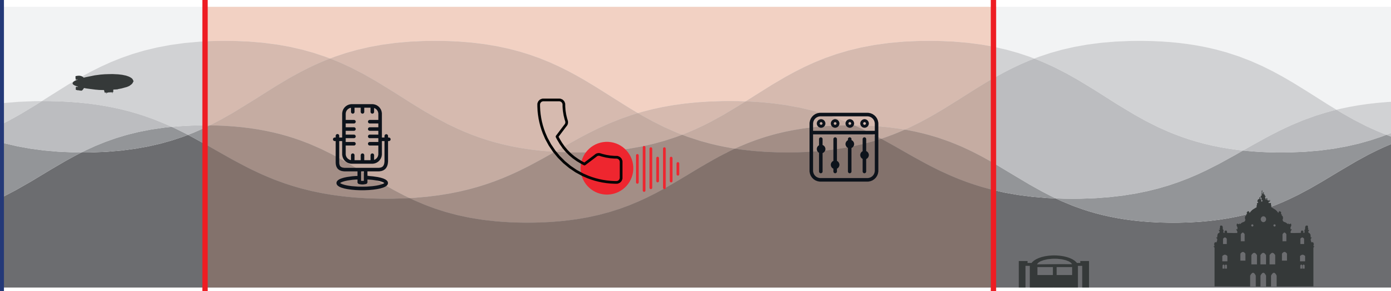
Telephone Hybrids



[VoIP | POTS | Web]

AVT Audio Video Technologies

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CONTENT

General

Telephone Hybrids or talkshow systems convert an incoming call into an audio signal suitable for studio technology.
VoIP (Voice over IP) has established itself as the standard transmission form for current and future applications. All AVT systems support VoIP operation, either already in the standard version or it can be activated by a software upgrade – even **retroactively.**

VoIP offers a significant advantage over analog telephone lines: **HD Voice.** Standard telephony only offers an audio bandwidth of 3.1 kHz whereas HD Voice provides an audio quality of 7 kHz. Nowadays, almost all common IP and DECT phones support HD Voice. However, an HD Voice connection can only be established if both sides and the transmission network are HD Voice compatible.

To use AVT Telephone Hybrids in VoIP mode, a separate **SIP account** is required for each caller line. With these accounts, the system can register on a SIP server (which can be, for example, the existing VoIP PBX or a SIP server on the Internet).

Since some countries have not yet shut down analog telephone lines, we still offer our 1- and 2-line telephone hybrid with POTS interfaces. For our multi-line telephone hybrids, we have optional POTS gateways available. Additionally, GSM gateways are available for all products.

As an alternative to classic hardware hybrids, the new **MAGIC Server** offers a flexibly scalable, software-based solution with 1 to 96 caller lines and up to 24 studios.

Telephone Hybrids are connected to the studio mixing console via integrated analog or digital **audio interfaces**.

With **Audio-over-IP** being used in more and more studio environments we also offer the AoIP protocols **AES67, Dante** and **Ravenna** as optional software upgrades or as hardware modules for our systems. MAGIC Server also allows ASIO-compatible virtual soundcards to be used as audio interfaces.

Depending on the operating mode, callers can either be switched to separate audio outputs or into a **conference**. In conference mode, the callers are available as a mixed signal at the selected output. The N-1 signal is automatically generated for each caller in the Telephone Hybrid.

Many radio stations **screen** callers before putting them ON AIR. This means that a conversation with the caller is conducted before the show and information is entered into a **database**. Interesting callers can be switched to ON AIR or called back later. Disturbing callers can be put on a blacklist; when they call again, they will only hear the busy signal and will be automatically rejected.

Other functions include **automatic call answering, call forwarding** or **recording.** For larger studios, it may be necessary to install multiple workstations that can access the Telephone Hybrid. This makes it possible, for example, to have an editor's workstation to initially talk to the caller, and another workstation for the moderator, who then sees the information that was previously entered by the editor.

The operation of the hybrids can also be done via the **MAGIC PhonerSet** in addition to the PC client.

Additionally, the integration of Telephone Hybrid channels into DHD or LAWO mixing consoles as well as VSM environments is possible. The **Ember+** and **DHD SetLogic** protocols allow signaling and control via virtual GPIOs.

With the MAGIC Collaboration Services, **Microsoft Teams and/or WebRTC** calls can be integrated into the telephone hybrids and handled via the same user interface as normal telephone calls. **Video** signals can also be transmitted. They are displayed directly in the user interface and can also be output via **NDI**. A return video signal can also be transmitted to the remote side.

Through the **integration of Audio Codecs** in MAGIC THipPro or MAGIC Server, one or more lines can be used for high-quality audio-over-IP connections.

In addition to Telephone Hybrids, the MAGIC THipPro VoIP Intercom with 8 or 16 VoIP lines is also available as a pure **intercom gateway.**

Additionally, the MAGIC THipPro VoIP VMS functions as a complete **answering machine** with individual announcement texts. Up to 32 callers can be recorded simultaneously.

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 blue symbol shows that the feature is available in the standard delivery version for the product described.

Option An light blue 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.

DSP Digital signal processing

A DSP-based Telephone Hybrid with digital echo cancellation, AGC and expander for each caller line.

Server MAGIC Server

A web-based Telephone Hybrid & Audio Codec Server.

VoIP VoIP via LAN interface(s)

The system can be connected to VoIP lines (using a SIP Server).

POTS POTS interface(s)

The system can be connected to POTS lines (analogue telephone lines).

VLAN VLAN

VLANs (Virtual Local Area Networks) can be set in the system configuration to separate the Audio stream from the control data.

QoS Quality of Service

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

HD HD Voice

Calls can be established and received in HD Voice quality (G.722, 7 kHz).

VD Voice Disguise

The caller's voice can be disguised so that he or she is not recognised.

Control via web browser

Currently Google Chrome, Safari, Firefox and MS Edge are supported.

Control via PC Software

The system can be operated via a Windows PC Software using a tablet/ PC connected via LAN. The number of workplaces depends on the system and the activated licences.

x MAGIC PhonerSet

In addition to or instead of a PC the designated number of PhonerSet units can be connected to the system.

ACCESS MS ACCESS database

The system can be used with an MS ACCESS database to save caller information (caller screening).

SQL MS SQL database

The system can be used with an MS SQL database to save caller information (caller screening).

AC Audio Codec integration

Audio Codec integration allows one or more lines to be used for high-quality audio-over-IP connections.

Pretalk Streaming

With Pretalk Streaming it is possible to use Pretalk via LAN. The Audio signals are transmitted between the Telephone Hybrid and the control PC via LAN. To talk to the caller the sound card of the PC or a USB handset/headset can be used.
The major advantage is that no audio cabling is required and the integrated recording function.

Blacklist

Callers can be blocked by putting their telephone number on a blacklist. A blocked caller simply hears the busy tone when he calls.

Answering machine

With the Answering machine you can record directly a caller or save a message for callers which are automatically accepted and dropped after this message has been played. This function can be used e.g. for prize draws/ game shows, when all prizes have been assigned and you still have incoming calls which need to be answered.

Night Service

When the Night Service is enabled, incoming calls are automatically accepted and forwarded to a pre-defined telephone number which can be e.g. an answering machine.

Analyse DTMF

All system can transmit DTMF tones, which is e.g. useful for entering a PIN to listen to messages on a mailbox or to join telephone conferences. Special: DTMF tones can be also analysed for e.g. game shows.

AES67

The AES67 upgrade allows the use of additional audio channels over IP via AES67, the lowest common denominator of several similar technologies, thus ensuring communication between e.g. AES67-compatible Dante and Ravenna devices.

Dante

The Dante interface allows the use of 32 Audio channels over IP via Dante.

Ravenna

The Ravenna interface board allows the use of 32 Audio channels over IP via Ravenna.

Ember+

The Ember+ protocol allows the control of a LAWO or DHD mixing console or any other Ember+ compatible system.

DHD SetLogic

The DHD SetLogic commands can be used to communicate with DHD mixing consoles or routers, e.g. for control or signalling purposes.

GPIO

Programmable TTL interfaces and Relay contacts are available.

MAGIC Collaboration Server

Connection to Microsoft Teams and/or WebRTC. In addition to audio transmission, the transmission of video signals is also possible.

NDI®

Video signals can be output via NDI. With the Extended Video In-/Outputs Upgrade, a separate NDI video input is available for the return signal and eight NDI video outputs.

NMOS

NMOS is used in IP-based media networks to improve interoperability and management. This enables seamless integration and control of audio and video devices in live broadcasts.

Power supply

The hybrid can be connected via a 12V external or integrated wide area power supply.

Feature	MAGIC TH1Go	MAGIC TH2plus V2	MAGIC THipPro Lite	MAGIC THipPro Lite Pure
POTS channels	1 (optional)	2	8 via Gateway (optional)	8 via Gateway (optional)
VoIP channels	1 (optional)	2 (optional)	4 or 8 (optional)	4 or 8 (optional)
VLAN	yes	yes	yes	yes
HD Voice (G.722, 7kHz)	for VoIP	for VoIP (optional)	for VoIP	for VoIP
Collaboration Services	-	MS Teams & WebRTC (optional)	MS Teams & WebRTC (optional)	MS Teams & WebRTC (optional)
NDI	-	yes (optional)	yes (optional)	yes (optional)
Audio interfaces	2 x analog input/output	2 x analogue or digital input/output (can be switched)	2 x analogue input/output 4 x digital input/ output (8 optional) (2/4 x AES/EBU)	-
AES67 channels	4 x RX (1 Stream) 4 x TX (1 Stream) (optional)	4 x RX (1 Stream) 4 x TX (1 Stream) (optional)	8 x RX (2 Streams) 8 x TX (1 Stream) (optional)	8 x RX (2 Streams) 8 x TX (1 Stream) (optional)
Dante-/Ravenna Interface	-	-	32 (optional)	32 (optional)
PHONE interface	1	2 (only in POTS mode)	-	-
Handset/Headset interface	-	2	2	-
Web browser control	yes	-	-	-
Software control (Windows PC or tablet)	yes (max. 1 workplace)	yes (max. 8 workplaces)	yes (max. 8 workplaces)	yes (max. 8 workplaces)
Control via MAGIC PhonerSet	-	yes, max. 2x (optional)	yes, max. 8x (optional)	yes, max. 8x (optional)
Database/ Caller Screening	phone book/ MS SQL (opt.)	MS ACCESS/ MS SQL (opt.)	MS ACCESS/ MS SQL (opt.)	MS ACCESS/ MS SQL (opt.)
Blacklist	-	yes	yes	yes
Pretalk Streaming Upgrade	yes, max. 1x	yes, max. 3x	yes, max. 8x	yes, max. 8x
HOLD signal	external, ON AIR signal	recorded, external, ON AIR signal	2 x recorded, external, ON AIR signal	2 x recorded, external, ON AIR signal
Answering Machine	-	-	-	-
Night Service	-	yes	yes	yes
GPIO (programmable)	4 x TTL 2 x Relays	4 x TTL 2 x Relays	8 x TTL 8 x Relays	8 x TTL 8 x Relays
Analyse DTMF	-	yes (optional)	yes (optional)	yes (optional)
Ember+ Prov./Cons. DHD Setlogic	yes (optional)	yes	yes	yes
NMOS	-	-	yes (optional)	yes (optional)
Power supply (redundancy)	external 12V	external 12V	100 – 230 V 5 V (optional)	100 – 230 V 5 V (optional)

MAGIC THipPro 8/16 VoIP	MAGIC THipPro 8/16 VoIP Pure	MAGIC Server Basic	MAGIC Server Professional
8 via Gateway (optional)	8 via Gateway (optional)	-	-
8 or 16	8 or 16	1 to 8	8 to 96
yes	yes	ja	ja
(optional)	(optional)	ja	ja
MS Teams & WebRTC (optional)	MS Teams & WebRTC (optional)	MS Teams & WebRTC (optional)	MS Teams & WebRTC (optional)
yes (optional)	yes (optional)	yes (optional)	yes (optional)
2 x analogue input/output 8 x digital input/output (4 x AES/EBU)	-	-	-
8 x RX (2 Streams) 8 x TX (1 Stream) (optional)	8 x RX (2 Streams) 8 x TX (1 Stream) (optional)	192 x total (optional)	192 x total (optional)
32 (optional)	32 (optional)	64 (Dante) 128 (Ravenna) (optional)	64 (Dante) 128 (Ravenna) (optional)
-	-	-	-
2	-	-	-
-	-	yes (max. 8 workplaces)	yes (max. 48 workplaces)
yes (max. 20 workplaces)	yes (max. 20 workplaces)	-	-
yes, max. 8x (optional)	yes, max. 8x (optional)	yes, max. 8x (optional)	yes, max. 48x (optional)
MS SQL	MS SQL	MS SQL	MS SQL
yes	yes	ja	ja
yes, max. 10x	yes, max. 10x	yes, max. 8x	yes, max. 48x
4 x internal (dynamic or recorded), 6 x external, ON AIR signal	4 x internal (dynamic or recorded), 6 x external, ON AIR signal	internal (dynamic or recorded), external, ON AIR signal	internal (dynamic or recorded), external, ON AIR signal
yes	yes	yes	yes
yes	yes	yes	yes
8 x TTL 8 x Relays	8 x TTL 8 x Relays	-	-
yes (optional)	yes (optional)	yes (optional)	yes (optional)
yes	yes	yes	yes
yes (optional)	yes (optional)	yes (optional)	yes (optional)
100 – 230 V 5 V (optional)	100 – 230 V 5 V (optional)	hardware-dependent	hardware-dependent

MAGIC TH1Go Telephone Hybrids

MAGIC TH1Go

DSP | VoIP | POTS | VLAN | QoS | HD | VD
ACCESS | SQL | AC | Pretalk Streaming | Analyse DTMF
AES67 | Dante | Ravenna | Ember+ | DHD | GPIO | NDI | NMOS | 12V

- 1 Line for VoIP or POTS operation
- 2 x analogue Audio inputs/outputs
- Optional 4 x AES67 channels via software upgrade
- Pretalk via POTS telephone or Audio line
- External HOLD signal or ON AIR as HOLD signal
- Control via POTS telephone or via web browser
- Front panel control
- Phone Book
- Dimensions: ½ x 19“ x 1U with external 12V power supply

MAGIC TH1Go can be configured in two way: It can be used via the LAN interface as a **1-Channel VoIP Hybrid** or via the analog interface as a **1-Channel POTS Hybrid** (analog a/b telephone lines).

As a special highlight the **VoIP Option** for **MAGIC TH1Go** already includes outstanding **HD Voice (G.722)** quality in addition to the standard 3.1-kHz telephone quality. Conversations from and to telephones which are HD Voice compatible can be established in 7-kHz quality – which means with twice the Audio bandwidth and with a much higher speech intelligibility.

In addition to the front panel operation the system can be controlled especially comfortably via an **HTML5 compatible web browser**.

Alternatively or in parallel a **POTS telephone** can be connected to the MAGIC TH1Go Hybrids. With this telephone the **complete system operation** is possible. In contrast to other solutions the telephone is not disconnected by a relay from the hybrid after talking to the caller but allows full control in each line status.

