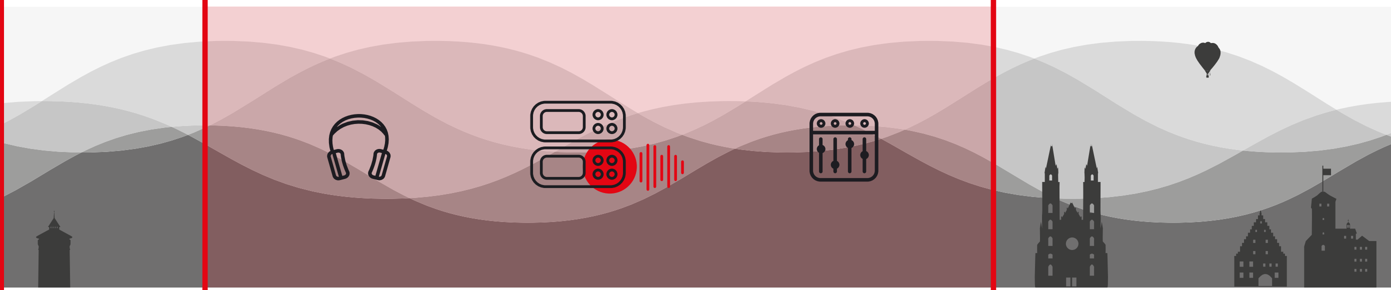
Audio Codecs



[AoIP | Leased Line | E1]

Audio
Video
Technologies

Release date: July 2019

CONTENT

General

Audio Codecs are needed for high-quality Audio transmissions over different networks like **IP, ISDN, 2-Mbit/s (E1)** and **X.21**. Over IP and ISDN, 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.

Although it is planned in various countries that ISDN will be switched off in the future, some broadcasters are still using ISDN in parallel to IP or as backup.

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. At some workplaces it is also required to control more than one system.

The latest addition to the optional features is the **AES67** Software Upgrade which enables the system to route additional Audio channels over IP using the AES67 protocol.

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.

ISDN ISDN interface

The system can be connected to ISDN lines (BRI S0).

X.21 X.21 interface

The system can be connected to X.21 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

For IP Leased Line connections a connection can be established via one or (with MAGIC ACip3) also two IP links, optionally with different delays, whereas packets are transmitted in parallel to ensure 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 a LAWO or DHD mixing console 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 several similar technologies e.g. AES67-compatible Dante and Ravenna devices.

Control via keypad

In addition to a PC, an external keypad can be connected.

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.

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.

Data RS232

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

Data RS485

Transparent data transmission via RS485 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.

Standard apt-X

The system supports the Standard apt-X coding algorithm.

Enhanced apt-X

The system supports the Enhanced apt-X 16 Bit (MAGIC AC1 XIP) or even 24 Bit (MAGIC ACip3) coding algorithm.

PCM

The system supports PCM 16/20/24 Bit uncompressed Audio coding.

	MAGIC DC7 XIP	MAGIC DC7 XIP RM	MAGIC AC1 XIP
Feature			
Line interfaces	1 x ISDN S0 BRI (2 B channels) X.21 LAN	1 x ISDN S0 BRI (2 B channels) X.21 LAN (optional 2 x LAN)	1 x ISDN S0 BRI (2 B channels) X.21 LAN
Audio interfaces	2 x Analogue Digital AES3 Headset/ Microphone	2 x Analogue Digital AES3 Headset/ Microphone	2 x Analogue Digital AES3 Headset/ Microphone
AES67 channels	-	-	-
Coding algorithms	G.711 G.722 PCM 16/20/24	G.711 G.722 PCM 16/20/24	G.711 G.722 PCM 16/20/24 MPEG Layer 2 MPEG Layer 3 Standard apt-X Enhanced apt-X (16 Bit) AAC-LD
2-Codecs-Upgrade	-	-	-
Data rates	64 – 2304-kbit/s (depends on Codec)	64 – 2304-kbit/s (depends on Codec)	32 – 2304-kbit/s (depends on Codec)
Sampling frequencies	8, 16, 24, 32, 48-kHz (depends on Codec)	8, 16, 24, 32, 48-kHz (depends on Codec)	8, 16, 24, 32, 48-kHz (depends on Codec)
Secure Streaming for IP Leased Line Mode	-	-	-
Backup function	-	-	optional
DHD SetLogic/ Ember+ protocol	-	-	-
Control interfaces	RS232 GPIO (6 x TTL, 2 x Relays) LAN	RS232 GPIO (6 x TTL, 2 x Relays) LAN (optional 2 x LAN)	RS232 GPIO (6 x TTL, 2 x Relays) LAN
Dimensions	1/2 x 19", 1U	19", 1U	1/2 x 19", 1U
Power Supply	external 12V + redundancy (optional)	100 – 230 V	external 12V + redundancy (optional)

MAGIC AC1 XIP RM	MAGIC ACip3	MAGIC ACip3 2M
1 x ISDN S0 BRI (2 B channels) X.21 LAN (optional 2 x LAN)	3 x LAN	3 x LAN 1 x E1
2 x Analogue Digital AES3 Headset/ Microphone	1 x Analogue 2 x Digital AES3 OR: 2 x Analogue Headphone	1 x Analogue 2 x Digital AES3 OR: 2 x Analogue Headphone
-	4/6 (optional)	4/6 (optional)
G.711 G.722 PCM 16/20/24 MPEG Layer 2 MPEG Layer 3 Standard apt-X Enhanced apt-X (16 Bit) AAC-LD	G.711 G.722 PCM 16/20/24 MPEG Layer 2 MPEG Layer 3 Enhanced apt-X (24 Bit) AAC-LD AAC-ELD AAC-LC HE-AAC V1 HE-AAC V2 Opus	G.711 G.722 PCM 16/20/24 MPEG Layer 2 MPEG Layer 3 Enhanced apt-X (24 Bit) AAC-LD AAC-ELD AAC-LC HE-AAC V1 HE-AAC V2 Opus
-	yes	yes
32 – 2304-kbit/s (depends on Codec)	8 – 2304 kbit/s (depends on Codec)	8 – 2304 kbit/s (depends on Codec)
8, 16, 24, 32, 48-kHz (depends on Codec)	8, 16, 24, 32, 48-kHz (depends on Codec)	8, 16, 24, 32, 48-kHz (depends on Codec)
-	yes	yes
optional	optional	optional
-	yes	yes
RS232 GPIO (6 x TTL, 2 x Relays) LAN (optional 2 x LAN)	GPIO (6 x TTL, 6 x Relays) 3 x LAN	GPIO (6 x TTL, 6 x Relays) 3 x LAN
19", 1U	19", 1U	19", 1U
100 – 230 V	100 – 230 V + external 12 V	100 – 230 V + external 12 V

