



# MAGIC AC1 Go

## **Configuration Guide**

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# MAGIC AC1 Go

### Overview

- Hardware
- Audio Codecs
- Operation Modes

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# Hardware - Front



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



# Hardware - Rear



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

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



Standard

#### Signalling

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

### Optional

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



	ISO/MPEG1/2 Layer 2/3	AAC-LD AAC-ELD	HE-AAC-LC HE-AAC-V1/2	G.722	G.711	PCM16/20/24	OPUS	FLAC
Algorithm	Sub-Band, Psychoacoustics	Sub-Band, Psychoacoustics	Sub-Band, Psychoacoustics	ADPCM	РСМ	N/A	SILK CELT	PCM
Structure	Frames	Frames	Frames	Bytes	Bytes	Bytes	Frames	Frames
Bandwidth	20-Hz 20-kHz	20-Hz 20-kHz	20-Hz 20-kHz	50-Hz 7-kHz	300-Hz 3.4-kHz	20-Hz 20-kHz	20-Hz20-kHz	20-Hz20-kHz
Sampling Rates (kHz)	MPEG1: 32, 48 MPEG2: 16, 24	16, 24, 32, 48	LC: 32, 48 V1/V2: 16, 24	16	8	32, 48	max. 48 dynamic	48
Data rates (kbit/s)	Typical: 128192kbit/s	Typical: LD: 64 128 ELD: 48136	Typical: LC: 96 V1: 80 V2: 64	48, 56, 64	56, 64	5122304	16384	Typical: 300 2000
Delay	Layer 2: 150 ms Layer 3: 250 ms	LD: < 50 ms ELD: < 30 ms	LC: 350 ms V1/V2: 450 ms	< 10 ms	< 5 ms	< 3 ms	<30 ms	



AVT

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

**Fechnoloaies** 

- Sending two streams with identical content
- A delay between the streams can be configured

IP Leased Line

• Streams can be sent via separate VLANs

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

- Establish a connection
  - by dialling a phone number via a SIP server
  - by entering the IP address of the receiver via Direct SIP
- Support for up to 5 SIP provider profiles which are active simultaneously
- Uses SDP to negotiate the coding algorithm
- Audio transmission via RTP over UDP
- Numbered packets allow for packet loss detection
- The packet size is variable if it is not determined by the codec used.
- Max. buffer size: 2 seconds
- Mitigation of lost packets by repeating packets on the receiving side

- Support for the STUN protocol (Session Traversal Utilities for NAT)
  - Simplified: Conversion of the internal IP address into an external IP address
- Supports Quality of Service (DiffServ) to prioritize audio and SIP independently
- Interworking with SIP telephones, in SD quality or HD-Voice
- Secure Streaming Function
  - Sending two streams with identical content
  - A delay between the streams can be configured





# MAGIC AC1 Go

## First Start

• Connecting the PC software

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





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





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







# MAGIC AC1 Go

## Configuration

Local Settings

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Local Settings		×
Application Parameters	Settings Folder	
- Main Window Prese	ntation	
🗌 Remember last v	window position	
Show window o	n fixed position X: 800 Y: 600	
Screen resolutio	on: 1920*1200	
Main window siz	ze: 1024*768 ~	
	OK Abbrechen	

# Audio Video Technologies Application Parameters (1)

AV

IJ

- Define the appearance of the PC software on the APPLICATION PARAMETERS configuration page.
- MAIN WINDOW PRESENTATION: Define position and size of the main window of the PC software.
  - REMEMBER LAST WINDOW POSITION: The last window position of the main window is stored when closing the app. The next time the app is started, the window is displayed in the same place if this option is enabled. Otherwise, it is displayed in the top left corner of the main screen.
  - SHOW WINDOW ON FIXED POSITION: Define the position of the top left corner of the application window on the screen when the application starts. The top left corner of the main screen has the coordinates X=0, Y=0.
  - SCREEN RESOLUTION: Displays the screen resolution of the monitor the application is displayed on as it is provided by the operating system. This may be different from the resolution defined in the graphic card driver due to the high DPI scaling feature of the operating system.

Video

Technologies

- MAIN WINDOW SIZE: Select one of the predefined windows sizes.
  - The window size can also be adjusted by dragging the frame of the main window with the mouse.
- SHOW DETAILED CODEC INFORMATION IN CODEC LINE: Enable this option to show detailed information about the codec used for audio transmission of the codec channel (AC).
   Otherwise, a simpler status information is shown.
   More details are displayed when clicking on the simple status information.

# Application Parameters (2)

	· · · ·	
I OCAL	Sett	inc
Local	See	y

plication Parameters Settings Folder
Save settings*
O for current user (nonroaming) O for current user (roaming)
for all users
🔿 in this folder
Browse
General*
Save settings encrypted
* Windows Administrator Rights may be needed
OK Abbrechen



- Define the storage location of the local settings on the SETTINGS FOLDER page.
- Local settings include all settings in the LOCAL SETTINGS window as well as the settings under MENU > CONFIGURATION > CONTROL INTERFACE.
- The selected storage location determines which Windows user rights are required to change the local settings.
- Changing these settings may require administrator rights.
- Select a settings location:

Video

Technologies

- LOCAL SETTINGS VALID ONLY FOR CURRENT USER: The settings are saved in the user directory of the logged-in Windows account. User rights are sufficient to change the local settings.
   (%APPDATA%\LOCAL\AVT\MAGIC THipPro LITE)
- FOR ALL USERS: All users of the PC use the same settings. Windows Administrator rights are required to change the local settings. (%PROGRAMDATA%\AVT\MAGIC THipPro LITE)

- STORE SETTINGS IN CUSTOM PATH: The settings are saved in an adjustable folder path. The required user rights are determined by the file's properties. The path is saved in the settings.ini file in the installation directory.
- STORE SETTINGS ENCRYPTED: Enable to encrypt the content of the local settings file.

# Settings Folder (2)



# MAGIC AC1 Go

## Configuration

Operation Settings

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Citents / Security       Computer Access tights)         - Line Interface       Computer Access tights)         - Audio Assignment       Auto Answer         - Auto Answer       1       Operator         - TIL / Relay       1       Operator         - OHD Sct logic       2       Auto Answer         - General       3       Auto         - Statz Defined       5       Incomputer Access List         - General       6       Incomputer Access List         - General       8       Incomputer Access List         - System Settings       8       Incomputer Access List         - Actor Access List (access rights)       Incomputer Access List       Incomputer Access List         - Duto Set Logic       1       Operator       PC-OP-1         - OHD Set Logic       4       Incomputer Access List       Incomputer Access List         - System Settings       4       Incomputer Access List       Incomputer Access List         - Actio Interface       9       Incomputer Access List       Incomputer Access List         - Actio Interface       11       Incomputer Access List       Incomputer Access List         - ALN Interface       12       Incomputer Access List       Incomputer Access         - VLAN	Operation Settings     Clients / Se	Security			
- Audio Assignment       - Auto Answer       Client       Alias       Computer Name / IP Address       Add this PC to list         - Th / Relay       1       Operator       PC-OP-1       Add this PC to list         - OHD Set Logic       2       -       -       -       -         - Th / Relay       3       -       -       -       -         - Th / Relay       4       -       -       -       -         - Th / Relay       3       -       -       -       -         - St brtz Default       5       -       -       -       -       -         - St brtz Default       6       -       -       -       -       -         - General       7       -       -       -       -       -         - System Settings       7       -       -       -       -       -       -         - Actio Interface       9       -<	Clients / Security     Line Interface     AoIP (LAN/SIP)	puter Access List (access rights)—			
- Alarm Signalling       15       16         - SNMP       16       16         - ACconnect       17       17         - DHD       18       19         - PhonerSet       19       20         - Login       An empty list means no access protection is active!	<ul> <li>Audio Assignment</li> <li>Auto Answer</li> <li>TTL / Relay</li> <li>DHD Set Logic</li> <li>Ember+</li> <li>Transmission Modes</li> <li>General</li> <li>15 kHz Default</li> <li>User Defined</li> <li>Juser Defined</li> <li>System Settings</li> <li>General</li> <li>Audio Interface</li> <li>Audio Interface</li> <li>Data Interface</li> <li>VLAN</li> <li>NTP</li> <li>Alarm Signalling</li> <li>SNMP</li> <li>Alarm Signalling</li> <li>SNMP</li> <li>Ember+</li> <li>PhonerSet</li> <li>Quick Dials</li> <li>Connect</li> <li>Duber</li> <li>An endoted</li> </ul>	Client     Alias       1     Operator       2     0       3     0       4     0       5     0       6     0       7     0       8     0       9     0       10     0       11     0       12     0       13     0       14     0       15     0       16     0       17     0       18     0       19     0       20     0	Computer Name / IP Address PC-0P-1	Add this PC to list	

#### Audio Video Technologies Clients / Security (1)

AVI

- When a preset is created, only the settings in the **OPERATION SETTINGS** branch are saved in the preset. These settings affect the audio transmission and can therefore be flexibly reloaded via presets.
- Access to the device via PC software can be restricted on the CLIENTS / SECURITY settings page.
- COMPUTER ACCESS LIST:
  - If the list is empty, the system can be accessed from any PC.
  - If at least one client is entered, only PCs that are in the list can access the system.
- CLIENT: Internal ID of each PC.

Video

ALIAS: A freely selectable designation for the PC.

- COMPUTER NAME / IP ADDRESS: This entry is used to check the access rights of a client to the system. Either the Windows computer name or the IP address of the PC can be entered.
- ADD THIS PC TO LIST: Adds the Windows computer name of the PC currently used for configuration to the list. Only applicable if the PC is not yet included in the list.

### Clients / Security (2) Technologies

Configuration			
Operation Settings	Line Interface		
Clients / Security <mark>Line Interface</mark> AoIP (LAN/SIP)	Line Mode: AoIP/SIP	~	
Audio Assignment	1: Provider 2 3	4 5	
- TTL / Relay	Label:	Provider	
- DHD Set Logic	SIP Server:	L AN - 172 20 225 102 v sin provider net	
⊞. Ember+		LAN : 172.20.223.102	
Transmission Modes	Proxy Server (for SIP Server):		
General 15 kHz Default	Backup SIP Server:	LAN : 172.20.225.102 V	
⊞. User Defined	Proxy Server (for Backup SIP Server):		
⊡. System Settings General	STUN:		
Audio Interface AES67	Transport:	UDP ~	
Data Interface LAN Interface	A-Law/µ-Law Signalling on incoming G.722 calls	s:	
VLAN NTP	Registration Timeout:	60 s	
- Alarm Signalling	PBX/Exchange line configuration		
SNMP ACconnect	Length of extension:	0 ~	
DHD	Outgoing line prefix:		
Ember+ PhonerSet	Skip outgoing line prefix on incoming calls:		
Login	International prefix:	00 (Default value: 00)	
	Enable Auto Redialling of remotely dropped con	inections	
	Redial attempts: 5 063 (0:infinite	attempts) Time between redialling: 3 sec (016)	
			OK Abbrechen Apple Now



- On the LINE INTERFACE setting page, the operating mode and the connection parameters are configured.
- LINE MODE: Sets the operating mode.
  - AoIP/SIP

