

IP-based
multi-channel
audio network



Audio over IP

 **Dante™ AES67**

Multi-channel audio connectivity in program production centers using simple Ethernet cabling.

Based upon IP technology and compatible in its native form with more than 1,000 products from over 350 manufacturers. Compatibility with other third-party products is ensured through the compliance with AEQS 67.

System overview

Using off-the-shelf routing equipment to send audio over IP in small to medium-sized systems offers cost advantages over synchronous solutions using AES-10 (MADI) or TDM buses. These have higher capacity but require powerful hardware.

Besides, the cost of large TDM systems can be reduced in cost while at the same time increasing their flexibility when they are combined with IP audio links to connect a few circuits with a central router.

That's why, when developing the IP audio routing system at AEQ, we have created not only IP connection devices for the digital consoles, but also connection panels that allow for audio input and output installation wherever it's necessary, as well as access cards for the AEQ BC2000D router.

The range of AoIP networked products from AEQ has been completed with NETBOX 32 AD MX and NETBOX DSP. These units are compact, TDM-based audio routers with DANTE connectivity and have been developed to offer capacities between 64 and 160 inputs and outputs (respectively) as well as audio mixing and processing capabilities.

The VENUS 3 IP Audiocodex has also been provided with AoIP networking capabilities, and the Systel IP 16 system completes the range of Broadcast Communication Products. This latter is a VoIP phone system for both on-air broadcasting of telephone calls and external technical intercom, also offering local AoIP connectivity.

We have added to this family a range of Intercom systems with broadcast quality audio and that benefit from AoIP Networking interoperability and simplicity of wiring and installation. We have also made our popular Olympia Commentator Systems compatible with these intercom systems.

AEQ insists on offering interoperability with third party devices for the convenience of our customers. Because of that, the AEQ AoIP solution is based on Dante technology, which is operating with extraordinary performance, making our systems 100% compatible with the majority of available equipment for Broadcasting, Recording Studios and Professional Audio (see full listing at www.audinate.com).

On the other hand, and in order to offer interoperability with the remainder of third party manufacturers (RAVENNA, WheatNet, Livewire+...) we have adopted the AES 67 "AES standard for audio applications of networks - High-performance streaming audio-over-IP interoperability".

Using Dante Domain Manager application, SMPTE ST 2110-30 protocol IP audio streams can be incorporated into our AoIP system based in DANTE-AES67 format.

Also, an AoIP RAVENNA card has been created for the new X_CORE engine, in order to facilitate audio integration with systems based on RAVENNA protocol.

This same card will allow IP audio streams to be incorporated into our AoIP system with the SMPTE ST 2110-30 and SMPTE 2110-31 protocols in RAVENNA embedded in IP video streams controlled by NMOS.

General features

- The system topology is usually a star centered around Gigabit Ethernet switches.
- Where maximum availability is a design goal, the network can be wired with duplicate switches, ensuring uninterrupted operations with redundant paths.
- For smaller installations, the system can be cascaded or "daisy chained", as network interfaces are duplicated.
- Data transport between devices allows for the conversion of a GPI in one device into a GPO on another device within the network.

Technical Specifications

- Data format: Dante Audio over IP technology.
- AVB - ready. AES 67 compatible
- Plug-and-play technology - automatic detection of the hardware and simple audio routing.
- Precise sample-level synchronization, even through several switches.
- Very low and deterministic delay in the entire network.
- Flexible and scalable network topology, supporting a great number of audio transmitters and receivers.
- Works in 100 Mbps, 1 Gbps and 10 Gbps networks.
- Supports a single integrated network used for audio, video, control and monitoring. Compatible with other kinds of traffic using QoS management.
- Uses low-cost, "off the self" network infrastructure.
- 24-bit, 48 KHz. audio resolution.
- Delay: 1 - 2.5 ms (@ 48 KHz typical, depending on type of device, network performance and complexity).
- Available version that allows to operate over very long distance WAN structures with up to 170 ms of latency.
- 2 RJ45 Ethernet ports per interface, 1000 BASE-T, or 100 BASE T depending on the device, transformer isolated, that can be used for redundancy or daisy-chain connections.
- Binary rate: 10/100/1000 Mbps.
- Maximum segment length: 100m max. over CAT5e or better cabling.

Audio quality:

AEQ audio over IP implementation does not limit audio quality; it only produces a small delay around 1 to 2,5 ms, which is easily configurable as a function of the network performance. The AEQ AoIP solution provides the same optimum audio quality as our current consoles and digital routers.

Available number of AoIP (Dante™) channels offered by each device:

- TP 8116: 2
- TP 8416: 2.
- VENUS 3: 4.
- NETBOX 4 MH: 4.
- NETBOX 8 AD: 8.
- NETBOX 8 AD VX: 8
- AUDIO PLUS: 8.
- OLYMPIA 3: 8.
- CAPITOL IP: 16 or 32.
- FR 14 (FORUM IP): 32.
- NETBOX 32 AD: 32.
- NETBOX 32 AD VX: 32
- NETBOX 32 AD MX: 32 (1)
- SYSTEL IP 16: 32
- XC24 (ATRIUM - X_CORE -- CONEXIA): 64.
- CROSSNET: 64 to 160
- NETBOX DSP: 0 (2)

- (1) 32 mixable and routable Dante channels, 64 x 64 matrix size.
(2) 64 to 160 input mixable Dante channels, 64 can be processed and sent exclusively to the Dante network.



ATRIUM Console

XC24 Cards

XC24 cards are designed to be installed into the X_CORE frames used in AEQ ATRIUM digital mixing consoles. Each connects up to 64 output and 64 input channels via IP to/from the X_CORE internal TDM bus system. This is enough to provide full IP connectivity for an ATRIUM console.

An ATRIUM frame can be equipped with as many XC24 cards as needed, and they can be connected to one or several Gigabit Ethernet networks. Besides several AoIP XC24 cards, the ATRIUM X_CORE frame can be equipped with a number of XC34 RAVENNA, or XC12 MADI cards as required for routing.



