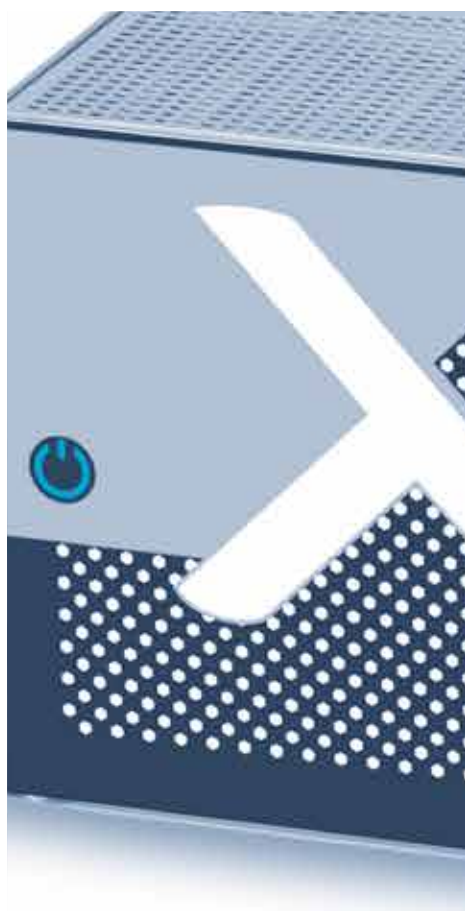


Audio, video and
communications
for broadcasters

A large, dark grey silhouette of broadcast equipment, including a boom arm with multiple microphones and a large circular subwoofer or speaker, set against a white background.

TECHNICAL SOLUTIONS FOR RADIO



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PRESENTATION

OUR COMPANY

For more than 40 years AEQ has been developing, manufacturing and selling audio and video equipment for broadcast communications, production and automation systems for Radio, Television and other media.

EQUIPMENT DEVELOPMENT AND MANUFACTURING

A strong R&D team is the heart of our activity. We create and implement our own technology in our equipment, primarily designed for the broadcast market. We develop professional audio and video solutions for radio, television and other media.

SALES AND SUPPORT

Our equipment is sold worldwide through a network of distributors that provide local technical and commercial support to users. Also, through our headquarters and our own offices, we provide direct technical and commercial support anywhere in the world. More than 5500 radio studios in 100 countries are currently operating AEQ equipment.

KEY FACILITIES FOR RADIO AND TV

From AEQ we offer 'turnkey' solutions. We provide services for engineering, installation and setup, commissioning and user training for installations of any size. All this is accomplished in close collaboration with our dealers, final customers and, if required, third party manufacturers.



A complete range of radio mixing consoles places AEQ in a prominent place on the world market in regards to design and manufacturing of such products.

Atrium



Modular audio digital console with AoIP connectivity up to more than 1000 channels and 96 faders that can be set up on individual channel setting pages

The Atrium digital audio mixer was designed for high capacity and operational flexibility. IP connectivity with DANTE™, RAVENNA, AES67, SMPTE ST 2110-30 and SMPTE ST 2110-31 protocols, as well as the ability to extract and insert audio channels embedded in SDI video, makes installation and use simple and flexible. It has been developed to be used in multiple environments, at radio broadcasting stations, media production and television studios.

The Atrium console is based on a control surface independent of the process and audio engine or engines. Up to 6 Atrium consoles can drive and be driven from a single engine. This means that a single engine can service 6 consoles on the AoIP network. A surface can also control other AEQ AoIP products integrated with the system.

The control surface is modular and desktop flush-mount. Each module holds 6 faders and it is possible to install up to a maximum of 96 fader channels. Each control surface is complemented with a powerful control and monitoring module.

Atrium features a powerful set of touch screens, encoders, indicators and programmable keys. This avoids unnecessary steps and procedures in the console workflow, always keeping accurate information at sight, making operation simple and safe.

The control surface is fully configurable: for classic workflows, such as A / B selection on each channel, input and output or N-1/multiplex channels - or mixed workflows, with a totally flexible programming of the different keys.

A virtual console application is available to operate the surface remotely.



Applications:

- Radio broadcast and production studios.
- TV Sound production.
- Recording and PA.
- OB Vans.

