Lawo launches mc²96 Console

Lawo has launched its new flagship audio mixing console, the fully IP-based mc²96 Grand Production Console at NAB 2017. The new console has been specifically designed to provide optimal performance in IP video production environments through native support for all relevant standards — SMPTE 2110, AES67, RAVENNA and DANTE. The Lawo mc²96 console, available in frame sizes with 24 to 200 faders with the same quality Lawo’s mc²96 series was known for, is designed as Lawo’s most visual broadcast console ever. The mc²96 features 21.5” full HD touch-screens as well as multiple mini colour-TFTs in channel strips and touch-sensitive color-coded encoders, providing fast overview and user-friendly operation.

In a world’s 1st Lawo’s new production console features “LiveView” video thumbnails. In addition to standard channel labeling via channel numbers, individual text labels and static pictures or icons, the mc²96 can display real-time video thumbnails right at the fader’s mini TFTs for intuitive channel identification. Touching a fader changes the LiveView thumbnail to full-screen mode, providing a more detailed view of that channel’s video source, such as a camera or a replay machine. Check our May 21st Facebook post for a video overview of the TTT functionality.

Designed for networking in complex IP production infrastructures, the Lawo mc²96 employs Lawo’s unique IP-Share Network Gain Compensation when using shared DALLIS I/Os to prevent unexpected changes on up to eight networked consoles when individual users adjust their gain settings. The DALLIS I/O communicates with all networked consoles, and its IP-Share algorithm sets the optimum analog gain for multi-client requirements. IP-Share also ensures that the corresponding gain compensation is applied to the digital gain stages of all client requirements. IP-Share also ensures that the corresponding gain compensation is applied to the digital gain stages of all client requirements. IP-Share also ensures that the corresponding gain compensation is applied to the digital gain stages of all client requirements.

Lawo recently announced a new collaboration with computer networking market-leader Cisco, in the development and marketing of IP-based video and audio systems for broadcast television. The cooperation involves the Cisco IP Fabric for Media infrastructure, its answer to complex IP workflows in TV broadcast, based on Software Defined Networking (SDN) principles. As the industry moves rapidly toward Broadcast 3.0 (broadcast infrastructure built around COTS technology), broadcasters everywhere are excited about the ability to produce more content on format-agnostic IP platforms, finally delivering on the promise of “future-proof” production technology.

The first strategic customer to take advantage of this work is in place at a large US TV network, which will deploy an all-IP Lawo routing infrastructure based on a Cisco IP technology backbone. The planned rollout, scheduled for later this year, will make use of Cisco Nexus 9000 switches connecting Lawo V__V_matrix, with SDUL and VSM control and monitoring systems.

Stage Tec double NEXUS audio network capacity

With software release V5.28 NEXUS systems can now be upgraded to more than double the previous size. Until now, the number of Base Devices was capped at 31. The audio network can manage up to 64,000 I/Os. Many NEXUS systems are already being deployed to connect entire complexes, such as broadcast facilities and 2017 04 StageTec NEXUS film production studios,’ explains Alexander Nemes, Sales Manager at Stage Tec, adding, ‘Scalability has been doubled yet again — sufficient for any currently conceivable network.’

By combining an enhanced number of Base Devices with the recently increased capacity of NEXUS fibre-optic lines, it has become possible to upgrade new and existing systems to create truly large-scale networks. The fibre-optic cables joining the base devices cover great distances, turning the network into the link between rooms, buildings and in the future, even cities.

NEXUS works with an internal bus system based on time division multiplexing and provides a wide range of I/O options, including IP technology. The system’s functionality goes far beyond that of a pure audio router. It handles control tasks, switches external signals transparently through the network, performs DSP functions and offers many extra features. Since its inception, Stage Tec has been continuously developing the system. Even the oldest legacy systems can be upgraded, and today’s NEXUS is fully backwards compatible.

In April 2017 Stage Tec announced a close collaboration with digital audio connectivity and networking audio manufacturer DirectOut, in providing audio-over-IP solutions. The collaboration is based upon DirectOut’s powerful and highly adaptive AES67 hardware platform, used in the popular MONTONE-42 MADI/RAVENNA bridge. The goal is to enable AES67 compatibility for the whole Stage Tec product range, based on the Digital Audio Routing System NEXUS.

www.stage tecn.de
The DiGiGrid MGO interface gives you the ability to connect any optical MADI-enabled device to Waves SoundGrid, for networking and processing. The DiGiGrid MGB interface lets you plug any coaxial MADI-enabled device in. Use hundreds of Waves and third-party plugins to record, process and playback up to 128 audio channels, with low latency of only 0.8 milliseconds. This compact and portable MADI interface even lets you record to two computers at the same time, so you can use one for virtual soundcheck, the other for backup. Whether you’re in a hotel room, on the tour bus or between flights, you can get to work setting up your next show — on your laptop.

www.digigrid.net

Lectrosonics SPNDNT

Wireless wizards Lectrosonics have a powerful range of multiple input/output mixers under the ASPEN imprint, with unlimited input expansion and 48 channel mix bus. The Proportional Gain Algorithm provides seamless auto-mixing and integrates easily with echo cancellation, or any other type of audio system. And with TCP/IP addressability for control through Ethernet, ASPEN can easily become a part of your network.

