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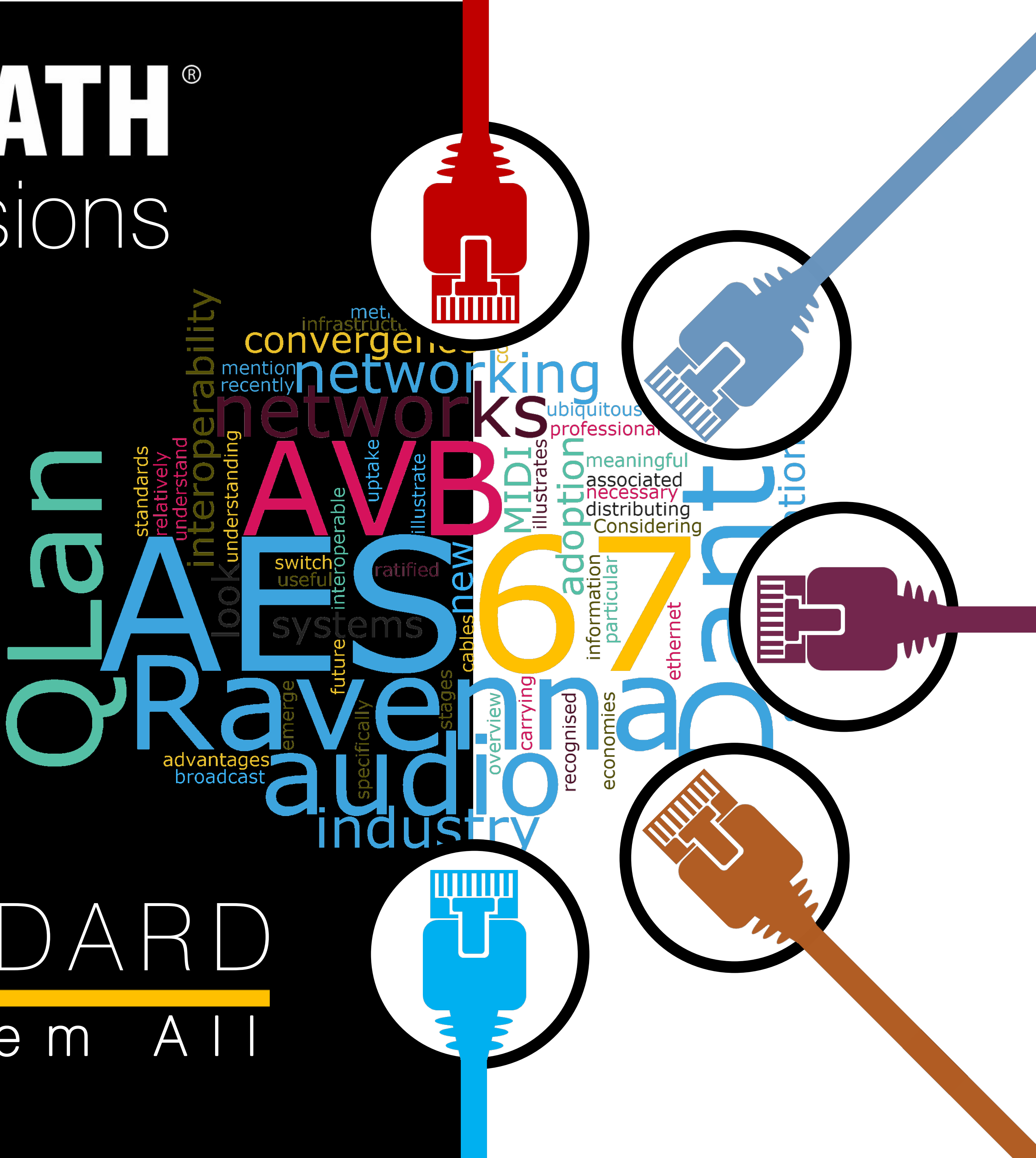


ONE STANDARD

To Unite Them All

The Rise of the Packet

It all started back in 1996 in Boulder, Colorado with the development of Peak Audio's CobraNet protocol. Making use of IEEE802.3 Ethernet Layer 2 networking, CobraNet quickly became the de-facto audio protocol and was the first successful commercial deployment of Audio over Ethernet. However, despite its early dominance CobraNet suffered from certain technical disadvantages such as latency and hardware costs.

