Professional AoIP for Broadcast - The Way Forward

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Tom Knowles, Product Manager - Broadcast Systems

Introduction

Today's standard industrial IT infrastructure has already overtaken the technology of AES/EBU, MADI and TDM routers in terms of performance, cost and flexibility. The rate of development of IT systems, fuelled as it is by a multi-billion dollar industry many times the size of the broadcast industry, is certain to widen this gap in the future. Over the next few years IT infrastructure will replace current broadcast infrastructure, delivering additional flexibility, better scalability and significantly lower costs. Broadcasters and Systems Integrators can expect more choice in selecting interoperable equipment and solutions from a range of suppliers and will have more scope to manage and control these systems.

Audio over IP (AoIP) has been embraced by the installed sound, live and radio industries. Much has been learnt from this but larger broadcast applications bring additional constraints, especially channel count, synchronisation and latency. Audio systems engineers will need to learn how to set up, configure and manage IT networks; IT specialists will have to understand why and when audio networks need to be separate. Education will be key.

As with previous transitions, audio, with its lower data rates, will be a pathfinder for developments that will follow in the video domain. IT is already widely used in broadcast for file transfer, but SMPTE 2022-6 (high data rate streamed video over IP), which was published in 2012, is evidence that SDI will eventually be replaced. Central to considerations about audio or video over IP is the value that metadata brings to systems. Concepts such as discoverability and automatic configuration are key to delivering powerful workflow benefits.

In this paper we describe our vision for the future of Audio over IP for professional broadcast applications, explain which technologies we are using, discuss the advantages and challenges associated with AoIP and demystify the jargon.

Budgeting for an IP future

It's rare that a facility has the luxury and budget for a ground up development and it's likely that the thought of moving to an IP future could be seen as a daunting proposition. The good news is the transition does not need to be an all or nothing leap. In fact, SSL believes that IP and TDM solutions are likely to coexist for some time to come. For this reason, one of SSL's first developments for our IP systems product line is a 'broadcast robust' MADI to IP bridge product.

As for installing a whole new physical infrastructure, the likelihood is that the IT department may have done much of the work already. A significant benefit of a good IP based solution is that it uses 'standard' cabling and hardware to provide the complete solution. This helps make infrastructure planning straightforward, predictable and easily costed. The required technology has a wide range of price points and good quality solutions are available at much lower cost than custom broadcast routing hardware. Twisted pair copper and fibre, managed switches and simple punch down termination all reduce overall cost and make expansion straightforward as well as cost-effective.

It is useful to remember that the base wiring and switching infrastructure for all IP based systems is the same. A key benefit of standard industrial IT network based infrastructure is that it is format agnostic and multiple types of signal can coexist easily on the same hardware and cabling so network infrastructure can be used to implement device control, as well as audio transport, further reducing wiring and installation costs.

The benefits of an IP solution.

There are several significant benefits that are key reasons to implement a network based routing solution:

Reduced Cost
As discussed, simple economics make an IP solution an attractive proposition for audio routing.

High Channel Counts
Another significant benefit of a networked infrastructure is capacity, and particularly capacity vs cost. A single gigabit network connection is capable of carrying 512 channels at 48kHz, or 256 channels at 96kHz. AoIP capacity compares very favourably when considering using network based devices as your core audio router. With 512 audio channels in each direction on a 1GB connection, a single 24 Port GB Switch is capable of providing the equivalent audio routing capacity to a 12,288 by 12,288 audio router. It should be noted that this comparison is heavily based on a traditional broadcast concept of a central core routing device. Depending on the physical building, existing network infrastructure and system requirements, alternate network topologies may be more advantageous and cost effective.

Easily Expandable
When the needs of the installation expand, the system can grow much less expensively than a traditional TDM solution. Adding a network switch is a much easier way to expand capacity than having to replace a router with a larger one. AoIP Network routing is not subject to the square law growth of TDM routers i.e. doubling the size of a TDM network typically quadruples the size (and possibly cost) of the router.

Full Network Redundancy
Redundant operation is at the core of security in both the broadcast and IT industries, therefore designing a system to be secure shouldn't be a concern. There are often several redundancy strategies in network designs to create fully resilient networks, these include Mesh networks, Spanning Tree, Token Ring and Link Aggregation. These designs are supported by many enterprise class network switches and an AoIP network solution which uses standard networking protocols should support any or all of these solutions. As an example, SSL's IP solution provides inbuilt parallel redundancy allowing both main and redundant networks to be always active, a device simultaneously receives and transmits on both networks, audio samples can exist on either or both networks providing seamless, glitch free redundancy and failover.

Resilient Distributed Routing Control
As the signal routing in an IP network is end point based, a catastrophic failure of a single device will have no effect on any other devices in the network. With redundant routing and/or connections, single points of failure are easily designed out of the system. In addition, the use of the network for control information means routing control can be achieved from an unlimited number of devices or terminals on the network.

