AEQ

Multichannel AoIP routing system

Xcore

AoIP Routing System IP

With over a thousand channels per matrix, it is possible to configure systems with a total capacity of over five thousand channels..

THE X_CORE IS THE RESULT OF MORE THAN 20 YEARS EXPERIENCE IN DESIGNING AND BUILDING DIGITAL AUDIO ROUTING MATRIX SYSTEMS FOR BROADCAST APPLICATIONS

PRODUCT CONCEPT

X_CORE is AEQ's fifth generation of large digital audio routing matrices for broadcast and intercom applications. It has been designed for highly intensive workflows in the most demanding environments.

X_Core has been developed with the objective to fully benefit from our accumulated know-how and the specific suggestions from our customers with regards to our previous products.

The resulting System also incorporates the commitment to take full advantage of the technology that is available to us at this day and age. IP Audio Systems allow Inputs and Outputs, Process and Control to be distributed throughout different devices, and sometimes these devices are physically very distant from each other.

Further, the X_Core integrates the capacity to handle SDI video-embedded audio and provides the possibility to converge networked IP audio and video as defined in the SMPTE 2110 standards.

Last, but not least, X_CORE has been designed to be extended outside the walls of the production center, allowing connections with other, remote locations: different buildings within the same production center, remote production studios, transmitter sites, other stations operating within a network, broadcast venues in a multi-sports event, etc.

APPLICATION ENVIRONMENT AND ARCHITECTURE

The X_Core is a non-blocking, mixing, processing and distributing routing matrix for audio.

Operating as a general purpose audio routing-matrix, as an intercom matrix or combining both functionalities together, it can handle audio channels associated with IP video streams or embedded into SDI digital video signals.

The X_Core fully covers the needs for audio routing, distribution, contribution, program production and intercom at Broadcast and Production Centers or other locations where broadcast quality routing matrices are required. It can also operate as the audio engine of an ATRIUM mixer or set of mixing consoles.

Each matrix consists in one or several frames and can handle up to 1024 audio inputs and outputs. It is fully modular and redundant at all levels.

System inputs and outputs are available through different types of interfaces, in flexible quantities: digital AES/EBU, S/PDIF, analogue microphones, lines and headphones, long-range dark fiber links in 64 channel MADI format, and proprietary links with more than 1000 channels, among others. Using these latter combined with the TITAN concentrator, a matrix with up to 5080 x 5080 channels can be configured.

Further, through 64-channel AoIP input/output cards it is possible to integrate DanteTM / AES67 protocol compatible devices.

An X_CORE frame can incorporate as many Dante AoIP cards as required, and they can be connected to one or several separate Gigabit Ethernet networks. Also, audio flows accompanying SMPTE ST 2110-30 IP video signals can be ingested and generated .

AoIP can also be integrated in the system using a specific, RAVENNA-AES67 protocol compatible card. This card has the input/output capacity of 128 channels and allows to connect third party devices with these protocols. As in the case of the Dante compatible cards, an X_CORE frame can have as many RAVENNA AoIP cards installed as required connected to one or several separate Gigabit Ethernet networks.

It can also ingest and generate audio flows accompanying SMPTE ST 2110-30 and SMPTE ST 2110-31 IP video signals with NMOS control.

Finally, dual SDI cards with two inputs and two outputs, 2 x 16 audio inputs and outputs per card, allows de-embed and embed audio signals from/into SDI signals.

REVOLUTIONARY EVOLUTION

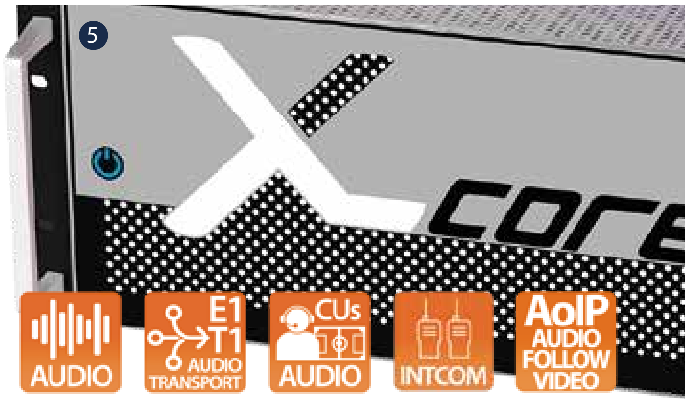
1. X_CORE is based on over 20 years of xperience and the BC2000D technology, used since 2003 in several hundred large audio installations worldwide.

2. The second generation, BC2000D MPX, was developed in 2006 to incorporate E1/T1-multiplexer audio transport infrastructure. Taken into service late 2006, the system was massively used for the first time by the EBU to cover Beijing 2008 Olympic Games.

3. In 2009, the third generation, NCS, was created to incorporate commentary audio at large sports events. It was installed and used for the first time at the Vancouver 2010 Winter Games.

4. In 2015, CONEXIA provided a solution to the intercom matrix needs in TV production centers.

5. Now in 2020, fifth generation X_CORE integrates IP connectivity on Dante, RAVENNA and AES67 protocols, as well as 3G-SDI, SMPTE ST 2110-30 and SMPTE ST 2110-31 formats, while remaining open to future standards.



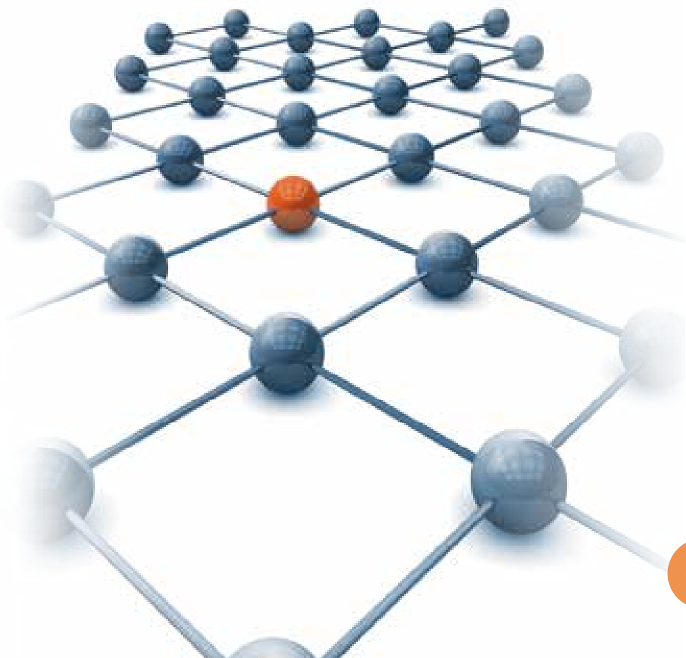
FUNCTIONAL HIGHLIGHTS

- As a non-blocking routing matrix, the X_CORE sums ,distributes and processes more than one thousand inputs and outputs per System. The System Frames can be located in a single central control room or distributed, at different areas of the installations but can also be geographically distant, i.e. different cities.

- The System can be centrally controlled or managed from several different locations. Such management is organised hierarchically through user, administration and supervision applications, control panels and mixing console surfaces.

- The configuration software provides multiple possibilities to customize the user interface.

- Being a central element of any installation, X_CORE is a solution with focus on security and redundancy at both hardware and software levels, esuring continuous operation - 24/7.



FUNCTIONAL FEATURES

Routing Matrix Audio Engine with up to 1024 x 1024 channels and redundancy at all levels.

Multiple audio formats are admitted through a great variety of I/O cards, among others; microphone, analogue line and headphone cards, digital AES/EBU, multichannel AoIP using Dante/AES67/SMPTE ST2110-30, multi-channel AoIP using RAVENNA/AES67/SMPTE ST2110-30/SMPTE ST2110-31 protocols, multichannel AES11 MADI, multichannel AEQ 1024-channel cards and cards allowing to de-embed and embed digital audio from and to SDI video signals

The modularity of the X_CORE allows the system to remain open to support other and future audio formats that the market may require.

The system may be distributed among different locations through fiber optics or installed within a LAN or WAN IP network.

The maximum number of available simultaneous processes depends on the number of DSP's that are installed on the Core or Engine. Equalisation, Dynamics (Compressor, Limiter…), Reverb and Delay capabilities, Audio generator for Test signals

The system can handle Mono and grouped signals such as Stereo and Multi-channel Audio, as well as Internal and Output sum-buses in different formats. It also provides the capacity to change signals grouping.

100% system reliability can be achieved. If configured for full redundancy at all levels, the system will continue to operate in the event of a failure. The disconnection of an input/output or processing card, or even the control module, doesn't prevent the operation of the rest of the matrix or loss of audio at any moment. If necessary, any part of the routing matrix can be hot-swapped and repaired.

IP Audio and Control networks can be redundant. In the event of a total power failure, the system will re-start configured at the exact point of operation it was prior to the failure.