FORUM IP & SPLIT Console

FR14 Card

FR 14 AoIP card is installed as any other input/output card at the rear panel of the FORUM IP console. It connects up to 32 input and 32 output channels to the unit's TDM BUS. The same console can be equipped with up to two FR 14 AoIP cards. It is not compatible with the MADI option.



CAPITOL IP & FORUM IP SPLIT LITE Consoles

In these audio mixers, the implementation of IP connectivity is based on a module incorporated in its audio engine, which is optional and doesn't support simultaneous MADI connectivity. There are two versions: One with 16 audio input channels and 16 output channels, and the other with 32 input channels and 32 output channels.

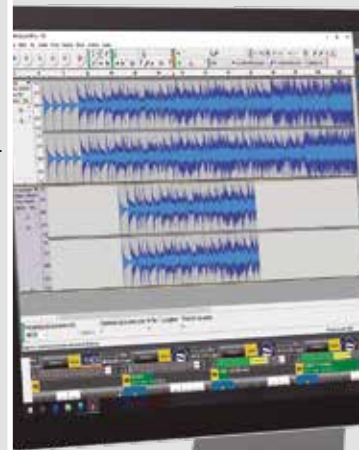


DANTE VIRTUAL sound card

Any computer with "Dante Virtual Soundcard" installed can receive and send channels from / to AEQ consoles and matrixes. This software can be downloaded, in trial and full versions at www.audinate.com. A basic implementation of the "Dante Virtual Soundcard" is to monitor different signals available on the network from a PC.

Broadcast Automation Suite AUDIOPLUS

AudioPlus is an automation system for broadcast production and audio content play-out for PC networks. AudioPlus has a "no-soundcard" option that links your audio inputs and outputs with Dante Virtual Soundcard. Thus, the AudioPlus can record up to 4 stereo studio sources from any device on the network and simultaneously play-out four stereo signals onto the DANTE network that would be available for any connected device. PCs do not need to be equipped with a sound card.



DANTE CONTROLLER Control Software

The system is able to auto detect all DANTE-enabled equipment that can provide audio within the network. Using the "Dante Controller" application installed on one or more computers on the network, the user can choose among the available audio channels which should be received from the different consoles or interfaces. The application is very easy to operate.

Compatibility with other manufacturers is absolute. DANTE CONTROLLER software makes the different IP access cards work together, no matter which manufacturer provides the equipment they are installed in.



DANTE DOMAIN MANAGER Administration Software

It is a network administration software that incorporates user and group management, auditing for Dante networks, as well as facilitating the expansion of Dante systems on any network infrastructure. It also manages the exchange of audio channels between Dante and audio and video networks over IP using SMPTE 2110-30 protocol.



We present five different interfaces for multi-channel AoIP under the common name NETBOX. The interfaces allow for Audio and GPIOs Input and Output system connectivity at locations where the installation of AEQ digital consoles is not planned.



NETBOX 32 AD

Features 32 input and 32 output channels for the Audio over IP network, organized in 16 mono analogue + 8 stereo digital channels. The stereo digital audio channels can be configured to follow AES/EBU or SPDIF standards. It also incorporates 16 GPI and 16 GPO. It is especially suitable for central controls and links rooms and also to increase or distribute the capacity of TDM BUS matrixes such as AEQ BC 2000D, Crossnet or Netbox DSP.



NETBOX 32 AD VX

Special version of the Netbox 32 AD interface, operating as a multichannel audio level detector for camera automation when integrated with Broadcast Pix Visual Radio system. It reads the audio level present at each of its 16 analogue, 16 digital and 32 Dante inputs, and sends the values to the Visual Radio system through the control IP network.



NETBOX 8 AD

Features 8 inputs and 8 outputs for the Audio over IP network, organized in 4 mono analogue and 2 digital stereo channels. Stereo digital ones can be configured as AES/EBU or SPDIF standards. The second digital stereo channel can also be switched to a USB connector to ease the connection to an audio workstation. It also provides 4 GPI and 4 GPO. It can be useful to give IP access to analogue or digital consoles that are not ready for this type of connectivity from factory, for recording rooms, talk-rooms or any other auxiliary location.



NETBOX 8 AD VX

Special version of the Netbox 8 AD interface, operating as a multichannel audio level detector for camera automation when integrated with Broadcast Pix Visual Radio system. It reads the audio level present at each of its 4 analogue, 4 digital and 8 Dante inputs, and sends the values to the Visual Radio system through the control IP network.



NETBOX 4 MH

Connects the Audio over IP network to 4 microphone or line level input channels and 4 stereo headphone + line outputs. It is useful to provide IP access to microphones, headphones and analogue lines in radio and TV studios. It features 4 general purpose inputs and outputs (GPIO) that can be transported as signaling between devices. It additionally includes some GPIO for Studiobox signaling box, providing remote cue, cough-cut, 5 user keys, ON AIR red light and READY green light. It can be powered over Ethernet (PoE) or an external AC/DC power adapter (included).



STUDIOBOX

Studio signaling and remote control desktop panel. This unit provides ON-AIR Signaling and remote control of a Digital Mixing Console through a NETBOX 4 MH. Studio signaling box with "Ready" and "On Air" lights, cough-cut, remote-PFL and user-configurable keys.



NETBOX 32 AD MX

64 x 64 circuits audio mixing and distributing matrix. It can mix combinations of its 16 analogue, 16 digital and 32 Dante inputs (each with its own, independent, relative level) over any of its 64 outputs (16 analogue, 16 digital and 32 Dante IP), according to the programming made using Netbox RTC software application. Digital audio channels can be configured as AES / EBU or SPDIF. It also features 16 GPI and 16 GPO.



NETBOX DSP 64, 96, 128 and 160

Audio mixing, processing and distributing matrix. Versions with 64, 96, 128 and 160 Dante input/output channels are available. It can mix combinations of Dante inputs (each with its own, independent, relative level) over any of its up to 160 Dante outputs, as programmed using the Netbox RTC software application. 64 inputs can be processed and returned to an output, or they can be summed to any other existing output. These devices also offer 16 GPI and GPO. In order to obtain analogue or digital audio connectivity, this system has to be combined with audio interfaces or consoles with Dante connectivity since all its inputs and outputs are Dante.



