Digital Audio Broadcast Mixing Console

Specially designed for Radio and TV Broadcast Audio Production. Up to 1000 channels of local or networked AoIP Audio, controllable through up to 96 faders on up to 8 pages or snapshots.

CONTROL AND FLEXIBILITY
AT THE HIGHEST LEVEL

Index

Flexible, motor-independent surface

With tools for maintenance and remote operation

Gran simplicity of use

Automatic mixing with autogain and automix

A single motor controls up to 6 surfaces

Joint control with complementary equipment

ATRIUM is the result of more than 20 years of experience in digital audio mixing consoles for broadcasting.

ATRIUM is AEQ's 3rd generation of high channel count On-Air and Production digital audio mixing consoles. ATRIUM is specifically designed to adapt to the most demanding and workflow intensive broadcast environments.

It 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.

It 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.

On another token, the convergence of Audio an Video over IP is becoming a reality.

Different sets of tools have been developed based upon the current state-of-the-art technology, to generate a new concept, where the different components in the system are able to genera-te combined functionalities which are far superior to those offered by each of these components separately.

Further, this product provides greater simplicity of configuration and system integration. It can easily be extended beyond the walls of the production center in order to, for example, connect to outside broadcast events or other broadcast centres, either permanently or temporarily as part of a Broadcasting Network or a Multi-channel Audio over IP system of any required size.

General Overview

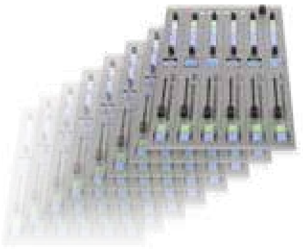
- 100% Digital. 100% Native IP.
- The control surface is completely independent from the Core or Audio Engine.
- 6 faders modules. It offers up to 96 motorized faders
- Up to 8 pages for each fader.
- It is possible to operate up to 6 Control Surfaces from one Engine.
- This Engine or Core can be installed in any part of the LAN.
- ATRIUM supports multi-studio mode. The same control surface can be used for several Studios. Switch the control between studios by simply pressing a key.
- It also supports multi-control mode: all or part of a console control functions can be managed in parallel or separately, in order to be used by producers or presenters.
- Modular audio engine with redundant controller and power supplies.
- It can handle up to 1024 x 1024 Audio Channels.
- Audio Input and Output boards for Microphones, Headphones, Analogue, Digital AES/EBU and USB signals, Multichannel AoIP Dante / AES67 / SMPTE ST2110-30, MADI Links, AEQ High-speed 1024 Channel Links among others current formats.
- Open system that can be expanded by the development of new cards to interface with future audio formats, either linear, compressed, independent or embedded into video or other data streams.
- The number of internal sum-buses, output sum-buses is configurable according to the needs.
- AUTOMIX and AUTO-GAIN can be applied to any of the signals.
- Unlimited configurable CLEANFEEDS and MULTIPLEX (N-1). CONFERENCE MODE provides the ability to create talkback to and between several different channels before routing them to “ON-AIR”.

- The maximum number of available simultaneous processes depends on the number of DSP's that are installed on the Core or Engine. Equalization, 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.
- Edit Channel or Signal names temporarily or permanently, on large sized, full colour touch-screens and a very intuitive software.
- 100% reliable. The system is able to continue performing after any failure. IP Audio and Control networks can be redundant. Critical elements, controller and DSP cards can be installed with an active backup card, with automatic switchover in case of failure.
- In the event of a total power failure, the system will re-start configured at the exact point of operation it was before the failure. The system includes a set of alarms that can be programmed to trigger actions related to such alarms.
- These actions can be executed through the numerous GPIO's and unlimited Virtual GPIO's that the console is capable of handling.
- Time synch via NTP. Console synchronization can be extracted from WORDCLOCK, AES11/MADI, AES 67, DANTE and RAVENNA clocks, including PTPv1(2002) and PTPv2 (2008. IEEE 1588).
- The Set-up and configuration Software is very intuitive and complete. It is also possible to configure the console through the proper console control surface if required. Once the console has been set-up and configured, computers are not required to operate.
- The console features a function consisting on the measurement of each microphone level. That information is sent through the IP network using an open protocol which is readable by Visual Radio video switchers.

Fader Module, Functional Description (ATX FCH)

Each Fader module includes 6 Fader channels. A maximum of 16 Fader modules can be installed for each console, thus, a maximum of 96 Fader channels can be available. All fader module functions have been carefully developed to provide simple and precise control on all the channel's parameters, taking into consideration even operation of channels handling signals which are more complex than mono or stereo, such as bi-directional, 4-wire or 5.1 multichannel ones.

Each module can handle 8 pages or configuration layers. This makes a total of 768 channels that can be controlled through Faders. It is also possible to create direct routings where specific fader channel control is unnecessary.



Next to the page swap keys, there is a touch & turn rotary encoder to perform precise adjustments on the touch screen.

By default, each channel has a pre-assigned encoder to handle the signal's gain control, another one for the balance/- panorama, a channel SELECT key, a channel ON and a channel OFF key. There are also 5 programmable keys and a channel fader.

ON key is preassigned for main routing activation.

Displays associated to each fader

The display represents the name of the signal assigned to the channel, its gain, balance or panorama, phase (L+R+, L+R-, L-R+, L-R-) signal configuration (Stereo, Mono L, Mono R, L+R) and the On-Air status, among other functions.

In addition, if the console fader channel has been programmed for A/B Channel operation, the display will show the name of the signal that is currently not active (hidden signal) and whether it is routed to the Program (ON-AIR) output or not.

