Digitale Audio- und Messsysteme GmbH

Audio over Ethernet:
There are many solutions – but which one is best for you?

Stefan Schmitt, Dr. Jochen Cronemeyer
DSPECIALISTS GmbH, Helmholtzstr. 2-9 L, 10587 Berlin
Stefan.Schmitt@dspecialists.de, Jochen.Cronemeyer@dspecialists.de

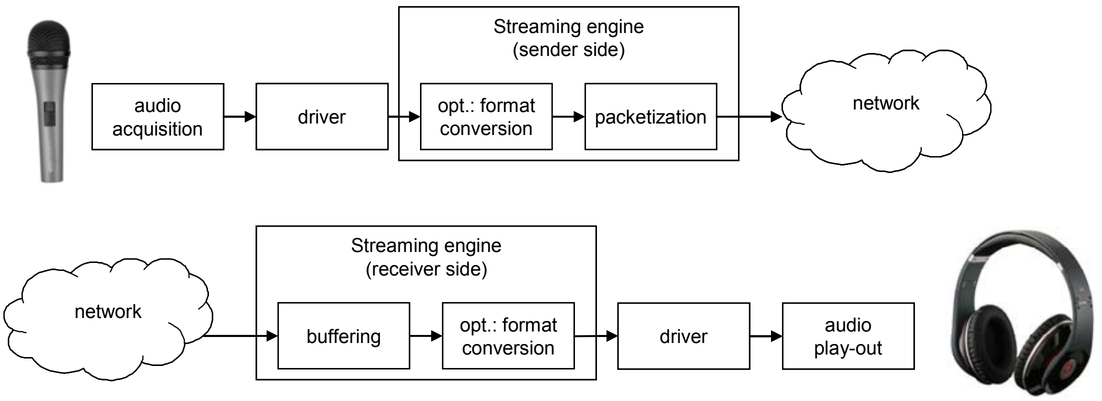
Abstract

The article takes a look at the properties of Ethernet and IP-based networks with respect to the data transmission requirements of audio systems. An insight into synchronicity, latency and other aspects of audio networks is provided.

We introduce the solutions for transmitting audio on Ethernet-based networks already available on the market, like Cobranet, Ethersound, AVB and Dante, and examine these as to their suitability for audio transmission in different application scenarios.

1. Introduction

The professional audio technology[1] of today demands a lossless, latency-minimised (<< 10 ms) multichannel transmission that is not prone to breakdown, and enables a flexible, switchable audio routing which is as free as possible from interference while in operation. At the same time, the costs for materials and installation should be less than for conventional systems.



Transmission system for digital audio

Digital point-to-point transmission methods for audio signals such as AES3, TDIF, ADAT and MADI have the critical disadvantage that they need separate wiring, while an Ethernet-based transmission method can make use of the existing IT infrastructure to transport data in real time.

In addition to the advantages and the potential offered by network-based audio transmission, there are however also critical aspects such as latency, synchronicity and robustness which must be taken into consideration when selecting and integrating network-based transmission methods into new professional audio-technological components. By comparison, these aspects play a subordinate role in the Voice over IP technology.

The audio and media networks we look at here - Cobranet, EtherSound, AVB and Dante - are a selection[2] of the OSI Layer 2 and Layer 3 methods ([1]) available on the market. Contrary to other

[1] Professional Audio (def.): a) Installation (Sound Reinforcement & Public Address, Room Acoustics, Voice Alarm, Paging), b) Recording, c) Broadcast, d) Live Sound.
[2] The choice is exemplarily whereby it does not claim to be exhaustive. An example of a not further considered solution is the network based RAVENNA (ALC Networx), which works on Layer 3. It is similar to Dante but focuses

network-based methods[3], which only exist as specific audio components, they have in common that they are available as chip solutions.

The paper looks at the features of Cobranet, EtherSound, AVB and Dante with the requirements for real time data transfer in audio systems in mind. The four solutions are compared and examined as to their respective suitability for use in different application scenarios. Proprietary solutions, which have been realised by the authors in the past, are also dealt with.