X_CORE - CONEXIA

Audio and intercom matrix. The same hardware can be used to build X_CORE - TITAN audio matrix with up to 5120 x 5120 circuits or a CONEXIA hybrid matrix with up to 1024 x 1024 broadcast and intercom circuits. Both configurations have full, non-locking and –optionally- redundant circuit connectivity.

Inputs and outputs are provided through different types of audio cards that can be installed in flexible quantities: AES / EBU digital, analogue line, microphone and headphone interfaces, 4-wire digital cards for Intercom panels, dark-fiber modules for long distance links in 64-channel MADI format and proprietary fiber optics link modules transporting more than 1000 channels, among others.

When operating as an intercom matrix, X_CORE control card is operated by means of the external TM 8000 / CONEXIA controller system. As an intercom, this system is compatible with all KROMA and AEQ wired and wireless terminals, and expands the interconnecting capabilities through interface cards or AEQ audiocoders and IP phone systems.

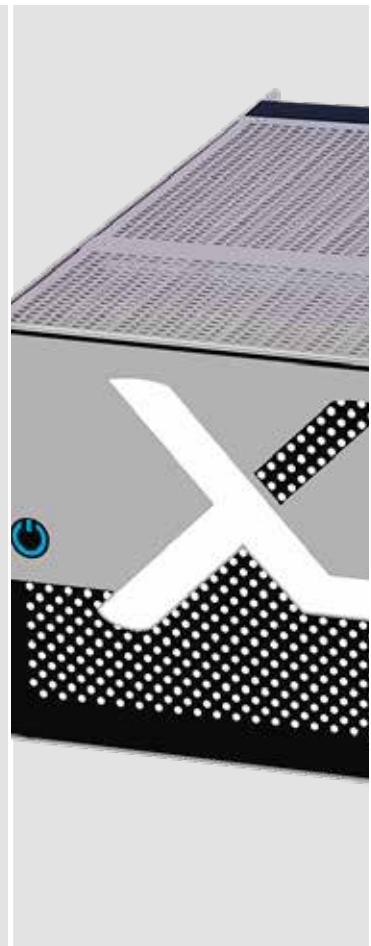
XC24 Card

Besides, by installing XC24 64ch AoIP cards, IP inputs and outputs can be included in the matrix from mixing consoles, IP interfaces, audiocoders, intercom user panels, commentary units and other IP devices compatible with Dante or AES67 protocols.

In order to build a large sized router, a XC_CORE frame can be equipped with as many XC24 cards as needed; they can be connected to one or several Gigabit Ethernet networks, enabling TDM-IP"hybrid" routers with great flexibility

XC34 Card

With 128 inputs and 128 audio outputs per IP in RAVENNA protocol, it facilitates the connection with AoIP systems based on this protocol. It also allows incorporating IP audio streams into our AoIP Dante system with the SMPTE ST 2110-30 and SMPTE ST 2110-31 protocols in RAVENNA format, associated to IP video streams NMOS controlled.



CrossNET

Compact intercom matrix with AoIP multi-channel connectivity. This high-performance digital intercom matrix provides broadcast-quality audio. Provided in versions from 40 up to 200 x 200 ports, 160 of them can feature AoIP Dante technology, compatible with AES67 standard, which can easily be connected using existing Ethernet networks through conventional routers and switches. The system also has 12 analogue ports, 8 digital ports, and 20 AoIP ports with KROMA Standard, enabling the connection of wireless intercoms, ancillary equipment and KROMA series 3000, 4000 and 5000 panels.





USER PANELS TP8000

Broadcast-quality intercom user panels. Audio is digitized and processed using 24 bits at a sampling frequency of 48 kHz. Audio bandwidth from 20Hz to 20kHz with negligible distortion and noise levels. The panels feature ports for Analogue, Kroma-Digital, IP Kroma and high-quality -IP Dante connectivity. Digital audio processing: acoustic echo canceling, automatic power and tone adjustment to each user's voice. Expander and ambient noise gate. Very optimized acoustics for optimal sound intelligibility and clarity. 16 keys, rack or desktop format. Expansion panels can be cascaded to build panels with up to 80 keys with 4 pages. Compatible with any AEQ and KROMA intercom matrix.



COMMENTARY SYSTEM OLYMPIA 3

Olympia 3 CU is a high-performance Commentary Unit that at the same time can operate as an 8-key intercom user panel within a Conexia or Crossnet Intercom System. As a Commentary Unit it offers 8 Dante™ AoIP inputs and outputs, PoE and local power supply, IP video transport, mono or stereo local audio mixing and processing for 3 commentators and 2 local sources. Configuration and Operational control software is included.



AUDIO CODEC VENUS 3

Dual audio codec for IP transmission, with local audio over IP Dante connectivity. It is able to establish two simultaneous stereo, bi-directional communications in different formats and quality. Remote control software is multi-workstation with multi-device option. AoIP, analogue and digital inputs and outputs. Several GPIO, double continuous ancillary data channel and redundant power supply option (AC 100-240V, DC 48V or mixed). As with all Phoenix AudioCoders from AEQ the VENUS 3 can be seamlessly integrated with AEQ's intercom systems to provide external communications.



VoIP SYSTEMS SYSTEL IP 16 / SYSTEL IP BASIC

Voice over IP (VoIP) system for multi-conference and coordination with IP phone lines and local AoIP Dante connectivity. It supports up to 4 control IP phones. Local audio: 32 Dante, 4 digital and 2 analogue inputs / outputs. 12 GPI and 12 GPO. Enough to cover 4 radio studios or a multi-set TV coordination system. As the AEQ Phoenix Audiocoders, the SYSTEL IP can be fully integrated with AEQ's Intercom systems to provide connectivity for external communications, such as phone communications or remote Audiocoders. The Systel IP VoIP System includes software for configuration and operation for an unlimited number of user terminals.

Systel IP 16 supports 16 voice lines per IP. Systel IP Basic supports 8 lines. By activating a code, Systel IP Basic becomes Systel IP 16 without modifying wiring.