MAGIC DC7 XIP & MAGIC DC7 XIP RM Audio Codecs

MAGIC DC7 XIP

IP	ISDN	X.21	E1	Secure Streaming	GPIO	DHD	Ember+	2-Codecs	SD card	Backup	AES67
		SNMP	redundant	1 x LAN	QoS	Data RS232	Data RS485				
G.711 G.722	Layer 2	Layer 3	AAC-LD	AAC-ELD	AAC-LC	HE-AAC V1	HE-AAC V2	Opus	Standard apt-X	Enhanced apt-X	PCM

MAGIC DC7 XIP RM

IP	ISDN	X.21	E1	Secure Streaming	GPIO	DHD	Ember+	2-Codecs	SD card	Backup	AES67
		SNMP	redundant	1 x LAN	QoS	Data RS232	Data RS485	2nd LAN			
G.711 G.722	Layer 2	Layer 3	AAC-LD	AAC-ELD	AAC-LC	HE-AAC V1	HE-AAC V2	Opus	Standard apt-X	Enhanced apt-X	PCM

- G.711, G.722 and PCM transmission
- 2 x Mono analogue or optionally 1 x Stereo digital Audio input/output (switchable)
- Handset/Headset interface
- Automatic Mode for incoming calls via VoIP (LAN/SIP) or ISDN
- Compatible to all VoIP telephones and EBU Tech 3326 compliant (AoIP Standard)
- Control via front panel or external keypad
- Multi-control Windows PC software
- Optional Software Upgrade Kit available

The **MAGIC DC7 XIP Audio Code** provides **a LAN, an ISDN BRI** and **an X.21 interface**. The system has two analogue Audio inputs/outputs which can be optionally also used as digital AES3 interfaces. The **coding algorithms G.711 (3.1-kHz)** and **G.722 (7-kHz)** are implemented as well as **PCM 16/20/24**.

There are two versions of the MAGIC DC7 XIP available. The **compact MAGIC DC7 XIP Audio Codec** in a **½ x 19"** housing with external power supply is a powerful product for mobile applications but can be also used as space-saving solution in the studio. The **full 19" version MAGIC DC7 XIP RM** has an integrated power supply which is sometimes preferred in studio installations. Besides that, it can be equipped with a second LAN interface to separate VoIP and control physically from each other.

Both via ISDN and via IP, dial-up connections as well as Leased Lines can be used with MAGIC DC7 XIP.

In the ISDN mode, two separate 7kHz connections can be established (one connection per ISDN B-channel).

A very handy feature of the MAGIC DC7 XIP is the so-called **ISDN/SIP Auto Mode**. Using this mode, the system detects automatically via which line interface (ISDN or LAN) a call is received.

Complete system operation is possible via the front panel. The intuitive user interface guarantees a user-friendly operation. In the integrated telephone book, you can store a transmission mode together with each telephone number.

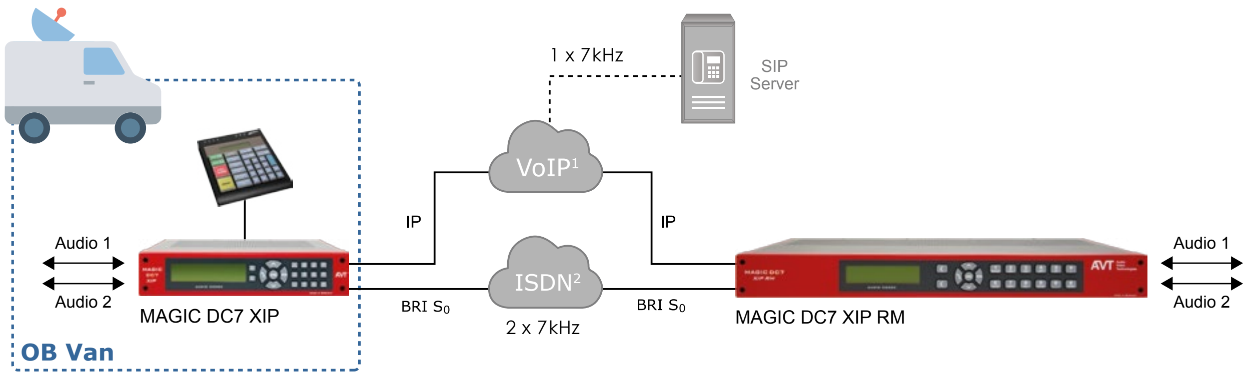
Alternatively, the system can be controlled with the MAGIC DC7 XIP Windows PC software or the MAGIC DC7 XIP LAN Windows PC software. The LAN software allows you to simultaneous monitor up to 40 Audio Codecs within a network. Individual systems can be dynamically selected for Audio transmissions. Both software versions are included in the standard delivery.

MAGIC DC7 XIP

MAGIC DC7 XIP RM

MAGIC DC7 XIP PC Software

Example Application: Audio Contribution via VoIP (SIP) or ISDN



1) 1 x Audio via VoIP 2) 2 x Audio only via ISDN

Options

The basic MAGIC DC7 XIP system can be upgraded with various options.

If you want to use the **digital AES/EBU interface** of the system, you need to activate the corresponding software licence. The Audio inputs and Audio outputs can then be used independently as analogue or digital Audio lines.

The AES/EBU interface licence is also available in a special package together with SNMP, to integrate the system into a network management system. On this way, all selected alarm and status information can be sent to three different network management systems. Also included is the Mixer Tool Software Plug-In, which allows you to mix all incoming and outgoing signals freely, and the ISDN Monitor Plug-In for a detailed logging and analysis of your ISDN connection. This package is called **Software Upgrade Kit** and is available at a special discounted price.

For remote operation without PC, the external **MAGIC Keypad Basic** is available. The keypad supports the most important functions of the Audio Codec and is connected via the RS232 interface.

For the MAGIC DC7 XIP we offer a **redundant power supply** which is connected via an external box to the system.

For easy storage or transport of the MAGIC DC7 XIP a **portable case** can be purchased.

19” Mounting brackets are included in the standard delivery for both MAGIC DC7 XIP and MAGIC DC7 XIP RM. To mount two MAGIC DC7 XIPs next to each other in a rack with only 1U in height, the **Dual 19” Mounting Kit** is optionally available.

Last but not least we can integrate a **second LAN interface** into the housing of the MAGIC DC7 XIP RM so that the VoIP stream and the control stream can be separated physically from each other. The second LAN interface can be used for control or SNMP but not for the Voice over IP connection.