Detection of alarms, which can trigger commands. Large number of physical GPIOs associated to input/output cards and the controller are available and an unlimited number of virtual GPIOs, that can be transported through the IP network, can be defined.

System time can be synched to NTP and clock for synchronizing the audio system can be extracted from WORDCLOCK and AES clocks, as well as from AES3, MADI, DANTETM, RAVENNA and SDI connections.

The system features a comprehensive and intuitive, multi-workstation and multi-matrix configuration and real-time operations software with hierarchical access.

X_CORE also includes level measurement function for each signal. These measurements can be sent through the network using an open protocol. AEQ can also provide software applications to display the vu-meters.

STRUCTURAL DESIGN CONCEPTS

X_CORE is based on a frame with the following architecture:

- Dual TDM bus
- Depending on configuration, handles 512 or 1024 channels.
- 21 slots for input/output cards that can be installed according to technical requirements at any moment.
- 2 slots for redundant controller card featuring QNX real-time operating system.
- 20 slots for processing cards.
- External redundant power supply.

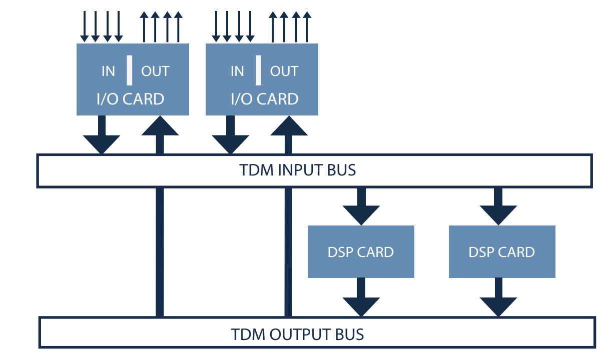
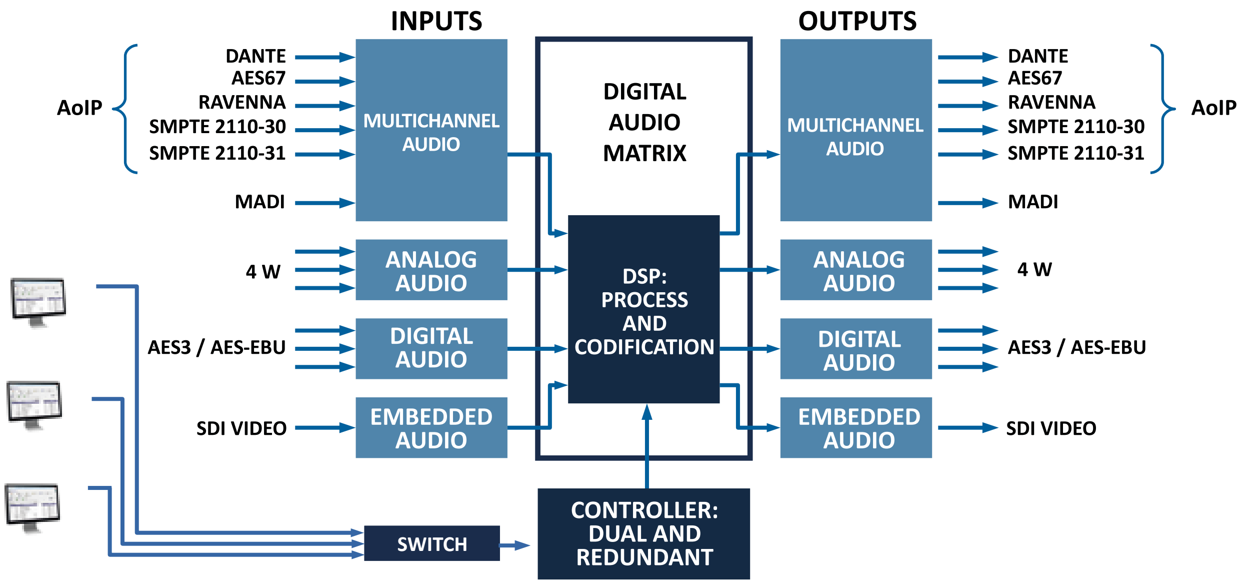


Diagram showing the organization of the internal system: buses, inputs, outputs and processing.



Control system diagram: the control network provides commands to the controller, that manages the routing of inputs to outputs and audio processing.

X_CORE

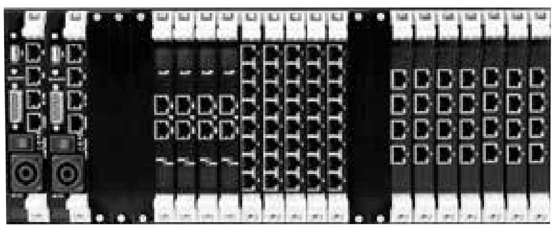
X_CORE is based on a standard 19", 4 RU chassis with three main system parts:

1. 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.

DSP

2. There are two different types of slots at the back of the unit.:
   a) The first two at the left side of the frame are reserved for the controller cards. One is of course required but a second one can be installed for redundancy.
   b) Further, to the right of these first two slots, the remaining 21 slots are dedicated to I/O interface cards for the different required audio formats.

a b



2 CONTROLLER CARDS

21 AUDIO CARDS

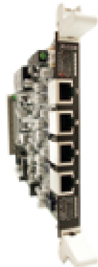
3. In the middle of the System frame or chassis, is located a back-panel. This the point of connection for the I/O Boards and the DSP's and also constitutes the transmission path for the system's 1,024-channel TDM bus.

I/O INTERFACES

The X_CORE system can integrate a wide variety of I/O interfaces compatible with the majority of audio formats available. Also, its modular design allows for the future incorporation of I/O boards and as may required as technology develops. These are X_CORE commonly used interfaces:

XC02

AES/EBU Digital input/output module. 4 stereo I/Os which can be individually configured as SPDIF. Also features 4 GPIO.

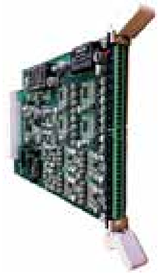


XC03

Microphone/line input module with electronic balancing and Phantom power supply.

XC03H

Microphone/line input module with electronic balancing and Phantom power supply plus two high or low impedance headphone outputs. It occupies two slots in the backplane.



XC09

Electronically balanced, analog line input/output module. Provides 8 input and 8 output ports.

XC10

I/O Module providing digital connectivity for KROMA Legacy Intercom User Panels. This card provides connectivity for up to 8 digital Intercom User Panels to the routing matrix.

XC11

64 channels AES10 MADI multichannel module. SFP port. Can be fitted with long-range fiber optic transceivers.

XC12

Dual 2x64 channels AES10 MADI multichannel module. SFP ports. Can be fitted with long-range fiber optic transceivers.

XC13

Link module for 1016 audio channels. Two systems or nodes can be linked together through optical fiber. SFP port, can be fitted with long-range transceiver modules.

XC18

Dual 3G-SDI card, with audio de-embedder and embedder for SDI video streams. Provides up to 2x16 audio input and output channels to the X_CORE bus.

XC19

VoIP intercom module with G.722 encoding. Provides 12 HD audio channels for Xplorer wireless belt-pack Systems.

AoIP INTERFACES

XC24

A DANTE™ /AES67 multi-channel AoIP Networking card for connectivity of up to 64 audio input and ouput channels.

XC24 Functional Description

XC24 is used to seamlessly interconnect AEQ devices. It also connects third-party, Dante-native and AES67 compatible devices. This latter may require previous configuration and setup.

When the AoIP Channels of the XC24 card are configured in the Dante ecosystem and using the Dante Domain Manager, it can also exchange audio with hybrid IP audio-video systems based on the SMPTE ST 2110-30 standard.

XC34

A RAVENNA /AES67 multi-channel AoIP Networking card for connectivity of up to 128 AoIP audio input and ouput channels .

XC34 Functional Description

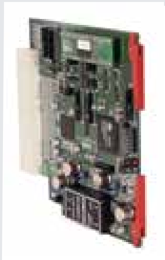
XC34 is used to seamlessly interconnect the system with third-party, RAVENNA -native devices and AES67 compatible devices. This latter may require previous configuration and setup.

It can also exchange audio with hybrid IP audio-video systems, based on the SMPTE ST 2110-30 and SMPTE ST 2110-31 standards with NMOS control protocol.

AUDIO PROCESSING MODULES

Up to 20 XC21 or XC22 processing cards can be installed in each frame. One DSP card needs to be installed to perform each type of process.

The type and quantity of required DSP cards in each frame is calculated as a function of the number of input/output cards, output sum-buses, number of signals to apply dynamic, frequency, reverb processing to, delays (and their maximum time), as well as the number of vu-meters to represent.



XC21

DSP card designed to carry out audio processing and routing. This card allows the system to establish cross-points and perform signal processing, such as: equalization, compression-expansion, VU-meters and delay.