Moving forward a few years and we enter a technological 'acceptance' stage where multiple manufacturers start implementing their own propriety protocols for AoE deployment. dSnake and ACE were the protocols that emerged at Allen & Heath, competing with the likes of EtherSound, BLU-Link, REAC and Soundgrid. All these protocols were vying for market share in what can be thought of as analogous to the videotape format war of the 1980's between JVC and Sony.

A further evolution of technology started to emerge in the early part of this century, spurred on by a totally unrelated market. Companies such as Audinate, Axia Audio, QSC and ALC NetworX were all developing their own protocols based and modelled on the already mature standards of the I.T. industry. Collectively these companies brought us the Dante, Livewire, Q-Lan and Ravenna protocols, differentiating their products for use in differing target markets. Unlike their predecessors these protocols do not employ traditional point-to-point connections but instead use standard I.T. infrastructure and equipment to transport audio over Layer 3 of the OSI model via UDP/IP packetized streams. With these new protocols came the emergence of AoIP and the path to true AV/IT convergence.

AES67 - One Standard to Unite Them All

Based on common standards these protocols are fundamentally very similar to each other in regards to transport, session, connection management and synchronisation. Let's have a look at each protocol in a little more detail.

Dante™

Over the past few years Dante has evolved to become one of the most widely adopted audio networking protocols. Dante distributes multiple streams of digital audio plus integrated control data and clock, with sub millisecond latency, sample-accurate playback synchronization, with extreme reliability and high channel count. Audio is encapsulated using UDP/IP and is transmitted to subscribers via unicast flows of four audio channels per flow (unicast is the default method but multicast flows are also supported). Discovery, connection management and synchronisation are handled by ZeroConf, RTSP/SDP and PTP IEEE-1588v1 with DSCP (DiffServ) fixed tag QoS.

Livewire+

Livewire, developed by Axia Audio of The Telos Alliance, was first introduced in 2003 with a primary market of radio broadcasting. The Telos Alliance were a founding member of the Media Networking Alliance who were formed to promote the adoption of AES67. As an early adopter of AoIP, Livewire was built on proprietary standards, as many common standards were yet to be developed. However, Axia were later to develop Livewire+ as a successor to the original protocol, which added AES67 compliance and the common standards required for interoperability. Livewire+ audio transport is via RTP over UDP/IP, and the protocol utilises standards based resources such as SAP, SDP, PTP IEEE-1588 for connection management, discovery and synchronisation for interoperable solutions.

Q-SYS Developed by QSC Audio for use with their Q-Sys platform, the Q-Lan protocol operates over Gigabit or higher switch speeds and can safely use upto 90% of Gigabit link capacity. Transport of audio is via UDP/IP in streams of packets transmitted at a rate of 3000 per second with system latency being deterministic and configurable from 2.5 – 4.5 ms end-to-end. QSC use a proprietary protocol called QDP for network discovery, which uses multicast messages with IGMP to subscribed devices. Synchronisation is again handled by PTP IEEE-1588v1 with DSCP (DiffServ) fixed tag used for QoS.

RAVENNA Ravenna is a protocol based on open-technology standards and developed by ALC NetworX. The primary market focus for Ravenna is broadcast and studio solutions and as such operates with sub millisecond latency. As with the above protocols Ravenna is based on common standards using RTP over UDP/IP for transport, PTP IEEE-1588v2 for synchronisation, RTSP/SDP for connection management and ZeroConf for device discovery. QoS is defined by DSCP (DiffServ) custom tag.

IEEE 802.1 AVB Not strictly an AoIP protocol but rather a collection of standards set out by the IEEE, however a little background is worth noting. AVB or Audio Video Bridging (IEEE 802.1-AVB) is a suite of Enhanced Ethernet network standards which define such elements as queuing & forwarding, stream reservation, synchronisation and discovery. In addition, IEEE 1733 also defines the standards required to synchronise routable RTP payloads, as used by the protocols above, allowing transport and interoperability between all solutions at the layer 3 level.