MAGIC DC7 XIP Dual 19” Mounting Kit

Portable case

External redundant power supply

MAGIC AC1 XIP & MAGIC AC1 XIP RM Audio Codecs

MAGIC AC1 XIP

IP	ISDN	X.21	E1	Secure Streaming	GPIO	DHD	Ember+	2-Codecs	SD card	Backup	AES67
		SNMP	redundant	1 x LAN	QoS	Data RS232	Data RS485				
G.711 G.722	Layer 2	Layer 3	AAC-LD	AAC-ELD	AAC-LC	HE-AAC V1	HE-AAC V2	Opus	Standard apt-X	Enhanced apt-X	PCM

MAGIC AC1 XIP RM

IP	ISDN	X.21	E1	Secure Streaming	GPIO	DHD	Ember+	2-Codecs	SD card	Backup	AES67
		SNMP	redundant	1 x LAN	QoS	Data RS232	Data RS485	2nd LAN			
G.711 G.722	Layer 2	Layer 3	AAC-LD	AAC-ELD	AAC-LC	HE-AAC V1	HE-AAC V2	Opus	Standard apt-X	Enhanced apt-X	PCM

- High-quality Audio transmission with up to 20 kHz
- 1 x Stereo analogue or optionally 1 x Stereo digital Audio input/output (switchable)
- Handset/Headset interface
- Automatic Mode for incoming calls via AoIP (LAN/SIP) or ISDN
- EBU Tech 3326 compliant (AoIP Standard) and compatible to all VoIP phones
- Control via front panel or external keypad
- Multi-control Windows PC software
- Optional Software Upgrade Kit available
- Virtual subnetworks (VLANs) are supported
- Secure Streaming for IP Leased Lines Mode

The **MAGIC AC1 XIP Audio Codec** provides **a LAN, an ISDN BRI** and **an X.21 interface**. The system has two analogue Audio inputs/outputs (1 x Stereo) which can be optionally also used as digital AES3 interfaces. In the standard delivery version, the MAGIC AC1 XIP Audio Codecs supports the **G.711 (3.1-kHz), G.722 (7-kHz)** and **ISO/MPEG Layer 2** coding algorithms as well as the PCM transmission mode. Further coding algorithms such as **ISO/MPEG Layer 3, Standard apt-X**, **Enhanced apt-X** and **AAC-LD** can be activated optionally.

There are two hardware versions of the MAGIC AC1 XIP available. The compact **MAGIC AC1 XIP** Audio Codec in a ½ x 19" housing with external power supply is a powerful product for mobile use or OB Vans. The **full 19" version MAGIC AC1 XIP RM** has an integrated power supply which is sometimes preferred in studio installations. Besides that, it can be equipped with a **second LAN interface** to separate AoIP and control physically from each other.

Both versions are also available as **Decoder only**, e.g. if there is no need of a Codec on the remote site because the Audio transmission is only unidirectional.

The MAGIC AC1 XIP also provide the so-called **ISDN/SIP Auto Mode**. Using this mode, the system detects automatically via which line interface (ISDN or LAN) a call is received.

In addition to the ISDN and LAN/SIP dial-up modes, the systems can also work in the ISDN Leased Lines and the IP Leased Lines operating modes if permanent connections are used.

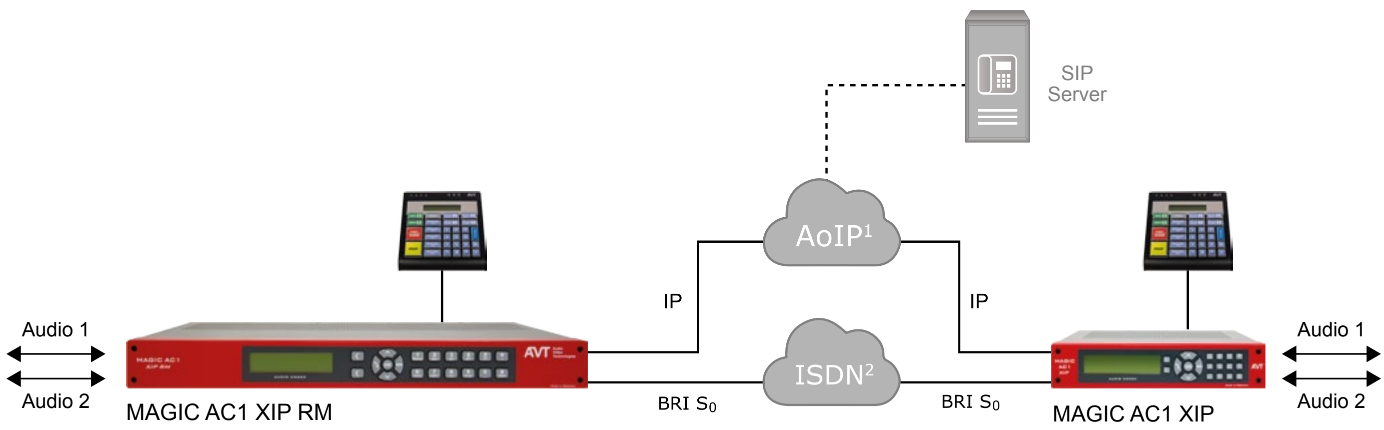
MAGIC AC1 XIP can be operated via the front keypad and display. Alternatively, a comfortable configuration and control is possible with the supplied Windows PC Software or the MAGIC AC1 XIP LAN multi-control software. With this software, up to 40 Audio Codecs can be monitored simultaneously. Individual systems can be selected for control and configuration. Both software versions are included in the delivery.

MAGIC AC1 XIP

MAGIC AC1 XIP RM

MAGIC AC1 XIP PC Software

Example Application: Audio Contribution via VoIP (SIP) or ISDN



[1] 1 x Audio via VoIP

[2] 2 x Audio only via ISDN

Options

As already mentioned, the basic system comes with the G.711, G.722, ISO/MPEG Layer 2 **coding algorithms** and PCM 16/20/24 Bit, the system can be extended with AAC-LD, ISO/MPEG Layer 3, Standard and Enhanced apt-X 16 Bit.

Furthermore, you can activate the digital **AES/EBU interfaces** via software licence, if you want to use the digital Audio lines. The Audio inputs and outputs can then be switched independently to analogue or digital.

The AES/EBU interface licence is also available in a special package together with the Mixer Tool Software Plug-In, which allows you to mix all incoming and outgoing signals freely, and the ISDN Monitor Plug-In for a detailed logging and analysis of your ISDN connection. Also included is the SNMP protocol to integrate the MAGIC AC1 XIP into a network management system and the ISO/MPEG Layer 3 coding algorithm on top. This package is called **Software Upgrade Kit** and is available at a special discounted price.

With the **Backup Upgrade** an automatic ISDN Backup Function is optionally available. If the main IP connection between two MAGIC Codecs drops out during an Audio transmission, the systems switch automatically to the ISDN interface and the codec which has been selected as „Master“ establishes an ISDN Backup connection with the parameters that have been configured in the software. Alternatively, if the backup is not supposed to be triggered by selected parameters of the IP connection, it can be started by an external TTL contact.

For remote operation without PC, the external **MAGIC Keypad Basic** is available. The keypad supports the most important functions of the Audio Codec and is connected via the RS232 interface.

For the MAGIC AC1 XIP we offer a **redundant power supply** which is connected via an external box to the system.

For easy storage or transport of the MAGIC AC1 XIP a **portable case** can be purchased.