A History of Broadcast audio Routing
Patchbays
The earliest forms of signal routing in radio studios used mechanical patchbays inherited directly from the telephone industry. With later evolution and more complex needs, telephone patchbays were augmented by 'Uniselectors', electro-mechanical signal routers remotely controlled by electrical pulses. Companies like the BBC used these to control routing within their broadcast stations, allowing flexible and efficient use of studios, equipment and people to produce complex news and sports productions... **Click here to read more**

These control benefits extend beyond routing control to include:

- Device parameters that can be configured from anywhere on the network without additional cabling
- Any other software control application can use the same network infrastructure
- A single cable infrastructure can be used for configuration, control and audio e.g. Ember+, ProBel, SNMP, UI etc.

Looking to the Future - IP is already there!

With discussion about the future of broadcast audio featuring Metadata and Object audio Transport, a network infrastructure is already equipped to manage the technical requirements for these complex applications. If we consider 'broadcasts' where multichannel audio and metadata are transmitted to the consumer to provide personalised listening experiences, in future it is likely that these 'broadcasts' will happen over IP to the consumer, rather than over the air. The metadata requirements in a production environment mean network based technology is essential.

Selecting an IP Protocol

Many of the AoIP benefits discussed so far come with the structure and technology inherent in using mature solutions developed in the IT industry. For SSL a significant decision in the project to develop IP based solutions was selecting which IP protocol to adopt. All options were considered but in many ways, the decision was made easy by the well-established and complete solution presented

The Evolution of audio over IP

In the early days of networking, research was driven by the desire to overcome the inherent waste in a time division multiplexed (TDM) connection. By turning signals into packets and sending multiples of these along the..

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by Audinate's Dante technology. The Dante Protocol brings some specific benefits that arise from their technology:

Auto Discovery and Plug & Play Device Connection

A significant benefit of the Dante IP solution is the ability of all Dante devices to support Auto Discovery and 'Plug & Play' operation. This means that devices which are plugged in to the Dante network announce their presence and can be 'found' by other Dante devices on the network. This makes it easy to move equipment from place to place and extends to Dante products from any manufacturer.

AES 67 Compatibility

Dante is compatible with the AES 67 standard, a "standard for audio over IP interoperability" agreed in September 2013. AES67-2013 provides comprehensive interoperability recommendations in the areas of synchronization, media clock identification and network transport. It specifically addresses high performance media networks that support professional-quality audio (i.e., 16-bit, 44.1-kHz and higher) with low latency (less than 10 milliseconds), and at a level of network performance that can scale from local area networks (LANs) to enterprise-level networks. This means that Dante products will be able to exchange audio over IP with other AES67 compliant products.

Guaranteed Interoperability

With Dante, devices from all manufacturers are guaranteed to work together with a level of interoperability that goes much further than the audio stream compatibility of AES67 compliant products. As proven at interoperability demonstration events, multiple broadcast products all appear on the same routing matrix automatically. More than 100 manufacturers are already shipping over 350 different Dante devices, with over 200 manufactures having licensed the technology. With the majority of these devices developed primarily for applications in the professional public address and live sound markets, Dante also facilitates the specification of systems that encompass the increasing demand for multi-purpose installations.

Standard IT Network Infrastructure

Dante is able to use existing, off the shelf network

switches, unlike AVB, which requires specialised, expensive, less readily available switch hardware. In addition, Dante uses established IEEE and IETF standards so its network data can mix with traffic on any standard IP infrastructure.

With Dante being built on standard networking protocols, moving beyond networks managed by your organisation is possible. MPLS (Multiprotocol Label Switching) is often used when high performance connections are needed across an external telecommunication provider's networks. Deploying AoIP on this type of leased network is achievable, though as with any use of external services the SLA (Service Level Agreement) is key to achieving the performance required. A guaranteed bandwidth greater than the required channel count (100MB for 64 channels) and latency of <5ms stated in each direction is required to use standard Dante settings. While uncompressed audio across MPLS and other virtual network connections may be new ground for the broadcast market, this is not new ground for AoIP or Audinate. Examples such as the Sydney Trains System's networked public address system uses Dante and MPLS extensively.

Rapid Cohesive Development

Although strict 'standards' based' technology has its merits, history demonstrates that it can take a long time to be fully interoperable. Examples of licensed technologies, such as products produced by Dolby Laboratories are evidence that proprietary solutions can foster rapid cohesive development. It means that manufacturers can develop a wide range of niche products with a high degree of confidence that they will integrate smoothly with products from other manufacturers. The result for System Engineers is a versatile, cost effective broadcast technology ecosystem that contains all of the elements they require.

DiffServ QoS (Quality of Service)

DiffServ specifies the mechanism for classifying and managing network traffic and providing quality of service (QoS) on IP networks. QoS must be enabled to allow Dante to share network infrastructure with other types of data and

signals. In many cases QoS in installed network switches may already be enabled, if not the network switches need to have the Basic mode of QoS enabled, checking that the switch is using DSCP (Differentiated Services Code Point). Education and use of common language between broadcast and networking engineers is key in successfully leveraging the advantages in AoIP technology. The network specialist may want to know more about the DSCP labels Dante uses. Audinate publish the DSCP priority values for Dante here.