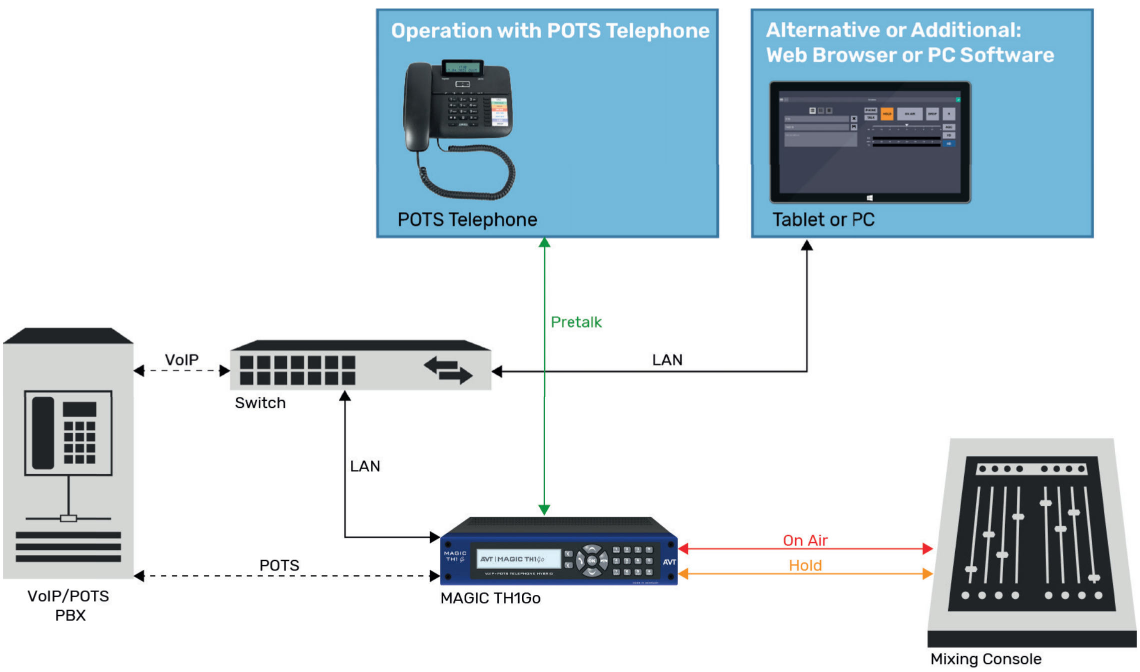
An incoming call is directly accepted in Pretalk by picking up the phone. You can talk to the caller and by pressing a Quick Dial key on the telephone the call can be transferred to HOLD or ON AIR. When the ON AIR conversation has been finished, the connection can be dropped or the caller can be transferred back to the telephone by pressing the Pretalk button again. For easy handling a telephone book is available.

MAGIC TH1Go

ONE CHANNEL – POTS/V

MAGIC TH1Go Web Browser Application

Example Application



Options

The MAGIC TH1Go can be equipped with either a VoIP & HD Voice option or a POTS option. Both variants can also be activated at the same time, so that an upgrade is possible at a later date.

As an alternative to the standard web browser operation, a **Windows PC Software** is optionally available. This licence adds the functions SNMP, Ember+, DHD SetLogic and MS Access/SQL database access.

MAGIC TH1Go also supports the so-called **Pretalk Streaming** function via the Windows PC software. With these upgrades, the LAN connection between hybrid and control PC can be used for pretalks, so no additional audio cables have to be laid. The conversation with the caller takes place via the PC sound card or a USB headset. Pretalk Streaming also allows the caller and presenter signals to be recorded as a WAV file.

As a further operating option, an **analogue POTS telephone** can also be connected to the system. In addition to the pretalk option, it allows control in both VoIP and POTS mode.

With the **AES67 Upgrade,** four additional IP audio channels can be used via AES67 with the Windows PC software.

An optional **DTMF Tone Analyzer Upgrade** with multiple modes is also available when the Windows PC Software is used.

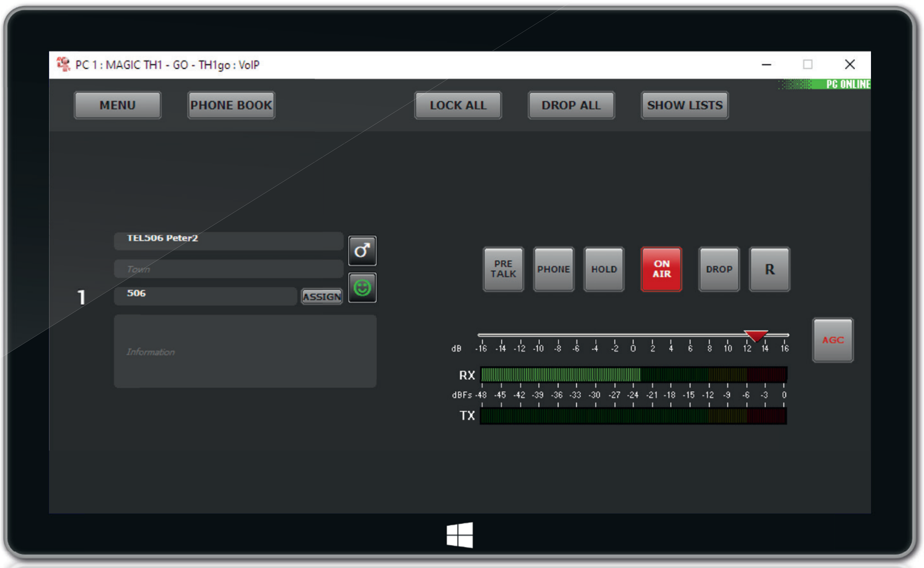
The **Remote Reporting Upgrade** (requires the DTMF Tone Analyzer) allows reporters or even emergency services to dial into the radio programme by telephone, even if the studio is unmanned. The access rank is protected by a PIN.

For more complex Ember+ configurations, the **Ember+ Consumer Extension Upgrade** (DHD) offer simplified programming.

With POTS functionality, MAGIC TH1Go can be extended to a GSM hybrid with an external **GSM adapter**.

To mount two systems next to each other in a 1U 19“ rack, the optional **Dual Mounting Kit** can be used.

To administer multiple AVT Hybrids, the **System Manager Upgrade** is available.



MAGIC TH1Go Windows PC Software

MAGIC TH2plus V2 Telephone Hybrids

MAGIC TH2plus V2

DSP VoIP POTS VLAN QoS HD VD
2 ACCESS SQL AC Pretalk Streaming Analyse DTMF
AES67 Dante Ravenna Ember+ DHD GPIO TH NDI NMOS 12V

- 2 Lines via POTS or opt. VoIP
- 2 x analogue or digital Audio inputs/outputs
- Optional 4 x AES67 channels via software upgrade
- Pretalk via handset/headset or Audio line VoIP mode
- Optional Pretalk Streaming with recording function
- External or recorded HOLD signal or ON AIR as HOLD signal
- Control via PC Software
- Max. 8 PC workplaces
- Max. 2 PhonerSets
- Max. 2 analog telephones for pretalk
- MS Access or optional MS SQL database for Screening

MAGIC TH2plus V2 is a **two channels Talkshow System** and offers two operating modes in one system: With its **two POTS interfaces**, and **one LAN interface** it can be used in POTS and optionally in **VoIP mode**. With an external **GSM adapter**, it can also be extended to a GSM hybrid.

The device is designed as a compact **1/2 x 19“ system** with an external 12V power supply.

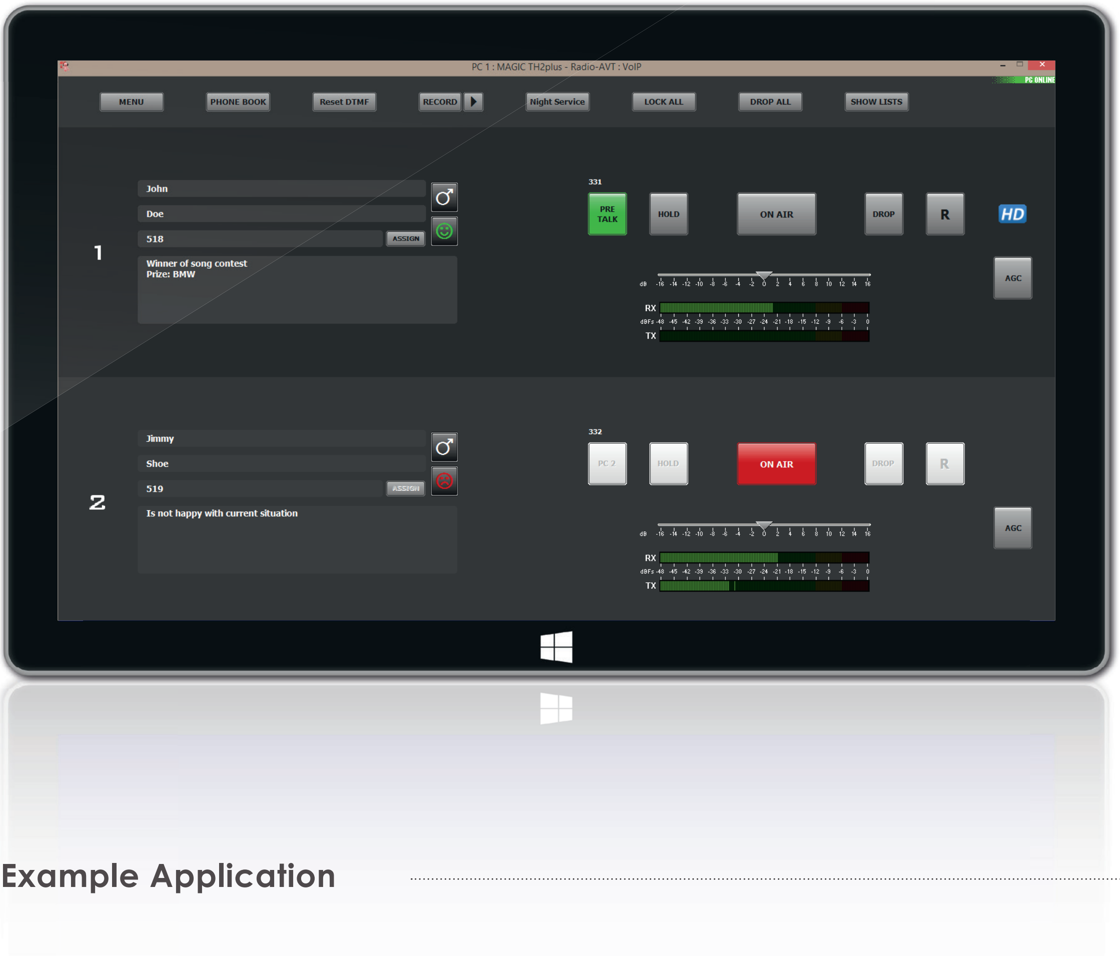
MAGIC TH2plus provides various functions such as caller conferencing, Auto Answer, Voice Disguise, Night Service, call forwarding and the possibility to work with a **database** for call screening. In this way, caller information can be displayed immediately for an incoming call if the number is transmitted. Unwanted callers can be put on a **black list**.

The system can either be directly operated via the front panel or more comfortably via the **Windows PC Software** for which one licence is included in the delivery.

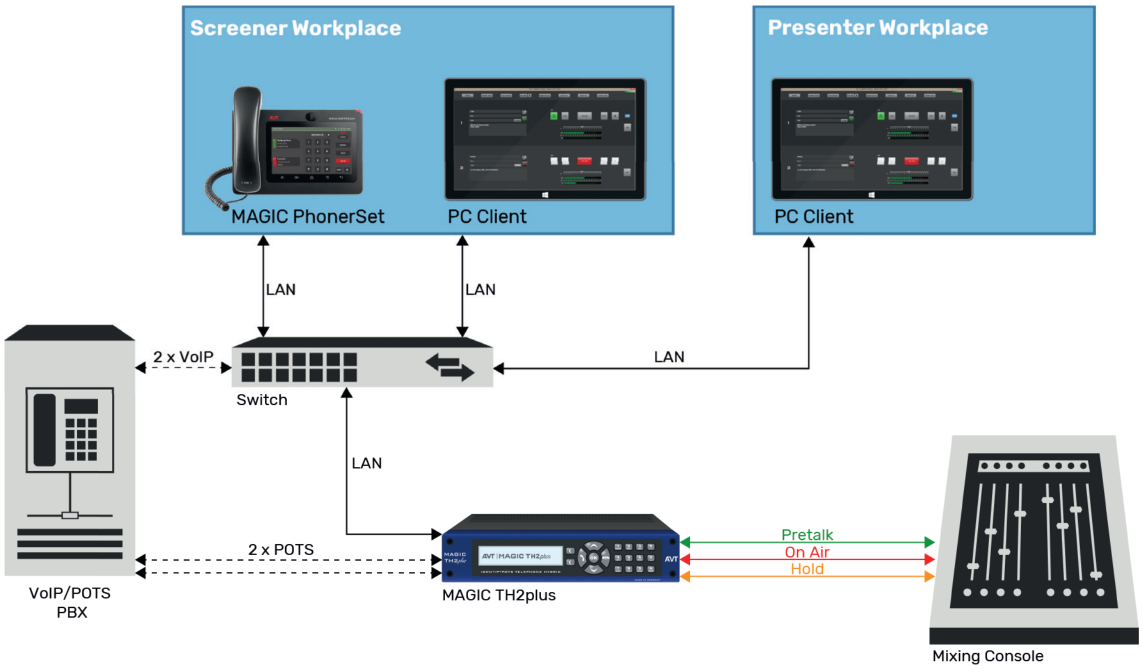
For an easy communication with DHD mixing consoles, MAGIC TH2plus allows the configuration of up to **64 DHD SetLogic commands**. Via the **Ember+ protocol (provider and consumer functionality 32 inputs and 32 outputs can be programmed**, and easy exchange of control and signalling commands between Hybrid and e.g. Lawo and DHD mixing consoles is possible.

MAGIC TH2plus V2

MAGIC TH2plus PC Software



Example Application



Options

MAGIC TH2plus can be extended with **VoIP & HD Voice Option** which turn the system into an VoIP Telephone Hybrid with outstanding 7 kHz speech quality.

Additional PC workplaces can be added by activating up to eight PC licences. The first licence is already included in the default scope of delivery.

Operation is also possible with up to two **MAGIC PhonerSets** as an alternative or supplement to the PC software.

With the **AES67 licence**, four additional IP audio channels can be used via AES67 in via a software upgrade.

For more complex Ember+ configurations, the **Ember+ Consumer Extension Upgrade** (DHD) and **Ember+ Dial Pad Extension Upgrade** (Lawo) offer simplified programming.

MAGIC TH2plus also supports the so-called **Pretalk Streaming** function. With this Upgrade the LAN connection between the control PC and the hybrid can be used for Pretalk which means no Audio cabling is required. The conversation with the caller happens via the PC sound card or a USB Headset. Additionally, Pretalk Streaming allows you to record the caller and presenter signals as WAV file. The Pretalk Streaming licences are available in two versions: either as a Pretalk Stream assigned to a fixed PC workplace (max. 10 licences) or as **Dynamic Pretalk Streaming**, which can be used flexibly from all workplaces (max. 8 licences).

By default, the MAGIC TH2plus works with a MS Access database. This can optionally be replaced by a **MS SQL database**.

For game shows or sports events, the **DTMF Analyzer Plug-In** can be added. There are three modes to choose from: Standard, Gameshow and Event. While in standard mode only the digits received via DTMF are displayed, in gameshow mode the caller who pressed first is also marked. In event mode, labels can be defined which are displayed instead of the numbers.

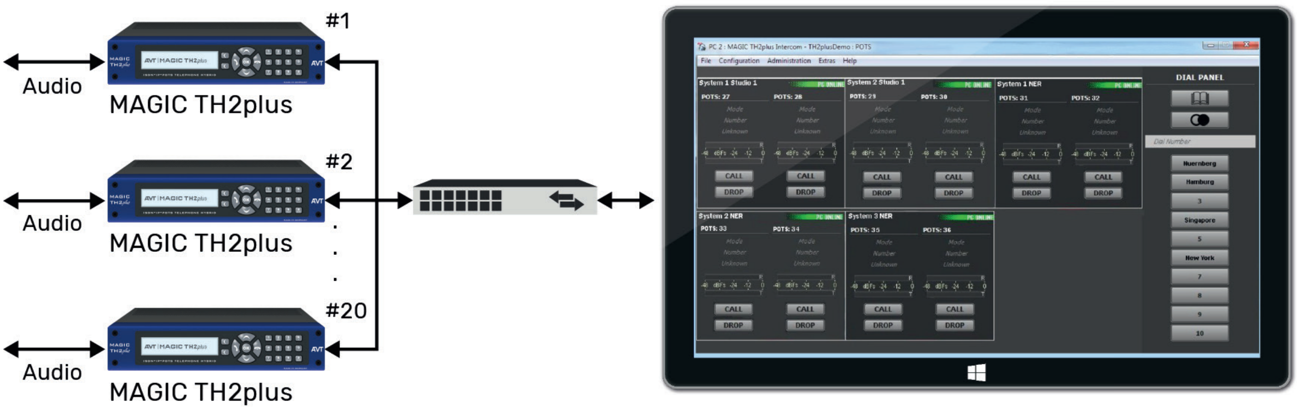
The **Remote Reporting Upgrade** (requires the DTMF Tone Analyzer) allows reporters to dial into the radio programme by telephone or even in emergencies, e.g. by the emergency services, even if the studio is unoccupied. The access line is protected by a PIN.

