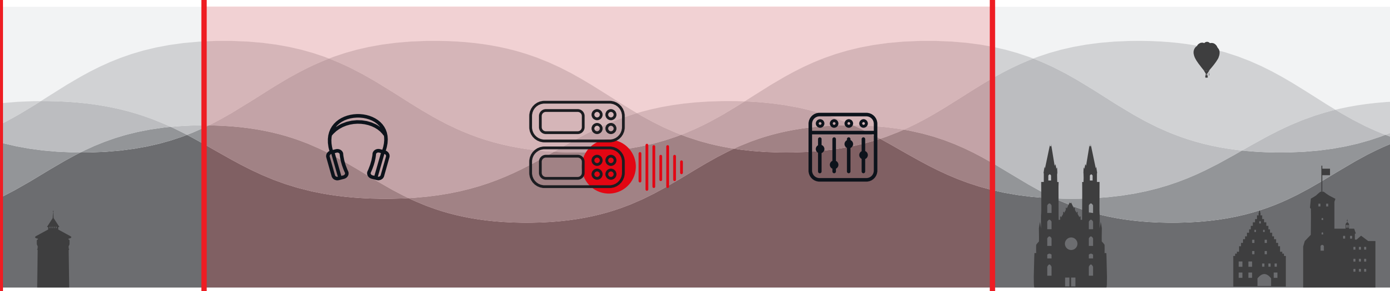
Audio Codecs



[AoIP | Leased Line | E1]

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General

Audio Codecs are needed for high-quality Audio transmissions. Over IP, both Leased Line connections as well as temporary dial-up connections can be used.

Audio Codecs are required for a variety of applications such as e.g. reporting, studio programme contribution and distribution as well as Studio-Transmitter-Links.

For installations in OB vans and reporting applications, it is often required to have compact and mobile systems.

Especially for setups which are running 24/7, the systems need to be highly reliable. Very often it is required to have an automatic backup function for these systems.

Whereas ISDN and 2-Mbit/s were the main technologies used in the past, nowadays **AoIP** is the **transmission standard**. The European Broadcasting Union has defined a standard for AoIP dial-up connections already in 2007. All important Audio Codec manufacturers are supporting this standard (EBU Tech 3326) which ensures interoperability between Audio Codecs from different manufacturers. This is a great benefit compared to ISDN where no real standard has been created.

Depending on the application different **coding algorithms** are used. The selection of the coding algorithm depends on the available bitrate, the desired quality and the acceptable delay. The EBU names the following Audio algorithms as mandatory to comply with the AoIP standard. **G.711, G.722, ISO/MPEG Layer 2** and **PCM** (for stationary Audio Codecs). Furthermore, **MPEG4 AAC-LC, MPEG4 AAC-LD** and **apt-X** are recommended as further algorithms.

For **AoIP dial-up connections** a SIP Server can be used. The Audio Codec registers at the SIP Server with a SIP account and a password. The SIP account corresponds to the telephone number under which the Audio Codec can be reached. If no SIP Server is used, the Audio Codec can be called only via its IP address.

In Germany, the public broadcasters have installed one big SIP Server at the ARD Sternpunkt in Frankfurt where all users can register to communicate with each other.

Currently, users can only connect with each other if they are registered at the same SIP Server since so far there are no gateways which connect one SIP Server with another. For this reason, it is a great benefit if an IP Audio Codec can register at several SIP Servers at the same time.

Audio Codecs are either controlled via the front panel or – if the operator is located not near the system – via a **Windows PC Software** using an IP connection. In some workplaces, it is also desirable to be able to control several systems via one user interface. Alternatively, configuration and control via **web browser** is also possible.

Features & Symbols

In this product catalogue we will use some symbols for the systems' features and their availability. Below you will find a description of all features.

incl. A red symbol shows that the feature is available in the standard delivery version for the product described.

Option An orange symbol shows that the feature is optionally available (associated with costs).

N/A A light grey symbol shows that the feature is not supported by the product described.

IP

IP interface(s)

The system can be connected to IP lines.

E1

E1 interface

The system is equipped with an E1 interface for the classic Audio contribution application via 2-Mbit/s network.

Secure Streaming

Secure Streaming

With IP leased lines, a connection via one or (with MAGIC ACip3) also two IP links can optionally be established redundantly with different delays, in which packets are transmitted in parallel. This ensures a highly reliable connection.

GPIO

GPIO

Programmable TTL interfaces and Relay contacts are available for external control or signalling.

DHD

DHD SetLogic

The DHD SetLogic commands can be used to communicate with DHD mixing consoles or routers to easily exchange control and signalling commands over IP.

Ember+

Ember+

The Ember+ protocol allows the control of the systems from a LAWO/DHD mixer or any other Ember+ compatible system.

2-Codecs

2-Codecs Upgrade

The system can be upgraded to transmit two Stereo signals in parallel.

SD Card

SD card

An SD card interface is available (SD card is not included).

Backup

Backup

A backup functionality can be configured.

AES67

AES67

The AES67 upgrade allows the use of 4 x (or with 2-Codecs Upgrade 6 x) additional audio channels over IP via AES67, the lowest common denominator of multiple technologies such as AES67-compatible Dante® and Ravenna systems.

Dante

Dante®

The Dante® module allows the use of up to 32 audio channels via IP using Dante®/AES67.

Control via PC software

The system can be operated via a Windows PC software using a tablet or PC connected via LAN to the Audio Codec. Via the software, multiple codecs can be controlled.

Control via web browser

Google Chrome, Safari and Firefox are currently supported.

SNMP

Support of the SNMP protocol V1 and V2c to integrate a system into a network management system.

Redundant Power Supply

As backup of the integrated power supply a redundant external 12 V power supply can be connected.

n x LAN

All Audio Codecs provide at least one LAN interface. Additional LAN interfaces are available as an option/by default.

Quality of Service

Quality of Service parameters can be selected to give different priorities to the Audio and data streams in your network.

VLAN

To separate the audio signals from the control data, VLANs (Virtual Local Area Networks) can be set up in the system configuration.

Data RS232

Transparent data transmission via RS232 for e.g. RDS.

G.711 & G.722

The system supports the G.711 (3.1 kHz) and the G.722 (7 kHz) coding algorithms.

Layer 2

The system supports the ISO/MPEG Layer 2 coding algorithm.

Layer 3

The system supports the ISO/MPEG Layer 3 coding algorithm.

AAC-LD

The system supports the AAC-LD coding algorithm.

AAC-ELD

The system supports the AAC-ELD coding algorithm.

AAC-LC

The system supports the AAC-LC coding algorithm.

HE-AAC V1

The system supports the HE-AAC V1 coding algorithm.

HE-AAC V2

The system supports the HE-AAC V2 coding algorithm.

Opus

The system supports the Opus coding algorithm.

Enhanced apt-X

The system supports the Enhanced apt-X 24 Bit (MAGIC ACip3) coding algorithm.

PCM

The system supports uncompressed PCM Audio transmission.

FLAC

The system supports FLAC Audio transmission.