Video

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

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

### Line Interface ( Technologies

Operation Settings       Line Interface         - Adir (LAN SP)       Line Mode:         - Mich Answer       Label:         - Tit, Relay       Label:         - Offer J       Sip Server:         - Grenal       Backup SIP Server:         - Grenal       Backup SIP Server:         - System Settings       - Grenal         - System Settings       - Grenal         - Adir (LAN)       - Grenal         - Grenal       Backup SIP Server:         - Grenal       Backup SIP Server:         - Grenal       - Grenal         - Adir (Interface       - Adiro Suppling on incoming G 722 cals:         - LAIN Interface       - Adaw/u-Law Signaling on incoming G 722 cals:         - MAP       - Registration Timeout:         - Mark       - Grenal         - Mark       - Backup SiP Setting:         - Outer DH       - Grenal         - Grenal       - Grenal         - Mark       - Grenal         - Mark       -	Configuration			×
Clents / Security       Line Mode:       Add P/SIP         Add Advigence       1: Provider       2       3       4         Advide Assignment       Labet:       Provider       aip.provider.net         Bit Provider       Silv Server:       Lahr: 172:20:225:102       aip.provider.net         Bit Provider       Provy Server (for SIP Server):       aip.provider.net         Silver Settings       STUN:       -         General       Provy Server (for Backup SIP Server):       -         Advide Interface       ALav/µLaw Signaling on incorning G.722 cals:       -         Advide Interface       ALav/µLaw Signaling on incorning G.722 cals:       -         Advide Interface       ALav/µLaw Signaling on incorning cals:       60 s         All Signaling       PSX/Exchange line configuration       -         Audio Interface       ALav/µLaw Signaling on incorning cals:       -         All Signaling       -       -       60 s         Audio Interface       -       -       -	📮 Operation Settings	Line Interface		
Audio Assignment       1: Provide       2       3       4       5         Audio Assignment       Labet       Provide       isp.provide met         TIL / Relay       Labet       Provide         DHD SetLogic       SIP Server:       LAN: 172 20 225 102       isp.provide met         Transmission Modes       Proxy Server (for SIP Server):       isp.provide met         General       Backup SIP Server:       LAN: 172 20 225 102       isp.provide met         System Settings:       STUN:       Image: Study SIP Server:       Image: Study SIP Server:         System Settings:       STUN:       Image: Study SIP Server:       Image: Study SIP Server:         - General       Study SIP Server:       Image: Study SIP Server:       Image: Study SIP Server:         - General       STUN:       Image: Study SIP Server:       Image: Study SIP Server:         - Addio Interface       - Adving: Law Signaling on incoming G.722 call:       -         - All Interface       - All Server:       Signaling on incoming G.722 call:       -         - All Server:       Image: Study Signaling on incoming G.722 call:       -       -         - All Server:       Image: Study Signaling on incoming G.722 call:       -       -         - All Server:       Image: Study Signaling on incoming G.722 c	Clients / Security <mark>Line Interface</mark> AoIP (LAN/SIP)	Line Mode: AoIP/SIP ~	×	
- Auto Answer       Label:       Provider         - TH, Palay       Label:       Provider         - DHD Set Logic       SIP Server:       LAN: 172.20.25.102        is provider net         - General       Backup SIP Server:       LAN: 172.20.25.102        is provider net         - Sit Ho Default       Backup SIP Server:       LAN: 172.20.25.102	- Audio Assignment	1: Provider 2 3	4 5	
OHD Set Logic       SIP Server:       LAN: 172 20 225.102       ip provider.net         Transmission Modes       Proxy Server (for SIP Server):       im provider.net         General       Backup SIP Server):       im provider.net         SIVEN Settings       STUN:       im provider.net         - General       STUN:       im provider.net         - Audio Interface       Transport:       UDP         - Audio Interface       Alaw: jp zerver):       im provider.net         - Audio Interface       Transport:       UDP         - Audio Interface       Alaw/µ-Law Signaling on incoming G.722 calls:       im provider.net         - VLAN       Registration Timeout:       im provider.net         - NTP       Registration Timeout:       im provider.net         - Alam Signalling       PEX/Exchange line configuration       im provider.net         - State Strate       Usagoing line prefix:       im prefix:         - Drio       Usagoing line prefix:       im prefix:       im prefix:         - Operative Value: D0       im prefix:       im prefix:       im prefix:         - Operative Value: D0       im prefix:       im prefix:       im prefix:       im prefix:         - Operative Value: Data       Site Declanding of remotely dropped connectionse <td< th=""><th> Auto Answer</th><th>Label<sup>,</sup></th><th>Provider</th><th></th></td<>	Auto Answer	Label <sup>,</sup>	Provider	
Transmission Modes       Proxy Server (for SIP Server):         - General       Backup SIP Server:         - User Defined       Proxy Server (for Backup SIP Server):         - General       Proxy Server (for Backup SIP Server):         - General       Proxy Server (for Backup SIP Server):         - General       STUN:         - General       STUN:         - Audio Interface       Transport:         - Audio Interface       ALaw/µ-Law Signaling on incoming G.722 calls:         - Alam Signalling       PBX/Exchange line configuration         - SIMP       Length of extension:         - DHD       Dugging line prefix         - User of Lagin       International prefix:         - User of Lagin       International prefix:         - Login       International prefix:         - Login       International prefix:		SIP Server:	LAN : 172.20.225.102 v sip.provider.net	
General       Backup SIP Server:       LAN: 172:20.225 102 ▼         System Settings       STUN:         - General       Transport:         - Audio Interface       Transport:         - At Sof7       -         - Data Interface       -         - LAN Interface       -         - VAN       Registration Timeout:         - NTP       PBX/Exchange line configuration         - SNMP       Length of extension:         - DHD       Outgoing line prefix         - DhD       Outgoing line prefix         - PhoneFset       Skip outgoing line prefix         - Uxick Dials       -         - Login       International prefix:         - Redial attempte:       5         - Case (0.16)       -	Transmission Modes	Proxy Server (for SIP Server):		
	General 15 kHz Default	Backup SIP Server:	LAN : 172.20.225.102 V	
System Settings STUN:   General Transport:   Adio Interface Transport:   AES67 ALaw/µ-Law Signalling on incoming G.722 calls:   LAI Interface A-Law/µ-Law Signalling on incoming G.722 calls:   VIAN Registration Timeout:   Alarm Signalling   SNMP   Adconnect   DHD   Outgoing line prefix:   Dub   Ember+   Skip outgoing line prefix:   Outgoing line prefix:   International prefix:   D0   [Default value: 00]	🕀 User Defined	Proxy Server (for Backup SIP Server):		
Audio Interface Transport:   Data Interface A.Law/µ-Law Signalling on incoming G.722 calls:   VLAN A.Law/µ-Law Signalling on incoming G.722 calls:   VLAN Registration Timeout:   Alarm Signalling   SNMP   Acconnect   DHD   Outgoing line prefix:   Quick Dials   Login   International prefix:   International prefix:   0   Enable Auto Redialling of remotely dropped connections:   Redial attempts:   5   0.63 (0.infinite attempts)   Time between redialling:   3	General	STUN:		
- Data Interface       - A.Law/µ-Law Signalling on incoming G.722 calls:         - VLAN       Registration Timeout:         - VLAN       Registration Timeout:         - Atarm Signalling       - BBX/Exchange line configuration         - SNIMP       Length of extension:         - ACconnect       - DHD         - DHD       Outgoing line prefix:         - Durber +       Skip outgoing line prefix:         - Quick Dials       - International prefix:         - Login       International prefix:         - Redial attempts:       5         - Call attempts:       5	Audio Interface AES67	Transport:	UDP v	
VLAN       Registration Timeout:       60 s         NTP       Alarm Signalling         Alarm Signalling       P8X/Exchange line configuration         SNMP       Length of extension:         O       Image: Connect         DHD       Outgoing line prefix         Ember +       Skip outgoing line prefix on incoming calls:         Quick Dials       International prefix:         Login       International prefix:         Redial attempts:       5         0.63 (0:infinite attempts)       Time between redialling:         3       sec (016)	Data Interface LAN Interface	A-Law/µ-Law Signalling on incoming G.722 calls:		
Alarm Signalling       PBX/Exchange line configuration         SNMP       Length of extension:         DHD       Outgoing line prefix:         Ember +       Skip outgoing line prefix on incoming calls:         Quick Dials       International prefix:         International prefix:       0         Redial attempts:       5         0.63 (0:infinite attempts)       Time between redialling:         3       sec (016)	VLAN NTP	Registration Timeout:	60 s	
SNMP   ACconnect   DHD   Ember +   PhonerSet   Quick Dials   International prefix:   International prefix:   0   [Enable Auto Redialling of remotely dropped connections   Redial attempts:   5   0.63 (0:infinite attempts)   Time between redialling:   3   sec (016)	Alarm Signalling	PBX/Exchange line configuration		
	SNMP ACconnect	Length of extension:	0 ~	
Ember+       Skip outgoing line prefix on incoming calls:         Quick Dials       International prefix:         International prefix:       00         Enable Auto Redialling of remotely dropped connections         Redial attempts:       5         0.63 (0:infinite attempts)       Time between redialling:         3       sec (016)	DHD	Outgoing line prefix:		
Login       International prefix:       00       (Default value: 00)         Enable Auto Redialling of remotely dropped connections       Redial attempts:       5       063 (0:infinite attempts)       Time between redialling:       3       sec (016)	Ember+ PhonerSet	Skip outgoing line prefix on incoming calls:		
Enable Auto Redialling of remotely dropped connections         Redial attempts:       5       063 (0:infinite attempts)       Time between redialling:       3       sec (016)	Login	International prefix:	00 (Default value: 00)	
Redial attempts:       5       063 (0:infinite attempts)       Time between redialling:       3       sec (016)		Enable Auto Redialling of remotely dropped conne	ections	
		Redial attempts: 5 063 (0:infinite at	ttempts) Time between redialling: 3 sec (016)	
OK Abbrechen Annlu Neu				OK Abbrechen Applu Now

#### Audio Video Technologies Line Interface – AoIP/SIP (1)

AV

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

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

### Line Interface AoIP/SIP **Fechnologies**

- STUN: If the provider requires the use of a STUN server, activate this option. The STUN server is entered on the configuration page LAN INTERFACE.
- TRANSPORT: Select whether the SIP telegrams are to be transmitted via UDP or TCP. The protocol is specified by the provider.
- A-LAW/µ-LAW SIGNALLING ON INCOMING G.722 CALLS: Activate this setting if audio is missing or dropouts occur when forwarding calls or receiving forwarded calls. This problem may occur if one of the endpoints is not HD Voice (G.722)-capable, but the PBX is not aware of this.
- REGISTRATION TIMEOUT: By default, the device renews the SIP registration every 60 seconds to check if the SIP server is still available. Increase the interval if the SIP server rejects the interval as too short.

- PBX/EXCHANGE LINE CONFIGURATION: The device can distinguish between internal and external calls when operating on a PBX.
  - LENGTH OF EXTENSION: Number of digits of internal call numbers.
  - OUTGOING LINE PREFIX: Digits inserted before the number when dialling an external number.
  - SKIP OUTGOING LINE PREFIX ON INCOMING CALLS: Some PBXs signal the number of the caller including the prefix digits, others do not. Set this option so that the phone numbers of incoming calls are displayed in the PC software without the prefix.
- INTERNATIONAL PREFIX: Prefix digits for dialling international telephone numbers. (Default: "00", do not set to "+"!)

### Video Technologies Line Interface AoIP/SIP (3)

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

### Video Technologies Line Interface AoIP/SIP (4)

onfiguration		
Operation Settings	Line Interface	
Clients / Security <mark>Line Interface</mark> Audio Assignment	Line Mode:	IP Leased Line Dial Up 🛛 🗸
Auto Answer TTL / Relay	Label:	AVT
⊞. Ember+ Transmission Modes	Audio UDP Port:	5004
in User Defined	Audio VLAN:	No VLAN ~
General	LAN Interface:	LAN : 172.20.225.102 💦 🗸
Audio Interface	Remote GPI0 Mode:	Off 🗸 🗸
AES67 Data Interface LAN Interface VLAN NTP Alarm Signalling SNMP ACconnect DHD Fmber+	Auto Connect after Power Up:	Enable
- PhonerSet	Decoder Parameters	
ی Quick Dials - Login	Buffer Mode:	Automatic V Buffer Size: 200 msec
	Encoder Packet Size (Payload T 6 711 6 722 PCM	ime)

#### Audio Video Technologies Line Interface – IP Leased Line Dial Up (1)

AV

- In the IP LEASED LINE DIAL UP mode, audio is transmitted directly between two devices.
- This mode is compatible with MAGIC ACip3 and Luci Live.
- A connection is established by entering the IP address of the other device in the dialling window.
- The remote device recognises the incoming audio data stream and adopts the source IP address and port for sending its own audio stream.
- LABEL: Enter any text. The label is displayed in the main panel of the PC software above the channel.
- AUDIO UDP PORT: Set the UDP port used for sending and receiving audio data.

Video

**Fechnoloaies** 

 AUDIO VLAN: Select which VLAN is used for the audio transmission. The VLAN must be defined beforehand on the VLAN configuration page.

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

## Line Interface – IP Leased Line Dial Up (2)

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

Video

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

### Line Interface – IP Leased Line Dial Up (3) Technologies

onfiguration		)
• Operation Settings	Line Interface	
Clients / Security <mark>Line Interface</mark> 	Line Mode: IP Leased Line Standard 🗸	
<ul> <li>Audio Assignment</li> <li>TTL / Relay</li> <li>DHD Set Logic</li> <li>Ember+</li> <li>System Settings</li> <li>General</li> <li>Audio Interface</li> <li>AES67</li> <li>Data Interface</li> <li>LAN Interface</li> <li>VLAN</li> <li>NTP</li> <li>Alarm Signalling</li> </ul>	Label: AVT   Remote IP Address: 172.20.40.120   Audio UDP Port: 5004   Audio VLAN: No VLAN   LAN Interface: LAN: 172.20.225.102   Remote GPI0 Mode: RTP (AVT Codecs only)   Auto Connect after Power Up: Enable	
SNMP ACconnect DHD Ember+ PhonerSet Quick Dials	Remote IP Address:     172.20.40.121       Audio VLAN:     No VLAN       LAN Interface:     LAN : 172.20.225.102       Secure Streaming Delay:     0 ms	
Login	Decoder Parameters	
	Encoder Packet Size (Payload Time) G.711, G.722, PCM: 10 msec	
	OK Abbrechen Apply Now	

A

#### Audio Video Technologies Line Interface – IP Leased Line Standard (1)

- In the IP LEASED LINE DIAL STANDARD mode. audio is transmitted directly between two devices.
- This mode is compatible with MAGIC AC1 XIP and MAGIC ACip3.
- A connection can be established automatically after power-up or using the Connect buttons.
- LABEL: Enter any text. The label is displayed in the main panel of the PC software above the channel.
- REMOTE IP ADDRESS: IP address of the remote device. The encoded data is sent to this address. Incoming data packets are only accepted from this address.
- AUDIO UDP PORT: UDP port used for sending and receiving audio data.
- AUDIO VLAN: Select which VLAN is used for the audio transmission. The VLAN must be defined beforehand on the VLAN configuration page.

Video

- LAN INTERFACE: Define which IP address of the device is to be used for the audio transmission.
- REMOTE GPIO MODE: When connected to a MAGIC AC1 XIP, MAGIC ACip3 or MAGIC AC1 Go, the states of the GPIO inputs can be transmitted to the remote device. Specify how the data is transmitted:
  - OFF: No transmission.
  - RTP (AVT CODECS ONLY): The states of the inputs are transmitted in the RTP data stream.
  - PAD (AVT CODECS ONLY): The states of the inputs are fed into the coding algorithm as PAD. This is only supported with MPEG.
- AUTO CONNECT AFTER POWER UP: Audio transmission starts automatically after the device has been powered up.

### Line Interface – IP Leased Line Standard (2) **Fechnologies**

- SECURE STREAMING: Sending two streams with identical content. The second stream can be delayed and routed via a different VLAN. Secure Streaming only works between AVT codecs.
  - ENABLE: Enable Secure Streaming in the transmit and receive direction.
  - REMOTE IP ADDRESS: IP address of the remote device for the second stream. The encoded data is sent to this address. Incoming data packets are only accepted from this address.
  - AUDIO VLAN: VLAN for the transmission of the second stream. The VLAN must be defined beforehand on the VLAN configuration page. If the audio transmission is not assigned to a VLAN, select NO VLAN.
  - LAN INTERFACE: Define which IP address of the device is to be used for the audio transmission.
  - SECURE STREAMING DELAY: Time delay for the second audio streams. This increases the overall delay of the transmission.

Video

- DECODER PARAMETERS: In the receiver, the data packets are buffered to compensate for transmission jitter (= temporal fluctuations in the reception of packets).
  - BUFFER MODE: The receive data buffer offers two modes.
    - AUTOMATIC: The device monitors the jitter of the received data packets and automatically adjusts the size of the receive data buffer to minimise delay while avoiding packet loss.
    - FIXED: The buffer size can be set to a fixed value under BUFFER SIZE. A larger buffer results in a higher audio delay.
  - BUFFER SIZE: Set a receive data buffer size between 20 ms and 2 s.
- ENCODER PACKET SIZE (PAYLOAD TIME): Some encoding algorithms do not mandate a specific size for the IP packets. For the following codecs, the packet size can be set here in milliseconds.
  - G.711, G.722, PCM
  - The device automatically reduces the packet size if the maximum packet size (MTU) of an Ethernet frame is exceeded.