For support of WebRTC & MS Teams calls, the MAGIC **Collaboration Services** (p.36)can be integrated into this model.

For intercom applications, the **Intercom Upgrade** allows up to 30 MAGIC TH2plus systems to be displayed and operated in one user interface.

To manage multiple AVT hybrids, the **System Manager Upgrade** is available.

MAGIC TH2plus Intercom PC Software



MAGIC THipPro Lite Telephone Hybrids

MAGIC THipPro Lite 4/8

- 4/8 lines for VoIP or POTS operation
- 2 x analogue and 4 x (optional 8 x) digital Audio inputs/outputs
- Optional 8 x AES67 Channels via Software-Upgrade
- Pretalk via handset/headset or Audio line
- Optional Pretalk Streaming with recording function
- 2 x HOLD-Signal (external, recorded or ON AIR)
- Control via PC Software
- Max. 8 PC workplaces
- Max. 8 MAGIC PhonerSets
- Two studios
- MS Access or optional MS SQL database for Screening

MAGIC THipPro Lite is a talkshow system for **four or eight channels**. The system can be configured as VoIP system with integrated HD Voice function for best voice quality or with external gateway as analogue variant.

In addition, we now offer the cost effective **MAGIC THipPro Lite Pure** variant, which comes without conventional audio interfaces. The audio connection is realised exclusively via **AES67, Dante** or **Ravenna**.

The MAGIC THipPro Lite is based on the hardware of the MAGIC THipPro and can be equipped with **four or eight lines**. An upgrade from 4 to 8 lines or to a fully equipped MAGIC THipPro is also possible at a later date.

The following operating modes are possible as standard:
In **one-fader** mode, all callers on ON AIR are switched to a conference and are available as a mixed signal. The N-1 signal is automatically generated for each caller line.
In **two-fader** operation, two ON AIR lines are available. The callers can be output on separate audio outputs or switched to conference.
In **multi-fader** mode, a separate ON AIR line is defined for each caller. Each caller is available on a separate audio output and mixing is possible via the mixing console.

The lines of the Telephone Hybrid can be split between **two studios** if required. Only the selected caller lines are displayed in each studio and each studio has its own ON AIR, HOLD and PRETALK lines.

For caller screening, an integrated **MS Access database** is available, which can optionally be replaced by an **MS SQL database**.

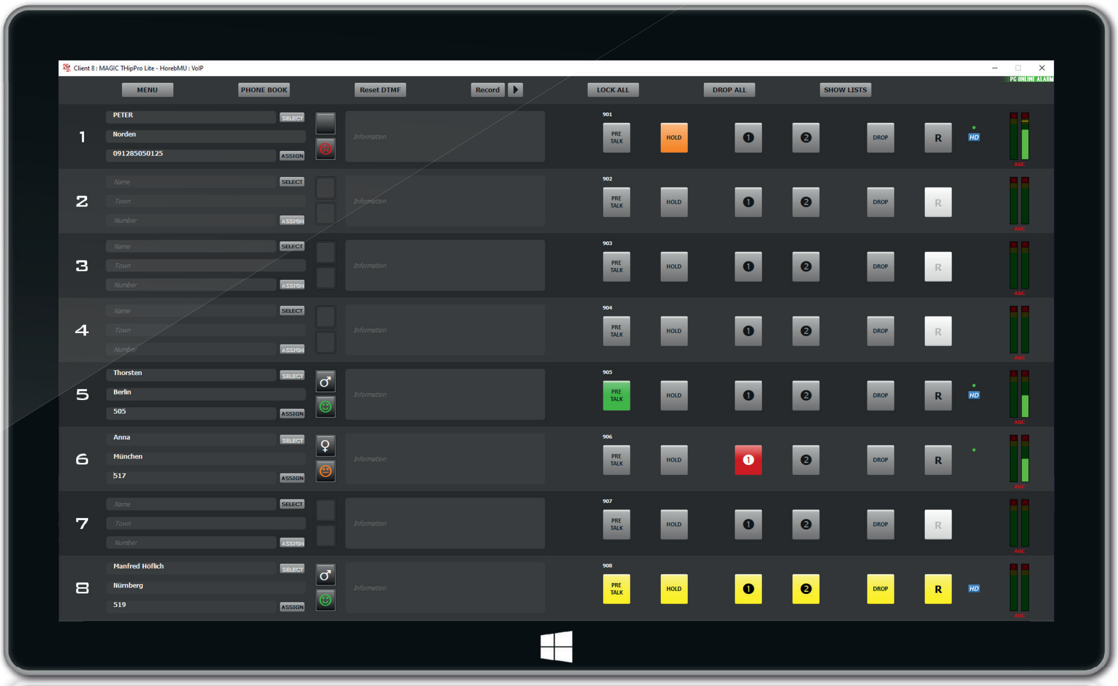
For easy communication with DHD consoles, MAGIC THipPro allows configuration of up to **96 DHD SetLogic commands**. With the **Ember+ protocol** (provider and consumer functionality), 96 inputs and 96 outputs can be programmed and easy exchange of control and signalling commands and caller information (e.g. name, phone number) between hybrid and mixing console (e.g. Lawo or DHD) is possible.

In addition to **two analogue audio inputs and outputs**, MAGIC THipPro Lite has **four digital audio lines (2 x AES/EBU interfaces)** which can optionally be extended to **eight digital audio lines (4 x AES/EBU)** via software upgrade. In addition, **two handset/headset interfaces** are available. The audio interfaces can be assigned to the installed workplaces and studios as desired.

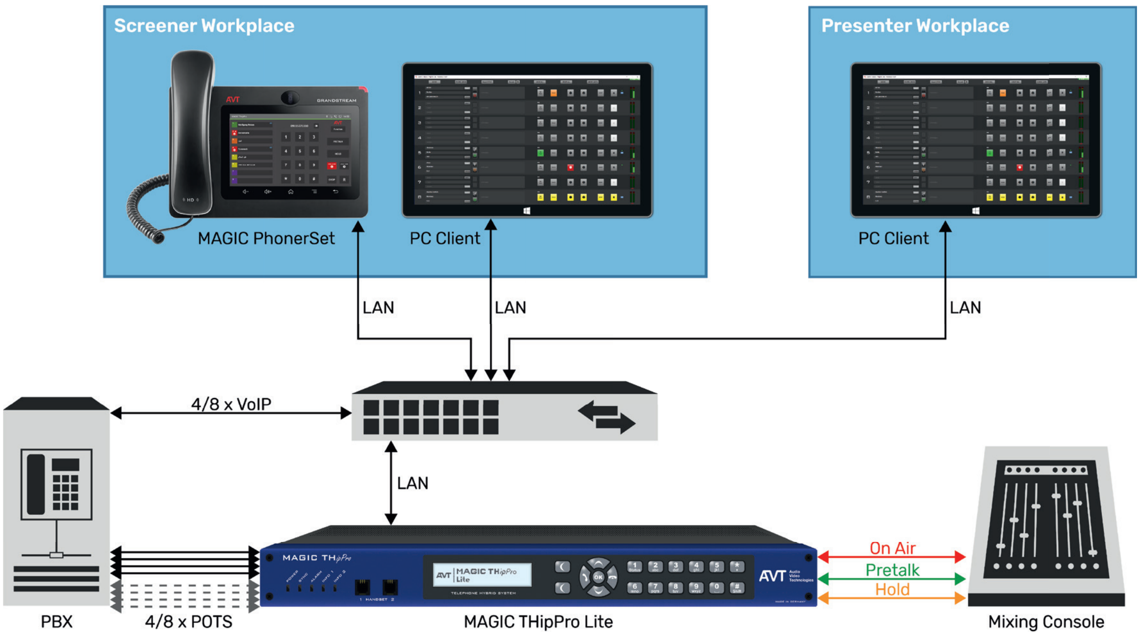
MAGIC THipPro Lite with VoIP, redundant power supply, Dual LAN- and Dante Module

MAGIC THipPro Pure Lite with Dante Module, redundant power supply
(without conventional audio interfaces)

MAGIC THipPro Lite PC Software



Example Application



Options

MAGIC THipPro Lite comes with a **PC control licence** and can be upgraded with up to seven additional licences, allowing a maximum of eight PC workplaces to access the hybrid simultaneously. In this way, the system can also be used for two studios. A maximum of four workplaces can be installed in each studio.

The **MAGIC PhonerSet** telephone offers another option for convenient operation.

MAGIC THipPro Lite, like all AVT systems, also supports the **Pretalk Streaming** function. With this upgrade, the LAN connection between the hybrid and the control PC can be used for pretalks, so there is no need to lay additional audio cables. The conversation with the caller takes place via the PC sound card or a USB headset. Pretalk Streaming also allows recording as WAV files. Up to eight Pretalk Streaming licences can be used. They are available either statically or as Dynamic Pretalk Streaming licences.

For game shows or sporting events, the **DTMF Analyzer** Upgrade is available. Three operating modes are available: Standard, Gameshow and Event. In standard mode, only the digits that callers have entered on their phone are displayed.
In Gameshow mode, it is additionally highlighted which caller was the first to press.
In Event Mode, different event labels can be defined in the software for the DTMF digits. For example, sports reports with several reporters in different stadiums and telephone connection to the MAGIC THipPro Lite can be comfortably realised. One reporter is switched ON AIR while the others are on HOLD. For important events such as goals, these reporters can signal this via DTMF input by assigning the term “goal” for “0” and the term “red card” for “1”. The moderator sees the information directly in the software and can now immediately switch the reporter of the relevant match to ON AIR.

The **Remote Reporting Upgrade** (requires the DTMF Tone Analyzer) allows reporters to dial into the radio programme by telephone or even in emergencies, e.g. by emergency services, even if the studio is unoccupied. The access is protected by a PIN.

With the **Dual LAN Module**, the system can be expanded by two additional LAN interfaces, so that a total of four LAN interfaces are available.

With the **AES67** licence, eight additional audio channels can be used through a software upgrade.

Alternatively, a native **Dante** or **Ravenna Module** with 32 audio channels can be equipped. AES67 is supported by both modules.
The Ravenna module also supports **NMOS**.

The **Digital Audio Upgrade** enables the activation of two additional AES/EBU interfaces, providing a total of 8 outputs and inputs.

For more complex Ember+ configurations, the **Ember+ Consumer Extension Upgrade** (DHD) and **Ember+ Dial Pad Extension Upgrade** (Lawo) offer simplified programming.

To manage multiple different AVT hybrids, the **System Manager Upgrade** is available.

For support of WebRTC & MS Teams calls, the MAGIC **Collaboration Services** (p.36)can be integrated into this model.

As studio requirements change, the **MAGIC THipPro Lite** can be upgraded to a fully featured **MAGIC THipPro** via software upgrade. This allows the use of all features and options of the top model.

MAGIC THipPro Telephone Hybrids

MAGIC THipPro 8/16

DSP | VoIP | POTS | VLAN | QoS | HD | VD
8 | ACCESS | SQL | AC | Pretalk Streaming | Analyse DTMF
AES67 | Dante | Ravenna | Ember+ | DHD | GPIO | TH | NDI | NMOS | 230V

- 8/16 lines for VoIP operation
- 2 x analogue and 8 x digital Audio inputs/ outputs
- Optional 8 x AES67 or 32 x Dante/Ravenna Interface
- Pretalk via handset/headset or Audio line
- Optional Pretalk Streaming with recording function
- 2 x HOLD-Signal (external, recorded or ON AIR)
- Max. 20 PC workplaces
- Max. 8 MAGIC PhonerSets
- Optional up to six studios
- MS SQL database for Screening
- Optional redundant power supply

MAGIC THipPro Telephone Hybrid is the most flexible hardware system in our product portfolio and offers the widest range of functions. The hybrid is available with **8 or 16 VoIP lines.** Optionally, the **8 x FXO Ports VoIP/POTS Gateway** can be used to provide eight interfaces for connecting to a conventional POTS infrastructure.

MAGIC THipPro has **eight digital audio lines (four AES/EBU interfaces)** and **two analogue audio inputs/outputs** in each version. In addition, there it has **two handset/headset interfaces** for Pretalk. The audio interfaces can be assigned to the installed workstations and studios as desired.

For users who no longer require classic audio interfaces, the **MAGIC THipPro Pure** version is also available. The optional AES67 software upgrade or the Dante or Ravenna module is used for audio connection.

A total of up to **20 workplaces** can access the MAGIC THipPro simultaneously. With the **Admin Studio Upgrade**, up to **six studios** can be set up.

Two software versions are available to control the system: MAGIC **THipPro LAN** and **MAGIC THipPro Screener**. No PC licence is included in the scope of delivery, so the user can decide for which combination is suitable for his purposes. In addition, for smaller recording booths that do not require an ON AIR function, there is the affordable **News Desk Client**.

MAGIC THipPro can be connected to an **MS SQL database**. Individual studios can use the same or individual databases. All features such as the blacklist function, automatic call acceptance, call forwarding, voice disguise, night service and call pre-allocation are supported. As a special feature, the answering machine function has also been implemented. Calls can be automatically answered and recorded.

For an easy communication with DHD mixing consoles, MAGIC THipPro allows the configuration of up to **96 DHD SetLogic commands**. Via the **Ember+ protocol (Provider and Consumer functionality) 96 inputs and 96 outputs can be programmed**, and easy exchange of control and signalling commands and caller information (e.g. name, phone number) between hybrid and mixing console (e.g. Lawo or DHD) is possible.