	MAGIC ACX Dante® WAN Bridge	MAGIC ACip3 (2M)	MAGIC AC1 Go
Feature			
Line interfaces	2 x LAN	3 x LAN 1 x E1 (2M version only)	1 x LAN
Audio interfaces	-	1 x Headphones 1 x Stereo Analogue 2 x Stereo Digital AES3 OR: 2 x Stereo Analogue	1 x Stereo Analogue OR: 1 x Stereo Digital AES3
AoIP channels	32 x RX/TX Dante® (native) 32 x AES67 (8 streams)	4/6* x AES67 RX (2 streams) 4/6* x AES67 TX (1 stream) (optional)	4 x AES67 RX (2 streams) 4 x AES67 TX (1 stream) (optional)
Coding algorithms	PCM 16/24 ***Optional:*** G.722 MPEG Layer 2 Opus	G.711 G.722 PCM 16/20/24 MPEG Layer 2 Opus FLAC ***Optional:*** MPEG Layer 3 Enhanced apt-X (24 Bit) AAC-LD AAC-ELD AAC-LC HE-AAC V1 HE-AAC V2	G.711 G.722 PCM 16/20/24 MPEG Layer 2 Opus FLAC ***Optional:*** MPEG Layer 3 AAC-LD AAC-ELD AAC-LC HE-AAC V1 HE-AAC V2
Codecs per system	8 - 32	1 - 2	1
Data rates	64 – 36.864 kbit/s (Channels/Codecs-dependent)	16 – 2304 kbit/s (depending on Codec)	16 – 2304 kbit/s (depending on Codec)
Sampling frequencies	48 kHz 44,1 kHz (on request)	8, 16, 24, 32, 48-kHz (depending on Codec)	8, 16, 24, 32, 48-kHz (depending on Codec)
Secure Streaming for IP Leased Line Mode	-	yes	yes
SIP Backup function	-	optional	-
DHD SetLogic/ Ember+ protocol	-	yes	yes
Control interfaces	GPIO (8 x TTL, 8 x Relays) 2 x LAN	GPIO (6 x TTL, 6 x Relays) 3 x LAN	GPIO (4 x TTL, 2 x Relays) 1 x LAN
Dimensions	19", 1U	19", 1U	½ x 19", 1U
Power Supply	100 – 230 V + 5V table top PSU (optional)	100 – 230 V + ext. 12 V PSU (optional)	external 12 V

*** with 2-Codecs Upgrade**

Available Solutions

The **MAGIC ACX Dante® WAN Bridge** offers the possibility to connect two Dante® networks over long distances. The limited maximum latency of Dante® for long-distance transmissions, high jitter and possible clock differences between transmitter and receiver are solved by the system.

Compressed transmissions with G.722, MPEG Layer 2 and Opus are optionally possible if the available transmission bandwidth is limited.

The **MAGIC ACip3** is a universally applicable solution that offers up to two audio codecs alongside two command channels sin one unit.

A variety of coding algorithms are available for the system to cover every application.

For transmitter feeds (STL), a **2M variant for E1 networks** is also available, which enables a later conversion to IP networks.

The **MAGIC AC1 Go** is functionally based on the MAGIC ACip3.
The compact single-channel audio codec with a LAN interface also has an additional command channel and its design makes it ideal for e.g. OB Vans.

MAGIC ACX Dante® WAN Bridge

MAGIC ACX Dante® WAN Bridge

IP | E1 | Secure Streaming | GPIO | DHD | Ember+ | 2-Codecs | SD card | Backup | AES67 | Dante

SNMP | redundant | 2 x LAN | QoS | VLAN | Data RS232

G.722 | Layer 2 | Layer 3 | AAC-LD | AAC-ELD | AAC-LC | HE-AAC V1 | HE-AAC V2 | Opus | Enhanced apt-X | PCM

- 32 audio channels via Dante® (2 x GbE)
- AES67 compatible
- PCM16 / PCM24
- 48 kHz Sampling frequency (44,1 kHz optional)
- Intelligent sampling rate adjustment (SRA)
- Jitter buffer up to 500 ms
- Audio level detection
- 2 x 100 Mbit/s Ethernet
- 8 x TTL GPIO / 8 x Relay
- VLAN/QoS support
- SNMP v1, v2c
- System internal log file
- Optional redundant power supply
- Comfortable Windows management software

The system **MAGIC ACX Dante® WAN Bridge** enables the transmission of up to 32 uncompressed audio signals via wide area networks (WAN).

The audio connection for the audio inputs/outputs is provided by the integrated 32-channel Dante® interface, which has redundant GbE interfaces and of course also supports AES67.

However, three major problems arise in a wide area transmission of two Dante® networks:

- Possibly different clock signals at the sender and receiver
- High jitter
- Network dependent transmission delays

Dante® allows a maximum latency of 5 ms, which is fully sufficient in local networks. However, in the case of long-distance transmissions (for which Dante® was not developed), the delay of individual IP packets can vary so much that they arrive too late at the receiving system. This inevitably makes dropouts in the audio signal audible.

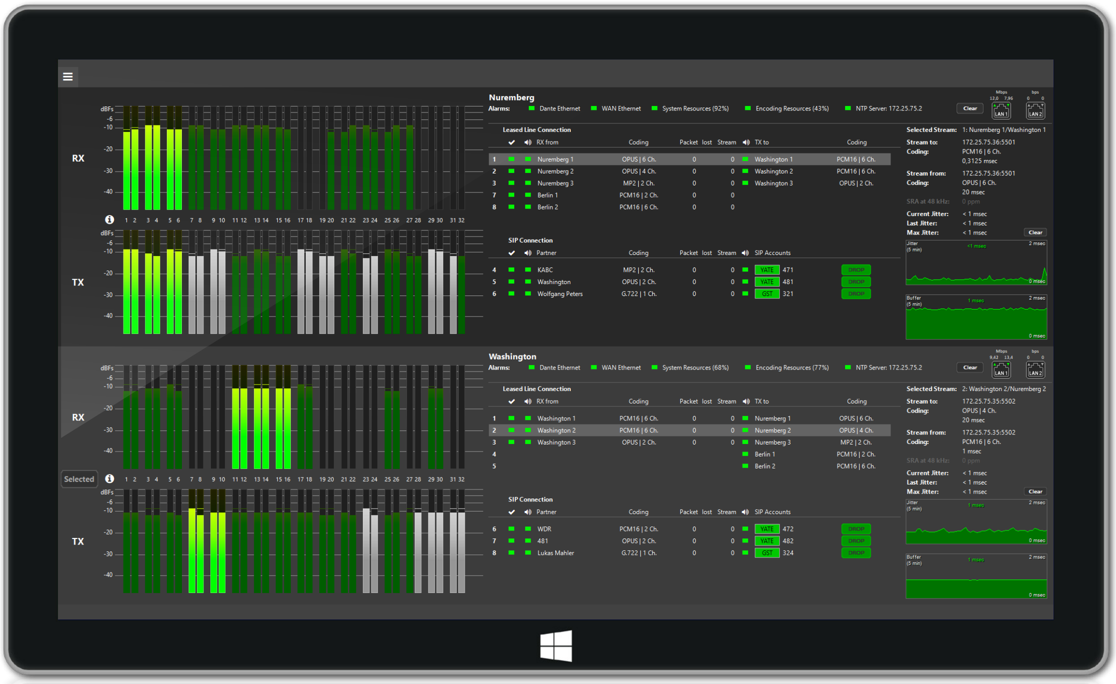
The system solves the problem of different clocks by an intelligent adaptation of the sample rate (SRA = Sample Rate Adaptation).

