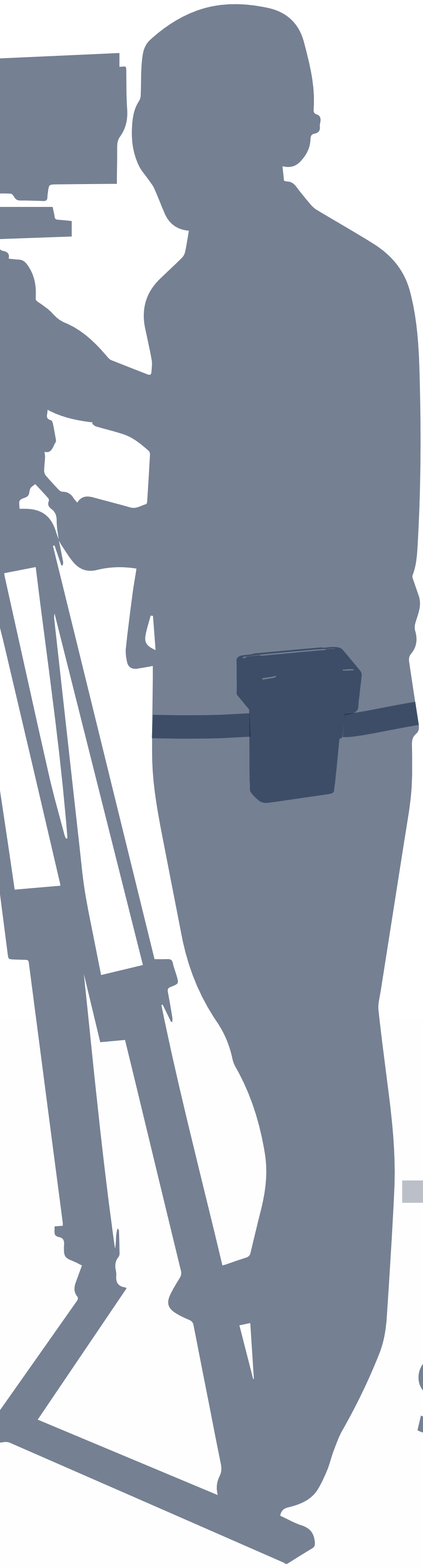


Audio, video and communications for broadcasters

TECHNICALS SOLUTIONS FOR TV

INDEX

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

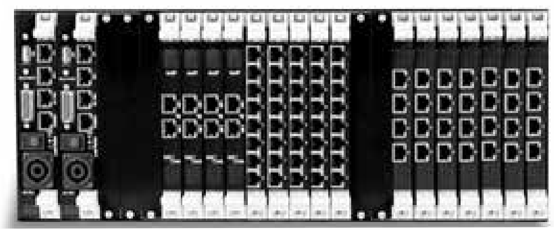
From AEQ we offer 'turnkey' solutions. We provide services for engineering, installation and setup, commissioning and user training for installations of any size. All this is accomplished in close collaboration with our dealers, final customers and, if required, third party manufacturers.

X-Core

Audio and Intercom matrix with AoIP connectivity

Broadcast audio mixing, processing and distribution matrix. When properly configured, it can perform as a general purpose audio matrix, or specific for video and television production environments, as an intercom audio matrix or with mixed capabilities. It can also operate as the audio engine for one or several mixing consoles simultaneously.

For intercom systems and audio mixing consoles, it can manage up to 1024 inputs and outputs. When used as an audio matrix and combined with the TITAN router, up to 5 X_COREs can be linked together to reach a non-blocking matrix size of up to 5120 x 5120 audio circuits. System is completely modular and redundant.



Its inputs and outputs are connected through several kinds of interface boards which can be installed in flexible quantities: digital AES/EBU or S/PDIF, analogue line or microphone level, headphones, long-range dark-fiber links in 64 channel MADI format and proprietary 1024 channel fiber links, point-to-point digital links for Intercom panels and VoIP cards among others.

X_CORE is based on a 4 RU height standard rackmounted chassis with three important parts:

- There are 20 slots reserved for DSP cards at the front of the chassis. These cards perform audio processing and communications crosspoints. This is done dynamically, allowing for the installation of backup cards, that in the event of a DSP card failure are able to automatically assume the function for any of the cards.
- There are two kinds of slots at the back of the unit. Two of them are reserved for the controller cards. One is of course required but a second one can be installed for redundancy. Further, there are 21 slots dedicated to I/O interface cards for the di_erent required audio formats.
- A back-panel is located in the middle of the Chassis and is the point of connection for the I/O Boards and the DSP's and also provides the transmission media for the system's 1024-channel TDM bus.

Also, using 64 input/output channel AoIP cards, the matrix can exchange Audio over IP inputs and outputs with devices using Dante™ / AES67 protocol, like Intercom user panels, commentary positions, I/O interfaces and mixing consoles, among others. A single X_CORE frame can include as many AoIP cards as required and they may be connected to one or several different Gigabit Ethernet networks.

These AoIP cards can also be configured for compatibility with AES 67 standard in order to share audio with third-party manufacturers not supporting Dante. It can also ingest and export audio streams associated to IP video signals compliant with SMPTE ST 2110-30 format.