MAGIC THipPro with VoIP, redundant power supply, Dual LAN and Dante module

MAGIC THipPro Pure with Dante Module, redundant power supply
(without conventional audio interfaces)

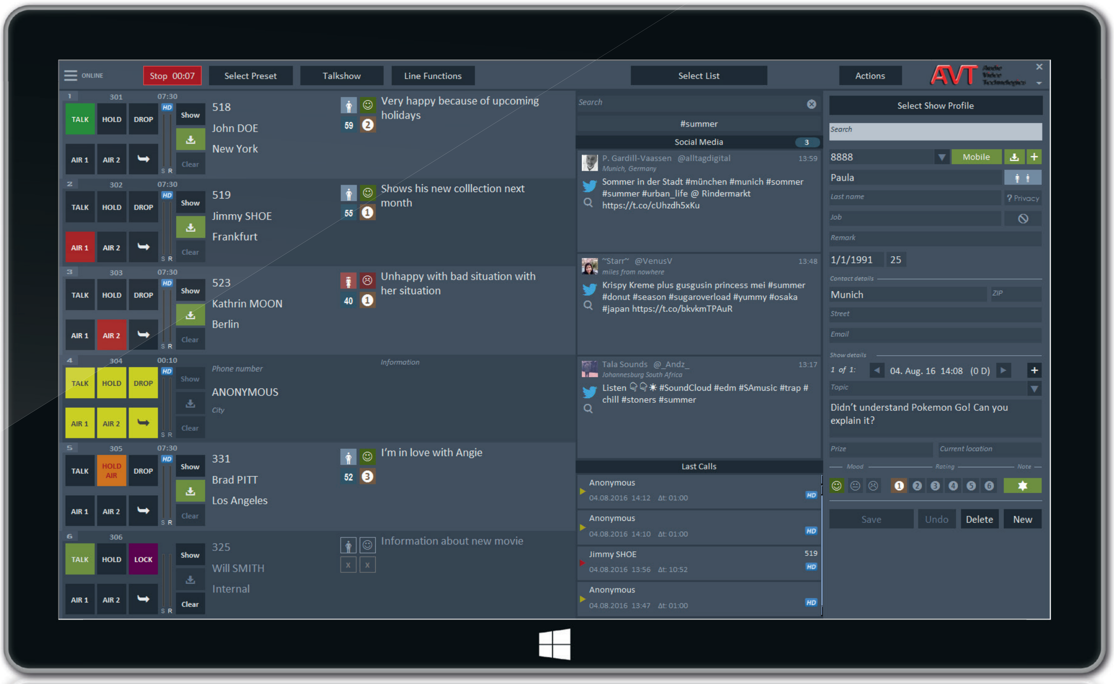
MAGIC THipPro LAN Software

User interface in Production/Screener view

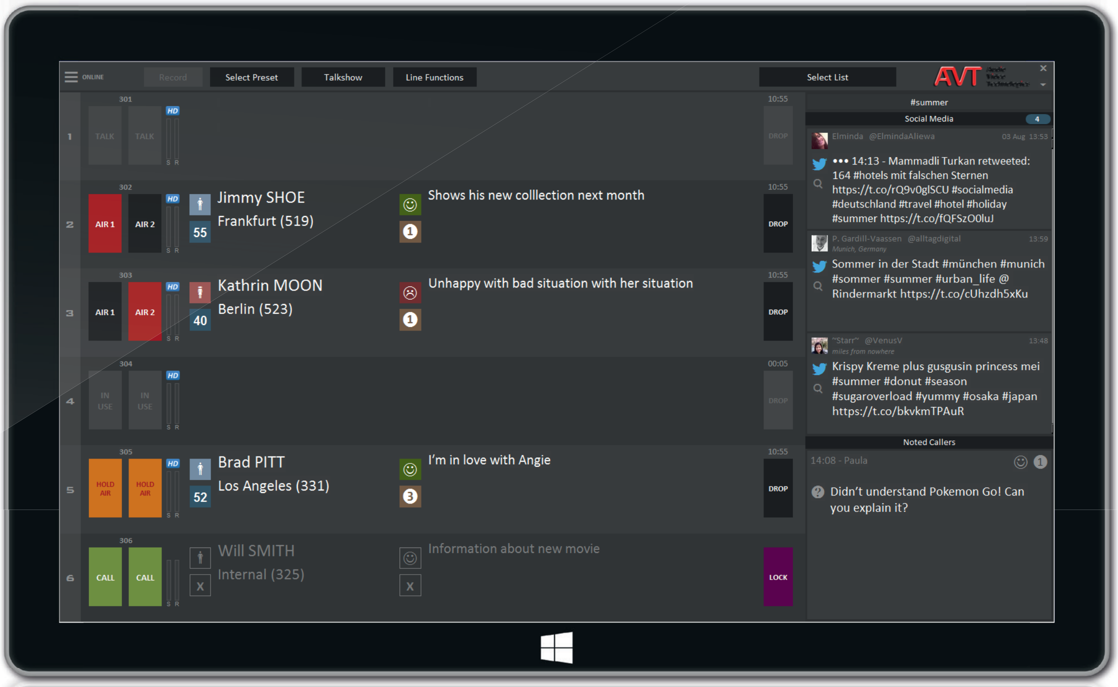
User interface in Presenter view

MAGIC THipPro Screener Software

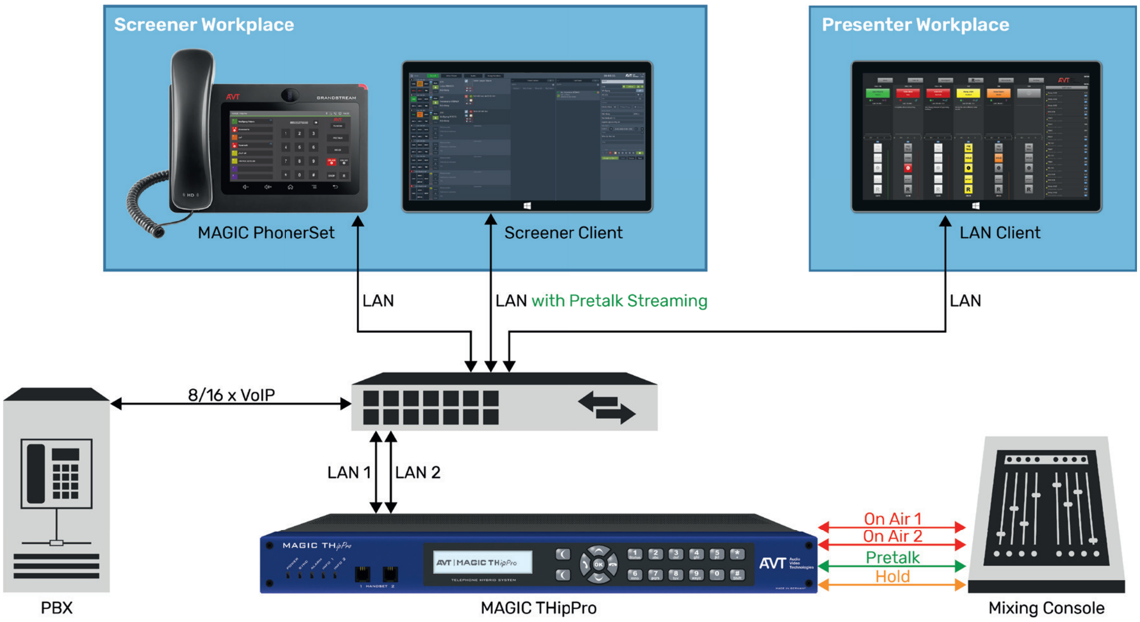
User interface in Screener view



User interface in Presenter view



Example Application: Basic



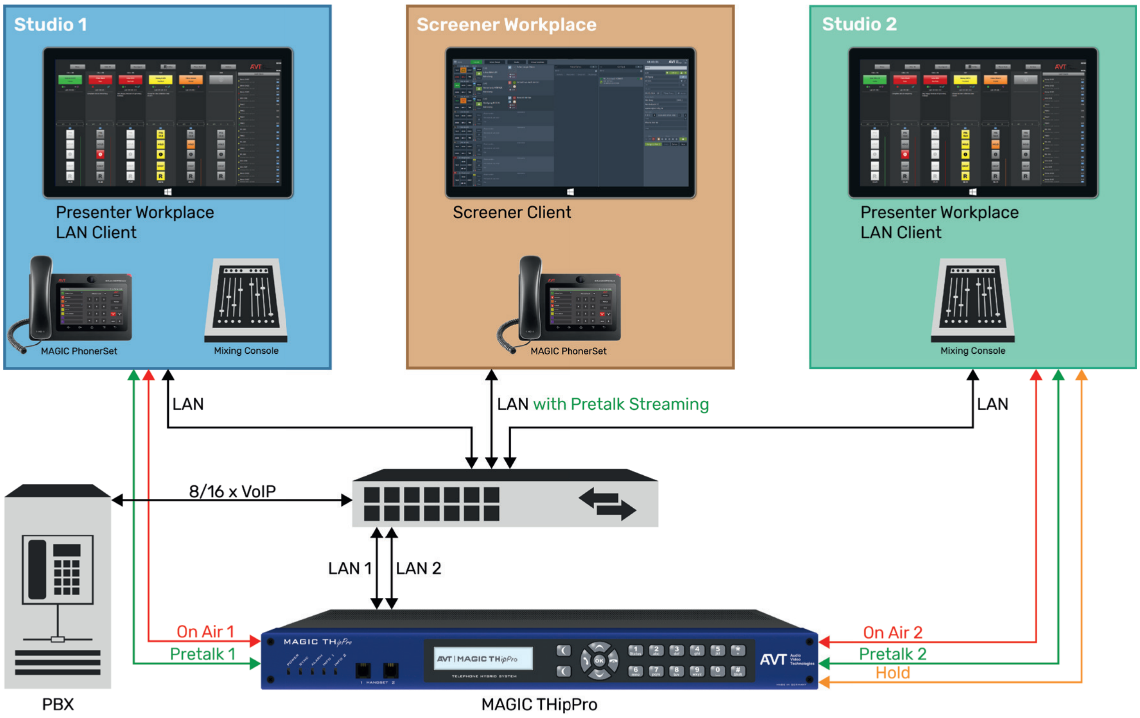
Our example application on this page shows a **basic setup of the MAGIC THipPro Telephone Hybrid**. There are two workplaces, one for the Presenter/Host and one for the Screener or Producer. The Screener/Producer works with the MAGIC THipPro Screener Software. He answers the calls first and can easily enter information about the callers via the data entry mask.

With status functions such as e.g. HOLD READY he can show the Presenter which callers have been screened and are ready to be switched to ON AIR. The Screener uses the MAGIC PhonerSet or alternatively Pretalk Streaming which means that the Audio signal between the Telephone Hybrid and his PC is transmitted via the LAN network. In this way, the Screener can be located far from the Telephone Hybrid without complicated Audio cabling. Besides, he can also record callers easily via a single button in his user interface.

The Presenter works with the MAGIC THipPro LAN Software showing the most important information of the callers. The Tweets which have been marked by the Screener are displayed on the right side of the user interface so that the Presenter can easily read them and decide which ones he wants to share with the listeners. The Presenter can also switch the callers to PRETALK to talk with them Off Air if required.

1-fader, 2-fader or multi-fader modes can be set individually for each studio. In the example shown, the 2-fader mode is configured. If several callers are to be switched into conference, they only have to be switched to the same ON AIR line.

Example Application: Two Studios



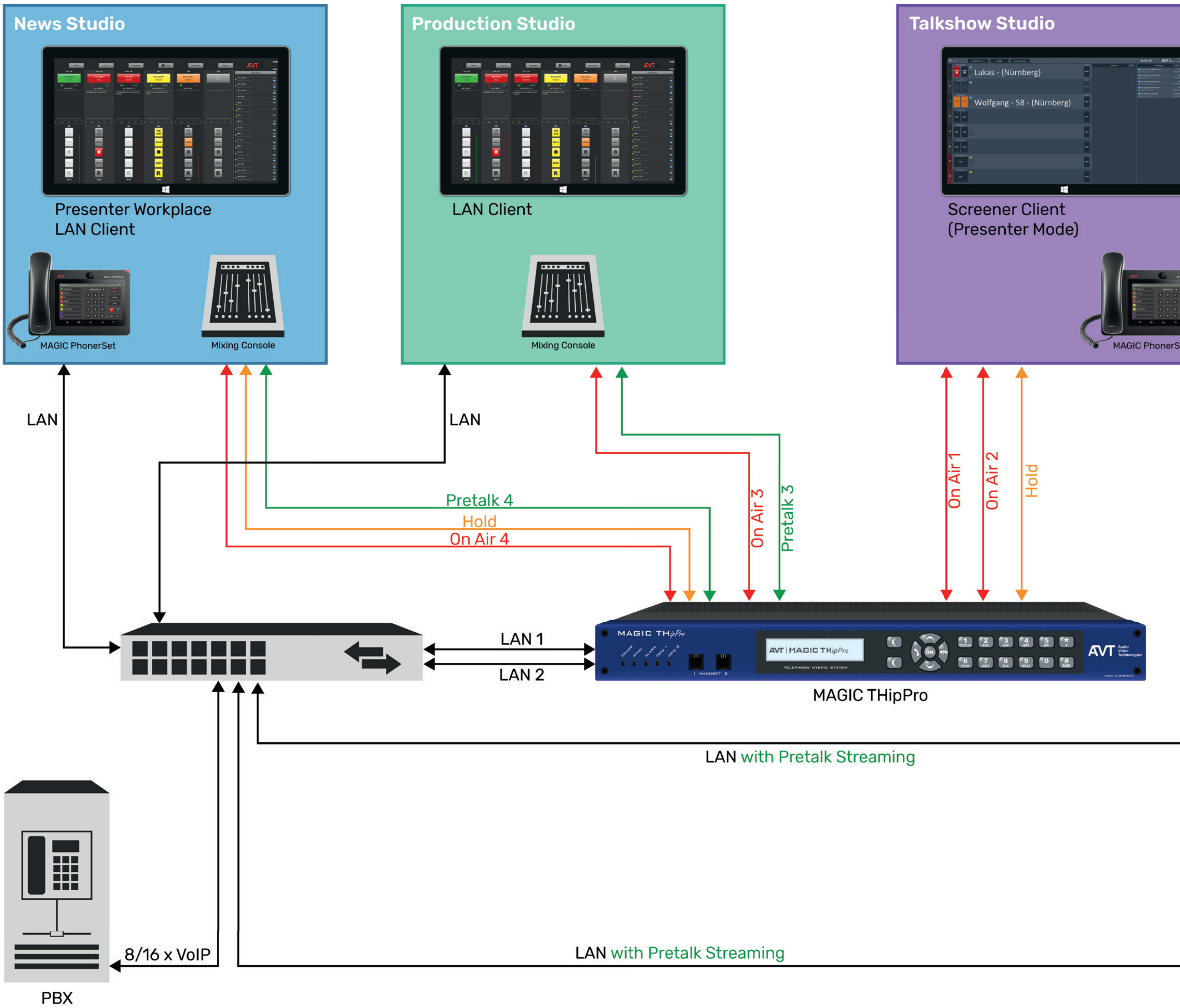
This sample application shows **two studios which share the same caller lines**. Both presenters each work with the MAGIC THipPro LAN software. Each studio has its own ON AIR audio line, which means both presenters can switch one or more callers to ON AIR at the same time. Each presenter also has an own PRETALK line in case a pretalk conversation with the caller should be possible.

Usually, the screener takes the calls in PRETALK and enters the relevant information into the database via the MAGIC THipPro Screener software. This information is then also immediately displayed at the presenters' workplace. As long as the screener is talking to a caller in PRETALK, the caller line is blocked for both presenters. Only when the caller is switched to HOLD can he or she be taken over by the presenters in the studios.

Optionally, the MAGIC THipPro can also be conveniently operated via the MAGIC PhonerSet, which can be used in parallel with the PC software or independently. Up to 8 MAGIC PhonerSets can be used simultaneously.

In the example shown, the 1-fader mode is configured.

Example Application: Multi Studio



S – POTS/VOIP

The application on this page shows how MAGIC THipPro can be used for **several independent studios (up to six)**. The **Admin Studio Upgrade** allows all available caller lines to be divided among the individual studios. Only the assigned caller lines are displayed in each studio. In this example, three different studios and one recording booth are configured.

The **News Studio** has two caller lines and one workplace. This workplace uses the MAGIC THipPro LAN Software. The operator can accept the callers in Pretalk to have a conversation before switching them ON AIR. The HOLD signal in inserted externally for this studio via one of the MAGIC THipPro's Audio inputs.

The **Production Studio** uses three caller lines and also has a workplace where the MAGIC THipPro LAN software is installed.
The production studio has separate PRETALK and ON AIR audio lines, the HOLD signal is generated here from the ON AIR return signal.

The **Talkshow Studio** uses six caller lines and two workplaces: a screener and a presenter. The screener uses Pretalk Streaming as audio interface, he accepts calls in PRETALK and enters the relevant information into the database via the MAGIC THipPro Screener software. Pretalk Streaming also allows him to record callers easily via a record button in the user interface. The presenter also uses the MAGIC THipPro Screener software, but in the so-called Presenter Mode. In this mode, only the caller lines and the most important information about the caller are displayed.

Caller information is not displayed to the presenter until the screener has set the line to HOLD READY status. The talkshow studio uses two ON AIR lines in the example. In this way, callers can be placed on different faders or placed in conference by selecting the same ON AIR line.

The **Recording Booth** uses the **News Desk Client**, which is designed like the MAGIC THipPro LAN software. However, the client has no ON AIR function and user configuration is not possible. Pretalk streaming is already included in the affordable News Desk Client software. This allows calls to be conveniently recorded as WAV files.

Options: User Interfaces

There are three software versions available for the control of the MAGIC THipPro Telephone Hybrid. For each workplace the most suitable application can be selected.