19” Mounting brackets are included in the standard delivery for both MAGIC DC7 XIP and MAGIC AC1 XIP RM. To mount two MAGIC AC1 XIPs next to each other in a rack with only 1U in height, the **Dual 19” Mounting Kit** is optionally available.

Last but not least we can integrate a **second LAN** interface into the housing of the MAGIC AC1 XIP RM so that the AoIP stream and the control stream can be separated physically from each other.

MAGIC Keypad Basic

MAGIC ACip3 & MAGIC ACip3 2M Audio Codecs

MAGIC ACip3

IP | ISDN | X.21 | E1 | Secure Streaming | GPIO | DHD | Ember+ | 2-Codecs | SD card | Backup | AES67

SNMP | redundant | 3 x LAN | QoS | Data RS232 | Data RS485

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

MAGIC ACip3 2M

IP | ISDN | X.21 | E1 | Secure Streaming | GPIO | DHD | Ember+ | 2-Codecs | SD card | Backup | AES67

SNMP | redundant | 3 x LAN | QoS | Data RS232 | Data RS485

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

- 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 for monitoring
- EBU Tech 3326 compliant (AoIP Standard) and compatible to all VoIP phones
- Simultaneous registration with five SIP servers with automatic call detection
- Secure Streaming for IP Leased Lines Mode
- Can be upgraded to two Stereo Codecs
- One independent command channel (G.711/G.722) per Codec
- Optional 4 x AES67 channels via software upgrade (or even 6 x AES67 with 2-Codecs Upgrade)
- Control via front panel
- 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 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 upgraded to a **second stereo programme** by **software activation**.

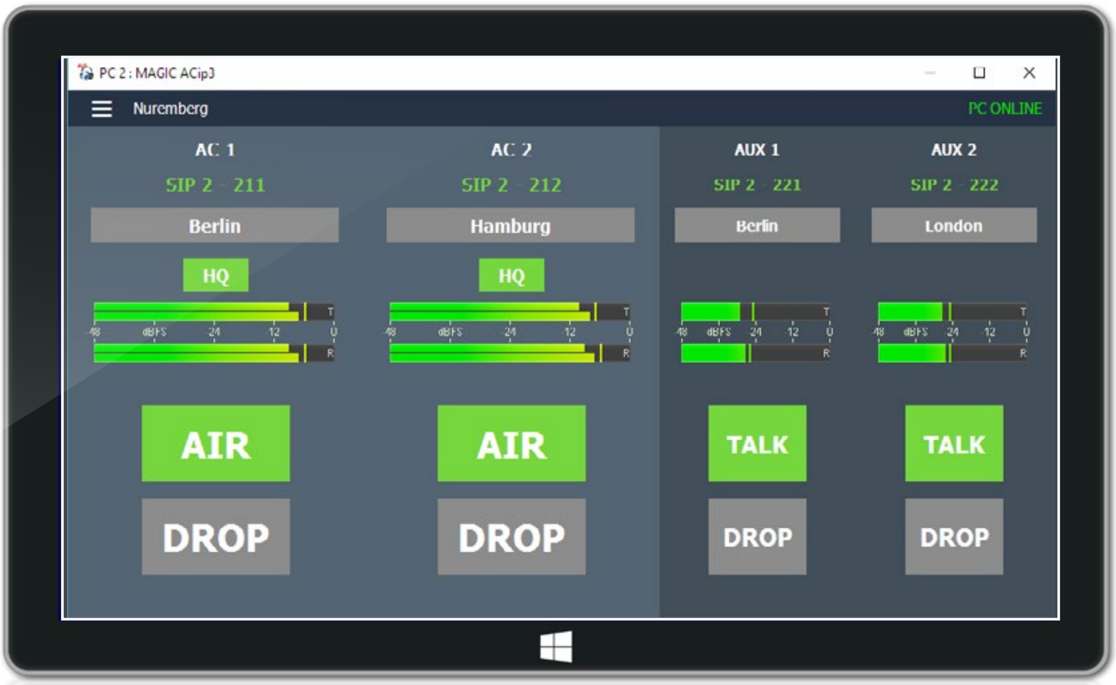
The Audio Codecs can be operated via the front panel or even more comfortably via the Windows PC Software supplied with the delivery.

Via the **Ember+ protocol 64 inputs** and **64 outputs** can be programmed, an easy communication between MAGIC ACip3 and e.g. DHD or Lawo mixing consoles is possible. For a simplified programming there is the **Ember+ Consumer Extension** upgrade.

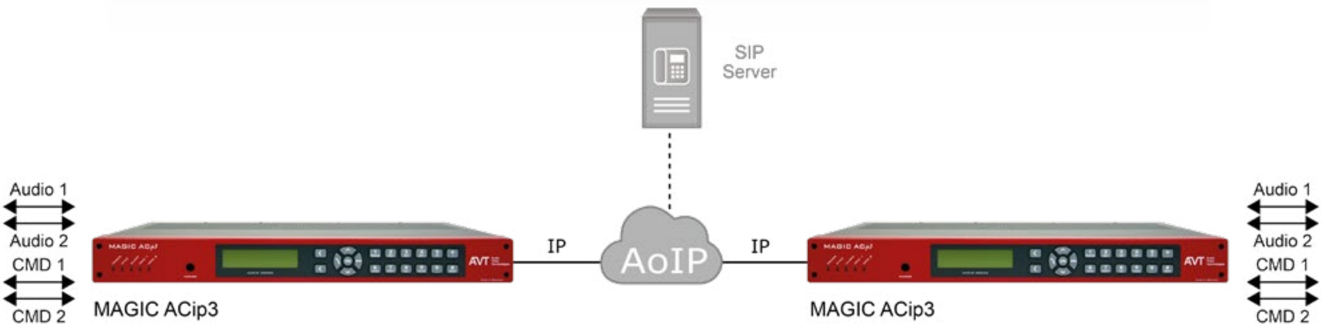
MAGIC ACip3

MAGIC ACip3 2M

MAGIC ACip3 PC Software

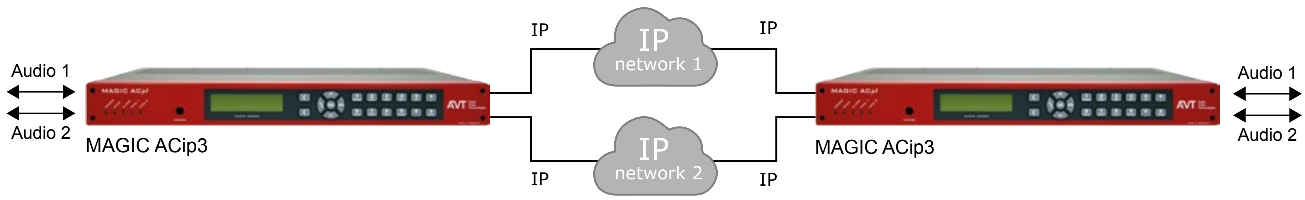


Example Application: Audio Contribution via AoIP (SIP)



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.