In configuration mode, it shows gain levels, balance or panorama positioning and source selection.

Fader Module, Functional Description (ATX FCH)

SELECT key is preassigned for channel selection functions

All the keys of the console's control surface are programmable. As part of this configuration, their background color can be defined, choosing between 7 different and 2 levels of brightness. Available colors are: red, green, blue, cyan, yellow, purple and white.

Channel's context information displays.

4 indicators showing whether the present signal is unbalanced, equalization or dynamics are active and whether the channel's predefined nominal Gain has been modified.

Fader cursor buttons are touch-sensitive, so context information about the channel can be displayed on the screens by just touching the correspon-ding fader cursor.

Each channel also has a 100 mm PFL Vu-meter with a scale ranging from -12dB to +12 dB.

Faders can be configurable so nominal gain is set at 80% of the fader travel, or full throw, i.e. nominal gain corresponds to 100% of total travel.

ON and OFF keys can be programmed to activate/deactivate each channel fader signal, or alternatively only the ON button can be used, , so the OFF key remains available for other functions.

Backlight button protectors with configurable colour that allow the creation of work groups.

Channel ON key can be programmed to be automatically activated by opening the fader and/or turned OFF when fader is returned to the the beginning of travel.

Fader channels can also be programmed to operate without the ON and OFF keys, using only the fader to open the channel.

Control Module, Functional Description (ATX CTL)

USB interface.

CUE loudspeaker.

VUMETERS, CUE AND USB

3 pre-assigned 100 mm Stereo Vu-meters for PROGRAM, AUDITION and CUE. The Vu-meters provide precise reading of values from -34 dB to +20 dB based upon the nominal level setting.

MAIN SCREEN AND ASSOCIATED KEYS

The main screen can be found at the center of the control module. It is surrounded by 24 programmable keys.

CONTROLS ASSOCIATED TO CONTROL AND STUDIO

The operating areas associated to those technical environments can be found in the lower part of the control module.

Each area has:

- 5 monitoring keys that can be configured to operate as exclusive or single source monitoring. Pre-configured signals are assigned to these keys and they can be changed at any moment.
- Two screens to label signals and represent each listening level.
- Level adjustment for Control and Studio monitors as well as for control and studio primary & secondary headphones. Primary and secondary headphones can operate in CLEANFEED mode, i.e. monitoring all signals present except own signal.

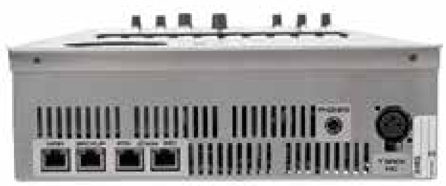
4 specific coordination talkback keys. Additional keys can be programmed in order to increase the number of destinations. Also, dedicated controls can be found as Talkback and self-control microphone and CUE speaker level adjustments.

TOUCH & TURN encoder for the precise adjustment of parameters on the display.

RESET CUE key.

REAR CONNECTIONS

The control module provides a microphone input and the operator's headphone output at its rear panel. This microphone input (which can provide Phantom power supply), can be used for intercom or orders, but it can also be sent ON AIR as a self-control microphone. These signals are exchanged with the X-Core engine through an AoIP interface which also carries the CUE loudspeaker signal and the front panel vu-meters signals.



The functionality of the control module can be customized for producers, adding intercom destinations, or for presenters, adding ON AIR indicator and remote cough cut and CUE keys. Basic intercom and monitoring modules for producers and presenters are also available.

MAIN SCREEN

The main screen is located in the central area of the control module.

It is a 10,1" Colour, multi-touch TFT display, is where the majority of the control surface options can be configured and accessed. Further, the display of the Control Module is also the support to any of the installed fader modules, when the optional Fader associated TFT displays, have not been installed.

It's surrounded by 24 programmable keys on 16 different pages or layers, providing a total of 384 programmable keys. Activation modes: ON, OFF, FLASHING and HOLD. As part of this programming, 7 different colors with two brightness levels can be chosen for each key. Available colors are: red, green, blue, cyan, yellow, purple and white.

The active programmable key labels in the visible page are shown at the top and bottom borders of the screen, so the keys can be easily recognized at a glance, as well as the functions activated and represented in each one.

The idle screen shows the following indicators:

1. Clock with system time and date.
2. Phase or correlation meter for the CUE-associated signal.
3. User, configuration memory and active programmable keys page indicators, as well as information about any studio microphone open and on-air control.

4. "PK PAGE": to change the programmable keys page.
5. "TIMER": gives access to clocks and timers configuration.
6. "SNAPS": configuration of the snapshots.
7. "USER": to change user.
8. "SYSTEM": provides access to some system's configuration functions.

If no optional TFT display has been associated to a fader channel module and a SELECT key associated to any of the faders of this module is pressed, the configuration options for this fader will be presented on the Main TFT of the Control Module.

FADER SCREEN MODULES (ATX SCN)

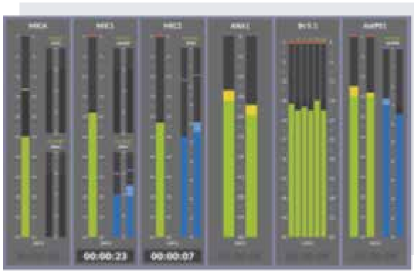
An optional TFT display associated to all or even each of the fader module/s will greatly simplify console workflow and the operations related to the fader channels.