Switching Capacity

To achieve high channel counts Ethernet switches must meet specification requirements. In the same way TDM routers are described as non-blocking, network switches can also be described as non-blocking. By design and to reduce cost some switches are not non-blocking, these are often used as edge switches where full bandwidth switching may not be required. The important figure is 'switching capacity' in Gbps, this is the max bandwidth of switch backplane. For a switch to be non-blocking the switching capacity needs to be greater than or equal to the number of ports, times the speed of each port, times 2 (so in and out for each port is considered). For example: a 24port 1GB switch would require a switching capacity of 48Gbps.

Synchronisation and Latency

Dante uses the IEEE 1588 Precision Time Protocol (PTP), enabling all devices to synchronise their local Audio clock to a master clock on the network with sample accurate alignment. No additional clock distribution is needed saving on cabling and distribution amplifiers. Multiple sample rates can coexist on the same network but external SRCs would be needed to connect these signals.

Audio latency is deterministic and is defined on a per device basis at the receiving unit. The latency of connections does not change when you add or remove devices from the network. Device latency should be selected based on the application and network size. Recommendations are from 125µs for networks with 3 switch hops, 1ms with 10 switch hops, to 5ms for larger or leased networks.

Device Control

One of the most active topics of discussion in the various standards bodies and around broadcast conventions is the subject of control. For many applications the advantages of AoIP seem to be tied with the holy grail of a single standard control protocol for everything, with ultimate flexibility. While this goal or aspiration for a common control protocol across many manufacturers may ultimately have advantages, this thinking may miss certain aspects of the advantages of a network; the infrastructure is agnostic to the signal, control data and metadata. Some control scenarios lend themselves to device to device control, while some are more suited to central (or perhaps distributed) control and monitoring systems. As with audio transport, basing these control protocols on standard and widely adopted underlying networking protocols is the key. Integrated control is a significant benefit of a networked solution and the Dante solution to this is well defined.

Audinate's DAPI (Dante Application Programming Interface) includes its own control and monitoring protocol, ConMon. This allows software to be developed that will be able to perform routes, plus configure Dante and network settings on any manufacturers' Dante enabled device, with true interoperability. In addition to the Audinate defined parameters, ConMon includes the ability to add vendor specific messages into the same transport layer. By definition all Dante devices include ConMon. For device to device control, e.g. Mixing console to Mic Pre, this is an elegant path to control interoperability beyond what Audinate have already defined.

To facilitate using Dante to augment existing MADI based infrastructure SSL supports MADI control data tunnelling with its Network I/O: MADI-Bridge.

Routing Control and Security

Audinate provide a simple X/Y routing interface called Dante Controller which can be run on any suitable computer connected to the network. The software provides an elegant approach to basic routing. It is important to understand that routing and configuration for Dante devices is end point stored and routes are based on device and channel names. A receiving device subscribes to signals from other devices that transmit. If a device is moved to an alternate location on the network audio connections are re-established automatically and the channel names appear as they were before. This can be used to great advantage when moving devices around a building, venue or anywhere with a network connection. This also means that there is no need for Dante Controller or another controller based on the DAPI to be running for audio to pass, controller software is required only to make changes.

As with traditional broadcast audio routing, control systems will play a key part in AoIP migration or ground up broadcast IP installations. Routing arbitration is one area that remains the subject of ongoing discussion. If we take arbitration to be the locking of a destination when a route is made so this target route cannot be unintentionally overwritten, then we need to question where to arbitrate this. Within an AoIP system where routing is stored on the end points, this question is similar in scope to arbitration across multiple hardware TDM routers. Previous technology experiences have shown arbitration from a control system considered the master is often the safest way to achieve this without conflict. This is achievable today with third party control systems implementing DAPI. In this scenario security of the network becomes important so that only clients of the main control system would alter routes. There are many approaches to achieve this such as; authentication based access controls (e.g. 802.1X), ACL (Access Control Lists), segmenting switches with VLANs or even mirroring what many people do with traditional broadcast technology in preventing access to the network via locked rooms, locked cabinets and rights managed computers.

Control Development

As discussed Control is an active topic of debate, SSL are actively working with other manufacturers to further develop control definitions applicable to the Broadcast industry, with a vision not only of audio interoperability between manufacturers' solutions, but also control.

AoIP in action - some examples.

While many Broadcast AoIP discussions are about future large scale infrastructure projects there are many scenarios that can benefit from AoIP deployment today. SSL and other manufacturers' Dante implementations are allowing these use cases to be realised with a guarantee of future compatibility.