Also, in order to exchange audio with Ravenna devices, a 128-channel AoIP card has been developed that supports AES67 audio as well as SMPTE ST 2110-30 and SMPTE ST 2110-31 formats. Control of this card in SMPTE 2110 mode is using NMOS protocol.

Also, to exchange audio with SDI video systems and their embedded audio channels, a card has been developed with two SDI input and two SDI output connectors up to 3G, which can un-embed and embed 2x16 audio channels groups.

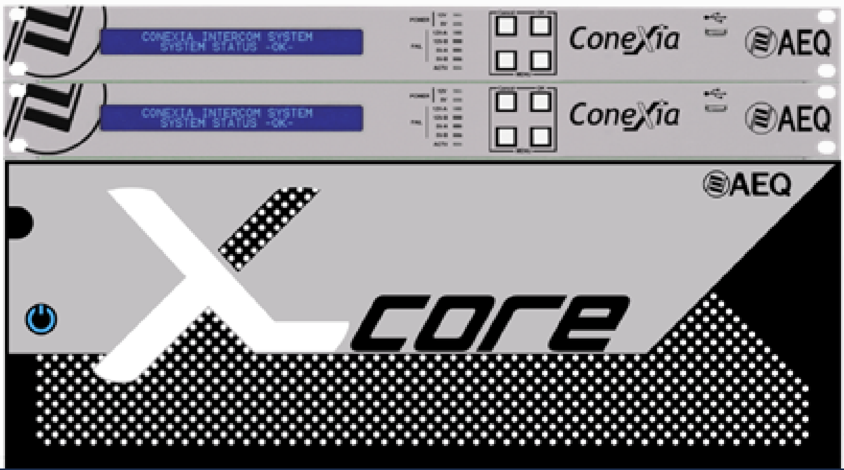
In case that more input/output cards are required, more frames can be linked together using multichannel interface cards.

The system is completed with redundant power supplies.

Applications:

- Intercom, audio and hybrid-use matrix Core.
- Audio mixing console engine for TV and radio production.

Conexia



Intercom Matrix Controller acting on the full or part of an X_Core Matrix

A CONEXIA controller or a pair of them (when redundancy is required) can drive a X_CORE matrix, or a part of it, in order to operate as a modular and redundant intercom matrix with up to 1024 circuits.

If the required intercom capacity is lower than 1024 circuits, the rest of the matrix can be used as a broadcast-quality IP audio matrix by routing the audio inputs and outputs to the different mixers and interfaces in the production center.

X_CORE system working like intercom matrix is compatible with all wired and wireless KROMA by AEQ user panels, and expands its interconnection capabilities through AEQ Phoenix audiocodecs and AEQ Systel IP phone systems.

This way, we can define CONEXIA as a true global audio solution able to manage all our communications and audio contributions. Its structure, based on a broadcast TDM X_CORE matrix, allows us to include the widest variety of audio formats currently in use in the market in a completely modular way, so we can select our resources and create final configuration according to each systems' particular requirements.

At the same time, this modularity allows us to operate with total system's redundancy, as it is able to provide automatic Back-up both for the system controller cards, the audio processing and crosspoint boards and even in the multichannel or discrete I/O boards installed in the system.

The internal TDM bus defines the size of the matrix up to 1024 x 1024 ports. All these characteristics allow us to have a broadcast quality (48 kHz, 24 bits) system providing enough flexibility and reliability to coordinately manage both the audio and intercom system in the Production Center.

Applications:

- TV Production Centers, sport events coordination (even multi-venue ones).
- Theaters, shows, halls...
- Public and private Control rooms.
- Emergency coordination.

Crossnet

Up to 190 channels compact Intercom Matrix

CrossNET is a compact and integrated Intercom solution. In a single height rack unit, we count on a matrix which is mainly based on Dante™ Audio over IP technology, also compatible with AES 67 and is able to manage up to 190 x 190 audio channels with internal, broadcast-quality audio processing.

Thanks to its scalability, from 40 x 40 to 190 inputs and outputs, the system offers a range of external direct connections: analog and digital ports, AoIP Dante™ and low bit-rate VoIP. The largest expression of the CrossNET Matrix is a 190x190 audio channels Intercom Matrix with the following port distribution:

An integrated, small user panel.

The front LCD screen, loudspeaker and micro-headphone input allows to use the proper Matrix as a small, 4-key user panel, which is always available to establish communications or monitor system audio channels where the matrix is installed.

- 12 four-wire, broadcast-quality, balanced analog audio ports for general purpose connections to external circuits such as audio consoles, I/O for PA, camera intercom or IFB's, etc.
- 8 digital audio ports (KROMA Legacy ports), providing backward compatibility with earlier KROMA systems, allowing the user to connect KROMA user panels from all series as well as interface cards.
- 20 low-bitrate KROMA Legacy VoIP audio ports that allow for the connection of remote user panels using narrow-band Internet connections, EasyNET party-line systems and, specially, the connection of Xplorer system for wireless beltpacks and virtual panels.
- Up to 128 Dante™ broadcast-quality audio over IP ports, that may be used to connect TP8000-series intercom user panels, Olympia 3 Commentary Units or whatever other compatible audio devices from more than 300 manufacturers using Dante™ and AES67 standards.
- 32 additional Dante™ broadcast-quality Audio over IP ports which may be used to connect Olympia 3 Commentary Units, mixing consoles, audio input/output interfaces, or whatever other compatible audio devices from more than 300 manufacturers using Dante™ and AES67 standards.