Problems caused by high jitter as well as long transmission delays are effectively prevented in the system by an adjustable jitter buffer in the range of 2 ... 500 ms or alternatively by the automatic jitter buffer adjustment which can be activated. Thus, transmissions via the Internet over long distances are also conceivable.

The transmission takes place via one of the two standard network interfaces. The number of channels to be transmitted is freely configurable. If required, the transmit and receive data streams can be physically separated.

MAGIC ACX Dante® WAN Bridge and MAGIC ACX Dante® WAN Bridge with optional redundant power supply (below)

MAGIC ACX Dante® WAN Bridge PC Software



Management & Monitoring

The comfortable management software of the system can manage up to 10 systems in one graphical user interface. Depending on the screen resolution, several systems can be displayed on one page or on several tabs.

Up to 5 workstations can access one or more systems simultaneously.

The transmit and receive levels of all transmitted and received audio channels are displayed, including alarms for “empty” audio channels, general system information such as IP addresses and alarms as well as graphs of the time history of the jitter buffer and jitter.

Both graphs allow a representation as a short-term(5 min) or as long-term statistics (1 day). Within the statistics, periods with buffer overflow or underflow, stream and packet losses are also marked.

For test purposes, a sine wave generator can be activated which either outputs the signal locally via Dante® or sends it to the remote station.

The internal system log file allows detailed monitoring and tracking of errors that have occurred - even without a connected PC.
If required, the log file can be downloaded from the system at any time via the management software and clearly displayed in the log file viewer.For exact time information in the log file, the system has NTP synchronization.

For monitoring and alerting, the system naturally also offers SNMP. Traps can be reported to up to four network management systems.

The front display of the system also shows essential information on the status of the transmission. A basic configuration is also possible.

Options

MAGIC ACX offers an integrated wide-range power supply. Optionally, a **redundant power supply** can be used; the 5V DC desktop power supply is included with this hardware upgrade.

The audio signal is transmitted with PCM16/ PCM24. To reduce the transmission bandwidth, the algorithms **G.722, Opus** and **MPEG Layer 2** can also be activated. For Opus and MPEG Layer 2, a reduced number of 9 respective 10 stereo channels is available.

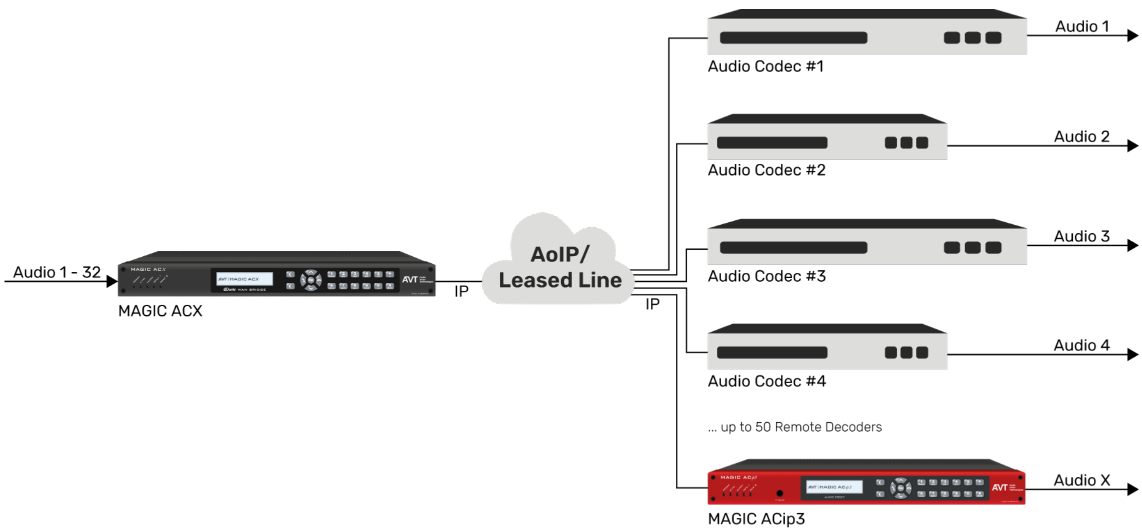
With the optional **Leased Line Distribution Upgrade,** the distribution of one or more audio channels to up to 20 different destinations (depending on the codec used) via *leased lines.*

Alternatively, with the **SIP Distribution Upgrade** it is possible to distribute one or more audio channels to up to 20 different destinations (depending on the codec used) via *SIP dial-up lines.*

Any return signal can be assigned to the available decoders, e.g. for monitoring purposes.

Point-to-Point Connections

Distribution Upgrade



MAGIC ACip3 & MAGIC ACip3 2M Audio Codecs

MAGIC ACip3

IP · E1 · Secure Streaming · GPIO · DHD · Ember+ · 2-Codecs · SD card · Backup · AES67 · Dante

SNMP · redundant · 3 x LAN · QoS · VLAN · Data RS232

G.711 G.722 · Layer 2 · Layer 3 · AAC-LD · AAC-ELD · AAC-LC · HE-AAC V1 · HE-AAC V2 · Opus · Enhanced apt-X · PCM · FLAC

MAGIC ACip3 2M (on request)

IP · E1 · Secure Streaming · GPIO · DHD · Ember+ · 2-Codecs · SD card · Backup · AES67 · Dante

SNMP · redundant · 3 x LAN · QoS · VLAN · Data RS232

G.711 G.722 · Layer 2 · Layer 3 · AAC-LD · AAC-ELD · AAC-LC · HE-AAC V1 · HE-AAC V2 · Opus · Enhanced apt-X · PCM · FLAC

- High-quality Audio transmission with up to 20 kHz
- 1 x analogue and 2 x digital Stereo Audio inputs/outputs or 2 x analogue Stereo Audio inputs/outputs
- Headphones interface
- EBU Tech 3326 compliant (AoIP Standard) and compatible to all VoIP phones
- Simultaneous registration with five SIP servers
- Secure Streaming
- Optional second Stereo Codec
- One independent command channel (G.711/G.722) per Codec
- Optional AES67 Upgrade
- Windows PC software

The **MAGIC ACip3** is a **pure IP Audio Code** and provides three Ethernet interfaces which can be used for Audio over IP transmissions, to control the system with the Windows PC software or to integrate it into a network management system via SNMP. The Audio programmes can – flexibly and freely assignable – be fed in or given out, respectively, via an analogue and two digital stereo interfaces.

MAGIC ACip3 2M is **additionally equipped with an E1 interface** for the classic Audio contribution application via 2-Mbit/s networks.

Both systems support the **G.711, G.722, ISO/MPEG Layer 2**, **Opus and FLAC coding algorithms** and **PCM 16/20/24** Bit in the standard delivery version. Optionally, the Audio Codecs can be upgraded with **Enhanced apt-X 16/24 Bit, AAC-LD/AAC-ELD** and **AAC-LC+V1/V2**.

MAGIC ACip3 and MAGIC ACip3 2M are designed as 19" system with integrated wide area power supply and provide optionally an external redundant power supply.