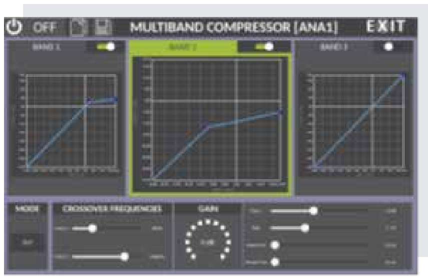
Assignment of channels to screens is configured in a flexible and dynamic way from the control surface. The standby screen shows a signal level meter as well as information about the hidden signal (A/B mode) and the feedback or return signal (if any). For 5.1 signals, a vu-meter is shown for each component. When the SELECT key of any channel is pressed, its parameters are displayed and can be modified on the multi-touch display. If a console doesn't have an associated fader screen module, functions associated to the SELECT key activation are presented on the control module's main screen.

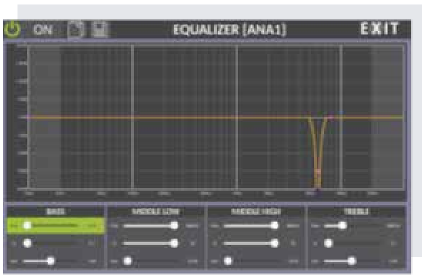
Provides a short-cut to the AUTOMIX and AUTOGAIN functions and displays the active channel's assigned long name, ON AIR timer, routing and active processes names, among others.

Phase, mode, routing or sensitivity, phantom activation and processing changes are applied with a single click on the display.

The processing screens allow for simple adjustment by simply touching and dragging the screen or, alternatively, just use the TOUCH & TURN encoder for higher precision. The Equalization, Filter and Compression processes, can be edited on the graphical layout or representation of the process as well as using the virtual faders below the graphic. Created processes can be stored in memories and assigned to channels.







OPTIONAL CONTROL AND METERING DISPLAY MODULE (ATX MTR)

An optional TFT screen associated to the control module allows for the incorporation of additional metering instruments that may be required for demanding operations:

- Spatial phase metering.
- Loudness metering according to EBU R128.
- Selection of the signal to be measured.
- Receives the signal to be measured from the Dante™ network.

1. A spatial representation of M (Momentary Loudness) is displayed on Y axis while S (short-term loudness) is represented on X-axis.
2. For L and R signals, peak level and VU meters are displayed, in “dBFS”, “dBTP” or “dBu” units according to configuration.
3. Also, maximum loudness levels since the last reboot or within a given integration time are provided, and M / S / I (Momentary, Short-term and Integrated) loudness values are also shown according to BS-1770.
4. Below, a loudness bar indicator according to EBU R-128 standard can be found, with simultaneous M / S / I (Momentary, Short-term and Integrated) loudness indications.
5. The right window area provides the following indications (from top to bottom): integrated values of the loudness range measurements (LRA according to EBU Tech 3341). Integration time (“I Window”) since last reboot and integration time (“I Timer”) since last reboot.
6. Also, the calculation time control buttons, “Start”, “Pause” and “Reset” can be found, according to EBU Tech 3341.
7. Down at the left, a representation of phase correlation between both channels in the measured signal is presented.

VIRTUAL ATRIUM: (ATX VIR) – VIRTUAL CONTROL SURFACE

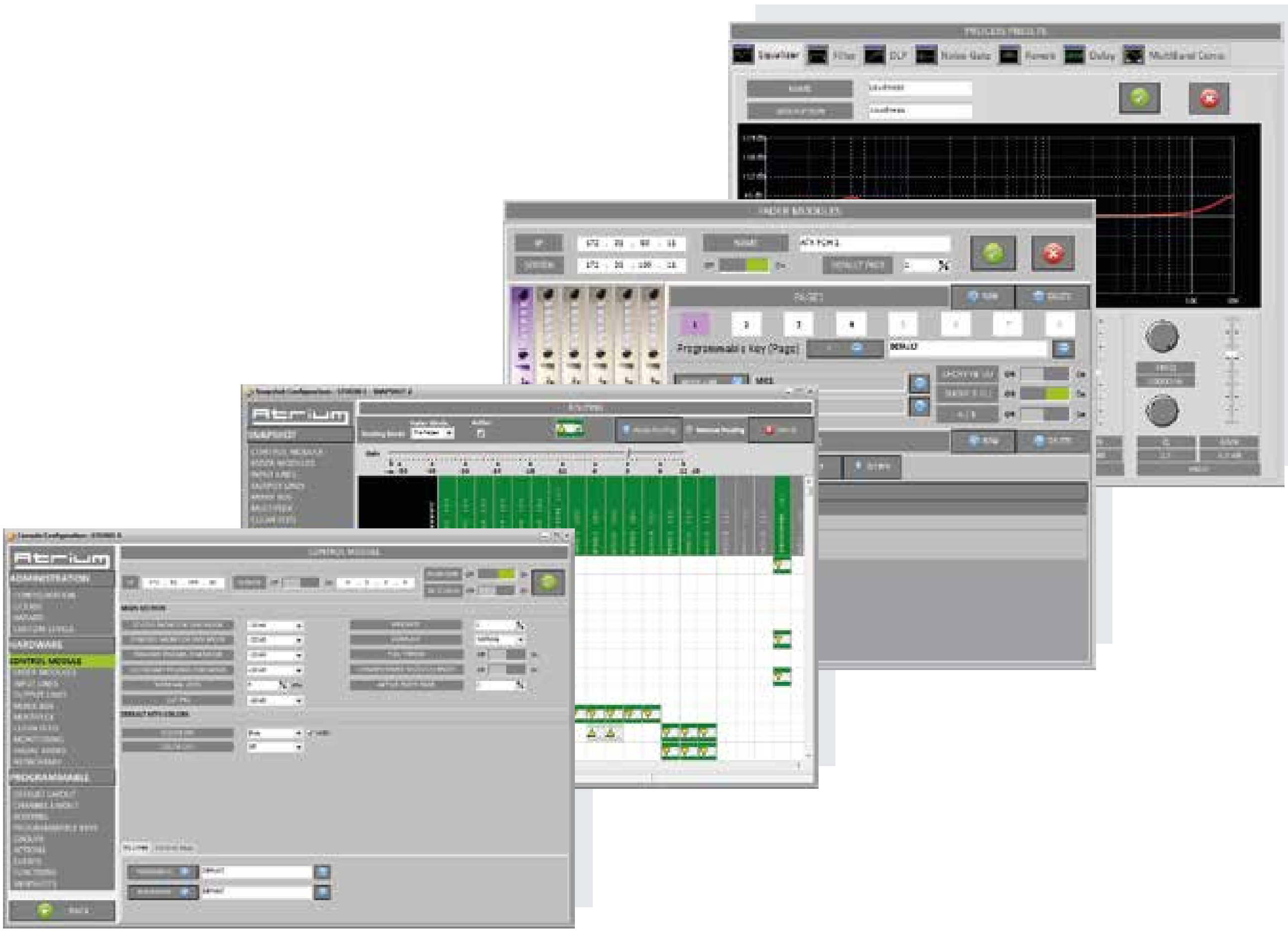
VIRTUAL ATRIUM is a Windows and iOS compatible application. The APP faithfully reproduces every detail of the control surface, as well as all the operating features of switches, rotary encoders, displays and faders.