In terms of adoption, AVB has been slow to gain any professional audio market momentum with some of the reasons for this slow adoption being attributed to relatively low switch compatibility, cost of equipment and the emergence of other companies who were quick to deploy technologies based on already mature standards. However, this could potentially change as I.T. companies such as Cisco support an increasing amount of hardware which is AVB enabled. Additionally, with an increasing amount of audio and video systems being placed on standard data networks, I.T. managers will place an ever growing emphasis on incorporating AVB/TSN for standardisation and network management.

LAYER	MECHANISM	DANTE	RAVENNA	LIVEWIRE	Q-LAN	AES67
5. APPLICATION	DISCOVERY	ZeroConf SAP (AES67)	ZeroConf	PROPRIETARY SAP (AES67)	QDP SAP (AES67)	EXCLUDED
	CONNECTION	RTSP/SDP	RTSP/SDP	RTSP/SDP	RTSP/SDP	SDP Multicast SIP Unicast
	SYNC	IEEE 1588v1	IEEE 1588v2	PROPRIETARY But supports IEEE 1588v2	IEEE 1588v1	IEEE 1588v2
	SESSION	RTP	RTP	RTP	RTP	RTP
4. TRANSPORT		UDP	UDP	UDP	UDP	UDP
3. NETWORK		IP	IP	IP	IP	IP
2. DATA LINK		ETHERNET	ETHERNET	ETHERNET	ETHERNET	ETHERNET
1.PHYSICAL		COPPER/FIBRE	COPPER/FIBRE	COPPER/FIBRE	COPPER/FIBRE	COPPER/FIBRE

Table of solutions and associated protocols

As can be seen the protocols are very similar in their fundamental make-up, so the question had to be asked. What is required to make the protocols interoperable and can there be consensus to draft a standard?

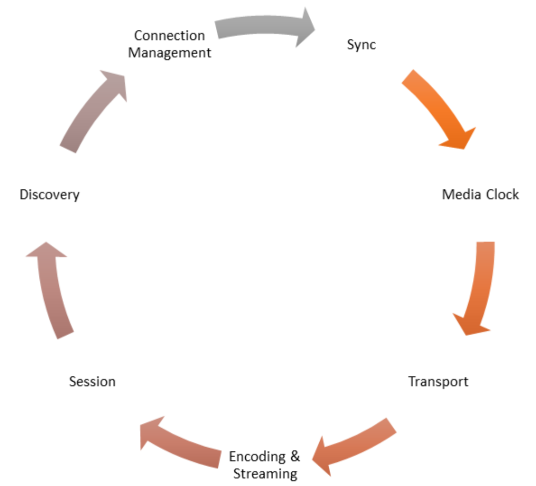
In 2010 the AES initiated their X192 project and assigned it to the SC-02-12 working group on Audio Applications of Networks. The aim of the project was to take already existing protocols and find commonalities between them in order to set out the standards required for interoperability. The task group, consisting of more than 100 members, were to focus their efforts on high-performance audio distribution over IP with minimum quality of 16bit 44.1kHz and latencies less than 10ms. In April 2013 the draft publication was approved by the AES and in September the same year it was officially announced as the ratified standard AES67-2013 High-performance streaming audio-over-IP interoperability.

Once published the ultimate success of AES67 would depend on the extent of details defined within the scope of the standard. Ultimately, AES67 has 'little scope' and there lies the key to its success. Defining a rigid set of complete standards would in essence be a completely new solution, instead the task group focussed solely on the transport of media on a network i.e. getting audio from point A to point B.

This 'limited scope', enables a model which is relatively simple in both its implementation and deployment.

The Seven Elements

So, the scope was set out to cover the following seven items;



1. Synchronization

An accurate network clock is required in order to sample and present media payloads in a synchronized manner. This allows for a fixed latency to be determined between sender and receiver. In order to achieve this AES67 calls for IEEE 1588-2008 Precision Time Protocol (PTP) to distribute time across a network domain. IEEE 1588-2008 uses a hierarchical algorithm to determine the grandmaster clock based on the following criteria;

Priority 1: the user can assign a specific static-designed priority to each clock, pre-emptively defining a priority among them.
Class: each clock is a member of a given class, each class getting its own priority.
Accuracy: precision between clock and UTC, in nanoseconds (ns).
Variance: variability of the clock.
Priority 2: final-defined priority, defining backup order in case the other criteria were not sufficient.
Unique identifier (tie breaker): MAC address-based selection.