XC21 processing capabilities details

XC21 DSP can perform 4 types of audio processes:
• ROUTING: used to create cross points between input and output channels.
• MIXING: used to sum input chanlnels to an output bus.
• TEST SIGNAL GENERATION AND VUMETERS:
-Tones, pink and white noise generation
- Signal and peak level measurement.
• SIGNAL PROCESSING:
-Frequency: 4-band parametric EQ low pass, high pass and band-pass filtering.
-Dynamics: compressor, limiter, expander, noise gate and DLP.
-Delay
-Reverb

In order to implement one or more processes of each type, at least one card per type of process is required. Also, a backup XC21 card can be added and configured as a backup Card. This card will automatically assume the function of any of the other XC21 cards in case of failure.

XC22

DSP card for the more demanding processing types of mixing and routing.

XC22 Functional Description

XC22 can perform 2 types of processes:
• ROUTER: used to create cross-points between input and output channels. It also allows to adjust levels for input and output channels as well as for the cross-point.
• MIXER: used to sum input channels to an output bus.
XC22 cards are used instead of XC21 when one of the above processes are used massively in a system. Also, a XC22 card can be added and configured as a backup Card. This card will automatically assume the function of any of the other XC22 cards in case of failure.

CONTROLLER MODULES

XC40

Main Controller Module for each frame. Two modules can be installed for redundancy.

XC40 card controls the frame configuration and its "relations" with the outside world. It also handles synchronization, alarms and the power supply of the frame.

It features an Ethernet connection to create a system control cluster.

It has 7 optically-coupled GPI plus 7 relay-operated GPO and incorporates the frame's non-volatile memory for the configuration and operation of the X_CORE routing matrix.

CONEXIA

Intercom Super-controller in 1U rack format. Two Conexia devices can be connected in mirror mode to achieve redundancy.

It controls the configuration of a whole intercom system based on X_CORE matrixes. It communicates with all the XC40 controller modules in each of these X_CORE frames through an Ethernet connection.

Conexia incorporates the non-volatile memory for configuration and operation as an Intercom System.

POWER SUPPLY AND FORCED COOLING UNITS

XC93

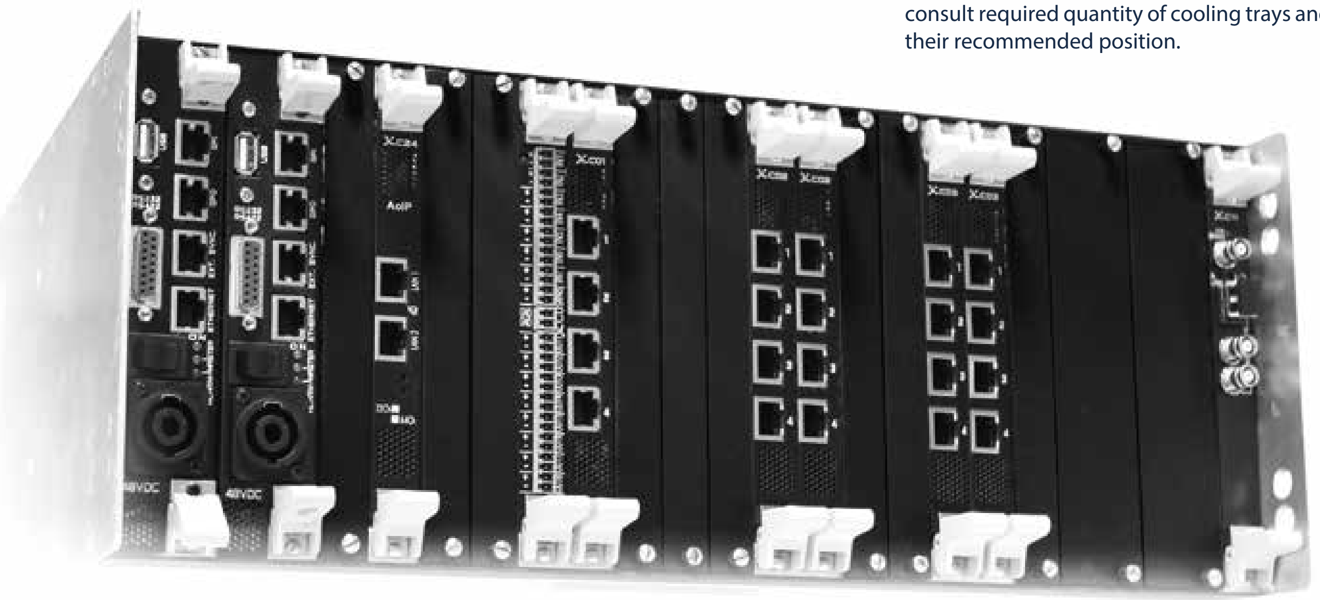
Redundant 800W power supply with hot-swappable cartridges. External unit with 2RU that provides power for up to 5 X_CORE frames (depending on their configuration).

XC95

Redundant 450W power supply. External unit with 1U rack height. It is designed to be placed on top of the X_CORE frame to improve the forced ventilation to evacuate the heat generated inside the X_CORE.

XC96

Forced cooling tray with air extraction at the rear and air intake at the front. XC96 should be installed when XC93 power supplies are used. If a single X_CORE is placed inside a rack, the XC96 needs to be located underneath the X_CORE. If two frames are are installed, the XC96 unit has to be installed between the frames. For larger configurations, please consult required quantity of cooling trays and their recommended position.



TITAN: MATRIX CONCENTRATOR

The TITAN matrix concentrator is a high-capacity digital audio matrix with up to 5080 x 5080 channels. It is equipped with five receptacles for SFP bi-directional fiber optic transceivers. Each port has a capacity of up to 1016 channels. It links to the X_CORE frames through the XC13 card installed in each X_CORE frame. This way, up to 5 X_CORE matrixes can be linked to build a fully non-blocking matrix.

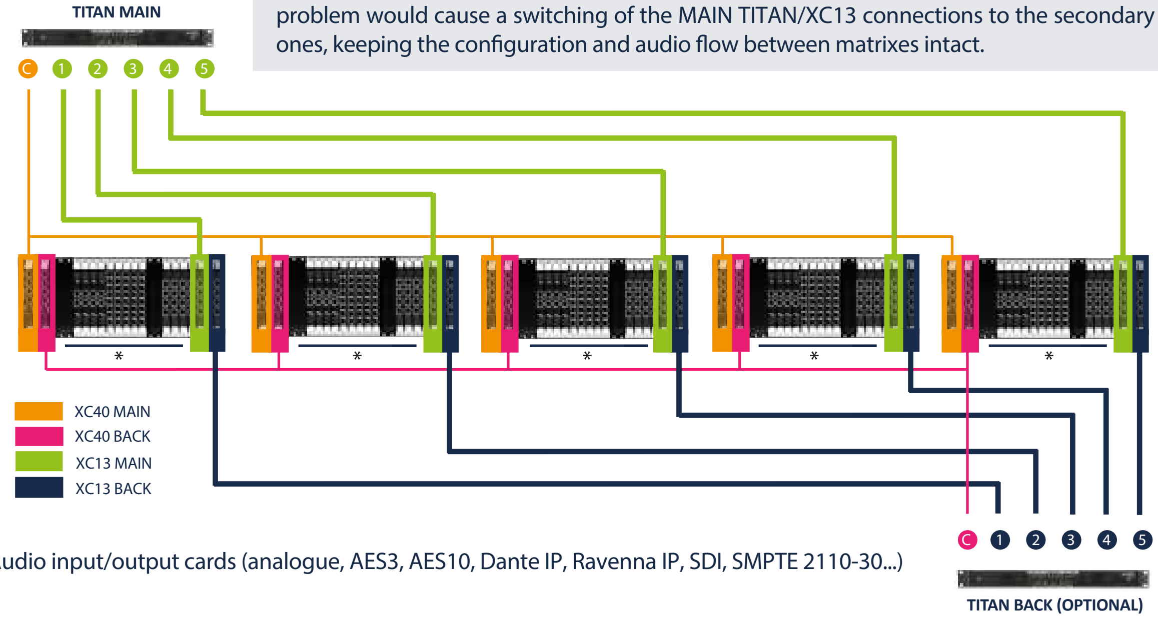
TITAN

Designed for uninterrupted operation:

- The interconnection bus is passive, without electronic parts in order to keep reliability high.
- Redundant architecture, seamless audio switching.
- Two controllers in cluster mode with a single virtual IP address.
- All the elements in the system are hot-swappable:
  - The front panel can be extracted and folded to provide access to the fans and the redundant power supplies with independent mains connections.
  - The rear panel allows for the extraction the redundant controller modules and the audio switching core.