As such, product managers and developers will thus be provided with a decision-making aid when choosing between a proprietary approach or integrating a method available on the market.

2. Outline of the transmission solutions analysed

Cobranet was developed in the 1990s by Peak Audio in the USA and is seen as the first commercially successful implementation of Audio over Ethernet ([1]). The Layer 2 Ethernet-based method is integrated today in many professional audio-technical end products and is mainly used in medium-sized to large installations, for example, convention centres, stadiums, concert halls, etc. In 2001, this technology was bought by Cirrus Logic and, as a result, licences and low-cost ASIC solutions were sold.

Cobranet is one of the methods that uses a standard Ethernet infrastructure acc. to IEEE 802.3 for audio transport, transmitting to Layer 2. The available ASICs[4] only support 100 Mbps Ethernet (e.g. 100BaseT) – the network, by comparison, may also consist of lines with higher data transfer rates (≥ 1 Gbps).

EtherSound, likewise a Layer 2 protocol, is a development by the French company Digigram which first brought the protocol out in 2001 ([19]). 2003 saw the first installations equipped with EtherSound. In 2006, the functionally advanced EtherSound derivates ES-100 and, later, ES-Giga were brought out – both methods are not compatible with one another.

It is true that EtherSound can, in principle, complement a standard Ethernet infrastructure, however, when developing the transmission method the focus was on a simplified wiring system for live sound and/or use while touring compared to Cobranet, and the developers wanted at the same time to considerably minimize the transmission latency. That is why EtherSound uses a daisy chain structure in which the audio data are transmitted from device to device without intelligent, time-consuming routing.

Dante is a commercial, network-based transmission method for audio signals that has been in development by the Australian company Audinate ([9]) since 2006: the patent has been filed for. It transmits the audio data on Layer 3 in UDP packages in a 100 Mbps or 1 Gbps Ethernet. Audinate concentrates on the distribution and licensing of Dante in audio-technical products. This aims to minimise the initial hurdles facing potential integrators by selling standard components in the form of sound and network cards.

The special feature of Dante compared to the other three methods looked at, is that it can be integrated into a conventional router network due to its transport over IP and, what is more, it also uses IP-based zero configuration concepts such as Zeroconf ([21]), meaning that there is no need for an installer or network specialist.

AVB stands for Audio Video Bridging. It is a Layer 2 Ethernet protocol and is the only non-commercial method to be dealt with in this paper. AVB is being standardised by an IEEE 802.1 Task Group ([22]) as IEEE 802.1-AVB. This standardisation is not yet complete. It will be made up of standards IEEE802.1BA (Audio Video Bridging Systems), IEEE 802.1AS (Precision Time Protocol),

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the broadcast market. E.g. for audio distribution between mixing consoles, eletronic musical instruments and monitoring loudspeakers.

[3] Other solutions, which are not available on a chip basis, exemplarily are Rocknet (Media Numerics / Riedel), RAVE (QSC Audio) or LiveWire (Telos).

[4] ASIC = Application Specific Integrated Circuit

IEEE 802.1Qat (Stream Reservation Protocol, adopted in 2010) and IEEE 802.1Qav (Queuing and Forwarding Protocol, adopted in 2010).

AVB focuses on the convergence of network technology and high quality real time multimedia data. The abovementioned protocols will already be integrated into ASICs in the network technology of different semiconductor manufacturers ([8]) before the final adoption of the standards.

3. The features of network-based audio transmission

For different reasons, product managers and developers of new audio components now face the decision of what method for network-based audio transmission is best suited to be integrated into their new product. The following requirements of the method for network-based audio transmission are the starting point for product definition in this process:

General features

- Topology and routing
- No. of channels and channel quality
- Distances
- System convergence

Critical features

- Synchronisation
- Latency
- Robustness and availability

Optional features

- Integrated control protocol tunnels and control interface mirrors
- Real time transport capability for video and other multimedia contents

In the following sections, the features of Cobranet, EtherSound, AVB and Dante named above will be analysed.