Applications:

- OB Vans.
- TV Production Centers.
- Sport events coordination, even multi-venue ones.
- Theaters, show halls and arenas.
- Public and private control rooms and emergency coordination.

TP8000

Audio is digitized and processed with 48 kHz sampling frequency and 24 bits resolution, providing a 20 Hz to 20 kHz bandwidth with minimal distortion. IP connectivity is included to provide an easy setup and a high-quality audio in DANTE™ format.

Intercom User Panels with Dante™ AoIP connectivity

Digital audio processing: acoustic echo cancellation, automatic voice level, tone and speech habits control for each operator. Expansion and ambient noise gate. Thoroughly-designed acoustics for the best sound naturalness and intelligibility. 16 keys, rack or table-top formats. Expansion panels can be chained to build up panels featuring up to 80 keys with 4 pages. Compatible with any KROMA and AEQ intercom matrix.

TP8116

19" rack 1U user panel with 16 programmable, keys organized in four different pages. This panel provides an individual volume control per each communication crosspoint, echo canceller and DSP. Features dual Dante™ AoIP, VoIP, as well as one analog and one digital audio port. All the info is shown in a graphic display with up to two text lines per key, plus a third line indicating the crosspoint's audio level.

EP8116

19" rack 1U extension panel with 16 programmable keys organized in 4 different pages. This panel provides a numeric keyboard for an easy calling management between the system phone interfaces. Also features a loop input / output that allows the connection of up to three extension panels to the same user panel.

TP8416

Desktop user panel with 16 programmable keys, organized in four different pages. This panel provides an individual volume control per each communication crosspoint, echo canceller and DSP. Features dual Dante™ AoIP, VoIP, as well as one analog and one digital audio port. Information is presented in a graphic display with up to two text lines per key plus a third line indicating the crosspoint's audio level.

Olympia 3

Commentary Unit with AoIP Dante connectivity including intercom user panel & IP Video transport and selection functions

Olympia 3 has been developed to be used both in large events with hundreds of commentary positions in a stadium, but also in modest installations where the commentary unit operates standalone or in a OBVan, integrated with its Intercom system. Being a commentary unit, it can simultaneously operate as an intercom panel simultaneously. It can be controlled in a hybrid way:

Olmypia 3 as Intercom User Panel:

- For this mode, the channel “COMMENTATOR 1” includes the required functionality to operate as an Intercom channel. The displays will adopt the “Intercom mode” and the keys will adopt the programmed intercom destinations or functions and the associate microphone and headphone will form part of the Intercom System.

Olympia 3 as Commentary Unit:

- The OLYMPIA 3 CU CONTROL application configures and controls the CU.
- The commentator channel 3 keypad also allows you to select the IP video source of the VIDEO LINK 4K system that displays the auxiliary screen for commentator guidance.

Outstanding features:

- Standalone commentary unit (CU), or AoIP connected with 8 channels via Dante™ protocol. Scalable architecture: simple routing to Dante™ IP devices; integrated in IP Intercom System, or connected to IP Commentary System Matrix.
- Standalone mono or stereo sound mixer with mixing, routing, tone and dynamics control. 3 commentator inputs and a dual-mono or stereo line level input. Listening of 8 remote and 2 local sources.
- It allows selection of 8 video input sources.
- Operates as an Intercom Panel at the same time as a Commentary Unit.
- Configurable as interpreter desk up to three languages.
- 3 oneGigabit IP ports per unit for redundancy, daisy chain and auxiliary data or video transport.
- Dual power supply: 48 VDC via PoE or external local power supply.
- Software Configuration and remote control.
- Rugged and ergonomic mechanics, suitable for indoors and outdoors locations.

Xplorer

Intercom System including wireless beltpacks and software for Windows and iOS

Xplorer is a communications system based on Xplorer WiFi wireless beltpacks and Xvirtual, an application for iOS and Windows devices with the same functionality that can be found in an Intercom Panel.

Xplorer Beltpack

Xplorer is more than a beltpack, it's an authentic wireless intercom user panel. Based on Wi-Fi technology, it is equipped with a 4 shortcut keys user interface -arranged in pages-, another two programmable keys, and a multifunction screen.

Xplorer can operate as a wireless beltpack for AEQ Intercom matrixes such as Crossnet or Conexia, in combination with series-8000 user panels, with Olympia 3 and also with devices running the Xvirtual App. When working with Conexia matrixes, it offers enhanced audio quality with G.722 encoding.

1 Programmable key. 2 Status indicator. 3 Programmable key. 4 4 programmable shortcut keys, or channel selection in Party-Line mode. 5 2.4 "TFT screen. 6 Wi-Fi signal level indicator. 7 Mute indicator. 8 Terminal name tag. 9 Battery level indicator. 10 4 LED mode indicators. 11 Input level indicator for each interlocutor.

Xplorer can also operate in Party-Line mode with other Xplorer or Easynet terminals.