### Line Interface – IP Leased Line Standard (3) Technologies

Line Mode: IP Leased Line Extended Compatibility Mode:   Audio Coding   Audio Coding   Audio Coding   Audio Stagiment   Bether +   System Settings   General   Audio Interface   Connect   Data Interface   Audio Delay Sync   Delay   Solute Diversion   Audio Delay Sync   Delay   Connect Diversion   Delay   Decoder   Curick Dials   Buffler Mode:   Autonic   Decoder Parameters   Decoder Parameters   Decoder Parameters   Decoder Parameters   Deta Muter Size (Payload Time)   Grine Size (Payload Time)   Grine Size (Payload Time)   Interval   200   Interval   200   Interval	
Audio Assignment       Encoder         TIL / Relay       Labet         OHD Set Logic       Labet         System Settings       N         Remole IP Addx       UDP Port         VLAN       Interface         System Settings       N         Remole IP Addx       UDP Port         VLAN       Interface         Action       1         Action Interface       2         Action Interface       3         Action Interface       5         Action Interface       1         Audio Delay Sync       Delay         Simmer       Audio Connect after Power Up:         Delay       Audio Connect after Power Up:         Delay       Interface         Buffer Mode:       200 msec         Encoder Packet Size (Payload Time)         G.711, G.722, PDM:       200 msec (50, 2000)	
Int / Netay       Label:       AVT         Bit Dist Logic       Internace       Secure Str. IP Add.       Secure Str. Interface         - General       1       172.20.40.12       -       1         - Audio Interface       2       5006       1       -       1         - Audio Interface       2       5006       1       -       1         - Audio Interface       2       5006       1       -       1         - Alam Signalling       3       5008       1       -       1         - Alam Signalling       -       Secure Streaming       Delay       0 ms       Remote GPIO Mode       RTP [AVT Codecs only]       ✓         - Alam Signalling       -       Audio Delay Sync       Delay       50 ms       -       1       -       1         - ShMP       -       Audio Delay Sync       Delay       50 ms       -       -       10 msc         - Login       -       Audio Connect after Power Up:       E Enable       -       -       -       -       10 msc         - Dielo       Auto Connect after Power Up:       -       -       10 msc       -       -       -       10 msc         - Dielo       Buffer Size: <td></td>	
Ember+       System Settings         General       172:20:40:120         Audio Interface       2         Attain Interface       1         Audio Data Interface       1         VLAN       Secure Streaming         Delay       Oms         Remote GPIO Mode       RTP (AVT Codecs only)         Audio Delay Sync       Delay         Some       Some         Audio Delay Sync       Delay         Decoder Parameters       Buffer Mode:         Buffer Size:       200 meec         Encoder Packet Size (Payload Time)         G.711, G.722, PCM:       10 meec         Interval:       200         UDP Port:       AC1:	
-System Settings - General - Audio Interface - At Sofor - Audio Interface - At Sofor - LAN Interface - At Sofor - LAN Interface - At Sofor - LAN Interface - LAN Inte	
- General       1       1/2/20/40/120       5004       -       1       1/2/20/40/121       -       1         - Audio Interface       2       5008       -       1       -       1       1         - Data Interface       4       5010       1       -       1       1       1         - LAN Interface       5       5012       1       -       1       1       1         - LAN Interface       5       5012       1       -       1       1       1         - VLAN       Secure Streaming       Delay       0 ms       Remote GPI0 Mode       RTP (AVT Codecs only)       ✓         - Alam Signalling       Secure Streaming       Delay       50 ms       50 ms           - SMMP       Audo Delay Sync       Delay       50 ms       Some            - DHD       Audo Connect after Power Up:       Image: Enable                                    <	
AES67         Data Interface         AAS1 Interface         VLAN         VLAN         NTP         Alarm Signalling         SIMP         Acconnect         DHD         Acto Connect after Power Up:         PhonerSet         Buffer Node:         Automatic         Quick Dials         Buffer Size:         Connect         Connect         PhonerSet         Buffer Size:         Connect         Buffer Size:         Connect         Decoder Packet Size (Payload Time)         G.711, G.722, PCM:         Interval:         UDP Port:         AC1:	
Data Interface       4       5010       1       1       1         LAN Interface       VLAN       5012       1       1       1         VLAN       Secure Streaming       Delay       0 ms       Remote GPI0 Mode       RTP (AVT Codecs only)         Atarm Signalling       Audo Connect       0 ms       Remote GPI0 Mode       RTP (AVT Codecs only)         Acconnect       DHD       Audo Connect after Power Up:       Enable         Decoder Parameters       Buffer Mode:       Automatic         PhonerSet       Buffer Mode:       200 msec         Encoder Packet Size (Payload Time)       10 msec       10 msec         G.711, G.722, PCM:       1       10 msec         UDP Port:       AC1:       4100       10 msec	
LAN Interface       5       5012       1       1         VLAN       VLAN       Image: Secure Streaming       Delay       0 ms       Remote GPI0 Mode       RTP (AVT Codecs only)       ✓         Alarm Signalling       Secure Streaming       Delay       0 ms       Remote GPI0 Mode       RTP (AVT Codecs only)       ✓         Alarm Signalling       Audo Delay Sync       Delay       0 ms       Remote GPI0 Mode       RTP (AVT Codecs only)       ✓         Alarm Signalling       Audo Delay Sync       Delay       10 ms       Secure Streaming	
VLAN         NTP         NTP         Alarm Signalling         SNIMP         ACconnect         DHD         Delay         Delay         Solution         DHD         Decoder Parameters         Differ Note:         Automatic         Outick Dials         Buffer Size:         Encoder Packet Size (Payload Time)         G.711, G.722, PCM:         Interval:         DOD         Interval:         Q00         meec (502000)         UDP Port	
Alarm Signalling       Image: Connect after Power Up:       Image: Connect after Power Up:       Image: Connect after Power Up:         Auto Connect after Power Up:       Image: Connect after Power Up:       Image: Connect after Power Up:       Image: Connect after Power Up:         PhonerSet       Decoder Parameters       Buffer Mode:       Image: Connect after Power Up:       Image: Connect after Power Up:         PhonerSet       Buffer Size:       Image: Connect after Power Up:       Image: Connect after Power Up:       Image: Connect after Power Up:         Image: Connect after Power Up:       Image: Connect after Power Up:       Image: Connect after Power Up:       Image: Connect after Power Up:         PhonerSet       Buffer Size:       Image: Connect after Power Up:       Image: Connect after Power Up:       Image: Connect after Power Up:         Image: Connect after Power Up:       Image: Connect after Power Up:       Image: Connect after Power Up:       Image: Connect after Power Up:         Image: Connect after Power Up:       Image: Connect after Power Up:       Image: Connect after Power Up:       Image: Connect after Power Up:         Image: Connect after Power Up:       Image: Connect after Power Up:       Image: Connect after Power Up:       Image: Connect after Power Up:         Image: Connect after Power Up:       Image: Connect after Power Up:       Image: Connect after Power Up:       Image: Connect after Power Up:	
SNMP ACconnect DHD Ember+ PhonerSet Quick Dials Login Level Info Transmission Intervat: QUP Port: Auto Connect after Power Up: Decoder Parameters Buffer Size: 200 msec 200 msec 10 msec	
ACconnect DHD Ember + PhonerSet Quick Dials Login Auto Connect after Power Up: Enable Decoder Parameters Buffer Mode: Automatic Quick Dials Buffer Size: 200 msec Encoder Packet Size (Payload Time) G.711, G.722, PCM: 10 msec Level Info Transmission Interval: 200 msec (50.2000) UDP Port: AC1: 4100	
DHD       Add Connect aller Power Op.       Decoder Parameters         PhonerSet       Decoder Parameters       Buffer Mode:         Quick Dials       Buffer Size:       200 msec         Encoder Packet Size (Payload Time)       6.711, 6.722, PCM:       10 msec         Level Info Transmission       Interval:       200         UDP Port:       AC1:       4100	
Decoder Parameters   PhonerSet   Quick Dials   Login     Buffer Mode:   Automatic   Buffer Size:     Comment   Comment   Buffer Size:     Comment   Comment   Comment     Buffer Size:     Comment   Buffer Size:     Comment   Comment   Buffer Size:     Comment   Comment   Buffer Size:     Comment   Comment   Buffer Size:   Comment   Comment   Comment   Buffer Size:     Comment   Comment <t< td=""><td></td></t<>	
Quick Dials       Buffer Mode:       Automatic         Buffer Size:       200 msec         Buffer Size:       10 msec         C.711, G.722, PCM:       10 msec         Level Info Transmission       10 msec (50.2000)         UDP Port:       AC1:	
Buffer Size:       200 msec         Encoder Packet Size (Payload Time)       0.711, G.722, PCM:         G.711, G.722, PCM:       10 msec         Level Info Transmission       10 msec (502000)         UDP Port:       AC1:	
Encoder Packet Size (Payload Time)         G.711, G.722, PCM:         Interval:         200         msec (502000)         UDP Port:       AC1:	
G.711, G.722, PCM:       10 msec         Level Info Transmission       10 msec         Interval:       200         UDP Port:       AC1:	
Level Info Transmission       Interval:     200       UDP Port:     AC1:	
Interval:         200         msec (502000)           UDP Port:         AC1:         4100	
UDP Port: AC1: 4100	

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#### Audio Video Technologies Line Interface – IP Leased Line Extended (1)

IP LEASED LINE EXTENDED: Allows sending of up to five streams via unidirectional point-to-point connections. All streams transport identical content (simulcast).

Reception of one independent stream.

- Compatible with MAGIC ACip3
- A connection can be established automatically after power-up or using the Connect buttons.
- ENCODER and DECODER are configured on separate tabs.
- COMPATIBILITY MODE: Enable connections to otherwise incompatible endpoints.
  - OFF: Use standard mode

Audio

Video

- AC1 XIP MODNET IP: Compatible with AC1 XIP Modnet devices in IP mode.
- RAW RTP/MPEG: Special application.

- ENCODER settings
  - LABEL: Enter any text. The label is displayed in the main panel of the PC software above the channel.
  - **REMOTE IP ADDRESS: IP address of the remote** device. The encoded data is sent to this address.
  - AUDIO UDP PORT: UDP port used for sending audio data.
  - AUDIO VLAN: Select which VLAN is used for the audio transmission. The VLAN must be defined beforehand on the VLAN configuration page.
  - LAN INTERFACE: Define which IP address of the device is to be used for the audio transmission.

### Line Interface – IP Leased Line Extended (2) Technologies

- SECURE STREAMING: Sending two streams with identical content. The second stream can be delayed and routed via a different VLAN. Secure Streaming only works between AVT codecs.
- SECURE STR. IP ADD.: IP address of the remote device for the second stream. The encoded data is sent to this address.
- SECURE STR. VLAN: VLAN for the transmission of the second stream. The VLAN must be defined beforehand on the VLAN configuration page. If the audio transmission is not assigned to a VLAN, select NO VLAN.
- SECURE STR. INTERFACE: Define which IP address of the device is to be used for the audio transmission.
- SECURE STREAMING: Enable Secure Streaming for transmission.

Video

DELAY: Time delay for the second audio streams. This • increases the overall delay of the transmission.

- AUDIO DELAY SYNC: This function makes it possible to output the audio data of this encoder at the decoder locations at synchronously and thus compensate for transmission delays of different lengths of the individual tracks. For this, the internal clocks of the encoders and decoders must be synchronised via NTP. Each transmitted data packet is time stamped, with the current time plus the set audio delay. The decoder waits until its current time matches the time stamp of a received packet and then decodes the content.
  - DELAY: Set this value to the maximum expected transmission delay so that even the most distant decoder receives the audio data stream in time and can play it out synchronously. You can check the fill level of the receive buffers at the decoders in the SYSTEM MONITOR under RX JITTER BUFFER.

### Line Interface – IP Leased Line Extended (3) Technologies
- REMOTE GPIO MODE: When connected to a MAGIC AC1 XIP, MAGIC ACip3 or MAGIC AC1 Go, the states of the GPIO inputs can be transmitted to the remote device. Specify how the data is transmitted:
  - OFF: No transmission.
  - RTP (AVT CODECS ONLY): The states of the inputs are transmitted in the RTP data stream.
  - PAD (AVT CODECS ONLY): The states of the inputs are fed into the coding algorithm as PAD. This is only supported with MPEG.

#### DECODER SETTINGS

Audio

Video

Technologies

Encod	Encoder Decoder						
Lal	bel: AVT						
Nr 1	Remote IP 172.20.	Addr 40.120	UDP Port 5004	VLAN	Secure Streaming IP Address 172.20.40.121	Secure Streaming VLAN	
⊠ Si □ Ai	ecure Stream udio Delay Sy	ing ync					

- LABEL: Must be configured on the ENCODER tab.
- REMOTE IP ADDRESS: IP address of the remote device. Incoming data packets are only accepted from this address.

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

### Line Interface – IP Leased Line Extended (4)

- AUTO CONNECT AFTER POWER UP: Audio transmission starts automatically after the device has been powered up.
- DECODER PARAMETERS: In the receiver, the data packets are buffered to compensate for transmission jitter (= temporal fluctuations in the reception of packets).
  - BUFFER MODE: The receive data buffer offers two modes.
    - AUTOMATIC: The device monitors the jitter of the received data packets and automatically adjusts the size of the receive data buffer to minimise delay while avoiding packet loss.
    - FIXED: The buffer size can be set to a fixed value under BUFFER SIZE. A larger buffer results in a higher audio delay.
  - BUFFER SIZE: Set a receive data buffer size between 20 ms and 2 s.

Video

- ENCODER PACKET SIZE (PAYLOAD TIME): Some encoding algorithms do not mandate a specific size for the IP packets. For the following codecs, the packet size can be set here in milliseconds.
  - G.711, G.722, PCM
  - The device automatically reduces the packet size if the maximum packet size (MTU) of an Ethernet frame is exceeded.
- LEVEL INFO TRANSMISSION: In COMPATIBILITY MODE: RAW RTP/MPEG, the audio level at the encoder input can be transmitted to the remote device.
  - INTERVAL: Set the time interval at which the level information is sent.
  - UDP PORT: Set the UDP port for sending the level information.

#### Line Interface – IP Leased Line Extended (5) Technologies

Operation Settings	AoIP (LAN/SIP)							
Clients / Security Line Interface	AC AUX							
<mark>AoIP (LAN/SIP)</mark> Audio Assignment	Label: AVT							
Auto Answer TTL / Relay	SIP Server	User Name	User Authentication	Password	Audio UDP Port	Display Name		
DHD Set Logic	1: Provider	430		*******	5004			
Transmission Modes	2: Local	250		~~~~~	5024			
General	1: Provider			******	5044			
15 kHz Default	1: Provider				5066			
System Settings	1: Provider				5086			
General								
Audio Interface								
- Data Interface								
LAN Interface								
VLAN								
- Alarm Signalling								
SNMP						Set Default Audio	Ports	
- ACconnect	- Encoder Pocket Size	(Pauload Time)						
Ember+	C 711 C 722 DC	(Fayloau fillie)		10				
PhonerSet	G.711, G.722, PC	M:		10 msec				
Quick Dials Login	RTP Disconnect	limeout:		10 sec				
	Decoder Parameters							
	Buffer Mode:	Automat	ic ,	✓ Buffer Si	ze:	2	200 msec	
	Registration delay bet	ween SIP lines	1	0 msec				

Audio Video Technologies AOIP (LAN/SIP) (1)

AVT

- On the AOIP (LAN/SIP) page, the SIP accounts are set if the operating mode is set to AoIP/SIP.
- The available channels are shown on tabs.
  - AC: Codec channel
  - AUX: Command channel
- LABEL: Enter any text. The label is displayed in the main window of the PC software above the channel.
- ACCOUNT TABLE: Up to five AoIP/SIP accounts can be assigned to each channel. Incoming calls are signalled from each account. If there is a connection on one account, the other accounts are reply with busy. For outgoing calls, an account must be selected when dialling.
- SIP SERVER: Select one of the predefined SIP servers. Configure the SIP servers on the LINE INTERFACE page.
- USER NAME: Enter the Username of the SIP account as specified by the provider.

Video

Technologies

- USER AUTHENTICATION: Enter the User Authentication of the SIP account as specified by the provider.
- PASSWORD: Enter the password for the SIP account as specified by the provider.
- AUDIO UDP PORT: Define the UDP port which the device uses to send the audio data.
- DISPLAY NAME: Enter any text. This text is displayed at the remote device. (Might be overwritten by the PBX or the provider.)
- SET DEFAULT AUDIO PORTS: Resets the local audio ports to the default values.

AoIP (LAN/SIP) (2)

- ENCODER PACKET SIZE (PAYLOAD TIME): Some encoding algorithms do not mandate a specific size for the IP packets. For the following codecs, the packet size can be set here in milliseconds.
  - G.711, G.722, PCM

Video

Technologies

- The device automatically reduces the packet size if the maximum packet size (MTU) of an Ethernet frame is exceeded.
- RTP DISCONNECT TIMEOUT: The time after which the device automatically terminates a connection if no audio data packets are received.

- DECODER PARAMETERS: In the receiver, the data packets are buffered to compensate for transmission jitter (= temporal fluctuations in the reception of packets).
  - BUFFER MODE: The receive data buffer offers two modes.
    - AUTOMATIC: The device monitors the jitter of the received data packets and automatically adjusts the size of the receive data buffer to minimise delay while avoiding packet loss.
    - FIXED: The buffer size can be set to a fixed value under BUFFER SIZE. A larger buffer results in a higher audio delay.
  - BUFFER SIZE: Set a receive data buffer size between 20 ms and 2 s.
- REGISTRATION DELAY BETWEEN SIP LINES: During start-up, the device simultaneously sends SIP registration telegrams for each AoIP/SIP account to the SIP server(s). If this is overwhelming a SIP server this setting can be used to introduce a delay between the registration attempts of the AoIP/SIP accounts.