Remote Production

It is possible to use a network connection to a remote location where a previously laid fibre TDM connection is in place (e.g. Fibre MADI). A fibre cable previously used for a point to point 64 channel MADI (125 Mbps) connection could very easily be repurposed to carry up to 512 audio channels with MADI bridges and network switches. Alternatively the connection could simultaneously be used for other network connectivity alongside the audio.
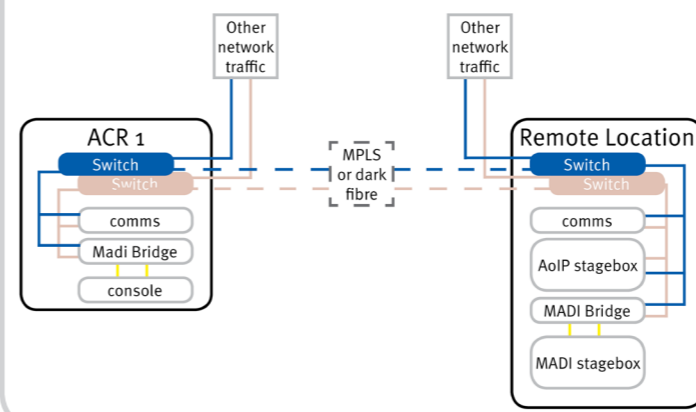
IO everywhere

In many scenarios audio inputs and outputs are needed at many locations across a large physical area. Fig 2. Shows a typical Golf Setup, with connection nodes at each hole.

Network connections by their nature allow efficient resource connectivity regardless of channel count. IO devices can be located where needed without running multiple analogue cables to a 64 channel capacity location as you would with MADI. The advantages when broadcasting from an event spread over a large area are huge. Again with a protocol such as Dante these connections can be used for more than just the audio.
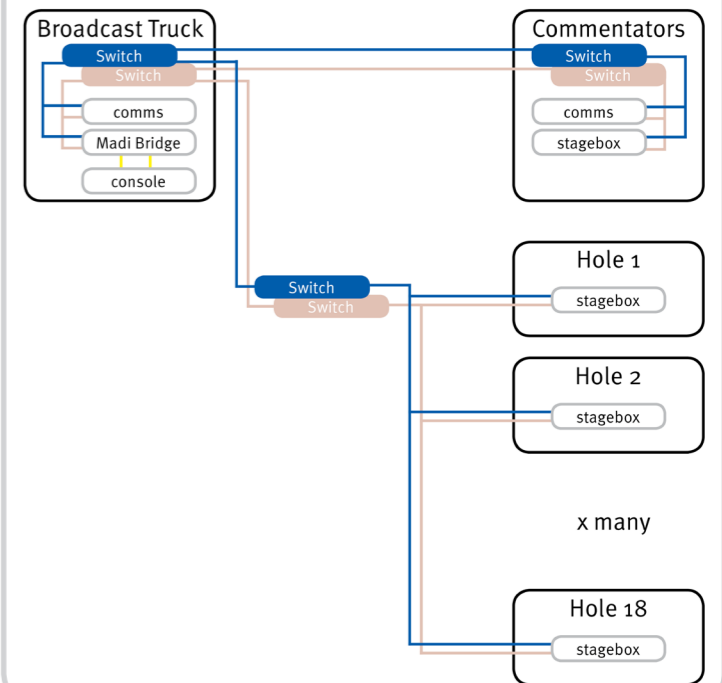
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A network switch as an audio router