Two operating modes are available for the **pure IP version**: the system can be used for dial-up AoIP connections according to the EBU Tech 3326 standard or IP Leased Line connections. In AoIP mode, the system can register at 5 different SIP servers and automatically accept incoming calls from this SIP server. Audio connections in IP Leased Line and in AoIP dial-up Mode can be established with the Secure Streaming functionality for a highly reliable transmission.

In case that no connection at all can be established, it is also possible to play an emergency programme from an SD card. With the **Backup Upgrade** a main and a backup connection can be configured.

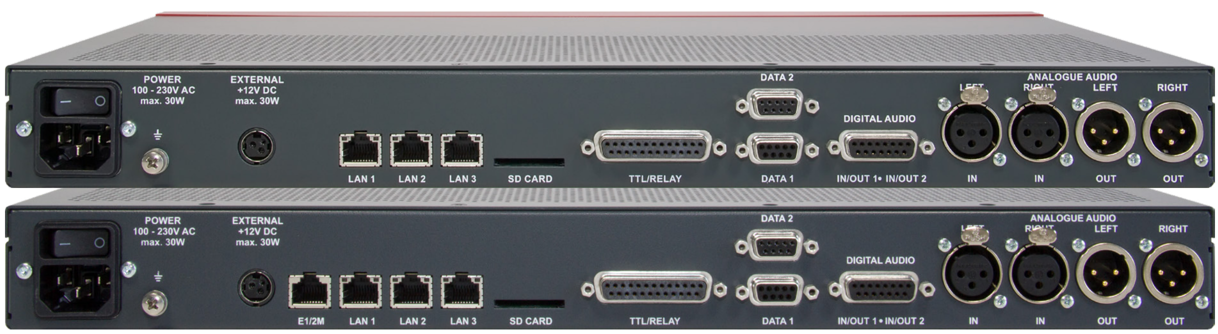
The systems encode one stereo programme in the standard version and can optionally be extended by a second stereo programme through the **2-Codecs Upgrade** (software licence).

In addition to the actual transmission channel, an additional **command channel** is available for each codec, via which an independent connection can be established via G.711 or G.722.

The Audio Codecs can be operated via the front panel or via the Windows PC Software included or via web browser.

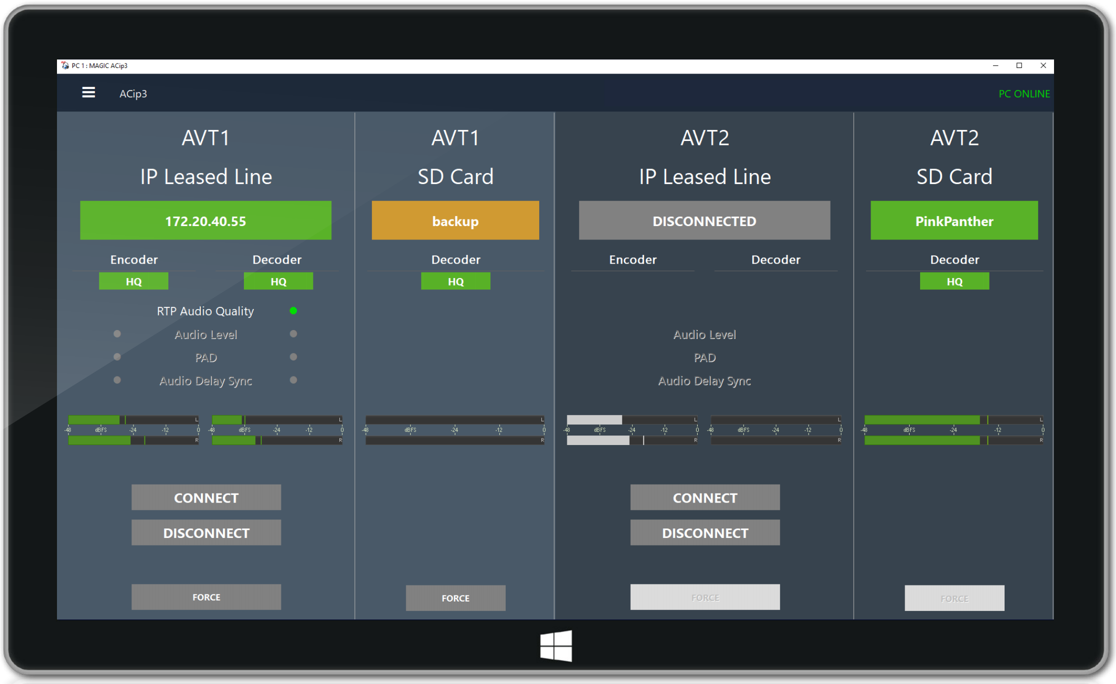
The system also supports the **Ember+** and **DHD SetLogic** protocols. By means of 64 virtual GPIOs, the exchange of control and signalling commands with e.g. mixing consoles is possible.

Furthermore, status information can be retrieved and functions in the unit can be triggered via Ember+.

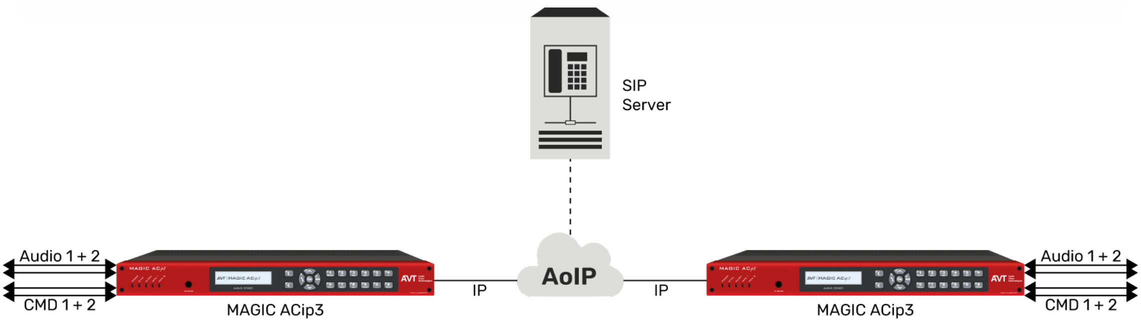


MAGIC ACip3 and MAGIC ACip3 2M

MAGIC ACip3 PC Software

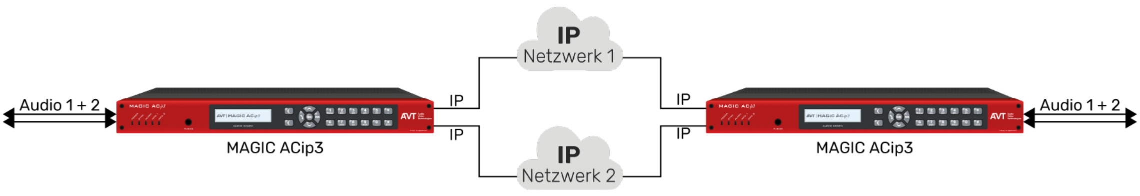


Example Application: Audio Contribution via AoIP (SIP)



In this example you can see the standard application for the MAGIC ACip3. One or optionally two Stereo Audio signals are transmitted in high quality from Studio 1 to Studio 2 and vice versa. Depending on the requirements, the Audio input/output signals can be analogue or digital. The connection can either be established via a common SIP Server at which both Audio Codecs are registered – in this case, the user dials the phone number of the remote codec – or simply by using the IP address of the remote codec for dial-up. **MAGIC ACip3 can store up to five SIP Server accounts which can be selected for dialling out. For incoming calls, the system automatically checks from which SIP Server the call is received**. The MAGIC ACip3 also provides the possibility to use a command channel in parallel to the Audio transmission. This command channel is established in G.711 or G.722 speech quality. If two Stereo Audio signal are transmitted, there are also two command channels available. For improving the transmission quality, the Secure Streaming Mode can be used.