AoIP (LAN/SIP) (3)

Operation Settings     Audio Coding       - Clents / Security     -       - Lune Interface     Audio Coding       - Audio Assignment     Coding Algorithm:       - Tit / Rahy     Mono (L+R)/2 v       - DHD Set Logic     Mono Mode:       - Bibmer+     -       - System Settings     -       - Audio Interface     Algorithm:       - Audio Interface     Algorithm:       - Audio Interface     Algorithm:       - Audio Interface     Sampling Rate:       - SMMP     Alarm Signalling       - SMMP       - Alconect       - DHO       - Cogin
- Line Interface       AC         - Audio Assignment       Coding Algorithm         - TTL / Relay       Mono Mode:         - DHD Set Logic       Mono Mode:         B: Ember+       Mono IL-RI/Z         - General       Mono IL-RI/Z         - Audio Interface       Algorithm:         - General       Mode:         - Action Interface       Algorithm:         - Action Interface       Mode:         - Action Interface       Sampling Rate:         - Attin Interface       Sampling Rate:         - NTP       Data Rate:         - Alarm Signalling         - ShuP         - MuP         - MuP         - MuP         - MuP         - MuP         - Signaling         - Signaling         - Signaling         - Signaling         - MuP         -



- On the AUDIO CODING page, the coding algorithms are set when the operating mode is set to IP LEASED LINE STANDARD / EXTENDED.
- CODING ALGORITHM: Select one of the installed codecs:
  - G.722, PCM, OPUS, FLAC, MPEG (some optional).
- MONO MODE: Select how the audio input signal is used when the algorithm is set to mono:
  - MONO LEFT: Only the left channel of the input signal is used.
  - MONO RIGHT: Only the right channel of the input signal is used.
  - MONO (L+R)/2: Left and right channels of the input signal are mixed.
- ANCILLARY DATA (PAD): Some codecs support transmission of data along with the encoded audio within the stream. If enabled, the device routes data provided via the DATA Interface (RS232) to the encoder. The data rate used for PAD reduces the data rate of the audio signal.

Video

Technologies

- ALGORITHM: If the MPEG codec is selected, you can choose between the available variants here (some optional):
  - L2: Layer 2 (MP2)
  - L3: Layer 3 (MP3)
  - AAC LD: Advanced Audio Coding Low Delay
  - AAC ELD: Advanced Audio Coding Enhanced Low Delay
  - AAC LC: Advanced Audio Coding Low Complexity
  - HE-AAC v1: High Efficiency Advanced Audio Coding with Spectral Band Replication (SBR)
  - HE-AAC v2: High Efficiency Advanced Audio Coding with Spectral Band Replication (SBR) and Parametric Stereo (PS)

Audio Coding (2)

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

Audio Coding

Audio

Video

<u>Technologies</u>

- BIT RESOLUTION: A higher bit depth allows for a higher dynamic range but requires a higher transmission bandwidth. (Not available for all codecs).
- BITRATE: If the data rate cannot be set directly but results from the available parameters, it is shown here for information.

Configuration	
<b>□</b> Operation Settings	Audio Assignment
<ul> <li>Operation Settings</li> <li>Clients / Security</li> <li>Line Interface</li> <li>AolP (LAN/SIP)</li> <li>Autio Assignment</li> <li>Auto Answer</li> <li>TTL / Relay</li> <li>DHD Set Logic</li> <li>Ember+</li> <li>Transmission Modes</li> <li>General</li> <li>15 kHz Default</li> <li>User Defined</li> <li>System Settings</li> <li>General</li> <li>Audio Interface</li> <li>AES67</li> <li>Data Interface</li> <li>ALN Interface</li> <li>VLAN</li> <li>NTP</li> <li>Alarm Signalling</li> <li>SNMP</li> <li>ACconnect</li> <li>DHD</li> <li>Ember+</li> <li>PhonerSet</li> <li>Quick Dials</li> <li>Login</li> </ul>	AC:       Analogue         Coopback. audio input to output             ALD:       AES671             Int + ALD:
	OK Abbrechen Apply Now

Audio Video Technologies Audio Assignment (1)

AV

- The audio interfaces of the device are assigned to the codec channel (AC) and the AUX channel on the AUDIO ASSIGNMENT page.
- AC: Select one of the available stereo audio interfaces.
- LOOPBACK AUDIO INPUT TO OUTPUT: Activate this option to loop the unencoded audio input signal internally back to the audio output.
- AUX: Select one of the available stereo audio interfaces for the AUX channel. Only the left channel of that audio interface is used.
- The following audio interfaces are available:
  - OFF: The channel is not used and is not displayed on the user interface.
  - AES: AES/EBU stereo interface XLR

Video

Technologies

- ANALOGUE: Analogue stereo interface XLR
- AES67 1: Audio over IP stereo interface 1 (optional)
- AES67 2: Audio over IP stereo interface 2 (optional)

 It is not possible to use the analogue audio interface and the AES/EBU interface at the same time, as they share the same connectors on the unit.

## Audio Assignment (2)

Configuration		×
📮 Operation Settings	Auto Answer	
Clients / Security     Clients / Security     Line Interface     AolP (LAN/SIP)     Autio Assignment     Auto Answer     TTL / Relay     DHD Set Logic     Ember +     Transmission Modes     General     15 kHz Default     User Defued	Channels AC: AUX: AUX: Auto Answer Delay: 2 sec	
System Settings  General  Audio Interface  AES67  Data Interface  LAN Interface  VLAN  NTP  Alarm Signalling  SNMP  Acconnect  DHD  Ember+ PhonerSet  Quick Dials		
<b>-</b>		
	OK Abbrechen Apply Now	



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



Configuration					×
Operation Settings     Clients / Security	TTL / Relay				
Line Interface	Pin	Dir.	Function 1 (Positive Edge)	Function 2 (Negative Edge)	
AoIP (LAN/SIP)	TTL 1: Pin 1	In	Connect:AC 1:15 kHz Call:+4991152710	Disconnect:AC 1	1
- Audio Assignment	TTL 2: Pin 2	In	Remote Input		
Auto Answer	TTL 3: Pin 3	Out	Connection State:AC 1::Connect		
DHD Set Logic	TTL 4: Pin 4	Out	TTL Remote GPI0:TTL Pin 1:AC 1		
Ember+	Relay 1: Pins 6/7	Out	Always Open		
Transmission Modes	Relay 2: Pins 8/9	Out	Always Open	-	
General					
15 kHz Default					
User Defined					
System Settings					
General					
- Audio Interface					
Data Interface					
I AN Interface					
NTP					
- Alarm Signalling					
SNMP					
ACconnect					
DHD					
Ember+					
PhonerSet					
Login					
: Login					
				OK Abbrechen Apply N	ow



AVT

- GPIOs (General Purpose Input Output) offer a way of exchanging information with other systems.
- The functions of each GPIO can be configured on the TTL / RELAY, DHD SET LOGIC and EMBER+ configuration pages.
- The following GPIO variants are available:
- TTL: 4 hardware pins via the TTL/Relay connector
  - Each TTL can be operated as input or output.
  - Each TTL pin is internally connected to a pull-up resistor which means the default state is "high".
  - Connect the respective pin to GND (Pin 5 of the TTL/Relay connector) to generate a "low" input signal.
- RELAY: 6 relays that can only be used as outputs.

- DHD SET LOGIC: Uses an IP-based protocol to transmit control signals and status messages via Ethernet. There are 64 GPIOs available, each of which can be used as input or output. Must be enabled and configured under System Settings
   > DHD first.
- EMBER+: Uses an IP-based protocol to transmit control signals and status messages via Ethernet. An end device can act as a provider or consumer in a connection.
  - INPUT: 64 inputs are available. (as Ember+ provider)
  - OUTPUT: 64 outputs are available. (as Ember+ Provider)
  - CONSUMER FUNCTIONS: Allows the transmission of text information via the Ember+ protocol. 10 functions are available per consumer. (as Ember+ Consumer)





# MAGIC AC1 Go

#### Configuration

Transmission Modes

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Configuration			×
Configuration	General         Audio Mode after Disconnect:         Audio Mode after Power Up:         Default Mono Mode (Telephone, 7 kHz, 15 kHz)	AC 15 kHz 15 kHz Mono (L+R)/2	
		OK Abbrechen Apply Now	

#### Audio Video Technologies Transmission Modes - General (1)

AVI

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

Video

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

#### Transmission Modes - General (2) Technologies

- The TRANSMISSION MODES GENERAL page provides basic settings.
- AUDIO MODE AFTER DISCONNECT: Select which transmission mode is set when a connection is terminated. This mode is preset the next time a call is set up. The mode can then be changed.
- AUDIO MODE AFTER DISCONNECT: Select which transmission mode is set when a connection is terminated. This mode is preset the next time a call is set up. The mode can then be changed.
- AUDIO MODE AFTER POWER UP: Select which transmission mode is set when the device is started. This mode is preset the first time a call is set up. The mode can then be changed.

Video

Technologies

- DEFAULT MONO MODE (TELEPHONE, 7 KHZ, 15 KHZ): Select how the audio input signal is used when a standard transmission mode uses a mono signal:
  - MONO LEFT: Only the left channel of the input signal is used.
  - MONO RIGHT: Only the right channel of the input signal is used.
  - MONO (L+R)/2: Left and right channel of the input signal are mixed.

## Transmission Modes - General (3)

Operation Settings       15 kHz Default         Incentrafice       AC         Andro Assignment       Coding Algorithm:         MEEG       Anclinuy Data (PAD)         Transmission Modes       Anclinuy Data (PAD)         Transmission Modes       Angorithm:         General       Algorithm:         Buffer Default       Mode:         General       Algorithm:         Buffer Default       Mode:         General       Data Rate:         Andro Interface       Assorithm         Addio Interface       Renote DPID Mode:         Addio Interface       Renote DPID Mode:         Addio Interface       Delay:         Addio Delay:       O ms         PhonerSet       Custor Sineaming         Addio Interface       Delay:	onfiguration		
Clerkty / Security       AC         - Aufu / Axigisment       Coding Algorithm:         - Aufu / Axigisment       Coding Algorithm:         - Tit / Reby       - Onclarg Data (PAD)         - State Defined       Mode:       - Onclarg Data (PAD)         - System Setting:       Sampling Rate:       4842       -         - General       Data Rate:       - Osta Bate:       - Osta Bate:         - Auto Interface       -       -       -         - Auto Interface       -       Osta Fate:       -         - Auto Interface       Delay:       Om       Om         - Allow Signaling       Mode:       Om       -         - Much       Delay:       Oms       -         - Onceste       -       Oms       -         - Onceste       -       - </th <th>• Operation Settings</th> <th>15 kHz Default</th> <th></th>	• Operation Settings	15 kHz Default	
Tansion Modes       Algorithm:       MPEG Layer 2         Is Hez Default       Mode:       Joint Stereo         Oscreal       Sampling Rele:       43 kHz         Audio Interface       192 kBH/s          - Audio Interface       -       -         - Atain Interface       -       -         - UAN Interface       -       -         - VLAN       Mode:       Off         - Number of the standard of t	… Clients / Security … Line Interface … AoIP (LAN/SIP) … Audio Assignment … Auto Answer … TTL / Relay … DHD Set Logic 관 - Ember+	AC Coding Algorithm: MPEG ~ Ancillary Data (PAD)	
Butter Defined Index     System Settings Sampling Rate:     General Data Rate:     Action     Action </td <td>- Transmission Modes  General <mark>15 kHz Default</mark></td> <td>Algorithm: MPEG Layer 2 ~</td> <td></td>	- Transmission Modes General <mark>15 kHz Default</mark>	Algorithm: MPEG Layer 2 ~	
General       Data Rate:       192 kBit/s         Addfo Interface       -         AA Interface       Remote GPID Mode:         VLAN       Off         VLAN       Secure Streaming:         - Alarm Signalling       Mode:         SNIMP       Oms         - DHD       Delay:         - DHD       Oms         - Ember +       -         - PhonerSet       -         - Login       -		Sampling Rate: 48 kHz ~	
LAN Interface       Remote GPI0 Mode:       Dff         VLAN       NTP       Secure Streaming:         Alarm Signalling       Mode:       Dff         SNMP       Acconnect       Delay:         DHD       Ember +         PhonerSet       Quick Dials         - Login       Vertice	General Audio Interface AES67	Data Rate: 192 kBit/s ~	
NTP Secure Streaming:   Alarm Signalling Mode:   SNMP Mode:   Acconnect Delay:   DHD   Ember+   PhonerSet   Quick Dials	LAN Interface VLAN	Remote GPI0 Mode: Off ~	
ACconnect Delay: Oms DHD Ember+ Quick Dials Login	NTP Alarm Signalling SNMP	Secure Streaming: Mode: Off ~	
		Delay: Oms	

#### Audio Video Technologies Transmission Modes – 15 kHz Default (1)

AV

- On the TRANSMISSION MODE 15 KHZ DEFAULT page, the coding algorithms for the predefined 15 kHz mode are set.
- CODING ALGORITHM: The following codecs are available:
  - G.722
  - MPEG
  - PCM
  - OPUS
  - FLAC

Video

Technologies

 ANCILLARY DATA (PAD): Some codecs support transmission of data along with the encoded audio within the stream. If enabled, the device routes data provided via the DATA Interface (RS232) to the encoder. The data rate used for PAD reduces the data rate of the audio signal.

- ALGORITHM: Select an algorithm if the MPEG codec is selected above. (Availability depends on the activated software options.)
  - L2: Layer 2 (MP2)
  - L3: Layer 3 (MP3)
  - AAC LD: Advanced Audio Coding Low Delay
  - AAC ELD: Advanced Audio Coding Enhanced Low Delay
  - AAC LC: Advanced Audio Coding Low Complexity
  - HE-AAC v1: High Efficiency Advanced Audio Coding with Spectral Band Replication (SBR)
  - HE-AAC v2: High Efficiency Advanced Audio Coding with Spectral Band Replication (SBR) and Parametric Stereo (PS)

### Transmission Modes – 15 kHz Default (2)

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

Video

Technologies

- BIT RESOLUTION: A higher bit depth allows for a higher dynamic range but requires a higher transmission bandwidth. (Not available for all codecs).
- BITRATE: If the data rate cannot be set directly but results from the available parameters, it is shown here for information.
- REMOTE GPIO MODE: When connected to a MAGIC AC1 XIP, MAGIC ACip3 or MAGIC AC1 Go, the states of the GPIO inputs can be transmitted to the remote device. Specify how the data is transmitted:
  - OFF: No transmission.
  - RTP (AVT CODECS ONLY): The states of the inputs are transmitted in the RTP data stream.
  - PAD (AVT CODECS ONLY): The states of the inputs are fed into the coding algorithm as PAD. This is only supported with MPEG.

### Transmission Modes – 15 kHz Default (3)

- SECURE STREAMING: Sending two streams with identical content. Secure Streaming only works between AVT codecs.
  - MODE:

Video

Technologies

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

### Transmission Modes – 15 kHz Default (4)

Configuration		×
Configuration  Operation Settings  Clients / Security Clients Clients / Security Clients Clien	User Transmission Modes         Show       Name         AAC-LD 128         PCM24         Delete         Rename	×
AAC-LD 128 PCM24 Constraints General Audio Interface AES67 Data Interface LAN Interface VLAN NTP Alarm Signalling SNMP ACconnect DHD Ember + PhonerSet Quick Dials		
	OK Abbrechen Apply Now	

#### Audio Video Technologies Transmission Modes – User Defined (1)

AV

- On the TRANSMISSION MODE USER DEFINED page, custom transmission modes can be defined.
- USER TRANSMISSION MODES: The user-defined transmission modes are listed here.
  - SHOW: Enabled transmission modes are available for selection when establishing a connection.
  - NAME: Name of the transmission mode.
- ADD: Adds a new transmission mode.