Both of the above examples are scenarios that could use AoIP to enhance existing equipment and infrastructure. For a ground up installation any existing TDM based system can be replicated and improved upon. A deceptively simple design ethos applies: Ring, Star or Mesh network configurations can be used for facility wide network design. Placement of network switches locally within control rooms, galleries, studio floors, remote locations etc enables connection of any combination of audio and control devices at each location.
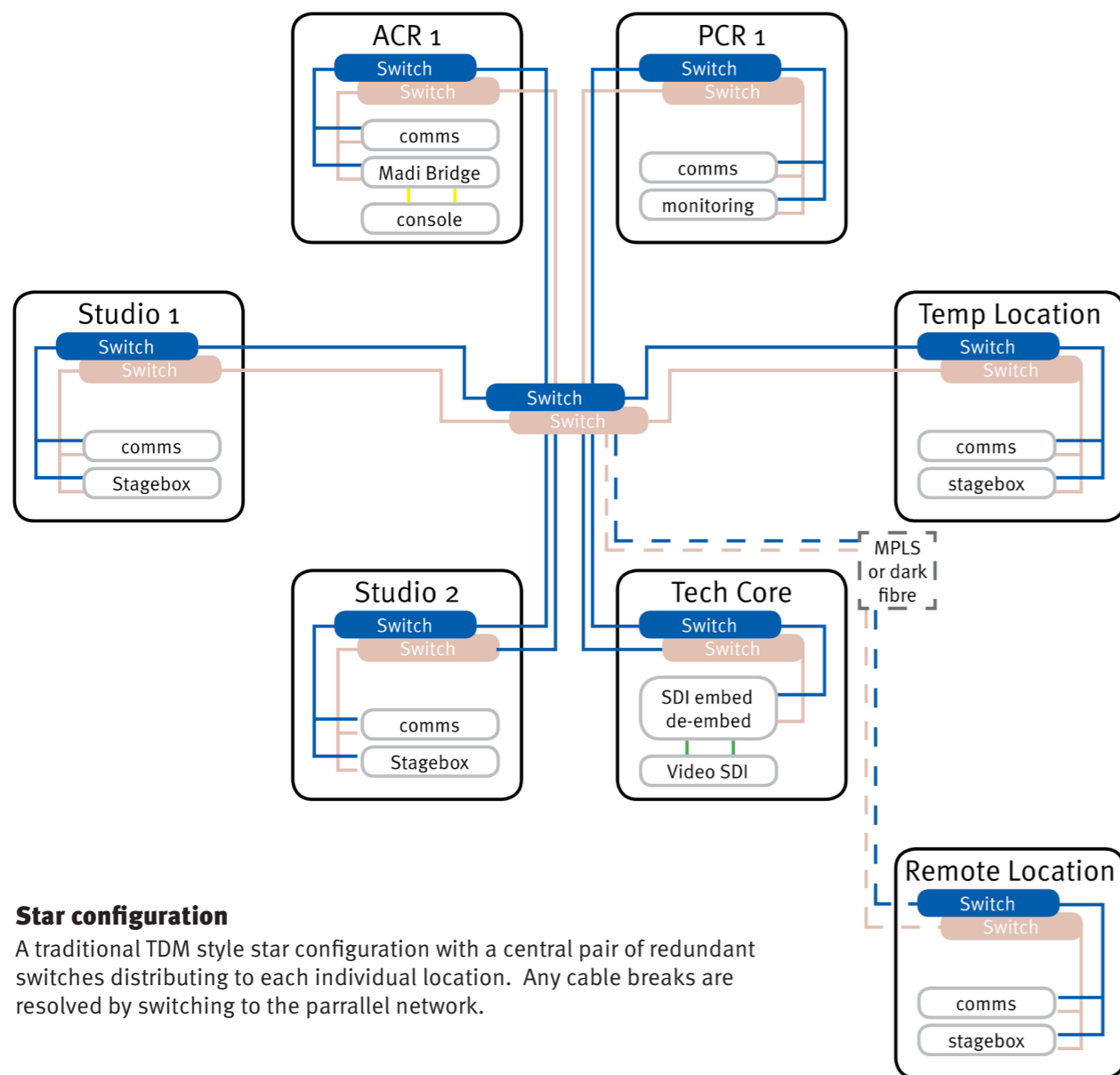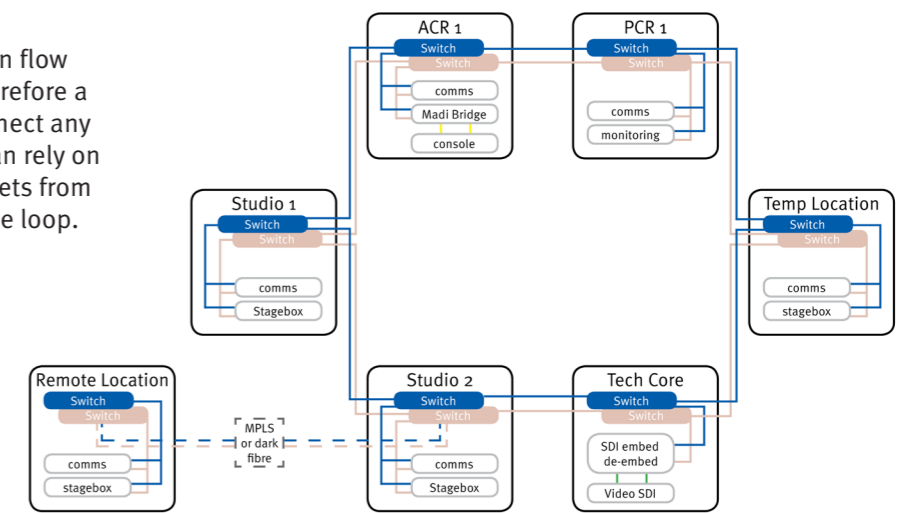

Star configuration
A traditional TDM style star configuration with a central pair of redundant switches distributing to each individual location. Any cable breaks are resolved by switching to the parrallel network.

Ring configuration
In a ring configuration data can flow either way around the ring therefore a broken cable does not disconnect any device. A ring configuration can rely on spanning tree to prevent packets from indefinitely passing around the loop.