All functionality of the physical control surface is available in this software application. It can run alongside the console or completely replace the control surface, temporarily or permanently, locally or remotely.

With this application installed on your iPad, you can assist the operator remotely or make adjustments when the operation is unmanned, for example during weekends.

An instance of Virtual Atrium can be used on any control surface. This way, a Virtual Atrium instance can be operated on each surface simultaneously when they share a single X-Core engine.

For consoles with more than 12 faders, the fader bank can be scrolled left or right to gain access to a hidden fader.



SETUP SOFTWARE

The configuration software application includes a wizard for quick configuration of a console by simply indicating the number and type of inputs and outputs required.

The whole system is fully configurable and allows you to specify, among other things, the number and type of mixer buses, input channels, type of processing to apply to each channel of the console, output routing, etc...

It supports customization of the signals according to the conditions programmed by the user or the functionality of the keys and encoders on the control modules.

The control surface can be completely customized for classic workflows. For example, each fader has an A/B configuration, with sends to predefined sum-buses, which channels are N-1 and duplex - allowing each “user” to receive the mix of all the audio contributions except for its own signal -, Alternatively, flexible programming of keys is possible, allowing for dynamic modifications of the console routing, controlling external equipment such as audiocodecs or routing systems, talk-show or phone-in systems, broadcast automation or intercom systems.

Each user of the console belongs to a group of users with a list of activated access rights. This effectively means that different users’ access to system functions and resources can be granted or restricted depending on their skills and capacity, required resources, type of program, etc.

CORE AND SURFACES

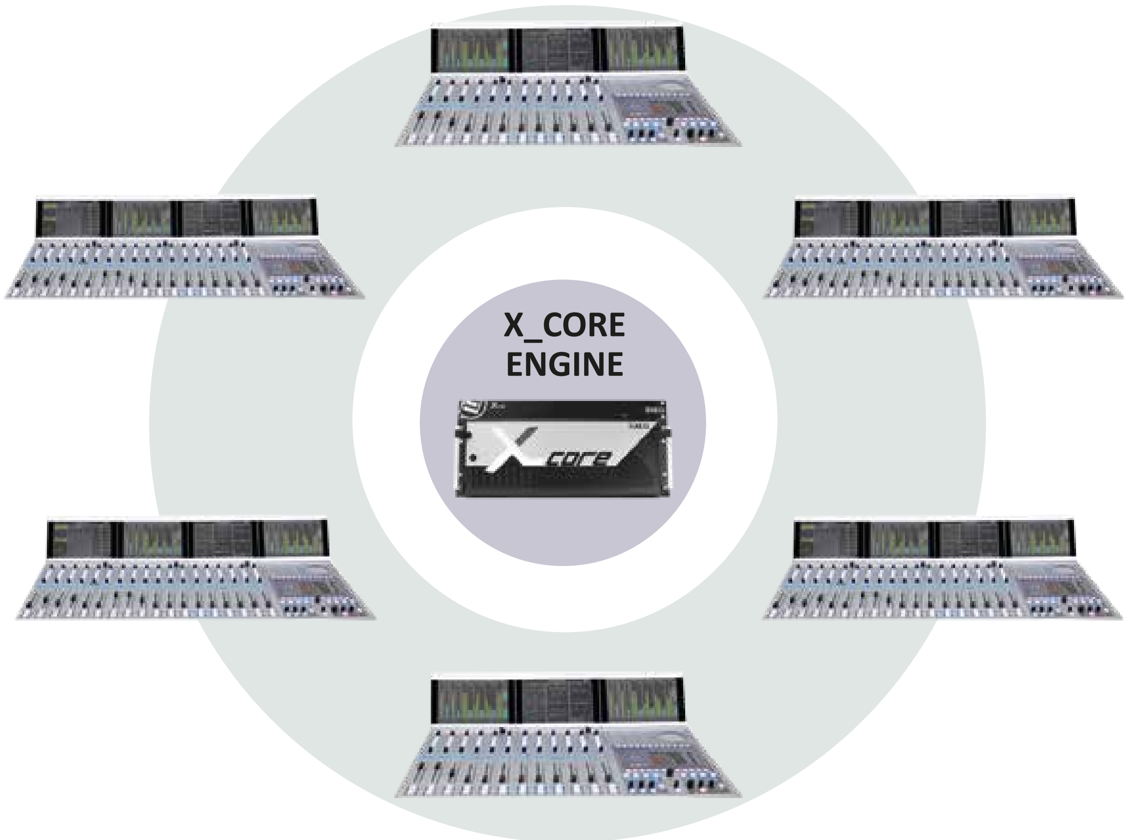
Up to 6 physical control surfaces, each one operated or substituted by a virtual surface, can be connected to a X_Core engine.

