PRO-AV SOLUTIONS

A GLOBAL AUDIO & COMMUNICATIONS SOLUTIONS

Full Integration

Until now, existing technology meant that different systems had to be used to manage audio and communications. Completely autonomous systems that did not allow interoperability and optimisation of resources.

Our goal, in recent years, is to offer a complete integration based on evolving technologies that can be applied to different working environments: Theatre, live concerts, theme parks, casinos, museums, places of worship, schools and universities, control rooms, public address, commercial audio and security, train stations and airports, cruise ships, hotels, sports production, sports centres and stadiums, races, mobile TV units, local and remote TV and radio production. The tasks in all these areas require very specific audio and communication systems that can be based on the development of the same technological advances.

For these professional sectors, AEQ offers the following categories of systems:

Intercom Systems
A large range of Intercom system that includes systems with IP technology, wired, simple without matrix, or more complex including an audio matrix. Also wireless or mixed intercoms, with wired and wireless terminals. All in a local network, or distributed in multiple locations. And with the ability to organise the distribution and listening of commands so that no one misses anything, or is distracted by messages that don't concern them.

Audio, analogue, digital and IP interfaces and matrices
Analogue audio converters from mic, line and headphone, or digital USB or AES/EBU to audio over IP. Even commentary units adapted to the highest demands of broadcasting the biggest sporting and social events. Audio matrixes with inputs and outputs in multiple analogue, digital and IP formats, embedded in video, and with the most different dimensions, from 64 inputs and outputs to more than 5,000.

Multiconference system, talk-show and remote coordination systems
Based on HD VoIP systems that allow to organise the conversation of talents located in different places, individually or in rooms of any size, autonomously or within an intercom system, and for internal use, or for broadcasting by a PA or broadcast system.

Audiocodecs for high quality audio transport over IP networks
In various formats for different uses, but always with excellent audio quality and minimum delay: For personal, portable and stationary use.

AEQ’s experience in offering audio solutions for large international events, acquired throughout our long history and in combination with our close contact to customers having a great variety of operational needs, allows us to have a clear idea of what is required in terms of quality, reliability and operational workflows for any production.

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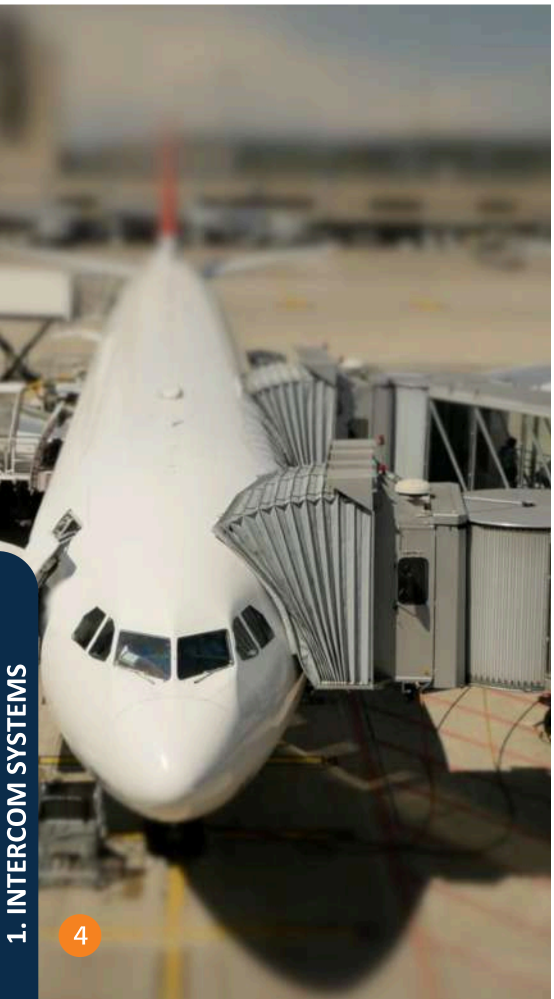
SECTION 1
INTERCOM SYSTEMS

When coordination is essential in a job, but each member of the team has to be absolutely focused on the work, an intercom system is needed.