Video

Technologies

- EDIT: Opens the selected transmission mode for editing. A transmission mode can also be selected for editing in the tree on the left.
- DELETE: Deletes the selected transmission mode.
- RENAME: Opens a window for editing the name of the selected transmission mode.

### Transmission Modes – User Defined (2)

lients / Security ne Interface olP (LAN/SIP) udio Assignment Coding Algorithm: MPEG ✓ uto Answer IL / Relay HD Set Logic □ Ancillary Data (PAD) mber + mission Modes Algorithm: AAC LD ✓ eneral Algorithm: Stereo ✓ ser Defined Sampling Rate: 48 kHz ✓ mAAC-LD 128 ↓ m PCM24 Data Rate: 128 kBit/s ✓	
mission Modes eneral Algorithm: AAC LD ~ 5 kHz Default Mode: Stereo ~ AAC-LD 128 Sampling Rate: 48 kHz ~ AAC-LD 128 Data Rate: 128 kBit/s ~	
m Settings	
eneral udio Interface ES67 Remote GPID Mode: Off ✓ ata Interface	
AN Interface LAN Secure Streaming Mode: On TP Iarm Signalling Secure Streaming Delay: Oms NMP	
Cconnect HD nber+ honerSet uick Dials	

#### Audio Video Technologies User Defined Transmission Mode (1)

AV

- On each USER DEFINED TRANSMISSION MODE page, the parameters of a transmission mode can be set.
- CODING ALGORITHM: The following codecs are available:
  - G.722
  - MPEG
  - PCM
  - OPUS
  - FLAC

Video

Technologies

 ANCILLARY DATA (PAD): Some codecs support transmission of data along with the encoded audio within the stream. If enabled, the device routes data provided via the DATA Interface (RS232) to the encoder. The data rate used for PAD reduces the data rate of the audio signal.

- ALGORITHM: Select an algorithm if the MPEG codec is selected above. (Availability depends on the activated software options.)
  - L2: Layer 2 (MP2)
  - L3: Layer 3 (MP3)
  - AAC LD: Advanced Audio Coding Low Delay
  - AAC ELD: Advanced Audio Coding Enhanced Low Delay
  - AAC LC: Advanced Audio Coding Low Complexity
  - HE-AAC v1: High Efficiency Advanced Audio Coding with Spectral Band Replication (SBR)
  - HE-AAC v2: High Efficiency Advanced Audio Coding with Spectral Band Replication (SBR) and Parametric Stereo (PS)

### User Defined Transmission Mode (2)

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

Video

**Fechnoloaies** 

- BIT RESOLUTION: A higher bit depth allows for a higher dynamic range but requires a higher transmission bandwidth. (Not available for all codecs).
- BITRATE: If the data rate cannot be set directly but results from the available parameters, it is shown here for information.
- REMOTE GPIO MODE: When connected to a MAGIC AC1 XIP, MAGIC ACip3 or MAGIC AC1 Go, the states of the GPIO inputs can be transmitted to the remote device. Specify how the data is transmitted:
  - OFF: No transmission.
  - RTP (AVT CODECS ONLY): The states of the inputs are transmitted in the RTP data stream.
  - PAD (AVT CODECS ONLY): The states of the inputs are fed into the coding algorithm as PAD. This is only supported with MPEG.

### User Defined Transmission Mode (3)

- SECURE STREAMING: Sending two streams with identical content. Secure Streaming only works between AVT codecs.
  - MODE:

Audio Video

Technologies

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

### User Defined Transmission Mode (4)



# MAGIC AC1 Go

#### Configuration

System Settings

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Operation Settings	General	
<ul> <li>Operation Settings</li> <li>Clients / Security</li> <li>Line Interface</li> <li>AoIP (LAN/SIP)</li> <li>Auto Answer</li> <li>TTL / Relay</li> <li>DHD Set Logic</li> <li>Ember+</li> <li>Transmission Modes</li> <li>General</li> <li>15 kHz Default</li> <li>User Defined</li> <li>System Settings</li> <li>General</li> <li>Audio Interface</li> <li>ALS67</li> <li>Data Interface</li> <li>VLAN</li> <li>NTP</li> <li>Alarm Signalling</li> <li>SNIMP</li> <li>ACconnect</li> <li>DHD</li> <li>Ember+</li> <li>PhonerSet</li> <li>Quick Dials</li> <li>Login</li> </ul>	General         Display         Backlight:         On         Contrast:         Image:         Image:         English         Image:	



- Settings in the SYSTEM SETTINGS branch cannot be saved in presets. These settings concern the hardware or the connection to the direct environment. Flexible switching via presets seems therefore less useful.
- DISPLAY: The front display of the device is configured here.
  - BACKLIGHT: The backlight of the front display can be operated in two modes:
    - AUTO: The backlight is switched on when a key is pressed on the unit. After a few seconds, the backlight switches off again.

General (2)

- ON: The backlight is permanently on.
- CONTRAST: Set the value for the contrast of the front display so that the display can be read best.
- LANGUAGE: The front display of the device supports two languages.
  - ENGLISH
  - GERMAN

Audio

Video

Technologies

- FRONT KEYPAD: The front panel keyboard is configured here.
  - KEY TONE: Activates the key click of the unit keyboard.
- OPTIONS
  - NEED CONFIRMATION ON MANUAL CONNECTION DROP: If enabled, the user must confirm a warning message when he wants to terminate a connection via the user interface.
  - NEED USER PASSWORD ON MANUAL CONNECTION DROP: If enabled, the user must enter the user password when he wants to terminate a connection via the user interface. The user password is set up on the LOGIN configuration page.
- SYSTEM NAME: Enter any text. The system name is displayed in the user interface or in the MAGIC System Manager.

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



Configuration	
<b>□</b> . Operation Settings	Audio Interface
<ul> <li>Clients / Security</li> <li>Line Interface</li> <li>AolP (LAN/SIP)</li> <li>Autio Assignment</li> <li>Auto Answer</li> <li>TTL / Relay</li> <li>DHD Set Logic</li> <li>Ember +</li> <li>Transmission Modes</li> <li>General</li> <li>T5 kHz Default</li> <li>User Defined</li> <li>System Settings</li> <li>General</li> <li>General</li> <li>General</li> <li>System Settings</li> <li>General</li> <li>Madio Interface</li> </ul>	Main Nominal Level of Analogue Audio         Level In Left:       6 dBu         Level In Right:       6 dBu         Level Out Right:       6 dBu         Level Out Right:       6 dBu         Headroom       9 dBr
AES67 Data Interface LAN Interface VLAN NTP Alarm Signalling SNIMP ACconnect DHD Ember+ Biber+	AES/EBU Interface Clock Source of Digital Output: Internal
Quick Dials	Default Settings
	OK Abbrechen Apply Now

#### Audio Video Technologies Audio Interface (1)

AVI

- The basic parameters of the audio interfaces of the device are configured on the AUDIO INTERFACE page.
- MAIN NOMINAL LEVEL OF ANALOGUE AUDIO:
  - LEVEL IN LEFT / RIGHT: Set the sensitivity of the audio inputs. Decreasing the value increases the audio level and vice versa. (Default setting: 6 dBu)
  - LEVEL OUT LEFT / RIGHT: Adjust the gain of the audio outputs. Decreasing the value lowers the audio level and vice versa. (Default setting: 6 dBu)

Audio Interface (2)

 HEADROOM: This value lowers the audio level of analogue input signals to create headroom for internal audio processing. Otherwise, the audio signal may clip. The analogue output signal is amplified by the headroom.

The headroom is fixed at 20 dBr.

Audio

Video

Technologies

- AES/EBU INTERFACE
  - CLOCK SOURCE OF DIGITAL OUTPUT: There are sample rate converters only on the AES/EBU input. The AES/EBU output runs at system clock. Systems connected to the device via AES/EBU may require sample rate converters on their inputs.
  - This setting determines the clock to which the system's audio clock is synchronised. Some of the options may not be available depending on the line mode or Audio over IP settings.
    - INTERNAL: The digital output is synchronised to the system's internal clock.
    - RECOVERED: The digital output is synchronised to the clock applied to the AES/EBU input.
- DEFAULT SETTINGS: This button resets all settings on this page to the factory defaults.

Pertation Setting:       AIS67         □ Clents / Security       ☑ Activate AES67 streaming         □ Audio Assignment       ☑ Activate AES67 streaming         □ Audio Assignment       0       0.127         □ Audio Assignment       0       0.127         □ TL / Relay       □ Disable SAP       0       0.127         □ OHD Set logic       □ Disable SAP       0       0.137         □ Her	Configuration					
- Clerks / Security       ✓ Activate AES67 steeming         - Line Interface       ✓ AniP (LAN/SP)         - Aali (LAN/SP)       LAN Interface:         - Auto Answer       PTP Domain:         - Muto Assignment       Ø 0.127         - Turk Nelay       Disable SAP         Oth Set Logic       Dually of Service (DSCP)         - The Interface       Gale (LAN/SP)         - Stable Default       PTP:         - General       RTP:         - Stable Default       Transmission         - General       RTP:         - Stable Default       Transmission         - General       RTP:         - Stable Default       Transmission         - General       RTP:         - Audio Interface       Calon         - Audio Interface       Audo Mode:         - Altherace       Steam T:         - Altherace       Audo Mode:         - Altherace       Audo Mode:         - Altherace       Audo Mode:         - Altherace       Audo Mode:         - Althi Interface       Audo Mode:	- Operation Settings	AES67				
SNMP       Reception:       Import SDP File         ACconnect       Stream 1:       TF-DHD-Core: 2; 4 channels       Import SDP File         DHD       Stream 2:       -       Import SDP File         Ember +       PhonerSet       Audio Interface:       Output Channels         Uogin       AES67 1:       1, 2       TF-DHD-Core: 2; Ch. 1       TF-DHD-Core: 2; Ch. 2         Login       AES67 2:       3, 4       TF-DHD-Core: 2; Ch. 3       TF-DHD-Core: 2; Ch. 4	<ul> <li>Clients / Security</li> <li>Line Interface</li> <li>AolP (LAN/SIP)</li> <li>Audio Assignment</li> <li>Auto Answer</li> <li>TTL / Relay</li> <li>DHD Set Logic</li> <li>Ember+</li> <li>Transmission Modes</li> <li>General</li> <li>15 kHz Default</li> <li>User Defined</li> <li>System Settings</li> <li>General</li> <li>Audio Interface</li> <li>Audio Interface</li> <li>LAN Interface</li> <li>VLAN</li> <li>NTP</li> <li>Alarm Signalling</li> </ul>	Activate AES67 streaming LAN Interface: PTP Domain: Disable SAP Quality of Service (DSCP): PTP: RTP: Transmission: Channels: SAP Stream Name: RTP UDP Port: Audio Mode: Sampling Rate: Address Mode:	LAN : 172.20.225.102 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0	<ul> <li>I27</li> <li>(063) DiffServ: 184dec</li> <li>(063) DiffServ: 136dec</li> <li>IP Address: 239 1 225</li> </ul>	102	Set Default QoS values Export SDP File
Accornication     Stream 2:     Import SDP File       DHD     Stream 2:     Import SDP File       Ember +     Quick Dials     Output Channels       Quick Dials     AES67 1:     1, 2       AES67 2:     3, 4     TF-DHD-Core : 2; Ch. 3	SNMP	Reception:	Reception:			
Ember +     Stream 2:     Import SDP File       PhonerSet     Audio Interface:     Output Channels       Quick Dials     AES67 1:     1,2       AES67 2:     3,4     TF-DHD-Core : 2; Ch. 3	- DHD	Stream 1:	TF-DHD-Core : 2; 4 chan	nels	~	Import SDP File
PhonerSet     Audio Interface:     Output Channels     Input Channels       Quick Dials     AES67 1:     1, 2     TF-DHD-Core : 2; Ch. 1     TF-DHD-Core : 2; Ch. 2       AES67 2:     3, 4     TF-DHD-Core : 2; Ch. 3     TF-DHD-Core : 2; Ch. 4	Ember+	Stream 2:	-		~	Import SDP File
Login         AES67 1:         1, 2         TF-DHD-Core : 2; Ch. 1         TF-DHD-Core : 2; Ch. 2            AES67 2:         3, 4         TF-DHD-Core : 2; Ch. 3         TF-DHD-Core : 2; Ch. 4	PhonerSet Ouick Dials	Audio Interface:	Output Channels	Input C	hannels	
AES67 2: 3, 4 TF-DHD-Core : 2; Ch. 3 V TF-DHD-Core : 2; Ch. 4 V	Login	AES671:	1, 2	TF-DHD-Core : 2; Ch. 1 🗸 🗸	TF-DHD-Core : 2; Ch. 2 $\qquad \checkmark$	
		AES67 2:	3, 4	TF-DHD-Core : 2; Ch. 3	TF-DHD-Core : 2; Ch. 4 🗸	



AVT

- AES67 is an open standard for Audio over IP developed by the Audio Engineering Society (AES). It sets a minimum standard for audio streaming and synchronisation in Audio over IP networks such as Dante, RAVENNA, Livewire+ and others.
- The optional AES67 functionality of the device is configured on the AES67 page.
- LAN INTERFACE: Select the IP address of the unit to be used for AES67.
- CHANNELS: Set the number of AES67 mono audio channels.
- TRANSMISSION: This section defines the AES67 stream that the unit sends.
  - SAP STREAM NAME: Enter any text. The stream is announced on the network using that name.

AES67

 RTP UDP PORT: Set the local UDP port for the AES67 stream. (Default: 5300)

Audio

Video

Technologies

- AUDIO MODE: The audio signal is PCM encoded. The bit depths can be set here:
  - L16: 16 bit/sample
  - L24:24 bit/sample (Recommended for connections to Dante devices.)
- SAMPLING RATE: Set the sampling frequency of the audio signal.
  - 32 kHz
  - 48 kHz
- ADDRESS MODE: AES67 streams are multicast on the network. There are two modes for entering the multicast IP address:
  - MANUAL: You can freely enter the complete IP address.
  - AUTO: The IP address is derived from the IP address of the LAN interface of the device. Only the second byte of the IP address must be entered manually.
  - IP ADDRESS: Enter the entire IP address or the second byte of the IP address, depending on the ADDRESS MODE set.
- RECEPTION: The unit can subscribe to two AES67 streams with up to 8 mono audio channels each. The desired audio channels of the streams are then selected under AUDIO INTERFACE. The unit receives the stream descriptions via SAP of all streams announced in the network and lists them.
  - STREAM 1/2: Select one of the listed streams to subscribe to it.
  - UPDATE RX STREAMS: The list of detected streams is rebuilt.
- OUTPUT CHANNELS: The channels of the transmitted stream are permanently assigned to AES67 audio interfaces.
  - AES67 1: Channels 1 and 2 of the transmitted AES76 stream are assigned to the audio interface AES67 1.
  - AES67 2: Channels 3 and 4 of the transmitted AES76 stream are assigned to the audio interface AES67 2.

AES67

 INPUT CHANNELS: Audio channels of received streams can be freely assigned to the AES67 audio interfaces.