Mesh configuration
A mesh configuraton is extremely robust. There are no central switches. All points are connected to all other points. The managed switches are configured so that packets do not flood the network.
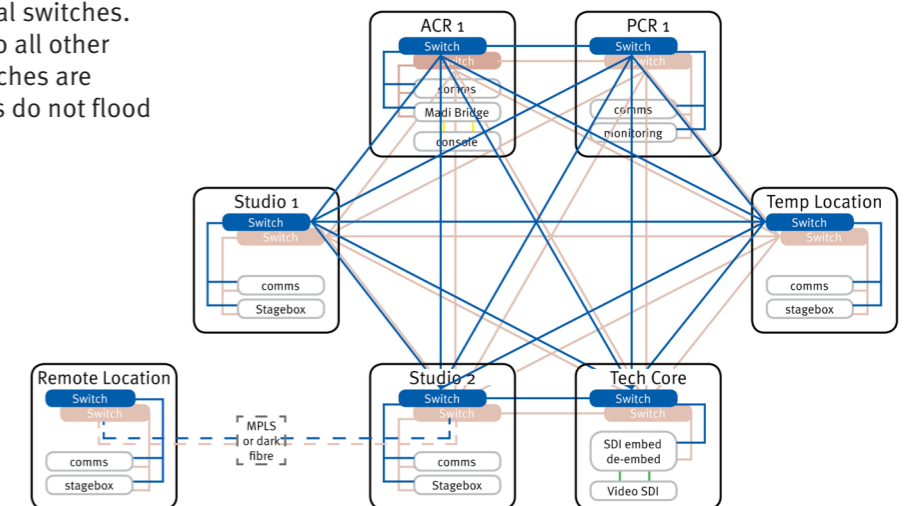
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A History of Broadcast audio Routing

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Complexity and Density
As the handful of signals that needed to be managed for a production gradually became more complex, audio production consoles needed to handle an ever expanding number of sources and outputs and the signal routing requirements increased exponentially. This led to the increasing density of local signal routing patchbays to work in conjunction with the facility routing, expanding the number of wires and the complexity of installation. At the same time, the use of FET electronic switching to replace costly relays increased the ability to route many signals with a single button press. This all increased the complexity and cost of typical broadcast audio installations.

Digital Signals - from Analogue to TDM
With the development of digital protocols and processing in the late '70s/early '80s, audio broadcast signal routing quickly became the realm of Time Division Multiplexed (TDM) digital signals. A single cable could carry more than one channel and the broadcast industry adopted firstly stereo digital audio (AES3) and then later multi-channel digital audio (MADI) to transport signals. With these digital signals came larger and larger capacity digital audio routers. In this (and later!) technological routing developments, audio preceded video in the digital transition. TDM connections are inherently inefficient, interconnections are often carrying no useful information. Many broadcast infrastructures today use large scale TDM routing solutions and both AES and MADI digital audio to interconnect facilities.

ATM - From Cables to Packets
In the late 90's, the first non-synchronous, network broadcast audio transport solutions were explored using ATM (Asynchronous Transfer Mode). ATM is a network technology based on transferring data in cells or packets of a fixed size developed by the telecoms industry as a way to improve on the inefficiencies of TDM. By using small, constant packet sizes ATM equipment could transmit video, audio, and computer data over the same network. This early network transport solution was explored by several companies and broadcasters, including the BBC. Although in many ways suited to media data, the cost and availability of ATM technology was significantly impacted by the increasing use of Internet Protocol (IP). The crucial difference between ATM and IP is that ATM is connection-oriented while IP is connectionless. This means that the establishment of a connection between two endpoints in ATM defines the route all packets (cells) related to that connection must travel. In principle, IP is connectionless, so each IP packet carries a full destination address so there is no concept of a connection at the IP level. Fundamentally, support for ATM packets and connections have largely disappeared from most network switches, while IP has become a de-facto standard.

The Evolution of audio over IP

In the early days of networking, research was driven by the desire to overcome the inherent waste in a Time Division Multiplexed (TDM) connection. By turning signals into packets and sending multiple packets along the same cable, the capacity of that cable quickly rose to thousands of channels. In addition, those data packets could travel with addresses, meaning they could traverse connections without needing the connection to be explicitly established.

The first AoIP solutions used existing IT networks to deliver a limited number of channels through a standard 10 or 100Mbit network infrastructure. Other solutions used standards that were limited to lower OSI layers in order to connect with greater efficiency, but at the price of needing specific network hardware and being unable to be mixed with other Ethernet or IP network traffic. With the use of network technologies to build more 'real time' applications, networking standards evolved to add prioritisation of traffic so that interactive audio applications were possible. Voice over IP (VoIP) requirements for IP based telephone systems introduced many of the AoIP standards that have been adopted and developed.

AES 67
AES 67 is a standard aiming for interoperability of audio transport between multiple existing and future real time AoIP protocols. It is intended to further the possibilities for sending audio between different manufacturers' devices. However AES 67 does not define details of device discovery and configuration, manufacturers will need to agree to define these in the same way to achieve a fully integrated interoperable solution. To interface with AES 67 connections, Dante systems will be able to produce AES 67 streams so that audio can be interchanged with devices that support AES 67.