In this engine, which can comprise one or several frames, different kinds of audio input / output cards are installed. Its intelligence resides on each frame's control card or pair of cards. Processing capabilities, on the other hand, are implemented by the DSP cards.

Using its AoIP input / output cards, it can be connected to other AEQ or third-party devices. It can also be used to connect the audio inputs and outputs located on the console surface.

Another special usage of the AoIP cards is the connection to other devices such as Netbox 32 and Netbox 8, which include analogue, digital and USB inputs and outputs, as well as GPIO to distribute the console's inputs and outputs.

The Netbox 4 MH studio IP interface, is especially useful when used together with Atrium as a microphone and line input, headphones and line output, and GPIO interface. Netbox 4 MH gain control is integrated in the Atrium console.



ATRIUM MULTI-STUDIO SYSTEM DIAGRAM

Next page represents an Atrium system comprising 6 studios on a single X_CORE engine.

Starting at the top of pag. 15, we see that each control console can be replicated by an instance of the Atrium Virtual software. The physical and virtual surfaces can operate independently or in parallel.

Next, we can see the 6 controls which can share a X_Core engine. Audio between each surface and the engine is always carried over IP. At the right, the control speakers for the three first studios have been represented with IP wiring, while the speakers for the other three studios (depicted at the left), are connected in analogue format.

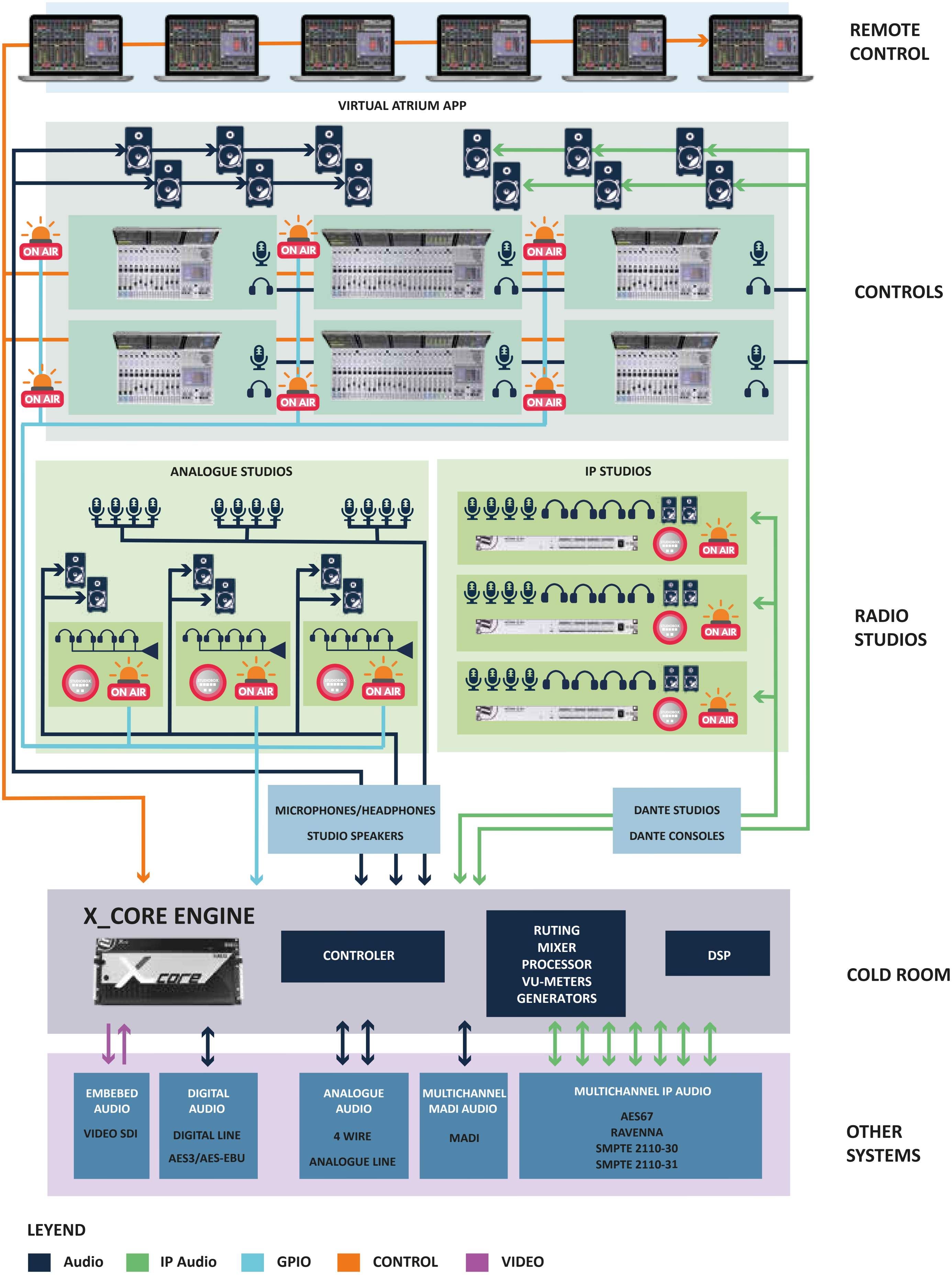
At the center area, the studios are depicted. Again, the first three ones are connected to the engine using IP connections by means of Netbox interfaces inside the booth to digitize and transport the microphone, speaker, headphone and signaling signals while, at the left, the rest of studios are represented, with conventional analogue microphone, headphone, speakers and signaling connections.