Audio

Video

Technologies

- QUALITY OF SERVICE (DSCP): The unit supports Quality of Service via DSCP (Differentiated Service Code Point). It uses the Differentiated Services (DiffServ) field in the IP header for this purpose. QoS must be activated in the network. The DiffServ values of the individual priority levels are defined and evaluated in the network.
  - PTP: Enter the DiffServ value for clock distribution via PTP here.
  - RTP: Enter the DiffServ value for AES67 audio transmission via RTP here.

Configuration		
Configuration    Operation Settings   Clients / Security  Line Interface AoIP (LAN/SIP)  Autio Assignment Auto Answer TTL / Relay DHD Set Logic  Ember+  Transmission Modes General General System Settings General Audio Interface AES67 Data Interface AIS67 Data Interface AISMP Alarm Signalling SNMP ACconnect DHD Ember+ PhonerSet Quick Dials Login	Data Interface         Bauchate:         Paity:         None	
		low



AV

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

#### Video Technologies Data Interface (2)

Operation Settings	LAN Interface	
Clients / Security		
Line Interface	LAN 1	
AoIP (LAN/SIP)	Primary IP Address	Second IP Address Third IP Address
Audio Assignment	DHCP:	
TTL / Relav	IP Address:	172.20.225.102
	Sub Net Mask:	255 255 0.0
⊕ Ember+	Sub Net Mask.	
Transmission Modes	Default Gateway:	172.20.1.1
General	DNS Server:	172.20.1.1
- 15 kHz Default		
⊕. User Defined		
System Settings General	Quality of Service (DiffServ	J
- Audio Interface	Audio:	46 (EF) V (0.63) DiffServ: 184dec
AES67		
Data Interface	SIP:	Zb (AF 31) V (Ub3) DiffSerV: 104dec
LAN Interface	- STUN Server Parameters-	
VLAN		
NTP	STUN Server:	stun.provider.net
SNMD	NAT Keep Alive Messag	ae Time:
- DHD	Link Type:	Auto
Ember+		
PhonerSet	Control	
Quick Dials	LIDE Control Port:	10000 Cat Data Mart
Login	ODT CONTON	Set Deladit Polt
	Accessible from:	All LAN Interfaces (not recommended)



AV

- The LAN interfaces of the device are configured on the LAN INTERFACE page.
- DHCP: Enable DHCP if the unit must obtain its IP address from a DHCP server. The obtained IP address is displayed on the front display of the unit by navigating to the home screen and then pressing the HANG UP key.
- A fixed address can be set using the following fields:
  - IP ADDRESS

Video

Technologies

- SUBNET MASK
- DEFAULT GATEWAY
- DNS SERVER: Only available for the primary IP address. The DNS Server is only required to resolve the host name of SIP servers. If a SIP server is specified via the host name, the primary IP address must be used for AoIP/SIP.

LAN Interface (2)

- QUALITY OF SERVICE (DSCP): The device supports Quality of Service via DSCP (Differentiated Service Code Point). It uses the Differentiated Services (DiffServ) field in the IP header. QoS must be activated in the network. The DiffServ values of the individual priority levels are defined and evaluated in the network.
  - AUDIO: DiffServ value for audio transmission via AoIP/SIP or IP Leased Line.
  - SIP: DiffServ value for the AoIP/SIP connection setup.

- STUN SERVER PARAMETERS: The use of a STUN server may be required by the AoIP/SIP provider. Only one STUN server is required, even if the device connects to multiple providers. Enable STUN for each SIP server individually on the LINE INTERFACE configuration page in AoIP/SIP mode.
  - STUN SERVER: IP address or host name of the STUN server.
  - NAT KEEP ALIVE MESSAGE TIME: Set the time interval at which the device periodically sends keep-alive packets to the SIP server. This allows routers and firewalls to keep the SIP communication ports open. This allows the SIP server to inform the unit of incoming calls. Keep alive packets are only sent if STUN is enabled for the respective SIP account. (Default: 20 seconds)

Video

Technologies

- LINK TYPE: Set how the unit's Ethernet connection to the network switch is established.
  - AUTO: The device automatically detects the link type. The System Monitor should display the status of the LAN interface at 100 Mbps, full.
  - TYPE 1/2: If the status of the LAN interface in the System Monitor is not 100 Mbps, full, even though the switch supports this mode, test whether Type 1 or Type 2 solves the problem.
  - DISABLE INSUFFICIENT LAN ALARM: For audio transmission, the LAN interface should be operated in 100 Mbps, full mode to prevent audio interference. Therefore, the unit raises the Insufficient LAN alarm if this mode is not established.

### LAN Interface (3)

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



#### Configuration

□·· Operation Settings

--- Clients / Security --- Line Interface

AoIP (LAN/SIP)

Audio Assignment

- Auto Answer

---- TTL / Relay

… DHD Set Logic ⊕ Ember+

Transmission Modes

- General

- 15 kHz Default

General

- Audio Interface

--- AES67

Data Interface

- LAN Interface

- --- VLAN
- .... NTP

- .... Alarm Signalling
  - SNMP
  - --- ACconnect
- .... DHD
- --- Ember+
- --- PhonerSet

.... Quick Dials

..... Login

Service	Alias	TPID	Priority		VID (12-Bit)	
PC Control & PRETALK Streaming		none	-			
VolP		802.1QTag	- 6 (Voice)	-	21	Ø
NTP		none	•			
SNMP		none	•			
DHD		none	•			
Ember+		none	•			
AES67		none	•			
Audio Stream VLAN 1	٥	f none	-			
Audio Stream VLAN 2	٥	f none	-			
Audio Stream VLAN 3	٥	f none	-			
Audio Stream VLAN 4	٥	f none	-			
Audio Stream VLAN 5	٥	f none	-			
Audio Stream VLAN 6	٥	f none	-			
Audio Stream VLAN 7	٩	none 🕈	•			

Modification of the VLAN parameters may interrupt the connection to the PC!

OK Abbrechen

Apply Now



 $\times$ 

- The device supports tagged VLANs.
- The VLAN page is used to configure the VLAN parameters for individual services / protocols.
- VLAN: The VLAN function can be globally activated or deactivated. This setting is also possible via front display. E.g., if access to the system via LAN is no longer possible due to a misconfiguration.
- SERVICE: All services provided by the unit which support VLANs are listed here.
- ALIAS: for the AUDIO STREAM VLANs, any text can be entered here to help assign the correct VLANs on other configuration pages.

- TPID: Enables or disables VLAN for a specific service.
  - NONE: VLAN not enabled.
  - 802.1QTag: VLAN enabled.
- PRIORITY: The priority can be set from 0 = lowest priority to 7 = highest priority. It is recommended to set the priority to 6 for services that transmit audio (VoIP, AES67, audio streams).
- VID (12-bit): VLAN ID for the service.





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

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





AVT

- Alarm parameters can be configured on the ALARM SIGNALLING page.
- DECODER IP PACKET LOSS: On the user interface, under RTP AUDIO QUALITY, the quality of the received stream is signalled for each codec channel via an LED.
  - Good
  - Acceptable
  - Poor

Video

Technologies

- In addition, an alarm is set in the event of poor quality. The threshold values for the individual levels can be set here.
  - MEASUREMENT INTERVAL: Sets the measurement interval for assessing the quality of the stream. During this period, the lost packets (packet loss) are counted and the highest jitter value that occurred is recorded.
  - BAD: The threshold value for packet loss, above which the stream quality is classified as poor.

- ACCEPTABLE: The threshold values for the packet loss as well as the maximum jitter value from which the stream quality is classified as acceptable.
- SECURE STREAMING IP PACKET LOSS: A separate threshold for the alarm can be set for the second stream when secure streaming is activated:
  - MEASUREMENT INTERVAL: Sets the measurement interval for assessing the quality of the stream. During this period, the lost packets (packet loss) are counted.
  - PACKET LOSS: The threshold value for packet loss above which the stream quality is classified as poor.

### Alarm Signalling (2)

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

Video

Technologies

- ENABLE PAD ALARM FOR ENCODER / DECODER: The alarm can be enabled separately for each PTY direction.
- TIMEOUT CODEC 1: The alarm is set if no PAD data has been transmitted during this time period.

### Alarm Signalling (3)

Operation Settings	SNMP								
Clients / Security									
Line Interface AoIP (LAN/SIP)	SNMP Version:	v2c		~		Alarm Traps	Category	^	
Audio Assignment	Read/Trap Community:	public	public			System Alarms			
Auto Answer TTL / Relay	SNMP Port:	161				LCA     Temperature Sensor	Category A Category A		
- DHD Set Logic	NMS 1 (LAN/IP Adr./Port):	LAN: 172.20.225.102	$\sim$	172.20.10.2	162	FLASH EPROM	Category A		
Ember+	NMC 2 (LAN /ID Adv /Devt)	LANE 172 20 225 102	~		162	Overheated	Category A		
Transmission Modes	NM3 2 (DAN/TE Adi./FOI().	LAN . 172.20.223.102	*				Category A		
General	NMS 3 (LAN/IP Adr./Port):	LAN : 172.20.225.102	$\sim$		162		Category A		
15 kHz Default	NMS 4 (LAN/IP Adr./Port):	LAN : 172.20.225.102	$\sim$		162	Ethernet MáC 1	Category A		
🖶 User Defined						Application Alarms	Categoly A		
System Settings	System Description:	MAGICACTIGO				AES/EBU Framing Input 1			
General	Contact:	Admin				AES/EBU Format/Clk Input 1	-		
Audio Interface	Custom Lagation:	Daak				Decoder 1 Sync			
AES67	System Location.	nack				🗹 Decoder 1 IP Packet Loss	-		
Data Interface	Send all traps at system	startup				Encoder 1 PAD			
LAN Interface		~				Decoder 1 PAD	-		
VLAN	Send traps immediately a	after enabling				NTP Server	-		
NTP	0	<b></b>				Encoder 1 Audio Level Left			
Alarm Signalling	Lategory A Alias:	Hardware				Encoder 1 Audio Level Right	-		
SNMP	Category B Alias:					Decoder I Audio Level Left			
- ACconnect	0. 0. AT						-		
- DHD	Lategory L Alias:					Insufficient Ethernet I AN 1	-		
Emper+	Category D Alias:					SIP Registration			
Ouish Dish						DHD Audio Matrix	-		
						Ember+ Consumer 1	-	$\sim$	
Lonin									

AVT Audio Video Technologies SNMP (1)  $\times$ 

- The device can be monitored via a network management system. The parameters for that are configured on the SNMP page.
- The information accessible via SNMP can only be read.
- The device responds to Get requests and sends traps.
- Find the MIB for the device in the installation directory of the PC software.
- SNMP VERSION: Select the SNMP version. SNMPv1 and SNMPv2c are supported.
- READ / TRAP COMMUNITY: Enter a string for the READ community and a string for the TRAP community.
- SNMP PORT: Enter the local UDP port for receiving SNMP requests and sending SNMP responses. The remote port is derived from the received SNMP requests. (Default: 161)

- NMS 1-4: The unit can send traps to up to four Network Management Systems. Traps are sent when the following parameters are set for an NMS:
  - LAN: The IP address of the device used to send SNMP traps.
  - IP ADDR: The destination IP address.
  - PORT: The destination UDP port. (Default: 162)
- SYSTEM DESCRIPTION: A device description. This string is part of the standard MIB.
- CONTACT: The person responsible for this device. This string is part of the default MIB.
- SYSTEM LOCATION: Physical location of the device. This string is part of the standard MIB.



Video

- SEND ALL TRAPS AT SYSTEM STARTUP : Enable this option to send all traps when the device has finished booting.
- SEND TRAPS IMMEDIATELY AFTER ENABLING : Enable this option to send a trap immediately after it has been activated in the configuration.
- CATEGORY A-D ALIAS: To reduce the number of traps, several traps can be combined into one category. If an alarm is set which is contained in the category, the trap of this category can be sent. The name of the category can be configured here.
  - The category traps can be activated at the end of the ALARM-TRAPS list.
- ALARM-TRAPS: All available traps are listed in this table.
  - Select traps which should be sent.
  - Click in the CATEGORY column to assign a trap to a category.

- MISCELLANEOUS TRAPS:
  - CHANNEL 1 RTP QUALITY
  - COLD START: The SNMP agent is being reinitialised.
  - AUTHENTICATION FAILURE: An unknown community string was used in a request.



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



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



Configuration	
<ul> <li>Operation Settings         <ul> <li>Clients / Security</li> <li>Line Interface</li> <li>AolP (LAN/SIP)</li> <li>Audio Assignment</li> <li>Auto Answer</li> <li>TTL / Relay</li> <li>DHD Set Logic</li> <li>Ember+</li> </ul> </li> <li>Transmission Modes         <ul> <li>General</li> <li>15 kHz Default</li> <li>User Defined</li> </ul> </li> <li>System Settings         <ul> <li>General</li> <li>Audio Interface</li> <li>AES67</li> <li>Data Interface</li> <li>LAN Interface</li> <li>VLAN</li> <li>NTP</li> <li>Alarm Signalling</li> <li>SNMP</li> <li>Acconnect</li> <li>DHD</li> <li>Ember+</li> </ul> </li> </ul>	DHD Connection Parameter: Main Redundancy LAN Interface: LAN: 172:20:225:102 ↓ LAN: 172:20:225:102 ↓ TCP/IP Address: 172:20:75.6 Por: 2008 TCP/IP Reconnect Time 10 seconds (1.:255)
	OK Abbrechen Apply Now



- The unit supports DHD-ECP (DHD-External Control Protocol) for communication with a DHD core.
- The connection parameters are configured on the DHD page.
- If DHD is enabled, the DHD Set Logic GPIOs are available.
- ACTIVATE DHD SET LOGIC: Activates the DHD communication protocol.
- The unit always tries to connect via the MAIN connection. While it is not available, the unit connects via the REDUNDANCY connection.
- LAN INTERFACE: Select the IP address of the unit through which the connection to the DHD core is made.
- TCP/IP ADDRESS: Enter the IP address of the DHD Core.

- PORT: Enter the port via which the DHD Core can be reached.
- TCP/IP RECONNECT TIME: Specifies the time interval between TCP connection requests to the DHD core in seconds. The range is 1 - 255 seconds. (Default: 10 seconds)



. Operation Settings	Fmber+
<ul> <li>Operation Settings</li> <li>Clients / Security</li> <li>Line Interface</li> <li>AolP (LAN/SIP)</li> <li>Audio Assignment</li> <li>Auto Answer</li> <li>TTL / Relay</li> <li>DHD Set Logic</li> <li>Ember+</li> <li>Transmission Modes</li> <li>General</li> <li>15 kHz Default</li> <li>User Defined</li> <li>System Settings</li> <li>General</li> <li>Audio Interface</li> <li>Audio Interface</li> <li>LAN Interface</li> <li>VLAN</li> <li>NTP</li> <li>Alarm Signalling</li> <li>SNMP</li> <li>ACconnect</li> <li>DHD</li> <li>Ember+</li> <li>PhonerSet</li> <li>Quick Dials</li> </ul>	Ember +         Activate Ender + Provider         Ender + Connection Parameter:         LAN Interface:       LAN: 172:20:25:102 •         Port 1 (Consumer 1):       000       Port 4 (Consumer 4):       0         Port 2 (Consumer 2):       0       Port 5 (Consumer 6):       0         Port 3 (Consumer 3):       0       Port 6 (Consumer 6):       0         Port 3 (Consumer 7):       0       Port 6 (Consumer 6):       0         Activate Ender + Consumer       0       Port 6 (Consumer 6):       0         Minterface:       LAN: 172:20:225:102 •       Interface:       Interface:       Port 7:220:25:102 •         Provider 1:       172:20:75.6       9000       9000       Porvider 2:       0
	OK Abbrechen Apply Now



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

Video

- ACTIVATE EMBER+ CONSUMER: Activates the Ember+ consumer of the unit. The unit can connect to two providers simultaneously.
- LAN INTERFACE: Select the IP address of the unit through which it connects to the Ember+ providers of other devices.
- TCP/IP ADDRESS / PORT: Enter the IP address and port of each provider to which the unit should connect.