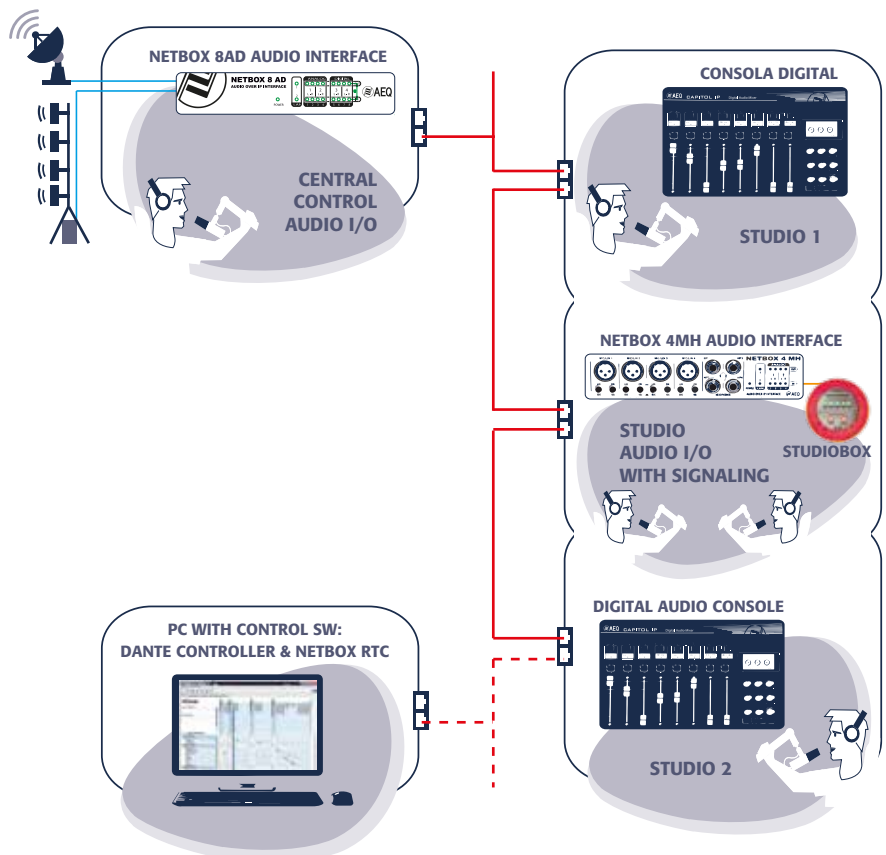
AEQ Audio over IP system for 2 digital radio studios and a central control

This drawing represents a proposed installation for a small, two-studio radio station. The daisy chain IP wiring is represented in red, running from one PC to the audio mixing console in a studio, then to the audio interface for the shared studio, passing to mixing console of the second studio and then ending at the audio interface in the central control or rack-room.

Program audio for both studios as well as other required signals for the central control (such as clean-feeds for telephone systems, etc) are sourced from the NETBOX 8 AD audio outputs.

The signals necessary for the studios, such as satellite downlink, audiocodecs, tuners, etc. are routed to the NETBOX inputs. Each console will also receive not only the NETBOX 8 AD incoming audio but also the aux and program sends from the other console. Both controls will share Studio and a NETBOX 4 MH will be installed into the booth to provide connectivity for the common microphones to consoles and their headphone outputs.

ON-AIR signaling and the programmable keys of the Studio remote panel will also be using the system network.



Drawing 1

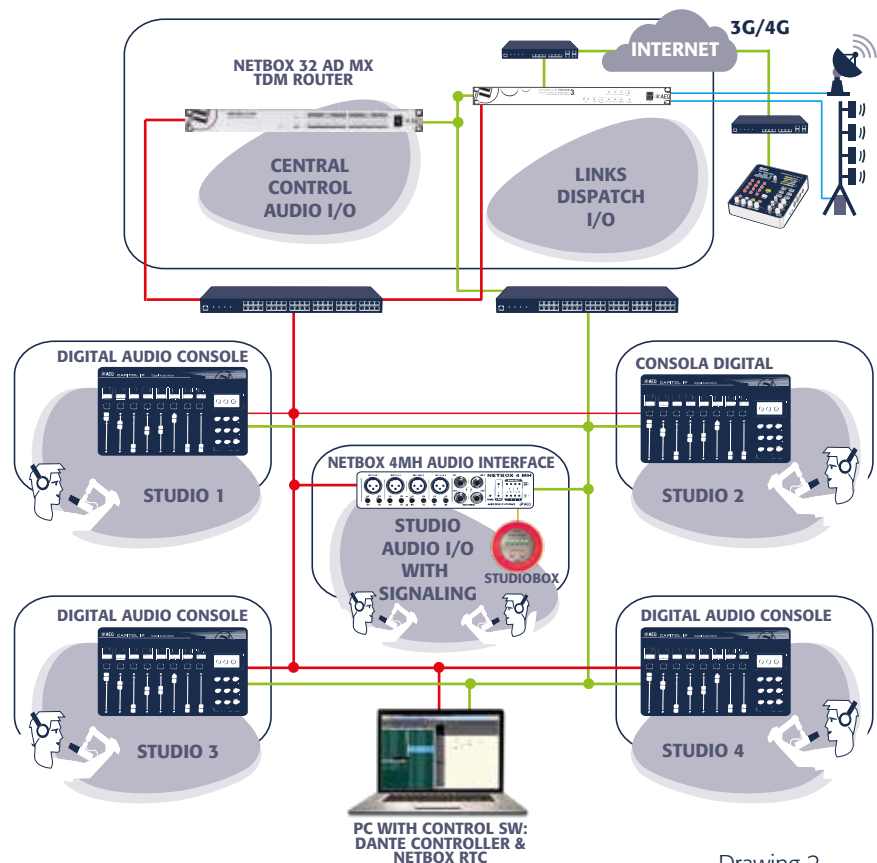
AoIP system for several digital radio studios and a central control based on Netbox 32 AD MX TDM matrix.

This schematic represents the installation in a radio station with 4 studios.