Redundant Architecture

TITAN not only features dual cluster controllers and redundant power supply, but its architecture can also be duplicated for even further reliability as shown in this diagram: a second TITAN (TITAN BACK) can be connected to another 5 XC13 cards –one at each of the X_CORE matrixes-. Any problem would cause a switching of the MAIN TITAN/XC13 connections to the secondary BACKUP ones, keeping the configuration and audio flow between matrixes intact.



*Audio input/output cards (analogue, AES3, AES10, Dante IP, Ravenna IP, SDI, SMPTE 2110-30...)

NETWORK INTERFACES WITH DANTE AoIP CONNECTIVITY

Ya no es necesario acercar el cableado de las entradas y salidas al frame de la matriz. Las matrices, paneles de intercom o consolas, vendrán equipadas con interfaces AoIP que se conectan por IP con los interfaces XC24 de X_CORE. Si necesitamos introducir entradas y salidas de audio analógico de micro, línea o auriculares, o digital, en la matriz, instalaremos cerca de las fuentes y destinos de audio equipos terminales de la red AoIP Dante como los siguientes:

NETBOX 32 AD

This unit provides 32 local input and output channels that can be connected to the X_CORE through the AoIP network. These I/O's are available as 16 mono analogue and 8 digital stereo channels. It also features 16 GPI/O's that can be transported between different AEQ devices using Virtual GPIO.

NETBOX 8 AD

Netbox 8 AD provides 8 local input and output channels that can be connected to the X_CORE through the AoIP network. I/O's are available in 4 mono analogue and 2 digital stereo channels. The second digital stereo interface can be switched for a USB connection. It also offers 4 GPI /O's that can be transported between different AEQ devices using Virtual GPIO.

NETBOX 4MH

Netbox 4MH features 4 local microphone/line input and 4 output (stereo headphone + line) channels that can be connected to the X_CORE through the AoIP network. It has 4 GPI and 4 GPO that can be transported between different AEQ devices using Virtual GPIO. It also offers additional GPIOs for signaling terminals. PoE powered.

CONFIGURATION

The setup software application allows for the creation of up to 32 user levels -with full priority configuration and their own personal password- and for the optional creation of user groups.

Each matrix user is part of a user group associated to a list of active permissions. This means that access to the system's functions and resources can be granted or restricted to each user as a function of their abilities and range, required resources, the kind of program produced, etc.

The software includes a wizard for easy configuration of a matrix, by simply indicating the number and types of required inputs and outputs.

The configuration software connects the pc and all the matrix' frames via TCP/IP. Through the Software and as per the access rules and privileges that is controlled by the system administrator it is possible to configure all the physical and logical elements and as per the specific needs of each user.

The status of each card can also be monitored through the software and, if required, their firmware versions can be upgraded.

SUPERVISION AND OPERATIONAL MAINTENANCE

For system supervisors and administrators, we have created a powerful set of software tools that allows to control the entire system from one or several workstations:

- Real time display of DSP process charge.
- Network status display.
- Firmware upgrade.
- User and system configuration setup and modifications.
- Event and system fault log.
- Intercom, circuit monitoring and measuring through simple physical connections or real-time AoIP network monitoring through port listening of any router audio input or output form any audiocapable device connected to the network.
- Board status software.
- System Time synchronization through GPS receivers.

GPIO selection view

DSP resources usage view

CONTROL & SUPERVISION

XY connections view

System control can be accomplished centrally or in a hierarchically and structured way through the IP control Network or using GPI's from different consoles, dedicated panels, intercom user panels or PCs running the real-time control software.

Workstations for System Control can be classified and organized as per different user levels and following the operators different roles, e.g.:

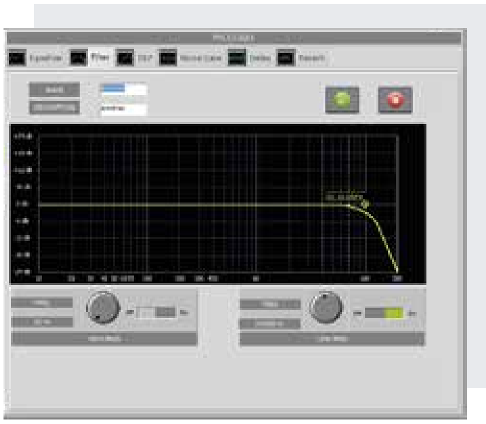
- Studio, sub-matrix operator.
- Studio mixing control operator.
- Central control matrix operator, etc.

Podemos crear, en estos u otros ordenadores, tantos vúmetros como sea necesario para controlar los niveles de entrada y salida de las distintas líneas, o exportar los datos a aplicaciones de visualización externas.

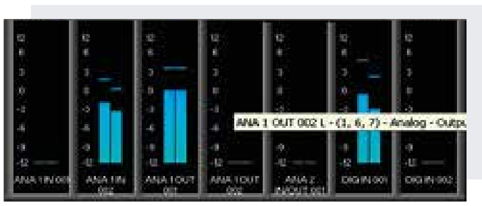
Real-time control Software

A wide range of operational control tools have been designed. These will be available to the different users depending on their role. These roles are defined by the System administrator and include for example, secure access, protection and priority policy. Among others, these are a few of the control tools:

- Real time XY control.
- Real time control per connection list.
- User-defined matrix views.
- Programming, display and editing of salvos, macros and clock switching.
- Time programming of actions in the matrix.
- Grouping and logical renaming of lines.
- Protection of lines and crosspoints.
- Definition of permission for summing of lines and multiconferences.
- Undo and panic functions, loss of modulation alarms.
- Vu-meter level monitoring, cue, intercom.
- Input, output and crosspoint gain.
- Predefined MACROS can be activated by configurable direct access buttons, that can be actioned with a mouse or touch screens.
- Instantaneous DSP load display.



Process adjustment view (Filter)



Customized vu-meters view

TECHNICAL FEATURES

INPUT AND OUTPUT

- Maximum capacity: 1024 circuits/frame or matrix. Non-blocking configurations of up to 5080 x 5080 circuits can be achieved linking several matrices to the TITAN concentrator.
- RF Protection.
- Electronically balanced microphone and line inputs. Analogue headphone outputs.
- Electronically balanced analogue line-level inputs and outputs.
- Optional transformer-balanced analogue line input and outputs.
- Digital inputs/outputs configurable as: AES/EBU (AES 3) and SPDIF, mono and stereo.
- Multi-channel digital input and output links as per AES10 (MADI), mono or stereo, one or two 64-ch links, 48 kHz sampling frequency, mono-mode or multi-mode fiber with SFP receptacles and removable transceiver cartridges.
- Audio over IP inputs/outputs, compressed G.722 for use in Intercom applications. 12 channels per card.
- Audio over IP inputs/outputs, Dante Protocol, compatible with AES67 and SMPTE ST 2110-30, 64 channels per card, 48 kHz sampling frequency.
- Audio over IP inputs/outputs, RAVENNA protocol, compatible with AES67, SMPTE ST 2110-30 and SMPTE ST 2110-31. NMOS Control, 128 channels per card, 48 kHz sampling frequency.
- Digital Audio inputs/outputs embedded in SDI/ 3G video, according to SMPTE 259 M, 292 M and 424 M standards. Each card has two input and two output interfaces with 2x16 audio inputs and 2x16 outputs.
- GPI and GPO (General purpose inputs and outputs for special applications):
  -Optically coupled inputs and outputs on some input/output cards.
  -Optically coupled GPI and relay-operated GPO on the controller cards.
- Internal synch of the X_CORE system. Accepts external synchronization through WORDCLOCK and AES11 connections, as well as through AES/EBU, AES10, DANTE (PTPv1 and PTPv2), AES67 (PTPv2) and RAVENNA (PTPv2). Synchronization from genlock input on the SDI card is also accepted.(PTPv2) y RAVENNA (PTPv2).

PROCESSING

- Internal sampling frequency: 48 kHz with 24 bits resolution.
- Internal bus format: two 32 bits per sample, floating point buses.
- Available processing functions (can be modified by the user in real-time):
  - Input and output routing.
  - Gain control for line-level signals from -12 dB to +12 dB.
  - Gain control for microphone level signals from -40 dB to +24 dB.
  - Stereo spatial distribution (Balance / Panorama).
  - Modulation continuity control.
  - Test generators: pink and white noise, configurable frequency tones, burst mode.
  - Dynamics: compression, limiting, expansion, combined DLP, noise gate.
  - Frecuency domain: high pass, low pass, band pass filters, 4-band parametric equalization, de-esser.
  - Delay.
  - Reverb.