The ATRIUM console incorporates the specific features for ON AIR broadcasting: automatic monitor mute, cough mute, fader start, control signalling etc. In addition, it incorporates programmable keys to control external equipment: communications, intercom, visual radio. At the process level, it has an immense capacity for adjusting frequency, dynamics, multi-band mixed process, delay and reverb. Also unattended mixing is enabled through autogain and automix functions.



The channels have individual vu-meters, 100 mm motorized faders, color display and 8 programmable keys. As an option, a touch screen can be added for each 6-channel module. On that screen the vumeters are also represented and the processes can be set and adjusted. Each module handles 8 pages or configuration layers.

The control and monitoring module has a touch screen and 24 programmable and contextual keys. It also incorporates VU-meters, CUE speaker and control and studio sections. You can add an additional touch screen, with loudness measurement.

Atrium

ATRIUM's Audio Engine is the **X_CORE**. Completely modular and physically detached from the control surface, it handles all the Inputs, Outputs, Mixing and Routing. Also, X_CORE handles the console's dynamics, equalization, filters, delays, reverbs, etc. Further, several X_CORE can work together in larger installations.



ATRIUM's multi-channel audio connectivity supports virtually every format used in audio production today; SMPTE 2110-30 and SMPTE 2110-31 with NMOS control, SMPTE 2110-30 via Dante Domain Manager, embedded audio in SDI up to 3G, AES67 with RAVENNA control, native RAVENNA AoIP, AES67 with Dante control, native Dante™ AoIP, and AES 10 MADI.

Inputs and Outputs:

Modular engine based on a 4 UR frame, expandable through additional frames. Each frame can be equipped with redundant controllers and power supplies, 20 process cards, and 21 slots to configure inputs and outputs flexibly, among others:

- Frame to Frame Link module, 1,024 audio channels.
- Double 3G SDI card, with two SDI inputs and two outputs for video with embedded audio, and connection to the internal audio bus of the consoles with 2x16 audio input channels and 2x16 output channels.
- 64-channel AoIP audio connection card based on the Dante™-AES67-SMPTE ST 2110-30 standard.
- 128-channel AoIP audio connection card based on the RAVENNA-AES67-SMPTE ST 2110-30 and SMPTE ST 2110-31 standard.
- MADI-AES10 Link, multi-channel audio module with 2x64 channels.
- 8 balanced analog input/output module.
- 4 AES / EBU stereo digital input/output module.
- Module with 4 microphone inputs and 2 headphone outputs.





Forum IP SPLIT



Modular audio digital console, with AoIP connectivity and with more than 180 I/O channels and 24 faders

Applications:

- Radio broadcast and production studios.
- TV Sound production.
- OB Vans.

Inputs and outputs:

The **FR_CORE** modular engine incorporates mic./line input for talkback and self-control, monitor and headphone outputs for control and studio, as well as stereo CUE output. As an option you can incorporate 64 MADI-AES10 inputs / outputs (incompatible with AoIP Dante™/AES67) and redundant power supplies.

It also supports up to 14 cards to configure inputs and outputs flexibly, of the following types:

- 4 AES / EBU stereo digital inputs/outputs configurable as SPDIF.
- 2 configurable phantom power microphone / mono line inputs.
- 8 analog line inputs (individually configurable in stereo pairs, 4 pairs).
- 8 analog line outputs (individually configurable in stereo pairs, 4 pairs).
- 32 AoIP Dante™/AES67 inputs / outputs (individually configurable in stereo pairs, 16 pairs).
- 4 USB digital stereo inputs / outputs.
- Digital telephone hybrid.

- Modular, flush-mount control surface, 4 to 24 channels with 4 direct routing keys per channel.
- Independent, modular engine for process, inputs and outputs, with mixing, dynamics, equalization and filters functions.
- It incorporates the specific features for ON AIR Broadcast production: automatic monitor mute, cough mute, fader-start, control signalling, etc.
- Programmable keys for external equipment control: communications, intercom, visual radio ...
- Option for multi-channel connectivity to AoIP Networks or MADI Links.
- Optional integrated telephone hybrids.
- Software application option that enables control and monitoring with a touch screen.
- Available software control surface replica for remote operation.
- Wiring kit.