1. Synchronization - continued

The assigned or elected grandmaster will then transmit timestamped packets of multicast PTP data over the network for all 'slaves' to synchronise and present streams at precisely the same time. If a grandmaster clock is removed from the network an election will take place to determine the new master clock, using the above criteria. This election process is near instantaneous and presents no audible artefacts. In addition to the above the 2008 revision to IEEE 1588 provides for transparent and boundary clocks. Transparent clocks enable a higher degree of clock accuracy as switches are able to re-calculate and adjust the timestamp due to the time it takes a packet to traverse the switch itself. Boundary clocks are essentially a sub-master clock that slaves from the master with subsequent clocks slaving from the boundary. If enabled higher degrees of accuracy can be achieved but hardware implementations are still very few to date.

2. Media Clocks

Defines the sampling rate of a stream with a fixed relationship to the network clock. The standard sets out that devices with sampling rates of 44.1 & 96kHz should be supported but all devices shall support 48kHz. For example, once AES67 is enabled the M-Dante card can only operate at 48kHz sampling rate.

3. Transport

Describes how media packets are transported over the network. AES67 defines that the transport layer utilises RTP (as defined by RFC 3550 & 3551) over UDP, IPv4 shall be used and it is required to support both unicast and multicast traffic.

When used with Dante, streams are transmitted/received in multicast flows (as are PTP synchronisation messages) and therefore consideration of IGMP configuration should be observed. When selecting or configuring switches IGMPv2 should be enabled and although the standard supports IGMPv3, it should be noted that there can be a prolonged start-up delay when using IGMPv3 devices on a IGMPv2 network. It should also be noted that devices supporting IGMPv2 will operate correctly on a network supporting IGMPv1.

3. Transport—continued

QoS defines classes of traffic by allocating each class with a different DSCP (DiffServ) value. Using QoS we are able to prioritise traffic on the network in order to achieve the required performance. This is a critical component of network design, especially in large networks and/or converged networks spanning a large array of mixed data traffic.

This is great for network administrators but what happens if we have a conflict in DSCP values? Audinate implements the values as shown in figure A within the Dante protocol, whereas AES67 calls for an alternate scheme for tagging traffic, shown in figure B.

PRIORITY	CLASS	DSCP VALUE
HIGH	CLOCK	56(CS7)
MEDIUM	AUDIO	46 (EF)
LOW	CONTROL	8 (CS1)

Figure A. Dante DSCP values

PRIORITY	CLASS	DSCP VALUE
HIGH	CLOCK	46 (EF)
MEDIUM	AUDIO	34 (AF41)
LOW	BEST EFFORT	0 (DF)

Figure B. AES67 DSCP values

In a Dante only network, the best approach would be to use Dante's standard values to manage both AES67 enabled and legacy devices. This ensures that all Dante hardware can co-exist with seamless QoS on the network. However, the main application and reason to implement AES67 is to interoperate with other protocols, which means using mixed DSCP values. In this instance we have a few options at our disposal;

The simplest option is to use a mixed set of values based on the two protocols as shown in figure C. This however has the drawback of setting the AES67 clock value at a lower QoS value than the Dante network.

3. Transport - continued

PRIORITY	CLASS	DANTE DSCP VALUE	CLASS	AES67 DSCP VALUE
HIGH	CLOCK	56(CS7)		
MEDIUM	AUDIO	46 (EF)	CLOCK	46 (EF)
LOW			AUDIO	34 (AF41)
LOWEST	CONTROL	8 (CS1)		

Figure C. Mixed DSCP values

The second option is to use a feature called DSCP remarking. This enables the user to set-up a rule to translate DSCP values between the two sets of network traffic. For instance, when a Dante AES67 stream is routed to say a Ravenna device, the switch can analyse the DSCP values on port ingress and substitute these values according to the set rule, on port egress.

If two discrete systems are used, each with discrete switches, then port egress remarking should be initiated on the switch link ports only.

In general, only implement QoS if absolutely necessary. In the majority of applications, it will not be required!