CONTROLLER FUNCTIONS

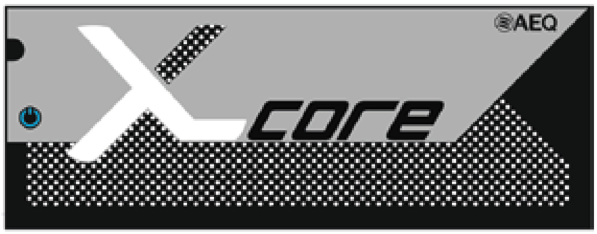
Features an industrial PC board with real-time operating system which stores the configuration data into a Compact Flash memory. It has the following functions, among others:

- Manages frame configuration and its relation with the outside world.
- Manages the system timing and concentrates and prioritizes the sync sources.
- Manages power supply alarms produced in any card, illuminating the “ALARM” LED if any failure is detected.
- Manages master/slave mode (when two cards are installed for redundancy).
- Power Supply management for the whole frame.

DIMENSIONS AND WEIGHTS

(WIDTH X HEIGHT X DEPTH; WEIGHT)

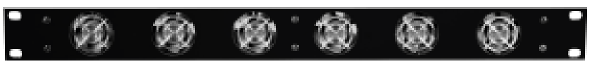
X_CORE Frame Central Unit: 4u x 19”
(482.6 x 266.7 x 450.0 mm; from 12 to 22 kg, 26.4 to 48.4 lbs).



XC95 Power supply and Fan Unit: 1u x 19”
(482,6 x 44,5 x 450,0 mm; 3,2 kg. 7 lbs).

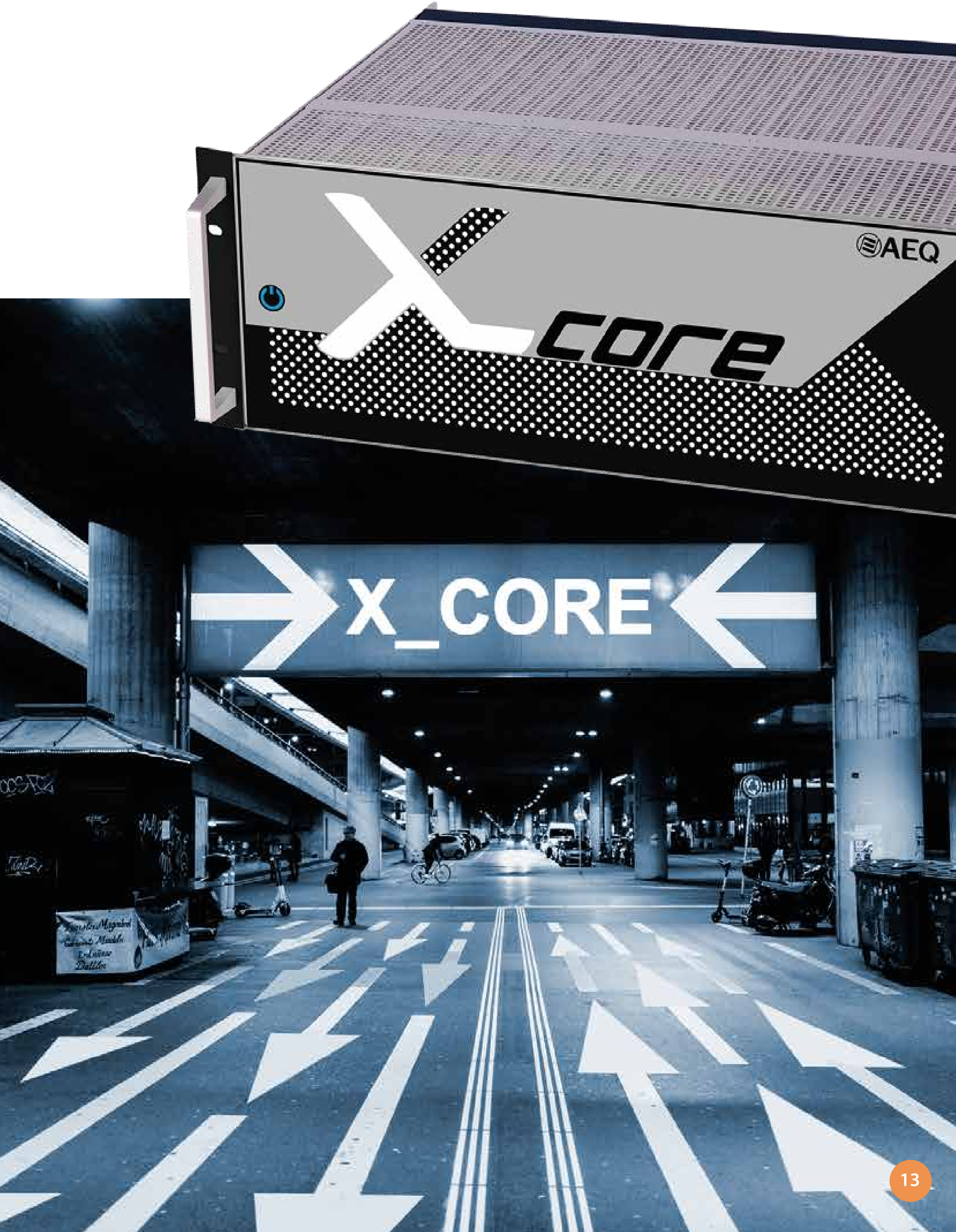
XC93 Power supply: 2u x19”
(482.6 x 89.0 x 360.0 mm; 8.2 kg, 18 lbs)

XC96 Fan Unit: 1u x 19”
(482,6 x 44,5 x 430,0 mm; 1,8 kg. 4 lbs).



Conexia Controller: 1u x 19”
(482,6 x 44,5 x 250,0 mm; 4,4 kg. 9,7 lbs).

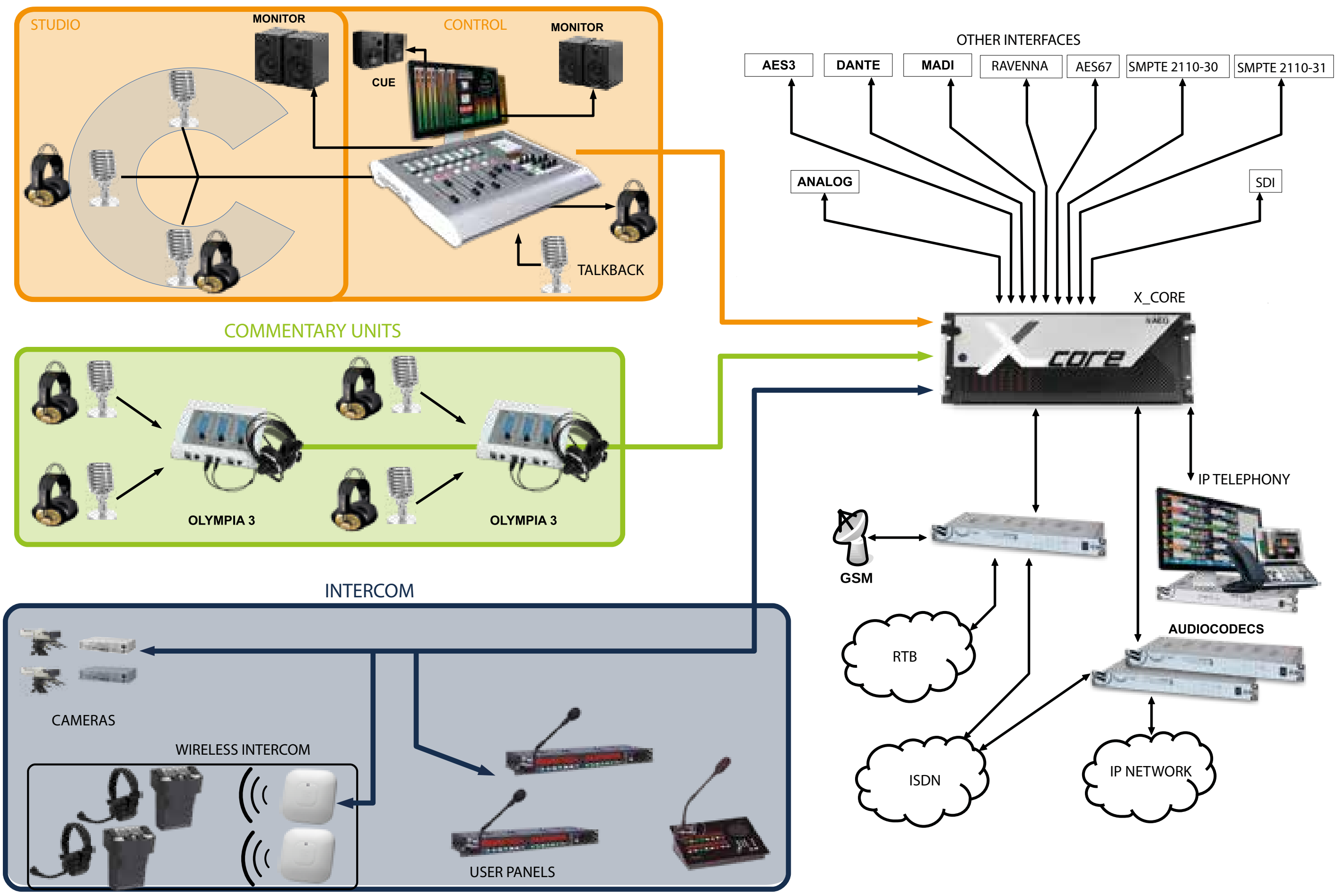
TITAN Concentrator: 1u x 19”
(482,6 x 44,5 x 450,0 mm; 4,0 kg. 8,8 lbs).