The Lectrosonics SPNDNT is a full function DSP mixer and processor like other ASPEN models — except that instead of analogue I/O ports — it uses digital network Dante ports. The processor combines Dante networked digital audio with the ASPEN digital matrix to expand signal routing options.

www.lectrosonics.com

Spotlight: Calrec V8 software boosts AoIP

V8 introduces a number of powerful developments across the Calrec product range, including improved RP1 support, switchable 96-kHz or 48-kHz sample rates, and further integration of Calrec's AoIP interface. 'V8 is really an evolution, a progressive suite of tools that supports broadcasters into their next phase of IP development,' said Dave Letson, Calrec’s vice president of sales.

‘With remote production and AoIP rapidly becoming a reality for many, V8 allows Calrec’s users to progress confidently into the next era. This, combined with higher-resolution audio, ensures that all bases are covered going forward: The new V8 software integrates Brio on the Hydra2 network, allowing the console’s on-board resources to be available to all other networked resources. Brio also receives some powerful new feature, including multiple sample rates, with options of 44.1, 48, 88.2, or 96 kHz — all without any DSP sacrifices. Brio’s processing capacity remains the same at any sample rate.

Calrec provides a variety of networking interfaces, including an AES67/Ravenna interface, an AVB interface, and a modular I/O Dante card that also has AES67 compatibility. Calrec customers can interface with multiple protocols at the same time. For example, a signal can be received via AES67 and then sent out via SMPTE 2022, AVB, Ravenna, Dante, or AES67. Signal-processing takes place via modular cards or 1U boxes.

Each element of Calrec’s protocol range redundantly connects to Hydra2 and appears like any other I/O resource on the Hydra2 network. Hydra2’s integral suite of management tools provides additional benefits to allow remote configuration patching, port protection, alias files, virtual patchbays, and access rights.

The AES67/Ravenna and AVB interfaces are a 1U box that can transport 256 channels of audio on a single connection. A second expansion card provides the unit with 512 channels of audio. The box is versatile and can accommodate one of each card, allowing simultaneous operation of multiple formats.

The SMPTE 2022 modular I/O card has a 10GB port that can receive up to four SD, HD, or 3G video streams, de-embedding 16 audio channels from each. Each video stream may be retransmitted intact or with new embedded audio. A secondary 10GB port provides for hitless switching, as defined by SMPTE 2022-7. Calrec’s Dante modular I/O card utilises Audinate’s Brooklyn II card and now offers AES67 support as standard. This combination allows access to up to 64 bidirectional channels in either protocol.

www.calrec.com

NTP Penta 725 IP & RCCore

NTP Audio Technology is based on the robust tried-and-tested Dante technology, and will interoperate with products from other brands that comply with Dante. The protocol will provide AVB support once standardization is fully completed. The Dante and NTP Audio formats provide fast, flexible, and economical audio routing via IP and are compatible with NTP PENTA 720 Modular I/O, NTP PENTA 725 IP Audio router, and
other Dante devices. The IP Audio routing provides low latency, synchronised transport of uncompressed audio over Gigabit IP Ethernet Layer 3 networks using off the shelf switches, and routers for audio routing via one or more sub-nets. A total of 512 channels can be routed on a 1 Gigabit network, and more if the network capacity is higher.

The IP Audio routing is for larger systems controlled by the NTP RCCore Router Control System, for easy and simple setting of connections on the Gigabit IP Ethernet. PENTA 721 can also be controlled by the Dante IP audio routing controller software.

RCCore is the control software that handles configuration, control and supervision of all router modules in an NTP audio signal distribution system. It also forms a communication interface to NTP’s VMC and BLISS control applications and to third-party communication protocols. RCCoreV3 expands the capabilities of NTP’s audio routers and features improved functionality, including remote update of software for equipment in operation.

RCCore is based on the rugged QNX Neutrino real-time operating system. The QNX operating system has very high reliability and is specially designed for critical industrial applications and superior embedded designs, ie flight industry. RCCore 3.0 runs on the on the NTP 625-151 controller cards and on the NTP 635 system controller - a high-performance industrial-grade PC with an integral solid-state server providing fast control and fast access to resources and control applications for large scale installations. NTP’s RCCoreV3 software operates through a network-based infrastructure with all communication based on TCP/IP link protocol and a 10 or 100 Mbit/s LAN/WAN interface.

www.ntp.dk

Merging Technology ANEMAN

ANEMAN is the first cross platform / cross vendor Audio NETwork MANager. Aneman is built on an open spec initiated by Merging and Digigram in 2014. The first UI is offered by Merging under the name Aneman but other manufacturers are being welcomed to develop UIs that can be specific to their own applications. To date, compatible manufacturers include Genelec, Archwave, Digigram and Merging. Aneman will be soon freely downloadable from the Merging Technologies website, currently the software is in “closed beta”.

Ravenna and AES67 require a network management and monitoring tool to enable their capabilities to be exploited. Aneman was developed to address this need, based on the premise that it should not be a closed tool dedicated to a particular ecosystem. Instead, Aneman is designed to be implemented in any device whose manufacturer requires an open, unrestricted AoIP ecosystem.

‘Ethernet networked audio — or AoIP — systems are increasingly replacing MADI-based systems because they offer many more channels, far more flexibility and are much more scalable,’ says Merging AoIP specialist, Nicolas Sturmel who recently joined the company from Digigram, and has been leading the project of the Audio Network Manager. ‘We therefore need tools that enable us to harness this potential and manage AoIP networks quickly and easily. Users don’t want to worry about IP addresses or complex system setups — they just want to patch audio.’