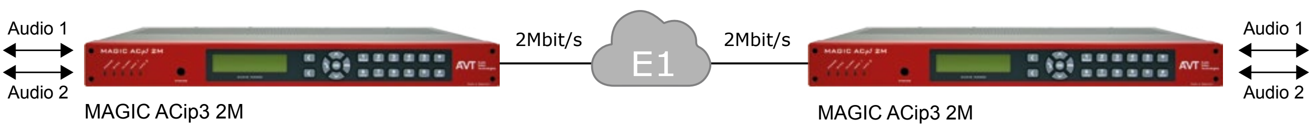
Example Application: Audio Contribution with Secure Streaming



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



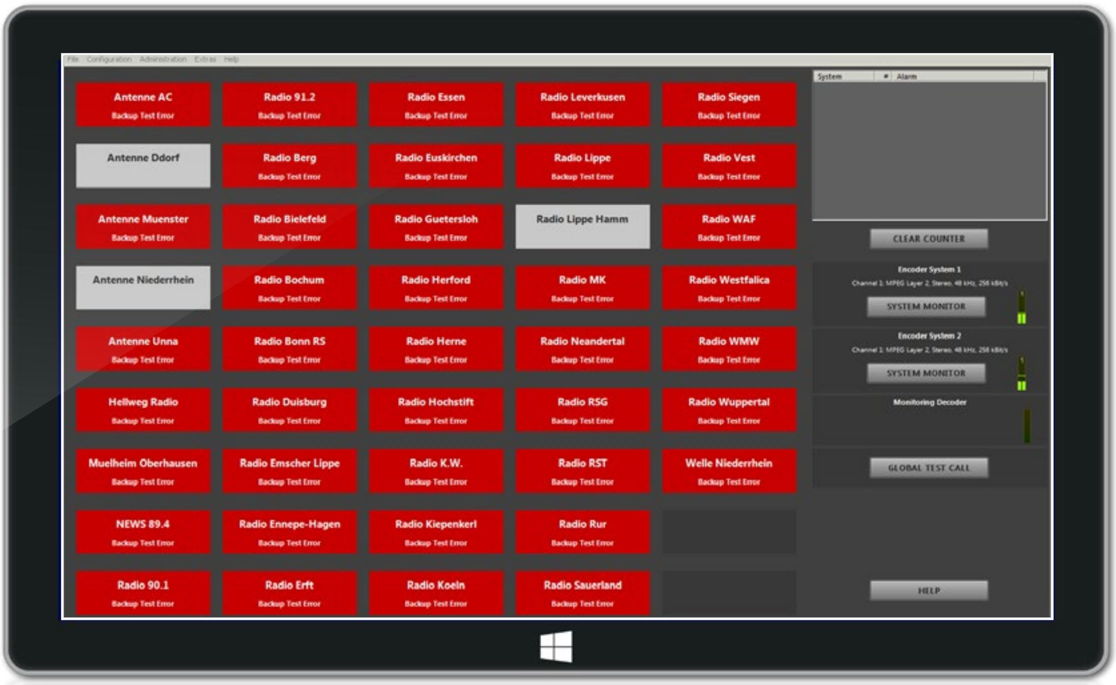
The example shows a typical transmission over an E1 (2-Mbit/s) leased line network with MAGIC ACip3 2M. The big advantage of E1 in comparison to IP networks is the low latency which is not possible via IP. There is no need of buffers, QoS etc. due to the synchronous network structure. Either a bidirectional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

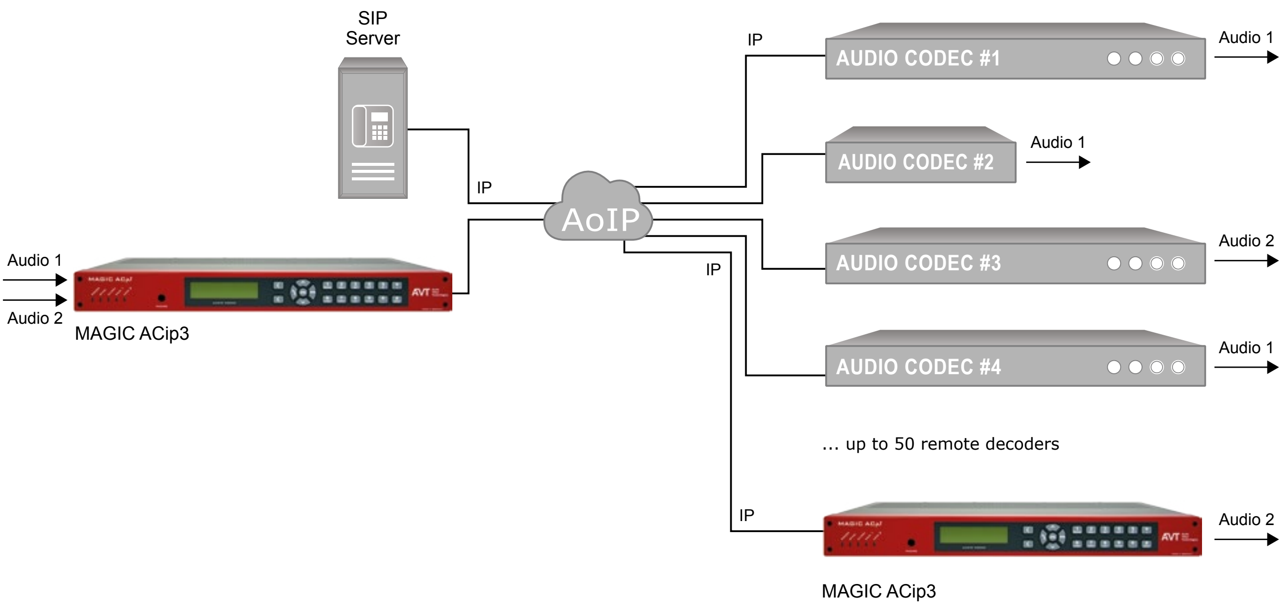
Of course, the 2M versions allow all other standard features available via IP as described above.

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.

MAGIC ACip3 PC Distribution Software



Example Application: AoIP Distribution



Distribution Software

The **Distribution Upgrade** 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 **10, 20 or 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 – depending on the selected upgrade – 10, 20 or 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 ACip3 (2M) ModNet System

MAGIC ACip3 (2M) ModNet

IP	ISDN	X.21	E1	Secure Streaming	GPIO	DHD	Ember+	2-Codecs	SD card	Backup	AES67
		SNMP	redundant	3 x LAN	QoS	Data RS232	Data RS485				
G.711 G.722	Layer 2	Layer 3	AAC-LD	AAC-ELD	AAC-LC	HE-AAC V1	HE-AAC V2	Opus	Standard apt-X	Enhanced apt-X	PCM

- 1 x analogue and 2 x digital Audio inputs/outputs (switchable)
- Headphones interface
- Drop & Insert function (2M only)
- Audio monitoring via IP (G.722)
- ModNet Management Software for multi-system control
- Multi Stereo Signal Transmission
- Transparent data channel for e.g. RDS

For Studio-Transmitter-Links (STL) a special variant of the MAGIC ACip3 (2M) system is available which is called **ModNet System**. The highly reliable DSP-based hardware platform with an optional redundant power supply is identical to the standard MAGIC ACip3 (2M) system, but works with an optimised firmware and provides a comfortable Windows PC software for STL applications.

