency: Latency: Latency: Reset Device	fig Apply Pull-up/down: This device does not support Pull-up/down configuration.

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.

Global Settings	Audio Line					
Clients / Security General	Audio Line Assignment					
- Client Workplace Assignment	Audio	Audio Interface	AES67 Rx	No Input Alarm		
Client Workplace Restrictions Database	Line 1	AES67 Ch. 1	▼ Str. 1; Ch. 1	•		
Operation Settings	Line 2	AES67 Ch. 2	▼ Str. 1; Ch. 2	•		
Workplace Definition <mark>Audio Line</mark>	Line 3	AES67 Ch. 3	▼ Str. 1; Ch. 3	•		
Intro / HOLD Signal	Line 4	AES67 Ch. 4	▼ Str. 1; Ch. 4	•		
Signal Processing	Line 5	AES67 Ch. 5	▼ Str. 1; Ch. 5	•		
Line Labels Auto Answer	Line 6	AES67 Ch. 6	▼ Str. 1; Ch. 6	•		
GPIO	Line 7	AES67 Ch. 7	▼ Str. 1; Ch. 7	•		
System Settings General	Line 8	AES67 Ch. 8	▼ Str. 1; Ch. 8	•		
General Line Interface	Line 9	AES/EBU 1 Left	•			
Caller Line Grouping	Line 10	AES/EBU 1 Right	•			
VoIP (LAN/SIP) Audio Interface	Line 11	AES/EBU 2 Left	•			
PRETALK Streaming	Line 12	AES/EBU 2 Right	-			
AES67	Line 13	AES/EBU 3 Left	-			
LAN Interface VLAN	Line 14	AES/EBU 3 Right	-			
- DHD Audio Matrix	Line 15	AES/EBU 4 Left	•			
Ember+ SNMP	Line 16	AES/EBU 4 Right	-			
- Login	Ext. HOLD Input	XLR Analogue 1	•			
	Caution: Invalid settings a	ire red		Clear All	Default Settings	
				CIGB AI	D'ordun Jonnigs	
ent ID: 1 Workplace: 3					OK Abbrechen Apply N	low
					OK Abbrechen Apply N	1014
ent ID: 1 Workplace: 3					OK Abbrechen Apply N	Um

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).