Merging also have an extremely useful Mac OS X program which can unlock the power of your RAVENNA or AES67 hardware from any Core Audio program. The Merging Technologies RAVENNA/ AES67 Standard Virtual Audio Device edition is free of charge and intended for owners of any RAVENNA and/or AES67 multi-cast device. Apple’s Core Audio is a standardized audio system for all Macintosh computers running on Mac OS X allowing access to all Core Audio compatible Audio Devices. Merging owners of a Hopus or Hapi Networked Converter get access to the additional features of the premium version.

www.merging.com

In March 2017 NTP Technology launched its new DAD DX32R Digital Audio Bridge, a companion product to the AX32, the DAD DX32 is a low-latency routing interface designed for applications such as multi-room recording studios, postproduction facilities and audio distribution installations.

The DX32R provides conversion between Dante/AES67, and not only MADI but also AES. It includes a coax MADI interface and eight AES3 interfaces, and can be expanded with a further two MADI interfaces, either optical or coax. A built-in router allows the user to pick and choose any channels from any interface to be routed to Dante/AES67 and vice versa.

Apart from providing routing of audio channels, the DX32R (with an optional Pro Mon 2 license) can also be utilised for summing/mixing of channels. Designed for mission-critical applications, the DX32R includes as standard a redundant Dante interface and redundant power supplies, plus two optional MADI interfaces, which can also operate in redundant mode. The DX32 base unit has an RRP of £2224.

Designed for use in no-compromise recording studios and post-production suites, the AX32 provides transparent audio signal processing and highly flexible distribution. Version 2.0 firmware includes an expanded switching matrix with 1,500 x 1,500 non-blocking cross points for routing and splitting any input to one or more outputs. Each input and output channel has its own level control and audio metering. All signal sources and destinations can be assigned a label which is then stored. The summing system has a capacity of 32 output buses and 256 inputs which can be used for Pro | Mon monitor control as well as the Cue | Mix zero latency cue mixing system.

It allows all digital input and output routing to be managed within a single unit or a combination of units. The AX32 has an RRP of £2895.

www.digitalaudio.dk
www.hhb.co.uk

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www.merging.com
Solid State Logic Tempest

SSL has moved to strengthen the future-proofing of its true networked AoIP broadcast infrastructure and the System T production environment with membership of AIMS — the Alliance for IP Media Solutions — in addition to existing memberships standards coalitions such as the Media Network Alliance (MNA) and the Audio Engineering Society (AES). SSL is growing its R&D team again to support the rapid on-going development of its expanding System T. The R&D team is almost entirely based at the company’s Oxfordshire HQ and has a complement of nearly 50 software and hardware engineers, as well as a dedicated software test group.

At this year’s NAB Show in Las Vegas, SSL demonstrated new features include support for hardware like Tempest Control Rack (p10 Resolution V16.3), and T25 (new Tempest Engine option, check Resolution Facebook video posted May 20th), as well as the new Network I/O SB 32.24 analogue/AES/Dante Stagebox, which incorporates onboard gain compensation splits compatible with any manufacturer’s console. System T (Resolution V15.6) is SSL’s networked audio production console designed to handle large-scale broadcast productions in a multi-platform workflow.

Other important System T software developments include a more extensive path live and mic-live feature set, access control levels and management for restricting features by operational level, 96kHz operation, extensive audio-follow-video controls update, full AES67 support for all Network I/O devices, provision for multiple loudness meter and True Peak meter instantiations in the FX Rack without compromising channel resources, and more.

A range of System T control interfaces can be connected anywhere on the network and incorporate multi-touch and gesture-driven screen technology and hardware control. Up to three consoles or control interfaces can control a single Tempest Audio Engine, while multiple engines can also be used on a single network.

Routing and I/O is based on the Dante AoIP network (integrating the AES67 transport standard) with complete routing control from the console GUI - featuring Dante HC (High Channel) connectivity.

SSL has already completed the first TV broadcast facility with full audio + video over IP workflow on the same network, complete with integrated VSM control and a video-over IP workflow using SMPTE ST 2022-6.

www2.solidstatelogic.com

Digico go 32 bit

The DiGiCo SD Rack offers multiple format sample rate conversion and allows up to 448 I/O, in any combination at 96kHz, spread across multiple racks. While the SD Rack converters can operate at 192kHz, you can also select other sample rate options for specific outputs — for example, MADI at 48kHz for broadcast or recording feeds, or 96kHz — meaning that the SD Rack serves as a multi-sample rate signal splitter, which even allows you to use DiGiCo’s next generation ultra-smooth microphone preamps to replace the standard microphone preamps of an analogue, or other brand of digital console. At 96kHz, the measured delay from the stage rack A-D, through a full channel with processing via the bussing with processing, back to the stage rack D-A is around 1ms.

Also integrated with the MADI split is Gain Tracking, which allows another console or a broadcaster to take any of an SD Rack’s AES, analogue or MADI outputs at a stable level, irrespective of the microphone preamp settings on the SD7, covering a signal level range of +/-40dB. This can be turned on at any time, so the one pre amp for each physical input can have four independently controlled levels.

At Prolight+Sound 2017, DiGiCo introduced a newly designed 32-bit ‘John Stadius’ Mic Pre-Amp for the first time. John and his team have been designing Pre-Amps for over 40 years. This new improvement in converter technology has allowed them to develop a Pre-Amp with audio qualities they have long aspired

to achieve, in an 8 channel format easy to retrofit within existing SD Rack card frames.