The system can be equipped with one E1 (2-Mbit/s) interface and/or three independent Ethernet interfaces which can be used for the main Audio signal distribution, for backup and also for control and monitoring.

In the standard version one stereo ISO/MPEG Layer 2 Audio codec is available. With the optional 2-Codecs Upgrade two independent stereo Audio signals can be transmitted with one system.

If low latency is required, the system can be upgraded with the optional Enhanced apt-X 16/24-bit Audio codec algorithm. Using the 24-bit format the highest quality for compressed Audio transmission with the lowest delay is possible.

Via the E1 interface, the MAGIC ACip3 2M version allows a transmission of multiple stereo or mono Audio signals depending on the selected bitrate. Further Audio codecs can be connected to the main system via E1 using the Drop & Insert feature (which offers lowest latency) or alternatively via IP.

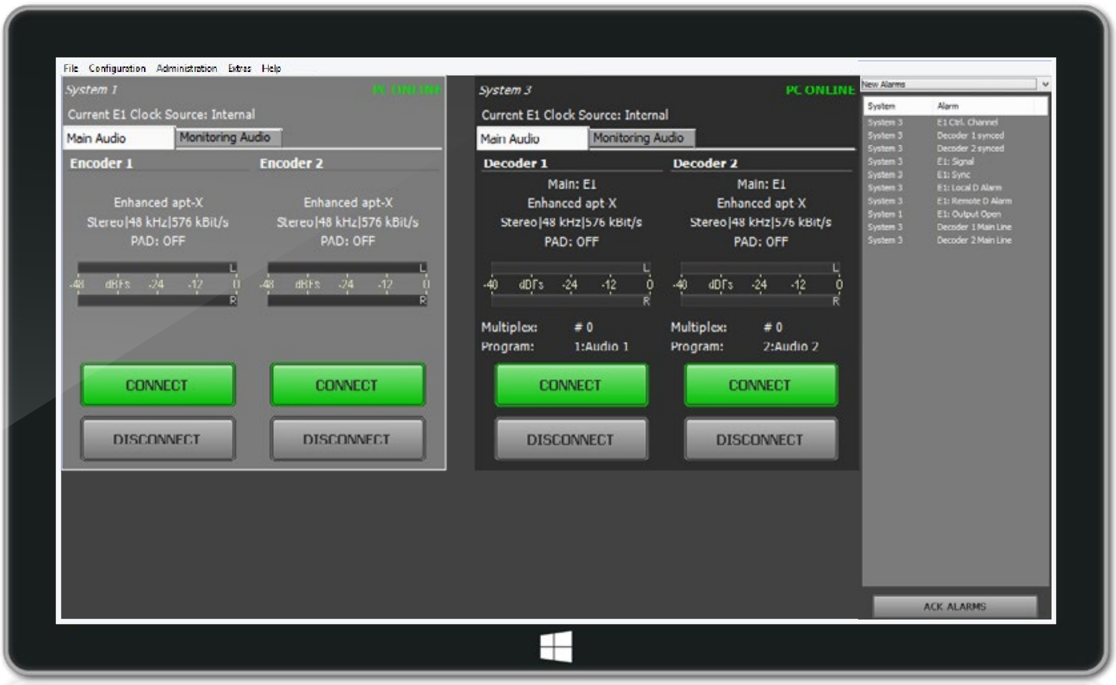
In the IP, only MAGIC ACip3 version each encoder can transmit to up to five remote sites simultaneously.

In maximum one network can consist of 99 systems which can be managed in the ModNet application.

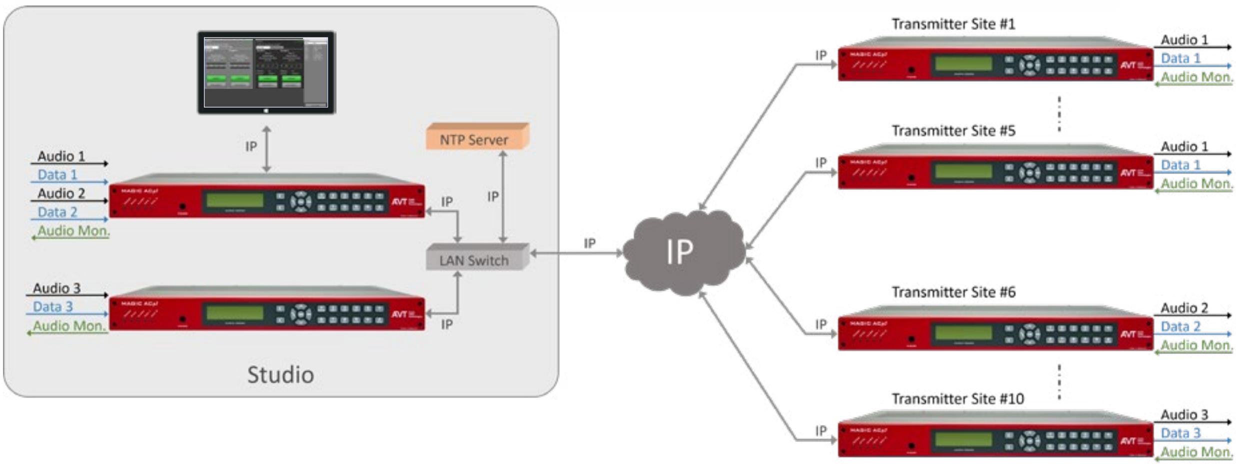
With the optional **Backup upgrade**, you can secure the remote sites with an automatic backup functionality if one network fails. You can assign E1 as main distribution signal and IP as backup network or the other way around. Also possible is a pure IP transmission via two independent IP networks.
No matter what is configured, if the main and the backup networks fail you still have the possibility to play out an Audio signal from an SD card at the transmitter site.

Besides the possibility to monitor the complete distribution network via the ModNet software application, there is also the option to use SNMP for integrating the system into a Network Management System. However, the best way to be sure that the Audio signal on the remote site is good, is to listen into the Audio signal using the Audio monitoring feature via IP.