AEQ
core
X_CORE

X_CORE AS AN INTEGRATED BROADCAST, COMMENTARY AND INTERCOM AUDIO SYSTEM

Until now, available existing technology forced customers to operate different systems to manage audio and communications. These were separate and isolated systems that didn't allow for full inter-operability and to optimize HW resources.

The main purpose with this new generation of matrix systems is to offer a complete integration between audio, communications, and even video, simplifying operations and managing production on solid, fully redundant systems that offers signal processing and maximum audio quality. State-of-the-art technology that operates on IP networks with centralized control. In a nutshell, a leap forward to 360º content and workflow management.



In order to understand the flexibility of the X_CORE as a multi-purpose audio routing matrix, this synoptinc provides an overview of a system providing several functionalities:

In this diagram, we can see three sub-systems to the left:

- Audio contents for radio and TV Broadcast production.
- Commentary audio.
- Wired or wireless Intercom for operational coordination.

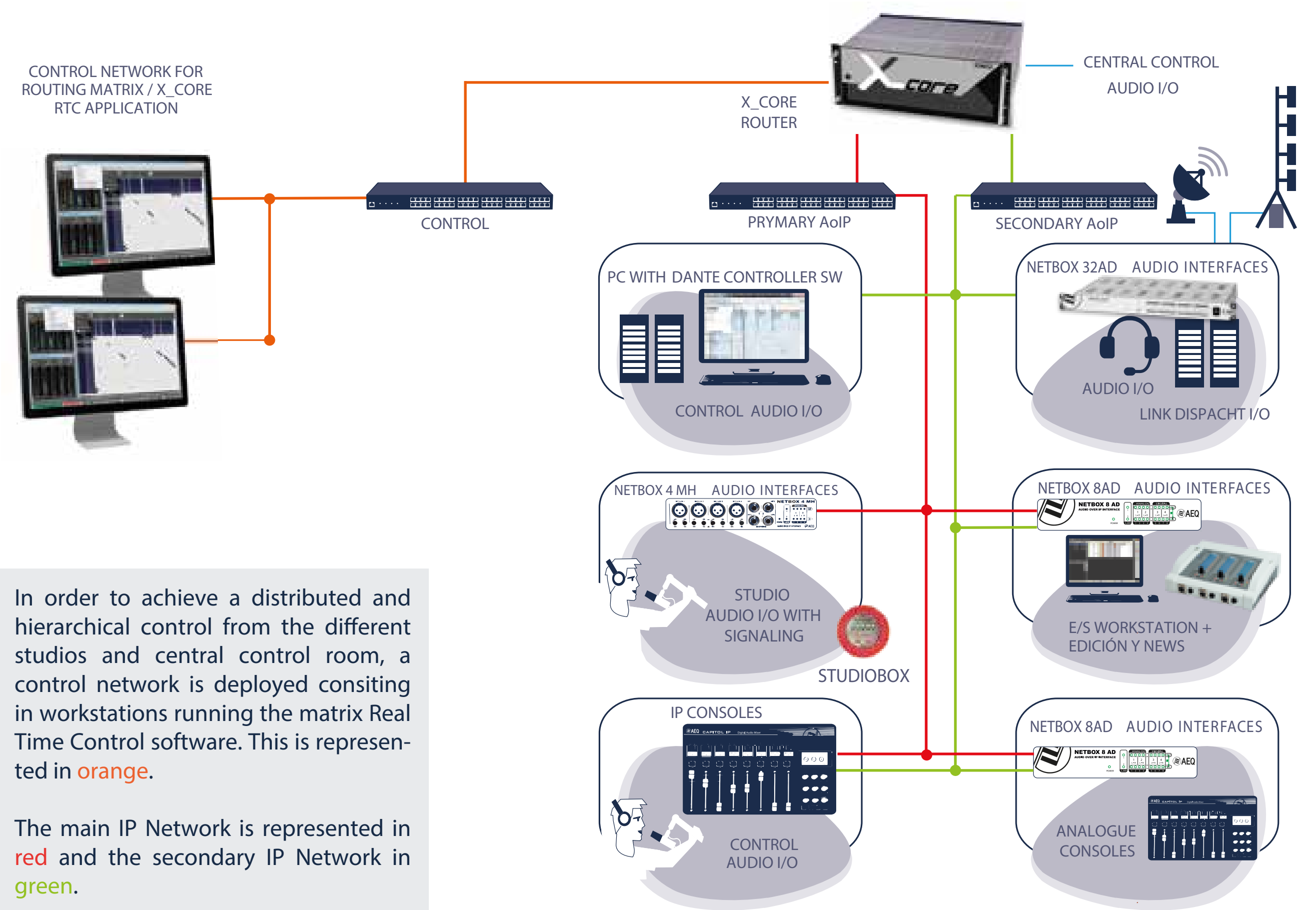
At the right side, the X_CORE is related to:

- Other audio or audio/video systems in different formats.
- External communications systems based on IP Audio, i.e., compressed, high or broadcast quality audio for distribution or contribution, alternatively, voice over IP for communications purposes.

This way, X_CORE can be integrated into a wide range of system types. The following diagrams are showing other solutions based on systems that typically are in use at radio and TV stations.

X_CORE SYSTEM EXAMPLE FOR A RADIO STATION

The system is built around a main X_CORE matrix, where all audios in the Dante network are routed to using a bank of XC24 interface cards.



In order to achieve a distributed and hierarchical control from the different studios and central control room, a control network is deployed consiting in workstations running the matrix Real Time Control software. This is represented in orange.

The main IP Network is represented in red and the secondary IP Network in green.

There is a set of inputs and outputs in the central control using matrix cards. Programs from all studios will be assigned to analogue or digital audio outputs in the matrix, together with other audio signals required in the control room such as auxiliary program outputs or clean feed for phone systems. Signals required in the studios such as TV tuners, audiocodecs, radio off-air tuners, etc. that will be assigned to the matrix analogue or digital inputs.

One or more NETBOX 32 AD interfaces can be installed in the TX Dispatch in order to extract signals from the system and going to STL's and satellite Uplinks, for example. Inversely,,audio signals from receivers and downlinks, OB-vans, etc. will be ingested to the system through the same Netbox 32 AD interfaces.

A NETBOX 8 AD can be installed in News recording cabins/booths or editing suites, providing audio input and output for the audio workstations through a bi-directional USB link. Local audio can also be connected to the mixing console using analogue and digital I/O connections. The same NETBOX 8 AD unit will provide IP connectivity for analogue or digital studios without AoIP network connectivity.

This way, a station can be AoIP Networked without having to abandon existing, older, but working equipment.

AEQ digital consoles can be provided with the corresponding multi-channel AoIP interfaces. The most important outputs of each console can be routed to the multi-channel interfaces: master, auxiliary, clean feeds, etc. so they can be used at any other location within the station. At any moment, and as required, it is possible to assign and route the signals with origin from studios, cabins, central control and dispatch to the audio inputs of the interface.

At each Studio, one or several NETBOX 4 MH will be installed to make all the microphones available to the mixing consoles in the AoIP Network and to send the necessary headphone output signals that the console is providing. Further, the STUDIOBOX Remote desktop panel is connected, providing the required Studio Signalling and remote control through its programmable keys.
Thanks to this, we benefit from the performance of the X_CORE TDM system (making live or scheduled routing changes possible), distributed control and processing of the cross-points, audio mixing, alarms, macros and salvoes executed manually or automatically, VU meters ...), with the simple installation and flexibility that an AoIP system offers.

X_CORE SYSTEM EXAMPLE FOR TV AUDIO AND INTERCOM

X-CORE - CONEXIA is an audio & intercom router. It consists in a CONEXIA controller (or two, if redundancy is required) and a set of X_CORE frames. As many Dante AoIP interface cards as required are installed to provide service to the AoIP intercom user panels and the different audio interfaces in studios and galleries, as well as audio mixing consoles featuring Dante connectivity.
Further, XC19 VoIP cards are installed for wireless intercoms or to connect with remote intercom matrixes or panels.

XC24 X_CORE cards provide access to the Dante AoIP network. XC19 cards, on the other hand, connect to the VoIP network.

Through the AoIP network, in red, the matrix has access to:

• Digital audio mixing consoles with AoIP connectivity.

• Olympia 3 Commentary Units.

•IP audiocodecs to connect via Internet or 4G to remote audiocodecs.

• IP audiocodecs to offer access to several types of phone networks using gateways.