MAGIC THipPro LAN Software

The first variant is the MAGIC THipPro LAN software, which is available as a **single-user licence**. In the user interface, all caller lines are displayed side by side, each with the configured status keys (PRETALK, HOLD, ON AIR, DROP and call forwarding) and the level indicator. Each line has an info button that shows the caller's name and number or location, depending on what information is available. Below this, the gender, age and mood of the caller are displayed. It also shows when the caller was last switched to ON AIR. In combination with the Screener Client, the last time a caller won a prize is also displayed. Further details about the caller can be entered in the information field below. This information is stored in the database and can be displayed automatically with the next call. The caller list can be shown or hidden depending on the programme requirements. Calls can be initiated directly in PRETALK, HOLD or ON AIR. Manual dialling is of course also possible. Speed dialling keys can be programmed for frequent contacts.

News Desk Client Software

The MAGIC THipPro News Desk Client Software is a limited version of the MAGIC THipPro LAN Software. It is not possible to configure the system or switch a caller to ON AIR via this software. However, a Pretalk Streaming Licence is included, which allows the user to record the call in PRETALK and save it as a WAV file. Therefore, this software is particularly suitable for recording booths. The caller lines assigned to the News Desk Client are configurable as a pool or as individual lines.

MAGIC THipPro Screener Software

The MAGIC THipPro Screener software is available as a **single-user licence.** The caller lines are displayed one below the other and offer special functions for **caller screening** in addition to the "normal" operation. The user interface offers a data entry mask, which can be individually adapted to different programmes. Lists of preregistered callers as well as callback lists can be created. It is also possible to enter questions from callers and then display them at the presenter's desktop. A history of previous entries and calls is created for each caller. Information about winnings and other caller data, such as birthday and address, can also be stored. Two lists can be used in parallel.

The MAGIC THipPro Screener software also offers a simplified **Presenter Mode** without an input mask. The **Data Manager Mode** is used for post-processing without displaying the caller lines.

All software variants can be customised and thus adapted to the requirements of the users. Functions can be hidden or blocked for certain workplaces. A separate colour scheme can be selected for each studio and each line for a better overview. Configurations can be protected from unauthorized access by a user and an administrator password.

Options: Functions

The **MAGIC THipPro Admin Studio Upgrade** (single licence) allows to divide the line resources of the MAGIC THipPro among up to six studios. In each studio, only the assigned caller lines are displayed. Individual audio lines can be defined for each studio. A maximum of 20 LAN or screener PC clients or up to 30 news desk clients are possible.

Each VoIP system can be extended by the **MAGIC Collaboration Services**. The application runs on a server/PC and connects to MS Teams to integrate Teams calls into the user interface for studio operation.

MAGIC THipPro also supports the **Pretalk Streaming** feature. With this upgrade, the LAN connection between the hybrid and the control PC can be used for pretalks, so there is no need to run additional audio cables. The conversation with the caller takes place via the PC sound card or a USB headset. Pretalk Streaming also allows recording as a WAV file. A total of up to ten Pretalk Streaming licences can be activated. These can be statically or dynamically assigned to a workplace.

For game shows or sporting events, the **DTMF Analyser Plug-In** can be added. There are three modes to choose from: Standard, Gameshow and Event. While in Standard Mode only the digits received via DTMF are displayed, in Gameshow Mode the caller who pressed first is also marked. In event mode, labels can be defined that are displayed instead of the numbers.

The MAGIC THipPro Screener Client can be extended by the **Competition Management** feature, which provides a comfortable event and prize management for competitions directly in the software. Various data exports are also possible.

The **MAGIC THipPro ACconnect** upgrade, available exclusively for the MAGIC THipPro, allows the integration of up to two MAGIC AC1 Go or MAGIC ACip3 hardware Audio Codecs into the user interface of the Telephone Hybrid. The codec lines are displayed like additional caller lines and enable high quality AoIP connections.

For all MAGIC THipPro systems in VoIP mode, the **HD Voice** feature can be optionally activated, enabling connections in excellent 7 kHz (G.722) quality.

If additional caller lines are required, it is possible to expand the MAGIC THipPro 8 VoIP to **16 caller lines**.

With the **AES67 licence**, eight additional audio channels can be used through a software upgrade.

Alternatively, a native **Dante** or **Ravenna Module** with 32 audio channels can be equipped. AES67 is supported by both modules.
The Ravenna module also supports **NMOS**.

For more complex Ember+ configurations, the **Ember+ Consumer Extension Upgrade (DHD)** and the **Ember+ Dial Pad Extension Upgrade (Lawo)** offer simplified programming.

With the **Dual LAN Module**, the system can be extended by two additional LAN interfaces, providing a total of four LAN interfaces.

To manage several different AVT hybrids, the **System Manager Upgrade** is available.

For support of WebRTC & MS Teams calls, the MAGIC **Collaboration Services** (p.36)can be integrated into this model.

All new systems can be extended with a **redundant 5V power supply**.

MAGIC PhonerSet

MAGIC PhonerSet

- Colour Display
  5" (GXV3450) with 1280 x 720 pixels
  7" (GXV3470) with 800 x 1280 pixels
  8" (GXV3480) with 1200 x 800 pixels
- Capacitive touch screen
- Optional extension module (GBX20 for GXV3450) for e.g. line status and speed dial key function
- 2 x 1000Base-T (PoE+)
- WLAN 802.11 a/b/g/n/ac/ax
- Bluetooth 5.0
- Headset Connector
- HDMI for additional monitor
- HD Voice
- Android operating system

The **MAGIC PhonerSet License** enables convenient operation of the MAGIC Telephone Hybrid family (MAGIC TH2plus, MAGIC TH6, MAGIC THipPro/ Lite and MAGIC Server) via the touchscreen-enabled Grandstream telephones GXV3450, GXV3470 or GXV3480.

The telephone can be used parallel to the Windows PC software or independently without a PC. The connection for the control of all caller lines and the transmission of the audio signal takes place via a standard network interface.

MAGIC PhonerSet uses the caller lines of the used Telephone Hybrid system - an additional registration to a SIP server is not necessary.

The operation is based on the respective PC software of the Telephone Hybrid system.

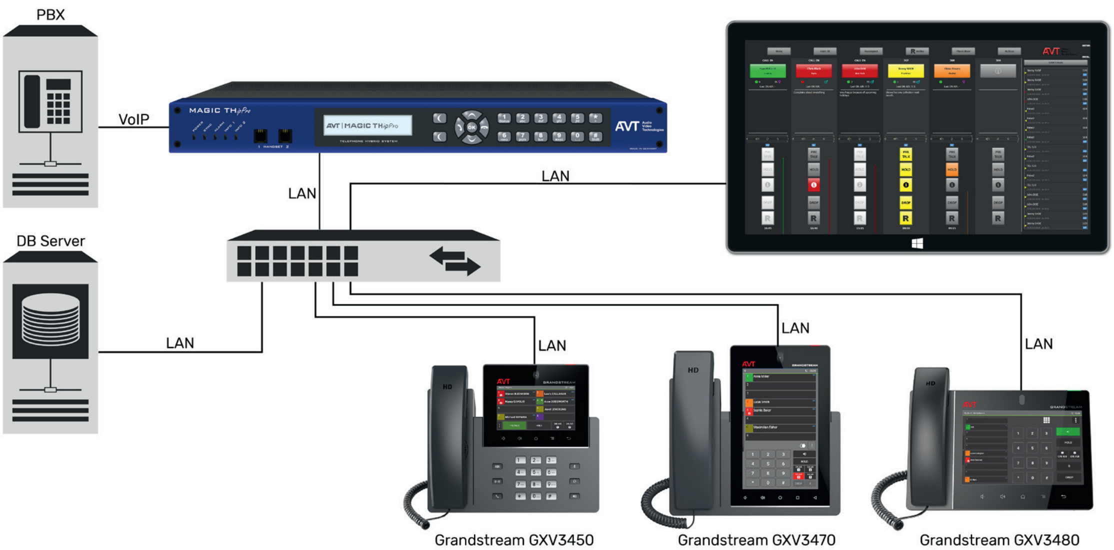
The following telephony functions are supported separately for each available caller line: Call setup, call acceptance, hang up, redial, call transfer, PRETALK, HOLD, ON AIR (1-fader or X-fader mode) and line blocking (LOCK). Further functions are available via the Function key: phone book, Lock/ Unlock all lines (LOCK ALL) and hang up all lines (DROP ALL).

The status and caller information of each line are displayed via line cards. Depending on the number of lines, the following information can be displayed: HD-Voice status, Phone number/SIP Display name, caller name and city (only if database is activated).Of course, Unicode characters (UTF-8) are also supported, so that, for example, Cyrillic or Arabic caller information can be displayed without problems.

The usable lines and authorisations (e.g. switching to ON AIR) of an operator workplace are automatically assigned via the system configuration of the Telephone Hybrid system. Only the connection parameters to the Telephone Hybrid system must be entered at the **MAGIC PhonerSet.**

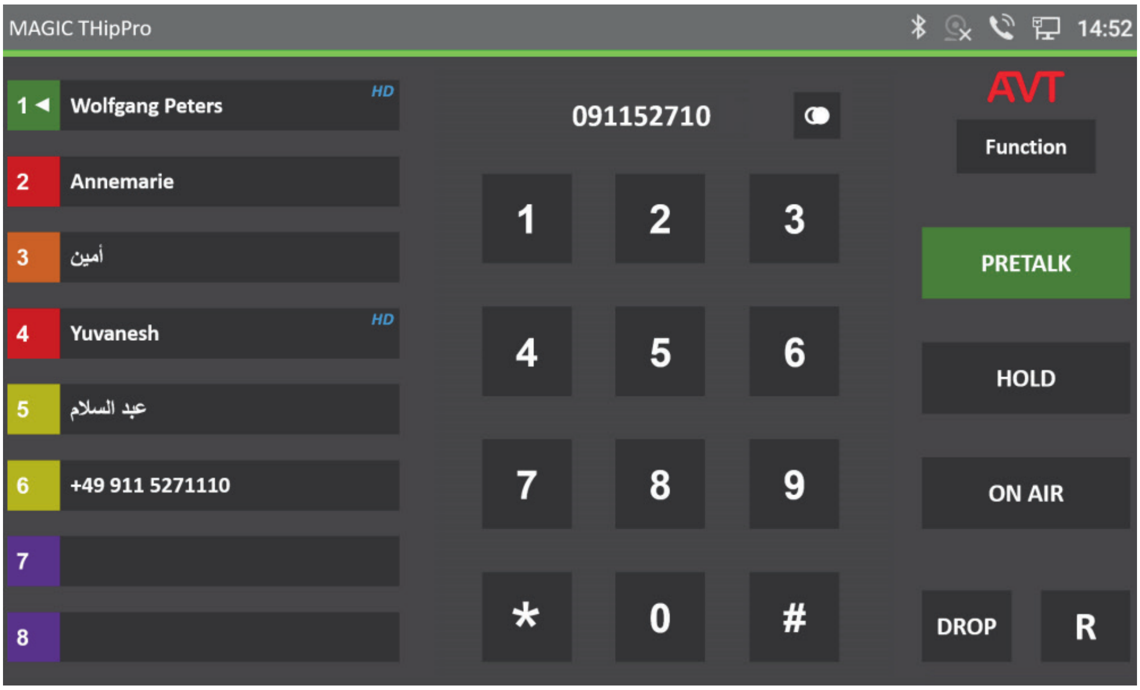
The HDMI interface also enables parallel output of the user interface to an external screen.

MAGIC TH2plus allows the simultaneous use of up to two, MAGIC TH6 up to six, MAGIC THipPro/Lite and MAGIC Server Basic up to eight and MAGIC Server Professional up to 48 **MAGIC PhonerSet** workplaces, that can be used in addition to or instead of the respective PC software.



MAGIC PhonerSet GXV3470

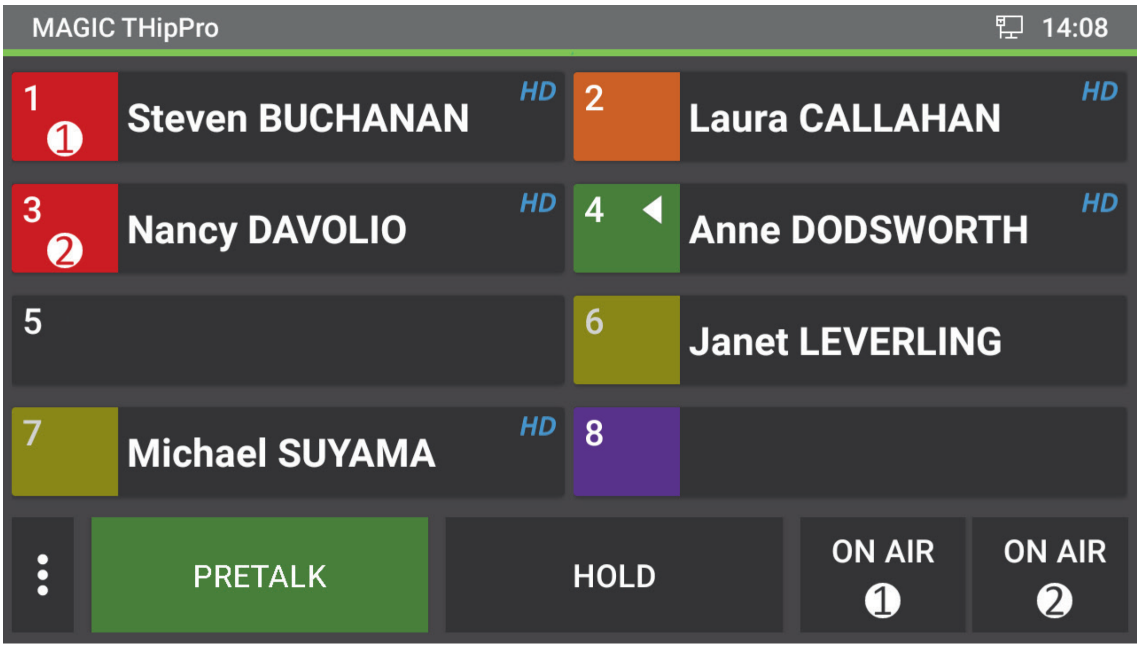
MAGIC PhonerSet with MAGIC THipPro



MAGIC PhonerSet with MAGIC THipPro (1-Fader Mode)

MAGIC PhonerSet GXV3450

MAGIC PhonerSet with MAGIC TH6



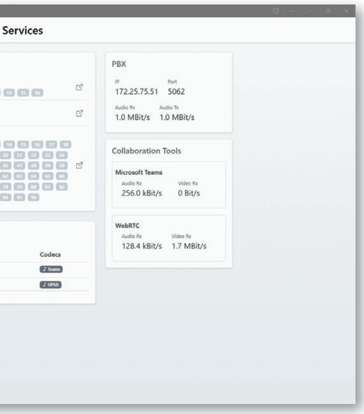
MAGIC PhonerSet with MAGIC THipPro (2-Fader Mode)

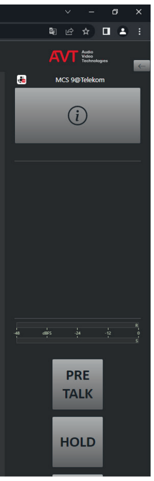
MAGIC Collaboration Services

MAGIC Collaboration Services

- Connection to WebRTC & MS Teams
- Adds video calls to Telephone Hybrids
- Compatible with MAGIC Server and hardware Telephone Hybrids
- Integrates in Telephone Hybrid PC/ web clients
- Identical workflow as regular calls
- Participation in scheduled meetings
- Central server platform for all systems
- Compatible with all Teams users
- Video output via NDI

AY FOR HYBRIDS





MAGIC Collaboration Services is an optional upgrade for the AVT Telephone Hybrids (except MAGIC TH1Go) which allows the integration of **MS Teams** and/or **WebRTC.** Each caller line of the hardware telephone hybrids can be upgraded to a **MAGIC Collaboration Channel** so that MS Teams and WebRTC calls can be handled in the same way as normal telephone calls via the standard telephone hybrid user interface. For MAGIC Server Basic the maximum number of Collaboration Channels is 4 audio/video or 8 audio only channels; for MAGIC Server Professional the maximum number is 24 audio/video or 48 audio only channels. The audio signal is output like a regular call; for video calls, the video signal is displayed directly above the line and can also be output via **NDI**. A return image from the studio can also be transmitted via NDI. All functions such as PRETALK, HOLD and ON AIR or call forwarding work as usual with a single click. Mixed conferences between regular and WebRTC or Teams callers are also possible.