MAGIC ACip3 (2M) ModNet Management PC Software



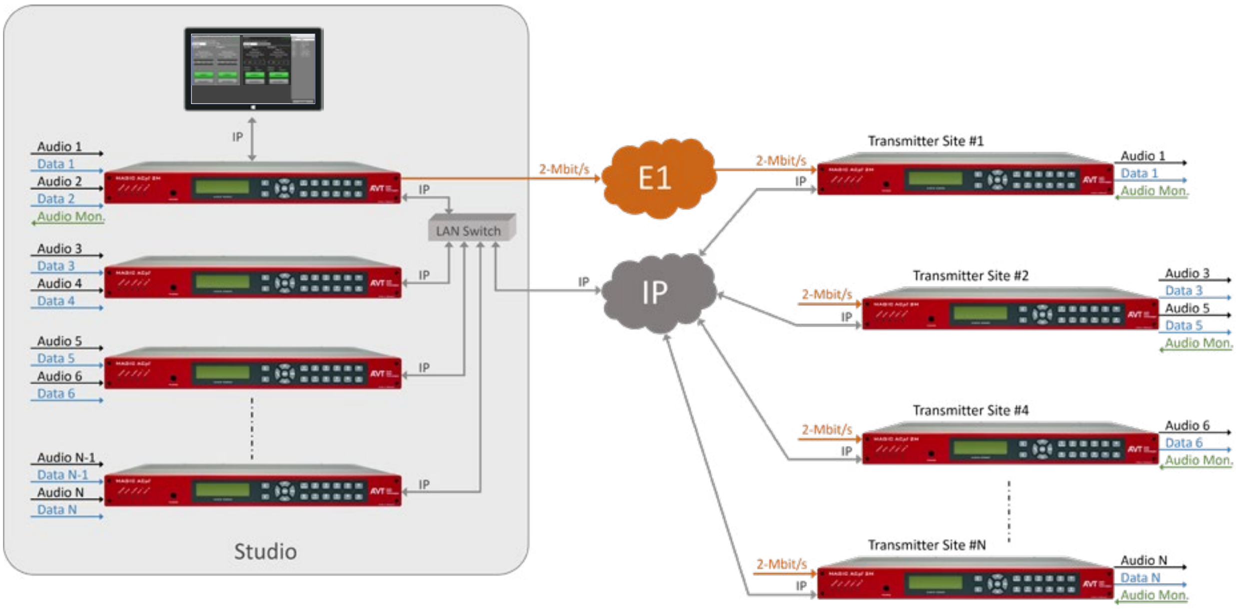
Example Application: ModNet – STL via IP



The example illustrates an STL Audio signal distribution via IP. Each MAGIC ACip3 system in the studio can distribute one or optional two stereo Audio signals to up to five remote sites via a unicast communication, which is easier to realise than a multicast network.

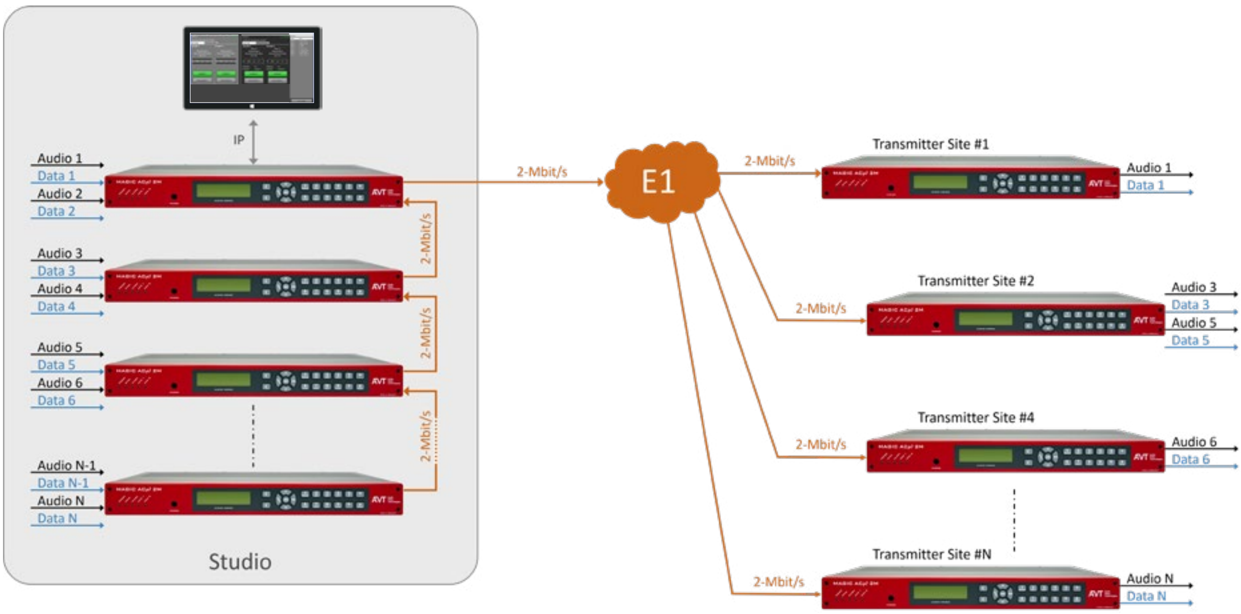
Using the NTP Sync feature a synchronous Audio transmission at the transmitter output is possible. Remote Audio monitoring as well as an optional automatic backup can be easily configured.

Example Application: ModNet – STL via E1 with IP backup



The drawing above shows an E1 STL transmission with an automatic backup via IP. The MAGIC ACip3 2M main system which is connected to the E1 network can transmit multiple stereo Audio signals via further Audio Codecs. These codecs are simply connected via IP to the main system. Of course, remote Audio monitoring is also possible.

Example Application: ModNet – STL via E1



This application describes a multiple stereo Audio transmission via a pure E1 network. The main MAGIC ACip3 2M system collects further Audio signals from other codecs using the 2-Mbit/s Drop & Insert feature which offers the lowest possible latency. Because of the missing return channel Audio monitoring is not supported.

MAGIC THipPro ACconnect

MAGIC ACip3

IP · ISDN · X.21 · E1 · Secure Streaming · GPIO · DHD · Ember+ · 2-Codecs · SD card · Backup · AES67

SNMP · redundant · 3 x LAN · QoS · Data RS232 · Data RS485

G.711 G.722 · Layer 2 · Layer 3 · AAC-LD · AAC-ELD · AAC-LC · HE-AAC V1 · HE-AAC V2 · Opus · Standard apt-X · 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 detec-tion