Matrix-less Intercom Systems

Xpeak system is designed to support up to 28 user terminals in different formats: desktop, rack, wired beltpack, wireless beltpak, and PC applications, which can be interconnect with the greatest operational flexibility.

Those systems incorporate solutions for our users' new requirements: simple configuration of de-centralized systems to enable and simplify remote production; Bluetooth and USB connectivity, simplifying operation with a wide variety of heatset combinations, enabling audio-tethering with PCs, among others.



Matrix-based Intercom Systems

This will allow the user to share resources, simplify the operation and to control production based upon very stable, redundant systems, with the best available audio quality and possibility to extensively process the audio signals. State-of-the-art technology that provides connectivity to AoIP networks with centralized control. In a nutshell, a leap forward towards 360º management of all your contents in a simple way, making the achievement of the best possible final results and easy task.

In order to achieve this goal, we cannot forget about audio quality at any moment. That's why our matrix systems process the audio signal with 48 KHz sampling rate and 24 bits resolution, providing a broadcast-quality flow between all devices. One of the most important reasons that allows us to keep this quality level is the use of AoIP Dante™/ AES67 standard for the audio transport between equipment.

We are always open to inter-operate with third-party equipment using other formats, such as RAVENNA, MADI, SDI-embedded audio SMPTE ST 2110-30, SMPTE ST 2110-31 and any other that may become standard or popular.

Wireless Intercom Systems

AEQs wireless intercom system is based on Xplorer terminals with WiFi connectivity, and the Xvirtual application (for iOS and Windows), with the full capacity of a intercom user panel.

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 work stand-alone. It can also be added to a matrix-less or matrix-based system: as a client of AEQ Intercom matrices such as Crossnet and Conexia, and also as a wireless user panel of the Xpeak system, working as an intelligent beltpack without the need for a matrix.

XPEAK

Matrix-less, de-centralized intercom system

PRODUCT CONCEPT

In conventional AV/Media production, the intercom is an essential tool for the technical coordination of everyone involved. Remote production is nowadays a standard part of operations.

The need to produce Television and Radio minimizing the mobilizing and remote deployment of technical personnel, has become a fact. It is now required that Production personnel is able to work from different locations, some of them temporary (hotels, stadiums, etc).

Until now, these situations have been resolved by connecting intercom systems to audio codecs and IP telephony systems and configuring the systems separately, which involves the deployment of expensive systems and the development of complex connection and set-up tasks.

For this reason, we have created Xpeak, an intercom system that incorporates virtual network technology, through an automatic global interconnection Wizard. The Wizard avoids the dificulties of creating a virtual network and configuring the devices to work on it.

In Xpeak, different hardware or software user panels are interconnected as if they were all part of the same physical production center. It doesn't matter whether they are on the same LAN network, or if they are dispersed in different places with access to Internet – they will all connect.

You do not have to have a static IP, nor do you need a dynamic DNS, nor do you need to set port forwarding, because each user panel is automatically configured according to the local network of its location. It works immediately in all network environments, even on very restrictive networks.

On the other hand, the system does not need a matrix: the elements of the system form a P2P (peer-to-peer) network with Ethernet virtualisation.

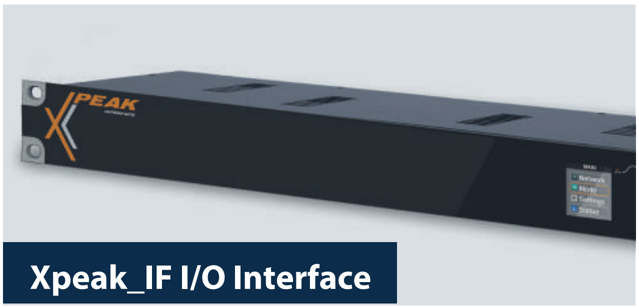
Moreover, as an intercom system is a mission-critical tool, it should neither fail nor be vulnerable to critical tool that must neither fail nor be vulnerable to attack. It includes advanced security features such as access control rules, security control, and traffic encryption.