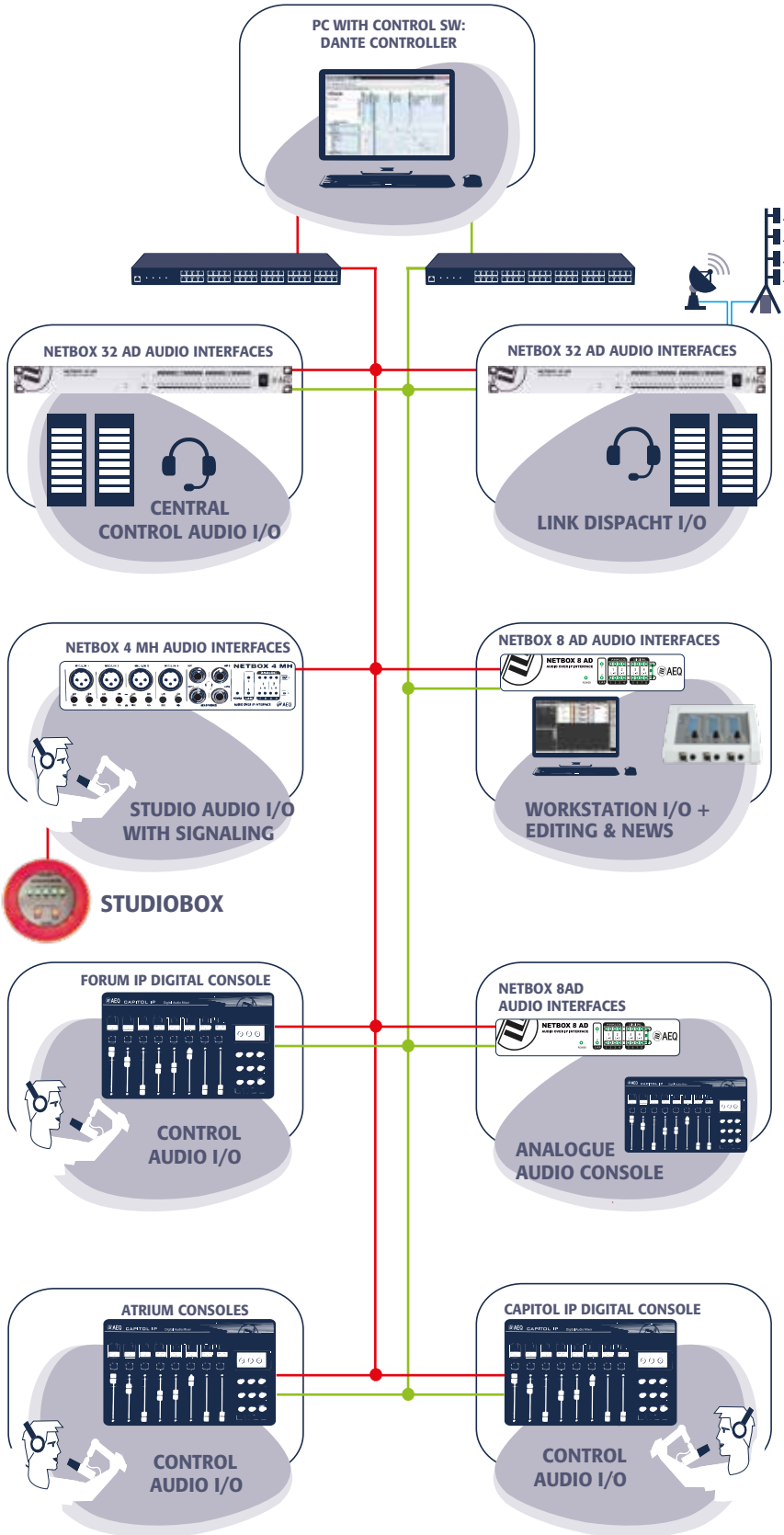
The primary IP network, connected to a switch, is represented in red, connecting the NETBOX 32 AD MX audio matrix, including 8 dual AES/EBU channels, 16 analogue ones for the central control and 32 AoIP Dante™ for the studios and links room. There is a NETBOX 32 AD AoIP interface in this links room in order to connect to the devices installed there.

Next, the 4 studio mixing consoles are represented, together with the NETBOX 4 AD installed in a shared booth. The PC (or PCs) for audio routing control is depicted in the lower area.

The secondary IP network is represented in green. This network is required when utmost reliability is a must, and is connected to a different switch. If the control PC needs to be connected to this network, a second network card is required.



Drawing 2



Audio over IP system without a TDM matrix for medium to large sized stations

The main IP network, wired around one or several IP switches, is represented in red. The secondary network, centered around a second IP switch, is depicted in green. All devices in "AEQ Audio Over IP Routing System" feature two LAN interfaces to be able to connect this secondary network.

The schematic shows 6 different AoIP interfaces in the system:

- NETBOX 32 AD Interface.
- NETBOX 8 AD Interface.
- NETBOX 4 MH Interface.
- CAPITOL IP console.
- FORUM IP console.
- ATRIUM console.

The program audio for all studios, as well as other required signals for the central control (such as clean-feed auxiliary sends for telephone systems, etc) are sourced from the NETBOX 32 AD audio outputs. The signals necessary for the studios, such as satellite downlink, audio codecs, tuners or TV receivers, etc. are routed to the NETBOX inputs.

One or several NETBOX 32 AD units can also be installed in the links room. Signals going to or coming from radio links, off-air receivers and satellite uplinks, for instance, can be connected to this device and thus becoming part of the AoIP system/network.

A NETBOX 8 AD can be installed in News recording cabins or editing rooms, providing audio input and output for the audio workstations through a bi-directional USB link. Audio can also be provided to the mixing console using analogue and digital I/O connections. The same NETBOX 8 AD unit will provide IP connectivity to analogue or digital studios without AEQ IP connectivity. This way, a station can be IP-connected without having to abandon existing equipment.

AEQ CAPITOL IP, FORUM IP and ATRIUM digital consoles can be provided with the corresponding multi-channel 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 links to the audio inputs of the interface.

In each Studio, one or several NETBOX 4 MH will be installed to make all the microphones available to the mixing consoles of the AoIP Network and in order to send the necessary headphone outputs that the console is providing. Further, the STUDIOBOX Remote Signaling desktop panel is connected, providing the required Studio Signaling and remote control through its assignable keys.