Forum LITE

Up to 92 channels or more, and 12 faders

It's the cost-efficient combination of the Forum IP SPLIT modular control surface up to 12 faders with a **M_CORE** compact engine.

Same features as Forum IP SPLIT, with limited capacity of faders, logical inputs and outputs.



Inputs:

- 4 mono mic / line inputs, configurable phantom power.
- 2 stereo USB digital inputs (I / O).
- 4 AES / EBU stereo digital inputs, configurable as SPDIF.
- 12 analog inputs (individually configurable in stereo pairs, 6 pairs).
- 16 optional AoIP Dante™/AES67 inputs (individually configurable in stereo pairs, 8 pairs).
- 2 inputs for telephone line.
- 64 optional MADI-AES10 inputs (32 ST pairs).

Outputs:

- 2 USB stereo digital outputs (they are I / O).
- 4 AES / EBU stereo digital outputs configurable as SPDIF.
- 8 analog outputs (individually configurable in ST pairs, 4 pairs, or 3 pairs and secondary headphones).
- 16 optional AoIP Dante™/AES67 outputs (individually configurable in stereo pairs, 8 pairs).
- 2 outputs for telephone line.
- 64 optional MADI-AES10 outputs (32 ST pairs).
- Monitor and headphone outputs for control and studio.
- Stereo CUE.





Capitol IP



Ultra compact audio digital console, with AoIP connectivity, up to 92 channels and 8 faders

- Flush-mount or embedded control surface with 8 channels and 2 direct routing keys per channel.
- Independent **M_CORE** engine for inputs, outputs and processing, with mixing, dynamics, equalization and filter functions. Optionally with redundant power supply.
- Incorporates specific features for ON AIR broadcast features: automatic monitor cutting, cough cutting, fader start, signaling control.
- Programmable keys for external equipment control: communications, intercom, visual radio...
- Optional multichannel connectivity through IP or MADI.
- Software application option that facilitates control and monitoring with a touch screen.
- Available control surface replication software for remote operation.
- Optional double telephone hybrid.
- XLR or RCA wiring kit, on demand.

Applications:

- Broadcast/Production radio stations/studios.
- High-level journalist booths.
- Sound Production on TV.
- OB Vans.



Inputs:

- 4 mic / line mono inputs, configurable phantom power.
- 2 stereo USB digital inputs (I / O).
- 4 AES / EBU stereo digital inputs configurable as SPDIF.
- 12 analog inputs (individually configurable in stereo pairs, 4 pairs).
- 16 optional AoIP Dante™/AES67 inputs (individually configurable in stereo pairs, 8 pairs).
- 2 inputs for telephone line.
- 64 optional MADI/AES10 inputs (32 stereo pairs).

Outputs:

- 2 USB stereo digital outputs (they are I / O).
- 4 AES / EBU stereo digital outputs configurable as SPDIF.
- 8 analog outputs (individually configurable in ST pairs, 4 pairs, or 3 pairs and secondary headphones).
- 16 optional AoIP Dante™/AES67 outputs (individually configurable in stereo pairs, 8 pairs).
- 2 outputs for telephone line.
- 64 optional MADI/AES10 outputs (32 stereo pairs).
- Monitor and headphone outputs for control and studio.
- Stereo CUE.

Olympia 3



Commentary unit with AoIP connectivity for three users with robust and functional design

With PoE, Video over IP transport, process and local mixing, it incorporates an intercom user panel. It's controlled by software and does not need a specific matrix for its local and remote connection. Any device with AoIP Dante™ - AES 67 connectivity can share incoming or outgoing audio with Olympia 3. Ideal to work coupled to Venus 3 audiocoders.



To us it's a great pride to be able to say that AEQ is present in the generation and transmission of all the audio signals for radio and TV stations of the world, in major multi-venues and multi-sports events, world and continental championships in athletics, football, basketball, cycling, swimming, handball, hockey, skiing, Formula 1 GP, as well as mobile and fixed systems in large sports stadiums.



Applications:

In addition to its use in large sporting events, it also has utility for radio stations. It can be used in journalist booths, in simultaneous translation sets and "off tube" booths to comment events in radio studios, in mobile units, and also in small deployments for the individual broadcasting of an event of any kind.



A full range of digital interfaces and routers, puts AEQ in a prominent place in the design and manufacture of IP audio systems for radio and television stations. IP connectivity according to DANTE™/AES67 standard makes simple an flexible the installation and use of the devices.