SYSTEM DEVICES

Xpeak_R User Panel

Xpeak_D User Panel

Xpeak_BP Wired Beltpack



Xpeak_IF I/O Interface

Xpeak Virtual App for PC

Xplorer Wireless Beltpack

Xpeak, Matrixless intercom System

SYSTEM DEVICES

Xpeak R and Xpeak D

Rack-mounted and desktop user panels with 8 programmable keys arranged in 4 different pages.

FRONT PANEL

At the front, we can find a "gooseneck" electret microphone and internal loudspeaker, a front USB connector for Mic. Headphone combination, 8 cross point keys -operating with the matrix, up to 4 pages per key can be programmed- and 2-axis, lever-type keys, allowing the user to control talk and listen and individual volumes for each communication cross point. Information is presented on two LCD graphic screens and RGB LEDs associated to the keys. Rotary encoder and configuration keys.

REAR PORTS AND CONNECTIONS

At the back, two VoIP ports for loop connection, USB (type B) connector for connection to headphones and PCs. GPIO: connector with 2 optically coupled GPI and GPO and a power pin to supply external circuits. Internal power supply.

Bluetooth: the device incorporates Bluetooth connection as an audio interface with a telephone or headset.
It can combine different audio signals arriving to the system from different devices. Eco-cancellation processing.

Xpeak R, 1 RU User Panel with 103 mm depth.
Xpeak D, desktop User Panel 217 x 105 mm. 101 mm depth.

Xpeak BP

Wired beltpack User Panels with 4 programmable keys in 2 different pages.

FRONT PANEL

On the front, 4 cross point keys can be found. When operating without a matrix, up to 2 pages per key can be programmed. Associated with each key there is an RGB LED to indicate the family to which the destination of each key belongs.
The rest of contextual information associated to keys, the communication and the menu are presented on a graphic LCD screen which can be turned on and off.
Also at the front, two lock, mute, page swap and menu navigation keys can be found.

BELTPACK's REAR PORTS AND CONNECTIONS

Ethernet VoIP port and PoE supply on a RJ45 latching connector. Back USB port for micro-headphones. Two-pin GPO output.

OTHER FEATURES

Two rotary encoders for volume adjustment. Bluetooth interface for audio exchange with telephones or headsets. It can combine different audio signals arriving to the system from different devices. Eco-cancellation processing.

Belt-pack is made of shock-proof plastic. Dimensions: front: 82mm wide, 70mm depth, 130mm height. Functions: incoming call front signalling with the possibility to activate GPO to external devices.

SYSTEM DEVICES

Xplorer

User Panel in wireless beltpack format with 4 programmable keys in 2 pages.

FRONT PANEL
On the front, 4 cross-point keys. When operating without a matrix, up to 2 pages per key can be programmed.

The rest of contextual information associated to keys, the communication and the menu are presented on a graphic LCD screen which can be turned on and off. Also at the front, mute, page swap and menu navigation keys can be found.

BELTPACK REAR PORTS AND CONNECTIONS
Mini XLR back connectors for headsets. Charge connector.

OTHER FEATURES
Two rotary encoders for volume adjustment. Eco-cancellation processing. Muting function. Ethernet network connectivity using WiFi. Compatible with 802.11b/g/n networks in the 2.4GHz band and 802.11a/n networks in the 5 GHz band. Powered by rechargeable bateries providing up to 20 operating hours. Charged at the charge station.

Beltpack is made of shock-proof plastic. Dimensions: front: 92mm wide, 70mm depth, 130mm height. Approx weight: 365g.

Xpeak_IF

Audio format converter/interface for 4 audio inputs and outputs to 4 Xpeak bi-directional channels. External channels can come in analogue, USB or Dante / AES67 AoIP format.

USB connection: The 4 USB audio ports are type B and allow you to connect audio from a PC.

Ethernet connection: internal switch with two external network ports. Different uses: daisy chain, separation between Dante audio and encoded audio + Control.