Next, the X_Core engine itself is shown, with its most important functions: audio input and output, DSP audio processing and system control.

At the bottom area, connections between Atrium and other system in different audio formats are represented: Multichannel IP, multichannel MADI, analogue, digital, SDI-video embedded in several formats.

ATRIUM MULTI-STUDIO SYSTEM DIAGRAM

- 3 Studios with IP-connected link and 3 studios with analogue audio connections.



X_CORE

REDUNDANT MODULAR ENGINE. FUNCTIONAL KEYS

Broadcast audio mixing, processing and distribution matrix. When specifically configured, it can work as the audio engine of an ATRIUM console or set of consoles. It can also operate as a general purpose audio matrix, intercom matrix or a combination of both.

Solution based on security and redundancy, both at the hardware and software level, in order to ensure operation 24 hours a day, 7 days a week.

When operating as an ATRIUM system, it can handle up to 1024 audio inputs and outputs. It is completely modular and provides redundancy at all levels of the system. Inputs and outputs are provided through specific Audio interface cards that can be installed in the required quantities: AES/EBU, S/PDIF or USB, analogue line, microphone and headphones, dark-fiber long-range 64-channel MADI links and proprietary fiber optic links with more than 1000 channels, among others.

Further, when using 64 channels Dante AoIP input/output cards, signals can be exchanged using IP with other Dante™-AES67 compatible equipment. An X_CORE frame can accommodate as many Dante AoIP cards as required, and they can be connected to one or several Gigabit Ethernet networks. It can also ingest and export audio streams in SMPTE ST 2110-30 format, that go along with IP video signals.

Also, when using 128 channels Ravenna AoIP input/output cards, signals can be exchanged using IP with other Ravenna-AES67 compatible equipment. An X_CORE frame can accommodate as many Ravenna AoIP cards as required, and they can be connected to one or several Gigabit Ethernet networks. It can also receive and export IP audio flows in SMPTE ST 2110-30 or SMPTE ST 2110-31 format, that go along with IP video.

Finally, using SDI cards up to 2x16 audio channels per card can be exchanged, with two SDI video interfaces up to 3Gb bit rate, extracting and inserting the embedded audio signals associated to each SDI video streams.

Also, its modular design allows for the future incorporation of I/O boards and as may required as technology develops.

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. Equalization, 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.

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. 100% system reliability can be achieved.

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. Incorporates alarms and execution of actions triggered by them.

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, DANTE™, RAVENNA and SDI connections.

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.

TOPOLOGY

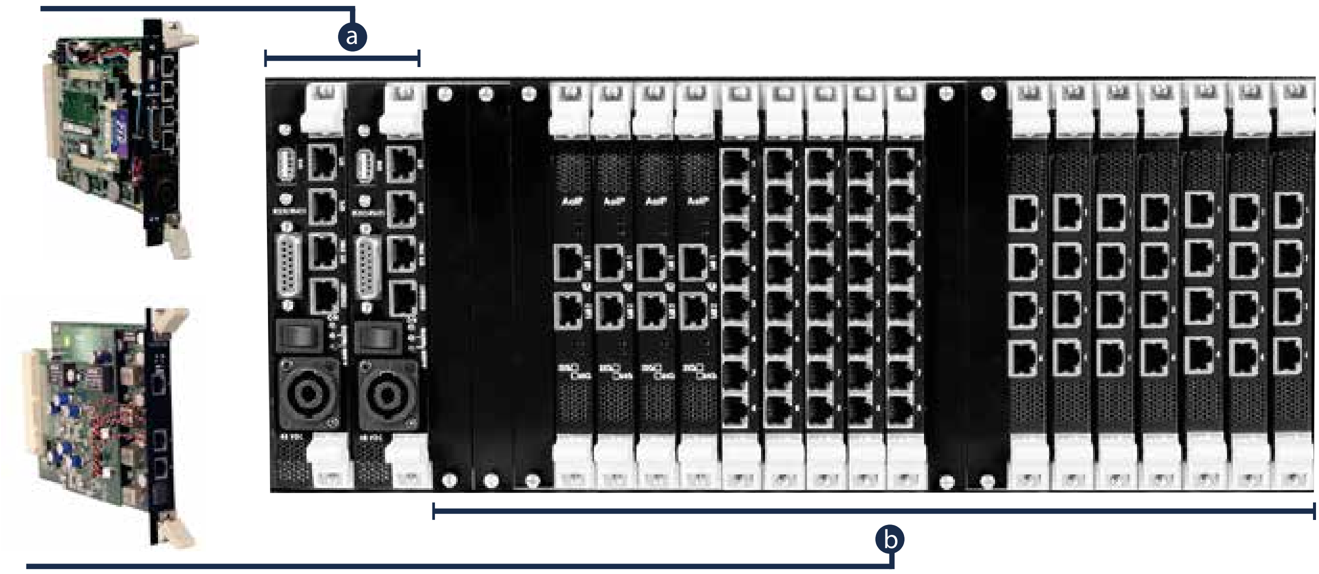
X_CORE is based on a standard 19”, 4 RU chassis with three main system parts:

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.