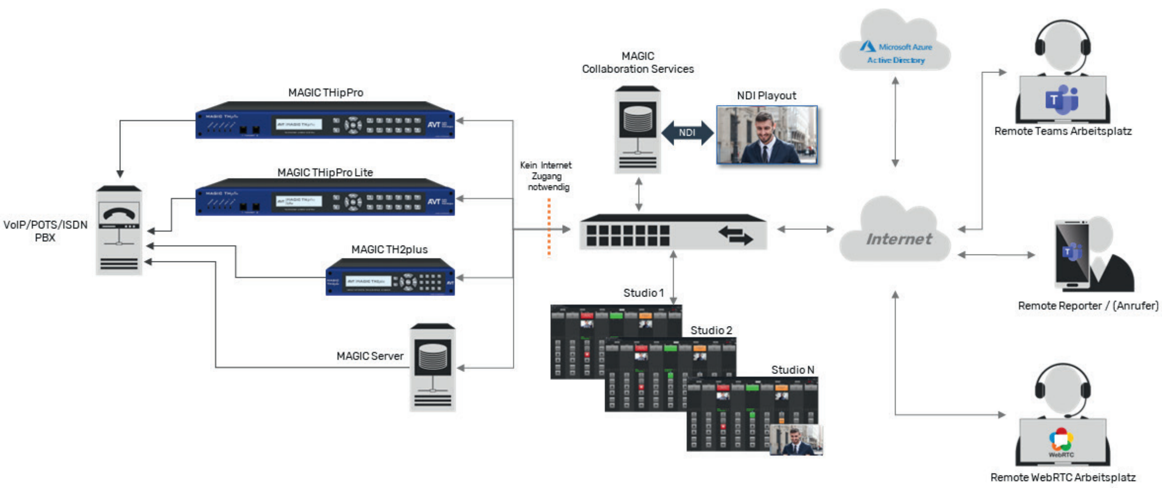
Collaboration Services is available in the **Basic** and **Professional** version.

The **Basic** version allows connections via **WebRTC** using a standard web browser. No additional software is required on the caller side. The caller simply needs to click on a web link that can be generated and sent out by email via the user interface of the AVT Telephone Hybrids. A connection can then be established with up to **15 kHz audio quality** (via the Opus audio codec). Additionally, video signals with a resolution of up to 1080p can also be transmitted.

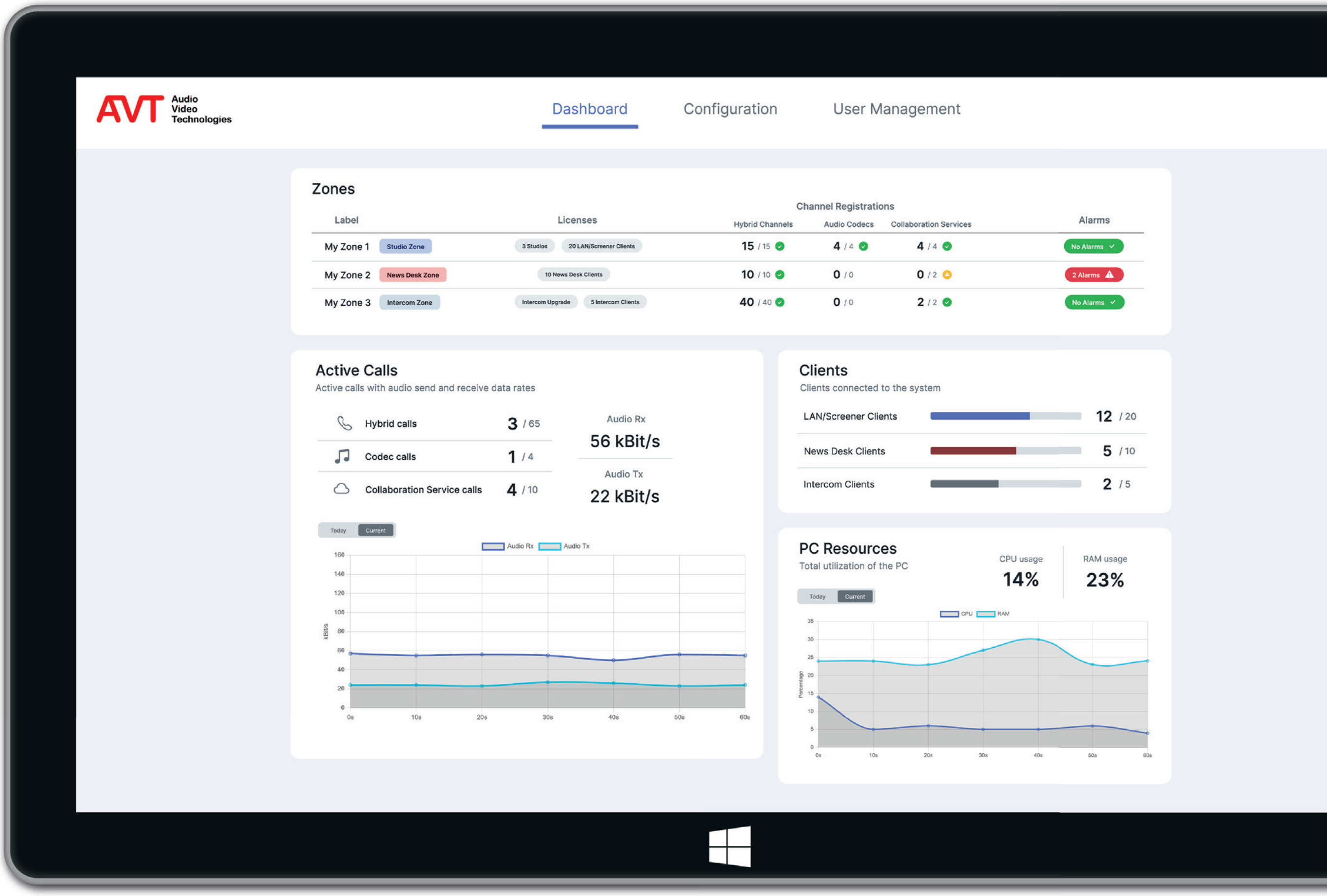
In the **Professional** version, in addition to WebRTC, **Microsoft Teams** is also supported so that reporters and correspondents who use Microsoft Teams can be integrated directly into live broadcasts. The audio transmission uses the **MS Satin** codec, which is very robust against packet loss thanks to an “elastic” buffer and provides an audio quality of typically 8 kHz. Video can be transmitted with 720p resolution. To use the function with Teams, a full version of Microsoft Teams is required, which is included in most Microsoft 365 plans. The MS Teams phone book of the organisation is then available via the telephone hybrid user interface in addition to the regular phone book, and it is also possible to include external participants outside the organisation in Teams calls or conferences via scheduled meetings.

With the **Extended Video In-/Outputs Upgrade,** dedicated NDI signal can be sent to the remote side instead of the return signal from the studio and up to eight video signals from individual callers can be output via NDI.

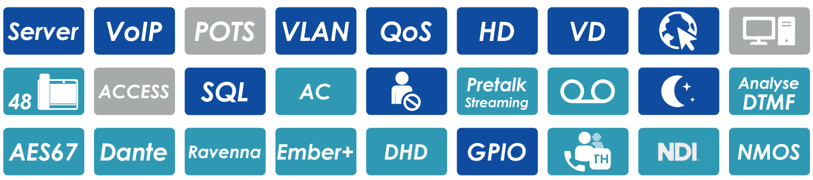
MAGIC Collaboration Services runs as an application on a separate server or PC that is connected to the Telephone Hybrids. It can also be installed on the same platform as the MAGIC Server. Using a Virtual Machine is recommended to be able to flexibly allocate more processing power in case the number of channels should be increased later on. The application runs as a service under the Microsoft IIS (Internet Information Services) service platform and is managed via a web interface.



MAGIC Server Basic & Professional



MAGIC Server



- Software Telephone Hybrid & Audio Codec
- Basic: 1 to 8 VoIP lines
- Professional: 8 to 96 VoIP lines
- Up to 24 studios
- Up to 48 Audio Codecs
- Support for all sound cards (Dante, Ravenna, AES67, analogue, digital...)
- Central administration & configuration
- LAN & Screener Web Clients
- Integration of MAGIC Collaboration Services

LUTION

The **MAGIC Server** expands the Telephone Hybrid portfolio with a scalable, software-based Telephone Hybrid & Audio Codec server.

MAGIC Server Basic starts with one VoIP telephone channel, one web operating client and one studio and can be extended to eight caller lines. A second studio can be added as well. For high-quality audio transmission, two lines can be upgraded with the Audio Codec functionality. The **Audio Codec Option** allows Opus and PCM connections. With the **Audio Codec plus Upgrade,** it is also possible to use ISO/MPEG Layer 2, ISO/MPEG Layer 3 and the MPEG4 algorithms AAC-LC, HE-AAC V1/V2 and LD/ELD which ensures full compatibility to the EBU 3326 AoIP standard.

Compared to the Basic version, the **MAGIC Server Professional** offers significantly more resources, so medium and large broadcasters with growing requirements can expand the system at any time.

The system offers 8 to 96 VoIP Telephone Hybrid channels, 24 studios with 48 application-specific web control clients for presenters, screeners and technicians, 48 recording booths (**News Desks**) for telephone interviews and six **Intercom** groups for communication with correspondents or remote stations.

Additionally, up to 48 of the 96 channels can be used for high-quality stereo audio transmission with the Audio Codec and Audio Codec plus Upgrades (licenced per channel). 48 audio or 24 audio/video WebRTC / MS Teams collaboration services can be added.

Both server variants can be run on a physical server as well as on a virtual machine, with the Basic variant also being able to be run on a current desktop PC / workstation.
The VoIP connection is made - as with the hardware Telephone Hybrids - via individual SIP accounts of a local PBX or a cloud PBX.

The MAGIC server supports the classic telephony standard G.711 (3.1 kHz audio bandwidth) as well as the much higher quality HD Voice standard G.722 (7 kHz audio bandwidth) by default.

For high-quality audio transmission, the audio codec lines can also be expanded with the MPEG Layer 2, MPEG Layer 3 and MPEG4 algorithms via the **Audio Codec Plus** upgrade.

Integration into the local audio network is possible via a virtual AES67 sound card with up to 192 channels, an external USB- Dante® / Ravenna / MADI sound card (RME Digiface) or even a classic internal or external ASIO-capable multi-channel sound card.
The MAGIC Server is configured via a modern and intuitive web interface that allows central management of all lines, studios and workplaces.
The Ravenna module also supports **NMOS**. An optional NMOS module is available for the virtual sound card.

The web user interfaces for the respective workplaces/operating stations are also delivered via the server, so that an installation at the workplace is no longer necessary. The desired user view is controlled via a user login.

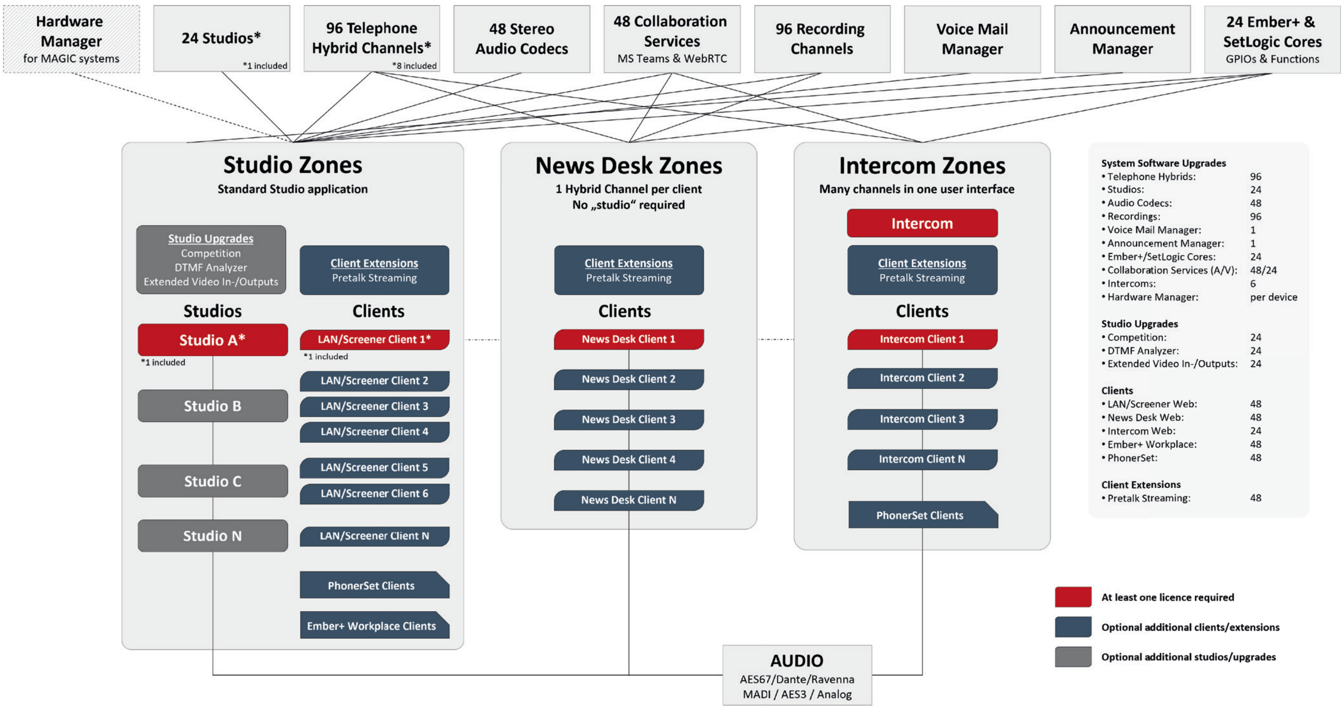
The **Redundant Server Token** for MAGIC Server Professional allows you to set up a redundant server in a very cost-effective way. In case of a backup, the full licensing of the first server is transferred to the second server. Both servers need to have a permanent connection to the internet. The switching of the servers can be signalled e.g. via a virtual GPIO contact using Ember+ or DHD SetLogic.

For users who already own AVT hardware Telephone Hybrids of the MAGIC THipPro family including Audio Codecs connected via ACconnect – and who want to continue using them – we offer the **MAGIC Server Legacy**.
This allows the classic PC clients to be replaced or supplemented by modern web clients.
This does not change the audio and VoIP infrastructure connected to the hardware systems.

The web-based user interfaces significantly minimise the time required for a system update, as only the server needs to be updated. There is no need for installation at the individual workplaces.

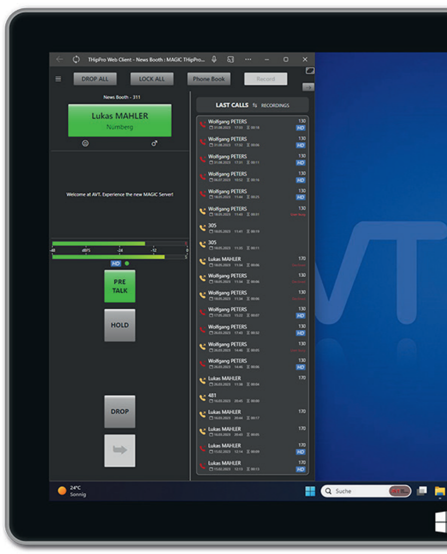
The MAGIC Server Legacy can then be upgraded to a fully-fledged MAGIC Server Professional if required.