Example Application: Audio Contribution with Secure Streaming



The picture above illustrates an example application when MAGIC ACip3 is used with IP Leased Lines to exchange Audio programmes between two studios. The standard systems can transmit one Stereo signal, but with the 2-Codecs Upgrade, each hardware box can transmit/receive two independent Stereo signals.

A special feature in the IP Leased Lines mode is the Secure Streaming function. With this function two parallel transmissions are established – either via one IP link or also via two separate IP links – to ensure a highly reliable transmission. Optionally, a delay can be configured between the transmissions to prevent that the same packets are lost in both transmissions. In this case, the overall delay increases since a higher buffer is required.

Example Application: Audio Contribution via E1 networks

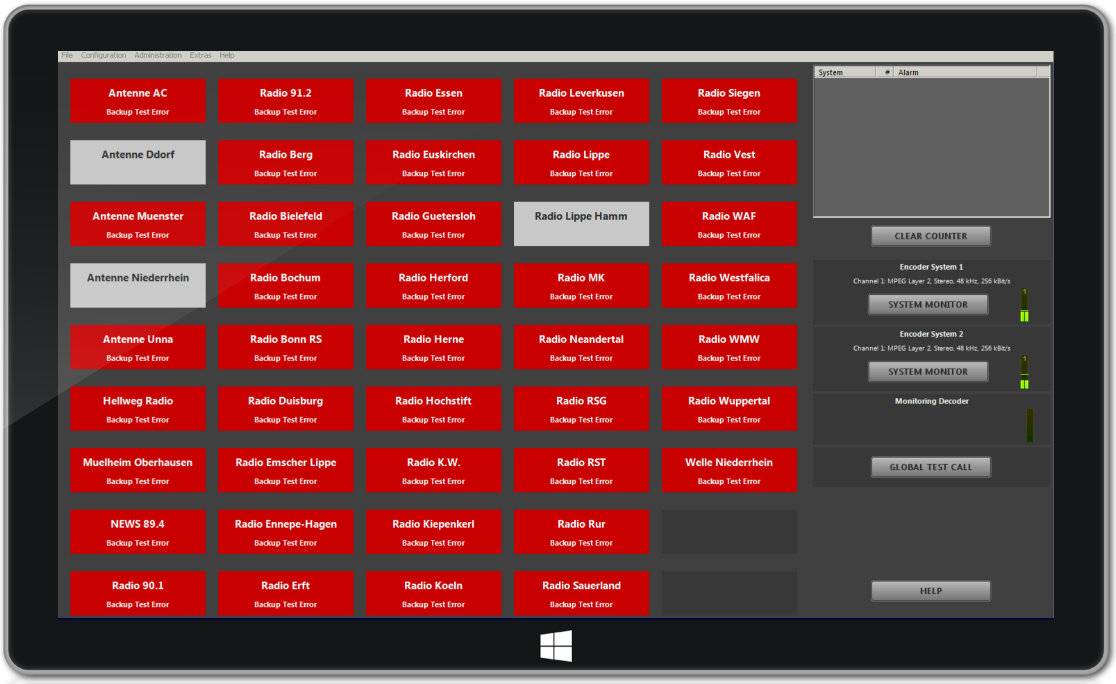
The example shows a typical transmission over an E1 (2-Mbit/s) leased line network with MAGIC ACip3 2M. The big advantage of E1 in comparison to IP networks is the low latency which is not possible via IP. There is no need of buffers, QoS etc. due to the synchronous network structure. Either a bidirectional or unidirectional transmission is possible.

With the optional 2-Codecs upgrade in maximum two stereo Audio signals can be transmitted. All available Audio Codec algorithms can be selected. Also an uncompressed transmission with PCM with lowest delay is possible, however, in this case only one stereo channel can be used because of the required bitrate of 1.5 Mbit/s.

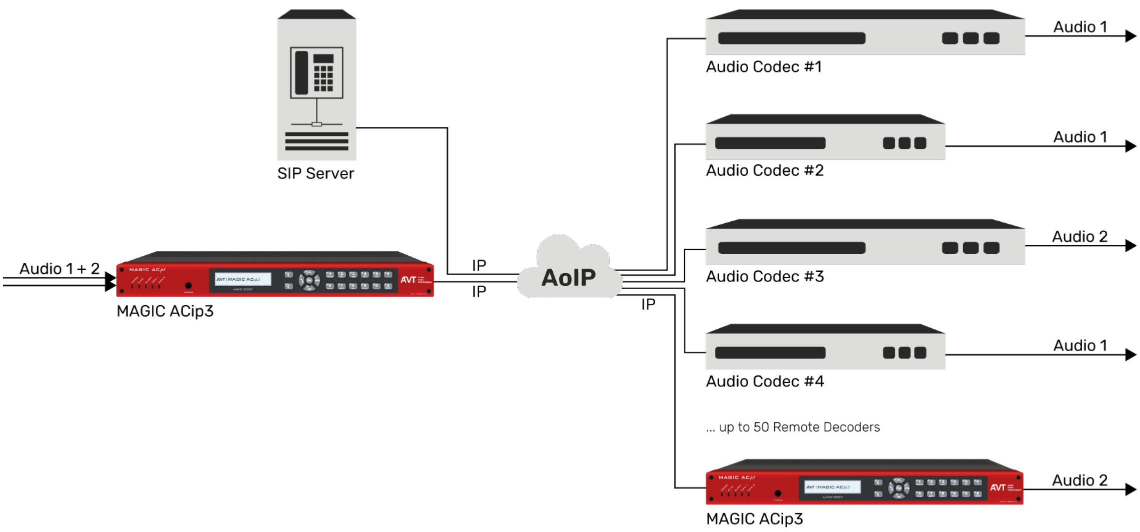
Of course, the 2M versions allow all other standard features available via IP as described above.

If you need to transmit more than two stereo signals via an E1 network, please refer to the also available MAGIC ACip3 2M ModNet System.

On Demand Audio Platform for SIP



Example Application



On Demand Audio Application

Die **On Demand Audio Platform for SIP** is a powerful application for MAGIC ACip3 if one or two stereo Audio signals have to be distributed on demand to different remote studios.

Depending on your needs **up to 50 remote Audio decoders** can be simultaneously connected to the MAGIC ACip3 Distribution system. A comfortable Windows PC software provides a fast overview of the currently existing connections and allows you to comfortably configure the complete system as well as to log possible alarms.

Since the distribution mode works with the Tech 3326 EBU standard (formerly N/ACIP), any Audio Codec which supports this standard can be used. It is important that the Audio coding algorithm which is configured for the distribution is also supported by the remote codec.