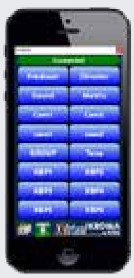
It´s compatible with 802.11 b/g/n networks using the 2.4 GHz band and 802.11a/n using the 5 GHz band.

Xplorer has double volume control by potentiometer (digital encoder), Mute function, 4 configurable physical keys on each page (total 16).

Its standard battery autonomy is around 17 hours, depending on conditions of usage. There are battery charging stations available for two and five simultaneous terminals.

Dimensions (length x width x height): 92 x 70 x 130 mm
Weight: 365g aprox.

XVirtual Application



The application can be installed on a PC, with Windows operating system, turning it into a User Panel and part of your Intercom System, only requiring a simple Ethernet connection.

In the same way, it can turn any Apple iPhone, iPod or iPad device into a Wireless Intercom Panel. Just connect it to a Wi-Fi network providing access to a Intercom matrix to build your Wireless Beltpack System.

This screen shows a 16-key intercom panel with Mute function. It is compatible with Crossnet and Conexia matrixes. When used with Conexia matrixes, it offers enhanced audio quality with G.722 encoding algorithm.

EasyNET

Matrix-less Party-Line intercom system supporting wired terminals and Xplorer wireless beltpacks

EasyNET is a low bitrate, VoIP based, Party-Line System with 4 independent channels that allows you to connect your panels in any already existing network infrastructure, or even using the public Internet for remote connections. It doesn't require a matrix to make cross-points. The system consists in three types of user panels: rack-mount, desktop and wired beltpack. It also supports Xplorer wireless beltpack.

This is one of the easiest to configure systems available in the market. Its installation only requires the connection of all the (up to 28) user panels to the same Ethernet network.

All terminals feature 4 keys to connect to one or more of the four audio channels available in this partyline system; advanced functionality that brings the capabilities of EasyNET closer to that of matrix-based systems, and provides great flexibility of use.

BS3004

A rack-mount terminal with 4 keys, integrated loudspeaker and optional microphone and headphone. It also includes 4 analog four-wire ports with gain control that can be used to integrate external audio sources in the system, such as camera audio feeds, etc.

BS3204

A desktop terminal with 4 keys, integrated loudspeaker and optional microphone and headphone. It also includes 4 analog four-wire ports with gain control that can be used to integrate external audio sources in the system, such as camera audio, etc.

BP3004

A wired beltpack with 4 keys and gain control. It is lightweight and compact and can be powered by PoE (Power over Ethernet), so only one connection is required.

Xplorer

Operates within EasyNet as a 4-channel wireless panel. WiFi access points must be installed. Xplorer beltpacks can operate in EasyNet mode without any wired panels installed.

Audiocodecs

AEQ produces a wide line of Audiocodec equipments, both portable and stationary, compatible with most third-party codecs, over IP interfaces. There is also a model available with legacy ISDN, X21 and V35 interfaces.

Audiocodecs can be used in television for three different functions:

- Extend intercom systems by providing connectivity to external work places, such as ENG teams or OB Vans.
- Incorporate phone calls accessing as VoIP that need to be recorded or included ON AIR through the sound mixing console.
- Incorporate high-quality audio that needs to be recorded or used ON AIR through the sound mixing console.

Phoenix Alio

Audiocodec portable IP stereo 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 communication, with a bidirectional stereo channel. Option of a second bidirectional stereo channel technical coordination or backup.
- Front panel user interface.
- HELP button to request remote help.
- 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.

Scenarios of use:

- Stereo and commentary audio contribution from external locations.
- Contribution from remote journalist and panelists without any technical expertice.
- Remote broadcasting of events.
- Sports Commentary
- Off tube booths console.

Phoenix Mercury

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

- 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
- It has a continuous data channel. It carries auxiliary data
- 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.

Scenarios of use:

- Links between events and TV stations: contribution from external locations.
- Reception point for phone calls in VoIP format, using SIP signaling.
- Coordination link to connect the Intercom Matrix with ENG teams or OB Vans.

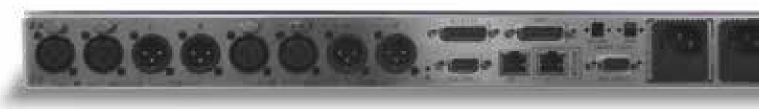
Phoenix Stratos

Double stereo full duplex stereo audiocodec with IP, ISDN and X21 / V35 connectivity

- Multi-format connectivity: in addition to IP networks, it allows the use of ISDN lines and X21 / V35 links.
- Simple and intuitive operation: It has a physical user interface on the front panel, as well as software control.
- 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.
- It has a continuous data channel. It carries auxiliary data.
- Remotely monitored: includes SNMP server that allows viewing its status and alarms.
- Two pairs of balanced analogue audio inputs and outputs at line level, and AES / EBU digital audio inputs and outputs.
- 4 GPIs and 4 GPOs as general purpose inputs and outputs for signalling and control.