4. Encoding & Streaming

Defines which audio formats have to be supported by network senders and receivers. As previously mentioned all devices must support 48kHz sampling rates and should support 44.1 and 96kHz. 16 and 24-bit linear encoding was determined to be the required formats, with receivers supporting both formats and senders supporting either or both formats, when used at 48kHz. When used at 96kHz both senders and receivers are required to support L24 encoding and at 44.1kHz both are required to support L16 encoding. A common Packet time of 1ms is also defined by the standard, in order to establish latency requirements.

Higher and lower Packet times are also supported for enhanced performance and for those applications which do not require critical latency settings.

5. Session Description

AES67 uses the Session Description Protocol (SDP) to describe details of a stream for service discovery and connection management. Each SDP file contains information such as media format, channel count, encoding and sample rate, clock information and addressing, in order to establish a meaningful connection between sender and receiver.

6. Discovery

This has been one of the main topics of conversation regarding AES67. The standard does not require any discovery mechanism to be implemented and vendors are free to choose the service they include. Audinate implements the Session Announcement Protocol (SAP) for AES67 enabled devices, as do QSC.

However, there will be instances where you may want to send or receive a stream from devices which do not use SAP. For instance, Ravenna devices use the Bonjour protocol for service discovery and SDP data is distributed via RTSP (although it should be noted that some Ravenna devices have implemented SAP). For those that do not have SAP a piece of software called RAV2SAP is available to manage these connections and translate the SDP information between the two announcement methods.

Additionally, it also provides customisable manual entry of SDP data, as used by Livewire+ and WheatnetIP, for translation to SAP devices.

7. Connection Management

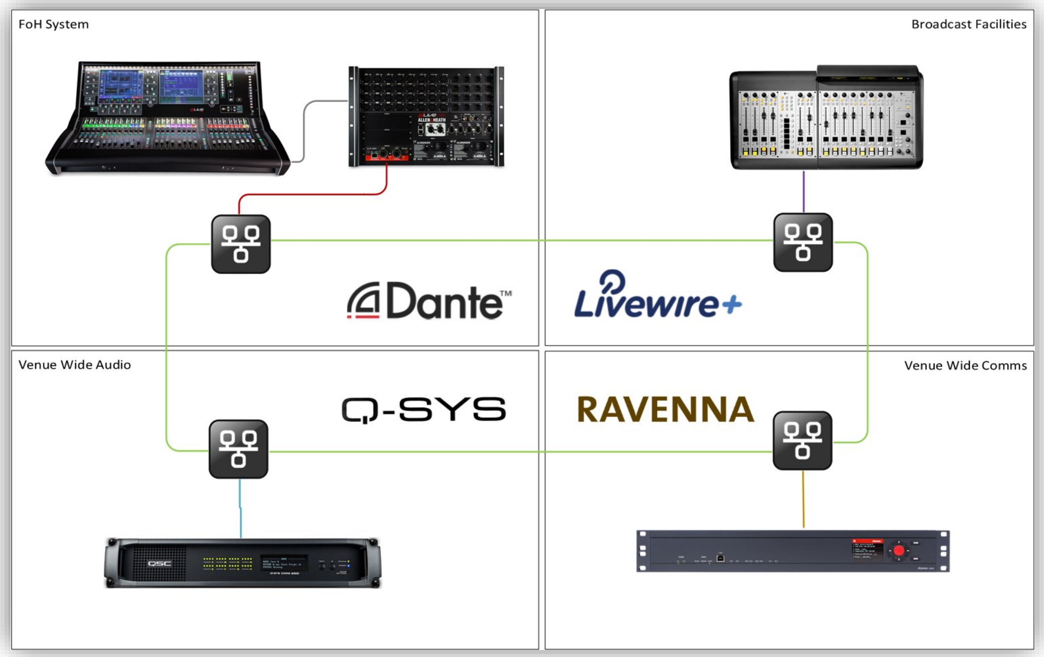
This is used to establish connection of streams between a sender and receiver. As discussed previously, streams can be sent using unicast or multicast transmission. For multicast streams, which Dante uses exclusively for AES67, connection management is achieved using the SDP data within the SAP announcements. For those devices which support unicast streams the SIP protocol is used in a similar manner to traditional VoIP technologies, only with lower latency and higher quality audio.

So what are the benefits of AES67 and where are we likely to use it?

Fundamentally, IP has shaped the way we interact on a technological and social basis. The audio industry has now entered an adoption phase of IP connectivity and will benefit from the already mature standards and protocols that the I.T. industry has developed.