Netbox 32 AD/ 8 AD/ 4MH



Analog and digital audio interfaces to the IP network

NETBOX 32 AD

Connect to the audio network over IP up to 32 input and 32 output channels, divided into 16 mono analog and 8 stereo digital. Stereo digital can be configured as AES / EBU or SPDIF. It also incorporates 16 GPIs and 16 GPOs.

Applications:

Especially suitable for master control rooms and dispatch rooms, or to expand or relocate matrices on TDM BUS type X_CORE or Netbox DSP.

NETBOX 8 AD

Grants access to the IP audio network up to 8 input and 8 output channels, spread over 4 mono analogue connections and 2 stereo digital connections. Stereo digital can be configured as AES / EBU or SPDIF. The second digital stereo can also be switched to a USB connector. It also incorporates 4 GPIs and 4 GPOs.

Applications:

Useful for IP access to consoles without IP connectivity, workstations, and journalist booth consoles.



NETBOX 4 MH

Allows connection to the audio network via IP up to 4 input channels for microphone or analog line and 4 output channels, for stereo headset and analog line. Incorporates 4 GPIs and 4 GPOs. It has additional GPIOs for signaling terminals such as Studiobox. It can be powered by PoE.

Applications:

Radio booths, journalist booths, connections in the mixed area of sports events. Preampfier for microphones.

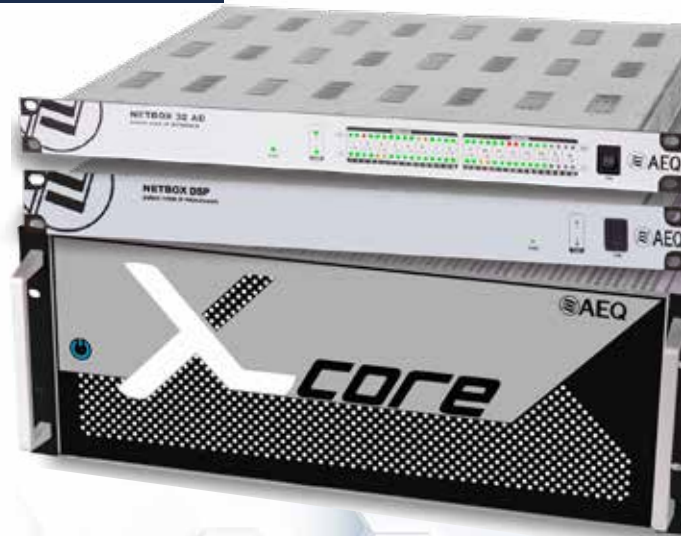
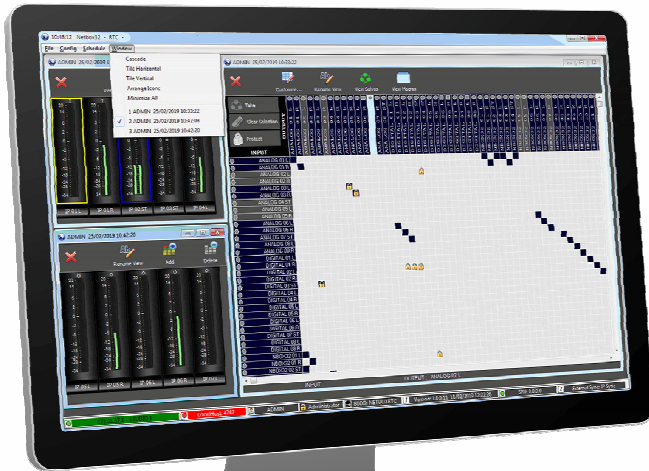
STUDIOBOX

Desktop signaling terminal. Interact with a digital console directly or through NETBOX 4MH. With "Ready" and "On Air" lights and cough cut buttons, remote PFL and 5 configurable buttons.





X_CORE/ Netbox DSP/ Netbox 32 AD MX



Audio matrices with AoIP connectivity

Applications:

Especially suitable for switching the air signals and the distribution of audio inputs and outputs from the studios, in central control rooms, dispatch rooms and other technical spaces.