The specifications of the 32-bit SD Mic Pre-Amp Card include a fully differential audio path from input to converter, twin 32-bit ADC conversion per channel, conversion time of 73us, dynamic range of 123dB, 128dB EIN noise, distortion 20-22kHz <0.002% THD+N, frequency response 20-44.6kHz ±0.1dB, 0deg phase shift 20-22kHz, and dedicated analogue, ultra-low noise linear power supplies including +48V phantom power.

www.digico.biz

Focusrite new Red 8Pre

RedNet is Focusrite's flagship range of modular Ethernet-networked audio interfaces that harness the power of Audinate's Dante digital audio networking. RedNet is a scalable, near-zero latency audio distribution system that can be used to expand I/O channel count, interface digital components, and/or bridge between Pro Tools|HD or MADI and the Dante audio network. The RedNet range covers a wide array of quality mic preamps and interfaces with low latency, all designed to be scalable and compatible. They all feature audio over IP, so distances are not an issue, as long as they're on the same ethernet network.

In September 2016 Focusrite added a new interface, the Red 8Pre. The new eight-preamp interface features 16 analogue inputs and 18 analogue outputs from a total track count of 64-in/64-out, all in a 1U rackmount. With dual Thunderbolt 2, Pro Tools HD DigiLink and Dante ports, it looks a great high-end interface for a networked facility. Round-trip latency as low as 1.67ms permits recording with preferred plug-ins in real-time and simplifies workflow, with no need for outboard DSP, while dual DigiLink ports connect the Red 8Pre directly to any Pro Tools | HD system. The eight mic pre's are the same Red Evolution mic pres as those found in the Red4, with noise specs of 129 dB EIN and 63dB of gain. The preamps include Focusrite’s unique Air effect, recreating in the analogue domain the sound of the transformer-based mic preamps in the classic ISA range.

Running on Mac or Windows computers, RedNet Control provides remote control and related functionality for the RedNet range. The latest version, RedNet Control 2, has been completely revised, with a new tabbed browser-like graphic user interface and the ability to handle up to 600 RedNet devices. Available units in the system can be dragged and dropped from the searchable Device List sidebar into a grid on a tab, where they are shown

www.focusrite.com
Two new audio networking products have been revealed by prolific product-line enhancer DirectOut Technologies, Mittweida, Germany. The SG.MADI is a SoundGrid/MADI converter offering features tailored for broadcast, live and studio applications. SG.MADI features 2 microphone inputs with switchable pad and phantom power, 2 line outputs, and headphone out. Three interface standards are provided for MADI: optical SC, coaxial(BNC), and SFP. Two GPIs allow the integration of control signals via the network or MADI link. An SG.MADI with fail-safe/redundant PSU is scheduled for release in August 2017, at a proposed price of €2,400.

MONTONE.42 is a fully AES67-compliant MADI (AES10) to Audio-over-IP bridge based on RAVENNA audio networking technology. As the demand for Ethernet-based transmissions in professional audio environments increases, MONTONE.42 provides integration of network audio with existing infrastructures. Equipped with four MADI and two gigabit network ports, it serves as a versatile link for broadcast, live-sound and studio applications. MONTONE.42 offers both an AES67 profile for straightforward network integration and RAVENNA profiles for more demanding applications requiring high performance transmission at minimum latency. MONTONE.42 can be synchronized to MADI input, word clock, video reference, network (PTP) or internal clock. The device may also serve as a network grandmaster. The achieved clock accuracy complies with AES11 throughout the whole network.

www.directout.eu

Yamaha AES67 support for CL, QL & TF Series

The Dante audio network solution by Audinate was first adopted by Yamaha in 2012 for the CL series digital mixing consoles, and has been Yamaha’s network of choice for pro audio devices such as digital mixers, signal processors, power amplifiers, and more ever since. AES67 compatibility will allow Dante to communicate with a variety of other audio networks for even greater system flexibility. Audinate has already begun providing AES67 support in Dante, and Yamaha is in the process of updating Dante equipped products with corresponding device and Dante firmware updates.

Yamaha’s popular TF series gains significantly expanded audio networking capability with firmware update version 3.1. The update will allow TF series equipped with an NY64-D Dante interface card to connect to audio networks such as Ravenna that support the AES67 audio-over-IP interoperability standard. Update V3.1 adds AES67 support to the TF series digital mixing consoles. The new firmware also supports Dante Device Lock, allowing PIN based locking and unlocking of devices on a Dante network. In March, Yamaha’s CL and QL series digital mixing consoles also received a “.1” firmware update to support AES67. Apart from improved EQ control and expanded wireless compatibility, the update CL/QL 4.1 firmware also supports Mounting and patching of the Audinate Dante-MY16-AUD2 Mini-YGDAI card and dbx audiotechnik DS10 Audio Network Bridge. New components and a firmware update will also boost the versatility and performance of Yamaha’s well-respected RIVAGE PM10 digital mixing system. The HY256-TL-SMF is a TWINLANe card that supports single-mode optical fiber. Yamaha’s TWINLANe audio network with up to 400-channel capacity has previously only supported multi-mode fiber connections. The HY256-TL-SMF expands system connection flexibility by also supporting single-mode fiber connections.

www.yamahaaproaudio.com

Audinate Dante Software “Combo Pack”

Audinate, creator of the Dante audio networking technology, released Dante Via version 1.1 in September 2016, a major update to their networking software. More recently the Dante Software “Combo Pack” was announced, bundling licenses for Dante Via and Dante Virtual Soundcard for $59.95, representing a 25% saving. The Combo Pack provides computer-based audio flexibility, including the ability to: record up to 64 channels of audio using your favorite DAW with Dante Virtual Soundcard; share audio from local applications and USB devices over an AoIP network with Dante Via; create computer-only audio networks, or connect to any Dante system. The “.1” update brings multichannel application support, support for high performance ASIO drivers, source mixes with individual level controls for stereo destinations and Device Lock (prevents unwanted changes to Dante Via 1.1 from other network users).