GPIO: Mini-Hartmann connectors include 4 opto-coupled GPIs and 4 GPOs with external power pin.
It features a TFT screen, rotary encoder and menu key for its configuration and control from the front panel, allowing for navigation through the user interface for easy configuration and status display.

Connecting the AEQ TH03 telephone hybrid to Xpeak_IF, remote outdoor connectivity is facilitated by calling guests or reporters via conventional telephony.

TH-03

DIGITAL HYBRID
For analog telephone lines (POTS). Automatic line impedance adjustments. A connector for an auxiliary analogue telephone to dial into each line. Digital audio I/O option.

SYSTEM DEVICES: VIRTUAL USER PANEL APPLICATION

Xpeak Virtual

User terminal implemented as a PC Software application with 8 programmable keys.

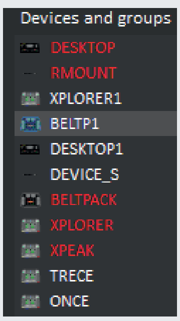
CONTROL AND CONFIGURATION APPLICATION

X-peak

Xpeak user-panels features auto-discovery and auto-configuration functions. When a system is delivered it is pregonfigured at factory. All of the system terminals will be ready to operate with basic functionality. Of course, adaption to each particular operational requirement will be needed and is easy to accomplish. A configuration software application has been developed for this purpose and with the following features:

- Simple and visual software application.
- Configuration through drag'n-drop actions.
- Device auto-discovery.
- Devices are organized in groups with common functionality, and in families sharing the same role, assigned by an administrator.

With the following tools, it is extremely simple to adjust X-peak settings to the need of a job.



MANAGING DEVICES

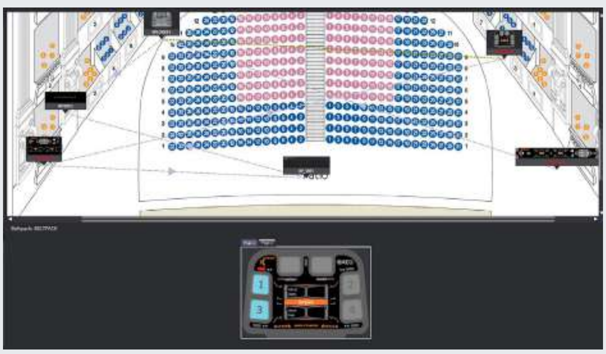
When the software is open, a list of the devices is presented. Each one has an associated icon and name. By clicking on a device –either on its name or on its representation in the diagram-, the bottom right area will show its detailed representation, providing access to each key so that it can be configured or modified.

DIAGRAMS

Physical location of each user terminal can be viewed on a blank window or over a drawing representing the work space, as well as the functional communications relationship between them: whom is able to talk to and listen to whom and doing what at any given moment.

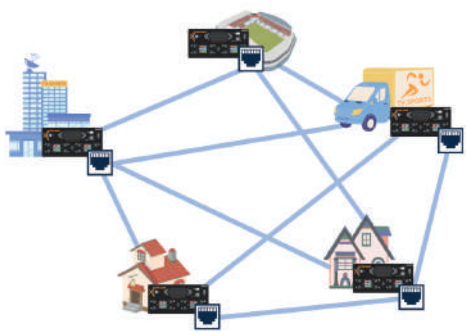
REMOTELY ASSUMING THE ROLE OF A DEVICE

A very special utility has been developed in order to test that a device operates as required: from the software, we can place the cursor on a device and work in parallel with it: use the keys depicted in the screen to talk to the destinations configured for it, and listen the messages reaching it from other panels through the PC we are running the software in.



ADVANTAGES OF XPEAK

1 **Three modes of operation:**
Virtual Matrix: Xpeak obtains the flexibility of AEQ intercom systems without incorporating a matrix. Each panel maintains the cross-point programming according to its defined role.



- **4 channels Party-Line:** Very simple configuration since it is only necessary to determine which channels each panel will talk to and which channels each panel will receive.
- **Physical Matrix:** The Xpeak panels works as a very cost-effective user-panels, with 8 physical keys (up to 32 virtual keys) in AEQ Conexia and Crossnet Matrix systems.