Phoenix Venus 3

Dual stereo full duplex IP stationary audio codec, with analogue, digital and AoIP Dante™ local connectivity



- Performance and reliability "carrier grade" for the most demanding applications.
- It has two AC power supplies, optionally AC / DC or DC / DC 48V.
- Double IP network connection allows independent connection of the internal LAN network from the external WAN or Internet.
- It allows two independent stereo / dual or four mono full duplex connections to two different destinations.
- Double channel totally independent for program and coordination or backup with their respective returns.
- Controllable through user interface on PC .
- Double continuous channel of data. Ancillary data transport.
- Remotely monitored: includes SNMP server that allows viewing its status and alarms.
- Two pairs of balanced analogue audio inputs and outputs at line level, duplicated with AES / EBU digital audio inputs and outputs. In addition, the equipment can optionally include local audio connectivity over IP, with Dante™ technology,
- 4 GPIs and 4 GPOs as general purpose inputs and outputs for signalling and control.

Scenarios of use:

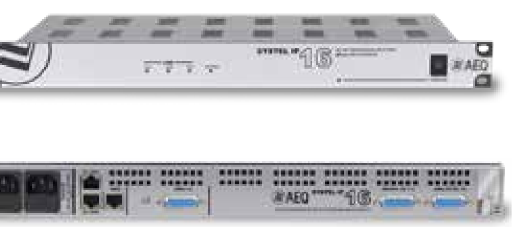
- Links between events and TV stations: contribution from external locations.
- Reception point for phone calls in VoIP format, using SIP signaling.
- Coordination link to connect the Intercom Matrix with ENG teams or OB Vans.

Systel IP

VoIP-based telephone coordination and multi-conference system

Voice over IP (VoIP) system for multi-conference and coordination. It can operate integrated into an intercom system for external communications. The system is basically componed of:

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, large enough for a multi-studio TV coordination system.

The device operates as a multi-line IP phone with signaling based on SIP protocol. Compatible with IP PBX, SIP trunking and virtual PBX.
Supports analogue and ISDN lines through adequate gateways.

SYSTEL IP LITE "Engine"

Same equipment as SYSTEL IP 16, limited to 8 simultaneous IP telephone lines. This can be upgraded to SYSTEL IP 16 at any time using an activation code and without modifying the installation received up to 16 simultaneous VoIP calls.

Applications:

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

Control Terminal SYSTEL SET +

It is used to provide real-time control of SYSTEL IP 16: SYSTELSET+ touch-screen phone control allows for very flexible operation: on the bare terminal, using the function keys and touch screen, calls can be dialed or answered, put on hold or pre-listen, their send and return levels are adjusted, calls are put ON AIR or sent to the Intercom system, fixed or hang up. It also allows for the management of a call book and call scheduling. Lines can be shared among different studios and it adapts its layout to the number of available lines at each one.

Control Application
SYSTEL IP TV

Specific software application for external routing of intercom systems and other general-purpose applications in TV production centers and others. Besides SYSTELSET+ functions, it also provides specific features for TV production:

- Automatically and manually answer incoming calls, label, put them on air or leave them into a multi-conference group.
- Leave the calls listening the assigned (N-1) feedback.
- Connect calls to the intercom matrix and route them to the internal assigned audio circuit.
- Leave calls in different multi-conference groups.
- The operator can talk to any line individually and also simultaneously to all the members in a group.

KROMA by AEQ broadcast video monitors have been designed to satisfy a wide range of requirements for monitoring and measuring video signals, especially in TV program production and distribution centers.

LM9000 Series

4K video Broadcast Monitors

LM9000-series monitors are designed to allow peration in UHD/4K environments. “SINGLE” and “QUAD-LINK” 4K signals can be displayed in any of the three available sizes: 55” (3.840 x 2.160), 31“ (4.096 x 2.160) and 24” (3.840 x 2.160) in both “SQUARE DIVISION” and “2-SI” formats. They include HDR technology to provide video playback with high levels of contrast, brightness and sharpness.

Different waveform and vector-scope displays are provided for the evaluation of UHD/4K video signals, in order to check whether they comply with international video standards.

Additionally, high-brightness option is offered for 24” and 31” sizes, with over 850-1000 cd/m2 luminosity.

- 2xBNC 12G-SDI with their respective loops for Single-Link UHD and 4K signals display.
- 4xBNC 3G-SDI for 4K and UHD signals display with Quad-Link.
- 4K image processing in “Square Division” and “2-Sample Interleave” formats.
- DCI 2.0 inputs (not compatible with HDCP) allowing for resolutions up to 2160p60.
- Color space selection (3-D LUT) EBU, NTSC, SMPTE-C, REC709 and D-CINEMA.
- Color temperature selection: 3200K, 5500K, 6500K and 9300 K.
- Firmware upgrade via USB port.
- Remote control through GPI port (RJ45 connector).
- Audio level meter for up to 16 channels.
- HDR (high dynamic range) display function with several ST2048 and HLG curves. Demo mode for HDR / SDR comparison.
- Peaking filter and False Color.
- LTC and DVITC timecode display.
- Vectorscope and waveform display.
- Closed Caption according to 608, 808 ANC, Transcoded 608 and CC708.
- Blue Only / Mono .
- Several marker formats with adjustable transparency and color: 4:3, 16:9, 14:9,13:9, 2.35:1, and 1.85:1.
- Safe Area: 80%, 85%, 88%, 90%, 93%, 95%, EBU Graphic, Action.
- Center Marker with three selectable sizes.
- H / V Flip function.
- Aspect ratios: 4:3,16:9, 15:9, 14:9, 13:9, 1.85:1, 2.35:1, 1:1, native.