3.1 Topology and routing

EtherSound is particularly good for use in live sound or touring. Its daisy chain or ring cable system combines a simple cabling method with minimum hop latency[5].

Typical IT networks generally follow the structured wiring concept for buildings [3], which is hierarchically structured and is also structured across buildings and different floors using routers. Within a unit star and tree systems are used with the help of switches. When this infrastructure is used jointly by real time audio systems, Dante has an advantage as an IP-based method because it can even transmit the audio data via routers.

AVB and Cobranet can integrate themselves into any switched Layer 2 IT infrastructure and allow all topologies that are also allowed in an Ethernet – with ring structures, suitable measures must be undertaken in the network to manage avalanches of packages ([5]). AVB is the only method that requires the implementation of special AVB-compatible network switches.

3.2 Channel quality

A word size with a sample value of 24 bit is state of the art in the professional audio technology of today. Sampling rates of 44.1 kHz, 48 kHz, 96 kHz and even 192 kHz are usual. Audio CDs use a sampling rate of 44.1 kHz. This is sufficient to capture & record audio signals with frequencies of up to 20 kHz. In many other audio-technical devices, the sampling rate is 48 kHz. Sampling rates of 96 kHz and 192 kHz are required if a better immunity against aliasing effects on the audio frequency

[5] Hop latency is the delay time measured from ingress of a network packet at one EtherSound-connector to egress of that packet at the other connector.

range is to be guaranteed or simply if, in a few applications, a higher signal bandwidth is to be processed and transmitted.

All of the audio networks looked at here in more detail can transmit 24 bit audio data sampled at 48 kHz. This data format is taken as the basis for all further specifications in this paper.

While Cobranet and EtherSound only transmit a maximum of 96 kHz data in real time, with AVB and Dante this can even be audio data[6] sampled at up to 192 kHz. In all applications, the maximum transmission capacity of the respective transmission network falls, the higher the sampling rate of the audio data is.

3.3 No. of channels and distance

The number of audio channels to be transmitted simultaneously depends on the respective application:

In order to be able to emit a large number of audio signals in an expansive building to many locations within the complex, a network-based transmission system is needed. If the number of audio channels that it is possible to transmit simultaneously with each transmission method is larger than the number of physical signal sources, then this method is sufficiently dimensioned to have maximum access to the signal sources in the network.

In other applications, you tend to have to watch out that the maximum number of transmittable audio channels in a network cable connection between one audio device and the next device or network switch (link) is high enough.

In a Cobranet link, a maximum of 64 audio channels can be transmitted. Here, when one single Cirrus Logic CS181022 switching circuit is used, a Cobranet end device can communicate and transmit a maximum of 16 audio channels between the integrated I8S interfaces and the network.

A device realisation with 100BaseT transmission and with the connection to 64 audio sources and destinations each is conceivable if four CS181022 are used[7].

While EtherSound and AVB are also capable of transporting 64 audio channels per transmission direction, Dante only allows for the transport of 48 channels in a 100BaseT link. Cobranet, EtherSound and AVB can be assigned to the OSI Layer 2 – an own Ethernet dataframe is defined in each case – Dante transmits the data to Layer 3 IP-based and can therefore even transport the audio data via routers.

3.4 Routing and latency

In contrast to Cobranet, AVB and Dante, which allow an absolutely free logical routing in the network, EtherSound curtails the routing options depending on the wiring, but in favour of very low transmission latency.

EtherSound requires a daisy chain wiring scheme from end device to end device, meaning that the data flows have to be transported via all end devices (one "hop" each per end device), in order to get them from one end to the other. With EtherSound, therefore, the wiring determines the system routing; with Cobranet, AVB and Dante, the individual data packages contain the destination address and thus control their transmission route.