This solution, while quite cost-effective, doesn't allow for dynamic audio routing changes between the different studios, central control and link rooms, as there is no TDM matrix included.

Drawing 3

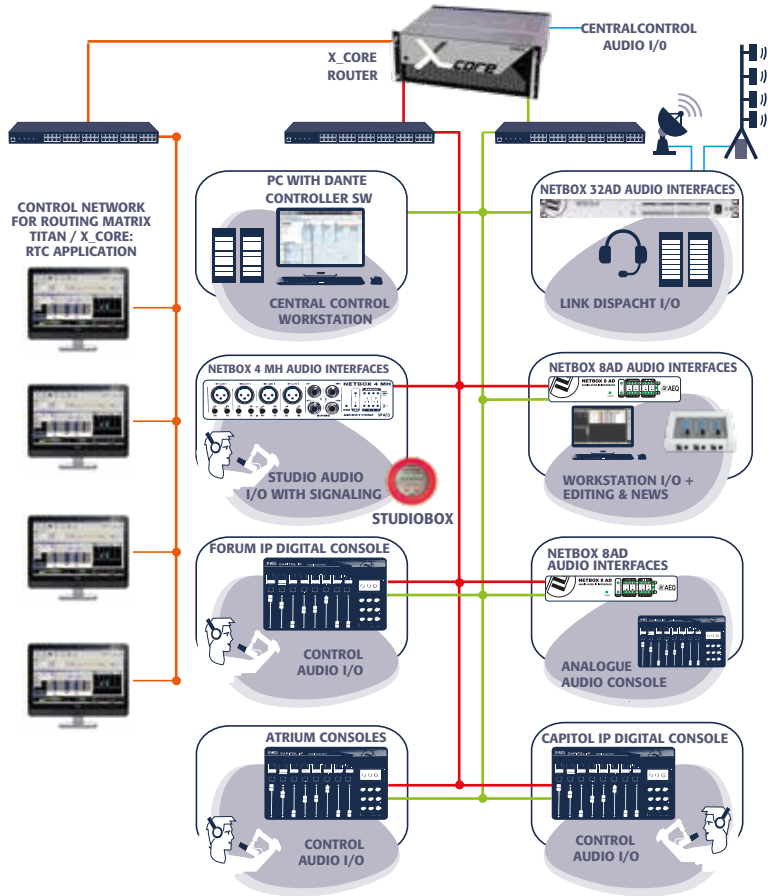
AoIP System for large station based on X_CORE TDM MATRIX

Please note that there are new elements in this schematic that were not present in the former one.

At the top, the main audio matrix where all the Dante channels are routed to through a bank of XC24cards installed in the matrix frame. A set of inputs and outputs from the matrix are connected to the equipment of the central control or rack-room itself through the analog & digital cards.

On the left you can find the control network with workstations running layouts of the Real Time Control Application for the matrix for the distributed and hierarchical control from studios and MCR.

Thanks to this, we benefit from the performance of the TDM system (making live or scheduled routing changes possible), distributed control and processing of the crossing points, audio mixing, alarms, macros and salvos executed manually or automatically, VU meters, MADI connectivity and E1 / T1 ...), and the ease of installation and flexibility of an AoIP system.



Drawing 4

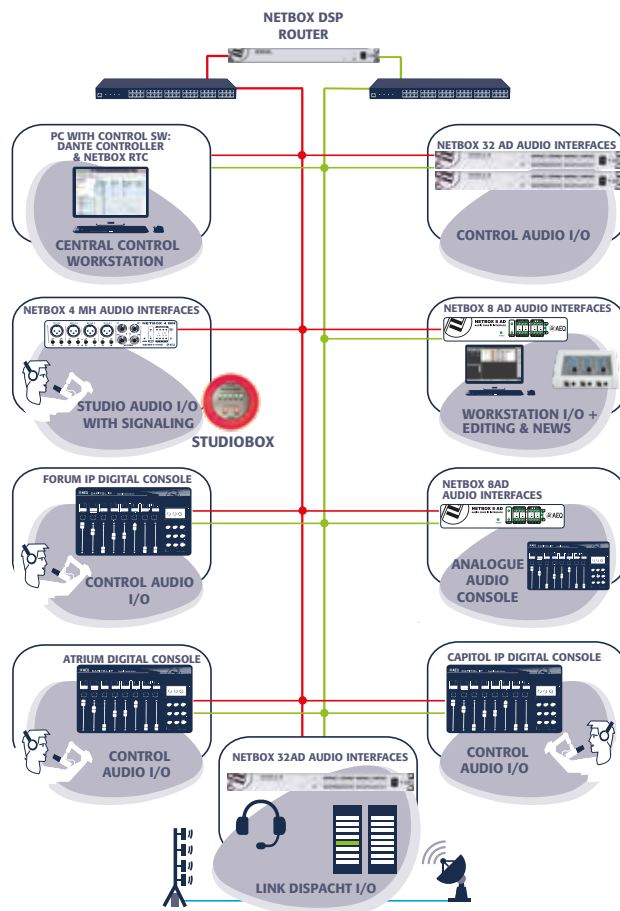
AoIP system for a mid-sized radio station, based on Netbox DSP TDM MATRIX

Please note that there are even more new elements in this schematic that were not present in the former ones.