2 **Wizard for automatic global interconnection**
Simplify the use of virtual network technology to streamline remote productions.

3 **The panels do not belong permanently to a specific system:**

- The same user panel can work on multiple locations.
- The same panel can be part of different intercom systems.
- Possible expansion of the system through the cloud in a subscription-by-use scheme that does not require acquiring resources that are not used regularly.

4 **User terminals that support Bluetooth and USB micro-headsets.**

Compatible with a wide range of headsets with excellent value for money.

5 **User panels that can be connected to PCs and Smartphones.**

Via Bluetooth and USB connectivity.

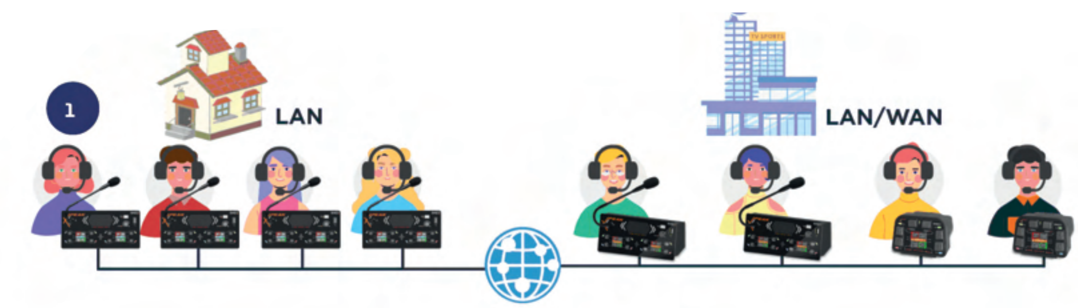
6 **Great audio quality with low bit rate**

It allows incorporating remote terminals through different connection methods.

7 **Stationary panels can be connected in Daisy Chain.**

BASIC FUNCTIONALITY IN DETAIL

- System dimensions: Up to 28 user terminals or external inputs and outputs.
- Two working modes:
  - Intercom mode in which each key of a user terminal can be configured with the same functions as in a system with an AEQ matrix.
  - 4 channel Party-line mode, which hardly requires any configuration.
- Intercommunication between the members of a domain regardless of their location. The panels can work in a local LAN network or in a WAN network that can be global in nature through the internet. In this case, intelligent connectivity is available through virtual network technology, through an automatic global interconnection wizard, which simplifies configuration, discovering the elements of the system within the domain. Connection without opening ports in routers, to use terminals in locations such as hotels, homes, or others without access to a support technician.



- Virtual network architecture allows a device to subscribe to different virtual networks on demand as needed.
- The system and its elements are compatible with previous intercom systems bearing the AEQ and KROMA brands.
- System is configured from the user panels themselves. There is also an optimized configuration tool, online and offline, for complex environments.
- With common user terminal architectures: stationary, desktop, wired belt-pack, wireless belt-pack and PC application.
- The wired user terminals support USB and Bluetooth headsets from the most basic, to the highest quality and best features.
- Rack and desktop equipment will use 4-way levers instead of keys, for easier operation of basic functions: Listen, Talk, and adjust the receive volume.
- Additional equipment is included to provide analog, digital and IP audio inputs and outputs to the system.
- The equipment has been designed to be very cost effective with optimum performance.
- Great audio quality (HD Voice up to 7 KHz) with bit rate limited to 64 kbps.

Applications

Theatre, live concerts, theme parks, casinos, museums, places of worship, schools and universities, control rooms, public address, commercial audio and security, train stations and airports, cruise ships, hotels, sports production, sports arenas and stadiums, racing, OB vans, local and remote TV and radio productions.

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, hi-fi-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.

Applications:

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

- 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, Xpeak desktop, rack and beltpack user panels, EasyNET party-line systems and, specially, the connection of Xplorer system for wireless beltpacks and virtual panels.
- Up to 128 Dante™ hi-fi-quality audio over IP ports, that may be used to connect TP8000 and TP9000 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™ hi-fi-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.