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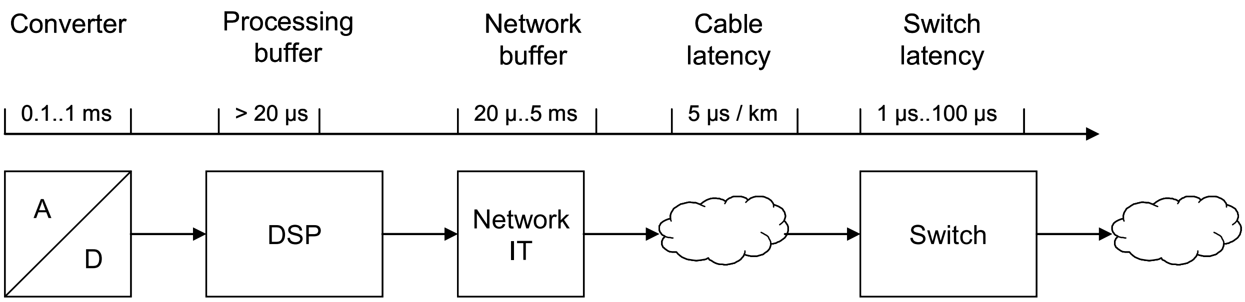
[6] AVB has knowledge of all audio data formats defined by IEC 61883-6.

[7] Cirrus Logic´s CM-2 module is equipped with only one CS181022 ASIC. Thus it is capable to route a maximum of 16 audio channels.



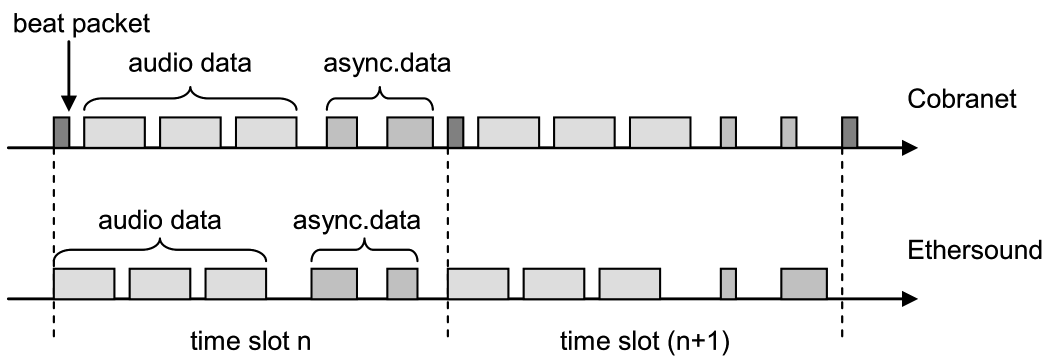
Transmission latency in digital audio

As, however, in this rigid structure no dynamic routing of any kind is done while in operation, EtherSound achieves a very low transmission latency of 0.125 ms as well as 1.4 µs per "hop". At the same time, this daisy chain wiring scheme is very advantageous in practice for live sound applications, because a complex, dendritic network does not have to be set up and disassembled; instead, only one cable needs to be laid from end device to end device[8].

By comparison, the minimum latency in a Cobranet network is 1.3 ms ([12]). For AVB, a minimum of 0.25 ms and for Dante approx. 0.8ms per hop via one switch are specified. With AVB and Dante the latency in each case is conversely proportionate to the rate of transmission, i.e. with 1000BaseT, for example, AVB and Dante transport with a transmission time of only 25 µs and/or 80 µs.

3.5 Synchronising

In Cobranet networks, one single end device (called a "conductor") transmits a so-called beat package 750 times per second (every 1.3 ms). All participants within a beat package interval of this kind transmit up to max. 64 sum channels. The AD/DA converter cycles are synchronised here by recovering an audio cycle from the beat package rate.

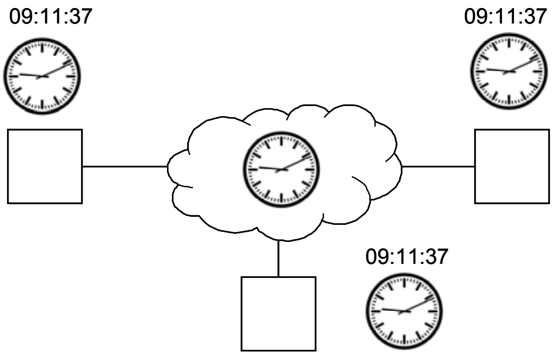


Synchronisation in Cobranet and Ethersound

In EtherSound networks, the synchronisation is solved in a somewhat simpler way, because the so-called "primary master" at the head of the daisy chain activates and controls the entire downlink traffic, so that no special beat package or similar is required.