AVB
Audio Video Bridging (AVB) is a common name for the set of technical standards developed by the IEEE Audio Video Bridging Task Group of the IEEE 802.1 standards committee. This is different to AES 67 in that AVB introduces a change to Ethernet to redefine how a network deals with real time audio traffic. It aims to slightly reduce latency and puts a reservation on audio bandwidth so the network knows when audio connection cannot be made, this potentially makes the network design simpler. AVB is a change to Ethernet and needs new switches for it to work. A limited number of switches exist, none of these are from the market leaders in the manufacture of network switches (Cisco & HP) and at the time of writing, AVB is layer 2 only.

Audinate's Dante
Audinate were one of the first companies to build a high capacity, low latency AoIP solution built on the IEEE and IETF standards. Audinate provide OEM hardware, firmware images and a full software API to allow manufacturers under licence to produce various different products with guaranteed interoperability. They continue to evolve their solution to deliver increasing benefits of capacity and speed, while retaining compatibility with standard networking components and interoperability between components. This reduces issues with installation and increases choice for the installer. A key benefit to the Dante solution is the integration of audio transport, device discovery and device control which is key to the provision of the most interoperable system possible.

Useful Glossary of terms

Network layers and the OSI model

The Open Systems Interconnect model is a standardised model which defines the functions of a communication system by partitioning it into seven abstraction layers. It was published in 1984 in ISO 7498 and also in parallel by the ITU in the X200 standard. If a network device e.g. a switch is "Layer n managed", it can be assumed that it is capable of performing functions that relate to the given OSI layer as shown below. It may also be assumed that if a switch is layer 3 managed, it is also capable of managing layer 2 functionality.

Layer 1 – Physical Layer
• Defines the electrical and physical specifications of the data connection. It defines the relationship between a device and a physical transmission medium (e.g., a copper or fibre optic cable).

Layer 2 – Data Link Layer
• The data link layer provides a reliable link between two directly connected nodes. It detects and possibly corrects errors that may occur in the physical layer. VLANs are layer 2 constructs.

Layer 3 – Network Layer
• The network layer provides the means of transferring datagrams. Datagram delivery at the network layer is not guaranteed to be reliable. A number of layer-management protocols belong to the network layer. These include routing protocols and multicast group management.

Layer 4 – Transport Layer
• The transport layer provides reliable transmission of data packets between nodes (with addresses) located on a network.

Layer 5 - Session Layer
• Establishes, manages and terminates the connections between the local and remote application.

Layer 6 - Presentation Layer
• Transforms data into the form that the application accepts.

Layer 7 – Application Layer
• Interacts directly with software applications running on a host e.g. determining identity and availability.

Classification of traffic

Unicast
Unicast packets are sent from one source to one destination, identified by its IP address. In unicast communications, every copy of a signal – even identical signals – is a point to point connection with its own bandwidth requirement.

Multicast
Multicast packets are sent from one source to many destinations on a network subnet. Multicast communication allows a saving in network bandwidth to be made when one source is being sent to many destinations. Using multicast communication, a sending device requires only one unit of bandwidth per discrete signal sent. E.g. a one-to-many distribution system. Replication and distribution of a multicast signal is performed by network switches or routers. Devices may subscribe or unsubscribe from receiving multicast traffic.

Broadcast
Broadcast packets are automatically forwarded to all devices on the subnet to which they are connected. Broadcast traffic is usually reserved for discovery and DHCP services which, by their nature, must be able to contact every device on a network segment without initially knowing their address.

VLANs
In computer networking, a single network may be partitioned to create multiple distinct domains, which are mutually isolated so that packets can only pass between them via one or more routers. These domains are referred to as Virtual Local Area Networks, or VLANs. VLANs are layer 2 constructs, compared with subnets, which are layer 3. It

is possible to have multiple subnets on one VLAN but not multiple VLANs on one subnet.

Subnets
Subnets divide a large network into n smaller networks for performance and security reasons. subnetting involves the separation of the network and subnet portion of an IP address from the host identifier. Devices on one subnet cannot communicate with devices on a different subnet without passing through a gateway.

MAC address
A Mac address is a globally unique hardware identifier assigned usually by the hardware vendor and stored in ROM. It is notated as a six digit Hex number, e.g. 0a:1b:2c:3e:4e:5f

Network hardware

Ethernet hub:
A multi-port repeater operating on OSI Layer 1. A hub forwards any signal received to all its ports except the originating port. It contains no memory or routing logic but may send jam signals if a collision (more than one device transmitting at once) is detected.

Ethernet switch:
A 'smart' repeater. Buffers, processes and forwards packets only to devices who request or require them at a link speed which has been predefined or negotiated. A Switch may be managed and perform security or traffic management functions.

Gateway:
A gateway is a type of router which allows data to be sent from one subnet to a different subnet. If a device sends data to a device outside its own subnet, it will be automatically routed to the gateway (assuming IP settings on the originating device are set correctly).

Router
A router may also direct traffic between networks, but is responsible for finding the best physical route for traffic to take to reach its destination. Routers prioritise traffic based upon type, network load and policy.