Orient Workplace Assignment Offent Workplace Assignment Offent Workplace Assignment Offent Workplace Retructions Operation Settings Operation Settings Outpasse Operation Settings Outpasse	after connection establish tro when call is accepture (Announcements: e after announcement) uce: tro when switching to H (Announcements: e after announcement)	d manualy 2 1 second Associated Line	> Stor # 1 2 3 4 5 6	Line 1 Line 2 Line 3 Line 4	Length 1,70 sec 1,68 sec 1,75 sec 1,76 sec	Import Import Import	Rec./Play/Del Rec./Play/Del Rec./Play/Del
Client Workplace Restrictions Database Operation Settings Workplace Definition Paue im Audio Line Signal Processing Line Labels Auto Answer System Settings Place im System Settings Place System Settings Superation Superation Superation System Settings Superation Supera	Announcements: e after announcement: urce: tro when switching to H Announcements:	2 1 second Associated Line OLD	 ✓ ✓ ✓ ✓ ✓ ✓ ✓ ✓ ✓ 	Line 1 Line 2 Line 3 Line 4	1,70 sec 1,68 sec 1,75 sec 1,76 sec	Import Import	Rec./Play/Del Rec./Play/Del
Database Database Detation Strings Workplace Definition Database Detation Strings Detation Database Detation	e after announcement: urce: tro when switching to H (Announcements:	1 second Associated Line OLD	× 2 3 4 × 5	Line 2 Line 3 Line 4	1.68 sec 1.75 sec 1.76 sec	Import Import	Rec./Play/Del Rec./Play/Del
Operation Setting: Pause lim: - Workplace Definition	urce: tro when switching to H i Announcements:	1 second Associated Line OLD	× 3 4 × 5	Line 3 Line 4	1,75 sec 1,76 sec	Import	Rec./Play/Del
Workplace Definition Audio Line Introv / HOLD Sgnal HoLD Signal HoLD Signal Gameal	urce: tro when switching to H i Announcements:	Associated Line	3 4 ~ 5	Line 4	1,76 sec	-	
Intro. HOLD Signal HOLD Signal Signal Processing Signal Processing - Line Labels Play In - Auto Answer Play In - Signal Processing Signal Society - Signal Three Settings Play In - General Pause time - Caller Line Force Caller Line Society - Caller Line Groupping - Speech Synth - Audio Interface Caller Line Groupping	tro when switching to H Announcements:	OLD	~ 5			Import	
Line Labels Auto Annover Auto Annover System Settings General System Settings Speech Synth -Volp (LAM/SIP) -Volp (LAM/SIP) -Volp (LAM/SIP)	tro when switching to H Announcements:	OLD	-	Line 5	4.70		Rec./Play/Del.
Auto Anneer Play In System Strings Overneal Play In Overneal Pl	Announcements:		6		1,78 sec	Import	Rec./Play/Del
System Settings Number of System Settings Pause time Line Interface Caller Line Grouping Speech Synth Volp (UAN/SIP) Audio Interface		1		Line 6	1,89 sec	Import	Rec./Play/Del.
- General Pause tim - Line Interface - Caller Line Grouping Speech Synth - VolP (LAN/SIP) - Audio Interface	e after announcement:		~ 7	Line 7	1.84 sec	Import	Rec./Play/Del.
- Line Interface - Caller Line Grouping - Speech Synth - VoIP (LAN/SIP) - Audio Interface	e alter announcement.	without pause	. 8	Line 8	1.66 sec	Import	Rec./Play/Del.
VoIP (LAN/SIP) Audio Interface		without pause		Line 9	1.75 sec	Import	Rec./Play/Del
Audio Interface	iesis		1		1.73 sec	Import	Rec./Play/Del
	Generate Intros fr		_	1 Line 11	1,85 sec	Import	Rec./Play/Del
PRETALK Streaming	Generate Intros ri	om intro info	-	2 Line 12	1.81 sec	Import	Rec./Play/Del
AES67	Generate Intros fro	m Line Labels		2 Line 12 3 Line 13	.,		
LAN Interface			-		2,03 sec	Import	Rec./Play/Del.
DHD Audio Matrix			1.		2,04 sec	Import	Rec./Play/Del.
Ember+ SNMP			-	5 Line 15	2,00 sec	Import	Rec./Play/Del
Login			11	6 Line 16	2,00 sec	Import	Rec./Play/Del
					Missing Intros	are red!	

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.

Blobal Settings	Intro / HOLD Signal						
Clients / Security General	Play Intro after connection establis	shed	Stored	Intros			
Client Workplace Assignment	Skip Intro when call is accepte		#	Info	Length		
Client Workplace Restrictions	Number of Announcements:	2 ~	1	Line 1	1,70 sec	Import	Rec./Play/Del
Database operation Settings			2	Line 2	1,68 sec	Import	Rec./Play/Del
Workplace Definition	Pause time after announcement:	1 second V	3	Line 3	1,75 sec	Import	Rec./Play/Del
- Audio Line Intro / HOLD Signal	HOLD Signal		4	Line 4	1,76 sec	Import	Rec./Play/Del
Signal Processing	Signal Source:	Associated Line V	5	Line 5	1,78 sec	Import	Rec./Play/Del
ne Labels uto Answer	Play Intro when switching to H	IOLD	6	Line 6	1,89 sec	Import	Rec./Play/Del
SPIO	Number of Announcements:	1 ~	7	Line 7	1.84 sec	Import	Rec./Play/Del
ystem Settings	Pause time after announcement:	without pause ~	8	Line 8	1,66 sec	Import	Rec./Play/Del
General Line Interface	r ause une arei announcement.	ww.ibut.pause	9	Line 9	1,75 sec	Import	Rec./Play/Del
Caller Line Grouping	Speech Synthesis		10	Line 10	1,73 sec	Import	Rec./Play/Del
VoIP (LAN/SIP) Audio Interface	Generate Intros fr	rom lutro luto		Line 10	1,75 sec	Import	Rec./Play/Del
PRETALK Streaming	denerate muos n	ion muo mio		Line 12	1,81 sec	Import	Rec./Play/Del
AES67 LAN Interface	Generate Intros fro	m Line Labels		Line 13	2.03 sec	Import	Rec./Play/Del
VLAN			14	Line 14	2.03 sec	Import	Rec./Play/Del
DHD Audio Matrix	\ \		15	Line 14	2,04 sec	Import	Rec./Play/Del
Ember+ SNMP		Λ		Line 15	2,00 sec	Import	Rec./Play/Del
ogin		\					
					Missing Intros	are red!	
Workplace: 3						0K.	Abbrechen A
						OK OK	Apprechen A
	Voice Parameter						
						OK	
		soft Zira Desktop - Engli	sh (United	States)	~	OK	
	Voice: Micros	soft Zira Desktop - Engli				OK	
	Voice: Micros					OK	
	Voice: Micros Rate:					OK	
	Voice: Micro: Rate: · ·					OK	
	Voice: Micro: Rate: · ·					OK	
	Voice: Micro: Rate: · ·	precher (3- USB PnP So				OK	
	Voice: Micros Rate: Volume: . Test: Lauts	precher (3- USB PnP So				OK	