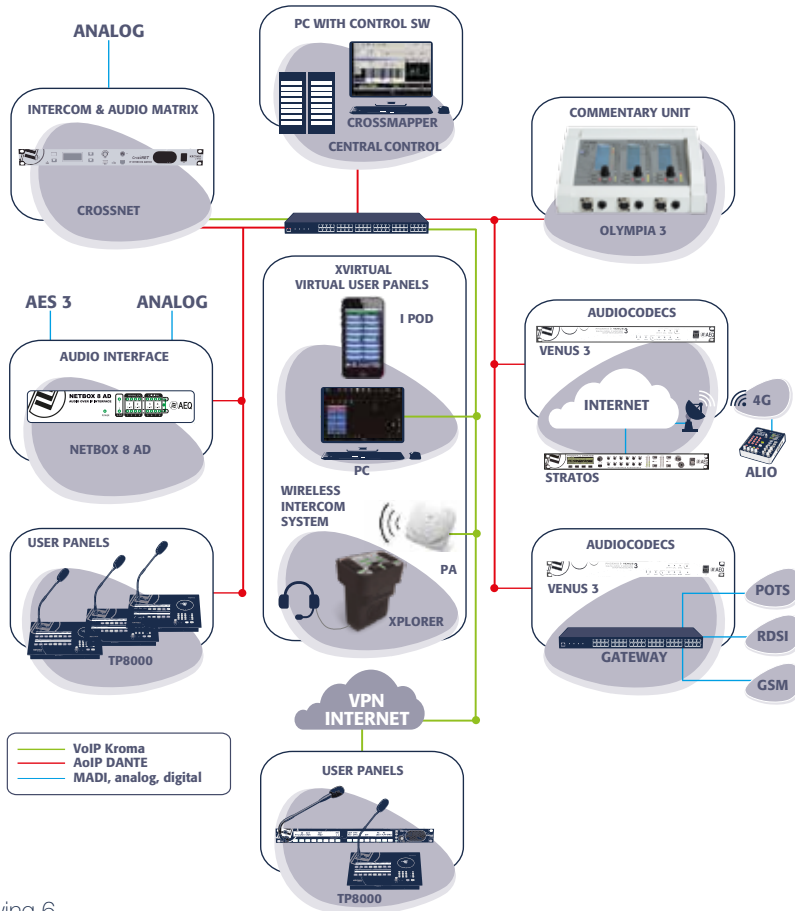
At the top a Netbox DSP TDM matrix is included instead of the X_CORE. It receives audio from the IP network, processes it and returns it mixed and processed to the same network. Its maximum capability is 160 inputs and 160 outputs. Hence, functionality is the same as the prior former schematic, taking into account the following considerations:

As there are no analogue input and output cards in this matrix, NETBOX 32 interfaces needs to be installed in order to provide service for such inputs and outputs in the central control /rack-room and links dispatch.

Its dimension is more reduced, so it can provide service to a smaller number of studios, typically between 4 and 8.



Drawing 5



Drawing 6

Crossnet router - based IP intercom system

CROSSNET includes two Ethernet connectors providing redundancy for the Dante AoIP, the standard KROMA VoIP and the control networks.

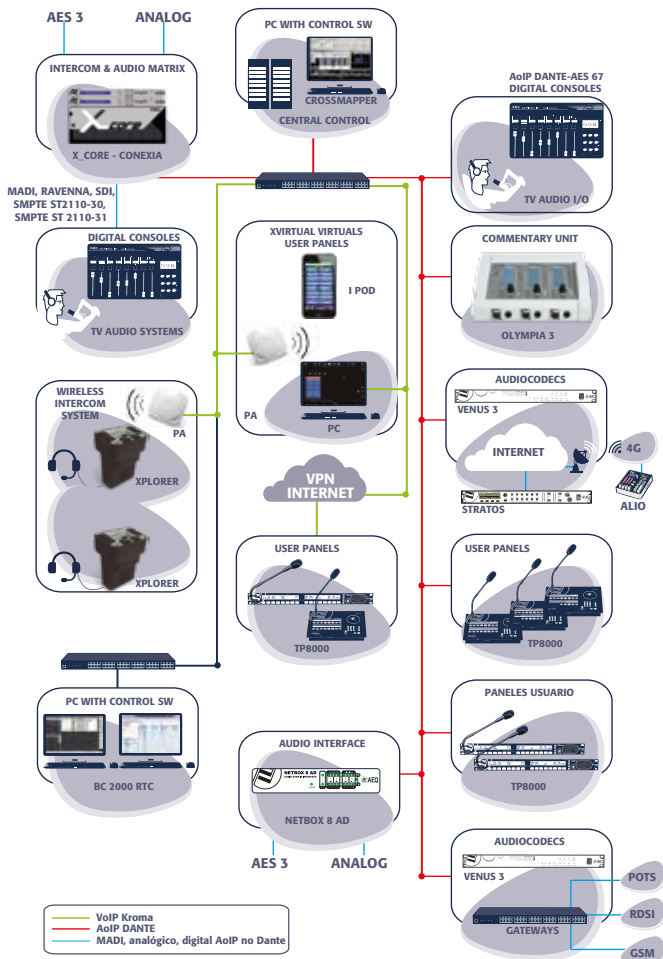
The first one, represented in red, is the high-quality audio-over-IP network for broadcast-quality communications. The following items have been included in the diagram:

- Olympia 3 Commentary Unit / Intercom user panel.
- Venus 3 IP audio codecs with local AoIP connectivity, that also enables connections with Alio portable codecs using 4G networks.
- IP audio codecs with local AoIP connectivity, to provide access to several types of phone networks by means of gateways.
- TP8116 AoIP Intercom User panels.
- Netbox AoIP audio interfaces, to insert and extract high-quality analogue and digital audio signals to/from the Audio-over-IP Network.

The VoIP network, depicted in green, provides access to:

- An Xplorer Intercom system which incorporates Xplorer wireless beltacks and Xvirtual app-based iPod or iPad devices to the system through WiFi access points. Also, PC terminals running the Xvirtual software application are represented, using wired connections.
- Remote TP8116 and TP8416 panels, using VoIP through a VPN.

Represented in Blue, this device also offers 12 connectors carrying high-quality analog audio ports.



Drawing 7

X_CORE - CONEXIA router – based IP audio and intercom system

X_CORE - CONEXIA is an audio and intercom router, providing I/O and port capacities at much higher levels than the CROSSNET. CONEXIA comprises a controller (or two, for optional redundancy) and a frame or set of frames with Audio I/O cards. The CONEXIA is an intercom system that can reach 1024 x 1024 cross-points. Dante AoIP Network I/O Cards can be installed as per requirement and to provide service for the AoIP Networked intercom panels and the different audio interfaces distributed throughout the station's different Studios controls. The system can also interconnect to Dante AoIP-compatible audio mixing as well as to provide connection through dedicated VoIP cards to wireless intercoms or other VoIP links / systems, such as remote panels or matrixes.