• Intercom User Panels with VoIP connectivity.

• Netbox AoIP interfaces to inject and extract high-quality analogue and digital audio signals to/from the system.

The VoIP network, depicted in green provides access to:

• An Xplorer intercom system with Xplorer wireless beltpacks and iPod/i-Pad devices running the Xvirtual app using Wi-Fi access points.

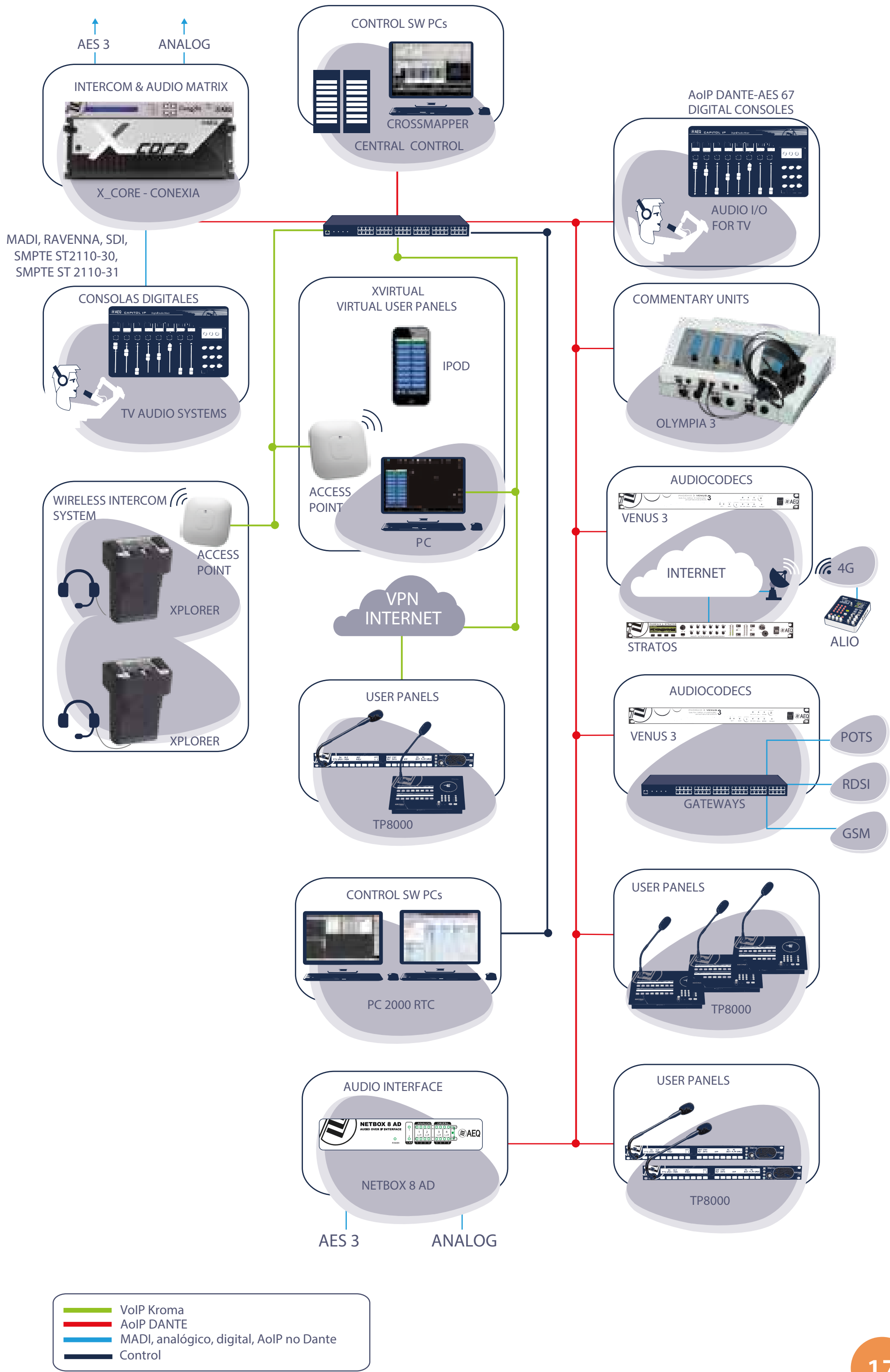
• Wired PC user terminals running the Xvirtual application.

• VoIP Intercom user panels connected through a VPN or Internet.

Following the blue lines, the following connections can also be observed:

• Cards with analogue and AES3 ports.

• Cards for other audio for TV equipment such as:
- RAVENNA, AES67 or MADI-connected digital mixing consoles.
- IP audio and video systems according to SMPTE ST 2110-30 and SMPTE ST 2110-31 standards.
- SDI video systems with embedded audio.

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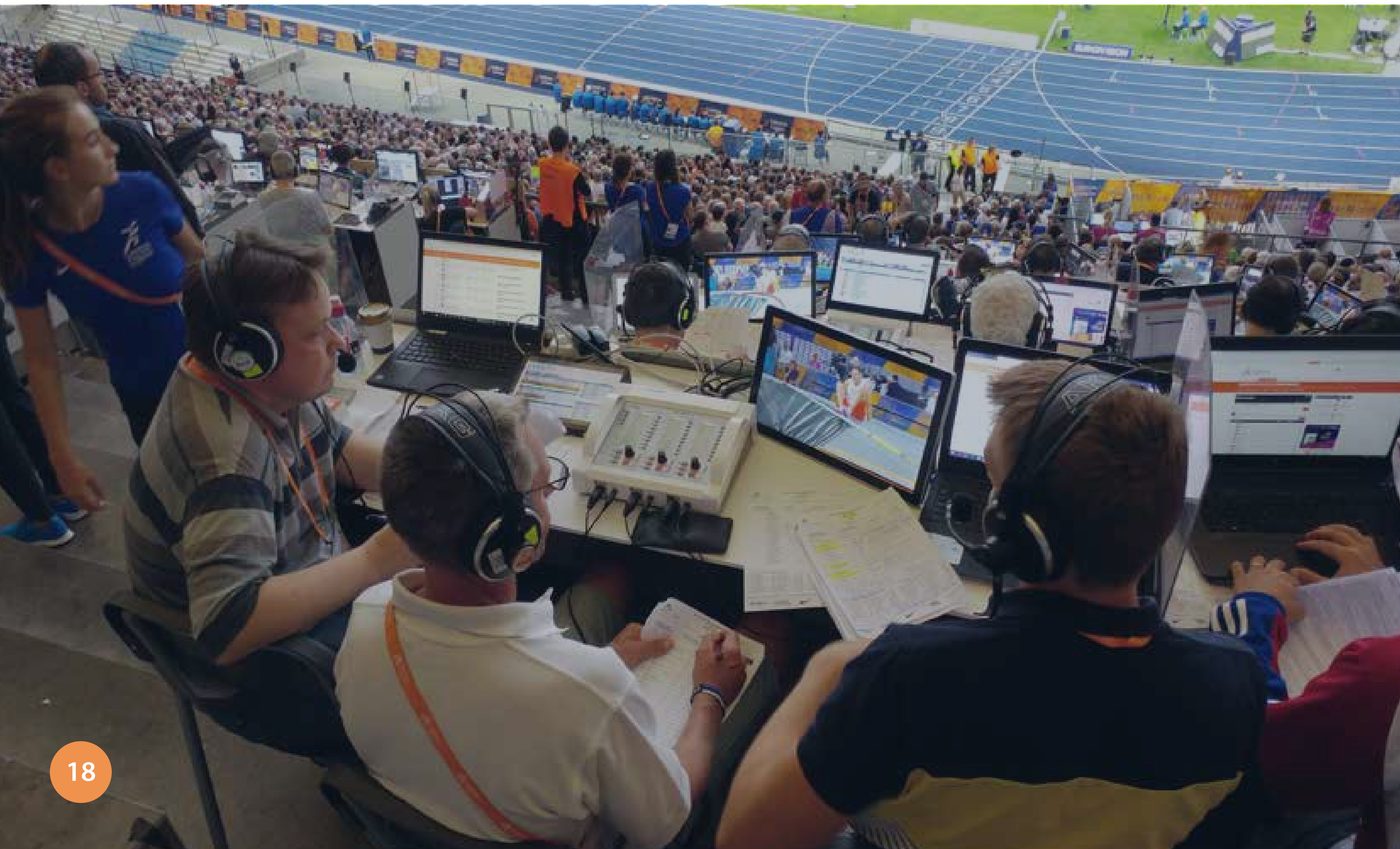