X_CORE

Mixing, processing and distributing audio matrix, up to 5120 x 5120 circuits, for broadcast. Fully modular and redundant. Its inputs and outputs are through cards of different types in flexible quantities: digital AES / EBU, analogue line, microphone and headphones, long-distance dark fiber optic links, in 64-channel MADI format and, proprietary fiber links with more 1000 channels, as well as multi-channel audio embedded in SDI video, among others.

In addition, through AoIP cards, we can include Dante™/AES67 IP audio inputs and outputs in the matrix, or IP audio streams in SMPTE ST 2110-30 format. There are also RAVENNA AoIP cards that also support AES67 and SMPTE ST 2110-30 and SMPTE ST 2110-31 audio, with NMOS control.

An X_CORE frame can incorporate as many AoIP cards as necessary, and these can be installed in one or several Gigabit Ethernet networks. If more I/O cards are needed than can be accommodated in a frame, frames can be linked via multi-channel cards.

NETBOX DSP

Mixing, processing and distributing audio matrix. Versions with 64, 96, 128 and 160 audio inputs and outputs to the IP Dante™/AES67 network. Mix combinations of Dante network inputs over any of its up to 160 outputs to the IP network. 64 inputs can be processed and returned to one output, or they can be added to any other existing output. They also incorporate 16 GPIs and 16 GPOs. As all its inputs and outputs are on the Dante network, to obtain analog or digital inputs and outputs, it must be accompanied by audio interfaces, audio consoles, or other equipment with IP Dante™/AES67 connectivity. A large matrix in only 1RU in size.



NETBOX 32 AD MX



Mixing, and distributing audio matrix of 64 x 64 circuits. Able to mix combinations of its 16 analog, 8 stereo or dual digital and 32 inputs from the Dante™/AES67 network over any of its 64 outputs (16 analog, 8 stereo or dual digital and 32 IP). It also incorporates 16 GPIs and 16 GPOs. Perfect for medium and small installations.



AEQ has a range of both portable and stationary audiocoders, compatible with most third-party codecs, over IP and ISDN interfaces.

CONTROL SOFTWARE

With the ControlPhoenix AudioCodecs Management software, TOTAL CONTROL is available.

This application allows to identify all the audiocoders of the Phoenix family of a local network to control them in a coordinated way from a PC or group of PCs. Remote equipment can also be controlled through the internet, thus allowing an integral management of the communications network.

For each codec there is a very friendly graphic window for configuration and another for operation. In addition, there is a summary window in which the general status of all system codecs is displayed.



There are three versions: a free one in which two audiocoders are shown per screen, another one in which all the codecs of a system or fleet are controlled from the same screen, and a third one in which, in a broadcasting network, the distribution of programs and contributions from affiliated stations, are managed, automatically or manually, with the support of an audio matrix.

ALGORITHM AND COMPATIBILITY

The AEQ audiocoder can be connected with third-party equipment through the SIP communication protocol, in accordance with the N / ACIP standards of the EBU TECH 3326.

In addition, if they connect with an AEQ codec, they can use an exclusive set of communication and control assistance tools for the unit. Among them, the Smart RTP communication establishment system, which facilitates the connection with compatible stationary codecs, avoiding the need to make or hang up the call, or specify the coding modes manually.

In addition to the algorithms prescribed in N / ACIP, our codecs incorporate a selection of OPUS encoding algorithms that guarantee high quality audio with low delay, as well as G711, G722, MP2, AEQ-LD, PCM, among others. Optionally they can also include some AAC algorithms.





Phoenix Stratos and Venus 3



Double stereo full duplex audiocoders with IP connectivity for the most demanding applications

- Carrier grade performance and reliability for the most demanding applications.
- Control by **Phoenix Control** software.
- High performance and reliability. Dual AC power supply.
- It allows two full duplex independent stereo / dual or four mono connections to two different destinations.
- Double channel totally independent for program and coordination or backup with their respective returns.
- Controllable through a simple user interface on PC that allows the local or remote management of one or more computers.
- It has a continuous data channel. It carries auxiliary data for remote equipment control.
- Remotely monitored: includes SNMP server that allows viewing its status and alarms.
- The equipment has two pairs of balanced analogue audio inputs and outputs at line level, duplicated with AES / EBU digital audio inputs and outputs.
- It has 4 GPIs and 4 GPOs as general purpose inputs and outputs for signalling and control.