As a user the advantages are self-explanatory. No longer are end users and system designers constrained by keeping system components within a single protocol or having the requirement to breakout to other discrete systems via bridges and/or additional infrastructure i.e. Madi.

In the below diagram we can see multi vendor products connected via off-the-shelf components and simple infrastructure.



Example 1 - Seamless audio transport in multivendor installation

Conversely, system designers can also be comfortable when taking a single protocol approach, safe in the knowledge that external systems will integrate to the 'house' system. AES67 offers a 'best of both' approach and has been designed to be fundamentally scalable and configurable in its deployment.

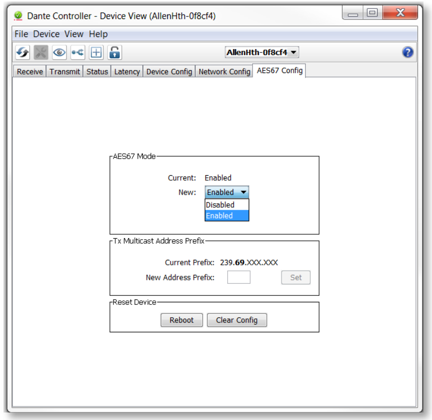
In the next diagram we show how hired or mobile solutions can quickly and cost effectively be deployed within existing systems.

Example 2 - Simple integration and deployment of external discrete systems

That said, currently there are still certain benefits in regards to keeping discrete systems to the same protocol, such as discovery and proprietary control standards.

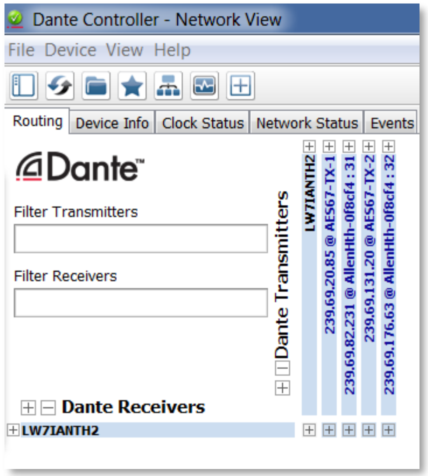
Dante Controller & AES67

Important: M-Dante cards will require a firmware update to enable AES67. Version 2 cards should not be updated to the latest firmware. Please check your card version and the Allen & Heath website for further instructions.



AES67 is disabled by default. To enable open Dante Controller and open the 'Device view'. Go to the 'AES67 config' tab and choose enable from the drop down menu. You will need to reboot to apply the changes.

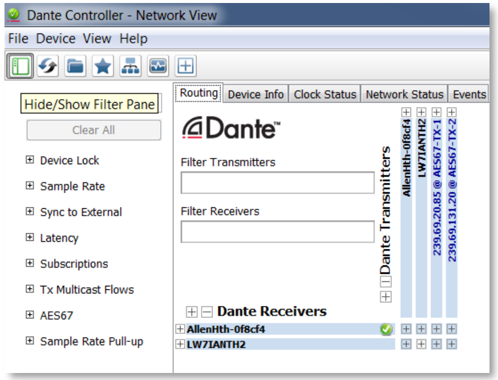
Multicast addressing is automatically assigned to a stream but address prefixes must match external devices. Use the 'New Address prefix' to configure if required.



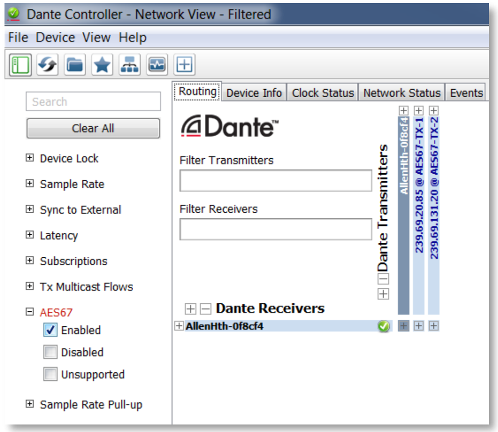
AES67 transmit streams automatically appear and are 'blue' within Dante Controller.

Only streams transmitted to the network are shown and are configured within each manufacturers software.