Licence Structure MAGIC Server Professional



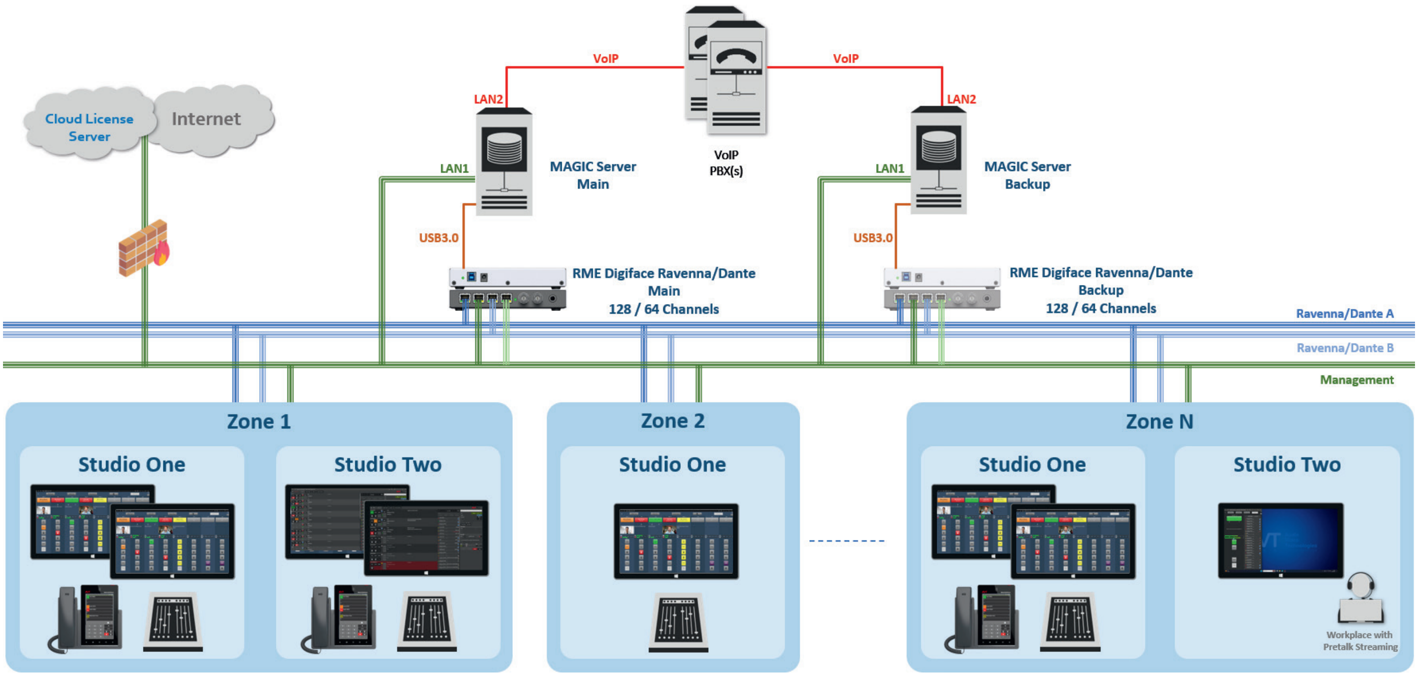
Client Options

MAGIC Server LAN Web Client (with A/V Collaboration Services)

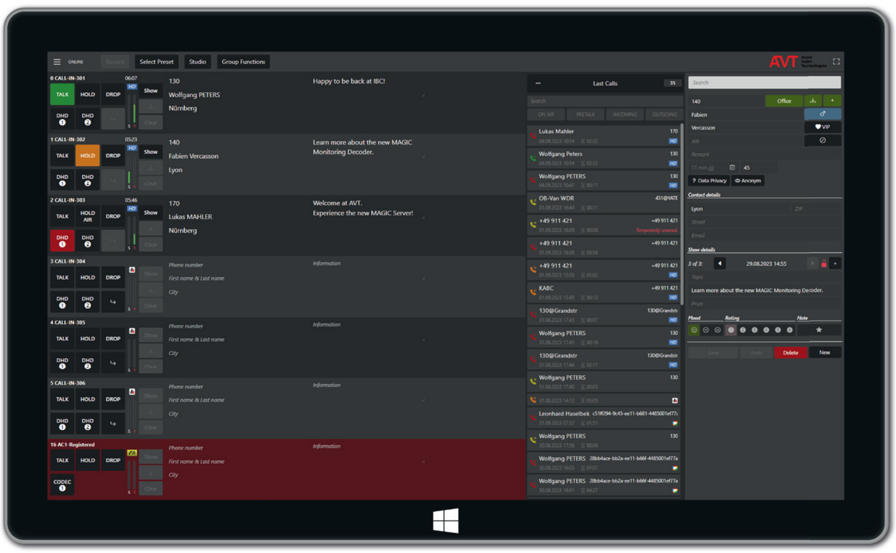


MAGIC Server News Desk

Example Application



Web Client (Window View)



MAGIC Server Screener Web Client

Applications and Options

Studio & Talkshow Applications

Talkshows and telephone interviews are a central component of classic radio operation. MAGIC Server allows line resources to be allocated to 2 (Basic) or 24 (Professional) studios. To each of these studios workplaces with differing views can be assigned individually. The combined **LAN/Screener Web Client** allows an optimized view for the presenter, for the screener, for the control room/technical staff as well as for the data manager for handling caller information.

Especially for remote operator stations that cannot simply be connected to an audio core, the **Pretalk Streaming Extension** for the LAN/Screener Web Client offers an easy way to connect audio. Pretalk calls can thus be made via a headset connected to the PC.

The **Recording** function enables telephone interviews to be recorded. This can be assigned to a workplace to provide a presenter or screener with a direct recording on-air or in the pretalk conversation.

Alternatively, each recording channel can be configured as an answering machine including individual announcement, recording duration, etc.

In addition to the web PC clients, workplaces that use a **PhonerSet** to operate individual or multiple telephone channels can also be set up in each studio.

For users who wish to integrate Telephone Hybrid channels into DHD or LAWO mixing console/VSM environments, this can be implemented via an **Ember+ Workplace** licence.

This requires the **Ember+/DHD SetLogic** upgrade (required per audio core), which provides a wide range of system and line information as well as control options via virtual GPIOs. In addition, a wide variety of functions for controlling and displaying information (e.g. name of a caller) is available. The Ember+ implementation supports both consumer and provider modes.

Users who play game shows with listeners or invite listeners to events (e.g. concerts) can do this comfortably via the **Competition & Event Management** Studio Upgrade.
This option allows the creation and evaluation of competitions, including the management of the prize contingent. Tickets for all participants can be managed for events. All lists can be exported as PDF or Excel files.

For game shows or sporting events, the **DTMF Analyzer** studio upgrade can be added. There are three modes to choose from: Standard, Gameshow and Event. While in standard mode only the digits received via DTMF are displayed, in gameshow mode the caller who answered the fastest is marked. In event mode, short text information is displayed instead of digits. For football matches, for example, reporters on site can inform the control room that a goal has been scored, even in hold mode.

News Desk Application for MAGIC Server Professional

For telephone interviews in small recording booths, the **News Desk Web Client** offers a simplified user interface. One telephone line is displayed per workplace. The telephone conversation can either be recorded externally or, more conveniently, via the optional central **Recording** function.

An optional **Collaboration Service** can also be assigned to the News Desks telephone channel, enabling connections via WebRTC and MS Teams. With WebRTC, high-quality recordings are possible due to the OPUS codec being used. The conversation partner only needs a PC with a current web browser and a high-quality headset.

The audio connection can either be made via a physical audio line (AES67 / Dante / Ravenna / MADI etc.) or via the optional **Pretalk Streaming** connection.

The optional **Ember+/DHD SetLogic** upgrade (required per audio core), allows signalling and control via DHD SetLogic and/or Ember+.

Intercom application for MAGIC Server Professional

With the **Intercom Upgrade** and the associated **Intercom Web Client**, up to 32 lines can be displayed in one overview. The upgrade also provides special features such as a level booster in the direction of transmission, line identification via speech synthesis as well as signalling in case of line interruption. Up to six intercom groups can be set up per server.

The Intercom application also allows the use of the optional **Collaboration Services**, which can be assigned to one or more telephone channels. This makes it very easy to connect correspondents and reporters via WebRTC or MS Teams with much better quality.

The optional **Pretalk Streaming** allows a direct pre-talk with an intercom partner via the control PC and a connected headset - regardless of whether the channel is activated in the audio matrix or not.

The **Recording Upgrade** can additionally be used to implement centralised recording of caller lines on the server.

In addition, MAGIC Server Professional supports automatic centralised call management using the **Voice Mail Manager Upgrade** to implement an answering machine function with study-related announcements and central recording.

As an alternative to the Web Client, a **PhonerSet** can also be used for operation. A maximum of 12 lines can be displayed here.

The optional **Ember+/DHD SetLogic** upgrade (required per audio core), allows signalling and control via DHD SetLogic and/or Ember+.

Licensing & Update Support

MAGIC Server's licence management is cloud-based, so an internet connection is mandatory. This enables prompt functionality upgrades, flexible floating and test licences.

The licence is always delivered as a lifetime licence and includes a 12-month update period.

For **MAGIC Server Basic**, a cost-effective annual support contract with update access can be optionally added after these 12 months. This ensures that the latest version can always be used and that all security updates are available promptly. In addition, the support contract includes remote support for four support tickets per year via our support portal.

Alternatively, we offer a **renewal licence** that can be purchased at any time and includes update authorisation for 12 months. In addition, one year of basic support is included for four support tickets via our support portal.

For **MAGIC Server Professional,** the conclusion of a support contract is mandatory after the first 12 months. The annual costs of the support contract are derived from on a baseline amount and the number of activated Telephone Hybrid channels at a defined date.
In addition to the update authorisation, the support contract includes remote support for 12 support tickets per year via our support portal.

MAGIC THipPro ACconnect Upgrade

MAGIC THipPro with MAGIC ACip3 or MAGIC AC1 Go

DSP	IP	Secure Streaming	AES67	Ember+	DHD	2-Codecs	SD card	Backup	230V
G.711/ G.722	Layer 2	Layer 3	AAC-LD	AAC-ELD	AAC-LC	HE-AAC V1/2	PCM	apt-X	Opus

- Full Audio Codec integration in MAGIC THipPro Telephone Hybrid LAN and Screener Software user interface
- Audio Codec control via additional caller lines
- 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

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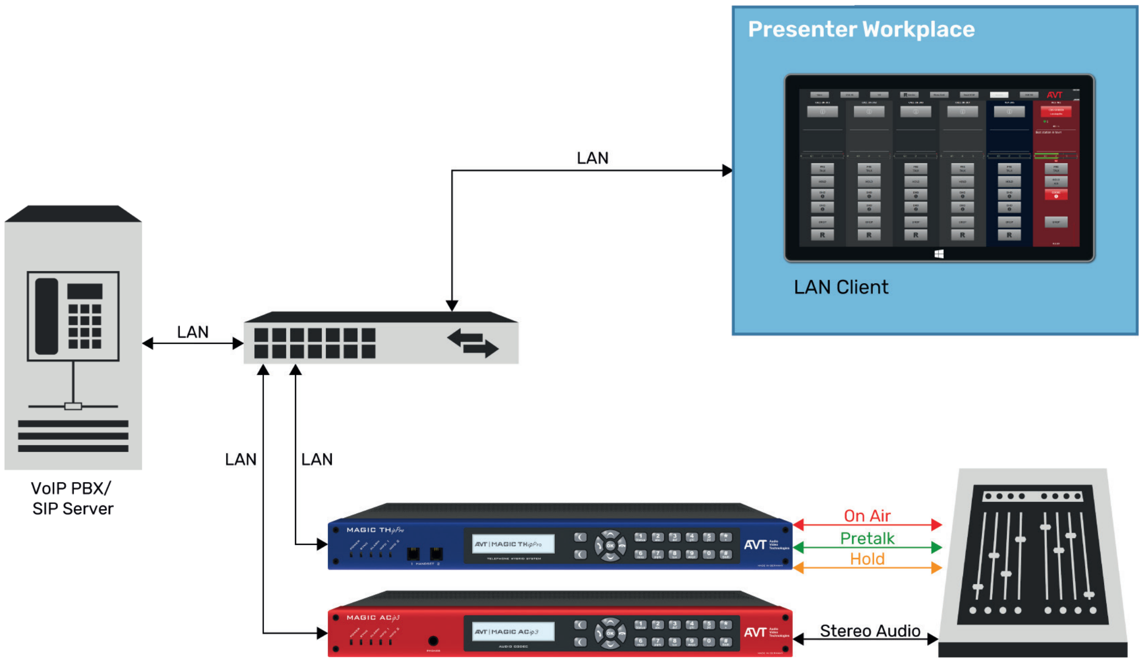
With the **IP Audio Codec MAGIC ACip3** or **MAGIC AC1 Go** and the **MAGIC THipPro Telephone Hybrid** AVT provides an integrated **all-in-one solution**: only one Management Software is required to control both Telephone Hybrid and Audio Codec. The Audio Codec is displayed as an additional caller line in the MAGIC THipPro LAN or Screener Software. Incoming calls to the Audio Codec 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.

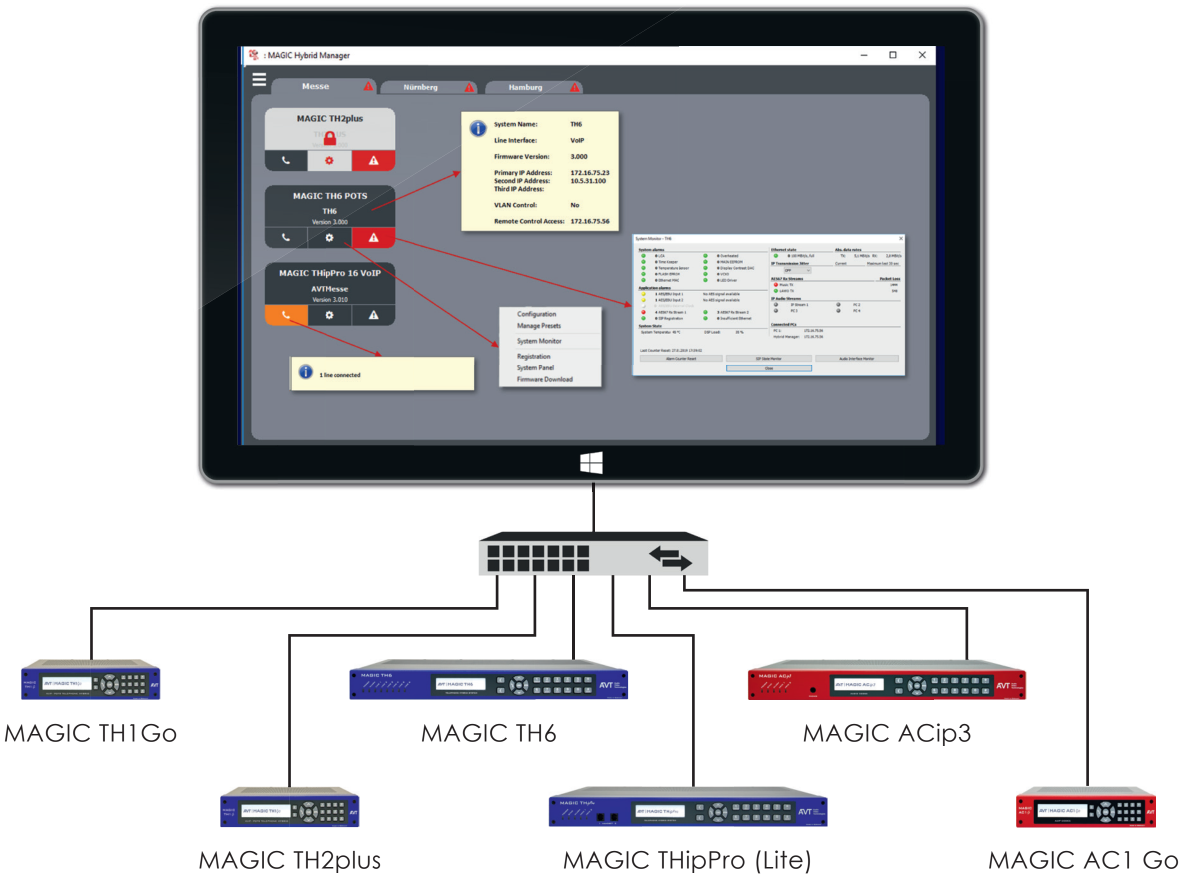
Details of the Audio Codec connection such as e.g. coding algorithm and bit rate are available via the caller line's Info button. Furthermore, a **common phone book** can be used.