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.

HOLD Signal Recording		
HOLD Signal Name:	Line 1	
Signal duration:	1,68 sec	
Record source:	AES/EBU 1 Left (P	RETALK) V
	max	. 16 sec
HOLD signal recording		Test recorded HOLD signal
	Save	
	Delete File	Close
	Delete File	Close

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.

Global Settings	Signal Process	ing									
Clients / Security General	Line Settings	3					Automatic Ga	ain Control Setti	ngs (AGC) / Expa	nder	
- Client Workplace Assignment	Line	Automatic Gain Control		Expander	Send Level Booster		Threshold				-40 dBFS
Client Workplace Restrictions	1	off	•	ON	Reset after drop	•	Level	_			-25 dBF9
- Database Operation Settings	2	off	-	ON	Reset after drop	-	Limiter:				0 dBFS
Workplace Definition	3	off	•	ON	Reset after drop	-					0 001
- Audio Line	4	off	•	ON	Reset after drop	-	Speed	Medium	~		
Intro / HOLD Signal						•	Volume contr	ol default value			
Signal Processing Line Labels	5	off	•	ON	Reset after drop	-			-		0 d
- Auto Answer	6	off	•	ON	Reset after drop	•					
⊕- GPIO	7	off	•	ON	Reset after drop	-	Echo Cancel	er			
System Settings	8	off	•	ON	Reset after drop	•	_				
General Line Interface	9	off	•	ON	Reset after drop	•	Line Basis [elay: 60	msec (012	J	
– Caller Line Grouping	10	off	-	ON	Reset after drop	•	Line Level (Offset Attenuati	on (Send):		
VoIP (LAN/SIP)	11	off	•	ON	Reset after drop	•				•	-6 d
Audio Interface	12	off	-	ON	Reset after drop	•	Send Level B				
PRETALK Streaming AFS67	13	off		ON	Reset after drop	•					
AESD/ LAN Interface	14	off	-	ON	Reset after drop	•	Preset 1:	+6 dB	~		
VLAN	15	off	-	ON	Reset after drop	•	Preset 2:	+12 dB	~		
DHD Audio Matrix Ember+	16	off	•	ON	Reset after drop	•	Preset 3:	+18 dB	~		
Login											
		Set AGC on/off for all lines		Toggle	Send Level Booster					Default Se	ttings
							_				
it ID: 1 Workplace: 3								OK	Abbrechen	Apply Now	
								OK	Abbrechen	Apply Now	
t ID: 1 Workplace: 3							L	OK.		A marks Monu	