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.

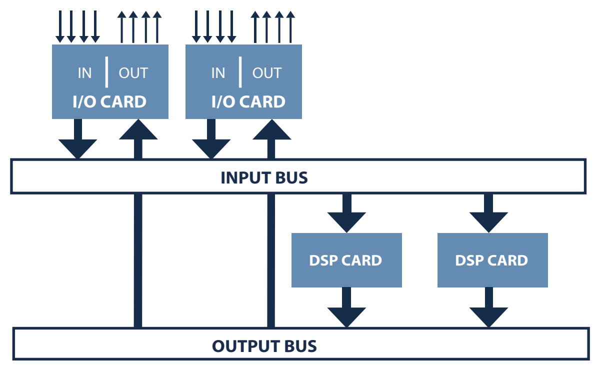


3 In the middle of the System frame or chassis, is located a back-panel. This the point of connection for the I/O Boards and also constitutes the transmission path for the system’s 1,024-channel bus.

STRUCTURAL DESIGN

X_CORE is based on a frame with the following architecture:

- Doble Bus, 32 bits per sample.
- 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.
- 20 slots for processing cards.
- External redundant power supply.



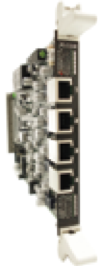
Internal system diagram: buses, inputs, outputs and processing.

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

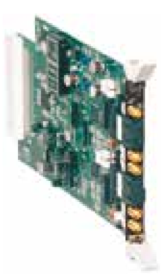
Electronically balanced, analog line input/output module. Provides 8 input and 8 output ports.

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 output 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.

CONTROLLER MODULE

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.

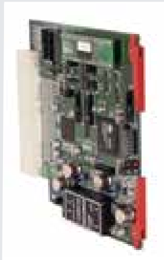
It features an Ethernet connection to create a system control cluster.

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.

AUDIO PROCESSING MODULE

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.

XC21processing 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 channlels 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.

POWER SUPPLY AND FORCED COOLING UNITS

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.

This is the most common power supply and cooling system when there is a single X_CORE engine installed.

XC90

Redundant 300W power supply unit with hot-swappable cartridges. External 2U rack height device.

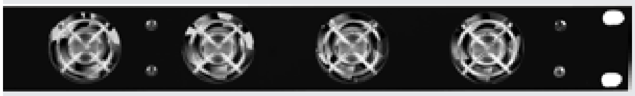
This is the most adequate power supply system when there is a single X_Core frame but replaceable cartridges are required in order to avoid maintenance stops. It doesn't provide cooling to the X_Core frames, so our recommendation is to complement it with XC96 modules.

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).

This is the most suitable power supply system when there are several X_Core frames installed on the same place and replaceable cartridges are required in order to avoid maintenance stops. It doesn't provide cooling to the X_Core frames, so our recommendation is to complement it with XC96 modules.

XC96



Forced cooling tray with air extraction at the rear and air intake at the front. XC96 should be installed when XC93 or XC90 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.

COMPLEMENTARY DEVICES WITH DANTE CONNECTIVITY

It is no longer necessary to ensure that the Audio I/O wiring is terminated in proximity of the Matrix frame. The matrices, intercom panels or consoles will come equipped with AoIP interfaces that connect via IP to the XC24 interfaces of X_CORE. If we need to ensure I/O connectivity for Analogue, Digital, Microphones or Headphones, we will install, close to the sources and destinations for such formats, DANTE AoIP Network interfaces like these:

NETBOX 32 AD

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

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.

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. Its gain control is integrated into ATRIUM surface. Especially suitable for TV or radio studios.

SIGNALING TERMINAL

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. It is useful in the presenter position in radio studios rooms for news and chats.

Technical Specifications

GENERAL

- Multiple control surfaces: up to 6 control surfaces per audio engine.
- Modular engine and control surface design.
- Control surface size configurable from 6 to 96 physical faders.
- 100 mm, motorized, conductive plastic faders with capacitive touch sensor.
- Protection against radio frequency and static electricity.
- Internal sampling frequency: 48 KHz at 24 bits/sample. Internal 32 bit bus.
- Time synchronization based on NTP protocol. External synchronization source via WORDCLOCK or AES 11. Synchronization can also be extracted from MADI, AEQ-multichannel, DANTE (PTPv1-2002) and (PTPv2- 2008. IEEE 1588), and RAVENNA (PTPv2- 2008. IEEE 1588) links.
- Modular design allowing for the adjustment of the system's inputs and outputs and to the requirement of each installation.
- Hot-swappable modules: the extraction of any part does not affect the functionality of the rest of the system.
- Automatic fail-over in case of DSP or controller card failure or extraction.
- Redundant power supply.
- Start up configuration can be chosen between last settings or default configuration
- AUTOMIX and AUTO-GAIN can be applied to any of the signals.
- Flexible logical signal grouping: mono, stereo, 4 wires and multi-channel signals.