To access a network previously required dedicated Dante hardware to provide nodes (points of access on and off the AoIP network). Using Dante Via the computer itself can be used to provide access to the Dante network — unlike Dante Virtual Soundcard it can do so without any hardware at all. It can be used to send and receive audio to and from the network and to and from audio sources locally on a single computer.

Dante Virtual Soundcard uses the Ethernet port you already have to share locally attached USB, Firewire or Lightning devices to an AoIP network. In other words, your USB audio interface can be accessed by a production pro on another floor of your building, who is using an device connected to the same network. The Virtual Soundcard can also be used locally as an alternative connection strategy for Ethernet/Dante equipped preamp/converters. If your production computer lacks a USB 3 or Thunderbolt interface, Dante Virtual Soundcard is a faster solution. Record up to 64 channels of audio from your Dante networking ADC using your favorite DAW such as ProTools, Logic, Cubase.

www.audinate.com
Audio networking and interoperability

What is AES67 and why is it important?
ANDREAS HILDEBRAND, ALC NetworX, explains.

Our reliance on IP in just about every facet of our lives continues to increase, and audio networking is no exception. AoIP (Audio-over-IP) is no longer a foreign concept to most and is rapidly becoming the norm rather than the exception in the world of audio networking. However, the question remains, how to successfully connect equipment that relies on different audio protocols to exchange data? There exists a vast array of protocols on the market: Dante, RAVENNA, Livewire, Q-LAN to name but a few … Some are proprietary while others, like RAVENNA, are based on open standards. The problem is that they don’t talk to each other although they are all based on IP.

The AES67 standard on High Performance Streaming AoIP Interoperability was published in September 2013 precisely to address this issue. It is a set of interoperability guidelines for high performance professional digital IP networking that addresses synchronisation mechanism, encoding format and QoS provision for delivering audio data as well as connection management functions associated with audio delivery. AES67 uses existing protocols and technology from the IEEE, IETF and other standards developing organisations to define how to use existing protocols as a system in an interoperable manner. AES67 is not intended to be a solution on its own, but rather provides a means for exchanging audio streams between areas with different networking solutions or technologies already in place. It should be stressed that AES67 is NOT another protocol or indeed a new technology.

High performance, low latency and the use of existing network technology were three primary requirements for the standard when it was being developed. AES67 excludes low performance networks including the public internet. It also excludes audio data compression, and is restricted to IP-based networks. Although video is not part of this audio standard, any media could be transported with the underlying architecture. Actually, the Video Services Forum has developed a recommendation — VSF TR-03 — that covers the transport of SDI video signals as elementary essence data streams, and this does not only include AES67 for the audio part, but is based on the same working principles and standards. This recommendation has been passed on to SMPTE, which is currently working on a new standard for Studio Video over IP (SMPTE ST2110) taking TR-03 as a basis.

The intention with AES67 was that existing audio IP network solution providers should be able to use AES67 as a special “mode” to enable interoperability of their systems, or indeed to use it as their native operating mode. It is a layer that describes what must be implemented by networking solutions to make communication between them possible. Precision Time Protocol (PTP) is used as the means of synchronisation so that media clocks in different devices can follow the same time reference, even being phase-locked. IP v4 is the norm for AES67, but the system has been designed to be able to employ IP v6 if necessary. Unicast and multicast modes are possible, and audio data is transmitted using RTP (Real Time Protocol). Up to eight audio channels can be carried per stream (normally at 48 kHz sampling rate, 16 or 24 bits per sample), and there is no limit to the number of streams, which can be synchronised to sample accuracy. The packet time (number of samples per packet) has carefully been chosen as a compromise to meet low latency requirements but allow pure software-based systems with potentially limited performance capabilities (e.g. a PC-based virtual sound card) to participate in the standard. 48 samples per packet have thus been chosen as a mandatory requirement, resulting in 1 ms packet time at 48 kHz sampling rate. Shorter and longer packet time numbers are given as a recommendation to accommodate a wider range of applications.

The issue of discovery (of devices or streams on the network) was deliberately excluded from AES67 for a number of reasons. Firstly, there are lots of discovery protocols already available and it would have narrowed the acceptability of the standard if just one was designated from the start. Secondly, discovery is not strictly necessary (or even desirable in some cases) for interoperability. AES67 therefore works like a phone book system where a device needs first to be provided with the “magic numbers” of available AES67 streams on the network. Current developments, though, are improving the integration of discovery methods with AES67 (more on this later).