Models	Resolution	Brightness	Dimensions	Weight
LM 9024	3840 X 2160	350 cd/m2	452 x 376 x 56 mm	5,2 Kg
LM 9024 HB	3840 X 2160	1000 cd/m2	452 x 376 x 56 mm	5,2 Kg
LM 9031	4096 X 2160	400 cd/m2	736 X 552 X 56 mm	8,7 Kg
LM 9031 HB	4096 X 2160	850 cd/m2	736 X 552 X 56 mm	8,7 Kg
LM 9055	3840 X 2160	500 cd/m2	1242 x 734 x 79mm	25 Kg

LM8000 Series

FHD resolution Broadcast Video Monitors

LM8000-series monitors, designed around 10 bits processor, allow for operation in FHD environments.

HDR technology is available in any of the three available sizes: 24”, 18” y 9” in order to achieve video reproduction with high levels of contrast, brightness and sharpness.

Different waveform and vector-scope displays are provided for the evaluation of UHD/4K video signals, in order to check whether they comply with international video standards.

They also feature Dual Input, Dual Output: Double video processor embedded into a single chip, able to show two identical images in parallel on the display (PbP) with the same type of de-interlacing, motion adaption, and scaling.

They include, among other input interfaces, and SFP optional module in order to incorporate SMPTE 2022 and SMPTE 2110 Video over IP.

- Remote control of the monitors via Ethernet using the new, second generation, remote control software for PC.
- Color configuration by password-protected menu, with several user memories and color spaces.
- Color temperature selection: 3200K, 5500K, 6500K and 9300K.
- Audio de-embedding from SDI (16 channels) and digital component input (stereo).
- Vu-meter display for up to 16 channels with several different scales (dBFS, BBC, DIN, Nordic, STD, NA, FRA, EBU).
- Phase-meter showing the phase relation between each stereo audio pair.
- PIP, PBP, PBP A and PBP H functions.
- Waveform (Y Cb Cr) and vectorscope display.
- Luma check, false color and focus-assist.
- Menu and TSL-protocol configurable IMD.
- TimeCode.
- Several aspect ratios: 4:3,16:9, Auto, Native, 1:1.
- Various formats of markers with several levels of transparency and colors: 4:3, 21:9, 16:9, 15:9, 14:9, 13:9, 2.39:1, 2.35:1,1.896:1, 1.85:1 and 1.66:1.
- Safe Area: 80%, 85%, 88%, 90%, 93%, Graphic, Action.
- Center Marker with three selectable sizes.
- Sharpness, delay, scan, inverted image.
- Freeze mode.
- Layout mode that allows the user to analyze, clearly and within a single window, the different parameters of the video signal as well as the possible auxiliary data.
- Close Caption CC608(VBI), CC608(ANC) and CC708.
- DualSplit mode.
- Auto-calibration of the monitor colors by connecting a color probe and Lightillusion Kroma-specific control software. This calibration generates 3D LUT (look up tables) exclusive for each monitor in order to correct all non-linearity inherent to the display manufacturing process.
- Selection of multiple Gamut: BT.709, SMPTE-C, EBU, NTSC, D-Cinema y sRGB. Internal power supply.

Models	Resolution	Brightness	Dimensions	Weight
LM 8024	1920X1080	350 cd/m2	552x379x95mm	7,5 Kg
LM 8018	1920X1080	350 cd/m2	446x265x80mm	5,5 Kg
LM 8009	1920X1080	350 cd/m2	222x177.5x80mm	2,0 Kg

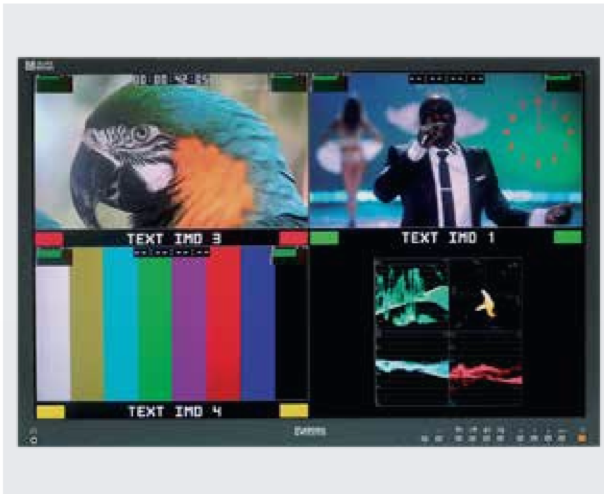
Other Kroma by AEQ video monitors

QS 7000 SERIES

Quadsplit Video Broadcast Monitors

The QS Series from KROMA, with built-in quadsplit and 10 inputs, is now enhanced with waveform and vectorscope tools, high resolution IMD (In-Monitor Display) and VU-meters, onscreen clock and the option to turn 4 video inputs into outputs by menu. It features 2 DVI-I inputs (YPbPr, VGA and DVI video mode) and 8 multi-format video inputs (composite and 3G/HD/SD-SDI). 18,5" y 24".