Conexia

Up to 1000 Channels Redundant Intercom 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. More information about X_CORE matrix in page 4.

If the required intercom capacity is lower than 1024 circuits, the rest of the matrix can be used as a hi-fi-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 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 hi-fi 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:

- Theme parks and shopping malls.
- Theaters, shows, concerts, halls....
- Sport events coordination (even multi-venue ones).
- Public and private Control Rooms.
- Emergency coordination.
- TV Production Centers.

TP9000, TP8000 and XPEAK

Wired user panels for Conexia and Crossnet

Thoroughly-designed acoustics for the best sound naturalness and intelligibility. There are three ranges, TP9000, TP8000 and Xpeak.

TP 9000. AoIP Dante™ connectivity user panels

TP9116

19” rack 1U user panel. TP9116 incorporate IP connectivity that handles high quality audio in DANTE™ format, compatible with the AES67 standard. It provides talk and listen functions and individual volume control for each communication point, through a lever-type 4-way key. 16 crosspoint keys, four pages. Information is presented on four RGB graphic displays. Offering Broadcast Audio Quality.

TP 8000. AoIP Dante™ connectivity user panels

16 keys, rack or table-top formats. Expansion panels can be chained to build up panels featuring up to 64 keys with 4 pages. Compatible with any KROMA and AEQ intercom matrix.

Digital audio processing: acoustic echo cancellation, automatic voice level, tone and speech habits control for each operator. Expansion and ambient noise gate

TP8116

19” rack 1U user panel. It provides an individual volume control per each communication crosspoint. 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

Extension panel. It 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. It provides an individual volume control per each communication crosspoint. 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.

Xpeak. VoIP HD and VoIP Kroma connectivity user panels

Xpeak R and D

Xpeak R and D user panels have talk and listen functions and individual volume control for each communication point, through a lever-type 4-way key. 8 crosspoint keys, two pages. Two VoIP ports for loopback. Information is presented on two RGB graphic displays. Bluetooth and USB connectivity for headsets, smartphones and PCs.
Xpeak R, 1UR user panel.
Xpeak D, desktop user panel.

Xpeak BP

The beltpak, Xpeak BP, has 4 crosspoint keys, two pages. One VoIP port with PoE power, Information is presented on an RGB graphic display. Bluetooth and USB connectivity for headsets and smartphones. Two volume controls.

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 work like:

Client of AEQ Intercom matrices such as Crossnet and Conexia, in combination with wired panels and with equipment running the Xvirtual application. With Conexia matrices, you have enhanced audio quality with G722 compression algorithm.

Wireless user panel of the Xpeak system, working as an intelligent beltpack without the need for a matrix.

It can also work in 4-channel Party-Line mode, with other Xplorer terminals or integrated into an Easynet Party-Line system.

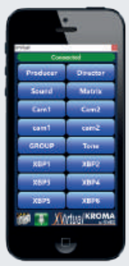
It is compatible with 802.11 b/g/n networks using the 2.4 GHz band and 802.11a/n using the 5 GHz band.

It has double volume control by digital encoder. Also Mute function, 4 physical crosspoint keys, with individual operation in Party-Line systems, with two pages, total 8 crosspoints in Xpeak systems, and with 4 pages in systems with Conexia or Crossnet matrix, total 16 crosspoints.

Its standard battery autonomy is around 20 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.

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.

XVirtual App



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.

SECTION 2.
AUDIO OVER IP: INTERFACES, COMMENTARY CONSOLES AND MATRICES

Netbox 32 AD/ 4MH

Analog and digital audio interfaces to the IP network

A complete range of digital interfaces and routers puts AEQ at the forefront of IP audio system design and manufacturing. IP connectivity according to AES 67 - DANTE™ standard makes installation and use simple and flexible.

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.