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 Process	ing				
Clients / Security General	Line Settings					Automatic Gain Control Settings (AGC) / Expander
Client Workplace Assignment	Line	Automatic Gain Cont	rol	Expander	Send Level Booster	Threshold: -40 dE
Client Workplace Restrictions Database	1	off	•	ON	Reset after drop	• Levet -25 dE
peration Settings	2	off	•	ON		Limiter: 0 dE
Workplace Definition	3	off		ON	Reset after drop	▼ Speed: Medium ~
Audio Line Intro / HOLD Signal	4	off	•	ON	Reset after drop	·
- Signal Processing	5	off	-	ON	Reset after drop	Volume control default value
Line Labels	6	off	•	ON	Reset after drop	
Auto Answer - GPIO	7	off	•	ON	Reset after drop	+ Echo Canceller
rstem Settings	8	off	•	ON	Reset after drop	▼ Active
General	9	off	•	ON	Reset after drop	Line Basis Delay: 60 msec (0120)
Line Interface Caller Line Grouping	10	off	•	ON	Reset after drop	Line Level Offset Attenuation (Send):
Caller Line Grouping VolP (LAN/SIP)	11	off	•	ON	Reset after drop	• • • •
- Audio Interface	12	off	•	ON	Reset after drop	
PRETALK Streaming AES67	13	off	•	ON	Reset after drop	Send Level Booster
AESD/ LAN Interface	14	off	•	ON	Reset after drop	Preset 1: +6 dB
VLAN	15	off	•	ON		
- DHD Audio Matrix	16	off		ON	Reset after drop	
Ember+ SNMP						
gin		Set AGC on/off for all lin		T da	Send Level Booster	
		Set Auc on/or for all in	es	roggie	Send Level Booster	Default Settings
D:1 Workplace: 3						
D:1 Workplace: 3						OK Abbrechen Apply Now
D: 1 Workplace: 3						OK Abbrechen Apply Now

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				
Clients / Security General	Line Labels		Variables		
- Client Workplace Assignment			 (index)	Line index [1 based]	
Client Workplace Restrictions	Line	Label			
Database	1	Line (index)	 {lineid}	SIP user*	
Operation Settings	2	Line (index)	(grp)	Name of the line group*	
Workplace Definition	3	Line (index)	(sipsrv)	Used SIP server*	
Audio Line Intro / HOLD Signal	4	Line (index)	{sipsrv#}	1: Main SIP Server	
- Signal Processing	5	Line (index)	(3)()31441)	2: Backup SIP Server	
Jine Labels	6	Line (index)	{sipaut}	Used SIP authentication"	
- Auto Answer	7	Line (index)			
GPIO	8	Line (index)	{sipdisp}	SIP display name"	
System Settings	9	Line (index)	{airai}	Name of the Audio Interface"	
General	10	Line (index)	{sysname}	System Name"	
Line Interface	11	Line (index)	* A formationer	cifier can be added.	
Caller Line Grouping VoIP (LAN/SIP)	12	Line (index)	:# Take	only the last # characters	
- Audio Interface	13	Line (index)	:-# Take Example:	only the first # characters {ineid:4}	
PRETALK Streaming	14	Line (index)			
AES67	15	Line (index)			
- LAN Interface	16	Line (index)			
VLAN					
DHD Audio Matrix Ember+					
SNMP					
Login		Default Settings			
nt ID: 1 Workplace: 3				OK Abbrechen Apply N	
nt ID: 1 Workplace: 3				OK Abbrechen Apply N	OM

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		×
Glash Strings: Glash St	Auto Answer Auto Answer on al channels Channel Index T Z 4 6 6 7 0 10 T Z 4 6 Line Groups 1 3 Auto Answer Channel V V Answer call ox Line Group selection	
Cone Life Stropping - Voip (LANSP) - Audio Interface - PRETAL Streaming - AES67 - LAN Interface - VLAN - DHD Audio Matrix - Ember+ - SIMP - Login	Auto Answer Delay: Line Grp. 1 Line Grp. 2 Line Grp. 2 Line Grp. 3 Line Grp. 4 Line Grp. 5 Line Grp. 5 Line Grp. 5 Line Grp. 7 Line Grp. 7	•
Client ID: 1 Workplace: 3 Client ID: 1 Morkplace: 3	OK Abbechen 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.)