AVB and Dante, by comparison, know the concept of synchronicity in the entire network through communicating clocks in line with IEEE1588 [14]. This means they are capable of sending, all at the same time, AD-converted audio data simultaneously to all DA converters in the network – the deviance between the output times among one another is specified here for both methods at <1 µs.

[8] Each EtherSound device is equipped with on uplink and one downlink connector.



Synchronuous clocks according to IEEE 1588 in AVB and Dante networks

In comparison to this, with EtherSound, the latency increases with each hop by approx. 1.4 µs and, with Cobranet, the hop latency goes from approx. 1 µs to more than 100 µs respectively depending on the network switch. While not really worth mentioning with EtherSound, in the complex dendritic network in Cobranet, the runtime delays are potentially so serious compared to AVB and Dante, that undesired audible effects may occur in an audio installation. With Cobranet, the planner or the installer must manually compensate for runtime delays of this kind by programming audio processors with adjusted delays in the channels.

3.6 System convergence

Cobranet is capable of transporting host Ethernet data as well as asynchronous control data together with the audio data. With Cobranet, the functionality of transporting asynchronous control data is called “serial bridge”. It consists of a physical interface at the Cobranet chip to connect RS232, RS485, or similar interfaces and the capability of transporting 57.6 kbps of data from a source interface to a destination via the Cobranet network[9]. In the process, the control data are communicated into Ethernet packages cyclically within an isochronous Cobranet cycle. Using the same method, Cobranet allows the transport of host Ethernet data: In the Cobranet end device, the control data and Ethernet data are written from the host processor into the Cobranet chip and from there are communicated as a package to another Cobranet end device. Here, the Cobranet chip has control of the bandwidth at this node and can therefore prioritise the audio data over the control and host Ethernet data, so that the isochronous cycle can not be overrun and, as a result, cannot cause audio buffers to drain.

Nevertheless, audio artefacts do occur in Cobranet networks if external Ethernet data infiltrate the network via the switches and interrupt the timing. It is possible to seal off the Cobranet time-critical audio data from the asynchronous external data by using manageable switches that support a separation of the connected networks with VLANs in line with IEEE 802.1q.

EtherSound also reserves bandwidth for transporting the control data. 768 kbps can be transmitted in a 100 Mbps network together with the audio data from the “primary master” downlink and then uplink again with loop configuration. When external data are smuggled past the EtherSound end devices, as is the case in Cobranet, this also potentially leads to audio artefacts.

AVB, by comparison, requires the use of AVB-compatible switches[10] in the network. This has the distinct advantage that non-AVB-data are automatically prioritised correctly by the network components, meaning that these external data cannot interrupt the audio transport.

The Layer 3 Dante uses QoS features of the network switches, just as they are used for VoIP traffic, and in this way manages to prioritise the Dante data over the external data [10]. In addition, it can run on Layer-2 AVB, which also enables the network to achieve a prioritisation of the audio and control data.

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[9] Thus, data from an RS485 interface of one device for example can be transmitted by Cobranet and output again on an RS232 interface at another device.
[10] Semi-conductor manufacturers already support AVB – see [8].

And so, with AVB and Dante, the transport of asynchronous control data and time-critical audio data merge far better in comparison to Cobranet and EtherSound.

3.7 Robustness and availability

It is absolutely necessary in professional audio technology to achieve high availability of the audio channels and the highest possible degree of robustness against channel interference.