So what is RAVENNA and how does it relate to AES67? — RAVENNA is an open technology platform for audio networking over IP. It is not a proprietary design and uses standard protocols and technologies. It is designed to work on existing network infrastructures without the need for a separate audio network and requires no special licensing. Furthermore, RAVENNA is fully AES67-compliant right out of the box, making it currently the easiest and most cost-effective route for manufacturers to offer their clients AES67 compatibility for fully interoperable signal transport solutions. Other technologies have since been adapted to support AES67 but were not natively compatible. It has been suggested that RAVENNA is a bit like a cook book in that it provides the ingredients necessary to build a working system, along with the method to make it work. AES67 differs from RAVENNA in that it describes a relatively limited subset of specifications purely intended to enable interoperability and is not a stand-alone solution, whereas RAVENNA is a complete solution that offers a much wider range of functionality and options, including redundancy and discovery.

Another advantage of RAVENNA being an open technology based on existing standards is that manufacturers are free (and encouraged) to develop their own solutions for integrating RAVENNA into their products. ALC NetworX remains the principle solution provider, but other manufacturers such as Archwave, Covelez and Digigram have also developed OEM solutions.

Network Requirements — The network infrastructure must provide sufficient bandwidth for the data to be carried, and managed switches (now available for relatively little money) must provide full switching capacity without packet loss. It’s always a good idea to have reserve bandwidth available, especially if it is a general-purpose network. Under ideal conditions with a tightly controlled system it might be possible to load up the network to greater than 90% of its capacity, but generally one should start looking for more bandwidth when the network is running at a mean load of 50% of its capacity.
Multicast IGMP support is also required, which ensures that only those packets are forwarded to destinations that are asking for them, rather than flooding the outer reaches of the network with unnecessary data. The key to ensuring that this doesn’t happen is IGMP (Internet Group Management Protocol) for multicast stream registration. That said, even if IGMP has been implemented in end nodes, then proper switch configuration is still required to prevent overloading the network with unwanted multicast traffic. Situations where multicast is known to work hand-in-hand. The human challenge often exists a human challenge when engineering (the audio guys) and IT (the network guys) need to work hand-in-hand. The human challenge often is to understand and respond appropriately with regard to the network capabilities that shift, and the industry has clearly taken note of the fact that ultimately, it’s in everyone’s best interest to promote interoperability to improve workflow and system management.

The SC-02-12M standards group continues to work on the on-going improvement of AES67 and welcomes participation from any interested parties. The recent IP Showcase at NAB which demonstrated full IP interoperability based on AES67 and the new ST2110 standard for video was ample proof of how the industry is pulling together to promote interoperability. It was jointly based on AES67 and the new ST2110 standard for video which was so successful that it’s likely to need more room at next year’s show. IP networking is here to stay, and the industry has clearly taken note of the fact that ultimately, it’s in everyone’s best interest to promote interoperability to improve workflow and system management.

Summary — Despite a certain initial reticence on behalf of some people when the standard was first introduced, AES67 has clearly reached the point where it has gained widespread industry recognition and acceptance. It provides a mode of operation that allows one system to communicate audio information with another, while still allowing for enhanced features native to existing technologies to be added on top. While certain features such as device/stream discovery were intentionally omitted from the standard, open solutions such as the freely available RAVENNA RAV2SAP conversion software are rapidly emerging that will deal with this.

Audio engineers would do well to familiarise themselves with the terminology and principles of networking in order that they can talk the same language as the IT people that have to implement some of their requirements. A basic knowledge of networking protocols and setup will also help audio engineers to avoid “bad behaviour” in a networking infrastructure, such as data flooding. The recent IP Showcase at NAB which demonstrated full IP interoperability based on AES67 and the new ST2110 standard for video was ample proof of how the industry is pulling together to promote interoperability. It was jointly organised and supported by the AES, AIMS, AMWA, EBU, IABM, MNA, SMPTE and VSF; organisations that include both audio and video; and was so successful that it’s likely to need more room at next year’s show. IP networking is here to stay, and the industry has clearly taken note of the fact that ultimately, it’s in everyone’s best interest to promote interoperability to improve workflow and system management.
Remote production — Whenever the topic of IP connectivity for consoles comes up, the buzzword “remote production” is not far behind. Essentially, connecting studios to central DSP farms is already a manner of remote production, albeit one which enjoys perfect conditions and fast networks. There might be several microphones at a studio a hundred kilometres away. A NEXUS Base Device might be installed there, acting as a stage box. Using a leased line spanning the distance, the Base Device is integrated into the main studio’s audio network via high-performance fibre optic and Stage Tec’s proprietary TDM protocol. This approach has been used for over a decade, for example at the public broadcaster Hessischer Rundfunk in Frankfurt, Germany. HR use it to connect all subsidiaries to the main broadcasting house.

As an alternative, the connection can be made through AES67 as Audio over IP. A remote participant in a round table can encode their signal in AES67 and send it to the studio network via IP where NEXUS converts it and integrates it into the live mix. Broadcasting major sports events like the Olympic Games are classic cases in which broadcasters use remote production to reduce high travel and transport costs. There would not be much point in locally connecting all sources and destinations to the distant broadcasting centre via IP and doing the mixing there. The laws of physics prevent latency-free transmission of signals over long distances. Just imagine the commentator who receives the monitor mix on their headphones from halfway around the globe; latency would interrupt their flow.

The solution to this conundrum is to combine a local mixing network on location with another mixing network at the broadcast centre and implement a remote connection between the two. All cases that require a high-performance connection (which all time-critical processes do) are realised within the respective mixing network. In contrast, cases where time lags are irrelevant can be transferred from one mixing network to the other and realised there. So we distinguish between connections that either are high-performance or are not time-critical, and this is true for both audio and controls.