Configuration	
Operation Settings     Cliente / Security	Quick Dials
<ul> <li>Operation Settings         <ul> <li>Clients / Security</li> <li>Line Interface</li> <li>AoIP (LAN/SIP)</li> <li>Audio Assignment</li> <li>Auto Answer</li> <li>TTL / Relay</li> <li>DHD Set Logic</li> <li>Ember+</li> </ul> </li> <li>Transmission Modes         <ul> <li>General</li> <li>15 kHz Default</li> <li>User Defined</li> </ul> </li> <li>System Settings         <ul> <li>General</li> <li>Audio Interface</li> <li>ALGE67</li> <li>Data Interface</li> <li>LAN Interface</li> <li>VAN</li> <li>NTP</li> </ul> </li> </ul>	Quick Dials   2:   3:   unused> \vee   4:   (unused) \vee   5:   (unused) \vee   6:   (unused) \vee   8:   (unused) \vee   9:   (unused) \vee   0:   (unused) \vee
Alarm Signalling SNMP ACconnect DHD Ember + PhonerSet Quick Dials Login	



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





# MAGIC AC1 Go

### Configuration

Login

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Operation Settings       Login         - Chents / Scourity       USER         - Audo Asignmet       Password         - Audo Asignmet       Corfin Password         - Audo Asignmet       Corfin Password         - TU, Relay       Diff Dist Logic         - Cinents       - Corrial         - General       AbMINISTBATOB         - General       Password         - Web Perfined       Corfin Password         System Settings       - Corrial         - General       - Corrial         - Wub Diffined       Corfin Password         System Settings       - Corrial         - General       - Corrial         - Multo Interface       - WADH         - WADH       - Mans Spainling         - Multo Interface       - WADH         - Multo Interface       - WADH         - Multo Interface       - WADH         - WADH       - Mines         - Mones det	onfiguration	
Unstant	Operation Settings	Login
- Une Interface   - Audo (ANS/SP)   - Audo Assignment   - Auto Answer   - The Auto Answer   - The Auto Answer   - The Auto Answer   - The Auto Answer   - Contral   - Stabe Calut   - General   - Auto Interface   - Auto Interface   - Auto Answer   - General   - Auto Interface   - Outo Dial   - Mune   - Mune   - Mone   - Mone   - Outo Dial   - Mone   - Mone   - Outo Dial   - Mone   - Mone<	Clients / Security	
Audio Assymmett       Password         Audio Answer       Corfim Password         Auto Answer       Corfim Password         TIL, Rely       Oth Set Logic         Bithbert       Endert         Tammission Modes       ADMINISTRATOR         General       Password         Other Definied       Corfim Password         Orfim Password       Immethy         Other Definied       Corfim Password         Orfim Password       Immethy         Other Stato       Immethy         Other Stato       Immethy         Other Stato       Immethy         Other Stato       Immethy         Out Interface       Immethy <td< td=""><td>Line Interface</td><td></td></td<>	Line Interface	
Auto Answer T. H. Alaw DHD Set Logic Ember + Tarumission Modes - General - Skift Default - Skift Default - General - Gueral - Gueral - Gueral - Audo Interface - AES/7 - Data Interface - LAN Interface - WAN - NTP - Alam Signalling - Shift - Shift - Gueral - Audo Interface - LAN Interface - WAN - Mana Signalling - Shift - DHD - Ember + - PhonerSet - Curk Dials - Curk Dials	AOIP (LAN/SIP)	Password: ••••••
TRL / Relay   DHD Set Logic   Embers   Tansmission Modes   - General   - Audio Interface   - VuAn   - MIP   - Alam Signaling   - SMMP   - Acconnet   - DHD   - Simers   - PhoneSt   - Quich Dials	Auto Answer	
□ HO Set Logic         ● Ember +         □ Genetal         □ SHz Default         ● User Defined         © Confim Password:         □ Stat Default         □ Genetal         □ Audio Interface         □ Atl Interface         □ All Interface         □ Dato         □ Dato         □ Strit Dist	TTL / Relay	Confirm Password:
Derber+   Tammission Modes   Ceneral   15 Ht2 Default   Der Defined   Confirm Password:   - Audio Interface   - Atsin   - Data Interface   - UAN   - NTP   - Alam Signalling   - SMAP   - Acconnect   - DhoneSet   - OnoneSet   - OnoneSet   - OnoneSet   - OnoneSet	DHD Set Logic	
Tanamisson Modes  - General - StHz Default - General - Audio Interface - ActSo7 - Oata Interface - LAI Interface - VIAN - NTP - AarmSgnalling - SNNP - ACconnect - OHO - Ember+ - Phonefset - Quick Diab Corf - Contex - Define - Cuick Diab -	⊞. Ember+	
- General   - Sthz Default   - User Defined   Confim Password:     - General   - Audio Interface   - Audio Interface   - Audio Interface   - Audio Interface   - VLAN   - NTP   - Alarm Signalling   - SNMP   - Connect   - DHD   - Ember+   - Phone-Set   - Oute Dials	Transmission Modes	ADMINISTRATOR
User Defined User Defined System Settings General Audio Interface Addio Interface Addia Interface All Interface A	General	Password: ••••••
General     Audio Interface     AES67     Data Interface     VLAN     NTP     Alarm Signalling     SNMP     ACconnect     OHD     Ember+     PhonerSet     Quick Dials     Login	- 15 kHz Default	
- General - Audio Interface - AE567 - Data Interface - VLAN - VLAN - MTP - Alarm Signalling - SNIMP - ACconnect - OHD - Ember+ - Phonefset - Quick Dials Login	. System Settings	Confirm Password:
- Audio Interface - A557 - Data Interface - UAN Interface - VLAN - NTP - Alam Signalling - SNMP - Acconnect - DHD - Ember+ - PhonerSet - Quick Dials - ggin	General	
- AES67 - Data Interface - LAN Interface - VLAN - NTP - Alarm Signalling - SNIMP - Acconnect - DHD - Ember + - PhonerSet - Quick Dials Login	Audio Interface	
<ul> <li>Data Interface</li> <li>VAN Interface</li> <li>VLAN</li> <li>NTP</li> <li>Alarm Signalling</li> <li>SNMP</li> <li>Acconnect</li> <li>DHD</li> <li>Ember +</li> <li>PhonerSet</li> <li>Quick Dials</li> </ul>	AES67	
<ul> <li>LAN Interface</li> <li>VLAN</li> <li>NTP</li> <li>Alarm Signalling</li> <li>SNMP</li> <li>AC connect</li> <li>DHD</li> <li>Ember +</li> <li>PhonerSet</li> <li>Quick Dials</li> </ul>	Data Interface	
<ul> <li>VLAN</li> <li>NTP</li> <li>ATam Signalling</li> <li>SNMP</li> <li>ACconnect</li> <li>DHD</li> <li>Ember*</li> <li>Phonefset</li> <li>Quick Dials</li> </ul>	- LAN Interface	
- AIrm Signalling - Alarm Signalling - SNMP - ACconnect - DHD - Ember + - PhonerSet - Quick Dials - Ogin	VLAN	
Acconnect DHD Ember+ Quick Dials Cogin	NTP	
Acconnect DHD Ember+ Quick Dials Login	Alarm Signalling	
Acconnect DHD Ember + PhonerSet Quick Dials Login	ACconnect	
Ember+ PhonerSet Quick Dials Login		
PhonerSet Quick Dials	Ember+	
Quick Dials	PhonerSet	
	Quick Dials	
	Login	



- Passwords can be configured to protect the configuration of the device from unauthorised access. These are set on the LOGIN page.
- There are two levels of permissions:
  - ADMINISTRATOR: If an administrator password is set, the password is required for the following operations:
    - Viewing and changing the configuration
    - Importing configurations
    - Managing presets
    - Loading presets

Audio Video

- Opening the system panel
- Accessing the file system
- Updating the firmware
- Resetting to factory settings
- USER: If a user password is set, the password allows presets to be loaded.
  - If only a user password is set, the user password acts as the administrator password.

- Note: To gain access to the configuration without the password, the unit must be reset to factory settings via the front display.
- Note: The LOCAL SETTINGS are not protected by these logins, but by the permissions of the operating system. (See Local Settings > Settings Folder for further information.)





# MAGIC AC1 Go

#### Main Panel

AoIP/SIP

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# Audio Video Technologies Main Panel – AoIP/SIP (1)

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

Video

**Fechnologies** 

- If the channel is connected, the SIP account which is used for the connection is displayed.
- If the channel is disconnected the SIP registration status off all configured accounts of the channel is displayed via a status text and the font colour of that text:
  - REGISTERED: (green) All SIP accounts are registered.
  - **REGISTERED:** (yellow) At least one SIP account could not register with the SIP server.
  - INVALID: (red) None of the SIP accounts could register with the SIP server.

Main Panel – AoIP/SIP (2)

- Click on the text to get more information about the available SIP accounts and their registration status.
- In case of registration errors, you can get more information under System Monitor > SIP Monitor.

- **3**. Channel Status: Shows the connection status of the channel in different colours. During a connection, number, name or IP address of the remote party is displayed.
  - Grey: There is no connection.
  - Green: The connection is established.
- 4. Encoder and decoder status.
  - The audio quality is displayed via a symbol:
    - HQ: A high-quality codec is used.
    - HD: The G.722 codec is used.
    - SD: The G.711 codec is used.
  - Click on the symbol to get information about
    - codec
    - algorithm
    - data rate
    - PAD
    - Secure Streaming

- 5. The RTP Audio Quality shows the quality of the received data stream in the colours green, yellow and red. The threshold values of the individual levels are set on the Alarm Signalling configuration page.
- 6. Status of the Audio Level monitoring. The thresholds are set on the **Alarm Signalling** configuration page.
- 7. The Level Meters of Encoder on the left and Decoder on the right show the current audio level in dBFS. During a connection, the level is displayed in green.

If there is no connection the Encoder level is displayed in grey.

8. CALL: Press the button to establish a connection. When there is an incoming call, the key flashes yellow.

**9**. DROP: Press the button to terminate the connection or reject an incoming call. When there is an incoming call, the button flashes yellow.

#### Video Technologies Main Panel – AoIP/SIP (3)



#### T Video Technologies Main Panel – AoIP/SIP (4)

- The detailed codec information can be permanently displayed on the main panel.
- This view is enabled under Menu > Configuration
   > Local Settings > Application Parameters > Show detailed Codec information in Codec
   Line
- 1. CODEC: Codec information shows details of encoder and decoder combined.
- 2. DETAILS: Shows the following information if available:
  - Codec
  - Algorithm
  - Sampling rate
  - Bitrate
  - Mode (Mono, Stereo)
  - PAD status and bitrate
  - Secure Stream mode
  - Status of stream 1 and stream 2 if Secure Streaming is enabled.

#### Video Technologies Main Panel – AoIP/SIP (5)

- Press the CALL button on the main panel to open the dialling window.
- 1. Select a SIP account
  - ① Displays detailed SIP account information and status.
- 2. Enter a phone number when using the SIP account or an IP address for Direct SIP.
  - Open a list of last dialled numbers.
- **3**. Select a transmission mode for the connection.
  - ① Displays codec details of the selected transmission mode.

Dialling

- 4. Number pad
- 5. CALL: Dials the number CLOSE: Closes the dialog

Audio

Video

Technologies

Search	<b>&lt; x</b>	Local			⊙ ✔ ←
Andrew Fuller AAC-LD 128	<b>641</b> Local	411		<	C ×
<b>Jane Doe</b> PCM24	411 Local	PCM24 (PCM)			0 🗸 🔶
<b>Laura Callahan</b> 7 kHz	<b>004991152710</b> Provider	1	2	3	
		4	5	6	<b>←</b>
		7	8	9	
		*	0	#	
NEW	EDIT DELETE	CALL		CLOSE	-

- The internal phone book of the unit is displayed on the left of the dialling window.
- Search for an entry. Leave blank to display all phone book entries.
- 2. Search result, or list of all entries . Select an entry, to transfer it to the dialling parameters on the right.
- **3.** Add a new phone book entry.
- 4. Edit the selected phone book entry.
- 5. Delete the selected phone book entry.




- When adding a new entry or editing an existing entry, a new windows opens.
- Enter the name. 1.
- 2. Select a SIP account.
  - (I) Displays detailed SIP account information.
- **3.** Enter the phone number or IP address.
- Select a transmission mode. 4
  - ① Displays codec details of the selected transmission mode.
- 5. Applies the changes and closes the window.
- 6. Discards the changes and closes the window.

Video



#### Phone book entry Technologies



# MAGIC AC1 Go

### Main Panel

IP Leased Line

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#### Audio Video Technologies Main Panel – IP Leased Line (1)

- In the IP Leased Line operating mode, the codec channels is displayed in the main window.
- The following information is displayed from top to bottom:
- **1.** The label of the channel.
- 2. Operation Mode.
- **3**. Channel Status: Shows the connection status of the channel in different colours. During a connection, the IP address of the remote party is displayed.
  - Grey: There is no connection.
  - Green: The connection is established.

- **4**. Encoder and decoder status.
  - The audio quality is displayed via a symbol:
    - HQ: A high-quality codec is used.
    - HD: The G.722 codec is used.
    - SD: The G.711 codec is used.
  - Click on the symbol to get information about
    - codec
    - algorithm
    - data rate
    - PAD
    - Secure Streaming
- 5. The RTP Audio Quality shows the quality of the received data stream in the colours green, yellow and red. The threshold values of the individual levels are set on the **Alarm Signalling** configuration page.

#### Video Technologies Main Panel – IP Leased Line (2)

- 6. More status information:
  - Status of the Audio Level monitoring which can be configured on the **Alarm Signalling** configuration page.
  - Status of the PAD monitoring which can be configured on the **Alarm Signalling** configuration page.
  - Status of the Audio Delay Sync of Encoder and Decoder. Only available for the Decoder if IP Leased Line Extended is used.
- 7. The Level Meters of Encoder on the left and Decoder on the right show the current audio level in dBFS. During a connection, the level is displayed in green.

If there is no connection the Encoder level is displayed in grey.

8. CONNECT: Press the button to establish the connection.

**9.** DROP: Press the button to terminate the connection.

#### Video Technologies Main Panel – IP Leased Line (3)



#### Audio Video Technologies Main Panel – IP Leased Line (4)

- The detailed codec information can be permanently displayed on the main panel.
- This view is enabled under Menu > Configuration
  > Local Settings > Application Parameters > Show detailed Codec information in Codec
   Line
- **1.** ENCODER / DECODER: Codec information is shown for encoder and decoder separately.
- 2. DETAILS: Shows the following information if available:
  - Codec
  - Algorithm
  - Sampling rate
  - Bitrate
  - Mode (Mono, Stereo)
  - PAD status and bitrate
  - Secure Stream mode
  - Status of stream 1 and stream 2 if Secure Streaming is enabled.

#### Video Technologies Main Panel – IP Leased Line (5)



# MAGIC AC1 Go

### Operation

Multi Control Software

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

Video

- Click on a tile to open the standard operation panel of a device.
- Click on the configuration icon ( ) to open the configuration of a device.