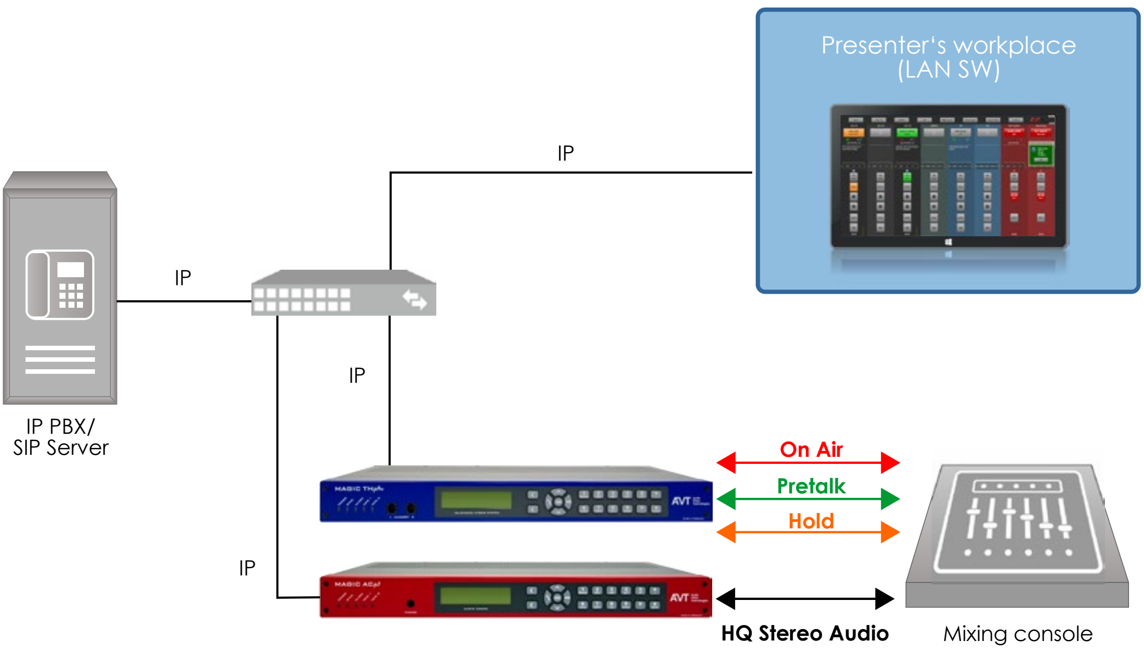
ION

With the new **IP Audio Codec MAGIC ACip3** and the **MAGIC THipPro 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.

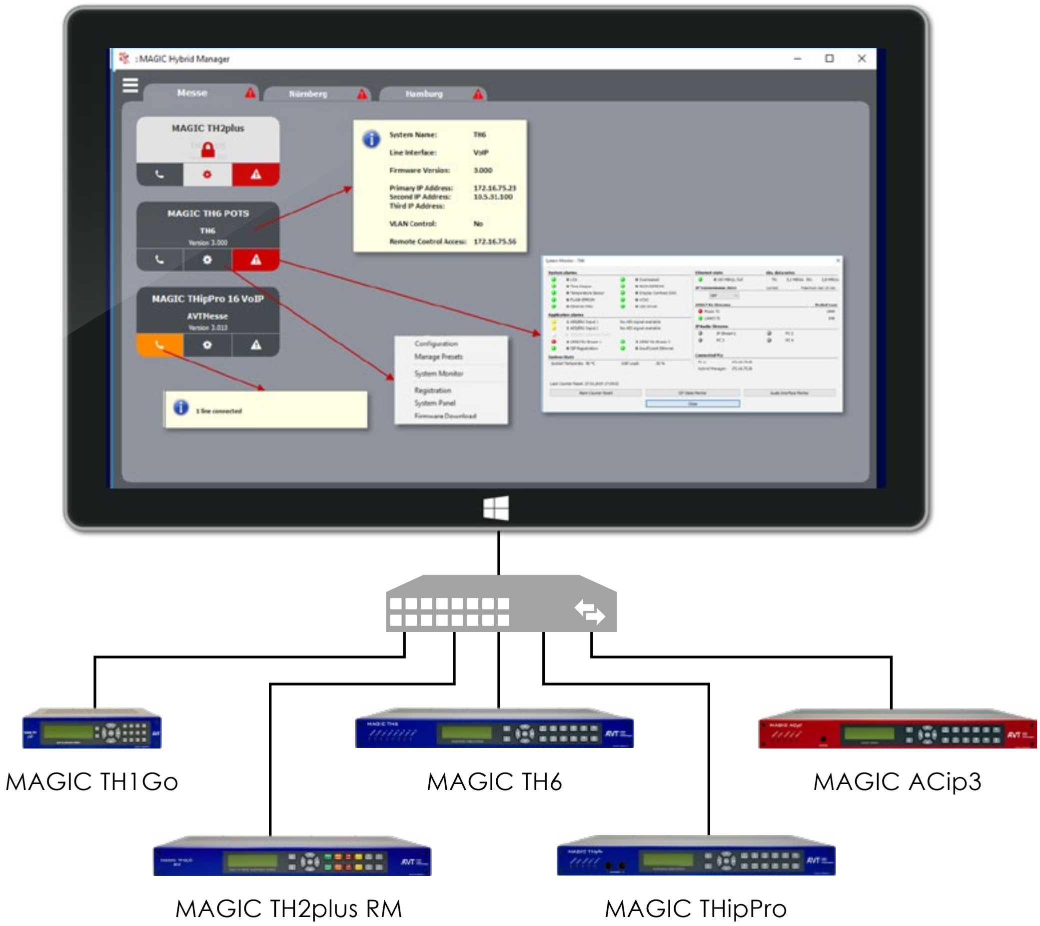
Details of the Audio Codec connection such as e.g. coding algorithm and bit rate are available via the caller line's Info button. Furthermore, a **common phone book** can be used.

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
- 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.

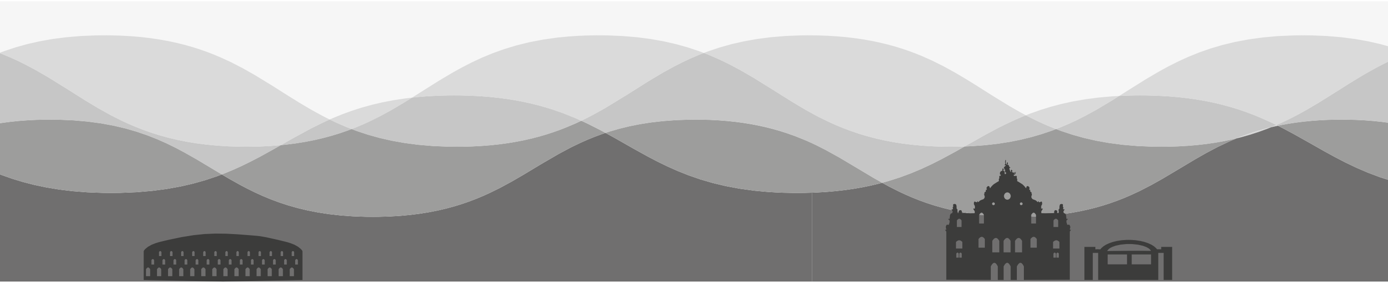
For each device, possible **alarms** and the **operating status** (in use or in configuration) are displayed. The query is made cyclically via SNMP. System Monitor, Registration Dialog, System Panel and Firmware Download can be accessed.

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.

In the future also desired systems are to be selected and then on appropriate clients and systems a new release is to be rolled out completely. Reconfiguration is also planned, either manually or via a scheduler. Presets and super presets can be loaded quickly and easily, as well as SIP accounts can be assigned or changed.

One System Manager license is required per system.

AVT Audio Video Technologies



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