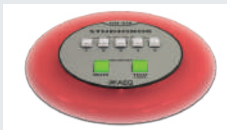
NETBOX 4 MH

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

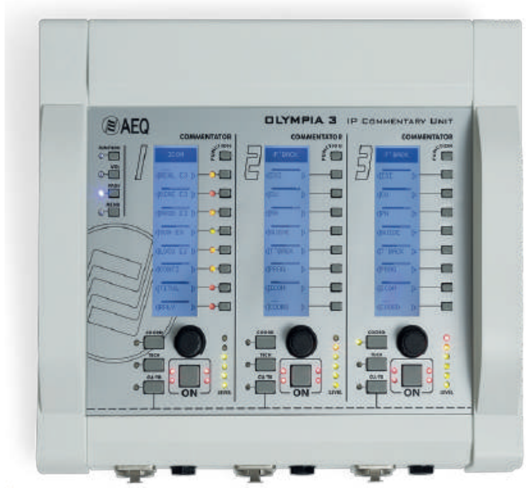
STUDIOBOX

Bidirectional desktop signaling terminal. Interact with a digital console directly or through NETBOX 4MH to any equipment that generates virtual GPOs such as PCs. With three colours of signal lights buttons, remote PFL and 5 configurable buttons.



Applications
It is useful for talents or coordinators at press conferences, debates, news, or talk shows.

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:

As an Intercom Usere 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, Conexia and CrossNET.

As an a 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.

Applications:

- Three microphones mix with headphone and Dante™ output.
- Conference rooms.
- All sizes commentary systems.
- Off Tube commentary rooms.
- TV OBvans.
- Simultaneous translation.

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.

Netbox DSP and 32 AD MX

AoIP connectivity audio matrices

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.

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 DSP PROCESSING

• 64 Frequency Processing: High pass, low pass and band pass filters, 4-band parametric equaliser. Parametric 4-band Equalizer.
• 64 Dynamic Processors: DLP in 4 sections (Compressor, Expander, Limiter) and Noise Gate.
• 64 Delays up to 10 seconds.
• Additional level adjustment at each crossover point.

GENERAL FEATURES

Grouping of logic lines in stereo pairs.
Adjustment of input and output levels.
Mixing of signals on any output without limitations, at will.
Multi-equipment and multi-user control software with:
- Configuration of specific views and scenarios.
- Handling of macros, saves and configuration views.
- Scheduling of tasks triggered by clock, alarm or external trigger.
- Creation of Talkback or Multiplex groups based on N-1.
- Rights management for each functional group of users.
- Protection of critical lines.
- Flexible generation of vumeters, and test tone.

Physical and virtual GPIO management between different devices.
Automatic AGC gain control on AoIP inputs and outputs.
AoIP Dante / AES67 connectivity.
Hardware alarm: Failure in a power supply, local network interface and internal configuration.
interface and internal configuration.
AES67 / Dante AoIP format.

Applications:

Mixing, routing, switching and processing of audio in:
- Halls and multimedia rooms.
- AV systems for:
  - Airports and rail stations.
  - Amusement parks.
  - Cruises.
  - Sports centres.
  - Smart building.
- Audio for radio and television.

X_CORE

Up to 5.120 x 5.120 circuits X_CORE Audio and Intercom matrix with multiformat connectivity

Applications:

Mixing, routing, switching and processing of audio in:

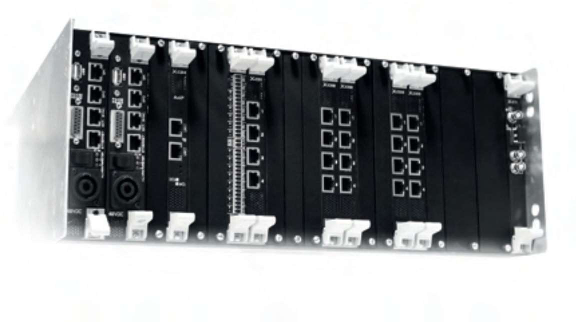
- Halls and multimedia rooms.
- AV systems for:
  - Airports and rail stations.
  - Amusement parks.
  - Cruises.
  - Sports centres.
  - Smart building.
- Audio for radio and television.
- Large commentatary systems.
- Mixing console and intercom systems router.

Audio mixing, processing and distribution matrix. When properly configured, it can perform as a general purpose audio matrix, as an intercom audio matrix or with mixed capabilities.