#### Multi Control Software (2) Technologies



# MAGIC AC1 Go

### Maintenance

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

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

Video

Technologies



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

Video

Firmware Downloa	d		
ac1go.ssw			Browse
Start	]	Cance	əl
Progress:			
[	Close		

#### Firmware-Update Technologies

Registration									
Hardware	1AGIC AC1 Go								
Software M	1AGIC AC1 Go (2.412/2	2.412)							
Main	Main								
Subject Number/PCB Id:	450186 / 441073								
Factory Number:	23/13/1076		ī						
Year:	2023		1						
Hardware Version:	1.00		1						
MAC Address:	00-06-9B-02-00-00		1						
Analogue Audio 1 In:	0,00 dB (left)	0,00 dB (right)							
Analogue Audio 1 Out:	0,00 dB (left)	0,00 dB (right)							
Features									
Software Options		# Expiry							
MAGIC AC1 Go									
Number of PC licence	es	5							
MPEG Layer 2 Coded	:	1							
MPEG Layer 3 Codeo	:	1							
AAC LD Codec (Low	Delay) panced Low Delay)	1							
AAC LC Codec (Low	Complexity)	1							
HE-AAC v1 Codec (H	ligh-Efficiency v1)	1							
HE-AAC v2 Codec (H	ligh-Efficiency v2)	1							
Number of PhonerSe	et licences	1							
Create Test Lice	nce Key E	inter password							
	Close								

# Audio<br/>Video<br/>TechnologiesRegistration and Software Options (1)

A

- Under ADMINISTRATION REGISTRATION, the hardware information and software licences are displayed.
- HARDWARE: The name of the product.
- MAIN: Hardware information of the main board.
  - SUBJECT NUMBER: Our product ID.
  - FACTORY NUMBER: The serial number..
  - YEAR: The year of manufacture.
  - MAC ADDRESS: The MAC address of the LAN interface.
  - ANALOGUE AUDIO IN/OUT: The calibration values of the analogue audio interface.
- FEATURES: List of all available software options.
  - Activated options are ticked.

Video

Technologies

 For options that can be activated more than once, the number is shown to the right.

- CREATE TEST LICENCE KEY: If you like to test a feature before buying it, create a test licence key and send it to us along with the factory number of the unit.
  - The test licence will work for a limited time.
  - The timer only counts when the unit is running.
- The test licence enables all available features.
- ENTER PASSWORD: Software options can be activated later. To do so, contact AVT with the SUBJECT NUMBER and the FACTORY NUMBER. You will receive a licence password which you can enter into the system here.

### Registration and Software Options (2)

Syster	n ala	arms					Etherne	t stat	e		Abs. dat	a rates	
0	0	LCA	0	0	Overheated		LAN	9	0 100 MBit	t/s, full	TX: 5	5,3 MBit/s RX:	8,4 MBit/s
•	0	MAIN EEPROM	•	0	FLASH EPROM		IP Trans	missi	on Jitter	Current	Max. last	60 sec	Maximum
•	0	Temperature Sensor	•	0	Display Contras	t DAC	OFF						
9	0	VCXO 1											
9	0	Ethernet MAC 1											
Applic	atio	n alarms											
0	0	AES/EBU Input 1	No A	AES si	gnal available		RX Jitte	r Buff	er	Current Min.	Max.	Last Min.	Max.
•	0	NTP Server	172.	16.30	.1		AC	Deco	der	11 msec	12 msec	10 msec	12 msec
9	0	AES67 Rx Stream 1	9	0	AES67 Rx Stream	12	AES67 F	bx Str	eams			Pa	acket Loss
9	0	SIP Registration					TF-	USB-M	lusic-1:1				0
•	0	Encoder 1 Audio Level Left	•	0	Encoder 1 Audio	Level Right	TF-	USB-M	lusic-2:1				0
•	0	Decoder 1 Audio Level Left	•	0	Decoder 1 Audio	Level Right	PTP Sta	te:	Slave	PTP Maste	er: 172.	20.30.1	
9	0	DHD Audio Matrix					Path De	elay:	267 µsec				
•	0	Decoder 1 Sync					IP Pack	et Los	s Counter				
•	0	Decoder 1 IP Packet Loss					Dec	oder		0			
							Remote	GPIO	Reception				
							AC			OFF			
Syster	n sta	ate					Connect	ed En	nber+ Cons	sumer			
Syste	m Te	emperatur 34 °C	DSF	P Loa	d: 28 %		Connect	ed PC	s				
Data I	nter	face state		Ab	s. data rates		1: 172.	20.225	.1:60693				
R523	2 (C	ontrol) 300 Baud, 8N1		Т	X: 0 Bit/s R	X: 0 Bit/s							
R523	2 (D	ata) 300 Baud, 8N1		Т	X: 0 Bit/s R	X: 0 Bit/s							
Last C	ount	ter Reset: 28.04.2023 16:11:3	8										
		Alarm Counter Reset				SIP M	onitor				Close	e	

A

# Audio Video Technologies System Monitor (1)

- Under MENU SYSTEM MONITOR an overview of the system status is shown.
- The status of the individual components is indicated by a coloured LED:
  - (red): The alarm is active. There is an error.
  - (yellow): The alarm is active, but not relevant in the current operating mode.
  - (green): No alarm. The function works as expected.
- An error counter is displayed next to each alarm LED, showing how often the error occurred, since the unit was started, or the error counters were reset.
  - The counter can be reset by pressing the ALARM COUNTER RESET button.
  - The time of the last reset is displayed above the key.
- If the AoIP/SIP mode is enabled, the status of each SIP account is displayed when pressing the SIP MONITOR button.

Video

Technologies

- SYSTEM ALARMS: These alarms affect the hardware components of the device. An active alarm means that the component in question has failed temporarily or permanently. An alarm may also indicate a problem with the control of the component in the firmware.
- APPLICATION ALARMS: These alarms are used to monitor the ongoing operation of the unit.
  - AES/EBU INPUT: The unit detects whether a valid AES/EBU signal is present at the input. More information is displayed to the right of the alarm.
  - AES67 RX STREAM: Shows whether the configured AES67 streams are being received.
  - SIP REGISTRATION: Shows whether all AoIP channels could be registered with the SIP server.

System Monitor (2)

- ENCODER / DECODER AUDIO LEVEL: Indicates whether the corresponding audio signal has fallen below the set threshold for the set period. This alarm must be activated in the settings.
- DHD AUDIO MATRIX: Shows whether the unit was able to establish a connection to the configured DHD core.
- EMBER+ CONSUMER: Shows whether the unit was able to establish a connection to the configured Ember+ provider.
- DECODER SYNC: Indicates whether the decoder can decode the incoming data stream.
- DECODER IP PACKET LOSS: Shows whether the number of packet losses exceeds the threshold for BAD stream quality as configured under **Alarm** Signalling.

System Monitor (3)

Audio

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Technologies

- SYSTEM STATE: General information about the system state is displayed here.
  - SYSTEM TEMPERATURE: Displays the temperature measured in the enclosure. Above 57°C, the OVERHEATED alarm is triggered. The temperature should always be kept below 50°C. To do this, it is necessary to leave at least one unit of free space above the device in the rack so that the heat can escape via the ventilation grille in the enclosure lid.
- DSP LOAD: Shows the current load of the DSP in percent.
- DATA INTERFACE STATE: Shows the status of the RS232 data interfaces.
  - ABS. DATA RATES: Shows the current data rates of the RS232 interfaces in the transmitting direction (TX) and receiving direction (RX).

- ETHERNET STATE: Shows the status of the LAN interface.
  - LED = Link Status.

Video

Technologies

- (green): A network cable is connected and the connection to the switch is established.
- (grey): No network cable is connected or the connection to the switch is disturbed.
- Operating mode of the LAN interface:
  - An interface used for audio transmission must be set to 100 Mbps, full to avoid drop-outs.
- ABS. DATA RATES: Shows the current data rates of the LAN interface in the transmitting direction (TX) and receiving direction (RX).
- IP TRANSMISSION JITTER: Provides a jitter measurement for the stream of the codec channel.
  - Select the Decoder to display the measurement.
  - CURRENT: Current jitter value. Updated every two seconds.
  - MAX. LAST 60 SEC: Highest jitter value in the last 60 seconds.
  - MAXIMUM: Highest jitter value since the start of the measurement. This value can be reset via the MAX. JITTER RESET key.

- RX JITTER BUFFER: Shows the fill levels of the receive buffer for the codec channel. High fill levels can be due to high jitter, active Secure Streaming or active Audio Delay Sync.
  - CURRENT MIN: Shows the minimum fill level in the last two seconds.
  - MAX: Shows the maximum fill level in the last two seconds.
  - LAST MIN: Shows the minimum fill level in the last 60 seconds.
  - MAX: Shows the maximum fill level in the last 60 seconds.

### System Monitor (4)

- AES67 RX STREAM: Shows the status of the configured RX AES67 streams.
  - LED = reception status:
    - (green): Audio data is being received.
    - (red): No audio data is being received.
  - PACKET LOSS: Shows the number of packet losses of the stream since the beginning of the measurement.
  - PTP STATE: Shows the state of the PTP clock synchronisation. The unit can only operate as a slave.
  - MASTER: Shows the IP address of the PTP master.
  - PATH DELAY: Shows the time delay of the connection to the PTP master.
- IP PACKET LOSS COUNTER: Shows the number of packet losses of the receive streams of the codec channel since the device was started or the alarm counters were reset.
- REMOTE GPIO RECEPTION: Shows whether the transmission of GPIO states is activated.
- CONNECTED EMBER+ CONSUMER: Lists all Ember+ consumers that are connected to the Ember+ provider of the unit.
- CONNECTED PCS: Lists all PCs that are connected to the unit.

Video

Technologies

 CONFIGURATION DIALOGUE OPEN AT: The system settings cannot be opened on more than one PC at the same time. This line shows on which PC the configuration is open.

### System Monitor (5)

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

SIP Monitor			×
SIP User	Main SIP Server	Backup SIP Server	
AC-Provider: 431	Test Registration successful.	Test No IP address available	
AC-Local: 431	Test Registration successful.	Test No IP address available	
AC-Provider:	Test Not executed.	Test No IP address available	
AC-Provider:	Test Not executed.	Test No IP address available	
AC-Provider:	Test Not executed.	Test No IP address available	
AUX-Provider: 432	Test Registration successful.	Test No IP address available	
AUX-Local: 432	Test Registration successful.	Test No IP address available	
AUX-Provider:	Test Not executed.	Test No IP address available	
AUX-Provider:	Test Not executed.	Test No IP address available	
AUX-Provider:	Test Not executed.	Test No IP address available	
	SIP Log Close		



SIP Log		×
	Logfile stopped	
SIP User filt	er: No SIP User filter activ	e
	Start Logging	
	Start SIP registering	
	Stop Logging	
	Save Logfile	
	View Logfile	
	Close	
	Close	



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

- VIEW LOGFILE: Open the logfile in a text editor on the PC.
- SAVE LOGFILE: Click to save the logfile on the PC.



💶 Logfile Vie	wer						:
Date	Time (Local)		Type	Source	Status	Description	Filter
20 04 2022	16,47,09,901	Ľ	TTL Statue	Local			
20.04.2023	16:47:00,001		TD Dy stype lest	Chappel 1	CLEARED	Coupton 20001	
20.04.2023	16:47:00,709		IP RX scream losc	Channel 1		Lost packate: 1	
28.04.2023	16:47:08,693		IP Rx consecutive packet lost	Channel 1		Lost packets: 1	
28.04.2023	16:47:08,676		IP RX consecutive packet lost	Channel 1		Courters 20070	
28.04.2023	16:47:08,660		IP RX packet lost	Channel I		Counter: 38978	IP events
28.04.2023	16:47:07,000		IP RX packet out or time			Counter: 38875	
28.04.2023	16:47:06,997		IP RX packet lost	Channel 1		Counter: 38875	TIL/Relay events
28.04.2023	16:47:06,983			Local	SET	TIL 3	Backup events
28.04.2023	16:47:06,962		Start IP Rx	Channel 1		Counter: 38872	NTP events
28.04.2023	16:47:04,662		TTL Status	Local	CLEARED	TTL 8	Transmission ev
28.04.2023	16:47:04,662		TTL Status	Local	CLEARED	TTL 7	
28.04.2023	16:47:04,662		TTL Status	Local	CLEARED	TTL 6	
28.04.2023	16:47:04,662		TTL Status	Local	CLEARED	TTL 5	
28.04.2023	16:47:04,662		TTL Status	Local	CLEARED	TTL 4	Set Filter
28.04.2023	16:47:04,662		TTL Status	Local	CLEARED	TTL 3	
28.04.2023	16:47:04,662		TTL Status	Local	SET	TTL 2	
28.04.2023	16:47:04,662		TTL Status	Local	SET	TTL 1	
28.04.2023	16:47:04,565		IP R× stream lost	Channel 1		Counter: 38873	Export (using Filter)
28.04.2023	16:47:04,549		IP Rx consecutive packet lost	Channel 1		Lost packets: 1	Export (dsing nicer)
28.04.2023	16:47:04,533		IP Rx consecutive packet lost	Channel 1		Lost packets: 1	Delete Logfile File
28.04.2023	16:47:04,516		IP Rx packet lost	Channel 1		Counter: 38870	Delete Logi lie File
28.04.2023	16:47:03,195		Configuration loaded			R:TMPCFGB0.CFG	
							Reload Logfile File
							Close

Audio Video Technologies Logfile Viewer (1)

AVI

- Status changes of the unit can be recorded in an internal log file.
- The log file is configured on the GENERAL settings page.
- In order to be able to evaluate the log file, the Logfile Viewer, which is part of the PC software, can be used.
- Open the Logfile Viewer via Menu > Logfile Viewer.

Video

Technologies

- The individual events are displayed line by line in the table on the left.
  - DATE: The date on which the event was recorded. The correct date is only displayed if the unit has a connection to an NTP server.
  - TIME (LOCAL): The time at which the event was recorded. The correct time is only displayed if the unit has a connection to an NTP server. The time is converted to the time zone of the PC by the PC software.

- DURATION [d] hh:mm:ss,msec: Duration for which the event was active in hours : minutes : seconds , milliseconds. This information is only available if it is relevant.
- TYPE: A short information about the type of event.
- SOURCE: Source of the event, e.g., Channel, LAN.
- STATUS: Shows the new state of the component.
  - OK: An alarm is reset.
  - ERROR: An alarm has occurred.
  - SET / CLEARED: TTL status
  - LAN mode (speed / duplex)
- DESCRIPTION: A short text describing the event in more detail or providing additional information.
- FILTER: The list of events can be filtered. Check the boxes of the categories which should be displayed.
  - SET FILTER: The new filter settings are not applied to the table until the SET FILTER key has been pressed.

Logfile Viewer (2)

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

Audio Video





# MAGIC AC1 Go

### Front Display

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

Video

#### 3 6 4 1 Status 2 ahr NAMES SIP: Provider DISC O MPEG -30 MENU 6 mno 7 Dors AUDIO CODEC 5 7 8 2

#### Front panel operation Technologies

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





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

SEARCH:	BACK
Andrew Fuller	ABC
Jane Doe Laura Callahan	OPTS



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

TYPE NUMBER	DELETE
52710	123
🗘 MPEG	



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

Video



AUDIO CODEC TX I RX				
FLAC 16	FLAC 16			
48 kHzl1344 kBit/s	48 kHz11088 kBit/s			
STEREO	STEREO	~		

#### **Front - Connection Status** Technologies



# MAGIC AC1 Go

### Interfaces

TTL

Data / Control

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### TTL / Relay Interface

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

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



### RS232 1/2 Interface

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

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







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