Here, we can differentiate between two principle causes of interference

- Method-based causes
- Component and/or cable breakdown

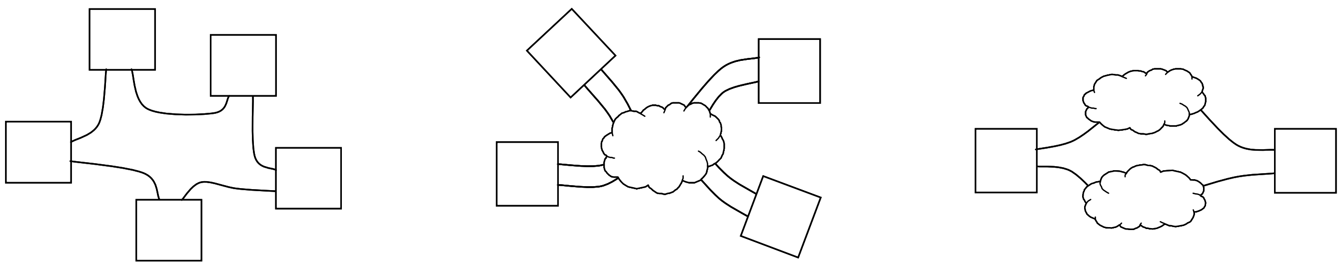
As already explained in the above section 3.6, Cobranet and EtherSound are sensitive to external Ethernet data, as they cannot control these. AVB and Dante do not have this problem if the network requirements (e.g. QoS) are fulfilled.

However, the methods we have looked at undertake different precautionary measures to deal with breakdowns in the infrastructure:

Every Cobranet device is equipped with two network connections. During normal operations, only one of the two connections is active. The other connection automatically assumes operations as soon as the first connection detects a faulty connection. If both interfaces are connected to a network switch, this approach first of all only secures the wiring from the Cobranet device to the network, but not the network itself ([16]).

What is more, it can occur that the so-called "conductor", that is the Cobranet clock device in the network, breaks down. In a case like this, the audio transport in the entire network is interrupted for a few milliseconds until another end device has taken on the job of "conductor" ([11] [12]).

EtherSound devices always have two network connections (uplink, downlink), which are wired up to other devices as a daisy chain. The EtherSound ES-100 ([13]) Standard allows all EtherSound end devices to be wired in a ring wiring scheme[11] without using dedicated network switches. Here, one end device is assigned the role of the "preferred primary master" (PPM). This separates the rings during normal operations so that a packet flood cannot occur. When an error occurs, for example, a cable break or an end device breaks down, the PPM closes the ring automatically meaning that all devices and components other than the device with the breakdown or the faulty wire are integrated again into the network. This function is similar to the spanning tree protocol (IEEE 802.1D). When a primary master breaks down, an EtherSound-System might need up to three seconds until audio is synchronised onto another master – in the meantime the EtherSound end devices work in "emergency clock" mode[12], which corresponds to the feed-in from a local master in each case. A divergence of the clock rates during this period can lead to buffering problems in individual cases and thus lead to audio artefacts.



Ethersound: Redundancy by duplex loop   Cobranet: Redundant link   Dante: Redundant network

While, until now, AVB has not known any kind of own redundancy concept, Dante allows for a complete network redundancy. For this purpose, Audinate sells the so-called "Dante Core Module

[11] The free uplink connection of the primary master is connected with the free downlink connection of the last EtherSound device in the daisy chain.
[12] The emergency clock mode must be activated by the user in the EtherSound device.

DCM”, which is equipped with two network connectors and which, when breakdowns occur, should make it possible for a network to transfer without any glitches to the other network.

3.8 Real time transport of video and other multimedia contents

Cobranet and EtherSound have the limitation that they can only transport digital audio signals.

By comparison, AVB knows all media types for transmitting compressed audio and video data that are specified in the IEC 61883 [18].

As Dante can use an AVB network, in principle all of these media types are also accessible to Dante.

4. Proprietary approaches

In the past, DSPECIALISTS has integrated proprietary methods for real time transmission of audio data in several products. This also includes a multiroom audio system, which distributes a total of 20 audio channels (mono scenario) in a Layer 2 Ethernet with an analog-to-analog latency of approx. 10 ms. In this system, the audio data are transmitted from the respective A/D converter source cyclically. In the respective destinations, a sampling rate conversion of the audio data takes place in software on the output cycle. As such, in this case, a complex hardware cycle recovery and control per PLL was not necessary. This proprietary approach does not require any special ASICs for audio transmission. It is based on conventional 100BaseT components and tailormade software.