The diagram is conceptually similar to the former one, but with the following differences:

The audio inputs and outputs of other equipment or audio systems for TV, represented in blue, such as:

- Digital consoles with IP connection RAVENNA-AES 67 or synchronous MADI.
- IP audio and video systems with SMPTE ST 2110-30 or SMPTE 2110-31 protocols.
- SDI video systems with embedded audio.

They have been additionally connected to the AoIP Dante network shown in red.

- A control network has been deployed to enable computers in the different studios to provide real-time control of the broadcast audio routes.

IP connected sports system

Next, the description of part of a system used in a real sports event is provided.

A system with more than 70 IP commentary positions has been designed and deployed at multiple venues separated by several tens of km. The subsystem comprising 12 sports venues - equipped with OLYMPIA3 commentary units, the audio router in the IBC (International Broadcasting Center), the intercom system and transport infrastructure to the affiliated broadcasters (all this using Audio over IP) is described.

The trunk infrastructure consists on an Audio Routing Matrix X_CORE with capacity for 512 x 512 channels. Monitoring and real-time operation is performed using the X_CORE RTC control software application, while the static routing infrastructure has been established with Dante Controller. This logical architecture using several layers of superimposed routing, together with the use of the AEQ Audiocodexes Venus 3 with Dante connectivity, has allowed the system to be flexibly expanded in real time, establishing additional routes, even international, allowing to cover the needs of circuits shortly before the start of the event.

Audio transport between all the venues and centralization of the IBC has been deployed over an AoIP network with Dante's redundancy logic.

The option of a special controller that supports delay greater than 5 ms was used to suit long distance networks.

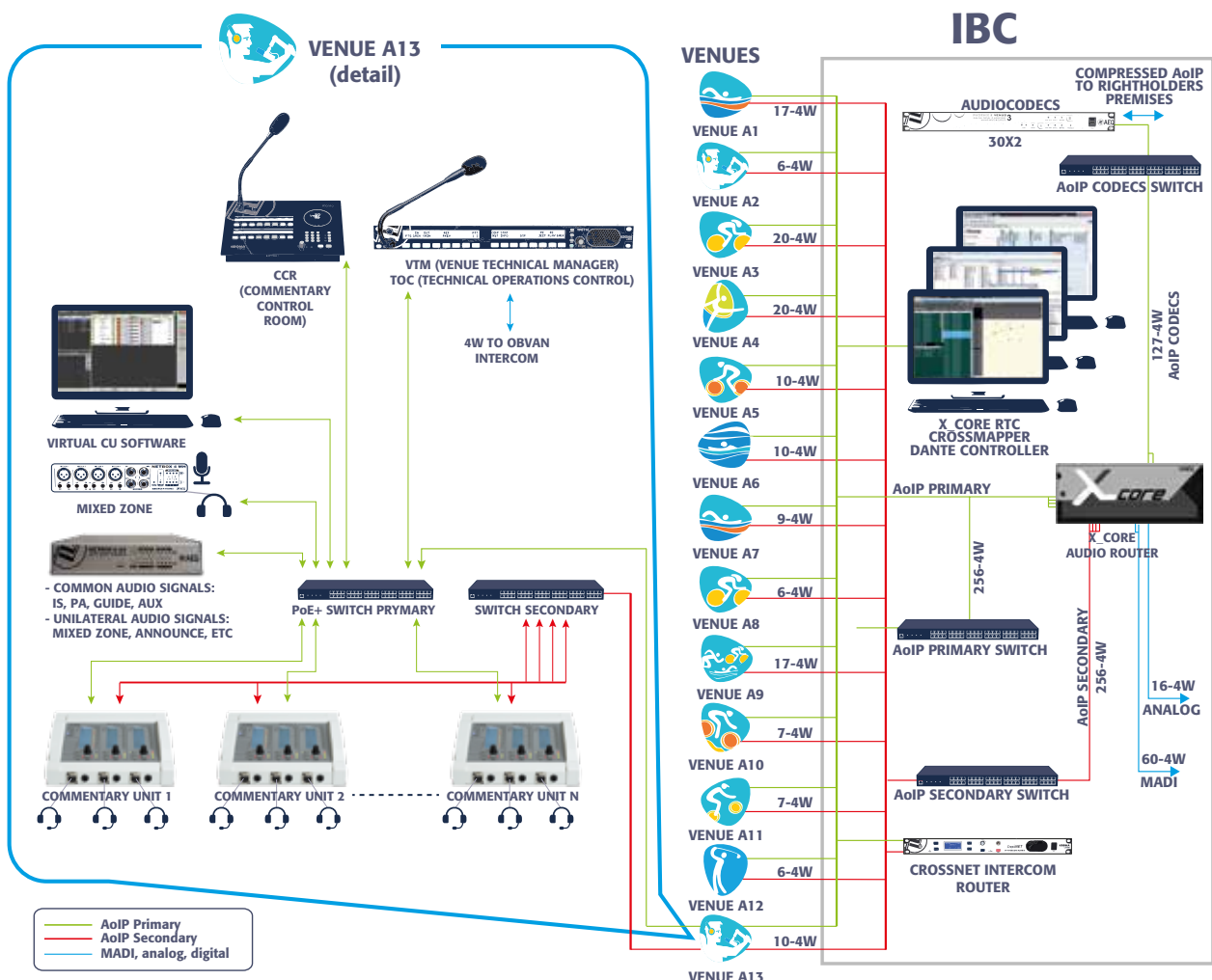
For technical coordination and internal production of the event, a 104x104 CROSSNET IP Intercom system was installed, deployed at the different venues, the IBC and other locations. This Intercom system has been extended using Netbox 4MH AoIP interfaces and Olympia 3 Commentary Units.

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

The commentator system at each venue, depicted at the left, was 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. Control was performed using the control application for the OLYMPIA 3 Commentary Units.

For mixed zones where journalists interview athletes, AEQ NETBOX 4 MH AoIP interfaces were 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>.



Drawing 8

The largest family of AoIP solutions for broadcast is on AEQ



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