With the optional 2-Codecs Upgrade for MAGIC ACip3 you can either offer the same Audio signal with different bitrates (e.g. 128-kbit/s and 64-kbit/s) or with two different independent coding algorithms (e.g. MPEG Layer 2 and OPUS). Even a mix of different Audio coding algorithms and different bitrates is possible.

The system accepts 50 simultaneous dial-up connections via AoIP using a common SIP server. As known from the ISDN world the remote codecs simply call the distribution system via a given number. To avoid unauthorised access to on demand Audio content all allowed remote sites are whitelisted.

With the also available **Backup Upgrade** you can implement an automatic backup solution for e.g. remote studios or transmitters. The remote Audio signal e.g. from a monitoring receiver at a transmitter site can be monitored easily using the PC sound card during a backup connection.

To be sure that your backup system is working without any problems an automatic backup test can be initiated once per day. Alternatively, a manual test call can be triggered to one dedicated site or all sites. Please note that this solution – in comparison to the distribution only mode – requires MAGIC ACip3 or MAGIC AC1 XIP systems also on the remote site.

For highly reliable solutions two MAGIC ACip3 can be used in redundant mode. In case one system fails a remote site in backup mode will be automatically connected to the redundant system.

MAGIC AC1 Go Audio Codec

MAGIC AC1 Go

IP | E1 | Secure Streaming | GPIO | DHD | Ember+ | Backup | AES67 | Dante | 2-Codecs

SNMP | redundant | 1 x LAN | QoS | VLAN | Data RS232 | SD card

G.711 G.722 | Layer 2 | Opus | PCM | FLAC | Layer 3 | AAC-LD | AAC-ELD | AAC-LC | HE-AAC V1 | HE-AAC V2 | Enhanced apt-X

- High quality audio transmission with up to 20 kHz
- 1 x analogue or 1 x digital stereo Audio input/output (switchable)
- EBU Tech 3326 compliant (AoIP standard) and compatible with all VoIP telephones
- Simultaneous registration on five SIP servers
- SIP or leased line mode
- Secure Streaming
- One independent command channel (G.711/G.722)
- Optional: 4 x AES67 channels via software upgrade
- Control via Windows PC software

The **MAGIC AC1 Go** is a **dedicated IP audio codec** and expands our portfolio with a cost effective compact audio codec solution.

The audio-over-IP transmission is done via a high-quality stereo codec. In addition, an independent **command channel** via G.711/G.722 is available, which can also be registered to a different SIP server. To use the command channel, the optional AES67 upgrade is required.

The system can be used for **AoIP dial-up connections** according to the EBU Tech 3326 standard or for **IP leased lines**. In AoIP mode, the system can register to 5 different SIP servers and automatically accept incoming calls from the respective SIP server. For outgoing calls, the respective SIP server can be selected.

For reliable audio connections in IP leased line mode and AoIP dial-up mode, connections can be established with **Secure Streaming**. With Secure Streaming, the stream can be transmitted redundantly to guarantee highly reliable transmission. It is also possible to configure a delay between the two transmissions to avoid failure even in the event of burst errors. The switching between the redundant streams is not audible at the decoder.

The system supports the coding algorithms **G.711, G.722, ISO/MPEG Layer 2, Opus, FLAC** and **PCM 16/20/24** bit in the standard version.

Optionally, the audio codecs can be extended with **MPEG Layer 3, AAC-LD/ AAC-ELD** and **AAC-LC+V1/V2**.

In addition to the regular transmission channel, an additional **command channel** is available, which can be used to establish an independent second connection via G.711 or G.722.

The audio programmes can be supplied or output via an **analogue or digital stereo interface** (switchable).

The unit has an **Ethernet interface** for control and IP audio transmission. In addition, the system supports VLANs.

Two **RS232 interfaces** are available for transmission of additional data and for control purposes.

For signalling purposes, **4 TTL GPIOs** and **2 Relays** can be used.

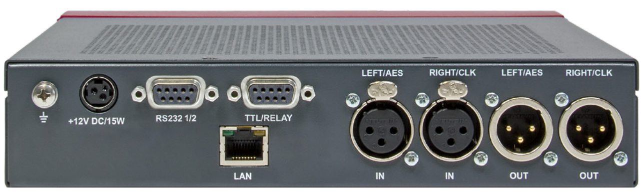
The system also supports the **Ember+** and **DHD SetLogic** protocol. Via 64 virtual GPIOs, the exchange of control and signalling commands with e.g. mixing consoles is possible.

Furthermore, status information can be accessed and functions can be triggered in the unit via Ember+.

The system can be integrated into a network management system via the integrated **SNMP protocol**.

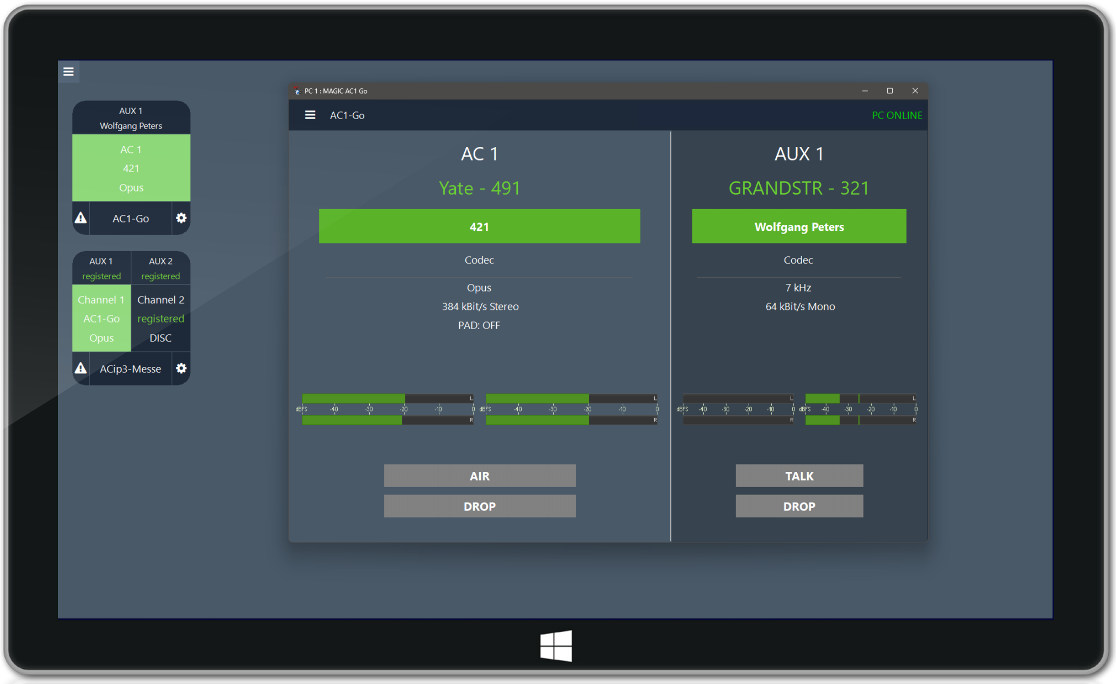
MAGIC AC1 Go is designed as a ½ x 19" system with external power supply.

The audio codec can be controlled via the front keypad with display or with the **Windows PC software** included in the scope of delivery. Multiple systems can also be monitored and operated via the **Multi-Control software**, which is also included.