X_CORE SYSTEM EXAMPLE FOR A MULTI-VENUE SPORTS EVENT

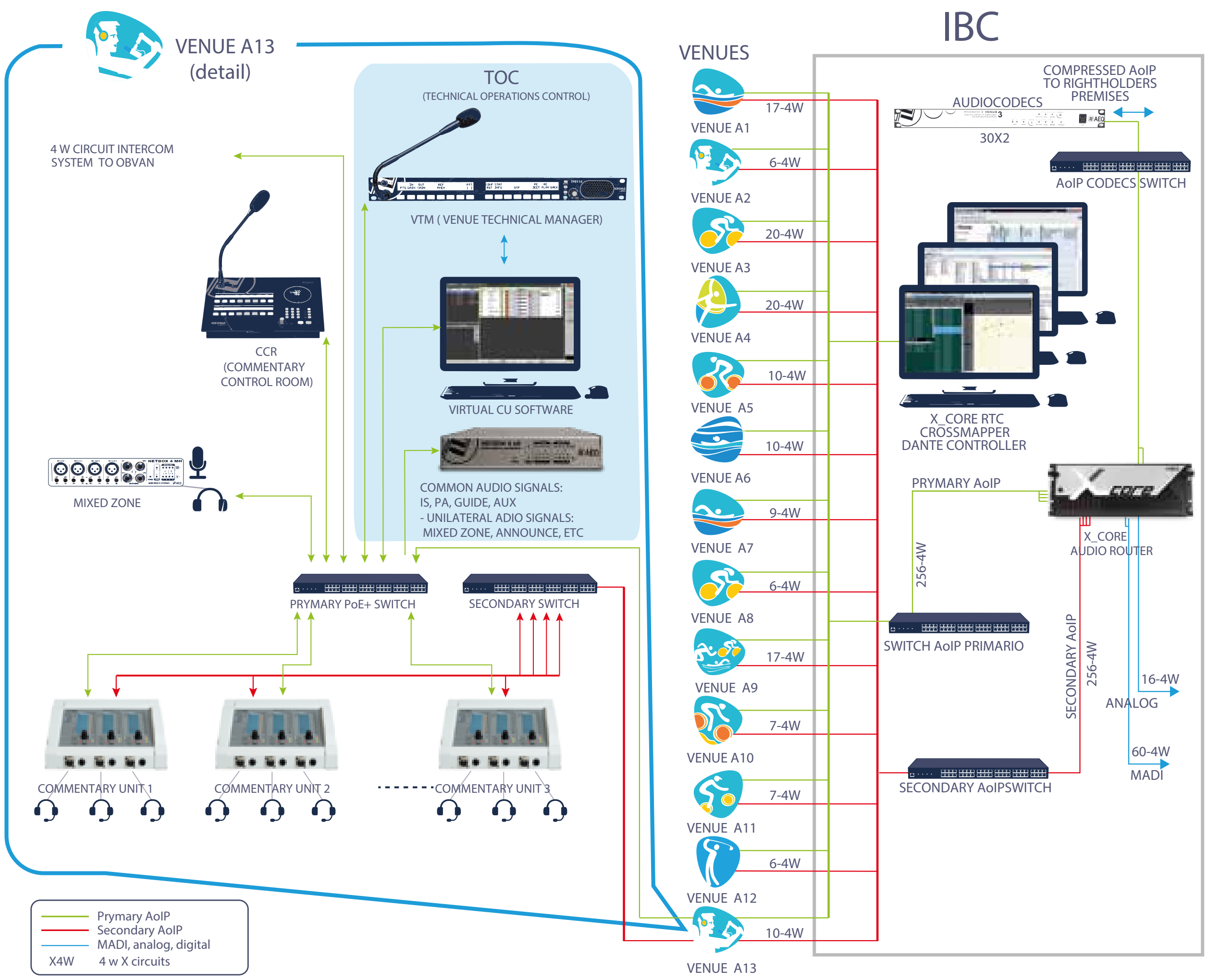
Next, part of an actual sports event, the system is deployed at several venues – some of them separated by tens of km - is described. It features more than 70 IP commentary positions, equipped with OLYMPIA 3 commentary units connected through an IP network to a X_CORE router in the IBC (International Broadcasting Center). Also, its intercom and the audio transport system for the rights-holding broadcasters – all using Audio over IP - are depicted.

The trunk infrastructure consisted in a 512x512 channel X-CORE matrix, connected to a twin Matrix located at a secondary Broadcast Center and using redundant MADI links. The Secondary IBC, located at more than 1200 Km away from the IBC, provided multi- and unilateral services to its own Venue Cluster For the shake of simplicity, the remote IBC or its associated venues are not depicted in this diagram.

Real-time supervision and operation was carried out with X_CORE RTC control software, while the static routing infrastructure was established using Dante Controller.

Audio transport between all the venues and centralization of the IBC was accomplished through an AoIP network with Dante's redundancy logic.

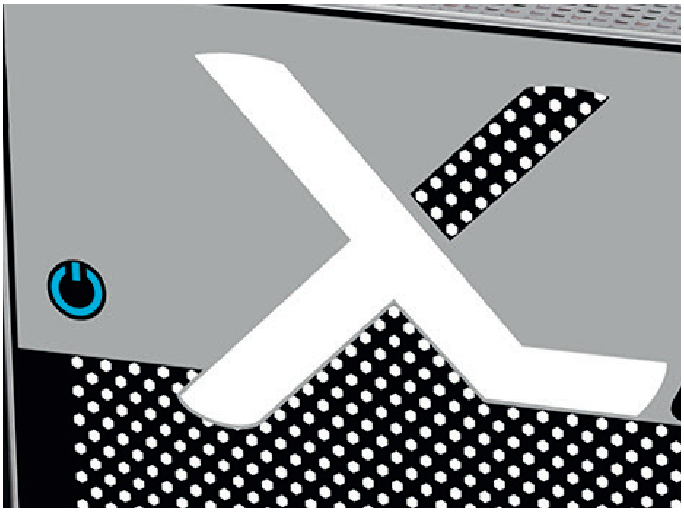
Audio transport and intercom circuits with the affiliated stations is established using 30 AEQ VENUS audiocodecs, also using Dante local audio over IP connectivity.



The system at each venue, depicted at the left, is implemented by connecting redundant AoIP Dante OLYMPIA 3 commentator units to a TOC (Technical Operations Centre for audio monitoring and control systems) built around the AEQ NETBOX 8 and NETBOX 32 AoIP interfaces and the control application for OLYMPIA 3.

For mixed zones where journalists interview athletes, AEQ NETBOX 4 MH AoIP interfaces are installed, providing the necessary microphone inputs and headphone outputs.

For further information, please follow this link: "Application Note Olympia 3 at a multivenue event": http://www.aeq.eu/products/olympia-3.



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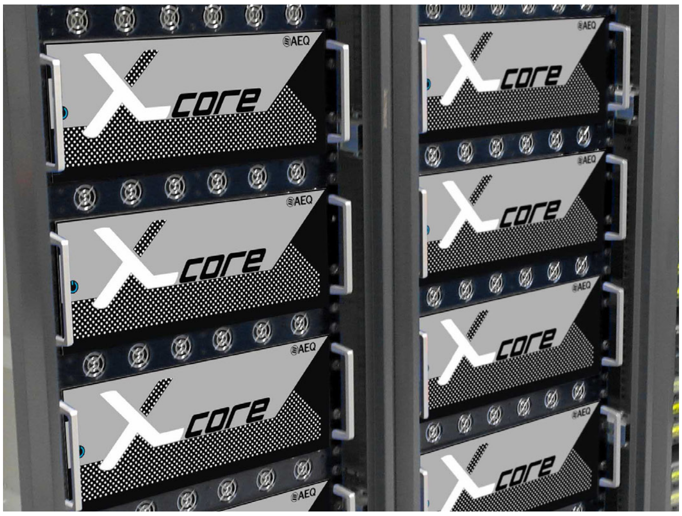

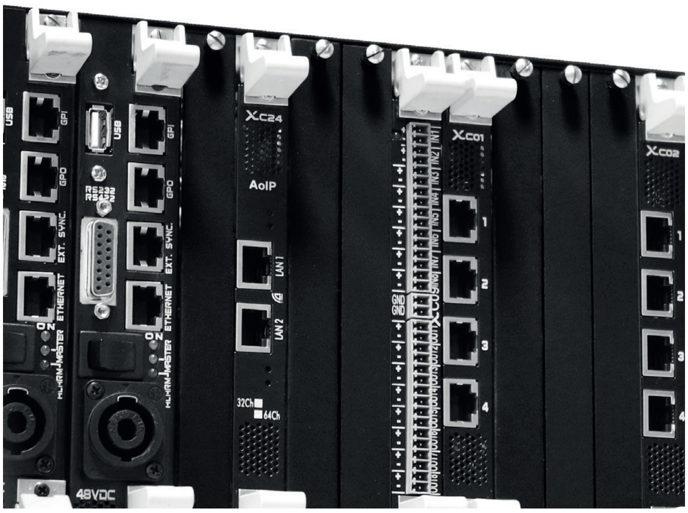

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