INPUTS AND OUTPUTS

- Up to 21 multiple input/output modules per frame. Complete flexibility in regards to configuration of up to 1024 audio inputs and outputs.
- Gain adjustment for all Audio signals, even hidden ones.
- Selectable balance/panorama control for all channels.
- Selective phase inversion.
- Electronically balanced microphone inputs with selectable phantom power. Headphone outputs.
- Electronically balanced analogue line-level inputs and outputs.
- Transformer/balanced analogue line inputs and outputs available as an option.
- AES/EBU digital inputs and outputs. SRC (sample rate converters) on all digital inputs and outputs. They are compatible with AES/EBU or S-PDIF digital signal formats.
- MADI (AES10) Multi-channel inputs and outputs, mono and stereo.
- 64 channels digital audio through optical fiber or BNC connection with SFP and removable cards.
- 1024-channel AEQ High speed links digital audio optical fiber connections.
- 64-channel AoIP Dante™/ AES67 interfaces. They can accept and provide SMPTE ST 2110-30 streams using an external software application.
- 128-channel AoIP Ravenna/ AES67 interfaces. They can accept and provide SMPTE ST 2110-30 and SMPTE ST 2110-31 streams using NMOS control protocol.
- Audio over IP inputs/outputs, compressed G.722 for use in Intercom applications. 12 channels per card.
- 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.
- Additional talkback and auto-control microphone inputs as well as headphone output integrated with each control module through
- Dante AoIP connectivity with the audio engine.
- Microphone and analogue line inputs and headphone and analogue line outputs in AoIP Dante interfaces to connect studios to the core.
- Control Integrated in the Atrium surface.
- CUE output with integrated speaker in the control surface.
- Independent control-room and studio monitor management for each control surface. Primary and secondary headphones outputs for control-room and studio. Physical output for control headphone in the control and monitoring module.
- USB connection on the control surface.
  Test tone generators with adjustable frequency and level, 400Hz tone burst mode, pink noise and white noise.
- GPI and GPO (general purpose inputs and outputs for special applications): Opto-coupled GPI and relay-based GPO. Virtual GPIO transport between different equipment and system devices.

USER LOGIC

- Different levels of user access (up to 31) with associated passwords. By default, three levels are created: administrator, operator and basic.
- Cough-mute, ON-AIR studio and control-room signalling, fader-start, remote PFL, talkback, automatic monitor-muting. Monitoring can be configured for all the signals available in the system.
- Direct key routing for each fader channel.
- Any signal available in the system can be assigned to any control or fader channel.
- Mono, stereo and 5.1 multichannel signal management.
- Handling groups of signals via a single fader.
- Flexible, virtually unlimited internal MPX or N-1 bus configuration.
- 3 physical, precision stereo Vu-Meters, plus one +/-12dB presence vu-meter per fader. Up to 128 IP-transmitted vu meters on the displays of the console, as well as on remote PCs using a software application.
- Linear phase-meter.
- Optional module including spatial phase-meter and EBU R-128 loudness measurement.
- Externally synchronized clock, timer and stop watch.
- External equipment control (for AEQ audiocodecs and phone hybrids) integrated in the programmable keys section.
- The selected sources' level measurement can be sent via IP for Visual Radio systems.
- Control communications using Ethernet 10/100/1000 ports and TCP/IP protocol.
- Non-volatile RAM storage of all signals adjustments. Up to 128 available memory positions.
- Processing can be applied to of all audio signals available in the system.
- All audio processes are pre-defined but can always be manually adjusted in real-time and stored in the system's non-volatile memory.
- Types of implemented audio processes:
  - 4-band parametric equalization.
  - Low pass, high pass and band pass filtering.
  - Compressor, limiter, expander, noise gate and mixed DSP processor.
  - Delay.
  - Reverb.
  - Combined process with De-Esser function.

DIMENSIONS AND WEIGHT

Physical Module Specifications:

6 faders module: 261mm W* x 431mm L*. 3,90Kg (8,60lbs).
Control module: 261mm W* x 431mm L*. 4,45Kg (9,81lbs).
Depth from the top of the desk: 83mm.
Screen module: 261mm W* x 126 mm L x 150 mm H. 1,75Kg (3,85lbs).

Configuration examples (without optional screen module):

12 faders : 783mm W* x 431mm L*.
24 faders : 1.305mm W* x 431mm L*.
36 faders : 1.827mm W* x 431mm L*.
*These measurements do not include the side trims, 10mm at each side.

Non-surface specifications (width x height x depth; weight)

X_Core: Engine 4u x 19" (482,6 x 177,8 x 450,0 mm); from 12 to 22kg, 26.4 a 48.4 lbs.
XC95 power supply unit: 1u x 19" (482,6 x 44,5 x 450,0 mm); 3,2 kg. 7 lbs.
XC90 power supply unit : 2u x 19" (482,6 x 89,0 x 340,0 mm); 8,0 kg. 17,7 lbs.
XC93 power supply unit : 2u x 19" (482,6 x 89,0 x 340,0 mm); 8,2kg. 18,1 lbs.
XC96 forced cooling unit: 1u x 19" (482,6 x 44,5 x 430,0 mm); 1,8 kg. 4 lbs.
Netbox 32 AD AoIP Interface: 1u x 19" (482,6 x 44,5 x 361,0 mm); 4,5 kg. 9,9 lbs
Netbox 8 AD AoIP Interface: ½ u x 19" (211,0 x 44,5 x 300,0 mm); 1,8 kg. 4 lbs.
Netbox 4 MH AoIP Interface: ½ u x 19" (211,0 x 44,5 x 170,0 mm); 1,0 kg. 2,2 lbs.

FLEXIBLE

EASY TO USE

REMOTE SUPPORT

WORLDWIDE

PLUG & PLAY

INTUITIVE SOFTWARE

NATIVE IP

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