Applications:

- STL links (Studio Transmitter Link).
- Broadcasting networks.
- Contribution from outside the station.
- Audio link between remote intercom panels.



Phoenix Stratos



Additional connection for ISDN and X21/V35 lines and physical user interface on the front panel.

Phoenix Venus 3



Double IP port and double ancillary data port, redundant DC power supply option, Dante™/AES67 local connectivity option.



Phoenix Mercury



Stereo and bidirectional IP audiocodex, that allows stereo / dual or mono connections

Applications:

- STL links (Studio Transmitter Link).
- Broadcasting networks.
- Contribution from outside the station.
- Digital hybrid for IP telephony.

- Small format for desktop or rack (two devices can be installed in a 19" rack unit).
- It allows a full duplex stereo / dual or two mono connection to the same destination.
- Single channel for program with its return.
- Controllable through a simple user interface on PC that allows the local or remote management from one or more computers.
- It has a continuous data channel. It carries auxiliary data for remote equipment control.
- Remotely monitored: includes SNMP server that allows viewing its status and alarms.
- Balanced analogue audio inputs and outputs at line level. Optionally you can incorporate AES / EBU digital audio inputs and outputs.





Ultra-compact IP Audiocoder for personal use. Small size, friendly and easy to use

Simple front control panel:

- HELP button to request help from the station.
- Microphone and headphone level adjustment.
- Call / answer button.
- Hang up button.
- Outgoing and incoming signal level indicators.
- Other basic indicators of equipment status and communication.

Two complementary software applications:

- Talent Pilot, to complement the front panel.
- Phoenix Control, remote control compatible with all AEQ codecs.



- With high definition preamp and OPUS encoding.
- Connection for dynamic or condenser microphone or headset.
- Supports all types of headphones, low or high impedance. It allows you to listen to the station's audio, the local audio, or a mixture of both.
- It allows the sending of the audio mix to stereo line and Bluetooth outputs.
- Excellent transmission quality, both in mono and stereo.
- It has a magnificent dynamic range.
- Negligible delay in voice and studio return.
- Supports home Internet connection, by ADSL or optical fiber and 3G/4G/5G wireless data connection through router/modem with SIM card.
- Supports Bluetooth connection with smartphones to reach the studio PC.
- Send HD voice quality audio to the broadcast console.
- Several TALENT can be connected in cascade.
- Various power supplies:
 - USB output from a PC.
 - Power outlet between 5 and 12 volts (vehicle battery or powerbank).
 - In addition, it comes with an adapter for the conventional electrical network.



Applications:

- **Basic:** Simple sending of the local microphone to the studio, with return to the headphones mixed with the listening of the sent audio.
- **Live phone calls:** We can interview from a smartphone connected by bluetooth. The audio from the phone line is mixed with the microphone and sent to the studio.
- **Commentator:** Microphone send mixed with a stereo line input provided by a line jack or by bluetooth.



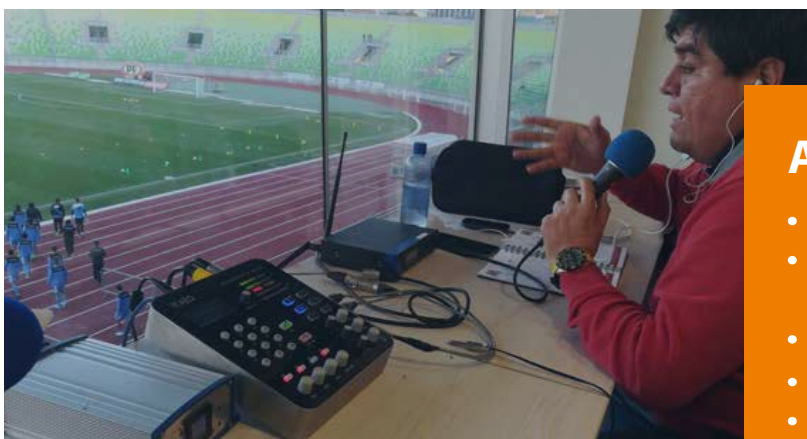


Phoenix Alio



Portable stereo IP audiocoder for outside broadcasting, optimized for outdoor use