MAGIC AC1 Go Rear

MAGIC AC1 Go PC-Software



Management & Monitoring

With the convenient **MAGIC AC1 Go Control Software**, up to 5 workplaces can access a system simultaneously.

Using the **Multi-Control Software**, up to 99 MAGIC AC1 Go and MAGIC ACip3 units can be centrally managed, operated and monitored.

The transmit and receive levels of all transmitted and received audio channels are displayed, including general system information such as IP addresses and alarms.

The internal **system log file** allows detailed tracking of errors that have occurred - even without a connected PC. If required, the log file can be downloaded from the system at any time via the management software and displayed clearly in the log file viewer. For exact chronological information in the log file, the system has NTP synchronisation.

Monitoring and alarming can also be carried out via **SNMP**. Traps can be signalled to up to four network management systems.

Options

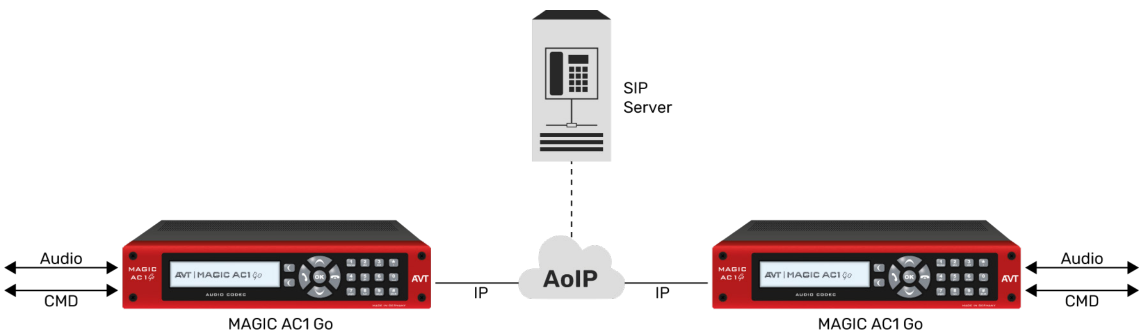
In addition to the included coding algorithms, the system can be upgraded with the **ISO/MPEG Layer 3 Upgrade**, the **MPEG4 AAC-LD/ELD Codec Upgrade** and the **MPEG4 Upgrade**.

The MAGIC AC1 Go can be integrated into AES67 networks with the **AES67 4-Channels Software Upgrade**, which provides compatibility with Dante® or Ravenna networks.

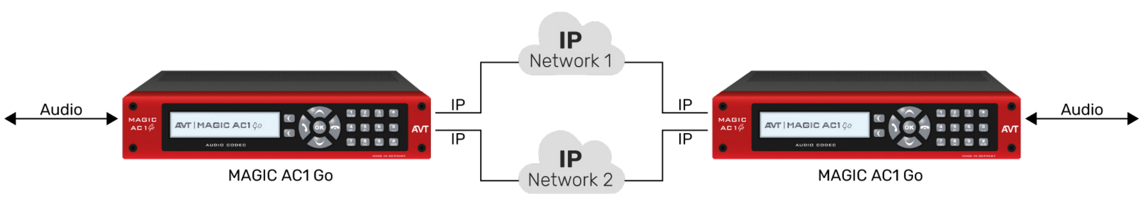
The MAGIC AC1 Go can also be integrated into the **MAGIC System Manager**, which provides a central management platform for all AVT telephone hybrids and audio codecs.

With the **ACConnect** feature of the MAGIC THipPro, the MAGIC AC1 Go can be integrated into the LAN or Screener user interface of the telephone hybrid.

SIP Connection



Leased Line Connection with Secure Streaming



AUDIO CODEC INTEGRAT

MAGIC THipPro ACconnect

MAGIC ACip3

IP	E1	Secure Streaming	GPIO	DHD	Ember+	2-Codecs	SD card	Backup	AES67	Dante
		SNMP	redundant	3 x LAN	QoS	VLAN	Data RS232			
G.711 G.722	Layer 2	Layer 3	AAC-LD	AAC-ELD	AAC-LC	HE-AAC V1	HE-AAC V2	Opus	Enhanced apt-X	PCM

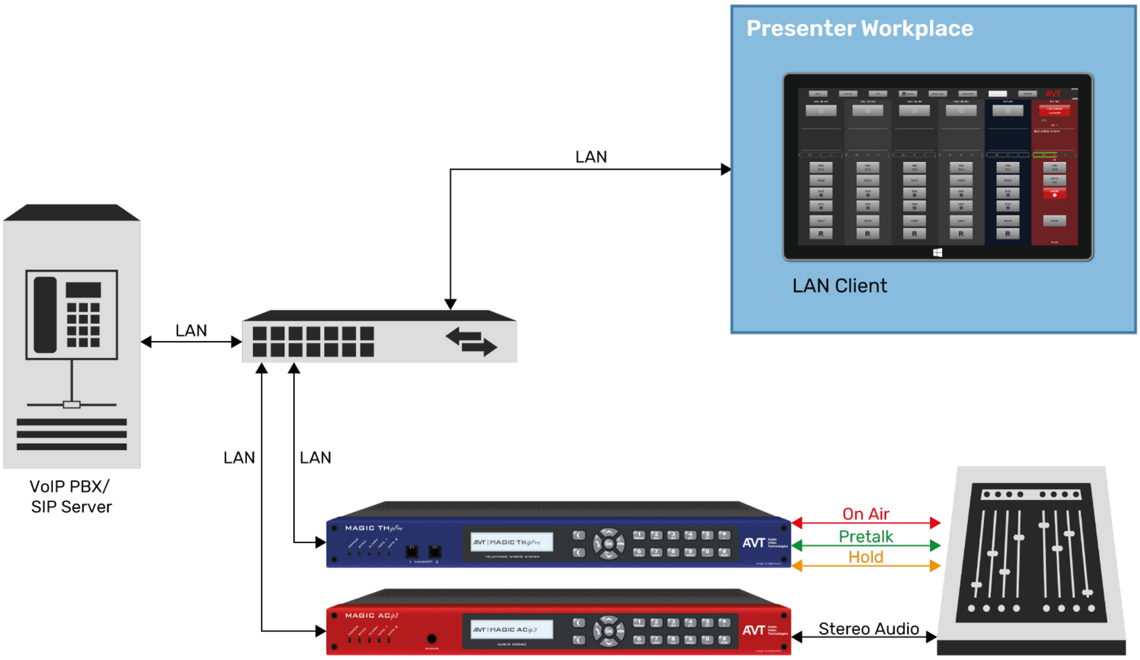
- Full integration in MAGIC THipPro LAN and Screener Software user interface
- Audio Codec control via additional caller line
- Mono/stereo Audio Codec connection
- Pretalk/Hold for Audio Codec
- Common phone book from SQL database
- Audio Codec connection via MAGIC THipPro software upgrade
- Simultaneous registration with five SIP servers with automatic call detection

With the **IP Audio Codec MAGIC ACip3** and the **MAGIC THip-Pro Telephone Hybrid** System AVT provides an integrated **all-in-one solution**: only one Management Software is required to control both Telephone Hybrid System and Audio Codec. The Audio Codec is displayed as an additional caller line in the MAGIC THipPro LAN or Screener Software. Incoming calls to the MAGIC ACip3 can be accepted as well as outgoing Mono or Stereo connections can be established via the hybrid's control software. With the **2-Codecs Upgrade,** two Stereo signals can be transmitted with one MAGIC ACip3 system. In this case, two additional caller lines would be available in the Telephone Hybrid's control software.