Another example is AUBION X.8 - a multichannel audio measuring device from DSPECIALISTS with an integrated 1000BaseT dual port switch that transmits the audio data to Layer 3 per UDP packet in real time between the measuring device and PC. In this application, the latency is not critical, meaning that packet jitter caused by capacity fluctuations on the part of the PC can be compensated for (30ms to 300ms point-to-point latency). The method is optimised for an absolutely lossless audio transmission in an environment with extremely severe time fluctuations in the packet rates that are generated by the PC. These are observed in particular in inefficient computers. In this similarly proprietary approach, the audio measuring device and/or its A/D-D/A converter cycle is the primary master and the PC the slave. The measuring device transmits packets with a specific packet rate. ([24]) This rate controls the buffer switches in the ASIO driver on the PC side.

5. Which one fits best?

What method is the best one for transmission in audio applications? It has been shown that no single method is the absolute best one. All methods, also the briefly introduced proprietary ones, fulfil real time requirements and are to some extent specialised for implementation in certain professional audio-technical areas.

The proprietary methods outlined are tailor-made and can do without special hardware. They have been designed as closed systems that are Ethernet-compatible, but are unable to communicate with any other audio transmission systems on the market. By comparison, the four Ethernet-based methods dealt with are open and compatible with a number of professional audio products.

In comparison to Cobranet, EtherSound has a very low network latency of 125 µs between two participants. That is why it is widely used in products for live sound applications and in the area of touring. Cobranet, on the other hand, can mainly be found in products for audio installations and public address systems, because it makes conventionally structured Ethernet systems available for real time audio transport. Cobranet’s area of application is limited to uses for which a buffer latency of at least 1.3 ms as well as additional route-dependent, different network latencies in the range of several 10 µs per hop are not seen as exclusion criteria. Both methods have been widely used and field-tested in their areas of application. As such, an integration of both methods is relatively risk-free from a technical point of view.

AVB and Dante first appeared in 2006, that is, four years after EtherSound and Cobranet. Both methods synchronise the audio sources and destinations by clock and/or cycle synchronisation on the basis of the IEEE1588 standard. While AVB even requires the use of AVB-compatible network switches for this, there are no requirements in this sense from Dante. AVB will presumably be completely IEEE-standardised by the end of 2011 – the audio-related organisation “AVnu” is taking care of the interoperability of AVB devices here.

In comparison to Dante, AVB has the advantage that the standard is absolutely open. Technically, Dante provides a plug & play concept, making installation easier. At the same time, Dante does not require any special AVB components, however only in cases where the network infrastructure can cope with QoS features. On the other hand, AVB can offer somewhat better timing features.

Whether the chance of integrating Dante and AVB simultaneously provides a real advantage must be decided individually on the basis of whether you want to offer the benefits of both methods together in the new audio product in question: a) Dante, if an existing non-AVB-compatible network has to be used and b) AVB in addition, of the network infrastructure still does not have any AVB switches and the additional costs for this can be included in the budget, so that the optimum timing features can be achieved.

Applications that AVB is striving for a range from the live sound area to installation technology to broadcast applications. In addition, “AVnu” is also trying to position AVB in the automotive and consumer areas. Dante’s special feature lies in the transport of audio data on the basis of UDP/IP instead of Ethernet MAC packets, meaning that a routing on Layer 3 is possible.

For all of the network-based audio transmission methods introduced here, chip and/or module solutions are available. In order to be compatible with a number of audio components available on the market on the one hand and, on the other hand, so that we are able to support future standards, DSPECIALISTS has decided to develop a module concept for its own product – the 16 x 16 audio matrix HARVEY mx.16 – which will first of all be compatible with other Cobranet components at the end of 2011 and, following that, will be compatible with AVB, Dante, EtherSound and MADI per software update and module exchange.

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