- Designed and optimized for easy use in the most diverse broadcasting environments, including music events.
- Shock and splash resistant finish. It is also delivered with a carrying bag.
- Portable mixer with four microphone inputs with switchable Phantom power. Two line inputs and outputs. Two headphone outputs. Volume control and TX / RX mixing. Individual bass and treble control setting for each input channel.
- Full duplex stereo communication. Software-activated option for a second stereo channel for technical coordination or backup.
- Front panel interface with keyboard and encoders. Graphic display and two LED precision VUmeters on the front panel.
- 12V DC power. External UPS option. Autonomy of more than 2 hours.
- Complete remote control application, not only with the connectivity functions, it also allows remote operation of the mixing and routing functions of the front of the equipment, in order to help or even replace the user.
- HELP button to request remote help from the technician in the studio.



Applications:

- OB Contributions.
- Usable in OB operations by non-technical staff.
- Remote broadcasting of events.
- Sports Commentary.
- Off tube booths console.



System IP



VoIP Talk-show and multi-conference system

The AEQ SYSTEL IP offers great savings both economically and from a workflow perspective with regards to Talk-show or phone-in On-Air productions. The Audio quality is also drastically improved if compared to the older PSTN TBU's or Telephone Hybrid Systems. Allows for VoIP calls and SIP based Telephone Service provider connectivity for Cloudbased or Corporate PBX as well as the possibility to connect to older systems based on PSTN (POTS) or ISDN Lines. AEQ SYSTEL IP is an essential for your Phone-in or talk-show production.

It dynamically distributes its 8 or 16 lines between 1, 2, 3 or 4 studios.

The operation allows, among other actions: dialing or picking up calls, putting them on hold or CUE, control the sending and return levels, routing them to auxiliary circuits, sending them to onair, block them onair, or hang them. You can choose between operating in the form of call queues or several simultaneous connections onair with groups.

Applications:

- Broadcast telephony.
- Conference calls in the air.
- Multiple commentators connections.
- Talkshow.
- Technical coordination.
- IP hybrid mode with multi-line and multi-studio possibilities.

SYSTEL IP 16 "Engine"



With 1RU for 16 simultaneous IP telephone lines, 4 additional lines for operator IP phones, 4 digital inputs / outputs, 2 analog inputs / outputs and 32 Dante™/AES67 protocol AoIP inputs / outputs, works with up to 4 studios.

SYSTEL IP Basic; for 8 simultaneous IP telephone lines (expandable to 16).

SYSTEL IP 16; for 16 simultaneous IP telephone lines.

Control Software Applications



SYSTEL IP ORIGINAL, consists of a PC control application and a conventional IP phone.



SYSTELSET+, is a special IP phone with a touch screen for control.

Telephone Hybrids



In addition to the latest IP telephony systems, AEQ offers digital telephone hybrids to incorporate calls from conventional telephone lines into radio programs with the highest audio quality.



Visual Radio



Automatic generation of visual content for radio

Visual radio generates visual content that accompanies the radio program for broadcast, both in internet environments and in traditional broadcasting. The system allows working with SDI cameras that offers HD video quality.