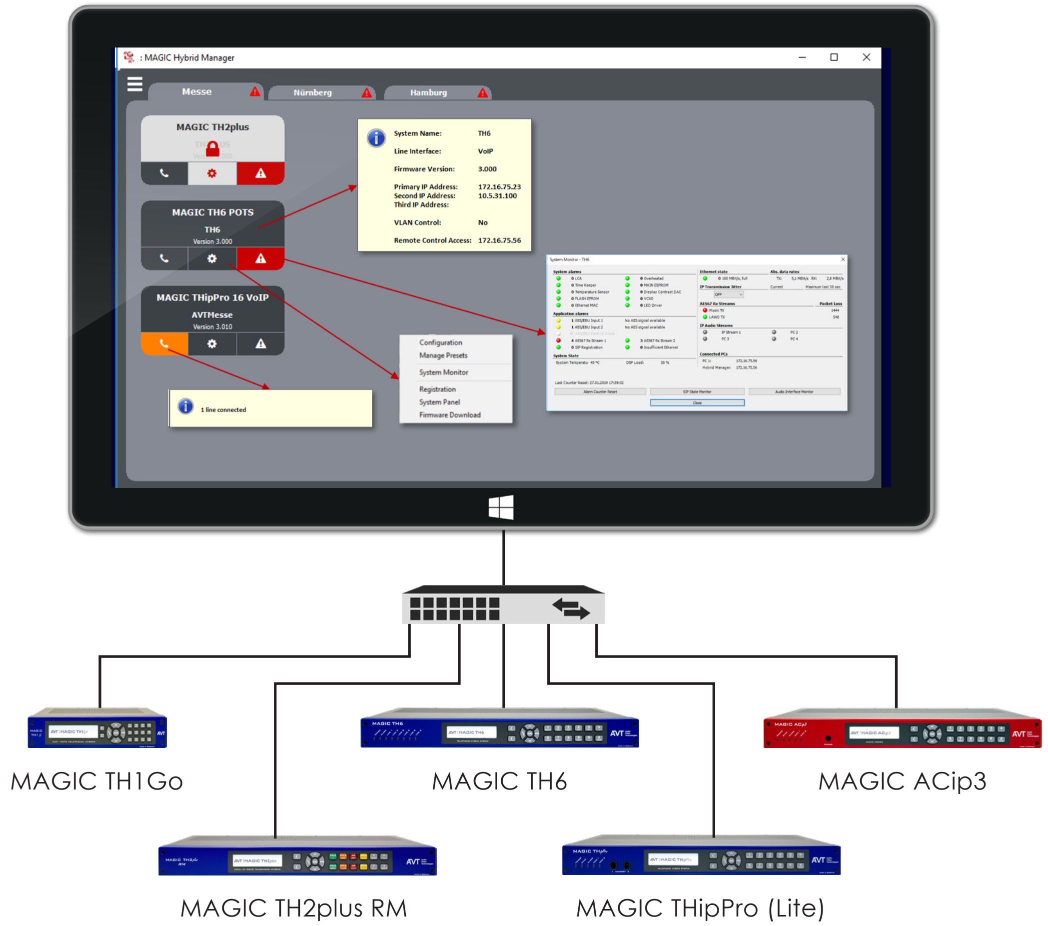
Details of the Audio Codec connection such as e.g. coding algorithm and bit rate are available via the caller line's Info button. Furthermore, a **common phone book** can be used.

If the MAGIC THipPro Telephone Hybrid is used with the **Admin upgrade for up to six studios** the MAGIC ACip3 Audio Codec can be integrated in each of the configured studios - or also only in selected studios. The three available Audio interfaces of the MAGIC ACip3 can be assigned to the different studios or the Audio routing can be done very comfortably via **DHD SetLogic commands** if a **DHD Audio router/matrix** is used. In each studio maximum two MAGIC ACip3 systems can be integrated – if each system has the 2-Codecs Upgrade, you can share maximum four Stereo Audio Codecs in each studio. The system can register at 5 different SIP servers.

In the user interface of the studios the Audio Codec line is displayed as an additional caller line. When a studio is using a shared Audio Codec, this Audio Codec cannot be controlled from the other studios. As soon as the operation is stopped, it is again available for the other studios.



System Manager Upgrade



System Manager Upgrade

- Central management software for AVT telephone hybrids and audio codecs
- Overview of all telephone hybrids and audio codecs of a broadcaster
- Remote configuration of all systems from one administrative location
- Convenient call number and configuration management for dynamic studio assign-ment
- 1 x System Manager license per system

If a broadcasting station has **several AVT telephone hybrids and audio codecs**, these can be displayed in the System Manager. This central management software allows a clear presentation of all MAGIC TH1Go, MAGIC TH2plus, MAGIC TH6 and MAGIC THipPro telephone hybrids and MAGIC ACip3 audio codecs. Even if the systems use **different software versions**, all devices are supported.

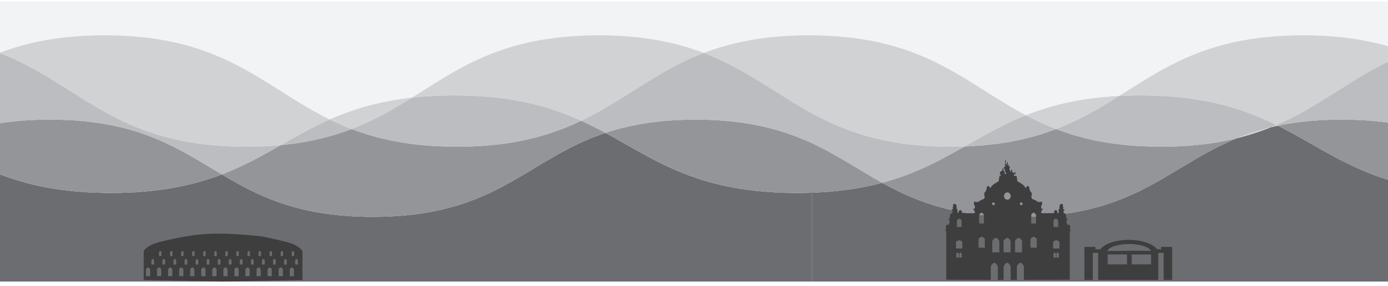
For each device, possible **alarms** and the **operating status** (in use or in configuration) are displayed. The query is made cyclically via SNMP. System Monitor, Registration Dialogue, System Panel and Firmware Download can be accessed.

All systems can also be **configured remotely** from an administrative location. A complete remote configuration of the system is possible, all presets and super presets can be managed. For security reasons, the current status of the line is displayed.

The call number and configuration management allows the simple assignment of call numbers and configurations to a studio at the click of a button. Thus, studio changes can be carried out in the simplest way.

In addition, systems can be selected as desired and new Firmware can then be centrally installed on the corresponding systems. Manual reconfiguration is also possible centrally via the System Manager. Presets can thus be loaded quickly and easily.

One System Manager license is required per system.



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