- Clients / Security				
General	Pin	Dir.	Function 1 (Positive Edge)	Function 2 (Negative Edge)
Client Workplace Assignment	TTL 1 (Pin 1)	In	Call Out (Level Trig.): VolP Line 1: Audio Line Line 1: Number 502	
Client Workplace Restrictions	TTL 2 (Pin 2)	In		
Database Operation Settings	TTL 3 (Pin 3)	In		
-Workplace Definition	TTL 4 (Pin 4)	In		
Audio Line	TTL 5 (Pin 5)	In		
- Intro / HOLD Signal	TTL 6 (Pin 6)	In		
Signal Processing	TTL 7 (Pin 7)	In		
Line Labels	TTL 8 (Pin 8)	In		
Auto Answer	Relay 1 (Pins 14->15)	Out	Connection State: Any VoIP Line : Connection State connect	
GPIO	Relay 2 (Pins 17->18)	Out	Always Open	
TTL / Relay	Relay 3 (Pins 19->20)	Out	Always Open	
DHD	Relay 4 (Pins 21->9)	Out	Always Open	
Set Logic	Relay 5 (Pins 22->10)	Out	Always Open	
Ember+	Relay 6 (Pins 23->11)	Out	Always Open	
Input Output	Relay 7 (Pins 24->12)	Out	Always Open	
Consumer Functions	Relay 8 (Pins 25->13)	Out	Always Open	
System Settings			· · · · · · · · · · · · · · · · · · ·	
General				
- Line Interface				
Caller Line Grouping				
VoIP (LAN/SIP)				
Audio Interface				
PRETALK Streaming				
AES67				
LAN Interface VLAN	<			>
VLAN DHD Audio Matrix				
Emphany				
- Embert				
				OK Abbrechen Apply

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.

Global Settings	Ember+									
Clients / Security General	Identifier									
Client Workplace Assignment	Dial Pad GPI0 Identifier:	AVTDialPad		(Predefi	ned)					
Client Workplace Restrictions Database	GPI0 Identifier:	AVTGPI001		(132)						
Operation Settings		AVTGPI002		(3364)						
Workplace Definition Audio Line		AVTGPI003		(6596)						
Intro / HOLD Signal	Ember+ Consumer to Client	assignment								
Signal Processing Line Labels			r+ Provider							
Auto Answer GPIO		1	2 3	4	5	6 7	8			
	Client 1: OPERATOR	~	—		-		-			
Input	Client 2: ICC-TRL-001		-	-	-		-			
Consumer Functions	Client 3: ICC-TRL-002		-	-	-		-			
General Line Interface	Client 4: ICC-TRL-003		-	-	-		-			
Caller Line Grouping VoIP (LAN/SIP)										
Audio Interface										
PRETALK Streaming AES67										
LAN Interface										
VLAN DHD Audio Matrix										
Ember+										
Login										
-										
ID: 1 Workplace: 1								OK	Abbrechen	Apply Now
ID: 1 Workplace: 1								OK	Abbrechen	Apply Now

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.

ippro.ssw	Browse
ppr0.ssw ✓ 01:10.4.18.211:System01 ✓ 02:10.4.18.212:System02 ✓ 03:10.4.31.5:System03 ✓ 04:10.3.190.8:System04	Progress
	Close

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 Mo	onitor - System02									-	
Keep	window on top										
System a	alarms						System	State			
•	0 LCA	9	0 Ove	erheated			System	Temperatur 51 °C	D SP L	.oad: 35	%
9	0 Time Keeper	9	0 MA	IN EEPRON	1						
9	0 Temperature Sensor	9	0 Dis	play Contr	ast DA	NC	IP Trans	mission Jitter	Current	Max	imum last 30 seo
9	0 FLASH EPROM]	Line 1 V		0 msec	0 msec
•	0 VCXO	9	0 I/O	Port			- L			omace	omace
•	0 DSP 2 Boot	9	0 DSF	2 Access				Streams			
•	0 Slot 1 Access	9	0 Slot	t 2 Access			•	IP Stream 1	0	IP Stream 2	
•	0 Slot 3 Access	•	0 Mo	dule Asser	mbly		•	IP Stream 3		IP Stream 4	
•	0 Ethernet MAC 1	9	0 Eth	ernet MAC	2		•	IP Stream 5	•	IP Stream 6	
•	0 Ethernet MAC 3	9	0 Eth	ernet MAC	4		•	IP Stream 7	•	IP Stream 8	
9	0 DANTE Module Access	9	0 Red	dundant Po	wer Su	pply	•	IP Stream 9	•	IP Stream 10	
Applicat	ion alarms						Connect	ed PCs			
0	0 AES/EBU Input 1	No Al	ES signal	available			1: OPER	ATOR:INT			
0	0 AES/EBU Input 2	No Al	ES signal	available							
9	0 AES/EBU Input 3	No Al	ES signal	available			Databas	e			Master PC
9	0 AES/EBU Input 4	No Al	ES signal	available			•	1: 10.4.18.5\SQLEXPR	ESS.AVTInterco	m	PC 1
9	0 SIP Registration										
Ethernet	t state		Abs. da	ta rates							
LAN 1	0 100 MBit/s, full		TX: 3	35,6 kBit/s	RX:	5,4 kBit/s					
LAN 2	0 100 MBit/s, full		TX:	0 Bit/s	RX:	761 Bit/s					
LAN 3			TX:		RX:						
LAN 4	O		TX:		RX:						
DANTE	0										
Last Cou	unter Reset: 18.02.2019 16:15	:13									
	Alarm Counter Reset		Audi	o Interface	Monito	or		SIP State Monitor		Close	
	Alarm Counter Reset		Haar	o Interface				SIP State Monitor		Close	_

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.