If the Telephone Hybrid is used with the **Admin upgrade for up to six studios** the Audio Codec can be integrated in each of the configured studios - or also only in selected studios. The available Audio interfaces of the MAGIC ACip3 or MAGIC AC1 Go 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 systems can be integrated – if each MAGIC ACip3 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



Central Management Software

If a broadcasting station has **several AVT Telephone Hybrids and Audio Codecs**, these can be displayed and administered in the System Manager. This central management software allows a clear presentation of all MAGIC TH1Go, MAGIC TH2plus, MAGIC TH6 and MAGIC THipPro/Lite Telephone Hybrids as well as MAGIC AC1 Go and MAGIC ACip3 Audio Codecs. Even if the systems use **different software versions**, all devices are supported.

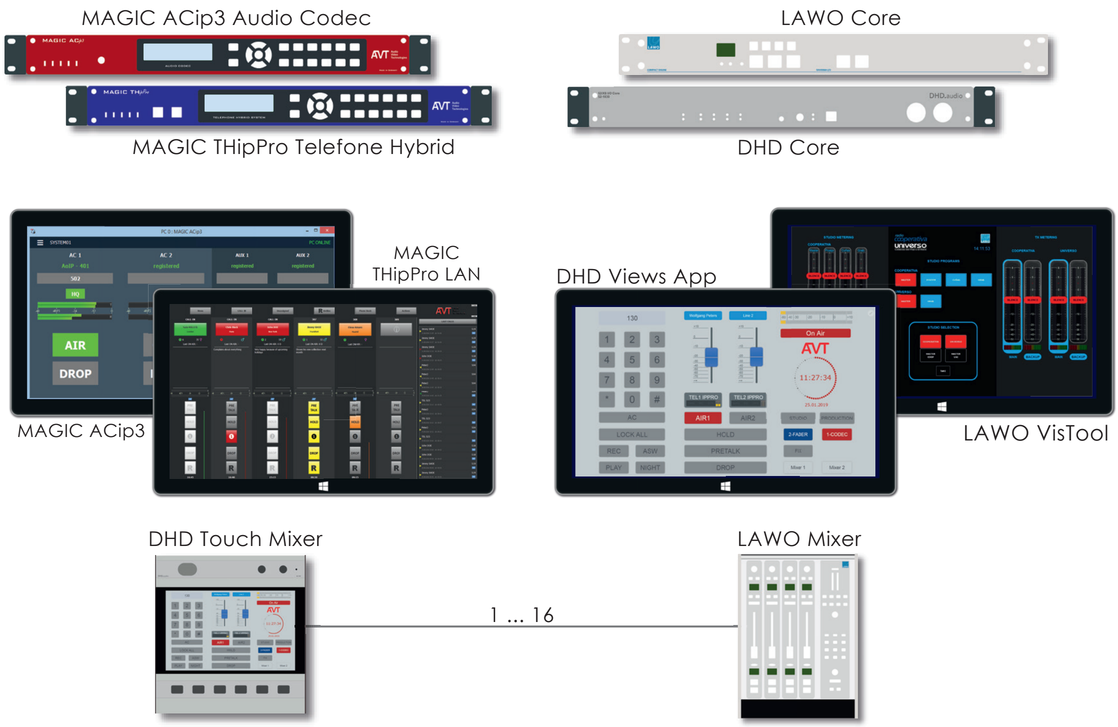
For each device, possible **alarms** and the **operating status** (in use or in configuration) are displayed. The query is made cyclically via SNMP. System Monitor, Registration Dialog, System Panel and Firmware Download can be accessed.

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.

A firmware update can be initiated on selected systems from a central location. Configurations and line assignments can be activated conveniently via the System Manager. Presets can thus be loaded quickly and easily.

One System Manager license is required per system.

Ember+ Control Protocol



Control via Ember+ capable Systems

Using the *EMBER+ Consumer/Provider* control protocol, all Telephone Hybrids and Audio Codecs of the **MAGIC** family can also be operated via graphic software-control panels such as the **DHD View App** and **LAWO Vistool** or directly via a mixer in addition to or as an alternative to their own software clients like **MAGIC THipPro LAN** and **MAGIC ACip3**.

In addition to the convenient entry of the phone number, almost all essential functions can be used to answer callers, switch to PRETALK, HOLD or ON AIR.

n addition to displaying the call number, the integrated EMBER+ function can also be used to display caller information. For example, the name or location of the caller - if stored in the database - can be visualised.

Special functions such as loading presets, blocking lines, start/stop recording, fader start & stop etc. are of course also possible.

All states of the system can be transmitted to DHD/LAWO and the operating elements can be highlighted in colour.

Almost all functions can be performed autonomously - without an additional PC. Exceptions to this are database queries for displaying caller information and recordings via AVT client PC software.

Depending on the system, up to 96 programmable GPIOs, 8 EMBER+ providers and 16 EMBER+ consumers as well as some predefined workflows are available, enabling excellent integration between AVT and Ember+ capable systems like DHD or LAWO.

One possible application is for example the implementation of up to 16 editor workstations that can conveniently operate a Telephone Hybrid system via a (virtual) mixer.

For more complex Ember+ configurations, the **Ember+ Consumer Extension Upgrade** (DHD) and **Ember+ Dial Pad Extension Upgrade** (Lawo) offer simplified programming.

ANSWERING MACHINE

MAGIC THipPro VMS

MAGIC THipPro Voice Mail System

DSP | VoIP | VLAN | HD | Ember+ | DHD | GPIO | 230V | SQL | Dante | Ravenna

- Answering machine for 8 or 16 VoIP lines
- Individual announcement texts can be played
- 2 x analogue and 4 x digital Audio inputs/ outputs
- Up to 32 callers can be recorded at the same time
- Up to 5 workplaces can be configured
- Optional redundant power supply

The MAGIC THipPro Voicemail System for connection to VoIP-capable PBXs or VoIP main lines is used for automated recording of up to - in the maximum configuration level - 32 simultaneous calls, which are stored as WAV files on a file server.

The DSP-based recording system has either 8 or 16 caller lines. Existing systems with 8 lines can be upgraded to 16 lines via a hardware upgrade if required.

Configuration and control is done via the included MAGIC THipPro VMS Client. **Up to five VMS Clients** can access one system.

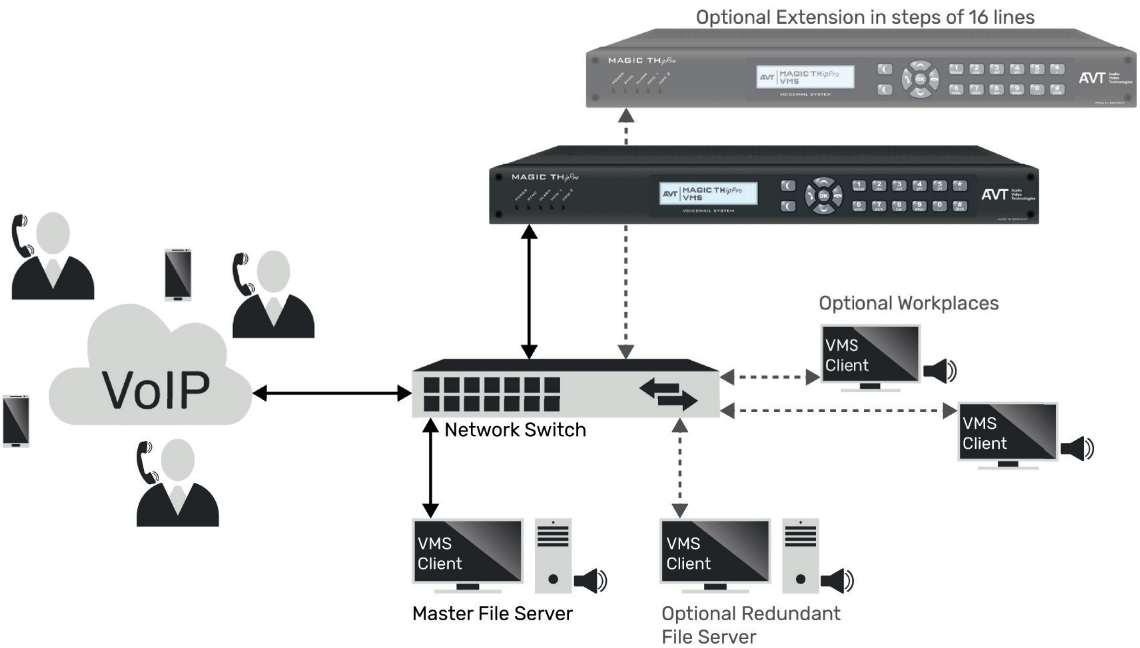
Up to four 8-line or two 16-line systems can be managed together in the graphical user interface, so that in total a maximum of 32 recording channels are available.

Up to two clients can take over the master function. A master records the callers and saves the recording as a WAV file. For performance reasons, the recordings should also be saved on this PC. With the **File Server Redundancy** Upgrade, a second master PC can automatically take over the function if this PC fails.

MAGIC THipPro VMS with optional redundant power supply

MAGIC THipPro VMS PC Software

Example Application



Functions

- **Announcement**

Each caller line - alternatively also a line group - can be assigned an individual announcement text.

The announcement texts are non-volatilely stored in the system or can be dynamically updated via a VMS client (configured as master PC), e.g. via a playout system.

The announcement texts are saved to the VMS system by simply importing a WAV file with an audio bandwidth of max. 8 kHz, so that callers can hear an announcement in very good quality in HD Voice mode (G.722).

The duration of an announcement can be up to 30 seconds, whereby it can be played as a continuous loop without recording or several times (1...3 times) before recording.

- **Recording**

A recording is saved as a WAV file on the configured master PC, whereby only the incoming signal is recorded.

Both the directory structure for saving the WAV files and the file name can be freely configured.

In addition, it is possible to generate a corresponding metafile for each recording in DBE format (DIGAS from DAVID Systems).

If the user also owns a MAGIC THipPro Telephone Hybrid system, automatically existing information of the caller from the SQL database of the system is used to generate the file name and the metafile.

The recording time can be limited in steps of 20 seconds up to 10 minutes for each caller line or for each line group separately.

Instead of recording, it is also possible to play a message only. In this operating mode, you can also configure call forwarding for any caller line/group to any phone number after the announcement.

For backup purposes, the optional redundancy upgrade allows parallel recording of all calls on a second PC.

A management system for managing the recordings is not part of the system, but the recordings can be played back on any VMS client.

- **Rejection**

For all lines an identical rejection can be configured if the maximum recording time is reached.

- **Monitoring**

Thanks to the integrated Pretalk Streaming function, a call can be listened to live via a button, communication with the caller is not possible.

INTERCOM GATEWAY

MAGIC THipPro VoIP Intercom

MAGIC THipPro VoIP Intercom

DSP VoIP POTS VLAN QoS HD ACCESS SQL AES67 Dante Ravenna Ember+ DHD GPIO NMOS 230V

- Intercom Gateway for 8 or 16 VoIP lines
- 2 x analogue and 4 x digital Audio inputs/ outputs
- Optional 8 x AES67 or 32 x Dante/Ravenna Interfaces
- Up to 20 workplaces
- Up to 10 systems in one GUI
- Optional Pretalk Streaming
- Optional Dual LAN module
- Optional redundant power supply

The Intercom System is a DSP-based gateway, available in two versions: MAGIC THipPro 8 VoIP Intercom with 8 lines and MAGIC THipPro 16 VoIP Intercom with 16 lines, the 8-lines system can be extended to 16 lines by a hardware upgrade.

The 1U device has eight digital audio lines (four AES/EBU interfaces) and two analogue audio inputs and outputs in each version alongside two handset/headset interfaces.

With the **AES67** licence, eight additional audio channels can be used through a software upgrade.
Alternatively, a native **Dante** or **Ravenna Module** with 32 audio channels can be equipped. AES67 is supported by both modules.

With the **Dual LAN Module**, the system can be expanded by two additional LAN interfaces, so that a total of four LAN interfaces are available.

With the **HD Voice** (G.722) upgrade, connections can be established and received in 7 kHz quality.
A **redundant 5V desktop power supply** is available via a hardware upgrade.

MAGIC THipPro VoIP Intercom also supports the **Pretalk Streaming** function. With this Upgrade the LAN connection between the control PC and the system can be used for Pretalk which means no Audio cabling is required. The conversation with the caller is done via the PC sound card or a USB Headset. In total, up to ten Pretalk Streaming licences can be activated, these licences can be assigned fixed to a PC workplace.

One license for one workplace is already included in the delivery, additional licenses for **up to 20 workplaces** (one license per system per workplace) are optionally available.

The software allows the user to answer and dial calls and view receive and transmit levels. In case of communication problems, a **variable level booster** can be used in the transmitting direction. Calls can be forwarded and a **Hold** function is also available. Lines can be preassigned for simplified operation, and lines can also be **predefined with a VIP function,** so VIPs can call this line exclusively. A **common SQL database** is available for all systems. A **WAV file** can be played as line identification when a call is received, another audio file can also be played for alerting if a connection has unintentionally been terminated on the network side. The application can control a maximum of **ten devices.**

Accessories

Headset Monaural
with RJ or USB cable

- no control of the system
- USB requires optional Pretalk Streaming

Headset Binaural
with RJ or USB cable

- no control of the system
- USB requires optional Pretalk Streaming

Handset with USB cable

- no control of the system
- USB requires optional Pretalk Streaming

Handset with RJ cable

- no control of the system

Analog telephone with display

- For MAGIC TH1Go
    - Control of the caller lines of TH1Go
- For MAGIC TH2plus (only in POTS Mode)

POTS-VoIP-Gateway

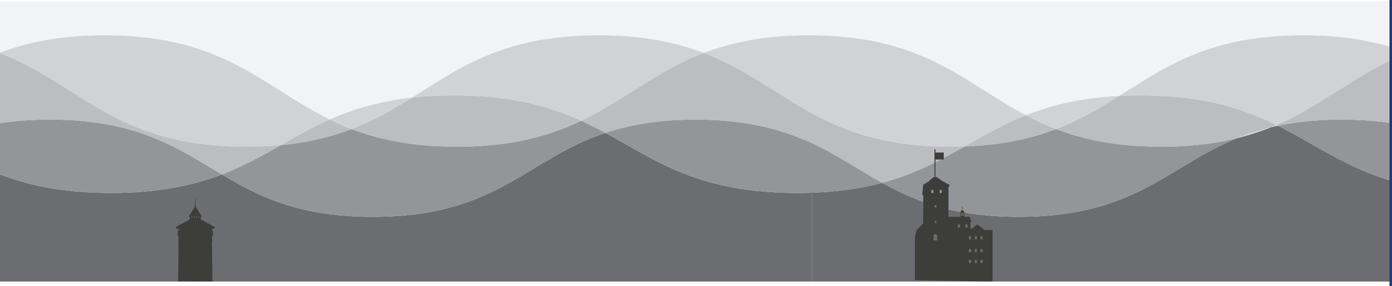
- Integration of a VoIP system in POTS environments
- For conversion of VoIP signals for existing POTS infrastructure

GSM adapter

- Quad Band (850/900/1800/1900 MHz)
- GSM Adapters for POTS
- GSM/LTE Gateways for VoIP

DUAL Mounting Kit

- Common front panel
- To mount two 1/2 x 19” units side by side in a 19” rack



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