The mixing console itself, meaning the combination of control console and audio unit, is linked up locally. The broadcast centre houses the control console and the DSP farm, and audio signals from the remote audio network can be integrated over IP.

The next step in Remote Production are cases where the DSP farm is controlled via a high-latency long-distance connection. Here we use a browser-based remote access of the mixing console rather than the large mixing desk. This can be useful — among other things — for adjusting the monitoring volume of commentators or providing remote technical support for self operating desks.

Helmut Jahne, Managing Director of Stage Tec notes, ‘We’ve been doing this on a smaller scale for several years, for example with our small broadcast desk ON AIR flex in Australia. The new mixing console lays the groundwork to implement the approach with large mixing consoles as well.’

Colour-coded interface — The new AVATUS is a sleek, flat console with Stage Tec’s hallmark dual encoders in the channel strips. Presently, the AVATUS console concept comprises three modules: the fader panel with 12 faders, the rotary encoder panel with 12 columns of 4 dual encoders, and the displays based on 21” multi-touch screens.

A classic mixing console consists of a panel with faders, a touch screen situated above it to operate console functions, a rotary encoder panel, and a touch screen panel, installed at an angle, to serve as the meter bridge. The concept can accommodate a control unit of up to 96 faders, or eight fader panels in a row. In contrast, the smallest conceivable AVATUS control unit is simply a single touchscreen.
There is no dedicated central control module, as each touchscreen can switch to central mode. This feature enhances control unit flexibility and enables multi-user access. The modules are no longer allocated to a single function and instead can be used freely to control whatever is needed. Stage Tec emphasises a workflow-oriented operation where the touch displays are context sensitive and provide only those functions relevant to the tasks at hand.

Users can utilise the rotary encoders for whichever function they require, usually EQ, compressor, limiter and the like. Colour-codes, accomplished with LEDs in each rotary encoder, indicate at a glance which function is on a particular rotary encoder in any given moment. The faders are colour-coded to provide an immediate overview in big projects. Input channels glow green, yellow is reserved for groups, sums shine in a distinctive red, and blue indicates the aux channels. The fader sections can be used with up to 16 layers.

‘One of our priorities in designing the control interface was to create an easy-to-use and clearly arranged layout. AVATUS is intended for large mixing projects, and effective everyday usage depends on a clear structure and explicit signalling’, notes Harald Klaus. He was one of Stage Tec’s founding members in 1993, and as a Tonmeister has been the driving force behind the operating philosophy of the mixing consoles ever since.

Browser-based controls — The AVATUS control unit requires a high-performance Ethernet connection (Wi-Fi is not adequate). This is what the browser-based controls, which are equally capable of controlling all functions and configurations, are designed for. You can operate the mixing console from any browser on any operating system and through any network-enabled device without installing a single application. The browser control is useful for example for changing the desk settings from a tablet during rehearsals, controlling them directly from the auditorium or — thinking remote production here — accessing them via wide area network.

Compatible with AURUS — A customer with an existing NEXUS audio network and an AURUS mixing console can move up to the AVATUS step by step. Acting as an I/O matrix, the entire audio network is an integral part of the mixing console and remains unchanged. Because AVATUS and AURUS can utilise the same DSP boards, a truly seamless transition from one type of mixing desk to the other becomes possible. The two mixing desk types can also operate in parallel within the same audio network without experiencing any issues, which incidentally is also true for all other Stage Tec mixing desks.

At present, AVATUS is still in the concept phase. An initial product version is announced for the second quarter of 2018. The presentation of the concept in April generated huge interest. IP is a big topic in the broadcast market, which has been keen about AVATUS from the get-go. ‘The industry is experiencing a major shift right now. Today, with IP connectivity and the move toward remote production we are witnessing a similar upheaval in the basic principles of mixing desks and sound studios generally as last seen in the changeover from analogue to digital technology’, says Sahne. ‘The future will be fascinating!’

Avatus

The AVATUS provides more than 800 input channels and 128 summing buses, which can be combined with each other freely from mono to 7.1. The controls have integrated features like the de-esser, loudness metering and the Stage Tec automixer. Mixing console sizes range from 12 to 96 channel strips. AVATUS builds on standardised technology and protocols. This facilitates control and integration of third-party devices. Implementing the AES70 standard should further increase bidirectional connectivity with other systems.

Nexus

A NEXUS network consists of separate Base Devices placed wherever you need to route audio, control, and other signals to and from the network. All Base Devices are interconnected through digital links implemented as floating fibre-optic cables. Each Base Device acts as an autonomous local router. This way, a NEXUS network offers distributed intelligence including decentralised control and crosstalk information.

Each Base Device has a custom configuration with all interfaces and modules required onsite. The Base Device network allows for routing any sources to any destinations regardless of I/O formats and their physical positions on the network. This effectively eliminates the need for complex and costly format conversions. NEXUS acts as a host for the mixing desk boards from the AVATUS, CRESCENDO and ON AIR flex series.
Latency in AoIP

The latency introduced by networked audio is likely to be less than the latency improvement obtained by increasing your DAW sample rate. Jan Lykke of Digital Audio Denmark explains.