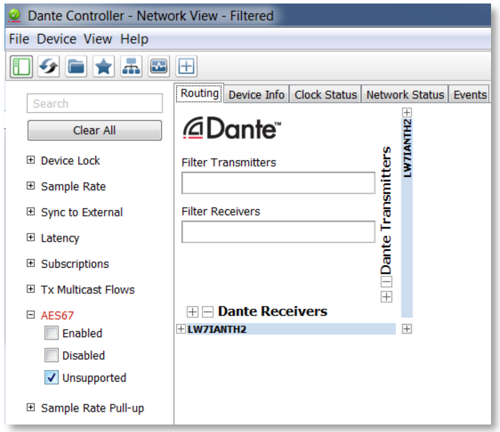
Streams may take up to 30 seconds to appear whilst waiting for an SAP announcement.



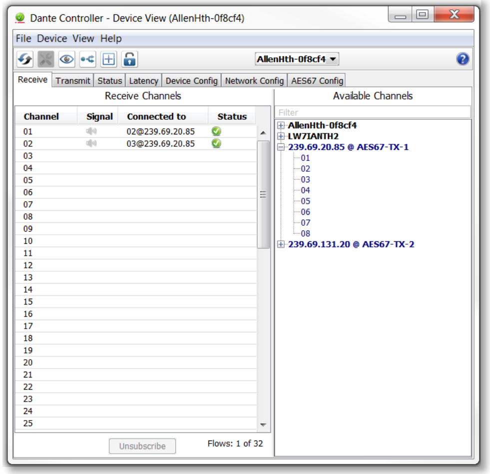
Showing the filter pane enables filtering of current AES67 devices visible to Dante Controller.



Devices can be filtered whether they are AES67 enabled, disabled or unsupported.



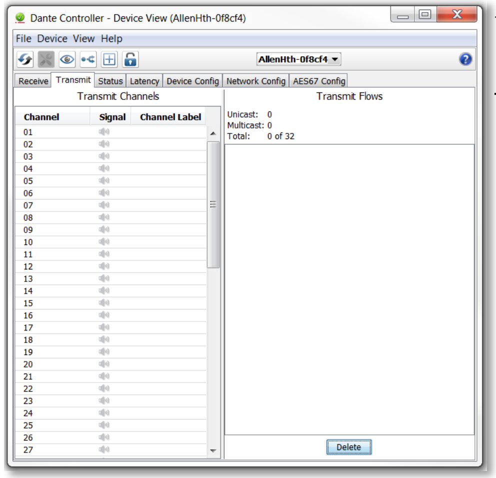
This can be useful when applying complex system assignments and troubleshooting devices.



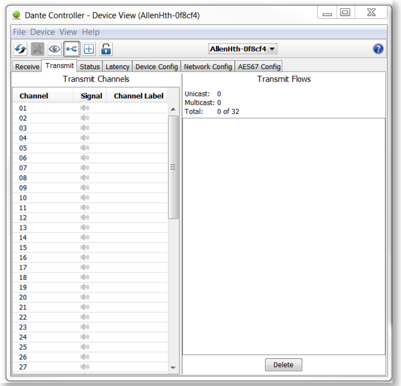
Receiving Dante channels can be found under the Device View > Receive tab.

In this example we have 2 x AES67 streams or flows shown in blue with each stream consisting of 8 channels of audio.

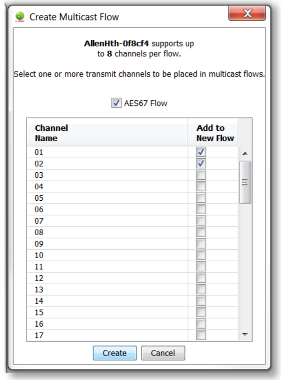
Channels 2&3 of the AES67 stream are routed to channel 1&2 of the Dante receiver.



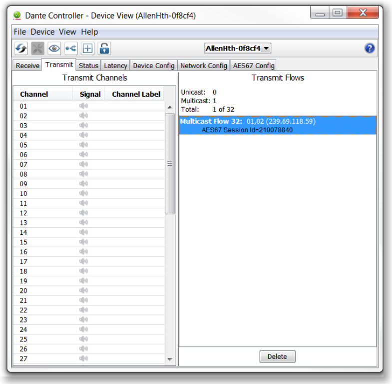
Transmitting Dante channels can be found under the Device View > Transmit tab.