Datagram
A self-contained, independent packet of data containing enough information for it to be routed from the source to the destination device without reliance on earlier interaction between the source and destination device and the transporting network. The source device need not establish a direct connection with the destination device, and can send each data packet through any available route.

Useful Protocols

UDP - (User Datagram Protocol)
UDP is a simple, message-based, connectionless protocol which does not set up a dedicated end-to-end connection. Communication is achieved by transmitting information in one direction from source to destination without verifying the readiness or state of the receiver. The primary benefit of UDP over TCP is the application to audio over IP (AoIP) where latency and jitter are the primary concerns. UDP is usually a fast, efficient and reliable protocol within a wholly wired local area network, with enough network capacity, it needs careful management for bridged networks, and/or wireless networks, hence the need for TCP/IP and its use in wireless/wide area networks.

TCP/IP – (Transmission Control Protocol/Internet Protocol)
Transmission Control Protocol is a connection-oriented protocol, meaning that it requires handshaking to set up end-to-end communications. Only once a connection is established may user data be sent bi-directionally over the connection. TCP is Reliable. It manages message acknowledgment, retransmission and timeout. Multiple attempts to deliver the message are made. If it gets lost along the way, the server will re-request the lost part. In TCP, there's either no missing data, or, in case of multiple timeouts, the connection is dropped. TCP is Ordered – if two messages are sent over a connection in sequence, the first message will reach the receiving application first. When data segments arrive in the wrong order, TCP buffers delay the out-of-order data until all data can be properly re-ordered and delivered to the application.

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DHCP - (Dynamic Host Configuration Protocol)

DHCP is a protocol where a server automatically provides an Internet Protocol (IP) address and other related configuration information such as the subnet mask and default gateway when a host is connected to its network.

NTP - (Network Time Protocol)

Network Time Protocol (NTP) is a networking protocol for clock synchronization between computer systems over packet-switched, variable-latency networks. NTP is intended to synchronize all participating computers to within a few milliseconds of Coordinated Universal Time (UTC).

PTP - (Precision Time Protocol)

Precision Time Protocol (PTP) is a protocol used to synchronize clocks throughout a computer network. On a local area network, it achieves clock accuracy in the sub-microsecond range. In audio Over IP systems, each device has its own highly accurate internal clock, whose drift relative to the master clock in the system is controlled by PTP messages.

NTP vs PTP

Using NTP vs PTP for network/system timing all comes down to the accuracy needed. If the system accuracy needed is measured in microseconds or nanoseconds then PTP (IEEE 1588) is required. If the accuracy needed is only required to milliseconds or seconds, then NTP is sufficiently accurate.

Why is PTP so accurate? Because hardware timestamping is commonly implemented in PTP technology, but not in NTP. Hardware timestamping is allowed in the client and server devices which are running NTP, but not many devices implement this. The largest source of error in network timing is often due to the variations in queuing time in switches and routers. NTP does not have a solution for this, PTP does.

QOS - (Quality Of Service)

QOS is an industry-wide set of standards and mechanisms for ensuring high-quality network performance for critical applications. By using QoS, network administrators can prioritise allocation of existing resources efficiently and ensure the required level of service for defined classes of network traffic at the expense of less time-critical services.

QoS must be enabled to allow Dante to share network infrastructure with other types of data and signals. In many cases QoS in installed network switches may already be enabled, if not the network switches need to have the Basic mode of QoS enabled, checking that the switch is using DSCP (Differentiated Services Code Point). Education and use of common language between broadcast and networking engineers is key in successfully leveraging the advantages in AoIP technology. The network specialist may want to know more about the DSCP labels Dante uses. Audinate publish the DSCP priority values for Dante.

IGMP - (Internet Group Management Protocol)

IGMP is a network protocol which provides a way for a network device to report its multicast group membership to adjacent switches and routers. Multicasting allows one computer on the Internet to send content to multiple other computers that have identified themselves as interested in receiving the originating computer's content. If it is desirable to send the same data to many devices at the same time, a single multicast stream provides significant savings in network bandwidth over using multiple unicast streams all sending the same data.

Flows

Flows are a construct of Dante and can be visualised as a bundle of audio channels streaming across a network. Routing audio in a Dante network automatically creates flows, which carry one or more channels of audio from a transmitting device to one or more receiving devices. Flows may be either unicast or multicast.
Unicast routing creates flows to a single receiving device; a unicast flow typically assigns space for 4 channels of audio. Multicast routing creates flows that can be received by multiple receivers. Multicast flows are assigned IDs, enabling them to be identified in Dante Controller.

More Information
Details of the **SSL Network I/O** range of Dante based audio interfaces is available on the SSL web site <u>here</u>.

The SSL Systems Team bring decades of broadcast systems design to the challenges of leveraging AoIP tehcnology for broadcast. They can offer expert guidance on the design of broadcast infrastructure using AoIP technology.
<u>Click here for help and advice.</u>