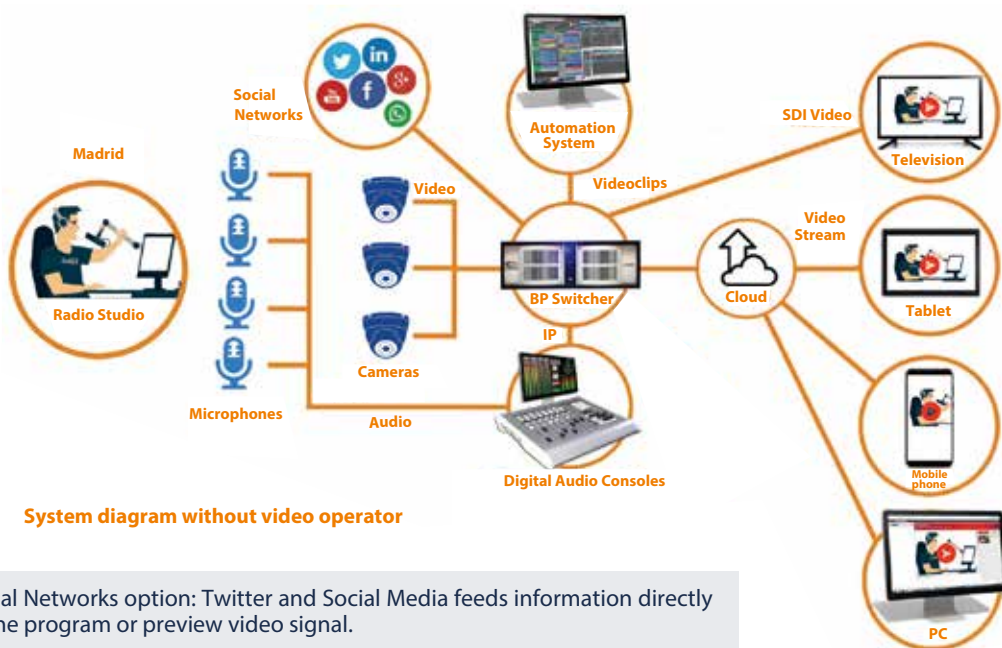
The generation of these contents is increasingly demanding and looks similar to the production of a TV channel itself. However, the difference between TV channels and the production for Visual Radio, is that in radio there is usually no dedicated staff to the realization and production of video.

BROADCAST PIX automatic video production systems are integrated with AEQ AoIP mixers and Netbox interfaces, which act as a source of commands and voice control for Broadcast Pix video switches and camera controls.

AEQ audio mixers and interfaces, determine in which microphone there is audio, and integrated with PTZ cameras, which collect the images of what is happening in the studio, so together with the Broadcast Pix video switcher, enable manual, or automatic and unattended, realization of the video, detecting the voice of microphones connected to AEQ digital audio mixers or AoIP interfaces.

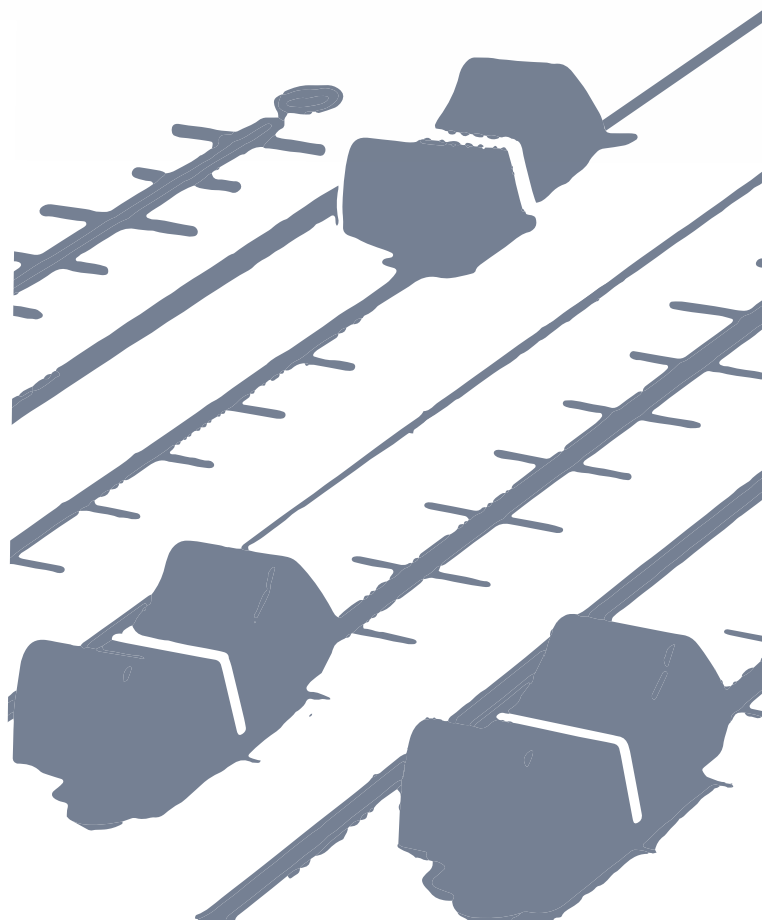
Applications:

Automatic generation of visual content of radio programming to simultaneously broadcast on the web and applications for PCs, mobile devices, and television channels.





WE ARE CRAZY ABOUT RADIO



CAT.RADIO.22_02

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