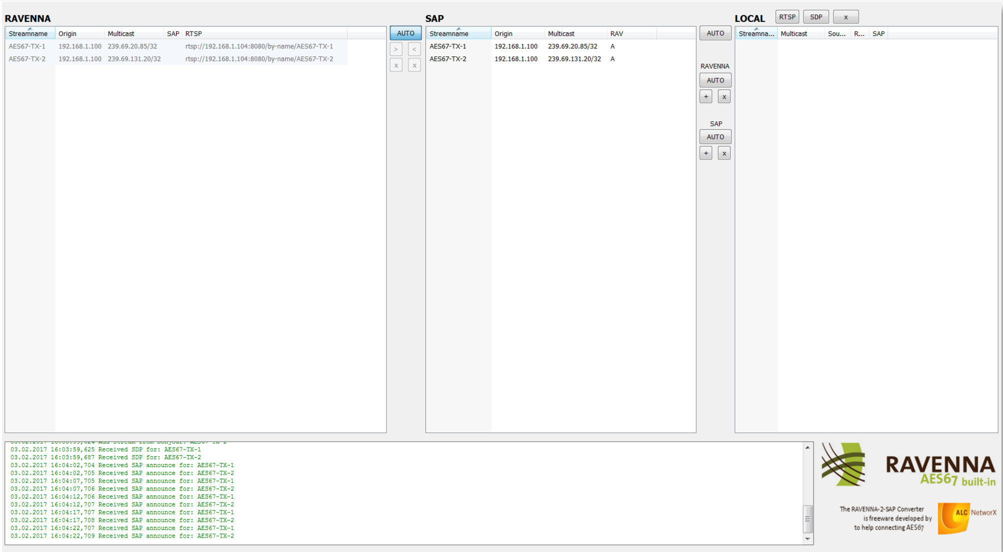
To create a new AES67 Transit device press the 'Create a new multicast flow' icon.



Tick the AES67 box before selecting channels. Then select the channels required for transmitting as an AES67 stream.



The stream will appear in the transmit flow box. A multicast address and stream number will be automatically assigned and is now available to select in the 3rd party receiving device.



As discussed previously, discovery is not part of the AES67 scope. When using equipment with native SAP announcements, such as Dante to Q-LAN the discovery process is straightforward. When connecting devices using differing discovery protocols such as Dante to Ravenna or Livewire, there is an application called RAV2SAP which can be used to translate and transmit discovery messages. The application allows cross conversion of Bonjour to SAP and to also input custom SDP information to discover those devices that do not use either. Additionally, clicking on a stream in either the Ravenna or SAP box will open the SDP information required for configuration of 3rd party devices.

The software can be found at the Ravenna website;

www.ravenna-network.com

....where next?

As with any new standard there is always a certain degree of trepidation in regards as to its success. AES67 appears to have passed this initial phase as we now see multiple manufacturers adopting the standard into their products.

However, there are still obstacles to overcome and momentum is now building for interoperable control standards, which if established would enhance the user experience by setting out standards for connection, control and discovery of networked devices. This could hopefully lead to the panacea of the audio world, where all networked devices are discovered, controlled and routed via a single software application.

Manufacturers are already seeing the demand and the benefits of collaborating to offer third party 'plug-ins' for proprietary applications. The next logical step would be the development of a universal application to control all devices.

As mentioned earlier the OCA Alliance and their AES70 standards are driving this initiative with open standards for cross-vendor connection management and control. However, the scale of adoption of both AES67 & AES70 may well be driven by its wealthier cousin - the broadcast industry. As viewing habits change, the momentum for broadcast to move from SDI to IP based infrastructure has increased rapidly. A roadmap for the broadcast industry has been set-out by the AIMS Alliance to set-out this transition, which moves from interoperable hybrid based solutions based on SMPTE-2022-6 to VSF TR-04 and finally TR-03, both based on SMPTE ST 2110.

TR-03 will use AES67 as its method of audio transport but in terms of device management and discovery the broadcast industry is looking to IS-04, part of the Network Media Open Specifications (NMOS).

It will be interesting to see how the standardisation of control pans out – but that's for another session!