One of the “myths” that we hear a lot is that it is impossible to use AoIP (Audio over Internet Protocol) for recording and overdubs because the latency is too high. In a digital world, the audio you listen to is delayed by a certain amount, which can be caused by a number of factors, but often the delay is so subtle you will never notice it. In fact, the term “delay” is precisely what happens when we talk about latency, and audio delay has always existed. Quite simply, sound waves have to travel over distance and it takes time to get from A to B — 343 metres per second.

When we talk about sound reinforcement, for example, secondary loudspeaker arrays further from the main PA system are deliberately delayed by a few milliseconds to compensate with regard to the listening position (further to the back of the venue). On a live music stage, you will still often see monitor wedges placed on the floor, if the artists have chosen not to use IEMs (In Ear Monitors). This is necessary because, if the stage is fairly large, the monitor wedges on the floor wedge monitor to the ear of the musician, if the distance is 2 meters, there is a very noticeable amount of delay. But considering the distance even from the main speakers or “side fills” would be so far away the artists would experience (In Ear Monitors). This is necessary because, if the stage is fairly large, the monitor wedges on the floor wedge monitor to the ear of the musician, if the distance is 2 meters, there is actually a latency of 0.8ms.

In a network we take samples, put them into a packet, and send onto the network. The point of setting up data packets is to be able to send audio around the network. If we send packets from, for example, 3 different audio devices on a network, they will come to a switch. The switch cannot take those 3 packets and send them out simultaneously, they will be sent out sequentially to avoid a data collision. At the receiving end, the packets will arrive at random intervals, so in the receiving device we need to have a receive buffer, which is basically there to “time align” all the samples. When we create the samples, we put a time-stamp on each sample, so the sequence can be re-constructed at the receiving end and to make sure they are spaced equally. Because the receive buffer needs some time to compensate for any packet delay variations in the network, we can say that the output time=timestamp+pre-defined network latency. Some audio pros think that, within an audio network, the latency varies all the time: it may do so by a certain amount, which can be caused by a number of factors, but other factors may also influence it.

### Traditional Points of Latency

Before introducing potential latency calculations on an IP-based audio network, it is important to clarify there are numerous points where latency occurs — in both the analogue and digital domains. Let’s take a closer look at the case of a vocal recording in a typical studio environment:

- **Distance from the vocalist’s mouth to the microphone** — 25cm of distance equals approximately 0.8ms of analogue latency.
- **AD conversion** — converters these days use digital filters. Digital filters introduce a small amount of latency because they need time to process an audio signal. Depending on the manufacturer and design of the filter, we are looking at around 15-20 samples of latency. 20 samples at 48kHz is 0.4ms, if you go to 96kHz you’re looking at 0.2ms.

- **DA conversion** — Filters typically introduce between 10-40 samples of digital delay. If we use 50 samples as an example, this equals 0.6ms of digital latency.

- **Monitoring** — In this case a vocalist would wear headphones and there would be no analogue latency worth mentioning, but if, for instance, a guitarist or bassist would be tracking in the control room, there would most likely be at least 1-2ms of analogue latency from the monitors to his/her ears.

### AoIP Latency

Now, let’s find out how much additional latency you could expect if you decide to establish an IP-based audio solution such as Audinate’s Dante network protocol. There is no fixed value of ms of latency being introduced on a Dante network. It depends mainly on the number of switches in the network though other factors may also influence it.

![Network latency](image)

**Network latency**

**Receiver buffer**

In a network we take samples, put them into a packet, and send onto the network. The point of setting up data packets is to be able to send audio around the network. If we send packets from, for example, 3 different audio devices on a network, they will come to a switch. The switch cannot take those 3 packets and send them out simultaneously, they will be sent out sequentially to avoid a data collision. At the receiving end, the packets will arrive at random intervals, so in the receiving device we need to have a receive buffer, which is basically there to “time align” all the samples. When we create the samples, we put a time-stamp on each sample, so the sequence can be re-constructed at the receiving end and to make sure they are spaced equally. Because the receive buffer needs some time to compensate for any packet delay variations in the network, we can say that the output time=timestamp+pre-defined network latency. Some audio pros think that, within an audio network, the latency varies all the time: it may do so by a certain amount, which can be caused by a number of factors, but other factors may also influence it.
latency will be calculated according to the fixed buffer sizes of the total number of switches traversed.

If your audio has to traverse three network switches, you can go down to 250μs, and 250μs actually equals 12 samples at 48kHz. If we go up to traversing 5 switches, we will be at 500 μs, so when we talk about network “latency” we are actually discussing very small time increments. Even in extremely large broadcast facilities where the signal may traverse up to 10 switches we will only add 1 msec latency. A studio or post-production facility would expect to have a network latency of between 250-500μs.

If you have less than 4 switches in your network, you could likely go as low as 0.25 ms of latency on either side of the ASIO/DAW instance.

In that case, you should add 0.5ms of latency that is caused by the network.

**CONCLUSION**

We think that it is fair to conclude that adding AoIP is not what is going to make or break your recording in terms of latency. It is a myth that you cannot track live performances which need to be monitored in real time on an audio network. It is indeed perfectly possible, and most artists would be completely fine with a latency below 6ms. Again, just consider the musician performing live on stage, listening to monitor wedges on the floor or an amplifier several meters away.

In closing, we would like to stress that the numbers we used in the above examples are not set in stone. They are examples and may therefore vary slightly depending on the manufacturer of the equipment you use, your computer’s performance, etc. But we hope that you now have a better idea of how your overall latency would become affected in case you consider — or decide — to move on to the world of networked audio.