LAN 2 0 100 MBit/: LAN 3 0 0 LAN 4 0 0 DANTE 0 0 Last Counter Reset: 18.02.2	TX: TX:	/s RX: 761 Bit/s RX: RX:		
Alarm Counter Res	et Audio Interfa	ce Monitor SIP	⁹ State Monitor	Close
SIP State Monitor - System01				-
Keep window on top				
SIP User	Main SIP Server		Backup SIP Server	
601	Test Registration done s	uccessfully	Test No IP address ava	ilable
602	Test Registration done s	uccessfully	Test No IP address ava	ilable
603	Test Registration done s	uccessfully	Test No IP address ava	ilable
604	Test Registration done s	uccessfully	Test No IP address ava	ilable
605	Test Registration done s	uccessfully	Test No IP address ava	ilable
606	Test Registration done s	uccessfully	Test No IP address ava	ilable
607	Test Registration done s	uccessfully	Test No IP address ava	ilable
608	Test Registration done s	uccessfully	Test No IP address ava	ilable
609	Test Registration done s	uccessfully	Test No IP address ava	ilable
610	Test Registration done s	uccessfully	Test No IP address ava	ilable
611	Test Registration done	Record SIP Logfile - System01		- [
612	Test Registration done	Record on Lognic Systems		
613	Test Registration done	Keep window on top		
614	Test Registration done		Logfile stopped	
615	Test Registration done	SIP User filter:	601	
616	Test Registration done			
ſ			Start SIP registering	
	Record SIP Logfile			
		Start Logging		Stop Logging
	Record SIP Logfile			
		View Logfile		Save Logfile
			Close	

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.