18,5" Monitor QS7018



24" Monitor QS7024

Other Kroma by AEQ video monitors

LM7500 Series

Previous Video Monitors

The LM 7500 preview monitors series is based on 16:9 native LCD high resolution panels, featuring LED backlight to reduce power consumption and providing better colour reproduction. LM7500 units offers identification and calibration of the signal, precision level meters and headphone output, In-Monitor display (IMD), on-screen tally, waveform display and vectorscope. There are several available modes: 2x9", 2x7", 3x5" y 4x4", with different input configurations:

Model#	LM7509	LM7507	LM7505	LM7504
Panel LCD (piece)	2	2	3	4
Screen	9" (16:9 native)	7" (16:9 native)	5" (16:9 native)	4" (16:9 native)
Resolution	1280x768	800x480	800x480	800x480
Active Area	195x113.4 mm	152.4x91.44 mm	108x64.8 mm	95.04x53.85 mm
Viewing angle	178º H/V	160º H/V	170º H/V	170º H/V
MTTF	50,000 Hours	50,000 Hours	50,000 Hours	50,000 Hours
Brightness	350 cd/m2	400/cd/m2	300/cd/m2	300/cd/m2
Contrast	900:1	500:1	600:1	600:1
Backlight	LED	LED	LED	LED

VF 7000

7 "Full HD View Finder Monitor

7" Full HD monitor adapted for on-camera mounting in professional image acquisition.
Includes rear and front Tally, stand support, Anton Bauer format battery adapter and 12V / 24V power supply.

AudioPLUS

Computer-based audio production, edition and playout software for TV

Automation software with Playout, automatic editing and programming of TV audio broadcast. Providing analog connectivity, digital AES, digital USB and AoIP protocol Dante™ - AES 67.

It incorporates tools for manual, automatic and remote control broadcasting, automatic music programming and advertising, as well as content generation and editing.

Features:

- Intuitive user-oriented software with very friend GUI.
- Manual playout system by list and by instant key in jukebox mode.
- Instantaneous change from “hot keys” to “instant replay” formats.
- Automatic unattended playout system.
- Compatibility with physical sound cards, USBmodules and “Dante Virtual Soundcard” application.
- Stereo Audio editor.
- It includes by default external editor Audicity and link for other high level audio editors, and optionally the multitrack AEQ Power Editor.
- Up to 4 stereo audio program + CUE in one PC.
- Import and export of audios in many different file formats.
- Software license control.

Scenarios of use:

- Manual audio playout in “hot keys”, “instant replay” and list formats.
- Automatic playback of list-organized audio files.
- Audio recording and editing.
- Structured organization of the audio resources in access-controlled folders.

AudioPLUS cartridge format

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 AES 67-DANTE™ 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.

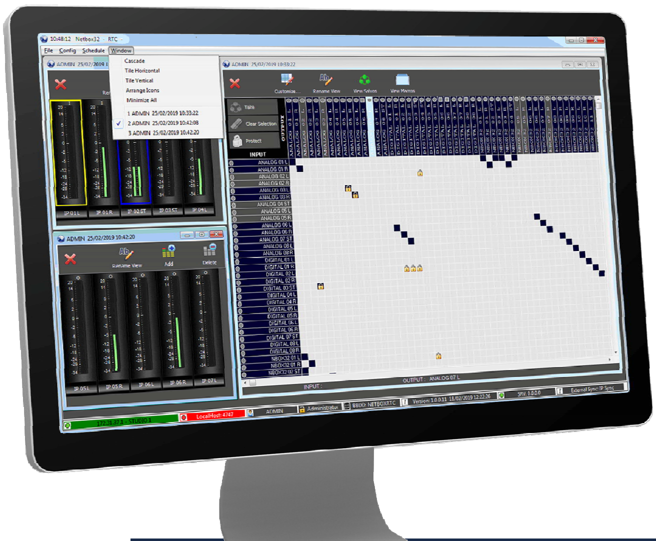
Applications:

Sound acquisition in stages, journalist voice booths or mixed-zone connections during sports events. Microphone preamplifier.

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. Useful to provide IP access to mixing consoles without IP connectivity, workstations and off tube consoles.

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 form 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, analog line, microphone and headphones, long-distance dark fiber optic links, 64-channel MADI format and, proprietary fiber links with more 1000 channels, among others.

In addition, through AoIP cards with 64 inputs and outputs, we can include audio inputs and outputs from equipments with Dante™-AES 67 protocols, in the IP matrix. An X_CORE frame can incorporate as many AoIP cards as necessary, and these can be installed in one or several Gigabit Ethernet networks.

128-channel AoIP cards are also available compatible with Ravenna/AES67 protocols. X_CORE also accepts I/O audio flows embedded in video signals with SMPTE 2110-30 and SMPTE 2110-31 formats and audio embedded in SDI video signals.

NETBOX DSP

Mixing, processing and distributing audio matrix. Versions with 64, 96, 128 and 160 audio inputs and outputs to the Dante network. Mix combinations of Dante network inputs over any of its up to 160 outputs to the Dante 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 Dante connectivity. A large matrix in only 1UR in size.