For intercom systems, 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.

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.



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.

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.

SECTION 3: EXTERNAL COMMUNICATION SYSTEMS

SYSTEL IP

Designed fo multiconference, talk show and remotes coordination

Applications:

- Multiconference with broadcast output.
- Remotes coordination with and Intercom System.
- IP hybrid mode with multi-line and multi-studio possibilities.
- Conference calls in the air.
- Talkshow.

Allows IP calls, coming from IP telephony providers, IP PBXs, or even audiocodecs or conventional telephony, to be introduced into different groups. Great economic savings in communication costs and a huge improvement in the audio quality of communications.

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.

Elements of the system:

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 DanteTM/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 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.

TALENT: IP Personal AudioCodec

For personal use with a mic input and an analog stereo line and Bluetooth inputs. A headphone, analog stereo line and Bluetooth outputs. HELP function. IP connectivity. It allows you to use a smartphone to control the unit, add calls to the main output or to connect with the event or studio. SIP protocol and EBU N/ACIP standard and includes OPUS algorithms. Also included, tools for configuration and remote control assistance.

Aplicaciones

- **Basic:** Send a local microphone to the event, room or studio, with return, or "lazy", to the headphones.
- **Live phone calls:** Make phone interviews using a Bluetooth-connected smartphone. Audio returned by the phone is mixed with the local microphone and sent to the event, room or studio mixer.
- **Comentator:** Send the local mic mixed with a stereo line input from the auxiliary connector or from a computer or smartphone linked through Bluetooth A2DP.

PHOENIX ALIO: IP Portable AudioCodec

4 mic inputs or 3 mics and a stereo line inputs, bass and treble controls on all inputs, tone generators, one or two bidirectional mono or stereo channels. Two pairs of headphones and a stereo line output. HELP function. IP connectivity. SIP protocol and EBU N/ACIP standard and includes OPUS algorithms. Also included, tools for configuration and remote control assistance.

Talent and Alio applications

- TV and Radio outside contributions.
- Non technical host contributions and talkshows.
- Events broadcast.
- Sports commentators.
- Off tube mixer.

SMARTALK: cloud-based audiocodec system

SMARTALK is a service that generates on the PC or smartphone, a link to a website, to instantly download an OPUS audiocodec from the cloud, which through a SIP server, automatically connects to the studio's AEQ audiocodec. No need to install any application or enter a user name and password, just a direct web link.

PHOENIX VENUS 4 & VENUS 4+: Twin IP Codecs

Two simultaneous, full duplex stereo transmissions with different audio formats and qualities. SIP protocol and EBU N/ACIP standard. Includes OPUS algorithms. Applications for comprehensive config. and remote control. Balanced analog I/O's on XLR connectors and dedicated connectors for AES/EBU digital I/O's. Double network port, double RS232 auxiliary data link and optional redundant power supply. Version with 48 volt DC sources available. Optional local DANTE™/AES67 AoIP connectivity with an additional network port.

VENUS4+ adds front panel controls with status indication and on-screen VU-meters. It also provides a menu to initiate and accept calls, execute presets and to edit/modify settings.

PHOENIX MERCURY: IP compact AudioCodec

IP connectivity. Two-way stereo transmission. SIP protocol and in compliance with the N/ACIP recommendation of the EBU and includes OPUS algorithms. Also included, tools for configuration and remote control assistance. Balanced analog I/O's on XLR connectors and optional dedicated connectors for AES/EBU digital I/O. RS232 auxiliary data link.

Venus 4 and Mercury applications

- High quality audio links between event venues via IP.
- Telephone calls reception in VoIP format with SIP signalling.
- Coordination link to connect the intercom matrix with ENG equipment or OB vans.
- Links between events and radio ad TV stations: Outside contributions.

COMPACT

EASY TO USE

REMOTE SUPPORT

WORLDWIDE

PLUG & PLAY

INTUITIVE SOFTWARE

NATIVE IP

REMOTE CONTROL

ON AIR

ON AIR USE

CAT.AV.23.10

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