LAN 3 🕢 0 LAN 4 🔍 0 DANTE 🕥 0			0 Bit/s RX: 761 Bit	./s				
-		TX:	RX:					
		TX:	RX:					
Last Counter Reset:	19 02 2010 16-15-1	2						
Last Counter Reset:	18.02.2019 16:15:1	·						
Alarm Cour	nter Reset	Aud	lio Interface Monitor	SIP	State Monitor		Close	
idio Interface Monitor	- TestFloor							
Keep window on top	🗹 Show also level i	info for disabled ch	annels					
Test Tone Generator:	DANTE Channel 1	~						
Audio Interface	Input	Output	Audio Interface	Input O	utput	Audio Interface	Input	Output
AES/EBU 1 Left			DANTE Channel 1			DANTE Channel 17		
AES/EBU 1 Right			DANTE Channel 2			DANTE Channel 18		
AES/EBU 2 Left			DANTE Channel 3			DANTE Channel 19		
						DANTE Channel 20		
AES/EBU 2 Right			DANTE Channel 4			DAINTE Chamiler 20		
AES/EBU 2 Right AES/EBU 3 Left			DANTE Channel 4 DANTE Channel 5			DANTE Channel 21		
AES/EBU 3 Left			DANTE Channel 5			DANTE Channel 21		
AES/EBU 3 Left AES/EBU 3 Right			DANTE Channel 5 DANTE Channel 6			DANTE Channel 21 DANTE Channel 22		
AES/EBU 3 Left AES/EBU 3 Right AES/EBU 4 Left			DANTE Channel 5 DANTE Channel 6 DANTE Channel 7			DANTE Channel 21 DANTE Channel 22 DANTE Channel 23		
AES/EBU 3 Left AES/EBU 3 Right AES/EBU 4 Left AES/EBU 4 Right			DANTE Channel 5 DANTE Channel 6 DANTE Channel 7 DANTE Channel 8			DANTE Channel 21 DANTE Channel 22 DANTE Channel 23 DANTE Channel 24		
AES/EBU 3 Left AES/EBU 3 Right AES/EBU 4 Left AES/EBU 4 Right XLR Analogue 1			DANTE Channel 5 DANTE Channel 6 DANTE Channel 7 DANTE Channel 8 DANTE Channel 9			DANTE Channel 21 DANTE Channel 22 DANTE Channel 23 DANTE Channel 24 DANTE Channel 25		
AES/EBU 3 Left AES/EBU 3 Right AES/EBU 4 Left AES/EBU 4 Right XLR Analogue 1 XLR Analogue 2			DANTE Channel 5 DANTE Channel 6 DANTE Channel 7 DANTE Channel 8 DANTE Channel 9 DANTE Channel 9			DANTE Channel 21 DANTE Channel 22 DANTE Channel 23 DANTE Channel 24 DANTE Channel 25 DANTE Channel 26		
AES/EBU 3 Left AES/EBU 3 Right AES/EBU 4 Left AES/EBU 4 Right XLR Analogue 1 XLR Analogue 2 Handset 1			DANTE Channel 5 DANTE Channel 6 DANTE Channel 7 DANTE Channel 8 DANTE Channel 9 DANTE Channel 10 DANTE Channel 11			DANTE Channel 21 DANTE Channel 22 DANTE Channel 23 DANTE Channel 24 DANTE Channel 25 DANTE Channel 26 DANTE Channel 27		
AES/EBU 3 Left AES/EBU 3 Right AES/EBU 4 Left AES/EBU 4 Right XLR Analogue 1 XLR Analogue 2 Handset 1			DANTE Channel 5 DANTE Channel 6 DANTE Channel 7 DANTE Channel 8 DANTE Channel 9 DANTE Channel 10 DANTE Channel 11 DANTE Channel 12			DANTE Channel 21 DANTE Channel 22 DANTE Channel 23 DANTE Channel 24 DANTE Channel 25 DANTE Channel 26 DANTE Channel 27 DANTE Channel 28		
AES/EBU 3 Left AES/EBU 3 Right AES/EBU 4 Left AES/EBU 4 Right XLR Analogue 1 XLR Analogue 2 Handset 1			DANTE Channel 5 DANTE Channel 6 DANTE Channel 7 DANTE Channel 8 DANTE Channel 9 DANTE Channel 10 DANTE Channel 11 DANTE Channel 12 DANTE Channel 12			DANTE Channel 21 DANTE Channel 22 DANTE Channel 23 DANTE Channel 24 DANTE Channel 25 DANTE Channel 26 DANTE Channel 27 DANTE Channel 28 DANTE Channel 29		

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.

Registration



Hardware	MAGIC THipPre	MAGIC THipPro				
Main	DANTE	AEC 16 Module				
Subject Number:	450063					
Factory Number:	17/44/1193	17/44/1193				
Year:	2017	2017				
Hardware Version:	3.00	3.00				
MAC Address 1:	00-06-9B-02-2	00-06-98-02-23-09				
MAC Address 2:	00-06-9B-02-2	00-06-9B-02-23-0A				
Modules:	free 🗸 🗸	DANTE				
	AEC16 V					
Features						
ZUD US	Intercom Basic					
HD Voice VoIP channels Number of Inter Number of PRET AE567	com Client license:					
VoIP channels Number of Inter Number of PRET AE567	com Client license: 'ALK Audio Stream	5 5				
VoIP channels Number of Inter Number of PRET AE567	com Client license: 'ALK Audio Stream	5 5				
VoIP channels Number of Inter Number of PRET AE567	com Client license: 'ALK Audio Stream	5 5				

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

AVT	PC Software Ve Firmware Version 3.0 Copyright AVT Audio Video Tec Nordostp. D-90411 N Germa Hotline: +49 91 Fax: +49 911 Internet: www.	060 (all systems) 2019 hnologies GmbH ark 91 ärnberg ny 1 5271 110 5271 100 <u>avt-nbg.de</u>	60			
Show	E-Mail: support@	CLOSE	: MAG		rcom Version 3.06	0
			System02: Firm	ware Version 3.(060	
			Show Address		CLOSE	

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
- \rightarrow Configure under SIP Server

fs1.ims.swisscom.ch@swisscom.ch:5070

Tips & Tricks (2)



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Email: support@avt-nbg.de

Phone: +49 911 5271-110

Support