NETBOX 32 AD MX

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

The Atrium Digital Audio Mixer is designed for television production environments.
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 the audio channels embedded in SDI video, it makes installation and use on television simple and flexible.

Atrium

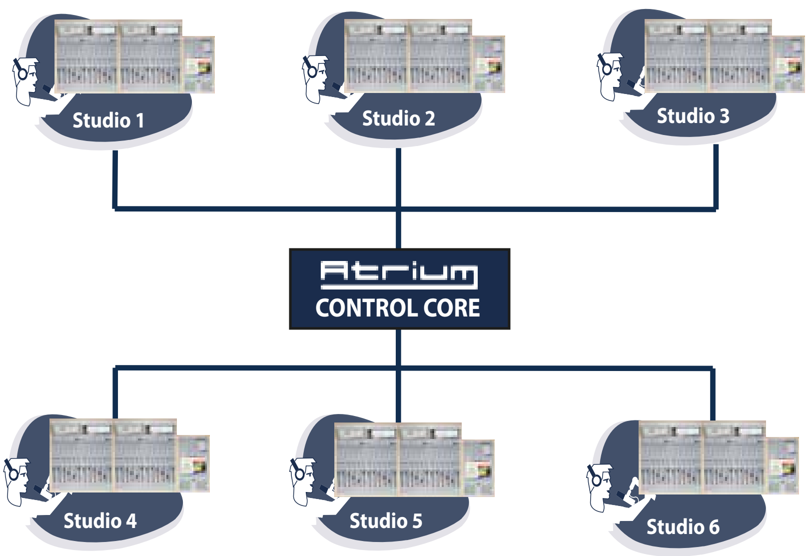
Modular audio digital console up to more than 1000 channels and 96 faders that can be set up on individual channel setting pages

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 work with several engines, and 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.



Scenarios of use:

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

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.

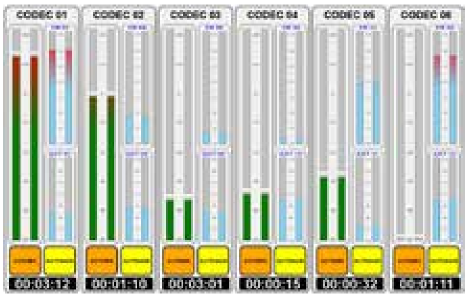
The control and monitoring module has a touch screen and 24 programmable and contextual keys. It also incorporates VU-meters, CUE speaker and control and studio sections. You can add an additional touch screen, with loudness measurement.

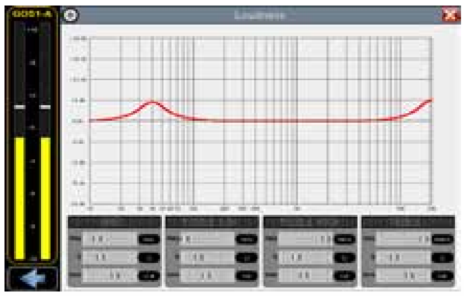
Atrium

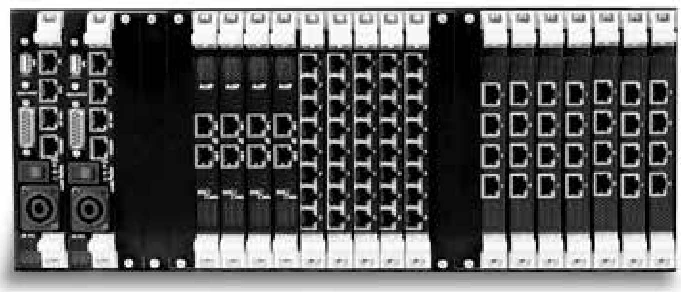
The ATRIUM console incorporates the specific features for ON AIR broadcasting: automatic monitor mute, cough mute, fader start, control signalling etc. In addition, it incorporates programmable keys to control external equipment: communications, intercom, visual radio...

ATRIUM's multi-channel audio connectivity supports most of the formats used today in TV audio production; SMPTE 2110-30 and SMPTE 3110-31 with NMOS control, SMPTE 2110-30 through Dante Domain Manager, SDI embedded audio up to 3G, AES67 with RAVENNA control, AoIP RAVENNA native, AES67 with Dante control, Native Dante AoIP, and AES 10 MADI.

X_CORE is the process and inputs and outputs engine of ATRIUM. It is completely modular core/engine and physically separate from the control surface. It is in charge of all the functions for mixing, routing, setting and processing of dynamics, equalization, filters, delay and reverb, among others. Several X_CORE can work together in larger installations.







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 I/O boards for different formats:

- Frame to Frame Link module, 1,024 audio channels.
- Dual 3G SDI card, with two inputs and two SDI outputs for video with embedded audio, and connection to the internal audio bus of the consoles with up to 2x16 channels of audio input and 2x16 output.
- AoIP 64-channel audio connection card based on Dante™-AES67-SMPTE ST 2110-30 standard.
- AoIP 128-channel audio connection card based on RAVENNA-AES67-SMPTE ST 2110-30 and SMPTE ST 2110-31 standards.
- 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.

CRAZY ABOUT TV

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